DATA COMMUNICATIONS

Contents

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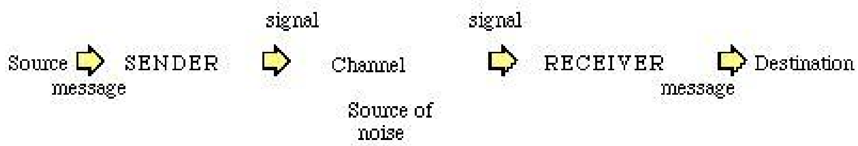
Part 1: Introduction to Communications

Overview of Communications

The need to communicate is part of mans inherent being. Since the beginning of time the human race has communicated using different techniques and methods. Circumstances and available technology have dictated the method and means of communications. Many early forms of communication were writing, depicted on cave walls. Then communication advanced by the development of language and the use of symbols. Papyrus and paper were used to record communication for later use. Smoke signals of the early american Indians; the drums of African tribes; and the towers of the chinese wall are indications of the desire to communicate beyond the immediate physical boundaries of space. Story tellers around the camp-fire are a good example of communication, using animation, gestures and sound to communicate their message to other members of the tribe.

In 1948, a model of communication was proposed by Claude Shannon. Shannon worked for the Bell Telephone Company in America, and was concerned with the transmission of speech across a telephone line. Warren Weaver, in association with Shannon, wrote a preface to this model and it was published as a book in 1949. Weaver saw the applicability of Shannon's model of communication to a much wider sphere than just telephony, and it has served as a basis for explaining communication since that time.



Shannons Model of Communication

In terms of oral communication between two people, the model is applied as follows.

Message	The idea, thought
Source	The brain
Sender	The transmitting device, the mouth
Channel	The medium the message travels over, air
Receiver	The receiving device, the ear
Destination	The brain

In any communication there is **noise**, which affects the message as it travels across the channel from the sender to the receiver. Shannon proposed building in**redundancy**, which was added to the transmitted message in order for it to be reliably detected at the receiver. Let us apply this model to a telephone conversation between two people. The person that initiates the call by lifting the telephone handset and dialing a number is the SOURCE, their telephone the SENDER, whilst the person who answers the ringing telephone is the DESTINATION and their telephone the RECEIVER. The CHANNEL is the Public Telephone Switched Network(PTSN), and the MESSAGE is the topic of conversation [speech] that was the reason for the call being made.

Exercise
Apply Shannon's model of communication to the following pictures, by identifying the source, sender, channel, receiver, destination and message elements.

Problems of Shannon's Model
However, Shannon's model has a number of problems as a model for explaining communication.

- it is one way (from source to destination)
- there is no feedback between the sender and receiver, it is non-interactive
- it does not translate appropriate to groups with many interactions
- it does not explain the process of how the message is generated by the source, or interpreted by the destination

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Modes of Communication
There are a number of commonly accepted divisions or layers of communication. The following table lists the more common layers that form the basis of modern courses in communications.

Communication Type	Explanation
Intra Personal	Communication one has with oneself, thoughts, daydreaming
Inter Personal	Communication one has with another person
Group	Communication one has with a group of people (group discussion, party)
Organizational	Communication within or between organizations (newsletters, memos)
National	Communication within or between nations (trade, war)
Global	Communication on a global scale that affects all people on the planet (greenhouse effect)

Look at the following images. Try to categorize each picture into the appropriate layer of communication outlined in the previous table.

   

 **Something to think about**

Spend a few moments thinking about some of the methods of communication used today, such as radio, television, news-papers, fax, phone and Internet. Are they one-way, two-way, multiple or interactive? How can user participation be encouraged (classic case is talk back radio, and music requests)? What reasons can you think of for encouraging user participation in traditionally one way methods of communication?

In addition, a message is sometimes delivered in different formats, for example, sound (as in clicks and whistles between dolphins), as printed text in newspapers, as sound waves in speech, sometimes as electromagnetic radio waves or light. What criteria do you think must exist if the sender and receiver are to inter-communicate effectively?

And another thought... under what occassions might you want the message to be secure so that other parties who might be listening in would not know the content of the message?

Summary

Communication takes place all around us. Mankind has communicated in various ways since the beginning of their existence, by means of drawings, writings, signals, symbols and language.

Shannon's model of communication serves as a basis for explaining communication, but it has limitations, an example is that it does not explain the meaning of the message.

There are many levels of communication, from inter, intra, group, organizational, national to global. We can also speak of animal communication and computer mediated communication, where the computer mediates the message. In the information age, much of the communication we receive each day is mediated in some way by technology.

Further Internet based references

Assyrian lion hunt frieze at the British Musuem. http://www.british-museum.ac.uk/highligh.htm#Dying
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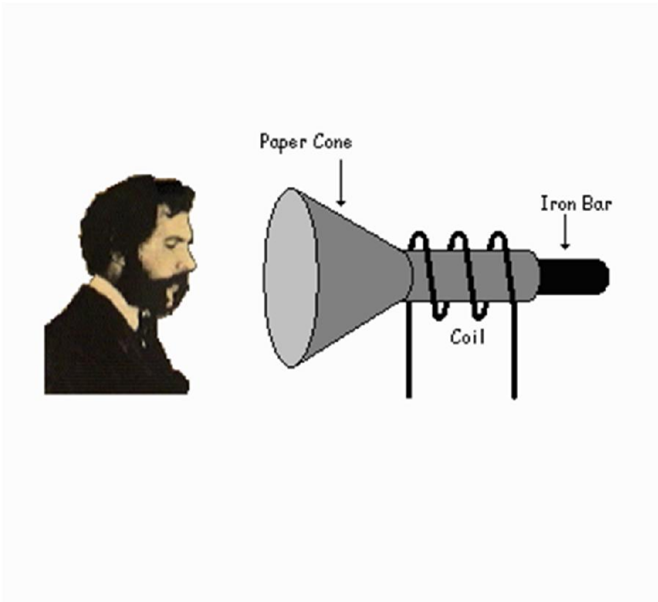
DATA COMMUNICATIONS

Part 2: The evolution of modern tele-communication networks

Evolution of Modern Communications Networks

Data communications concerns itself with the transmission (sending and receiving) of information between two locations. In the information age, this means sending information between machines that are connected together by physical wires or radio links.

The history of modern electronic communications begun with Alexander Graham Bells telephone experiments, where speech was able to be converted into electrical energy, transmitted along physical wires and reconstructed at the receiver. Speech, which is actually vibration of the air, vibrated a paper cone to which a small coil was attached. This induced an electrical signal into the coil, which was proportional to the vibration of the paper cone.



As you can see in the animation on the left, the sound waves caused by speech make the paper cone move. The paper cones movement is directly proportional to the strength of the air waves caused by speech.

As the cone moves, the coil of wire also moves, being attached to the cone. Inside is an iron bar which helps to increase the effect.

A corresponding electrical signal is created in the coil, which can then be sent along a pair of wires to a receiving device that would convert the electrical signal back into sound waves (vibration of the air).

Alexander Graham Bell, born in Edinburgh, Scotland, 1847, emigrated to the United States, settling in Boston. Bell, interested in the education of deaf people, invented the microphone, and later, in 1876, an *electrical speech machine*, called the telephone.

By 1878, Bell set up the first telephone exchange in New Haven, Connecticut, and in 1884, long distance connections were made between Boston, Massachusetts and New York

City.

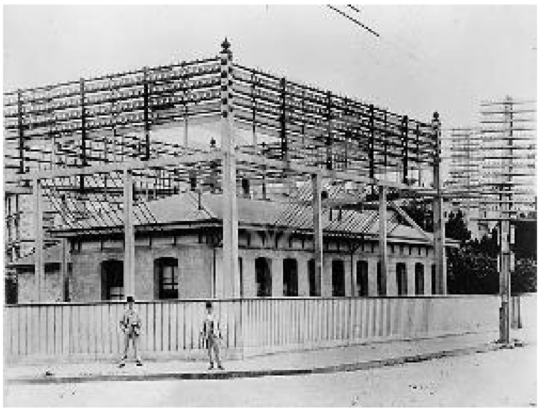
In these early experiments at transmitting speech. copper wire was used as the connection media over which the signals traveled. This is due to copper wire being a very good conductor, which lets electrical signals flow down the wire easily.

As the telephone became popular, more and more people wanted to communicate with each other, so a **switching center** (telephone exchange) evolved. Each customer was connected to the telephone exchange via a pair of wires, which carried the signal from their telephone.

As the need to inter-connect telephone customers grew, they were connected via overhead wires to a central switching center, where the physical wires from each customer was connected to the physical wires of another customer via a manual operator.

As more and more customers were connected, the need for more and more operators to connect calls were required. This quickly became unworkable, so development began on automating the connection process between customers, hence, automatic telephone switching exchanges became a reality and replaced local operators, who were still used to connect toll (long distance) calls.

The customers wires from their home to the telephone switching center were **overhead**, installed in the same way as a lot of existing electrical (power) wires are today, on top of long poles inserted into the ground, about 15 or so feet high. This quickly became very cumbersome and difficult to manage. Transmission quality was affected by the weather, as the rain created leakage paths for the signals to go to ground instead of along the wire.

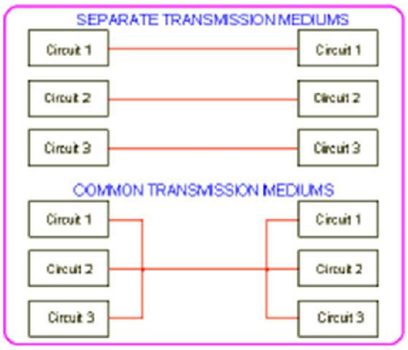


File Print: *Wellington. Telephone Exchange. 1894*
Creator: Photographer Unknown
Description: The manual telephone exchange at the north end of Stout Street, Wellington.
Date: 9 January 1894
This photograph must be acknowledged to the
J M Lander Collection
Reference Number: F-115589-1/2-
Collection Reference No: PAColl-0715
Alexander Turnbull Library, National Library of New Zealand, Te Puna Matauranga o Aotearoa.

At the same time, demand grew to connect customers who lived further and further away, in rural areas. Electrical signals can only travel so far, and to provide service to rural customers meant using better cable that allowed the signals to travel longer distances. Unfortunately, this was costly, so something had to be done to provide them with service. In addition, customers also wanted to be able to talk to other people in different cities, so there developed a need to interconnect telephone exchanges together.

Rural customers began sharing cables, so that one cable supported a number of customers, either one at a time (which was known as *party-line*), or at the same time using radio signals to separate each conversation. To interconnect telephone exchanges in distant cities together, work began on using different cable and the use of radio signals over cable such as *coax.* For rural customers, who lived longer distances from the central switching center, it was too expensive to have a single cable for each customer. The need arose for a single cable to carry the conversations of more than one customer, in this way, customers could share a single cable and thus save the telephone company money. So, within a short time, by using radio signals, each open wire cable could carry up to 12 separate speech conversations, with an optional telegraph (telex) signal. Distance then became a limiting factor, so work began on increasing the distance over which the signals could be transmitted and received. One method was to increase the diameter of the cable (which also makes the cable heavier).

The cost of cable was very high, and before long special equipment was used to carry more than one speech conversation on one pair of wires. This was done by a technique known as **multiplexing** (specifically Frequency Division Multiplexing, which separates each conversion by frequency).



This bought down the cost of providing speech circuits (one circuit=one speech conversation) to customers. Whereas previously one cable equated to one speech circuit, now a cable could be equated to hundreds of speech circuits. This allowed an increase in revenue to be collected per cable by the telephone companies.

For a given cable size and type, the speech signal is affected by loss. In other words, as the signal travels down a cable, part of it is absorbed by the cable and thus the signal arrives at the end of the cable with less strength than when it started (this is called **attenuation**). It thus follows that if the cable is too long, no signal will arrive at the end, or the signal will be so small that we cannot hear it.

A thicker cable affects the signal less than a thinner cable, so you can go greater distances with a thicker cable before the signal becomes too weak. However, thicker cables cost more and are heavier. Open wire systems were originally made from copper, but the cost became very expensive (and difficult due to the first world war where materials were in sort supply).

Attenuation is a measure of how much loss a signal experiences when it travels down a communications medium. Part of the signal is dispersed as heat, or absorbed by the communications medium. Not all of the transmitted signal arrives at the receiver. Attenuation is measured in decibels (dB). A loss of 3db means half the signal is lost by the time it reaches the receiver. The longer the distance that the signal travels, the greater will be the attenuation that the signal suffers.

Open wire copper systems gave way to open wire systems using aluminum. These cables had to be a great deal thicker because aluminum is not as good as copper (it affects the signal more), but it was lighter and cheaper. The other problem is that aluminum suffers from oxidization when exposed to the air, and this also affects the signal as it travels along the cable. Aluminum cables began to be used during the second world war, when copper became scarce.

Open wire systems, being exposed to the weather, often failed during wet weather and storms. They were often struck by lightning and falling trees. It wasn't long before open wire systems serving rural customers were replaced by other forms of communication systems.

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Part 3a: Cables and Network Communication Systems

Open Wire | Twisted pair | Coax | Microwave
satellite | Fiber Optic | Cellular Radio | Satellite Cellular Radio
Pagers | Summary | References

Transmission Media
Transmission media refers to the many types of cables and other mediums that carry the signal from the sender to the receiver. Copper based cable is the most common form of medium, and is used for virtually all links except long distance. A cable medium often introduces unwanted changes to the signal, which limits the speed (how fast we can send information) and frequency range of the signals that can be transmitted.

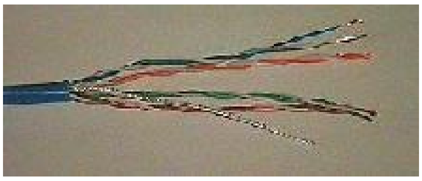
Open Wire
Seldom used nowadays, but was used extensively in rural areas to provide 12 speech channels over a single wire atop telephone poles. Open wire cable has been replaced by micro-wave systems or newer technologies. Being exposed to the weather, dust particles and salt spray were deposited on the insulators that secured the cable to the telegraph pole. When storms and wet weather arrived, the dust turned to mud, and provided a path of least resistance for the signal, thus the signal was diverted down the pole to ground and significantly reduced in strength. Lightning strikes were also a hazard and special ceramic arrestors were used to protect equipment attached to open wire lines.

Twisted Pair Cable
Before long, the open overhead wires servicing users from the telephone centers were replaced by multi-strand cable , which was buried underground and protected by many sheaths of polyurethane plastic. This prevented water seeping into the cable and affecting the signal. It also helped overcome some of the limitations that open wire systems had. The major problem now is people digging them up! These multi-strand cables used a pair of wires for each user, and there were about 500 pairs or more per cable. Each wire was twisted around each other wire in order to try and reduce unwanted noise (hence the term *twisted pair*).

Cable was laid by the telephone company along each street, from the telephone center, to the customers house. Thus each customer had a physical set of wires which ran from their telephone set all the way into the telephone exchange. The use of multi-strand underground cable, still in widespread use today, made the delivery of low-cost telephone services to the general public.

Twisted pair cable is the most common form of cable today, used to connect telephone subscribers to exchanges (switching centers) and wire buildings. Two insulated wires are twisted around each other, and combined with others into a cable. In general, each twisted pair supports a single voice channel. Twisted pair is also used to interconnect PC's on a Local Area Network (LAN).

Twisted pair used in Local Area Networks has several ratings. Category 3 has a speed rating of 10 million bits per second (the speed of Ethernet), whereas category 5 has a speed rating of 100 million bits per second. CIT is cabled using Category 5 UTP cable. UTP (unshielded twisted pair) is cable which has no ground shield. Cables are often provided with a ground shield which helps to reduce signal interference from external sources, thus making the signal travelling down the cable less prone to alteration. Twisted pair cable is provided in two forms, UTP and STP (shielded twisted pair).



The picture on the left is UTP Category 5 cable, suitable for transmission speeds up to 100 million bits per second.

The picture on the right is STP Category 3 cable, suitable for transmission speeds up to 10 million bits per second.

Unshielded twisted pair cable is the predominant cable used today. Two conductors are coated with a plastic sheath then twisted around each other. These pairs are then twisted around other pairs to make a multi-pair cable. The twisting of the wires around each other helps to reduce unwanted signals being induced into the wires. It is used for telephone wiring inside buildings, as telephone cables which link customer houses and buildings to telephone switching exchanges, and for implementing local area networks.

UTP has the advantages of

- a high installed base
- cheap to install
- easy to terminate

Its disadvantages are,

- very noisy
- limited in distance
- suffers from interference

Coaxial Cable Systems

There became a need to inter-connect telephone centers in different towns together. The use of open wire or multi-strand cable for this was unsuitable, as the capacity of each cable was too low (number of simultaneous speech conversations per cable) in order to make it economically viable. Hence the introduction of coaxial cable. Till recently, coaxial cable was extensively used to support toll traffic and long distance links. Today, it is being replaced by micro-wave, satellite orfiber-optic links. Coaxial cable is a two wire conductor with a larger bandwidth than twisted pair cable. It is used in television, radio, and Ethernet LANs. In voice communication systems, each coaxial cable supports about 60 speech channels.

It has a single core, with an outer conductor which acts as a shield. The signal is transmitted on the inner core. The inner core and the outer shield are separated by an insulator, either plastic or mica. The cable is enclosed in polyurethane to protect it and give it some strength.

It is important not to bend the cable too tightly, as this damages the insulator which separates the inner core from the shield. The transmission qualities of coax are affected by the properties of the type of insulator used. Coaxial cable could carry up to 1200 speech circuits per cable. Coaxial cable operates better than open wire, as it is buried in the ground and not subject to the elements. The systems used to implement multi-speech channels on coaxial cable are called **broad-band** systems.

When used in Local Area Networks to interconnect computers, the most popular form is RG-58AU cable, commonly called *thin Ethernet*. The coax cable connects to each PC using a special T connector, and up to a maximum of 30 connections can be made in tandem, from PC to PC.

Thin Ethernet is cheap to install and is rated at 10 million bits per second. Each end of the cable is terminated using a 50ohm terminator. Failure to terminate each end of the cable, or a break in the cable, causes the network to fail.

Coaxial cable is used extensively in networking and data communications. A center conductor is separated from an outer conductor by an insulator medium. The cable cannot be crushed or bent sharply, as this damages the insulation between the conductors and thus alters the electrical characteristics of the cable. When used for local area networking, it links PC's together. The networking protocol commonly used with coaxial cable is ETHERNET, which describes how data is formatted and transmitted along a shared cable system.

As open wire gave way to better systems, so coaxial cable has given way to others. The problem with coaxial cable is the number of speech channels available per cable. With the high demands for data communications between computers, increased telephone circuits between cities and countries, not to mention television channels, other mediums have taken over from coaxial cable. Coaxial cable became popular in the 1980's as a method of interconnecting computers, specifically Local Area Networks (LAN's), because it was cheap and easy to install. In addition, some cable TV systems use coaxial cable to supply programming content to subscribers. In New Zealand during 1995, Kapiti Coast Television introduced a cable TV subscription service on the Kapiti Coast which used coaxial cable to supply TV programming content to paying customers.

Coaxial cable has the advantages of

- cheap to install
- conforms to standards
- widely used
- greater capacity than UTP to carry more conversations

Its disadvantages are,

- limited in distance
- limited in number of connections
- terminations and connectors must be done properly

Micro-wave systems

As the demand for more and more speech circuits grew by customers wanting to make long distance calls, the telephone companies had to expand the capacity to meet this demand. One such system which was used was Microwave, which does not use cable as a transmission medium, rather it uses the air. Using very highfrequency signals, microwave support thousands of telephone channels and several television channels on the one circuit. Microwave is a radio system which uses very high frequencies to send and receive data. Because of the high frequencies involved, stations are located about 30 kilometers apart and in line of sight (visible to each other). Microwave systems have sufficient bandwidth capacity to support a large number of voice channels and one or two TV channels.

Transmitters and receivers must be located within sight of each other, about 30kilometers apart. Microwave does not bend round corners or jump over hills.

Dishes and towers were expensive to construct and with the distance limitations, meant it was expensive to go very long distances.

Nowadays, many companies use microwave systems to interconnect buildings at high speed digital links of 2 million bits per second or greater. Sometimes, this is a cheaper solution than linking buildings using fiber optic cable, specially in inner cities where cabling is a problem, or across rivers etc where terrain prohibits the use of existing physical cabling methods. This allows companies to link their networks in different buildings together into one common network, allowing the sharing and accessing of information. Today, microwave systems are used in a number of areas, such as linking local area networks together between campus buildings, sending radio signals from a radio station to its transmitter site, and the sending of video or audio signals from an outside sports event back to a TV broadcasting studio.

Microwave systems have the advantage of

- medium capacity
- medium cost
- can go long distances

Its disadvantages are,

- noise interference
- geographical problems due to line of sight requirements
- becoming outdated

Satellite systems

Ground stations with large dishes communicate with a communications satellite in geo-stationary orbit around the earth. Each channel is managed by a transponder, which can support thousands of speech channels and about 4 TV channels simultaneously. The cost of satellite links is still very expensive (about $4M per transponder). It is primarily used for intercontinental links.

Satellite systems are comprised of ground based transmitter and receiver dishes, with an orbital satellite circuit (called a transponder). Signals are transmitted to the orbiting satellite, which relays it back to another ground station. The footprint coverage of a single satellite system is very large, covering thousands of square kilometers (enough for entire countries such as New Zealand). For example, in New Zealand in 1997, the SKY television network announced plans and introduced a pay per view satellite service for New Zealander subscribers. Satellite is used to carry television channels and telephone conversions between countries.

These use an orbiting satellite to communicate with ground stations. They support tens of thousands of speech channels and tens of television channels.

The cost is very high per circuit, and signals are received using a special dish. Satellite TV is an example of such a system.

Satellite systems have the advantage of

- low cost per user (for PAY TV)
- high capacity
- very large coverage

Its disadvantages are,

- high install cost in launching a satellite
- receive dishes and decoders required
- delays involved in the reception of the signal

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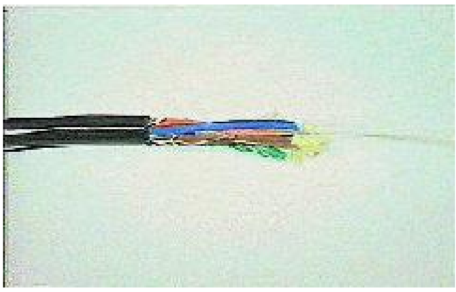
Part 3b: Cables and Network Communication Systems

Open Wire | Twisted pair | Coax | Microwave
satellite | Fiber Optic | Cellular Telephone | Satellite Cellular Telephone
Pagers | Summary | References

Fiber Optic Cable

This is cable made from fine strands of silica (glass), coated with a plastic sheath. The signals are converted to light pulses using a laser. Each fiber optic strand can support thousands of speech channels and multiple TV channels simultaneously. It is used mainly for long haul links and intercontinental links.



A strand of silica glass fiber (thinner than a human hair), is coated with a reflective surface. When light (provided by a laser) is shown into the strand, it travels along the fiber strand (the reflective layer prevents it from escaping).

Fiber optic cable is used for long haul telecommunications links, high speed data communications links for computers, and information services to homes (e.g., PAY TV).

Fiber optic cable has the advantages of

- high capacity
- immune to interference
- can go long distances

Its disadvantages are,

- costly
- difficult to join

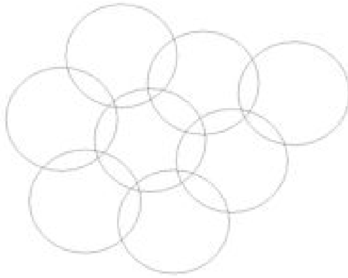
Cellular Telephone

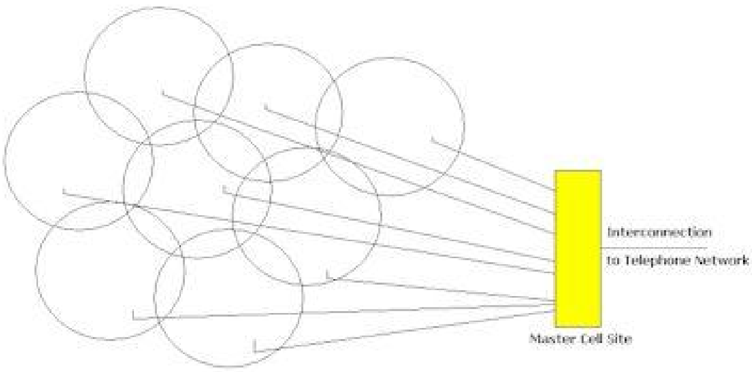
Also known as mobile phones, cellular telephones are a recent technology designed for mobile users who need to make telephone calls from various locations. The telephone is portable and carried with the user. A cellular phone uses radio frequencies to talk to a nearby cell site (a site that handles cellular calls).

The cellular phone regularly communicates with the nearest cell site to inform the network that it is connected.

A **Cell Site** is a circular geographical area that handles cellular phones within its defined physical space. Larger coverage is obtained by overlapping cell sites to form a cellular network.

As a user moves location from one cell site to another, the call is transferred to the nearest cell site responsible for that physical area.





Each cell site is linked back to a master site that provides an interconnection to the regular telephone network. Calls handled by each cell site are relayed back to the master cell site which then relays it to the telephone network.

Frequencies can be reused by other cell sites making for efficient sharing of facilities. Many calls can be handled by one frequency (especially where digital phones are used)

Cellular telephone suits large geographical areas, including remote sites. It is cheaper to install that traditional copper cable, and is making large inroads into countries with high dense populations that cannot upgrade their existing infrastructures (such as pacific/asia).

Satellite Cellular Telephone

This is cellular phones using low earth orbiting (LEO) satellites. There are currently two systems in place, the Iridium and Teledesic satellite networks. The advantage of a satellite cellular network over a land based one is much wider geographical coverage, especially in mountainous terrain and at sea. One of the problems of the land based cellular coverage is the amount of cell sites required, and their accurate positioning to avoid blind spots where calls cannot be made.

The Teledesic Network
The Teledesic network is another constellation of low earth orbiting satellites, 288 (plus spares) in total. Its major sponsor is Motorola. Its purpose is to provide high speed data access to services such as the Internet, video conferencing and high quality voice and data connections.


© www.teledesic.com

The Teledesic network will support millions of simultaneous users, with speeds up to 64Mbps on the downlink path and up to 2Mbps on the uplink path.

Pagers
Pagers are small hand held devices that allow one way communication between two parties. Paging was first developed by Charles Neergard in 1949. Neergard working as a radio engineer, and was annoyed with the way in which doctors in the hospital he was at were continually called over the loudspeaker system. His proposed paging system allowed a much quieter method of informing people that a message was waiting for them.

A ground based radio transmitter sends out a constant stream of messages on a particular frequency. Pagers, which are essentially radio receivers, monitor this stream of messages sent by the transmitter. Each pager has a built-in address code (sometimes called a cap code). When an incoming message is detected that has the pagers built-in address associated with it, the pager decodes and displays the message.

There are 4 main types of pagers

Cap Code	These pagers (which were the first on the market) beep when the built-in address is received by the pager.
Tone voice	These pagers were developed in the 1970's and allow the sender to record and send a short voice message.
Digital display	Introduced in the early 1980's, these are the most widely used pagers. A call back number is entered

	by the sender, which then appears on the pagers display unit.
Alphanumeric	These were introduced in the late 1980's and allow a text message to be displayed on the pagers display unit. This often has the advantage of not requiring the receiver to dial back the sender.

Summary

Technology has happened at a fast pace. Since modern communications has begun, early forms of media quickly become outdated and are replaced with newer more modern methods.

For example, in New Zealand, we have seen the Public Telephone Switched network evolve from the early use of coaxial cable to interconnect main centers, then its replacement with the use of microwave stations, and now, today, the heavy use of fiber optic cable as a preferred means of interconnecting main centers together.

Each media is suited to different purposes and each has their place. In the early days of telephones, the telephone companies wired each persons home with a twisted pair cable. What limitations do you think such a connection now poses? In addition, as people become more and more involved in the technological revolution, what additional services might they want to access from their home (one such example is the Internet) and how do you think the existing telephone connection limits the provision of those additional services?

Make a list of some additional services you think people might be interested in and the benefits of having access to such services from their home might bring.

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http://www.teledesic.com/ The Teledesic satellite network

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Part 4: The Public Telephone Switched Network

Public Switched Network | ISDN | Nominal Voice Channel | Summary

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The Public Telephone Switched Network (PTSN)

The Public Telephone Switched Network (PTSN) refers to the publicly available dial-up telephone network (offered in New Zealand by Telecom New Zealand). It is an interconnection of switching centers and connections to subscribers, that offers voice dial-up between any two subscribers connected to the PTSN. In addition, overseas connections to other countries is possible.

As telephone communications evolved, switching centers were used to manually connect users together. A pair of overhead wires was run from the switching center to each subscriber, and the wires were terminated on a switch-board. A manual operator was used to interconnect subscribers. With more and more subscribers, the number of operators required became a problem. In addition, the sheer amount of cabling and interconnection became unmanageable. The development of automatic exchanges (using Electro-mechanical devices) simplified the interconnection of subscribers by placing it under the control of the subscriber (provide a telephone with a numbered dial).

The problem of subscriber wires was overcome by the use of multi-strand underground telephone cable (500/1000 pair). Each switching center (called a CENTRAL EXCHANGE or METROPOLITAN EXCHANGE) was interconnected to the next, and located geographically center of the existing population to minimize the amount of cable required to reach all subscribers.

As data communications evolved, the only connection mechanisms available was via **DIAL-UP lines** provided over the switched telephone network. These dial-up lines were suitable for voice use, but severely affected digital data. They were prone to noise from the Electro-mechanical switches used to interconnect subscribers, and they placed limits on the speed of which data could be transmitted.

Special circuits that are suitable for data are available. These circuits are an END-TO-END connection, and no switching is involved (a direct hard-wired connection that bypasses the switching mechanism in the exchange). These are called **LEASED LINES**, and are suitable for data communications at a number of different speeds (customers pay a monthly rental based on the speed of the line they lease). A typical use of leased lines is to provide a permanent connection to the Internet.

The need for dial-up data connections has necessitated other services being made available. In addition, the computerization of switching centers has reduced the interference and noise associated with voice channels, and high grade (low error rate), high speed (9600bps or higher) communication is now possible over most dial-up voice circuits.

The Integrated Services Digital Network (ISDN)

The Integrated Services Digital Network (ISDN) is a new integrated network design that allows a network to support voice, data, video and other data over twisted pair cable. It is fully digital, and requires voice users to convert speech into digital signals before connection. Rather than a company having three separate lines to handle three machines like phone, fax and computer, it can now have a single line and connect all three simultaneously. The single line is a digital line that connects into the telephone companies digital network. ISDN is not the same as the PTSN. The PTSN is an analgue network designed for speech, whereas ISDN is a digital network designed for transferring digital data.

The basic level of services available for ISDN (the single line) are,

```
      2B + D
where B = a 64Kbps data channel
      D = a 16Kbps secondary channel used for signally or low speed data
use.

Higher capacity circuits are also offered (called H channels).
```

The advantages of ISDN are,

- reduced cost
- reduced error rate
- access to different services over a single link
- standard network interface

Typical uses of ISDN are,

- desktop conferencing, where two users use the same application though separated geographically
- video conferencing
- high quality audio connections for sports events (radio)
- temporary Internet connections for retrieving and sending email

The ISDN connection is digital end-to-end. Calls are setup to a destination on demand, just as in the PSTN. Once the connection is made, the user has access to the full digital capacity of the link. Users are often charged on combination of time and the amount of data transferred.

The Nominal Voice Channel

The voice channel supplied by telecommunication companies for speech circuits meets internationally defined standards. These define such characteristics as,

- bandwidth (300Hz to 3.4KHz)
- attenuation (signal loss)
- phase shift

Voice circuits are suited to conveying speech. When used to carry digital data, they limit the speed at which signals can be transmitted, and also introduce errors as the voice circuits generally suffer from noise. Each speech circuit provided by the telecommunication companies is measured against the nominal speech channel standard. This guarantees it suitable for voice transmission only.

Summary

A dial up line is a non-permanent switched connection. It is suited to telephone conversations (analogue speech). In data communications, it is often used by users to connect to the Internet using a modem.

A leased line is a permanent non-switched connection. It is suited to higher speed communication and attracts a monthly fee. However, users have exclusive use of the leased line and can send as much data over it as they want. It is used to establish a permanent connection between two sites, such as connecting a company to the Internet 24 hours a day.

ISDN provides much higher speeds than dial-up, less errors in transmission of data, but currently has slower speeds than that available for leased line. It is a digital temporary dial up service, and the basic ISDN connection is two lots of 64Kbps and one 16Kbps. If a company is using a connection more than 5 hours per day, a leased line will be cheaper, as ISDN charges include call connection charges and time charges.

DATA COMMUNICATIONS

Part 5: The Internet

What is it? | History | Services | Providers
Intranets | Thinking Exercise | Summary | References

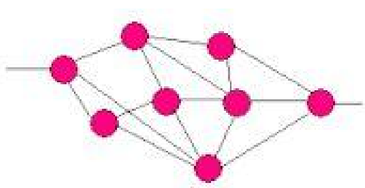
What is the Internet?

The Internet is a global connection of computers. These computers are connected via a huge network of telecommunications links. The Internet allows you to access to a whole resource of data and information stored at different sites (called **hosts** or **servers**) and locations all around the world. The communication links that interconnect each host computer use a common method of transmission, known as TCP/IP, which stands for Transmission Control Protocol/Internet Protocol.

Each computer connected to the Internet (by the way, it is always spelt with a capital I) can act as a host. A host computer provides information for other people to access and retrieve.

History of the Internet

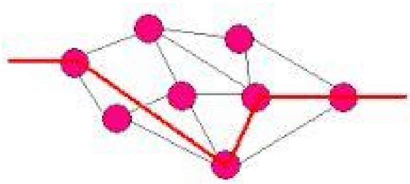
The Internet had its origins in the cold war between Russia and America during the 1960's. Concerned about the survivability of its communications in the event of a nuclear strike, the US air force needed to ensure that it could still communicate with its forces.

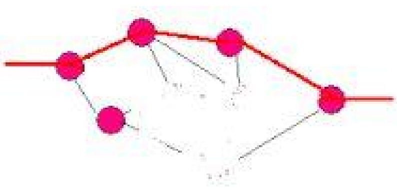


The RAND corporation proposed a system with no centralized authority, as any centralized system would be a target of any possible attack.

The proposed system, developed by Paul Baran, suggested a decentralized system that would still operate even if parts of it were destroyed.

All interconnections in the network could send and receive messages, forwarding them onto other interconnection points (called **nodes**) until the message reached its destination.





Information would be sent in little packets, each packet would be self contained and have its own address information. Packets would travel from node node, each node deciding how to send the packet to the next available node. Even if some nodes were destroyed, the message could still be sent by an alternative route.

In this way, the network would withstand a nuclear strike. After implementing the network, it was known as ARPANET and used by the US military and US universities. Gradually, as more and more connections were made, it has evolved to the Internet.

Services available on the Internet

The Internet provides for a wide range of services. Some of these are listed in the table below.

Service	Description of Service
EMAIL	Electronic mail. Permits the sending and receiving of messages to other users connected to the Internet.
FTP	File Transfer Protocol. A means of sending and receiving files from one computer to another.
GOPHER	An early form of representing information as graphical icons or symbols, that could be displayed in a window and then downloaded. It has been replaced by the WWW
USENET	A number of discussion groups that allow users to post questions and replies, sorted by topic. Also known as news.
WWW	World Wide Web. Accessed using a web browser such as Netscape Navigator or Internet Explorer, a means of locating and displaying information located on the Internet.

Internet Service Providers

Internet Service Providers (ISP's) are companies which provide others with access to the Internet. This can be via dial-up connection using a modem, or using an

ISDN or permanent high speed connection. Various charging levels may exist, but a popular method for home users is flat rate (per month unlimited time and data amount).

Each user can access the Internet through connection on an existing network or via a modem from a remote site such as a private residence. The data and information that can be accessed on the Internet comes in numerous different formats and there a wide range of applications that interpret the information for the user.

Intranets

An Intranet is an internal TCP/IP network which is not visible outside the company or department where it resides, essentially being used as a repository of local documents. It uses the same technology as web servers that the Internet uses. Companies today deal with masses of information. This information is in many forms, typically, to name a few

- sales reports
- product information
- training manuals
- company procedures and practices
- price lists
- press releases
- newsletters

Companies can no longer justify printing this information and keeping it up to date. Using an Intranet solves this problem, by providing a single document which can more readily be kept up to date. By storing this data on an Intranet, users get access to the information and can search this information faster and with greater ease than before.

Something to think about

Think of some other applications that could be used on a company Intranet. What value might a company have in establishing an Intranet? What problems do you think a company Intranet might suffer from? Consider issues of user support, hardware and software requirements, updating of information, and security of the information on the Intranet.

Summary

Traditionally the Internet was purely a text based global pool of information and access was either limited or required a certain specialised knowledge. The development of the Internet today has ensured that information now comes in other formats such as graphical, audio and animated images, and the interface for such information is now a lot more dynamic and user friendly.

Commercialization of the Internet is evidenced by the increasing presence of large multinational corporations. The Internet continues to grow each year, with more and more people becoming connected and more and more services becoming available.

Further Internet based references

History of the Internet and WWW. http://members.magnet.at/dmayr/history.htm

DATA COMMUNICATIONS

Part 6: Analogue and Digital Signals

Analogue | Digital | Common Terms | Speech Circuits
Problems in sending data over telephone circuits
Summary | Test 2

Introduction
In this section, we look at the nature of signals such as voice and data. In data communications, understanding how signals are constructed will help us understand the problems involved in sending them from one place to another.

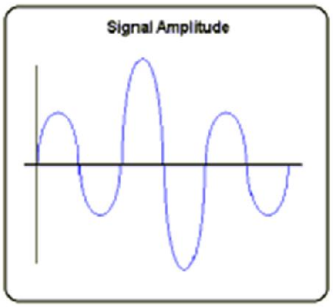
Analogue Signals
The public dial-up service supports analogue signals. Analogue signals are what we encounter every day of our life. Speech is an analogue signal, and varies in amplitude (volume), frequency (pitch), and phase.

The three main characteristics of analogue signals are,

- **Amplitude**
  This is the strength of the signal. It can be expressed a number of different ways (as volts, decibels). The higher the amplitude, the stronger (louder) the signal. The **decibel** (named in honor of Alexander Graham Bell) is a popular measure of signal strength.

Sound level	Type of Sound
40db	normal speech
90db	lawn mowers
110db	shotgun blast
120db	jet engine taking off
120db+	rock concerts

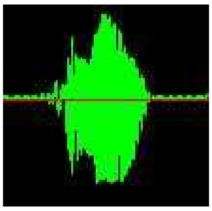
- It has been discovered that exposure to sounds greater than 90db for a period exceeding 15 minutes causes permanent damage to hearing. Our ability to hear high notes is affected. As young babies, we have the ability to hear quite high frequencies. This ability reduces as the aging process occurs. It can also be affected by too much noise over sustained periods. **Ringing** in the ears after being exposed to loud noise is an indication that hearing loss may be occurring.



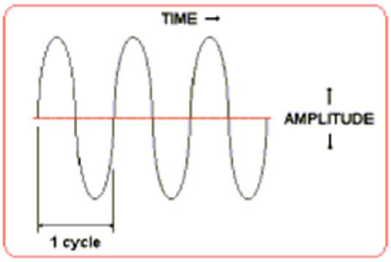
This diagram shows a single signal of various amplitudes. The base line indicates a steady state, in this example, the signal amplitude rises both above and below the steady state.

The measurement of the two extremes is called the **peak to peak** measurement.

This diagram illustrates a speech signal, in this instance, the word "Hello". Speech is a very complex signal, and contains many thousands of different combinations of signals all mixed together. Note that it looks much more complicated than the single signal shown above.



- **Frequency**
  This is the rate of change the signal undergoes every second, expressed in Hertz (Hz), or cycles per second. A 30Hz signal changes thirty times a second. In speech, we also refer to it as the number of vibrations per second. As we speak, the air is forced out of our mouths, being vibrated by our voice box. Men, on average, tend to vibrate the air at a lower rate than women, thus tend to have deeper voices.



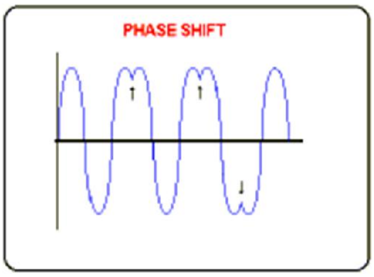
A cycle is one complete movement of the wave, from its original start position and back to the same point again.

The number of cycles (or waves) within a one second time interval is called cycles-per-second, or **Hertz.**

An example is sitting on the beach, counting the waves as they come in on the shore. If we counted the number of waves that crashes on the beach over a minute period, then divided that number by 60 (because there are 60 seconds in one minute), we would have the number of waves per second... (i.e. frequency!).

Humans can hear from reasonably low frequency tones (about 100Hz) all the way up to about 12KHz. As we get older, our ability to hear the higher notes is lessened, due to the aging process making the bones in the ear harder and less able to vibrate. In addition, human speech contains a great deal of redundant information, and can be compacted within the range of 300Hz to 3400Hz whilst still retaining approximately 80% of its content.

- **Phase**
  This is the rate at which the signal changes its relationship to time, expressed as degrees. One complete cycle of a wave begins at a certain point, and continues till the same point is reached again. Phase shift occurs when the cycle does not complete, and a new cycle begins before the previous one has fully completed.



The human ear is insensitive to phase shift, but data signals are severely affected by it. Phase shift is caused by imperfections in cable media, such as joins and imperfect terminations.

- In a practical sense, imagine you are in the bath. If you drop the soap, it forms ripples where the waves travel outwards towards the edge of the bath. When the wave reaches the edge, it hits the wall of the bath and bounces back. This is like phase shift, which is an abrupt change in the signals relationship.

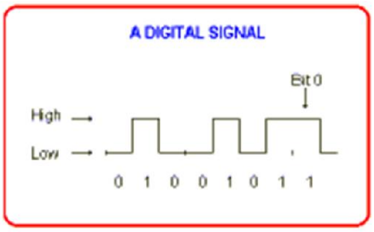
Analogue signals are sent via the PTSN. Digital signals cannot be sent via the PTSN without being first converted to analogue.

Digital Signals

Digital signals are the language of modern day computers. Digital signals comprise only two states. These are expressed as ON or OFF, 1 or 0 respectively. Examples of devices having TWO states in the home are,

- Light Switches: Either ON or OFF

- Doors: Either OPEN or CLOSED



Digital signals require greater bandwidth capacity than analogue signals, thus are more expensive to communicate. This diagram shows a digital signal.

SOME COMMON TERMS

Baud Rate
Baud rate is the reciprocal of the shortest signal element (a measure of the number of line changes which occur every second). For a binary signal of 20Hz, this is equivalent to 20 baud (there are 20 changes per second). For telephone cables, the limiting factor in speed is the number of line changes per second. A line change is defined as switching from one state to another, for instance, switching from a 1 to a 0, or from a 0 to 1 for a digital signal. If the number of line changes per second are exceeded, errors occur and the signal at the receiving end cannot be reliably reconstructed.

Bits Per Second
This is an expression of the number of bits per second. Where a binary signal is being used, this is the same as the baud rate. When the signal is changed to another form, it will not be equal to the baud rate, as each line change can represent more than one bit (either two or four bits).

Digital signals sent via the PSTN need to be converted to analogue first (by using a device called a modem). Digital signals can be sent via the ISDN unmodified.

Bandwidth
Bandwidth is the frequency range of a channel, measured as the difference between the highest and lowest frequencies that the channel supports. The maximum transmission speed is dependant upon the available bandwidth. The larger the bandwidth, the higher the transmission speed. A nominal voice channel has a bandwidth of 3.1KHz. In reality this equates to about 1200bps maximum for a binary digital signal.

Dial Up Speech Circuits provided by Telephone Communication Companies

The voice channel was designed to handle analogue voice in the range 300Hz to 3.4Khz. The nature of voice traffic is

- periodic - talk and then listen
- varying in intensity - talking softly, shouting

In addition, the voice channel was implemented using two way amplifiers, which meant special devices were used to prevent echoes or unwanted oscillations (the circuit suffering from feedback). These devices are called **echo suppressors**, and affect the signal by reducing the available bandwidth of the channel. The amplifiers are designed to boost low level signals and attenuate high level signals, with the intention of trying to maintain an average signal level on the channel.

Data signals are digital in nature and are

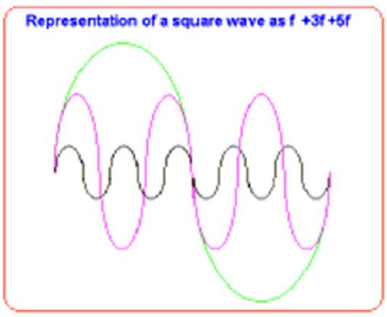
- continuous
- constant in intensity
- require greater channel bandwidth than analogue signals

This causes several problems. The two way amplifiers tend to get overloaded, with the net result of putting too much signal level on the channel. This overflows onto other channels, affected the signals on them also (called **crosstalk**). The second problem is the signals are affected by the bandwidth of the channel, such that only some of the original signal will appear at the other end. This effect becomes more pronounced as the speed of the data signal is increased.

Problems of using Voice Channels for Digital Transmission

A digital signal is comprised of a number of signals. Specifically, the signal is represented as follows,

signal = f + f3 + f5 +f7 +f9 +f11 +f13 ....f(infinity)

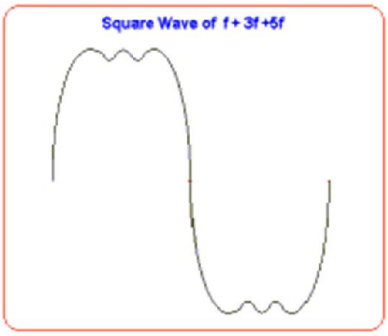


This means a digital signal has a base frequency, plus another at three times the base frequency, plus another at five times the base frequency etc. f3 is called the third harmonic, f5 the fifth harmonic and so on.

The third harmonic is one third of the amplitude of the base frequency (called the fundamental frequency), the fifth harmonic is one fifth the amplitude of the fundamental and so on.

In order to send a digital signal across a voice channel, the bandwidth of the channel must allow the fundamental plus third and fifth harmonic to pass without affecting them too much.

As can be seen, this is what such a signal looks like, and is the minimum required to be correctly detected as a digital signal by the receiver.



Lets consider sending a 2400bps binary digital signal down a voice channel rated with a bandwidth of 3.1KHz. The base frequency of the digital signal is 1200Hz (it is always half the bit rate), so the fundamental frequency will pass through the channel relatively unaltered. The third harmonic is 3600Hz, which will suffer attenuation and arrive severely altered (if at all). The fifth harmonic has no chance of passing the channel.

In this case it can be seen that only the base frequency will arrive at the end of the channel. This means the receiver will not be able to reconstruct the digital signal properly, as it will require f3 and f5 for proper reconstruction.

This results in errors in the detection process by the receiver.

▲ Summary

The Public Switched Telephone Network has, for a long time, provided users with dial up telephone connections on demand. Each connection has supported analogue speech in the voice range of 300Hz to 3400Hz. The signals provided by computers are digital, and are not designed to travel across the dial up telephone connections. In order for digital signals to be sent across a telephone connection, they must be converted to analogue voice tones within the frequency range 300Hz to 3400Hz (this is done by a modem).

Analogue signals have three main characteristics which define them, being amplitude (a measure of how loud they are), frequency (a measure of how often they change) and phase. Speech is an example of an analogue signal. Analogue signals can be sent across a telephone connection. Digital signals comprise two states, and must be converted to an analogue form before being sent across a telephone connection.

DATA COMMUNICATIONS

Part 7: Standards Organizations

Organizations | Summary | Test

Introduction
This section briefly outlines some common standard organizations that are responsible for protocols like RS232 and other issues.

Standards Organizations

Because of the wide number of hardware manufacturers, a standard is essential in order to connect one computer to another computer if a different type. There are recognized and widely accepted standards governing how data is to be transmitted, whether asynchronously, parallel, or synchronously. Standards govern the format of the data, and also specify the hardware details like voltages to use, bit durations, speeds etc.

The major organisations responsible for standards are

- **Electronics Industries Association (EIA)**
  Made up by manufacturers in the USA, and is responsible for RS232 and similar standards.

- **Institute of Electrical and Electronic Engineers (IEEE)**
  Professional organization of engineers. An example is the IEEE-754 standard for representing floating point numbers.
  http://www.ieee.org/

- **American National Standards Institute (ANSI)**
  Represents a number of US standards organizations. Member organizations submit their standards for acceptance. An example is the ANSI standards for representing ASCII characters.
  http://www.ansi.org/

- **International Organization for Standards (ISO)**
  Has standards covering a wide range of computer related topics. The US representative is ANSI. An example is ISO9000 standard for quality assurance.
  http://www.iso.ch

- **International Telecommunications Union (ITU)**
  The ITU co-ordinates international communications and recommends standard interfaces and policies for the interconnection of national networks. It is also involved with allocation of satellite frequencies and orbits. Members of the ITU are industry representatives from member countries.
  http://www.itu.ch

Summary
There are a number of organizations responsible for standards. From a company view, standards are important because they ensure inter-operability of products and reliability.

DATA COMMUNICATIONS

Part 8: RS232 Serial Communications

Introduction
This section introduces the RS232 Serial standard. This is a standard that is used to connect serial devices like Modems to a personal computer. The main features and signals of the standard will be covered.

Electrical Interfaces

An electrical interface is the connection between two devices. There are two common data interfaces that specify international standards for low speed data communication.

- **CCITT V.24**
  Does not specify pin numbers. Details the types of signals to be exchanged. Specifies the absolute voltage levels, impedance's, and timing characteristics for each line.
- **EIA RS232-C**
  Uses V.24 recommendations and further specifies the mechanical connector type and pin numbers to be used (DB-25 pin connector).

THE EIA RS232-C STANDARD

Specifies a 25 pin connector as the standard interface in data communication networks, with lettering pin designations for ground, data, control and timing circuits. The table below shows the designations for each of the 25 pins of the standard.

INTERCHANGE	CIRCUIT No.	PIN No.	DESCRIPTION
AA	101	1	Protective Ground
BA	103	2	Transmit Data
BB	104	3	Receive Data

CA	105	4	Request To Send
CB	106	5	Clear To Send
CC	107	6	Data Set Ready
AB	102	7	Signal Ground
CF	109	8	Receive Line Signal Detect/Carrier Detect
--	--	9	Reserved
--	--	10	Reserved
--	--	11	Unassigned
SCF	122	12	Secondary RLSD
SCB	121	13	Secondary CTS
SBA	118	14	Secondary TD
DB	114	15	Transmitter Signal Element Timing
SBB	119	16	Secondary RD
DD	115	17	Receiver Signal Timing Element
--	--	18	Unassigned
SCA	120	19	Secondary RTS
CD	108.2	20	Data Terminal Ready
CG	110	21	Signal Quality Detector
CE	125	22	Ring Indicator
CH/CI	111/112	23	Data Signal Rate Selector
DA	113	24	Transmit Signal Element Timing
--	--	25	Unassigned

This diagram shows an RS232 female connector, with the designated signals shown for each pin.

DATA COMMUNICATIONS EQUIPMENT (DCE)
An example of a DCE is a modem. A DCE is fitted with a 25 pin female connector.

DATA TERMINAL EQUIPMENT (DTE)
An example of a DTE is a computer terminal. A DTE is fitted with a 25 pin male connector.

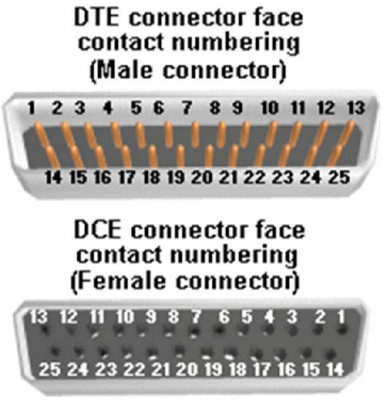
There are two main parts, a mechanical and an electrical standard.

- **EIA RS232-C Mechanical Standard**
  Female connector is connected to DCE and male connector to DTE. Short cables of less than 15 meters (50 feet) are recommended. The pin assignments detailed above must be used.
- **EIA RS232-C Electrical Standard**
  All circuits carry bi-polar low-voltage signals, measured at the connector with respect to signal ground (AB), and may not exceed ±25 volts. Signals are valid in the range ±3 volts to ±25 volts. Signals within the range -3 volts to +3 volts are considered invalid.

  For data lines, binary 1 (a high) is represented by -3 volts to -25 volts, whilst binary 0 is +3 volts to +25 volts.

  For control lines, OFF is represented by -3 volts to -25 volts, whilst binary 0 is +3 volts to +25 volts.

RS232 Connectors
The following diagram shows the 25-pin connector used for the DTE interface. It is a MALE connector, which has 25 pins. Beneath it is the 25 pin FEMALE connector used on the DCE interface. Note that the connectors have a longer side at the top and a shorter side at the bottom. This is to prevent the plugs being inserted into the connectors the wrong way round.



▲ RS-232 Signal Descriptions

The interface transfers data between the computer and the modem via the TD and RD lines. The other signals are essentially used for FLOW CONTROL, in that they either grant or deny requests for the transfer of information between a DTE and a DCE. Data cannot be transferred unless the appropriate flow control line are first asserted.

The interface can send data either way(DTE to DCE, or, DCE to DTE) independently at the same time. This is called FULL DUPLEX operation. Some systems may utilize software codes so that information may only be transmitted in one direction at a time (HALF DUPLEX), and requires software codes to switch from one direction to another (i.e., from a transmit to receive state).

The following is a list of common RS232 signals.

- **Request To Send (RTS)** This signal line is asserted by the computer (DTE) to inform the modem (DCE) that it wants to transmit data. If the modem decides this is okay, it will assert the CTS line. Typically, once the computer asserts RTS, it will wait for the modem to assert CTS. When CTS is asserted by the modem, the computer will begin to transmit data.
- **Clear To Send (CTS)** Asserted by the modem after receiving a RTS signal, indicating that the computer can now transmit.
- **Data Terminal Ready (DTR)** This signal line is asserted by the computer, and informs the modem that the computer is ready to receive data.
- **Data Set Ready (DSR)** This signal line is asserted by the modem in response to a DTR signal from the computer. The computer will monitor the state of this line after asserting DTR to detect if the modem is turned on.

- **Receive Signal Line Detect (RSLD)** This control line is asserted by the modem, informing the computer that it has established a physical connection to another modem. It is sometimes known as *Carrier Detect (CD)*. It would be pointless a computer transmitting information to a modem if this signal line was not asserted. If the physical connection is broken, this signal line will change state.
- **Transmit Data (TD)** is the line where the data is transmitted, a bit at a time.
- **Receive Data (RD)** is the line where data is received, a bit at a time.

A lot of signals work in pairs. Some signals are generated by the DTE, and some signals are generated by the DCE. If you were measuring the signals on a computer which was NOT connected to a modem, you could only expect to see those signals that the DTE can generate.

The following table lists some of the signal pairs and the device responsible for generating them.

DTE	DCE
TD	RD
RTS	CTS
DTR	DSR

Something to think about

You have connected an external modem to a computer using an RS232 cable. After loading the application software, it reports "Modem is not turned on". You check, and find the modem is turned on. Gaining access to a multi-meter device, which is used to read the state of pins on the RS232 connection, which pin do you think you should check to verify that the modem is turned on?.

How to exchange information between a DCE and DTE

Now, lets look at the sequence that occurs when data is transferred between a DTE and a DCE. The data can only be transferred after the correct sequence of signals is followed, for instance, there is no point sending data if the modem is turned off. Lets go through each of the steps involved (i.e., signal line assertions required) to transmit and receive characters across the RS232 interface.

TRANSMITTING DATA (DTE to DCE)

- 1: Assert DTR and RTS
- 2: Wait for DSR

- 3: Wait for CTS
- 4: Transmit the data

Step 1 and 2 are essential to ensure that the modem is on-line and connected to another modem. Waiting for DSR checks that the modem is on-line.

RECEIVING DATA (DCE to DTE)

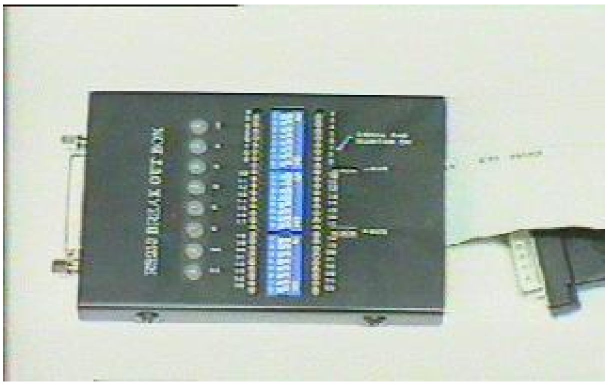
- 1: Assert DTR
- 2: Wait for DSR
- 3: Receive the data

Break-out Boxes

An RS232 breakout box is a device that allows you to monitor the RS232 connection, and connect various signal lines together. It is placed between the DTE and the DCE, so you can see the state of the various signal lines and perform interconnection if required.

Using the breakout box is a matter of determining the signals being asserted and performing interconnection of signal lines if required.

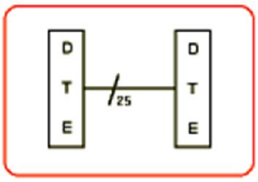
LED's are used to indicate the state of the signal line. RED indicates an active signal (high), and GREEN an inactive signal (low).



CONNECTING TWO DTE DEVICES TOGETHER

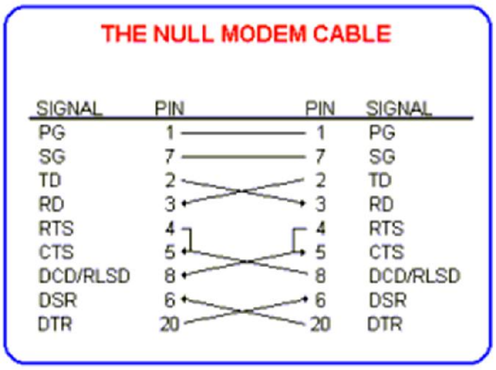
Often, two DTE devices need to be connected together using a serial link. This is for file transfer or printer access. The problem is that DTE devices expect to talk directly to DCE devices, not another device of the same type. DTE's cannot generate signals like DSR and CTS, so connecting two DTE's together will result in neither getting permission to send, and thinking that the modem is off-line (by not receiving DSR).

To allow the interconnection of two DTE devices without using DCE's, a special type of cable must be used. This is called a *Null Modem Cable*, which fools the DTE into thinking that it is connected to a DCE device. In this case, modems are not used, so the connection looks like.



DESIGNING A NULL-MODEM CABLE

In designing a NULL-MODEM cable, the DTE signals from one computer are swapped over as inputs to supply the DCE expected signals on the other DTE.



As you can see from the diagram, when two DTE's are connected together, the signal lines from one DTE are transposed to the other DTE, fooling it into thinking that it is communicating with a DCE.

RS232D (9 pin Connector)

The following table illustrates the 9 pin serial connector as found on most PC's today. This has all but replaced the previous 25 pin connector found on earlier PC's.

SIGNAL	PIN No.
Carrier Detect	1
Receive Data	2
Transmit Data	3
Data Terminal Ready	4
Signal Ground	5
Data Set Ready	6
Request To Send	7
Clear To Send	8
Ring Indicator	9

Summary

The RS232-C interface connects a DTE (computer) to a DCE (modem). It is a 25 pin interface, where some signals are generated by the DTE, and some by the DCE. It supports slow speed data communications. The DTE uses a male connector and the DCE a female connector.

A breakout box is a device used to monitor or change the state of the interface lines. It is handy in troubleshooting problems. A null modem cable transposes signals so that a DTE can exchange data with another DTE.

DATA COMMUNICATIONS

Part 9: Baudot and ASCII Codes

Baudot | ASCII | Summary

Introduction
This section introduces the concept of data codes. This refers to the way in which bits are grouped together to represent different symbols. There are a number of different codes, but the most common code in use today is the ASCII code.

DATA CODES
This refers to the way in which data is represented. The sender and receiver must use the same code in order to communicate properly. Here, we will briefly look at two common codes, one which was developed earlier on and was widely used in early telegraph systems, and the other, which is in widespread use today.

The Baudot Code
The Baudot code was used extensively in telegraph systems. It is a five bit code invented by the Frenchman Emile Baudot in 1870. Using five bits allowed 32 different characters. To accommodate all the letters of the alphabet and numerals, two of the 32 combinations were used to select alternate character sets.

Each character is preceded by a start bit, and followed by a stop bit. It is an asynchronous code, and thus suited for low speed data communication.

Value	LTRS shift	FIGS shift	Value	LTRS shift	FIGS shift
3	A	-	23	Q	1
25	B	?	10	R	4
14	C	:	5	S	.
9	D	Who are u	16	T	5
1	E	3	7	U	7
13	F	!	30	V	;
26	G	&	19	W	2
20	H	#	29	X	/
6	I	8	21	Y	6

11	J	Bell	17	Z	"
15	K	(	0	BLANK	BLANK
18	L	)	31	LTRS	LTRS
28	M	.	27	FIGS	FIGS
12	N	,	4	SPACE	SPACE
24	O	9	8	CR	CR
22	P	0	2	LF	LF

For instance, lets consider coding the phase "JAMES BOND 007 SAYS HI!" using the Baudot code. To switch between the LTRs and FIGs requires the use of a LetterShift or a FigureShift. Once switched, you stay in that mode till you want to switch back again. So, here is the phrase encoded in Baudot.

J	A	M	E	S		B	O	N	D		0	0	7		S	A	Y	S		H	I	!
31 11	3	28	1	5	4	25	24	12	9	4	27 22	22	7	4	31 5	3	21	5	4	20	6	27 13

ASCII (American Standard Code for Information Interchange)

The ASCII code is the most popular code for serial data communications today. It is a seven bit code (128 combinations), and thus supports upper and lowercase characters, numeric digits, punctuation symbols, and special codes. The table below lists the values for each character in the ASCII set.

	00	**01**	**02**	**03**	**04**	**05**	**06**	**07**	**08**	**09**	**0A**	**0B**	**0C**	**0D**	**0E**	**0F**
00	NUL	SOH	STX	ETX	EOT	ENQ	ACK	BEL	BS	TAB	LF	VT	FF	CR	SO	SI
10	DLE	DC1	DC2	DC3	DC4	NAK	SYN	ETB	CAN	EM	SUB	ESC	FS	GS	RS	US
20		!	"	#	$	%	&	'	(	)	*	+	,	-	.	/
30	0	1	2	3	4	5	6	7	8	9	:	;	<	=	>	?
40	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
50	P	Q	R	S	T	U	V	W	X	Y	Z	[	\	]	^	_
60	`	a	b	c	d	e	f	g	h	i	j	k	l	m	n	o
70	p	q	r	s	t	u	v	w	x	y	z	{	|	}	~	DEL

To work out a particular value from the table, you first determine the row value, then add the column value. For example, the character **A** has a value of 41, being a row value of 40 and a column value of 1.

ASCII is also used as the data code for keyboards in computers. **Control Codes** have values between 00 and 1F (hexadecimal). Control codes are used in binary synchronous communication, and device control codes in communicating with devices such as printers or terminals.

A control code can be generated from a keyboard by holding down the Ctrl key and pressing another key. For instance, holding down the Ctrl key and pressing the A key generates the control code SOH.

Summary

Data codes have always been in widespread use from mankinds early history. From the use of hand signals to mirrors flashing signals across the land, to smoke signals of the American Indians, information has been coded and sent by a variety of means.

ASCII is the most widespread data code in use today. It is a seven bit code, but with the world rapidly shrinking and global boundaries becoming blurred, the necessity to communicate across language barriers has exposed the limitations of this code. Another code, **unicode**, offers some promise in this multi-language area.

DATA COMMUNICATIONS

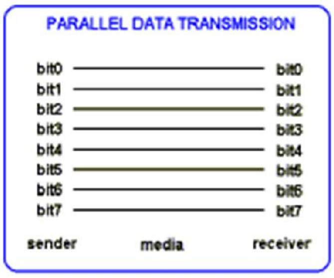
Part 10: Channel Organization

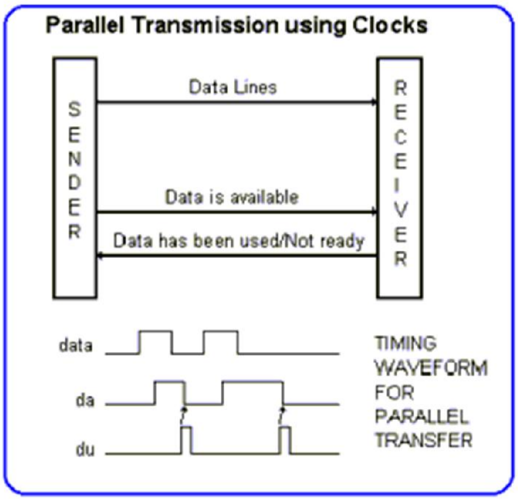
Parallel | Serial | Summary

Introduction
This section briefly discusses how data might be sent from one point to another, namely via a parallel method, or via a serial method. Each technique has its own advantages and disadvantages.

Channel Organization
Data may be transmitted between two points in two different ways. Let's consider sending 8 bits of digital data (1 byte). These bits may be sent all at once (*in parallel*), or one after the other (*serial*).

- **Parallel**
  Each bit uses a separate wire. If there is eight bits sent at a time, this will require 8 wires, one for each data bit. The organization looks like,





To transfer data on a parallel link, a separate line is used as a clock signal. This serves to inform the receiver when data is available.

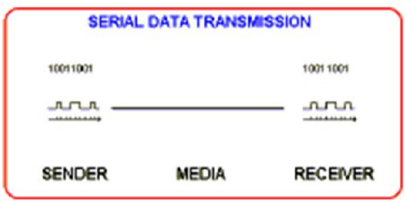
In addition, another line may be used by the receiver to inform the sender that the data has been used, and its ready for the next data.

In the above diagram, the sender places the data on the data lines, then signals the receiver that data is available by asserting (sending a pulse) on the **DA** (Data is available) line. After reading the state of the data lines, the receiver signals back to the sender that it has processed the data and is now ready for some more data by asserting the **DU** (Data used) line. The sender, upon getting the DU signal, removes the data and sends the next data element in the same manner.

This exchange of signals such as DA and DU between the sender and the receiver is called a **handshake**. These handshake signals allow the sender and receiver to keep synchronized (work on the same data at the same time in the proper sequence).

Parallel transmission is obviously faster than serial, because more than one bit is sent at a time. Parallel transmission is good only for short links, and examples are found in all computers. The address, data and control buses which interface the processor to other peripherals inside the computer are all parallel buses. In addition, most printers on PC's (LPT1/LPT2) use a parallel interface, commonly called the *Centronics Interface*.

- **Serial**
  Each bit is sent over a single wire, one after the other. The organization looks like,



No signal lines are used to convey clock (timing information) and handshake signals. There are two methods (asynchronous and synchronous) in which timing information is encoded with the signal so that the sender and receiver are synchronized (working on the same data at the same time). If no clock information was sent, the receiver can mis-interpret the arriving data (due to bits being lost, going too slow). In asynchronous, each character is synchronized using a start and stop signal. In synchronous, each group or block of characters is synchronized using a synchronize flag.

In the following sections, serial transmission will be further investigated. A common standard for serial communications is **RS232**

Summary

Parallel transmission sends each bit using a separate wire. In addition, extra wires are needed to transfer the data between the sender and receiver. These handshake signals allow the data to be transferred in the correct sequence. Computers often send data in parallel form because it is fast. An example of a parallel format is the Centronics parallel interface.

Serial data is slower than parallel, but suited to long distances. There is no need for extra wires to convey handshake signals, as the data is packaged in a variety of different ways (prefixed and suffixed with additional information). An example of a serial format is RS232.

DATA COMMUNICATIONS

Part 11: Asynchronous and Synchronous Protocols

Protocols | Asynchronous | Synchronous | Summary

Introduction
This section briefly discusses the differences between two different methods of Serial transmission, namely, asynchronous and synchronous. A protocol establishes a means of communicating between two systems. As long as the sender and receiver each use the same protocol, information can be reliably exchanged between them. We shall look at two common protocols used in Serial data communications, the first is known as **Asynchronous**, the second as **Synchronous**.

PROTOCOLS
A protocol is a set of rules which governs how data is sent from one point to another. In data communications, there are widely accepted protocols for sending data. Both the sender and receiver must use the same protocol when communicating. One such rule is . . . .

BY CONVENTION, THE LEAST SIGNIFICANT BIT IS TRANSMITTED FIRST

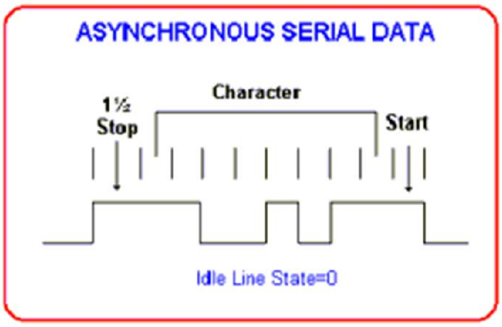
Asynchronous Transmission
The asynchronous protocol evolved early in the history of telecommunications. It became popular with the invention of the early tele-typewriters that were used to send telegrams around the world.

Asynchronous systems send data bytes between the sender and receiver by packaging the data in an envelope. This envelope helps transport the character across the transmission link that separates the sender and receiver. The transmitter creates the envelope, and the receiver uses the envelope to extract the data. Each character (data byte) the sender transmits is preceded with a start bit, and suffixed with a stop bit. These extra bits serve to synchronize the receiver with the sender.

In asynchronous serial transmission, each character is packaged in an envelope, and sent across a single wire, bit by bit, to a receiver. Because no signal lines are used to convey clock (timing) information, this method groups data together into a sequence of bits (five - eight), then prefixes them with a start bit and appends the data with a stop bit.

The purpose of the start and stop bits was introduced for the old electromechanical Tele-typewriters. These used motors driving cams that actuated solenoids that sampled the signal at specific time intervals. The motors took a while to get up to speed, thus by prefixing the first data bit with a start bit, this gave time for the motors to get up to speed. The cams generate a reference point for the start of the first data bit.

At the end of the character sequence, a stop bit was used to allow the motors/cams etc to get back to normal. In addition, it was needed to fill in time in case the character was an end of line, when the Tele-typewriter would need to go to the beginning of a new-line. Without the stop character, the machine could not complete this before the next character arrived.



It's important to realize that the receiver and sender are re-synchronized each time a character arrives. What that means is that the motors/cams are restarted each time a start bit arrives at the receiver.

Nowadays, electronic clocks that provide the timing sequences necessary to decode the incoming signal have replaced the electromechanical motors.

This method of transmission is suitable for slow speeds less than about 32000 bits per second. In addition, notice that the signal that is sent does not contain any information that can be used to validate if it was received without modification. This means that this method does not contain error detection information, and is susceptible to errors.

In addition, for every character that is sent, an additional two bits is also sent. Consider the sending of a text document which contains 1000 characters. Each character is eight bits, thus the total number of bits sent are 10000 (8 bits per character plus a start and stop bit for each character). This 10000 bits is actually 1250 characters, meaning that an additional 250 equivalent characters are sent due to the start and stop bits. This represents a large overhead in sending data, clearly making this method an inefficient means of sending large amounts of data.

▲ Summary for Asynchronous

Transmission of these extra bits (2 per byte) reduce data throughput.

Synchronization is achieved for each character only. When the sender has no data to transmit, the line is idle and the sender and receiver are NOT in synchronization. Asynchronous protocols are suited for low speed data communications, and there is no method of error checking inherent in this protocol.

Synchronous Transmission

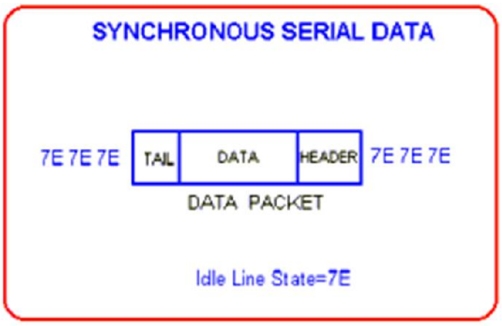
One of the problems associated with asynchronous transmission is the high overhead associated with transmitting data. For instance, for every character of 8 bits transmitted, at least an additional overhead of 2 bits is required. For large amounts of data, this quickly adds up. For example, to transmit 1000 characters, this requires 12000 bits, an extra 2000 bits for the start and stops. This is equivalent to an overhead of 250 characters. Another problem is the complete lack of any form of error detection. This means the sender has no method of knowing whether the receiver is correctly recognizing the data being transmitted.

In synchronous transmission, greater efficiency is achieved by grouping characters together, and doing away with the start and stop bits for each character. We still envelop the information in a similar way as before, but this time we send more characters between the start and end sequences. In addition, the start and stop bits are replaced with a new format that permits greater flexibility. An extra ending sequence is added to perform error checking.

A start type sequence, called a header, prefixes each block of characters, and a stop type sequence, called a tail, suffixes each block of characters. The tail is expanded to include a check code, inserted by the transmitter, and used by the receiver to determine if the data block of characters was received without errors. In this way, synchronous transmission overcomes the two main deficiencies of the asynchronous method, that of inefficiency and lack of error detection.

There are variations of synchronous transmission, which are split into two groups, namely character orientated and bit orientated. In character orientated, information is encoded as characters. In bit orientated, information is encoded using bits or combinations of bits, and is thus more complex than the character orientated version. Binary synchronous is an example of character orientated, and High Level Data Link Control (HDLC) is an example of bit orientated.

In asynchronous transmission, if there was no data to transmit, nothing was sent. We relied on the start bit to start the motor and thus begin the preparation to decode the incoming character. However, in synchronous transmission, because the start bit has been dropped, the receiver must be kept in a state of readiness. This is achieved by sending a special code by the transmitter whenever it has no data to send.



In bit orientated protocols, the line idle state is changed to 7E, which synchronizes the receiver to the sender. The start and stop bits are removed, and each character is combined with others into a data packet.

User data is prefixed with a header field, and suffixed with a trailer field which includes a checksum value (used by the receiver to check for errors in sending).

The ***header field*** is used to convey address information (sender and receiver), packet type and control data. The ***data field*** contains the users data (if it can't fit in a single packet, then use multiple packets and number them). Generally, it has a fixed size. The ***tail*** field contains checksum information which the receiver uses to check whether the packet was corrupted during transmission.

Summary

Asynchronous transmission is suited for low speed serial transmission and does not include error checking as part of the protocol. Each character is contained in an envelope of a start and stop bit. It is inefficient.

Synchronous transmission can achieve higher speeds than asynchronous. In addition, it is an error checking protocol, and much more efficient because i t groups characters together into blocks.

DATA COMMUNICATIONS

Part 12: Binary Synchronous Protocol

Binary Synchronous | Summary

Introduction
This section briefly discusses the Binary Synchronous protocol, which uses characters of the ASCII set to exchange data.

SYNCHRONOUS PROTOCOLS

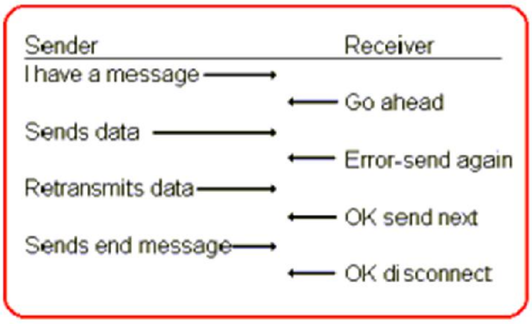
Synchronous protocols involve sending timing information along with the data bytes, so that the receiver can remain in synchronization with the sender. When the sender has no data to transmit, the sender transmits idle flags (a sequence of alternating 0's and 1's) to maintain sender/receiver synchronization. Data bytes are packaged into small chunks called packets, with address fields being added at the front (header) and checksums at the rear of the packet.

The envelope which holds the data and transports it across the link between the sender and receiver is changed, which allows more characters to be sent together, and for error checking to be an inherent feature of this protocol. This overcomes the two main deficiencies of the asynchronous protocol.

Character Orientated Protocols (COP)
In character orientated protocols, each character has significance. In other words, when a character arrives at the receiver, the character has two meanings, it's either a data byte, or it's a control byte (which is used as information signals between the sender and receiver). The main COP in use today is known as BI-SYNC, or **binary synchronous**.

Each character sent is transmitted using the ASCII code. Control bytes obviously have values in ASCII of between 00 and 1F, whereas data bytes have values between 20 and 7F.

Communication takes the form of a hand-shake between the sender and receiver. Communication of a message from sender to receiver takes the following format,



As you can see, this is a HALF-DUPLEX (which means only one side talks at once) method of communication. Long messages are broken up into a series of smaller data packets, and transmitted one at a time across the link. Each packet is acknowledged before the next packet is transmitted.

If a packet is not acknowledged, the sender will time out and then retransmit the packet. If the packet is acknowledged by the receiver, the sender sends the next packet and so on until the entire message has been sent. If a packet is received and contains errors, the receiver will send a negative acknowledge, which requests the sender to send it again.

Data bytes contain data according to the ASCII code (for text), or simply a value between 0-255 for binary data. Control bytes determine the behavior of the communication link, and are used for a range of different purposes.

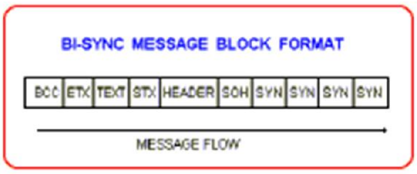
Some examples of control bytes are,

Code	**Meaning**
SYN	Synchronize character, it establishes and maintains character synchronization prior to a message block and during transmission. Also used as a fill when there is no messages to be sent
STX	Start of Text, transmitted before the first data characters. Signifies that a block of data bytes follows.
ETX	End of Text, terminates a data block begun with SOH or STX and terminates the end of a sequence of blocks. The receiver will then send ACK or NACK depending upon the correct receipt of the message blocks.
EOT	End of Transmission, this concludes the transmission.
ACK	Positive Acknowledge, this is sent by the receiver to indicate successful receipt of the previous message block, or as a successful response to a selection (multipoint) or line bid (point to point).
NAK	Negative Acknowledge, this is sent by the receiver to indicate the unsuccessful receipt of the previous message block. It is also used to indicate a negative response to a selection or line bid.

SOH	Start of Header, transmitted before header characters, which specify routing or priority information for the message.
ETB	End of Transmission Block, indicates the end of the text block which started with STX or SOH. The receiver will then send ACK or NACK depending upon the correct receipt of the message blocks.

Binary Synchronous Message Blocks

Messages are sent in blocks. Message blocks have the following format,



Each message block can contain up to three parts,

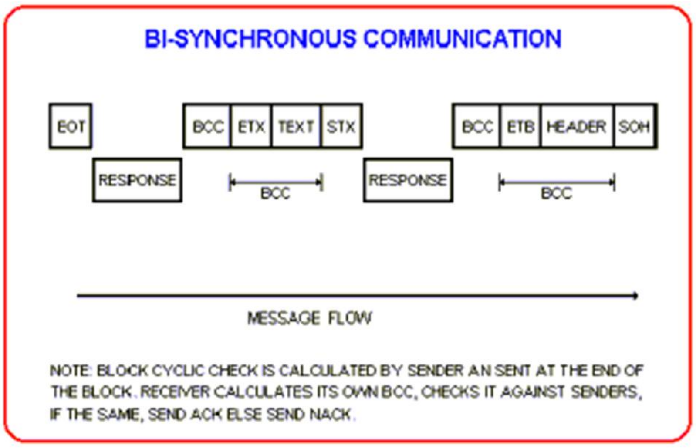
- an optional header
- the text
- a trailer

The control characters used to identify these parts are,

- SOH indicates the header follows
- STX indicates the text follows
- ETX indicates the end of the text block

SYN characters are used to establish synchronization between the sender and receiver. The message block follows the SYN characters.

A long message is split up into a number of small message blocks by the sender. The trailer for each block consists of a block check character (BCC). As the message block is transmitted, both the sender and receiver each generate their own BCC. At the end of receiving the trailer, the receiver compares its own BCC against that of the senders. If they are the same, this indicates the block has been successfully received without errors, so the receiver will reply using a positive acknowledge (ACK). If the BCC of the receiver does not match that of the sender, the receiver knows an error has occurred during transmission, and will instruct the sender to re-send the block by replying with a negative acknowledge (NACK).



A preset number of repeated attempts to re-transmit a corrupted message block (upon receiving a NACK) will be made before the sender will abort the transmission.

There is one major problem associated with this protocol. Consider the sending of a binary file, which has not only text, but also contains values which would be interpreted as control codes. What happens if a control character such as ETX occurs in the text field? The receiver would interpret this as the end of the text field and take the next character as the BCC. This is incorrect.

To prevent this problem and allow any characters to appear in the data field (say to transmit a binary file), *data transparency* is used. This means preceding each control character with the Data Link Escape control character (DLE). If the receiver gets a DLE code, it knows the next byte is a control code. The DLE control character is discarded by the receiver.

What happens if the sender has a DLE code as part of the text block? In this case the sender precedes it with a DLE, and the receiver discards the first and uses the second as a data byte.

Summary

Binary synchronous is a well established protocol and has been around for a number of years. It uses characters to convey both data and control information.

It provides error checking by the inclusion of a Block Check Character, and has the ability to send binary files.

DATA COMMUNICATIONS

Part 13: Bit Orientated Protocols

Bit orientated protocols | HDLC frames
Sliding windows | Summary

Introduction
This section briefly discusses Bit Orientated Protocols, which uses bits to exchange data.

Bit Orientated Protocols (BOP)

Character orientated protocols are still inefficient. This is because a character is used to convey meaning. As the number of meanings increase, the overhead involved also increases, as a character is used to signal the meaning.

In bit orientated protocols, each bit has significance. The position and value of each bit in the data stream determines its function. Thus, a single character can hold 256 different meanings in a bit orientated protocol. This reduces the information needed to convey additional information, thus increasing the efficiency of the protocol.

Examples of these types of protocols are,

- X.25 CCITT standard for packet data transmission
- HDLC high level data link control (adopted by ISO in 1970's)
- SDLC synchronous data link control (developed by IBM)

Links between sender and receiver can be either half duplex, full duplex or both. Information can be sent across the network in two different ways, travelling different routes to the receiver (*datagram*), or travelling the same route (*virtual circuit*).

Information is packaged into an envelope, called a FRAME. Each frame has a similar format

- header containing routing and control information
- body
- tail containing checksum data

Frames are responsible for transporting the data to the next point. Consider data that is to be sent from a source to a destination. This involves several intermediate points (called stations). The data is placed into a frame and sent to the next station, where the frame is checked for validity and if valid, the data extracted. The data is now repackaged into a new frame and sent by that station to the next station, and the process repeats till the data arrives at the destination.

When a station transmits a frame, it keeps a copy of the frame contents till the frame is acknowledged as correctly received by the next station. When a station receives a frame, it is temporarily stored in a buffer and checked for errors. If the frame has errors, the station will ask the previous station to resend the frame. Frames that are received without errors are also acknowledged, at which point the sending station can erase its copy of the frame.

A receiving station has a limited amount of buffer space to store incoming frames. When it runs out of buffer space, it signals other stations that it cannot receive any more frames.

Data is placed into frames for sending across a transmission link. The frame allows intelligent control of the transmission link, as well as supporting multiple stations, error recovery, intelligent (adaptive) routing and other important functions.

For the purposes of sending data on an HDLC link, there are two types of stations,

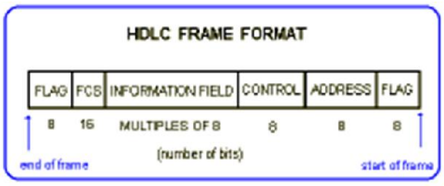
- primary station (issues commands)
- secondary station (responds to commands)

HDLC Primary Station
The primary station is responsible for controlling the data link, initiating error recovery procedures, and handling the flow of transmitting data to and from the primary. In a conversation, there is one primary and one or more secondary stations involved.

HDLC Secondary Station
A secondary station responds to requests from a primary station, but may under certain modes of operation, initiate transmission of its own. An example of this is when it runs out of buffer space, at which point it sends RNR (receiver not ready) to the primary station. When the buffer space is cleared, it sends RR (receiver ready) to the primary station, informing the primary that it is now ready to receive frames again.

▲ HDLC FRAMES

Data is packaged into frames to be sent across the HDLC link. The typical frame format used is,



Frames are a transport mechanism. Their sole purpose is to transport the data across one link, not end to end. As the frame arrives at the other end of the link, it is checked to errors, and if its okay, the data is stripped out of the frame, a new frame generated for it, the data inserted into the new frame, and then transmitted on the next link and so on until the data reaches its destination.

The various fields of a frame are

- **Frame Start/End Flag** Each frame begins and ends with a special 8 bit sequence, 7E hexadecimal (binary 01111110). The end flag can also be the start flag of the next frame.
- **Address Field** In command frames, this contains the address of the secondary (destination) station. In response frames (a reply to a primary station), the address specifies the secondary station sending the response.
- **Control Field** This field holds commands and responses. Examples are sequence numbers (frames are numbered to ensure delivery), and poll (you must reply) and final (this is the last frame) bits.
- **Information Field** This field contains the data. It can contain any sequence of bits. It normal practice it is a multiple of 8 bits.
- **Frame Check Sequence Field** This field contains a 32 bit Cyclic Redundancy Check (CRC) which covers the A, C and I fields.

▲ *HDLC FRAME TYPES*

There are three types of frames

- information frames, which contain data (the information field)
- supervisory frames, which contain commands and responses
- un-numbered frames, which contain commands and responses and sometimes data

Information frames

- are sequentially numbered
- carry data, message acknowledgements, poll and final bits

Supervisory Frames

- perform link supervisory control
- message acknowledgements
- retransmit requests
- signal temporary hold on receipt on I frames (if secondary is busy)

Unnumbered Frames

- provide flexible format for additional link control
- do not have sequence numbers

Sliding Windows

Because frames are numbered, it is possible for a primary station to transmit a number of frames without receiving an acknowledgement for each frame. The secondary can store the incoming frames and reply using a supervisory frame with the sequence number bits in the control field set so as to acknowledge a group of received frames.

If the secondary runs out of buffer space to store incoming Information frames, it can transmit a supervisory frame informing primary stations of its status. Primary stations will thus keep their Information frames and wait till the secondary is again able to process Information frames.

When a secondary cannot process Information frames, it must still be able to process incoming supervisory and unnumbered frames (because of status requests).

Summary

A bit orientated protocol sends information as a sequence of bits. An example of a bit orientated protocol is HDLC. Frames are used as a transport mechanism to transport data from one point to another. A frame contains error checking information which allows data to be sent reliably from a sender to a receiver.

Three frames types are defined, and data is normally send using Information frames. At any one time, a number of Information frames can be unacknowledged by a secondary station, and this is called the *sliding window value*, which defaults to 2, but can be negotiated when a call is first established.

DATA COMMUNICATIONS

Part 14: Simplex, Half-Duplex and Full Duplex

Simplex | Half Duplex | Full Duplex | Summary

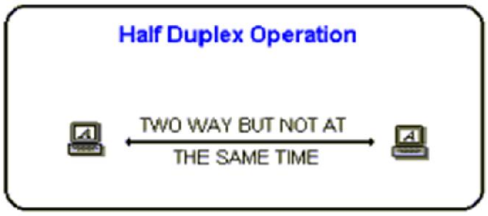
Introduction
This section briefly discusses the modes of channel operation, namely, simplex, half-duplex and full-duplex operation. Each is suited a particular type of application, and has its own advantages and disadvantages.

Modes of Channel Operation

Simplex
Data in a simplex channel is always one way. Simplex channels are not often used because it is not possible to send back error or control signals to the transmit end.

It's like a one way street. An example of simplex is Television, or Radio. The simplex channel also corresponds directly to Shannon's model of communication discussed earlier.

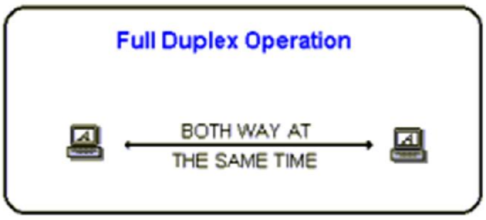
Half Duplex
A half-duplex channel can send and receive, but not at the same time. It's like a one-lane bridge where two way traffic must give way in order to cross. Only one end transmits at a time, the other end receives. In addition, it is possible to perform error detection and request the sender to retransmit information that arrived corrupted. In some aspects, you can think of Internet surfing as being half-duplex, as a user issues a request for a web document, then that document is downloaded and displayed before the user issues another request.



Another example of half-duplex is talk-back radio, and CB Radio (Citizens Band). You might have seen movies where truckies (drivers of very big trucks) communicate to each other, and when they want the other person to speak they say "over". This is because only one person can talk at a time.

Full Duplex

Data can travel in both directions simultaneously. There is no need to switch from transmit to receive mode like in half duplex. Its like a two lane bridge on a two-lane highway. Have you ever watched these television talk shows where the host has a number of people on the show, and they all try to talk at once. Well, that's full duplex!



Of course, in the world of data communications, full duplex allows both way communication simultaneously. An example can be a consumer which uses a cable connection to not only receive TV channels, but also the same cable to support their phone and Internet surfing. All these activities can occur simultaneously.

Summary

The three modes of channel operation are simplex, half-duplex and full-duplex.

Simple is a one way communication and there is no means of informing the sender to retransmit data in case of errors. There is however a good example of the retransmission of data, and that is TeleText, which sends text based data on top of a Television signal. A special decoder displays the Teletext data as a series of pages. These pages are sequenced and repeated, so if a page arrives corrupted, the user just needs to wait a little while till it is resent.

Half-duplex and full-duplex are the other two methods. As telephone companies become more aware of the added services that customers require, such as Internet access and Television, it is probable that a single connection to your home will provide you with a range of services, which you can use. This would require a full-duplex connection.

DATA COMMUNICATIONS

Part 15: Modems

Modems | Smart Modems | AT Commands | Standards | Summary

Introduction
This section briefly discusses modems, what they are, what they are used and a summary of the standards associated with them.

Modems
Modems are devices that allow digital data signals to be transmitted across an analogue link. As has already been discussed, the connections provided by telephone companies for the use of speech via dial up telephones is analogue based. However, in recent times, with the development of on-line information boards (bulletin boards) and the Internet, it is now standard practice to want to use the dial-up telephone connection to access these services.

As the computer uses digital information, and the telephone line uses analogue signals, a device is needed which converts the digital data from the computer and converts it to analogue tones (within the voice channel range of 300Hz to 3400Hz) so that the signals can travel across the dial up speech connection. At the other end, the signals are converted back to digital. The device that converts digital signals to analogue for transmission across a dial-up telephone connection, and then converts them back again, is a modem.

Modem stands for *modulator demodulator*. A modem changes the digital signal to an analogue frequency, and sends this tone across the analogue link. At the other end, another modem receives the signal and converts it back to digital.

A modern modem looks like,

Smart Modems

This is a standard modem with a micro-processor to provide both data communications and automatic dialing in the one unit. These modems offer a range of features,

- auto dial, the modem can seize the phone line and dial a number
- auto answer, the modem can answer an incoming call
- auto fallback, the modem can use a lower speed if the line is noisy
- accept commands, the modem can be reconfigured

Modem Commands

Modems can receive and act on commands from the computer terminal. Commands begin with the symbol AT and are terminated with a carriage return.

The range of commands is called the AT Command Set. This is detailed in the reference listed above. Depending upon how the modem is configured (whether to echo results of commands), the modem will return the symbol OK in response to a command request if it is performed.

The AT Command Set

Hayes Corporation developed a smart modem that accepted AT type commands. This is now a widely accepted standard. The following is a brief list of the AT command set.

- ATA Answer call
- ATA Repeat last command
- ATC Turn modems carrier signal ON (ATC1) or OFF (ATC0)
- ATD Dial a telephone number (ATDT255-0789)
- ATE Enable (ATE1) or disable (ATE0) the echo of characters to the screen
- ATH Hang up the phone (ATH0) or pick up the phone (ATH1)
- ATM Turn on modem speaker (ATM1) or turn off speaker (ATM0)
- ATO Place modem on-line

- ATP Pulse dial
- ATS Set values in modem 'S' registers
- ATT Touch tone dial
- ATZ Reset the modem

ITU V.XX Modem Standards

The International Telegraphic Union (ITU) is an international organization responsible for the establishment of the world wide V standards.

V Series Standard	Details
V.22	1200bps full duplex 2 wire dial-up for use in general telephone networks
V.22 bis	2400bps dial-up or 2 wire leased line
V.23	600/1200bps dial-up for use in general telephone networks
V.26 bis	2400/1200bps dial-up for use in general telephone networks
V.27	4800bps leased line
V.27 bis	4800/2400bps leased line with automatic adaptive equaliser
V.27 ter	4800/2400bps dial-up for use in general telephone networks
V.29	9600bps leased line
V.32	9600bps dial-up for use in general telephone networks
V.32 bis	14400bps dial-up for use in general telephone networks or leased line, synchronous or asynchronous full-duplex
V.33	14000bps over 4 wire leased line, synchronous full-duplex
V.34	28800bps dial-up for use in general telephone networks or leased line, synchronous or asynchronous full-duplex
V.34 Enhanced	33600bps dial-up for use in general telephone networks or leased line, synchronous or asynchronous full-duplex
V.35	48Kbps using 60-108Khz group-band circuits

Summary

A modem is required to send digital data over an analogue connection such as a dial up telephone line. Modems change the digital data to analogue tones that fit within the frequency range of the voice channel. A modem can also convert these analogue tones back to the original digital data.

Nowadays, it is common for the home user to access the Internet via a dial up telephone connection using a modem. Typically, these modems run at 28.8Kbps or 33.3Kbps, although 56.6Kbps modems are now becoming available.

DATA COMMUNICATIONS

Part 16: Modulation

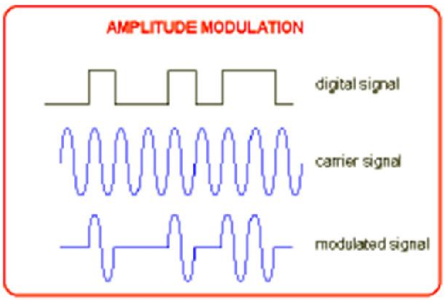
Amplitude | Frequency | Phase | Summary

Introduction
This section briefly discusses modulation, the process a modem uses to convert the digital data into analogue tones which are sent over the dial up connection.

MODULATION
Is the process used to describe how the digital signal is changed so it can be transmitted across the analogue link.

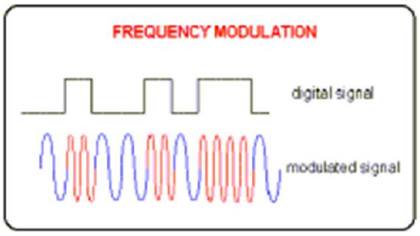
Modulation methods
This refers to how the digital signal is altered so that it can be sent via the analogue PTSN. There are a number of different methods. The more complex methods allow much higher transmission rates (bits per second) than the simpler methods.

- **Amplitude Modulation**
  This uses a single carrier frequency, on for high, off for low. Note that there is no carrier signal being sent for approximately half the time.



This modulation method is suitable only for low speed transmission. The frequency of the carrier signal used depends on the protocol standard being used.

- **Frequency Modulation**



This method uses two carrier signals, one for high one for low. The higher frequency is associated with binary 1, the lower frequency with binary 0. Also called frequency shift keying, this method is used for 1200 bps modems or slower speeds.

- Higher speeds are not possible. If you increase the number of bits per second of the digital signal, the corresponding carrier tones are on for a shorter duration. Eventually, the stage is reached where only a few cycles of the carrier tone is being sent for each digital bit. In order to reliably detect the carrier signal at the receiver, up to 5 complete cycles are required. Less than this, and the carrier signal may be misinterpreted.
- To compensate for this, you might consider the option of increasing the frequency of the carrier signals. But, You just cannot raise the carrier frequency higher as the voice channel will cut it off (remember the bandwidth limits of the voice channel).
- **Phase Modulation**
  This method uses a single carrier frequency and alters the phase of the carrier. Variations of this method allow coding of multiple bits (i.e. two or four bits) to represent a single carrier change.

  Normally, a change from binary 1 to binary 0 is represented as a phase shift of 180 degrees.

  By coding two bits per phase change, this doubles the number of bits per second. This is called two level coding. This method is suitable for 2400 bps modems (CCITT V.26).

Bit Pattern	Degrees Phase Shift
00	45º
01	135º
10	315º
11	225º

Remember that baud rate is the number of changes per second. By moving to two bits per phase change, the number of changes per second is still the same (baud rate), but the number of bits per second is doubled. Baud rate is only equal to bit rate where single level encoding is used.

Three bit encoding is used for CCITT V.27 modems at 4800 bps. Higher speeds are obtained by using four bits, eight phase changes, with two or four different amplitudes. Where changes in amplitude are combined with phase changes, this is called **phase amplitude modulation**.

Summary

Modulation is the mechanism that a modem uses to convert digital signals to analogue signals for sending across a telephone connection. The three different type of modulation are amplitude, frequency and phase, although phase amplitude modulation is a combined technique of different phase and relative signal levels that offers the greatest speed of the methods discussed above.

The limitation with sending data over a telephone connection is related to baud rate, which is a measure of the line changes per second. Once this limit is exceeded, the signal is lost and cannot be reliably detected at the receiver.

In amplitude and frequency modulation, each line change represents a bit change in the input signal. Thus, as the input signal is sped up, the baud rate limit is reached, and at that point, no increase in speed is possible.

In phase modulation, groups of bits represent a line change. The limiting factor is always the number of line changes per second, it is just that in phase modulation each line change represents more than one bit of the input signal, and in this way, higher speeds are possible.

DATA COMMUNICATIONS

Part 17: TCP/IP and Inter-Networking

Inter-networking | Features | History | Relationship to OSI
Addressing | Address Resolution | IP | ICMP | UDP
TCP | Domain Names | SNMP | Bootp | Network Services
Sockets | Network Commands | Network Files | Summary

Introduction
This section briefly discusses TCP/IP, a common protocol method used to interconnect computers together, and also serves as the default protocol for accessing information over the Internet.

Inter Networking

UNIX systems are usually interconnected using TCP/IP (transmission control protocol, Internet protocol). This is a protocol mechanism that is widely used by large networks world wide to interconnect computers of different types.

A protocol is a set of rules that govern how computers talk to each other. TCP/IP is a widely used and very popular protocol. With TCP/IP, different computer systems can reliably exchange data on an interconnected network. It also provides a consistent set of application programming interfaces (API's) to support application development. This means that software programs can use TCP/IP to exchange data. An example of this is web servers and web browsers, software applications which use TCP/IP to exchange data.

Features of TCP/IP

Below are a few of the common features of TCP/IP.

- **File Transfer**
  The file transfer protocol (FTP and remote copy (RCP) applications let users transfer files between their computer systems.

- **Terminal Emulation**
  Telnet and rlogin provide a method for establishing an interactive connection between computer systems.

- **Transparent distributed file access and sharing**
  The Network File System (NFS) uses the IP protocol to extend the file system to support access to directories and disk on other computer systems.

- **Remote command execution**
  Using the remote shell (rsh) and remote execution (rexec) programs, users can run programs on remote computers and see the results on their own computer. This lets users of slow computers take advantage of faster computers by running their programs on the faster remote computer.

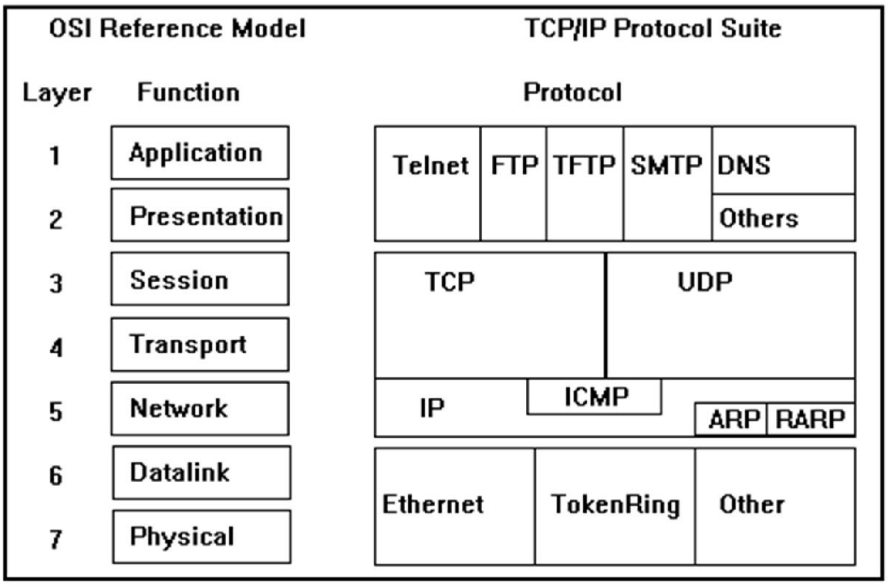
- **Remote Printing**
  The UNIX command lpr provides remote printing services.

▲ TCP/IP History

The concept of connecting dissimilar computers into a common network arose from research conducted by the Defense Advanced Research Projects Agency (DARPA). DARPA developed the TCP/IP suite of protocols, and implemented an internetwork called ARPANET, which has evolved into the INTERNET.

▲ TCP/IP and OSI

The protocols used closely resemble the OSI model. The Open Systems Interconnect model is a model of 7 layers, which deal with the exchange of data from one computer to another.



Applications developed for TCP/IP generally use several of the protocols. The sum of the layers used is known as the *protocol stack*.

User Application programs communicate with the top layer in the protocol stack. This layer passes information to the next subsequent lower layer of the stack, and son on till the information is passed to the lowest layer, the physical layer, which transfers the information to the destination network. The lower layer levels of the destination computer passes the received information to its higher levels, which in turn passes the data to the destination application. Each protocol layer performs various functions which are independent of the other layers. Each layer communicates with equivalent layers on another computer, e.g., the session layer of two different computers interact.

An application program, transferring files using TCP/IP, performs the following,

- the **application layer** passes the data to the transport layer of the source computer
- the **transport layer**
  divides the data into TCP segments
  adds a header with a sequence number to each TCP segment
  passes the TCP segments to the IP layer
- the **IP layer**
  creates a packet with a data portion containing the TCP segment
  adds a packet header containing the source and destination IP addresses
  determines the physical address of the destination computer
  passes the packet and destination physical address to the datalink layer
- the **datalink layer** transmits the IP packet in the data portion of a frame
- the **destination computers datalink layer**
  discards the datalink header and passes the IP packet to the IP layer
- the **destinations IP layer**
  checks the IP packet header and checksum
  if okay, it discards the IP header and passes the TCP segment to the TCP layer
- the **destinations TCP layer**
  computes a checksum for the TCP segment data and header
  if okay, sends acknowledge to the source computer
  discards the TCP header and passes the data to the application

---

Physical Addresses and Internet Addresses

Each networked computer is assigned a *physical address*, which takes different forms on different networks. For ETHERNET networks, the physical address is a 6 byte numeric (or 12 digit hexadecimal) value (e.g. 080BF0AFDC09). Each computers Ethernet address is unique, and corresponds to the address of the physical network card installed in the computer.

Internet addresses are *logical addresses*, and are independent of any particular hardware or network component.

The TCP/IP protocol implements a logical network numbering, stored in configuration files, which a machine identifies itself as. This logical numbering is important in sending information to other users at other networks, or accessing machines remotely. Internet addresses are logical addresses, and are independent of any particular hardware or network component. It consists of a 4 byte (32-bit) numeric value which identifies the network number and the device number on the network. The 4 byte IP address is represented in dotted decimal notation, where each byte represents a value between 0 and 255, e.g., 127.46.6.11

When a computer wants to exchange data with another computer using TCP/IP, it first translates the destination IP address into a physical address in order to send packets to other computers on the network (this is called *address resolution*).

In addition, computers in a TCP/IP network each have unique logical names like ICE.CIT.AC.NZ. These logical names are connected to their IP address, in this example, the IP address of ice.cit.ac.nz is 156.59.20.50. The logical name is also referred to as the domain name**.**

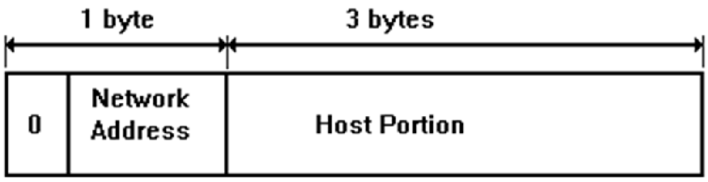
When a client computer wishes to communicate with the host computer ICE, it must translate its logical name into its IP address. It does this via a domain name lookup query, which asks a domain name server the IP address of the destination host. The domain name server has a set of static tables that it uses to find the IP address. Notably, the domain name server is a mission critical piece of hardware, and if it fails, all lookup requests cannot be answered and thus you will not be able to connect to any computer using its domain name. Once the IP address is known, an address resolution is performed to return the physical address of the computer.

The IP logical numbering is comprised of a network number and a local number. For sites connected to the Internet (a global computer network of universities, databases, companies and US defence sites), the network portion is assigned by applying to a company responsible for maintaining the Internet Domain Names.

The construction of an IP address is divided into three classes. Which class is used by an organization depends upon the maximum number of work stations that is required by that organization. Each node or computer using TCP/IP within the organization MUST HAVE a unique host part of the IP address.

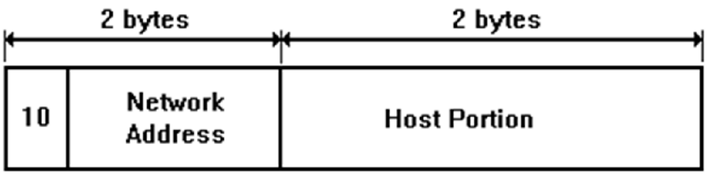
Class A Addressing

- first byte specifies the network portion
- remaining bytes specify the host portion
- the highest order bit of the network byte is always 0
- network values of 0 and 127 are reserved
- there are 126 class A networks
- there are more than 16 million host values for each class A network



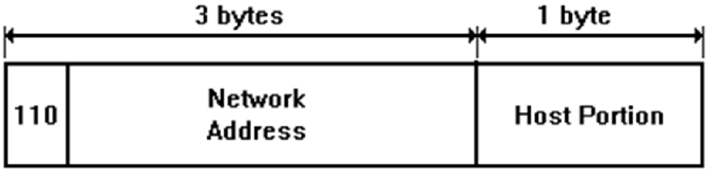
Class B Addressing

- the first two bytes specify the network portion
- the last two bytes specify the host portion
- the highest order bits 6 and 7 of the network portion are 10
- there are more than 16 thousand class B networks
- there are 65 thousand nodes in each class B network



Class C Addressing

- the first three bytes specify the network portion
- the last byte specifies the host portion
- the highest order bits 5, 6 and 7 of the network portion are 110
- there are more than 2 million class C networks
- there are 254 nodes in each class C network



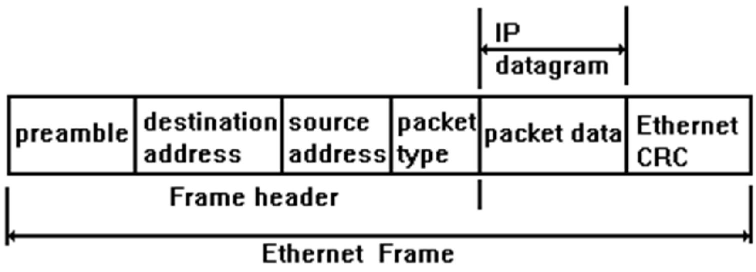
Reserved IP Addresses

The following IP addresses are reserved for special purposes, and must NOT be assigned to any host.

- Network Addresses : The host portion is set to all zero's (129.47.0.0)
- Broadcast Address : The host portion is set to all one's (129.47.255.255)
- Loopback Addresses : 127.0.0.0 and 127.0.0.1

Internet to Physical Address Translation

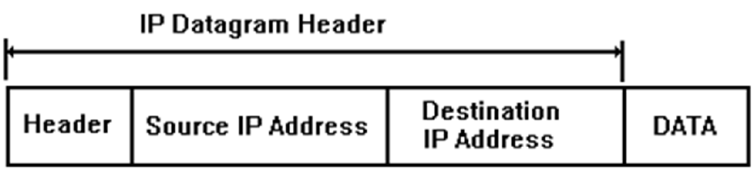
When an IP packet is sent, it is encapsulated (enclosed) within the physical frame used by the network. The IP address is mapped onto the physical address using the Address Resolution Protocol (arp) for networks such as Ethernet, token-ring, and Arcnet.



When a node wants to send an IP packet, it determines the physical address of the destination node by first broadcasting an ARP packet which contains the destination IP address. The destination node responds by sending its physical address back to the requesting node.

The Internet Protocol (IP)

This defines the format of the packets and how to handle them when sending or receiving. The form of the packets is called an *IP datagram*.

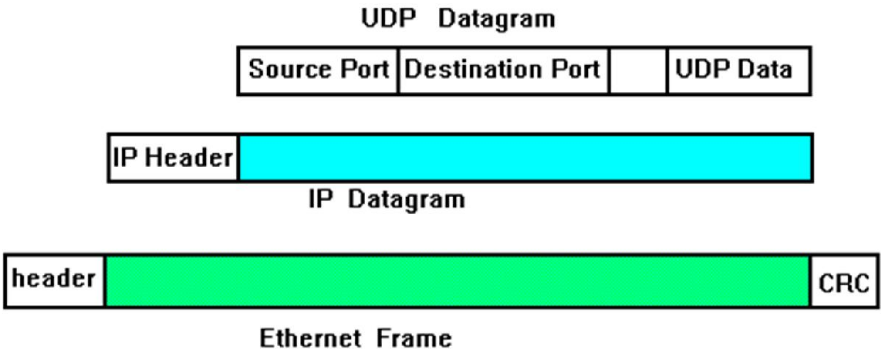


▲ The Internet Control Message Protocol (ICMP)

ICMP packets contain information about failures on the network, such as inoperative nodes and gateways, packet congestion etc. The IP software interprets ICMP messages. ICMP messages often travel across many networks to reach their destination, so they are encapsulated in the data portion of an IP datagram.

▲ The User Datagram Protocol (UDP)

This permits users to exchange individual packets over a network. It defines a set of destinations known as *protocol ports*. Ports are numbered, and TCP/IP reserves 1 to 255 for certain applications. The UDP datagram is encapsulated into one or more IP datagrams.



▲ The Transmission Control Protocol (TCP)

TCP is a reliable stream delivery protocol. It establishes a virtual circuit between the two applications, and sends a stream of bytes to the destination in exactly the same order as they left the source.

Before transmission begins, the applications at both ends obtain a TCP port, similar to that used by UDP.

TCP segments are encapsulated into an IP datagram. TCP buffers the stream by waiting for enough data to fill a large datagram before sending it.

TCP is full duplex, and assigns each segment a sequence number, which the receiving end uses to ensure all segments are received in the correct order. Upon arrival of the next segment, the receiving end sends an acknowledgement to the sending node.

If the sending node does not receive an acknowledgement within a certain time, it retransmits the segment.

Domain Name Servers

This is a hierarchical naming system for identifying hosts. Each host name is comprised of domain labels separated by periods. If your machine is connected to the Internet, you assign local domain names to host computers only, and your higher level domain name is assigned to you by the organization that controls the domain names. Domain names must be registered, so they don't conflict with an existing one.

For example, the domain name assigned to CIT is,

```
cit.ac.nz
```

An example of the host computers at CIT are called cit1, cit2, and mail. Their host names in the domain are

```
cit1.cit.ac.nz
cit2.cit.ac.nz
mail.cit.ac.nz
```

Users are also assigned names. Consider the user joe, who has an account on the host machine mail. The domain name for this user is,

```
joe@mail.cit.ac.nz
```

Hosts in your domain can be referred to by host name only. One host acts as a name resolver (host domain name server), which resolves machine names. For example, if you want to ftp into the local host ftp.cit.ac.nz, it will send a request to the name domain server, which will send back it's IP address.

The name domain server uses a special file called hosts to resolve host names and their IP addresses. This file is a static file that must be periodically updated every time changes are made.

Simple Network Management Protocol (snmp)

This provides a means for managing a network environment. Each host, router or gateway running SNMP can be interrogated for information related to the network.

Examples of information are

- host names

- packets transmitted and received
- errors
- routing information

Boot Protocol (bootp)

This service allows a local machine to get its Internet address from a designated boot server. The bootp server has a list of Ethernet addresses and IP addresses stored in a file (bootptab). When it receives a request from a machine, it looks at this file for a match and responds with the assigned IP address. The bootp server uses static tables to maintain a link between the Ethernet addresses and IP addresses for computers on the network. Obviously, this requires continual updating as network cards are changed and computers moved within the organization.

Network Services

All of the above network services like snmp and ftp are enabled on the host machine by running the system daemon process to support the service.

If the daemon process is not started, the service is not available at that host. In other words, you cannot ftp into a host which is not running the ftp daemon service.

When a UNIX host starts up, it usually runs an inetd service, which reads the file inetd.lst which contains a list of the networking services for the host to start.

Sockets

Sockets are an end to end numbered connection between two UNIX machines communicating via TCP/IP. Standard packages are assigned special socket numbers (telnet is port 23). The socket numbers for various protocols and services are found in /etc/services.

A programming socket interface provides calls for opening, reading, writing and closing a socket to another host machine. In this way, the programmer need not be concerned with the underlying protocol associated with the socket interface.

Networking Commands

Below is a discussion of some of the more common networking commands.

arp (address resolution protocol)

This command displays and modifies the internet to physical hardware address translation tables.

Examples

```
arp  -a                ; show all ARP entries on host kai
arp  -d 156.59.20.50   ; delete an ARP entry for the host ice
arp  -f                ; delete all ARP entries
```

netstat (network status)

This command displays the network status of the local host. It provides information about the TCP connections, packet statistics, memory buffers and socket information.

Examples

```
netstat  -s     ; show socket information
netstat  -r     ; show routing tables
netstat  -a     ; show addresses of network interfaces
netstat  -?     ; show help
```

ping

This command sends an echo request to a host. It is a diagnostic tool for testing whether a host can be found. When the request reaches the host, it is sent back to the originator.

Examples

```
ping  ice.cit.ac.nz     ; send an echo request to host ice.cit.ac.nz
ping  156.45.208.1      ; ping host at IP address 156.45.208.1
```

```
c:\winnt\system32> Ping ice.cit.ac.nz

Pinging ice.cit.ac.nz [156.59.20.50] with 32 bytes of data:

Reply from 156.59.20.50: bytes=32 time<10ms TTL=128
Reply from 156.59.20.50: bytes=32 time<10ms TTL=128
Reply from 156.59.20.50: bytes=32 time<10ms TTL=128
Reply from 156.59.20.50: bytes=32 time<10ms TTL=128
```

route

This command manually manipulates the network routing tables which are used to connect to other hosts.

Examples

```
route  add  net  129.34.10.0   129.34.20.1  1
; add a new network 129.34.10.0 accessible via the gateway 129.34.20.1 and
; there is one metric hop to this destination
```

▲ tracert

This command lists all the connections (links) between the current computer and the destination computer.

```
c:\winnt\system32> tracert www.vuw.ac.nz
Tracing route to totara.its.vuw.ac.nz [130.195.2.249]
over a maximum of 30 hops:

1 <10 ms 10 ms <10 ms 156.59.20.1
2 <10 ms 10 ms <10 ms portal.cit.ac.nz [156.59.220.2]
3 430 ms 90 ms 411 ms 203.97.0.149
4 81 ms 40 ms 60 ms 203.97.0.74
5 60 ms 60 ms 60 ms feba-aotearoa.waikato.netlink.net.nz [140.200.128.52]
6 * 1783 ms 1442 ms wngw1f01-hn-1m.netlink.net.nz [203.97.191.5]
7 90 ms 90 ms 80 ms vuw.netlink.net.nz [203.97.191.226]
8 100 ms 140 ms 231 ms totara.its.vuw.ac.nz [130.195.2.249]

Trace complete.
```

▲ Networking Files

The following files are associated with networking and are generally found in the /etc subdirectory on UNIX systems.

/etc/gateways
Used by routed and identifies the accessible gateway machines.

/etc/hosts
Used by clients and servers to resolve host names if a name server is unavailable.

/etc/netrc
Used by ftp and rexec as an alternative source for a username and password.

/etc/resolv
Used by clients and servers to provide a domain name and name server address.

/etc/trusers
Used by ftpd to verify users and their passwords.

▲ Summary

TCP/IP is the protocol used by computers on the Internet. Each computer has an IP address, which is a set of four digits joined using dots, and a logical name, which identifies it.

Many applications can be built on top of TCP/IP, such as file transfer (FTP) and Web services (WWW). In an organization which is connected to the Internet using

TCP/IP, a domain name server resolves logical names of host computers to IP addresses.

DATA COMMUNICATIONS

Part 18: Open Systems Interconnect [OSI] Model

OSI Model | Sending Data | Summary

Introduction
This section briefly discusses OSI, which is a 7 layer model for the exchange of data between computers.

OSI Model

In 1983, the International Standards Organization (ISO) developed a model that would allow the sending and receiving of data between two computers. It works on a layer approach, where each layer is responsible for performing certain functions.

When we think of how to send data from one computer to another, there are many different things involved. There are network adapters, voltages and signals on the cable, how the data is packaged, error control in case something goes wrong, and many other concerns. By dividing these into separate layers, it makes the task of writing software to perform this much easier.

In the Open Systems Interconnect model, which allows dissimilar computers to transfer data between themselves, there are SEVEN distinct layers.

7. **Application Layer**
   Provides Applications with access to network services.

6. **Presentation Layer**
   Determines the format used to exchange data among networked computers.

5. **Session Layer**
   Allows two applications to establish, use and disconnect a connection between them called a *session*. Provides for name recognition and additional functions like security which are needed to allow applications to communicate over the network.

4. **Transport Layer**
   Ensures that data is delivered error free, in sequence and with no loss, duplications or corruption. This layer also repackages data by assembling

long messages into lots of smaller messages for sending, and repackaging the smaller messages into the original larger message at the receiving end.

3. **Network Layer**
This is responsible for addressing messages and data so they are sent to the correct destination, and for translating logical addresses and names (like a machine name FLAME) into physical addresses. This layer is also responsible for finding a path through the network to the destination computer.

2. **Data-Link Layer**
This layer takes the data frames or messages from the Network Layer and provides for their actual transmission. At the receiving computer, this layer receives the incoming data and sends it to the network layer for handling.

The Data-Link Layer also provides error-free delivery of data between the two computers by using the physical layer. It does this by packaging the data from the Network Layer into a **frame** that includes error detection information. At the receiving computer, the Data-Link Layer reads the incoming frame, and generates its own error detection information based on the received frame data. After receiving all of the frame, it then compares its error detection value with that of the incoming frames, and if they match, the frame has been received correctly.

A frame looks like,

Destination ID	Sender ID	Control	Data	CRC

The Data-Link Layer actually consists of two separate parts, the **Medium Access Control** (MAC) and **Logical Link Control Layer** (LLC). Example MAC layers are Ethernet 802.3 and Token Ring 802.5

Bridges are an example of devices which works at the MAC layer.

1. **Physical Layer**
Controls the transmission of the actual data onto the network cable. It defines the electrical signals, line states and encoding of the data and the connector types used. An example is 10BaseT. Repeaters are an example of devices that work at the Physical Layer.

For Ethernet 802.3, the Physical Layer can be represented as

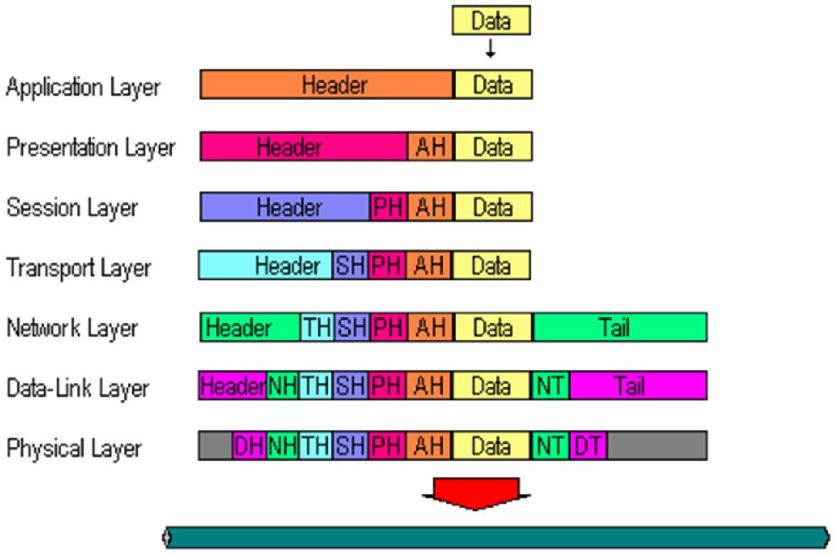
- 10Base5
- 10Base2

- 10BaseT
- 10BaseF

▲ Sending Data Via the OSI Model

Each layer acts as though it is communicating with its corresponding layer on the other end.

In reality, data is passed from one layer down to the next lower layer at the sending computer, till its finally transmitted onto the network cable by the Physical Layer. As the data it passed down to a lower layer, it is encapsulated into a larger unit (in effect, each layer adds its own layer information to that which it receives from a higher layer). At the receiving end, the message is passed upwards to the desired layer, and as it passes upwards through each layer, the encapsulation information is stripped off .



▲ Summary

The OSI model defines a model consisting of 7 layers. This allows the exchange of data between different computers using the OSI model.

DATA COMMUNICATIONS

Part 19: WIDE AREA NETWORKING - 1

Analog Modems | Dedicated or Leased Lines | Packet Switching
Integrated Services Digital Network | Frame Relay
Asynchronous Transfer Mode | Digital Subscriber Line

The following is a brief discussion on the merits of the technologies used to implement Wide Area Networks.

Analog Modems

Analog modems use the existing telephone infrastructure to link sites together. The telephone cabling supports analogue frequencies in the range 300Hz to 3400KHz, and is primarily designed for speech. The available bandwidth of the speech circuits provided by telecommunication companies imposes limits on the available speed in bits per second that can be transmitted.

The modems implement a dial-up connection. A connection is made between the two modems by dialing the number assigned to the other modem, using the existing dial up telephone network. Generally, connections are established for limited duration's. This suits remote access users who might want to dial into their network after hours, or small offices which dial into their internet service provider at regular intervals during the day to exchange (upload and download) email.

Current Modem standards are

Standard	**Speed in bps**
V.21	300
V.22	1200
V.22bis	2400
V.32	9600
V.32bis	14400
V.FC	19200
V.34	28800
V.34+	33600

The speeds stated above are maximum speeds, and often, modems fail to achieve this. Errors caused by noise on the telecommunication lines often cause modems to fall back to a much lower speed, in order to reduce the number of errors. Thus a high speed modem rated at 33600bps often achieves a throughput of 9600bps due to the existing phone lines being too error prone to support the higher rate.

Another problem that occurs is with modems that utilize compression techniques. Often, compression is measured on the transmission of uncompressed files like text files. When these same compression modems are asked to deal with the transfer of compressed files like .ZIP files, they do not perform well, and effectively either transfer at a much reduced rate or no compression at all. Some typical compression type modems are MNP4 and MNP5. In addition, modems utilizing the different compression schemes often fail to communicate properly with compression enabled. This is due to variances in manufacturers implementations of compression algorithms.

Advantages	**Disadvantages**	**Common Usage**
Widely available	Low speed	Remote access
Low Cost	Error Prone	Low bandwidth requirements like email
Most interoperate reliably	Technology changing rapidly	Roving users
Portable		

Dedicated Lines (Leased Line)

Dedicated lines are fixed connections which do not involve dialing. They are permanent end to end connections. The telecommunications company provides a dedicated high speed connection between the two desired locations, at speeds ranging from as low as 9600bps to as high as 45Mbps. The higher the speed, the greater the cost, which is usually a fixed monthly rental charge (does not include data charges, only rental charges).

The connection is available 24 hours a day, seven days a week, and is thus suited to companies who want permanent connections between their office branches, or perhaps to a company who wants a permanent connection to the Internet (they are providing a WWW server for people to access).

The basic unit of measurement for dedicated lines is a **T1** connection, which supports 1.544Mbps. A T3 connection supports 45Mbps. Fractional T1 circuits are available in units of 64Kbps, with connections of 384Kbps, 512Kbps and 768Kbps being common.

The connection is implemented with two units

- Channel Service Unit (CSU)
  This provides the interface to the dedicated line
- Data Service Unit (DSU)
  This interfaces between the CSU and the customers equipment, using RS232 for low speeds up to 56Kbps, and V.35 (RS-422/499) for higher speeds

It is common to have the units as a single component. The CSU/DSU is normally the demarcation zone which defines where the customers responsibility ends the the telecommunications company begins. Most telecommunication companies provide the ability to perform real-time monitoring of the connection via the CSU/DSU.

Advantages	**Disadvantages**	**Common Usage**
Private and secure	Locked into Tele-communications pricing regime	Connecting large sites
Cost effective for regular transfer of large amounts of data	High monthly rental	Establishment of a permanent internet presence
Fixed costs easier to budget for than if you pay for data transferred		

Packet Switching (X.25)

Packet switching has been around for some time now. It is an established technology which sends data across a packet switched network in small parcels called packets. If the data packets travel the same path to the destination, this is called *virtual circuit*, if packets can travel any path, not necessarily the same as each other, this is called *datagram*.

Packet switched connections are normally in the speed of 19.2Kbps to 64Kbps, though some higher speed connections may be available in certain countries. It is a dial-up switched connection, in that the user pays connection charges, traffic charges and time charges. As such, its not suitable for permanent connections.

X.25 was designed to be implemented over noisy analogue phone lines, thus has a lot of built in error control. With today's relatively low error links, this can result in an unnecessary overhead.

An X.25 connection supports a number of *virtual circuits* which are each numbered. These represent a time division of the available bandwidth of the connection. This division into virtual circuits allows each VC to support a single device. X.25 uses the lower 3 levels of the OSI model. The virtual circuit is a full duplex connection which is established for the duration of the call.

Devices which do not have built in packet switched support can be interfaced to a packet switched network using a *Packet Assembly/Disassembly (PAD)* unit. This allows existing computers or terminals to be connected.

Integrated Services Digital Network (ISDN)

ISDN was developed in order to provide the user with a single interface which supported a range of different devices simultaneously. The basic ISDN connection is a 2B + D connection, that is, 2 B channels each of 64Kbps, and a single D channel of 16Kbps. The B channels are designed to carry user data, whilst the D channel is meant to carry control and signaling information. This format is known as the ***Basic Rate Interface*** (BRI), which also provides for frame control and other heads, which gives an overall capacity of 192Kbps per BRI ISDN connection.

Higher capacity circuits are available. ISDN uses the existing telecommunications dial-up infrastructure, though special ISDN connection interface boxes are required at the users premises. Each B channel can be used separately or combined with other B channels to achieve higher speeds.

The ***Primary Rate Interface*** (PRI) offers 23B channels and one D channel at 64Kbps (North America and Japan) giving a total of 1.544Mbps. The PRI for Europe, Australia and some other parts of the world is 30B channels and one D channel at 64Kbps giving a total of 2.048Mbps.

Advantages	**Disadvantages**	**Common Usage**
Low fixed cost	Not available in all centers or countries	Periodic Internet Access (for email etc)
Scalable (B circuits can be combined for greater speeds)	Not suited to mobile users (users dialing in via remote access)	LAN-LAN remote connections which are not permanent
Fast call set up times		

DATA COMMUNICATIONS

Part 20: WIDE AREA NETWORKING - 2

Analog Modems | Dedicated or Leased Lines | Packet Switching
Integrated Services Digital Network | Frame Relay
Asynchronous Transfer Mode | Digital Subscriber Line

Frame Relay

Sometimes referred to as Fast packet, it is designed for modern networks which do not need lots of error recovery (unlike packet switching). Typical Frame relay connections range from 56Kbps to 2Mbps. Frame relay is similar to packet switching X.25, but is more streamlined giving higher performance and greater efficiency.

Frame relay, like X.25, implements multiple virtual circuits over a single connection, but does so using statistical multiplexing techniques which yields a much more flexible and efficient use of the available bandwidth. FR includes a *cyclic redundancy check* (CRC) for detecting corrupted data, but does not include any mechanism for corrected corrupted data.

In addition, because many higher level protocols include their own flow control algorithms, FR implements a simple congestion notification mechanism to notify the user when the network is nearing saturation.

Frame Format

The format of FR frames is shown in the diagram below. Flags define a frames start and end. The address field is 16 bytes long, 10 of which comprise the actual circuit ID (*Data Link Connection Identifier*). The DLCI identifies the logical connection that is multiplexed into the physical channel. Three bits of the address field are allocated to congestion control.

FR also supports multi-casting, the ability to send to more than one destination simultaneously. Four reserved DLCI values (1019 to 1022) are designated as multicast groups.

Advantages	Disadvantages	Common Usage
Low incremental cost per connection (PVC)	Relatively high initial cost	Interconnecting lots of remote LAN's together
Exploits recent advances in network technology		
Supports multicasting		

Additional References
Frame Relay Resources
Basics of Frame Relay

Asynchronous Transfer Mode (ATM)

ATM breaks data into small chunks of fixed size cells (48 bytes of data plus a 5 byte overhead). ATM is designed for handling large amounts of data across long distances using a high speed backbone approach. Rather than allocating a dedicated virtual circuit for the duration of each call, data is assembled into small packets and statistically multiplexed according to their traffic characteristics.

One problem with other protocols which implement virtual connections is that some time slots are wasted if no data is being transmitted. ATM avoids this by dynamically allocating bandwidth for traffic on demand. This means greater utilization of bandwidth and better capacity to handle heavy load situations.

When an ATM connection is requested, details concerning the connection are specified which allow decisions to be made concerning the route and handling of the data to be made. Typical details are the type of traffic [video requires higher priority], destination, peak and average bandwidth requirements [which the network can use to estimate resources and cost structures], a cost factor [which allows the network to chose a route which fits within the cost structure] and other parameters.

UNDER SONSTRUCTION

155Mbps

622Mbps

Additional References
[ATM Frequently Asked Questions (FAQ)]
[A Very Cool Java-based ATM Signalling Simulator (requires Java-capable browser)]
[IETF IP over ATM Working Group]

Digital Subscriber Line (xDSL)

xDSL is a high speed solution that allows megabit bandwidth from tele-communications to customers over existing copper cable, namely, the installed telephone pair to the customers premises (called the *local loop*). With the high penetration and existing infrastructure of copper cable to virtually everyone's home (for providing a voice telephone connection), xDSL offers significant increases in connection speed and data transfers for access to information.

In many cases, the cost of relaying fiber optic cable to subscriber premises is prohibitive. As access to the Internet and associated applications like multi-media, tele-conferencing and on demand video become pervasive, the speed of the local loop (from the subscriber to the telephone company) is now a limiting factor. Current technology during the 1980's and most of the 1990's has relied on the use of the analog modem with connection rates up to 56Kbps, which is too slow for most applications except simple email.

xDSL is a number of different technologies that provide megabit speeds over the local loop, without the use of amplifiers or repeaters. This technology works over**non-loaded** local loops (loaded coils were added by telephone companies on some copper cable pairs to improve voice quality). xDSL coexists with existing voice over the same cable pair, the subscriber is still able to use their telephone, at the same time. This technology is referred to seamless.

To implement xDSL, a terminating device is required at each end of the cable, which accepts the digital data and converts it to analogue signals for transmission over the copper cable. In this respect, it is very similar to modem technology.

xDSL provides for both symmetric and asymmetric configurations.

Asymmetric	**Symmetric**
Bandwidth is higher in one direction	Bandwidth same in both directions
Suitable for Web Browsing	Suitable for video-conferencing

Variations of xDSL

There are currently six variations of xDSL.

xDSL Technology	Meaning	Rate
DSL	Digital Subscriber Line	2 x 64Kbps circuit switched 1 x 16Kbps packet switched (similar to ISDN-BRI)
HDSL	High-bit-rate DSL	2.048Mbps over two pairs at a distance up to 4.2Km
S-HDSL/SDSL	Single-pair or Symmetric High-bit-rate DSL	768Kbps over a single pair
ADSL	Asymmetric DSL	up to 6Mbps in one direction
RADSL	Rate Adaptive DSL	An extension of ADSL which supports a variety of data rates depending upon the quality of the local loop
VDSL	Very High-bit-rate asymmetric DSL	Up to 52Mbps in one direction and 2Mbps in the other direction.