CT2108 – Nets and Comms 1

Physical Layer

Content

- Theoretical basis for data communications
- Transmission media
  - Guided, wireless and satellite
- Examples of telecommunicating systems used in practice for WANs:
  - Fixed phone system
  - Mobile phone system
  - Cable television

The Theoretical Basis for Data Communication

- Fourier Analysis
- Bandwidth-Limited Signals
- Maximum Data Rate of a Channel
- Channel Organization
- Types of data communication

Fourier Analysis (1)

- A periodic signal with period T can be constructed as a sum of a number of sine and cosines

$$g(t) = \frac{1}{2}c + \sum_{n=1}^{\infty} a_n \sin(2\pi nft) + \sum_{n=1}^{\infty} b_n \cos(2\pi nft)$$

- Where:
  - **f**=1/T
  - $\mathbf{a_n}$ and $\mathbf{b_n}$ are the sine and cosine amplitudes of the n^{th} harmonics
  - **c** -s a constant

Such a decomposition is called FOURIER SERIES. The original function of time can be reconstructed (by performing the sum) if the period T is known and the amplitudes an, bn and the constant c are given.

Fourier Analysis (2)

- The $\mathbf{a_n}$ amplitudes can be computed for any g(t) by multiplying the previous formula (Fourier series) by sin (2πkft) and integrating from 0 to T

$$a_n = \frac{2}{T}\int_0^T g(t)\sin(2\pi nft)dt$$

SINA * SINB = ½[COS(A-B) - COS(A+B)]

SINA * COSB = ½[SIN(A+B) + SIN(A-B)]

COSA * SINB = ½[SIN(A+B) – SIN(A-B)]

COSA * COSB = ½[COS(A+B) + COS(A-B)]

Ssin = -cos

Scos= sin

Fourier Analysis (3)

- The bn amplitudes can be computed by multiplying the Fourier series by cos(2πkft) and integrating from 0 to T

$$b_n = \frac{2}{T}\int_0^T g(t)\cos(2\pi nft)dt$$

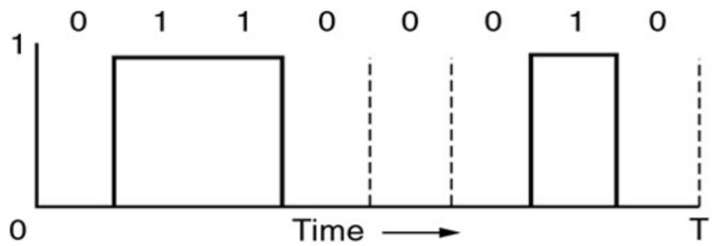
Fourier Analysis (4)

- The constant c can be computed by just integrating form 0 to T

$$c = \frac{2}{T} \int_0^T g(t)dt$$

Bandwidth limited signals (1)

- Let's consider ASCII character 'b' encoded in a byte. The bit pattern to be sent is 01100010
- The way to deal with data signal that has a fixed duration (like our example) is to imagine that the entire pattern repeats over again forever



Bandwidth limited signals (2)

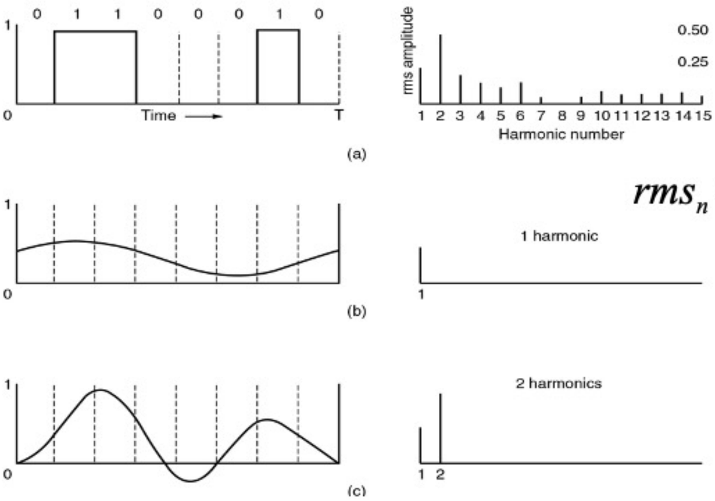
- The Fourier analysis of the signal yields to the coefficients:

$$a_n = \frac{1}{\pi n}[\cos(\frac{\pi n}{4}) - \cos(\frac{3\pi n}{4}) + \cos(\frac{6\pi n}{4}) - \cos(\frac{7\pi n}{4})]$$

$$b_n = \frac{1}{\pi n}[\sin(\frac{3\pi n}{4}) - \sin(\frac{\pi n}{4}) + \sin(\frac{7\pi n}{4}) - \sin(\frac{6\pi n}{4})]$$

$$c = \frac{3}{4}$$

Bandwidth-Limited Signals (3)



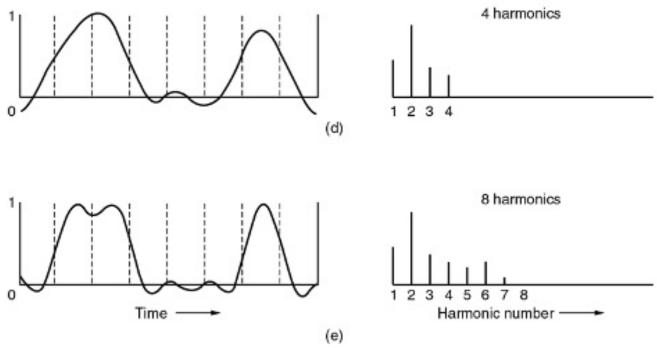
$$rms_n = \sqrt{a_n^2 + b_n^2}$$

Our signal and its root-mean-square Fourier amplitudes.
(b) – (c) Successive approximations to the original signal.

The root mean square amplitudes **(RMS)** are of interest because those values show the **ENERGY** transmitted at the corresponding frequency. No transmission facility can transmit signals without loosing some power in the process. If all Fourier components were equally diminished than the resulted signal would be reduced in amplitude, but not distorted. Unfortunately, all the transmission medium diminish different Fourier components by different amounts, resulting in a distortion of the signal at the other end. Usually, the amplitudes are transmitted undiminished between 0 and a frequency Fc (F cut, measured in Hz), with all frequencies above this Fc attenuated. The range of frequencies transmitted without being attenuated it is called **BANDWIDTH.** In practice the cut of frequency is not really sharp, so usually this bandwidth is the range from 0 to the frequency where half of the power of the signal gets through.

The bandwidth is a physical property of the transmission medium and usually depends on the construction, thickness and length of the medium. Consider our signal (ASCII 'B', (a)) and look how the signal would look if only the lowest frequencies where transmitted (i.e. if the function was approximated by only a few terms).

Bandwidth-Limited Signals (4)



(d) – (e) Successive approximations to the original signal.

Bandwidth-Limited Signals (5)

Bps	T (msec)	First harmonic (Hz)	# Harmonics sent
300	26.67	37.5	80
600	13.33	75	40
1200	6.67	150	20
2400	3.33	300	10
4800	1.67	600	5
9600	0.83	1200	2
19200	0.42	2400	1
38400	0.21	4800	0

Given a rate of b bits per second. The time to send one byte is 8/b, which is the period for the first harmonic. The frequency of the first harmonic would be b/8 Hz.

A (non-DSL) phone line has an artificially introduced cut off frequency of 3000Hz. That means that the number of the highest harmonics passed through is about 3000/b/8 = 24000/b. It is clear that on a (non-DSL) phone line signals can't be sent beyond 9600 bps, since the reconstruction of the signal at the other end would be extremely difficult. It is obvious that higher data rates, for binary signals, it is impossible to receive and reconstruct the original signal without somehow increasing the available bandwidth. DSL enabled phone lines now provide much wider wider bandwidth than the original 3KHz that was available, so much higher data rates are now possible. Sophisticated coding and modulation do exist and can achieve higher data rates.

Maximum rate of a channel (1)

- Noiseless channel – Nyquist theorem:

$$MaxDataRate = 2B\log_2 V \qquad [bits/\sec]$$

- Where:
  - B – noiseless channel bandwidth
  - V – number of discrete levels of the signal transmitted through the channel

Maximum rate of a channel (2)

- If random noise is present, then the situation deteriorates rapidly.
- In reality, there is always random noise, due to the motion of the molecules in the system
- The amount of the thermal noise is measured by the ratio of the signal power to the noise power, called **signal-to-noise ratio**

$$SNR_{dB} = 10\log_{10}(\frac{S}{N}) \qquad [db]$$

- Signal to noise ratio is given in decibels (dB) and the ratio itself is not usually quoted.
- A ratio of 10 is 10dB, a ratio of 100 is 20dB, a ratio of 1000 is 30dB and so on

Maximum rate of a channel (3)

- Shannon theory:

$$MaxDataRate = B\log_2(1+\frac{S}{N}) \qquad [bits/\sec]$$

- Where:
  - B – bandwidth of the noisy channel
  - S/N – signal to noise ratio of the channel
- Shannon's theory demonstrates that the maximum data rate through a channel is limited by the amount of noise present, no matter how many or few signal levels are used and no matter how often or infrequently samples are taken
- This limit is the upper limit and real systems rarely achieve it

Example

- A channel of 3000Hz bandwidth (phone line) has a SNR ratio of 30dB. The transmitted signal has 8 levels. What is the maximum data rate that this channel can accommodate?
- What if the SNR of the channel would be only 10dB?

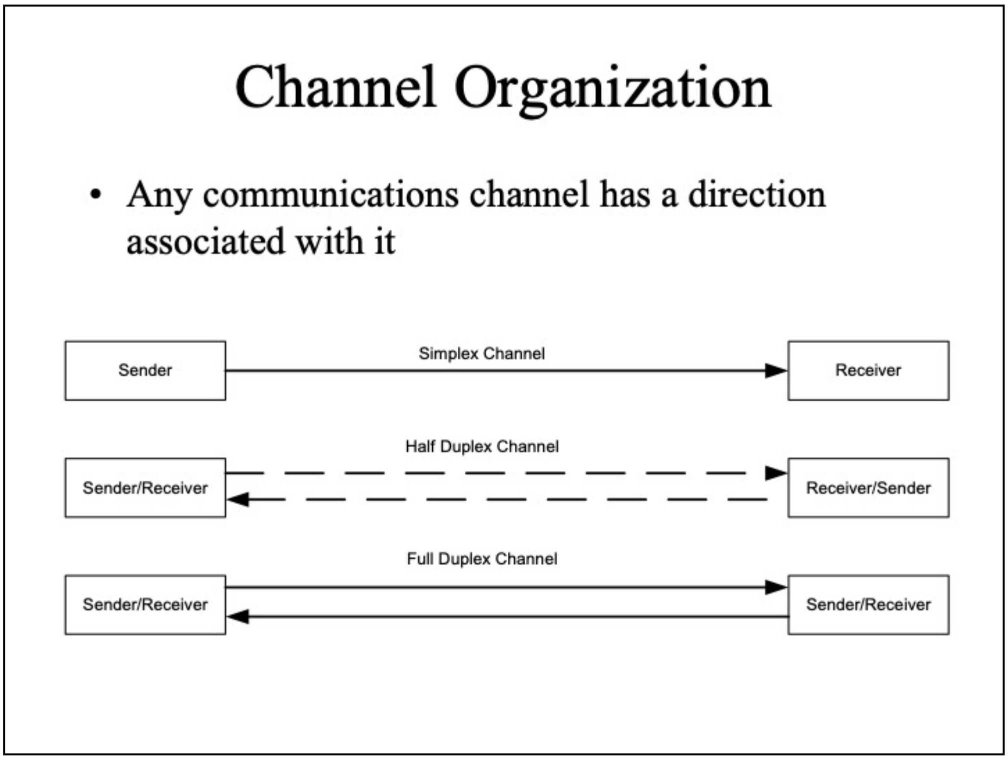
Compute both Nyquist and Shannon:

Nyquist – max = 18 kbps

Shannon – max = 30kbps

Take the lowest, in our case is the Nyquist limit. So the maximum throughput through the channel would be 18kb/s

If the channel would have 10dB, than the Shannon theory gives aabout max = 10kbps, therefore the maximum throughput would be 10kbps. Always take the smallest between the two (Nyquist and Shannon)



The message source is the transmitter, and the destination is the receiver. A channel whose direction of transmission is unchanging is referred to as a simplex channel. For example, a radio station is a simplex channel because it always transmits the signal to its listeners and never allows them to transmit back.

A half-duplex channel is a single physical channel in which the direction may be reversed. Messages may flow in two directions, but never at the same time, in a half-duplex system. In a telephone call, one party speaks while the other listens. After a pause, the other party speaks and the first party listens. Speaking simultaneously results in garbled sound that cannot be understood.

A full-duplex channel allows simultaneous message exchange in both directions. It really consists of two simplex channels, a forward channel and a reverse channel, linking the same points. The transmission rate of the reverse channel may be slower if it is used only for flow control of the forward channel

Channel Organization

- Synchronization
  - the receiver should know the exact moment when data is valid
- Synchronous channels
  - Data and timing information are sent separately (through separate channels or same channel)
  - The timing channel transmits clock pulses to the receiver
  - Upon receive of clock pulse, the receiver reads the data and latches it
  - The data is not read again until next clock pulse arrives
- Asynchronous channels
  - No separate timing information is used
  - Transmitter and receiver must agree in advance on timings
  - Start and stop conditions are used
  - Accurate oscillators will measure the bit widths.

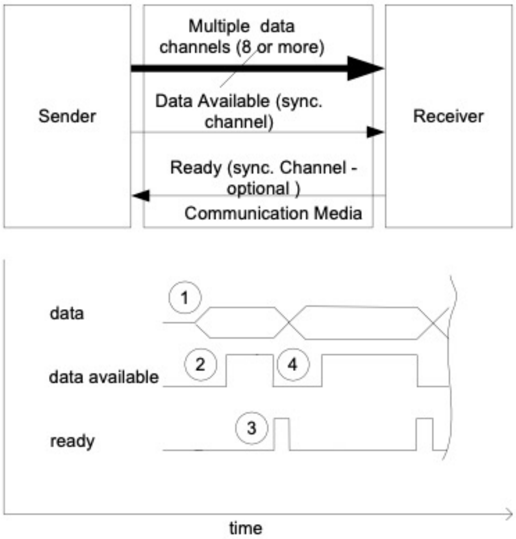
Data is not generally sent at a uniform rate through a channel. Instead, there is usually a burst of data followed by a pause, after which the data flow resumes. Packets of binary data are sent in this manner, possibly with variable-length pauses between packets, until the message has been fully transmitted. In order for the receiving end to know the proper moment to read individual binary bits from the channel, it must know exactly when a packet begins and how much time elapses between bits. When this timing information is known, the receiver is said to be synchronized with the transmitter, and accurate data transfer becomes possible. Failure to remain synchronized throughout a transmission will cause data to be corrupted or lost.

Two basic techniques are employed to ensure correct synchronization: synchronous and asynchronous communication

In synchronous systems, separate channels are used to transmit data and timing information. The timing channel transmits clock pulses to the receiver. Upon receipt of a clock pulse, the receiver reads the data channel and latches the bit value found on the channel at that moment. The data channel is not read again until the next clock pulse arrives. Because the transmitter originates both the data and the timing pulses, the receiver will read the data channel only when told to do so by the transmitter (via the clock pulse), and synchronization is guaranteed. Techniques exist to merge the timing signal with the data so that only a single channel is required. This is especially useful when synchronous transmissions are to be sent through a modem. Two methods in which a data signal is self-timed are non-return-to-zero and biphase Manchester coding. These both refer to methods for encoding a data stream into an electrical waveform for transmission.

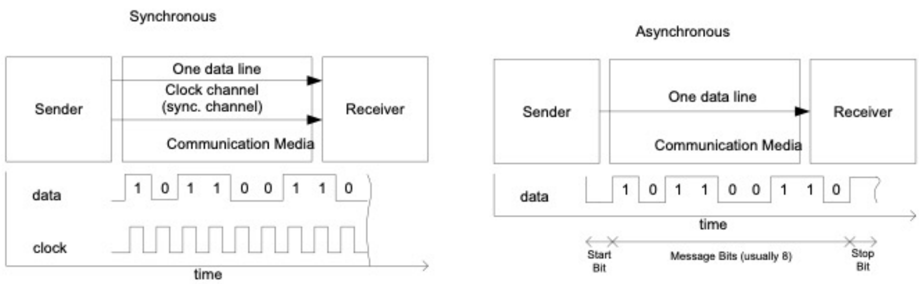
In asynchronous systems, a separate timing channel is not used. The transmitter and receiver must be preset in advance on timings. A very accurate local oscillator within the receiver will then generate an internal clock signal that is equal to the transmitter's within a fraction of a percent. For the most common serial protocol, data is sent in small packets of 10 or 11 bits, eight of which constitute message information. When the channel is idle, the signal voltage corresponds to a continuous logic '1'. A data packet always begins with a logic '0' (the start bit) to signal the receiver that a transmission is starting. The start bit triggers an internal timer in the receiver that generates the needed clock pulses. Following the start bit, eight bits of message data are sent bit by bit at the agreed upon baud rate. The packet is concluded with a parity bit and stop bit

Channel Organization – Parallel Communication



- 1 – Sender places data on the channel
- 2 – Sender asserts "data available"
- 3 – receiver reads the data and asserts "ready" signal
- 4 – Sender de-asserts "data available" signal and it will be ready for a new data transfer

Channel Organization – Serial Communication

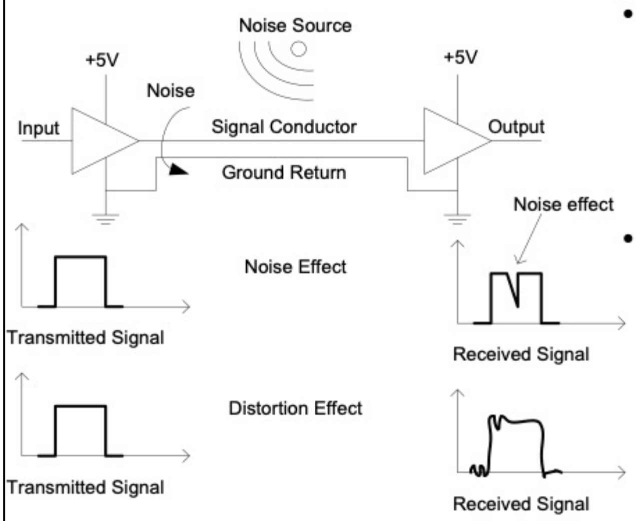


- Synchronous
  - Data and sync info are sent separately
  - Examples: I2C, USB, SPI
- Asynchronous
  - Packet length is short (to minimize local oscillators drift)
  - Examples: RS232

In synchronous systems, separate channels are used to transmit data and timing information. The timing channel transmits clock pulses to the receiver. Upon receipt of a clock pulse, the receiver reads the data channel and latches the bit value found on the channel at that moment. The data channel is not read again until the next clock pulse arrives. Because the transmitter originates both the data and the timing pulses, the receiver will read the data channel only when told to do so by the transmitter (via the clock pulse), and synchronization is guaranteed. Techniques exist to merge the timing signal with the data so that only a single channel is required. This is especially useful when synchronous transmissions are to be sent through a modem. Two methods in which a data signal is self-timed are nonreturn-to-zero and biphase Manchester coding. These both refer to methods for encoding a data stream into an electrical waveform for transmission.

Asynchronous serial transmission: for the most common serial protocol, data is sent in small packets of 10 or 11 bits, eight of which constitute message information. When the channel is idle, the signal voltage corresponds to a continuous logic '1'. A data packet always begins with a logic '0' (the start bit) to signal the receiver that a transmission is starting. The start bit triggers an internal timer in the receiver that generates the needed clock pulses. Following the start bit, eight bits of message data are sent bit by bit at the agreed upon baud rate. The packet is concluded with a parity bit and stop bit. The packet length is short in asynchronous systems to minimize the risk that the local oscillators in the receiver and transmitter will drift apart. When high-quality crystal oscillators are used, synchronization can be guaranteed over an 11-bit period. Every time a new packet is sent, the start bit resets the synchronization, so the pause between packets can be arbitrarily long.

Noise and Distortion



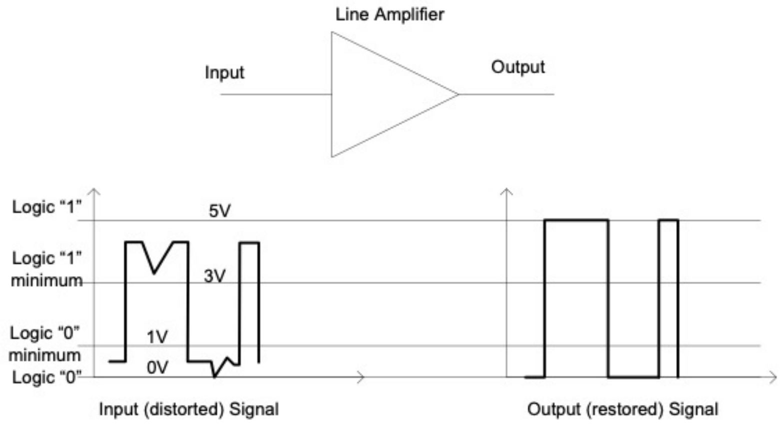
- Long conductors act like receiving antennas for electrical noise radiated by motors, switches and other electrical circuits
- Electrical signals get distorted when passing through metallic conductors due to limitations of bandwidth and internal thermal noise of the transmission media

Because of the very high switching rate and relatively low signal strength found on data, address, and other buses within a computer, direct extension of the buses beyond the boudaries of the main circuit board or plug-in boards would pose serious problems.

First, long runs of electrical conductors, either on printed circuit boards or through cables, act like receiving antennas for electrical noise radiated by motors, switches, and electronic circuits.

A second problem involves the distortion of electrical signals as they pass through metallic conductors. Signals that start at the source as clean, rectangular pulses may be received as rounded pulses with ringing at the rising and falling edges.

Noise Compensation

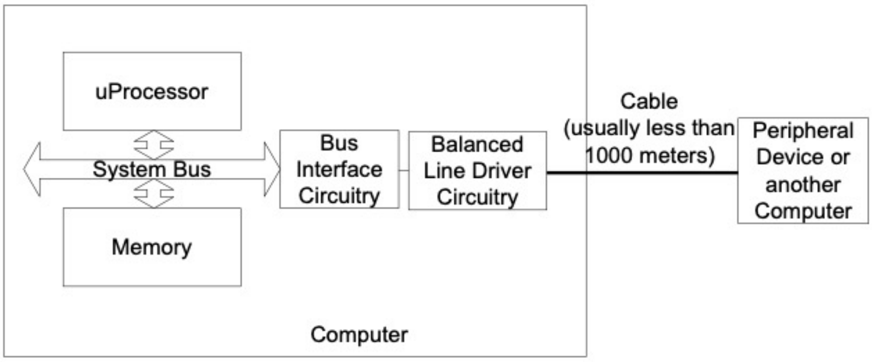


- For digital signals transmitted over short and medium distances

These effects are properties of transmission through metallic conductors, and become more pronounced as the conductor length increases. To compensate for distortion, signal power must be increased or the transmission rate decreased.

Special amplifier circuits are designed for transmitting direct (un-modulated) digital signals through cables. For the relatively short distances between components on a printed circuit board or along a computer backplane, the amplifiers are in simple IC chips that operate from standard +5v power. The normal output voltage from the amplifier for logic '1' is slightly higher than the minimum needed to pass the logic '1' threshold. Correspondingly for logic '0', it is slightly lower. The difference between the actual output voltage and the threshold value is referred to as the noise margin, and represents the amount of noise voltage that can be added to the signal without creating an error.

Data Transfer over Long Distances



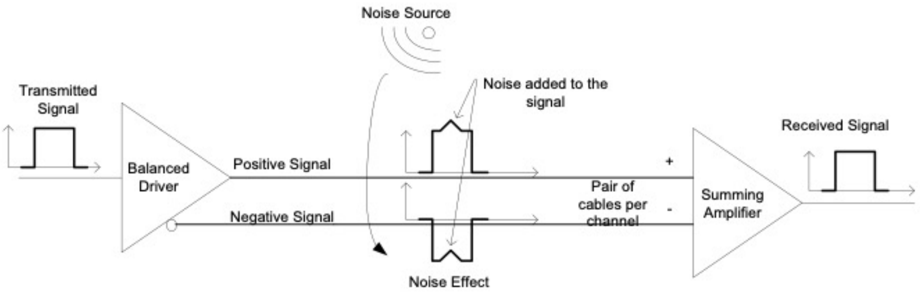
- Bit serial transmission is almost always used when distance exceeds 10 meters
- Balanced circuitry is used to send the signals

When relatively long distances are involved in reaching a peripheral device, driver circuits must be inserted after the bus interface unit to compensate for the electrical effects of long cables (noise and distortion).

This is the only change needed if a single peripheral is used. However, if many peripherals are connected, or if other computer stations are to be linked, a local area network (LAN) is required, and it becomes necessary to drastically change both the electrical drivers and the protocol to send messages through the cable. Because multi-conductor cable is expensive, bit-serial transmission is almost always used when the distance exceeds 20 feet.

In either a simple extension cable or a LAN, a balanced electrical system is used for transmitting digital data through the channel. This type of system involves at least two wires per channel, neither of which is a ground. Note that a common ground return cannot be shared by multiple channels in the same cable as would be possible in an unbalanced system.

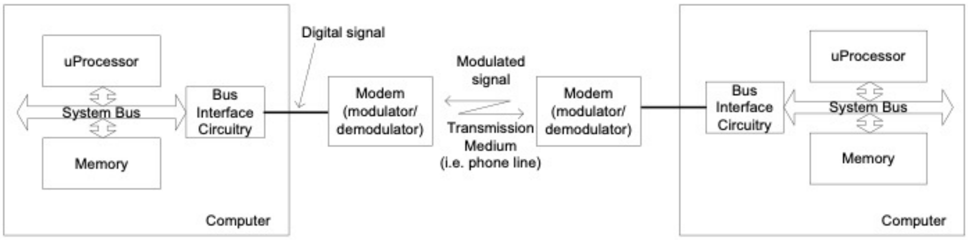
Balanced Line Transmission



- The signal is sent over two wires simultaneously
  - One wire is carrying the positive signal
  - One wire is carrying the negative (inverted) signal
- At the receiver, the signals are subtracted by a summing amplifier producing a stronger signal than any of the signals swinging on the individual lines
- When the signals are subtracted, any induced noise cancels

The basic idea behind a balanced circuit is that a digital signal is sent on two wires simultaneously, one wire expressing a positive voltage image of the signal and the other a negative voltage image. When both wires reach the destination, the signals are subtracted by a summing amplifier, producing a signal swing of twice the value found on either incoming line. If the cable is exposed to radiated electrical noise, a small voltage of the same polarity is added to both wires in the cable. When the signals are subtracted by the summing amplifier, the noise cancels and the signal emerges from the cable without noise.

Data Transfers over Long Distances



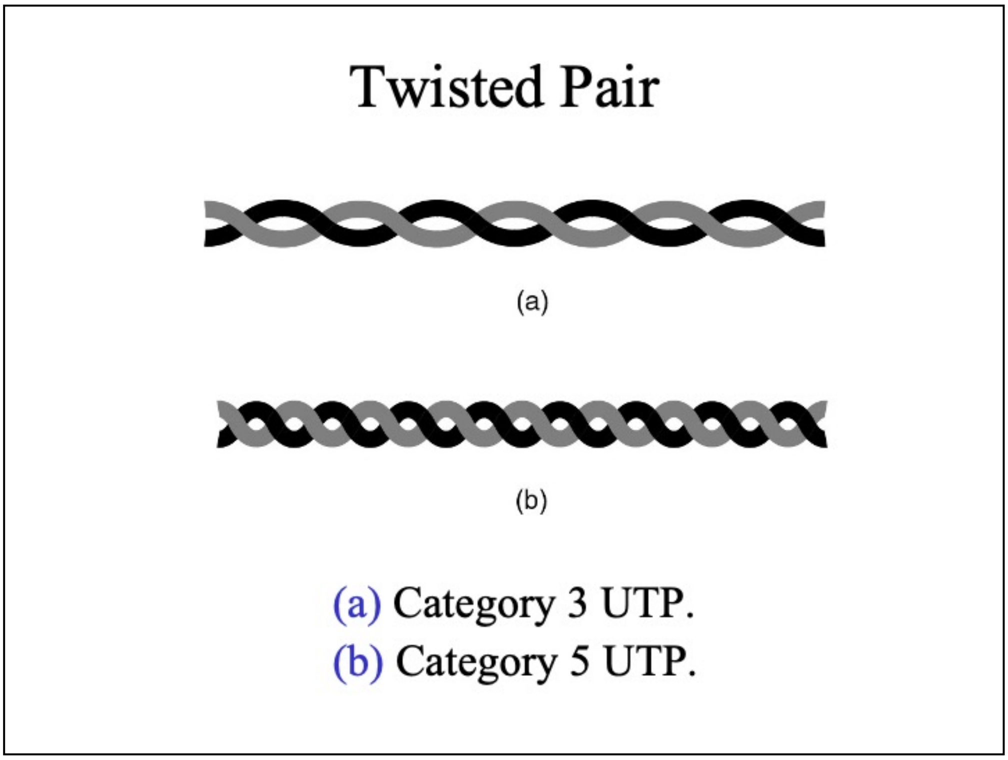
- Special devices are used, call modems, **MO**dulator/**DEM**odulator, connected to the computer system through techniques covered in communication over short distances
- Modulation – analogue signals are used as carriers to carry digital signals
  - Amplitude, frequency, phase and other types of modulation are possible

A good example of data communications over longer distance using copper wire are the use of the telephone network for your home internet connection. Transmissions over such distances are not generally accomplished with a direct-wire digital link, unless you have fibre to the home, but rather with digitally-modulated analog carrier signals, as these are easier to transmit intact over a bandwidth limited link to the ISP. This technique makes it possible to use existing phone lines for digital data, although at possibly reduced data rates compared to a direct digital link. Transmission of data from your home over phone lines requires that data signals be converted to modulated carrier waves by a modem. One or more sine wave carriers are used, and, depending on the baud rate and protocol, the modem will encode data by varying the frequency, phase, or amplitude of the carrier. The receiver's modem accepts the modulated sine wave and extracts the digital data from it. Several modulation techniques are typically used in encoding digital data for transmission, some of these will be looked at latter in this presentation.

Guided Transmission Data

- Twisted Pair
- Coaxial Cable
- Fiber Optics

Magnetic media is another (not related to networking) media used to transfer data.



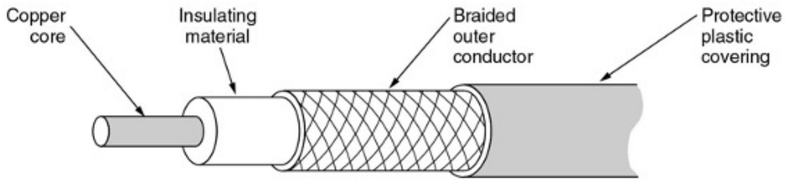
Consists of two insulated copper wires, typically about 1mm thick. The wires are twisted together in a helical form (just as DNA molecules). Twisting is done because two parallel wires constitute a fine antenna. When the wires are twisted, the waves radiated from different twists are canceling each other. The wires are usually bundled together and encased together in protective shields, when coming from a block of apartments to a phone company. If the wires wouldn't be twisted, then the interference between the wires part of same bundle would be big.

Twisted pair wires can be used for transmission of both digital and analog signals. The bandwidth depends on the thickness of the wire and the distance, but usually several Mbps can be easily achieved.

Cat 3 UTP (16 MHz) – used until 1988 to wire telephone systems. Latter replaced by Cat 5 UTP and later variants (more twists per centimeter, resulting in less cross-talk and better-quality signals over longer distances), able to handle about 100Mhz. Later categories are 6 and 7, able to handle signals of 250MHz and 600 MHz and therefore higher data rates over longer distances.

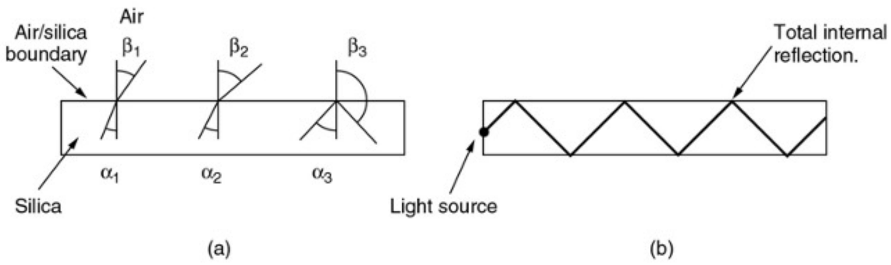
Coaxial Cable

A coaxial cable.



It is better shielded than twisted pair cable and it can span over longer distances at higher speeds. 50 ohms (for digital transmission) and 75 ohms (for analog transmission) are available. Due to the construction and the shielding process, the coax cables can have large bandwidth (up to 1GHz). They used to be largely used by phone companies for long distance lines, but now they have been replaced with fiber optics.

Fiber Optics



(a) Three examples of a light ray from inside a silica fiber impinging on the air/silica boundary at different angles.

(b) Light trapped by total internal reflection.

The optical fiber achievable bandwidth is in excess of 50 000 Gbps (50 Tbps). The current practical signaling limit is 10Gbps and it is not limited by the characteristics of the optical fiber but by our inability to convert electrical signals into optical signals any faster.

OPTICAL TRANSMISSION SYSTEM has three components: light source, the transmission medium and the detector. Conventionally a pulse of light indicates a 1 and absence of light indicates 0. Transmission medium is thin optical fiber, and the detector generates an electrical pulse when light falls on it. By attaching the source of light at one end of an optical fiber and the detector at the other end, we have a unidirectional data transmission system, that accepts electrical signal, converts it into light, transmits it over the optical fiber, it is received by the detector and transformed back into electrical signal.

The angle at which the light is injected into the optical fiber is very important. Because of refraction, the light can escape the optical fiber, or it can be "trapped" inside, with virtually no loss.

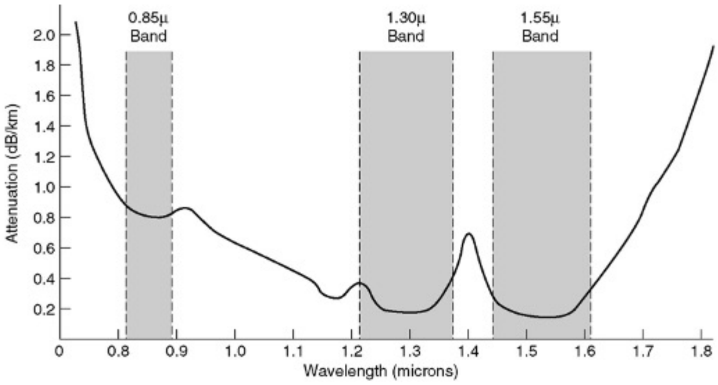
MULTIMODE FIBER – many different rays could bounce inside of the fiber, at different angles.

SINGLE MODE FIBER – the diameter of the fiber is reduced to a few wavelengths of light, which causes the light to travel (propagate) only in straight line.

Single mode fibers are more expensive and are used for transmission on very long distances. 50Gbps for over 100Km are possible, without any amplification.

Transmission of Light through Fiber

$$Attenuation_{dB} = 10\log_{10}\frac{TransmitedPower}{ReceivedPower}$$



Attenuation of light through fiber in the infrared region.

The attenuation of light through glass (the raw material used for optical fiber cable) depends on the wavelength of the light but also on the physical properties of the glass.

For example, an attenuation of 2 is given by 10log2 = 3dB

Visible light is from 0.4 to 0.7 microns (400 to 700 nm).

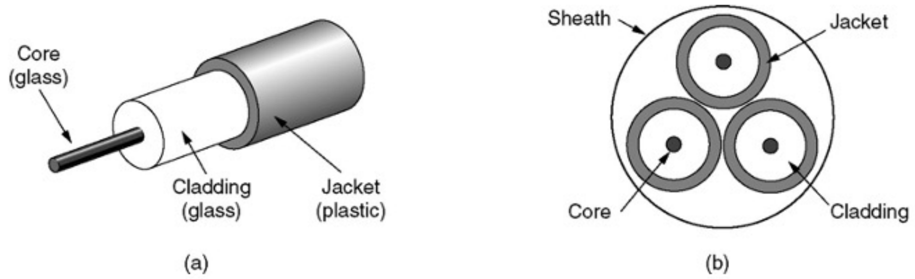
Three wavelength bands are used for communication, centered on: 0.85microns, 1.30microns and 1.55microns .

As the ray lights travel down the fiber, CHROMATIC DISPERSION is happening (the process of spreading of the wavelength). By making the pulses of a special shape, nearly all the dispersion effects can be canceled out. These pulses are called SOLITONS.

Fiber Cables (1)

(a) Side view of a single fiber.

(b) End view of a sheath with three fibers.



Multimode fibers, the core is 50 microns. For single mode fibers, the diameter of the core is 8 to maximum 10 microns.

Fibers can be connected in three ways: connectors, mechanically connected (spliced) and fused (melted) together.

Fiber Cables (2)

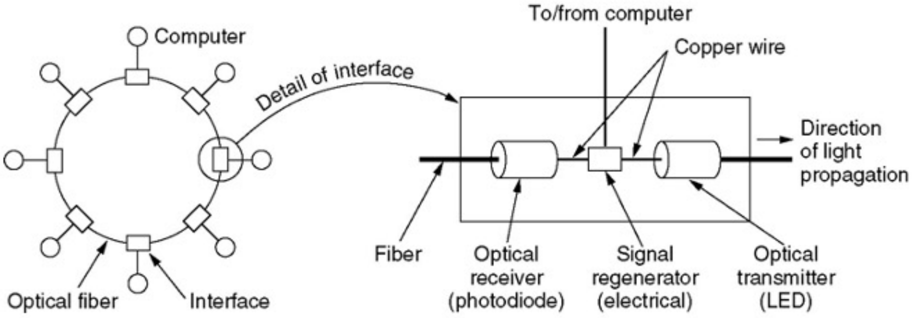
A comparison of semiconductor diodes and LEDs as light sources.

Item	LED	Semiconductor laser
Data rate	Low	High
Fiber type	Multimode	Multimode or single mode
Distance	Short	Long
Lifetime	Long life	Short life
Temperature sensitivity	Minor	Substantial
Cost	Low cost	Expensive

Two kind of light sources can be used: LED (Light Emitting Diode) and semiconductor lasers. The receiver is a photodiode, which gives and electrical pulse when stroke by light. The response time is usually 1ns, which limits the data rates at about 1Gbps.

Fiber Optic Networks

A fiber optic ring with active repeaters.



A ring network is just a collection of point to point links. The interface on each computer passes the light pulse stream through to the next link and also serves as a T junction to allow the computer to send and accept messages. The interfaces could be passive (tapping an LED and a photodiode on the fiber) or active repeater (presented in the figure). If an active repeater fails, the ring is broken and the network goes down. The passive interface is loosing signal, therefore, the light can't travel to far, so it limits the size of the network.

Wireless Transmission

- The Electromagnetic Spectrum
- Radio Transmission
- Microwave Transmission
- Infrared and Millimeter Waves
- Lightwave Transmission

The Electromagnetic Spectrum (1)

- The number of oscillations per second of a wave is called its **frequency** (f), measured in Hz
- The distance between two consecutive maxima (or minima) is called wavelength (λ), measured in meters.
- In vacuum all electromagnetic waves travel at same speed – speed of light (300 X 10^6 m/s)
- In copper all waves travel at about 2/3 of the speed of light.

The Electromagnetic Spectrum (2)

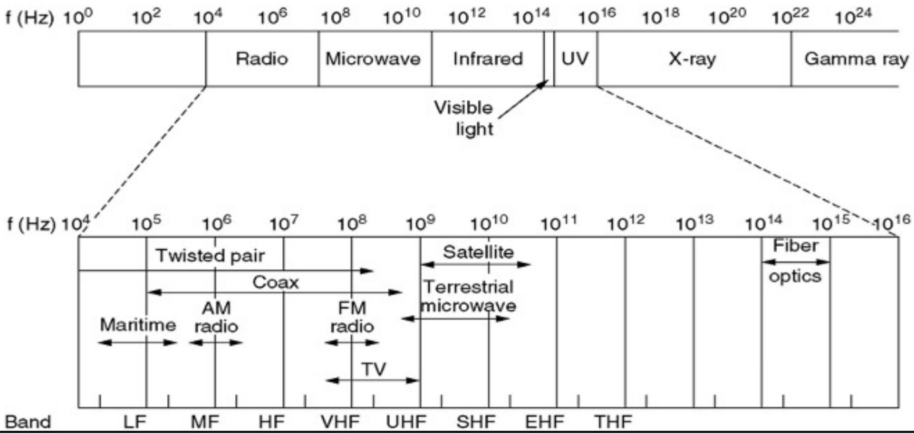
- The relation between the frequency and wavelength of an electromagnetic wave is given by the relation (in vacuum):

$$\lambda * f = c$$

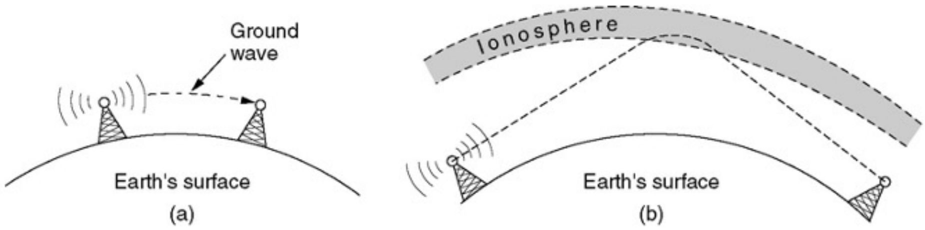
- Since c is a constant, if we know the frequency we can compute the λ and vice versa
- Example: compute the wavelength of a signal having 100MHz.
  - $100*10^6 * \lambda = 300 * 10^6$
  - $\lambda = 3$ meters

The Electromagnetic Spectrum (3)

The electromagnetic spectrum and its uses for communication.



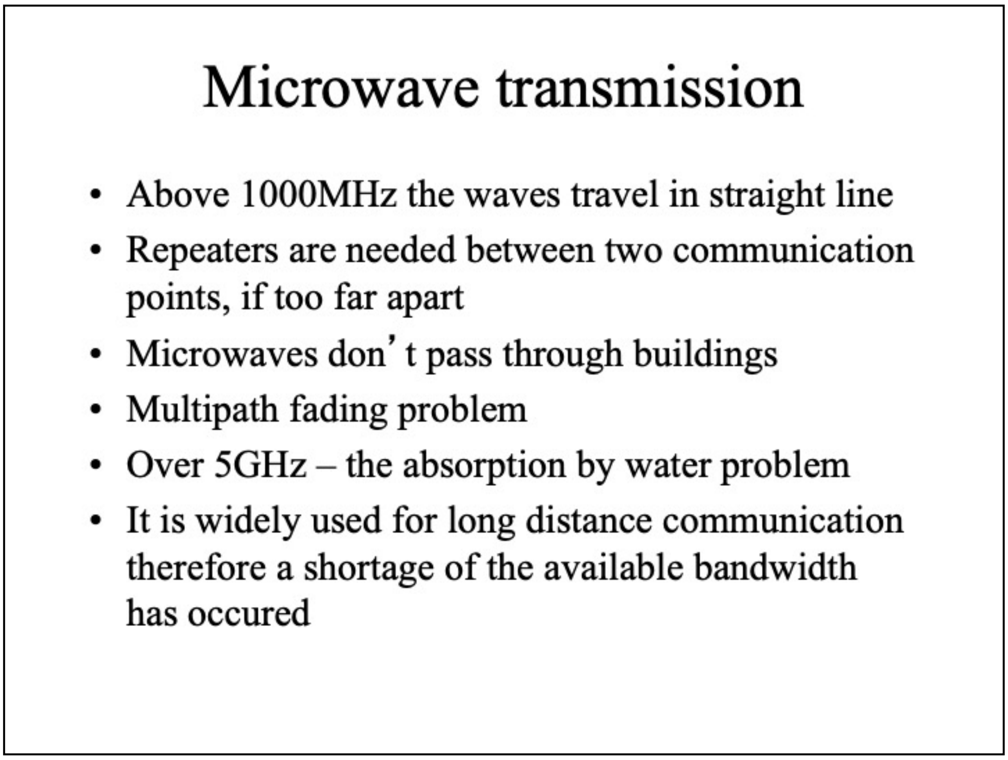
Radio Transmission



(a) In the VLF, LF, and MF bands, radio waves follow the curvature of the earth.

(b) In the HF band, they bounce off the ionosphere.

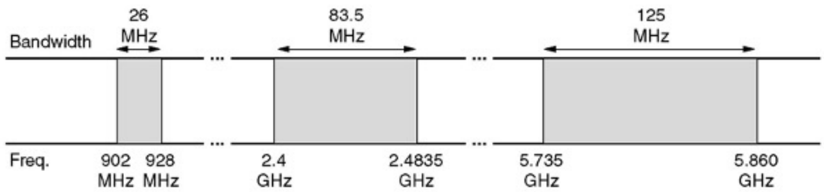
Almost all transmission use a narrow frequency band (with few exceptions – frequency hopping spread spectrum and direct sequence spread spectrum). For the moment we assume all transmission do use a narrow frequency band.



Multipath fading refers to the fact that delayed waves may arrive out of phase with the direct wave and thus cancel the signal. Some operators keep their channels idle as spares to switch on when multipath fading wipes out some frequency band temporarily. It is a phenomena that is dependent on weather and on the operating frequency.

Politics of the Electromagnetic Spectrum

The ISM bands in the United States.



Due to radio waves ability to travel long distances, interference between users is a problem. For this reasons, governments license the use of radio transmitters.

ISM – Industrial, Scientific and Medical. The ISM bands is different somehow from country to country.

Infrared and Millimeter Waves

- Widely used for short range communication (i.e. used in remote controls)
- Directional and easy to build but don't pass through objects
  - This is also a plus – it means no interference with similar systems sitting in next room, no security issues, etc..
- No government license is needed to operate infrared systems
- It has some usage in interconnecting some devices as well (i.e. mice to computers, mobile phones to other equipment, etc…)

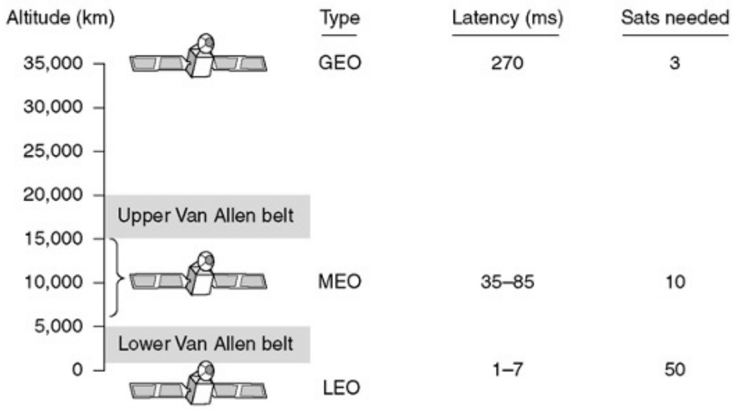
The more we go from long wave radio towards visible light, the electromagnetic wave behaves more and more like light and less and less like radio.

Communication Satellites

- Geostationary Satellites
- Medium-Earth Orbit Satellites
- Low-Earth Orbit Satellites

In its most simple form, a communication satellite can be seen as a big microwave repeater up in the sky. It contains several transponders, each of which listens to some portion of the spectrum, amplify the incoming signal and then rebroadcasts it to another frequency to avoid interference with the incoming signal. The downward beam can be broad (covering a large area of Earth's surface) or can be narrow (covering only a few hundreds of KM in diameter).

Communication Satellites (1)



Communication satellites and some of their properties, including altitude above the earth, round-trip delay time and number of satellites needed for global coverage.

Issues with satellites:

1. Orbital period – the highest the satellite, the larger the period (at an altitude of about 36 800 KM, the period is about 24 hours, at 384 000 the period is about one month)
2. Presence of the van Allen belts – layers of highly charged particles trapped by earth's magnetic field. Any satellite flying inside of those belts would be destroyed quickly.

GEO – Geostationary Earth Orbit satellites that sit at about 35 800 in a circular equatorial orbit (Arthur C. Clarke – science fiction author – computed that at this altitude, the satellite would appear to remain motionless in the sky).

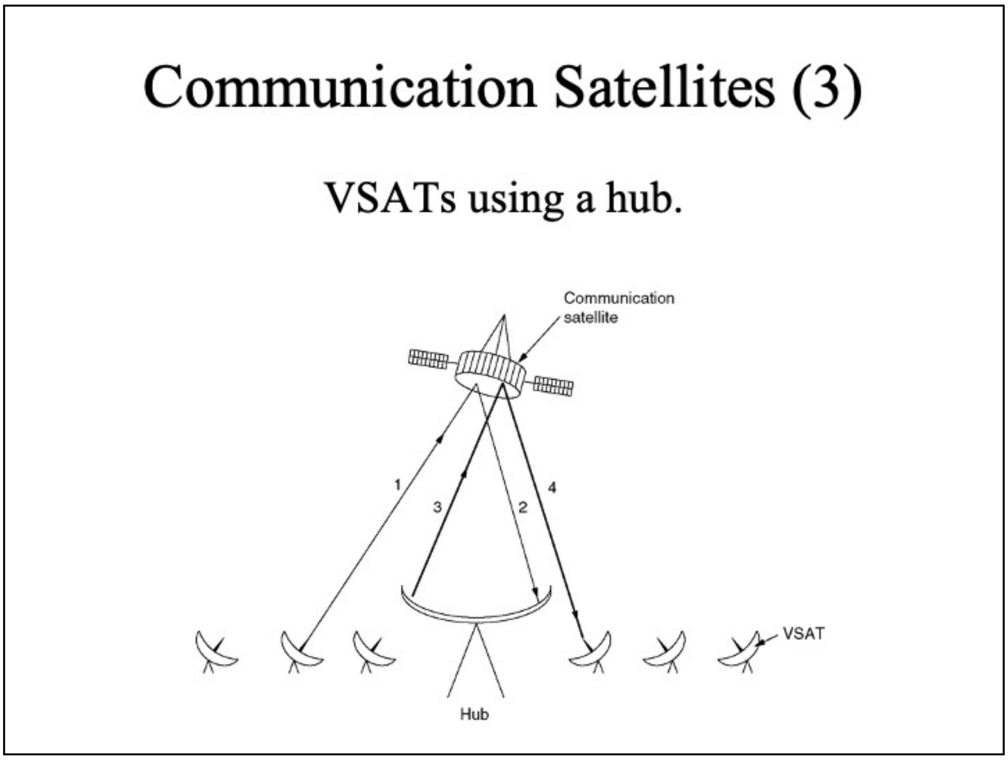
MEO – Middle Earth Orbit satellites – between the two van Allen belts

LEO – Low Earth Orbit satellites

Communication Satellites (2)

The principal satellite bands.

Band	Downlink	Uplink	Bandwidth	Problems
L	1.5 GHz	1.6 GHz	15 MHz	Low bandwidth; crowded
S	1.9 GHz	2.2 GHz	70 MHz	Low bandwidth; crowded
C	4.0 GHz	6.0 GHz	500 MHz	Terrestrial interference
Ku	11 GHz	14 GHz	500 MHz	Rain
Ka	20 GHz	30 GHz	3500 MHz	Rain, equipment cost



VSAT – VERY SMALL APPERTURE TERMINALS – 1 meter or smaller compared to 10 meters required for GEO satellites communication.

Middle Earth Orbit Satellites

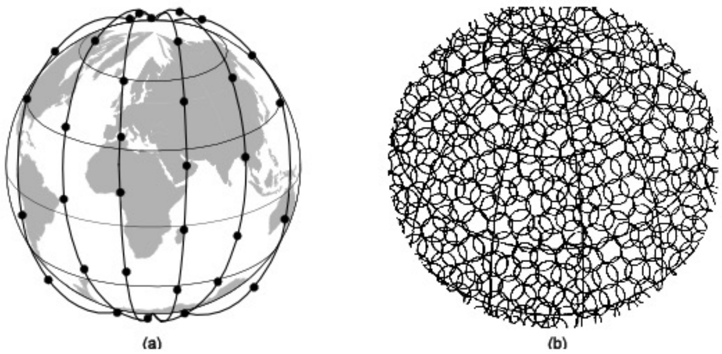
- Must be tracked as they move the sky
- Less powerful transmitters to reach them
- They are not usually used for communication
- One example is the GPS (General Positioning System) satellites
  - 30 satellites used for the GPS system are orbiting at about 18000KM. They are MEO satellites

GPS needs a minimum of 24 to operate and the typical number active is about 30. GPS receivers released since 2018 have much higher accuracy, pinpointing to within 30 centimeters.

Low Earth Orbit Satellites

- Due to their motion, large number of them are needed for a complete coverage
- Because are low orbit, the ground stations don't need much power and the round-trip delay is only a few milliseconds
- Examples
  - Iridium
  - Starlink

Iridium



(a) The Iridium satellites from six necklaces around the earth.

(b) 1628 moving cells cover the earth.

In 1990 Motorola started so called Iridium project. Its goal was to use 77 low orbit satellites (the element 77 is Iridium) to cover completely the whole surface of the Earth. When one satellite went out of view, another one would replace it. Latter, the project was revised and instead 77 satellites, only 66 where used.

The satellites were launched in 1997 and the service begun in 1998. Iridium wasn't profitable, being one of biggest fiascos in history (this partially because of the cellular telephony developing so rapidly).

The Iridium service was restarted in March 2001, providing worldwide communication services using hand-held devices communicating directly with satellites. It provides voice, data, fax and navigation services everywhere on land, water or air.

Iridium satellites are positioned at 750KM altitude, in circular polar orbits. They are arranged in North-South necklaces, with one satellite very 32 degrees of latitude. With six satellite necklaces, the entire earth is covered.

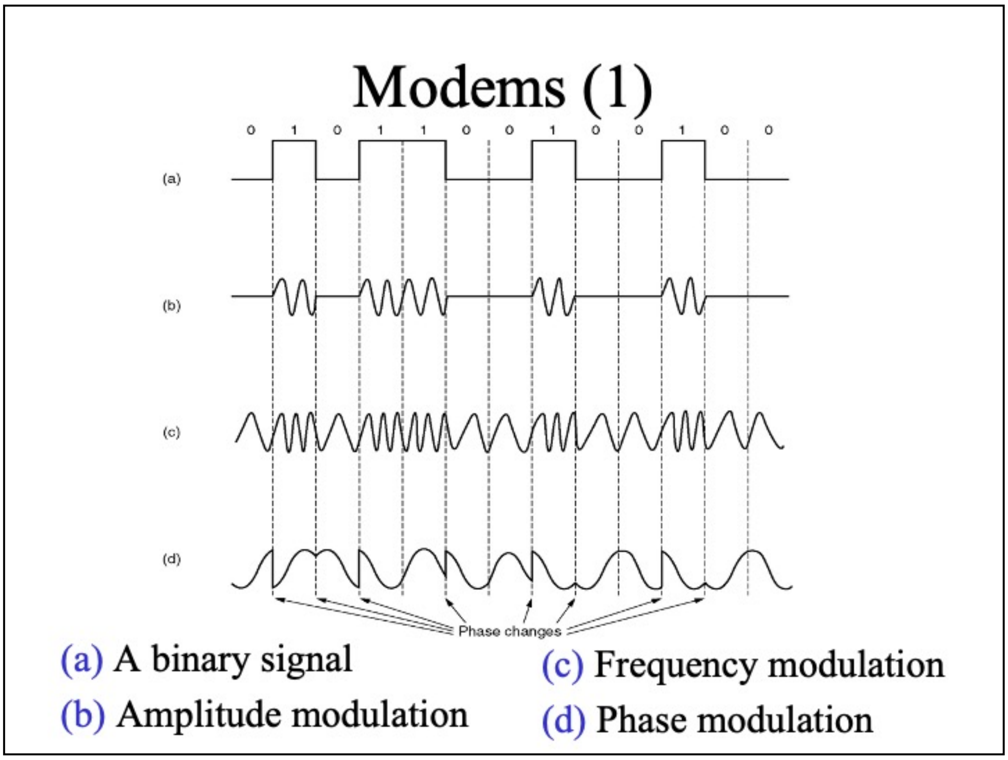
Each satellite has a maximum of 48 cells (spot beams) with a total of 1628 cells over the surface of the earth. Each satellite has a capacity of 3840 low bandwidth phone channels.

Modulation (1)

- Due to attenuation (and other discussed problems) square waves (digital signal) can't be used for long distance transmission
- AC signaling is used instead (continuous tone, around 1000 – 2000 Hz, called sin wave carrier) is introduced
- Its amplitude, frequency or phase can be **modulated** to transmit information

Modulation (2)

- **Amplitude modulation**
  - Two different amplitudes are used to represent 0 or 1
- **Frequency modulation** (or frequency shift keying)
  - Two (or more) different tones are used
- **Phase modulation**
  - The carrier wave is shifted 0º or 180º at bit intervals, to show a transition.
  - A better scheme is to use shifts of 45, 135, 225 or 315 º to transmit two bits of information



A modem is a device that accepts a serial stream of bits as input and produces a carrier modulated by one (or more) of these methods (modulator/demodulator). The modem is inserted between the (digital) computer and the (analog) telephone system.

- – binary signal
- - amplitude modulation using absence of carrier to encode a ‘0’ and presence to encode a ‘1’
- - frequency modulation using waves with different frequencies to encode ‘0’ and ‘1’
- - phase modulation using phase shifts to show a change from ‘0’ to ‘1’ and ‘1’ to ‘0’. No phase shift means the signal stays the same (0 or 1 during the bit intervals).

Modems (2)

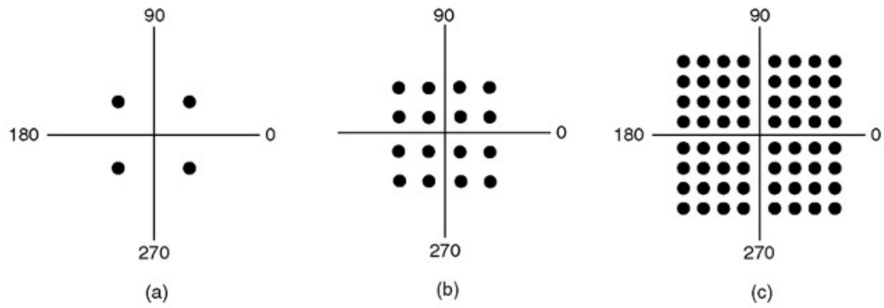
- A phone line has a band of 3000Hz, no point in sampling the line at faster rates than 6000 times per second (according to Nyquist theorem)
  - In practice most modems sample at 2400 times per seconds and focus in getting more bits per sample
- The number of samples per second is measured in **baud**. During one baud one symbol is sent
- One symbol can consist of one or more bits

N baud line transmits n symbols per second. A 2400 bps line transmits one symbol about every 416.667 microseconds. If the symbol consists of one bit, than the line is 2400 bps. If the symbol consists of 2 bits per symbol, then the bit rate is 4800 bits per second.

Bandwidth, baud rate and bit rate

- Bandwidth of a medium is the range of frequencies that pass through with minimum attenuation
  - Physical property of the medium (0 to some maximum frequency, measured in Hz)
- Baud rate – number of samples made per second
  - Per sample is sent out one symbol, therefore the baud rate and symbol rate are the same
- Bit rate – amount of information sent over the channel and is equal to the number of symbols/sec times the number of bits/symbol

More about Modems (1)



(a) QPSK.

(b) QAM-16.

(c) QAM-64.

QPSK – QUADRATURE PHASE SHIFT KEYING – four phases are used

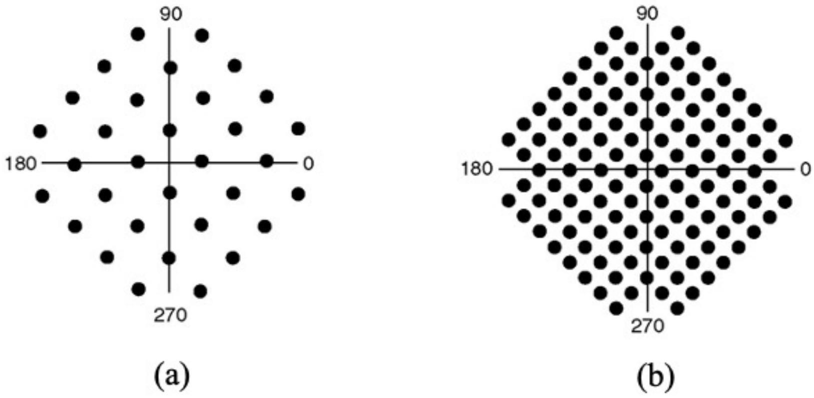
QAM-16 – QUADRATURE AMPLITUDE MODULATION – four amplitudes and four phases to form a total of 16 possible combinations. In other words, a symbol can encode 4 bits. 9600 bits per second over a 2400 baud line

QAM-64 – QUADRATURE AMPLITUDE MODULATION – a total of 64 possible combinations. In other words, a symbol can encode 6 bits.

Higher order QAMs are also used.

Those diagrams are called CONSTELLATION DIGARAMS – show the legal combinations of amplitude and phase. Each modem is using its own constellation diagram and can talk only with modems implementing same constellation diagrams.

More about Modems (2)



(a) V.32 for 9600 bps.

(b) V32 bis for 14,400 bps.

With many points in the constellation diagram, even a small amount of noise would result in transmission errors.

TCM (TRELLIS CODED MODULATION) is used, where beside the data, a symbol contains parity checking.

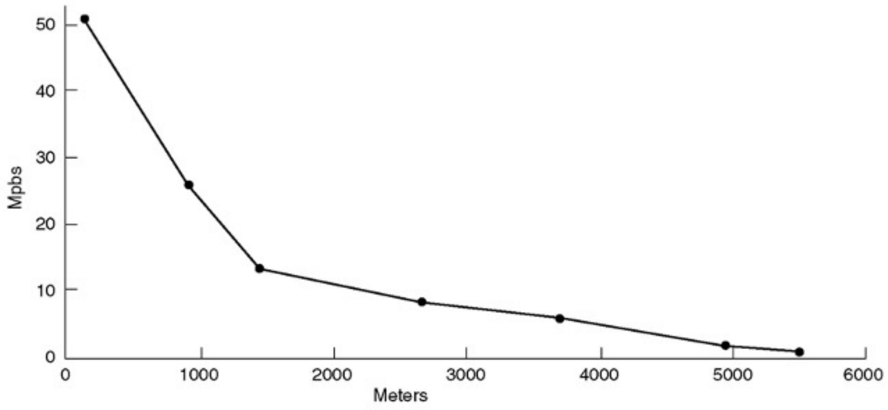
The V32 modem is using 32 points constellation to send 4 bits of data and 1 bit of parity per each symbol, at an effective bit rate of 9600 bps.

Next step up from **V32** is **V32.bis**, where 128 points constellation is used. 6 bits for data and one for parity in a 7 bit symbol, at a line of 2400 baud. It achieves 14400 bps effective data bit rate.

V34 runs at 28800 at 2400 baud channel with 12 data bits/symbol. V34.bis runs at 33600 over a 2400 baud line, using 14 data bits/symbol.

Modems are bidirectional devices, by using different frequencies for different directions. Bidirectional simultaneously connections are called full duplex connections.

Digital Subscriber Lines (1)



Services with more bandwidth than standard telephone services are called BROADBAND (more marketing than technical).

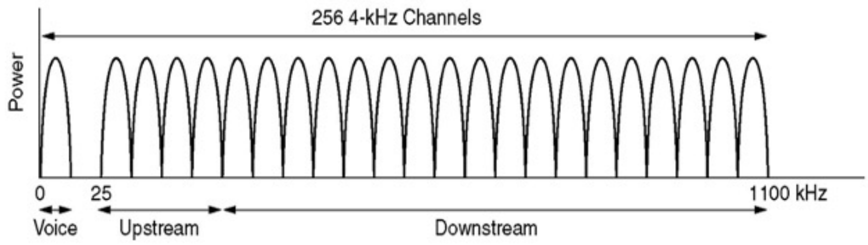
DSL (DIGITAL SUBSCRIBER LINE) services started to be offered by phone companies, to be competitive on the broadband services market (TV and satellite companies were already offering broadband services).

The reason that modems are so slow is because phone lines were invented to carry human voice not data. Therefore, frequencies bellow 300 HZ and over 3100Hz were artificially filtered out. Cutoff filters were added on the local loops to filter out other frequencies than voice. The cutoff is not sharp, so the bandwidth is usually quoted to be 4000Hz even thought the distance between the 3dB points is 3100 Hz. Data is thus restricted to this narrow band.

The trick that made DSL (or ADSL – Asymmetric DSL) work is that when a customer subscribes to it, the incoming line is connected to a different kind of switch (in the end office), one that doesn't have this voice filter installed, thus making the entire capacity of the local loop available.

The capacity of the local loop depends on the length and thickness of the cable and an plot of it is presented here. Therefore the ADSL services are limited to customers that are within certain radius from the end office (phone company switch).

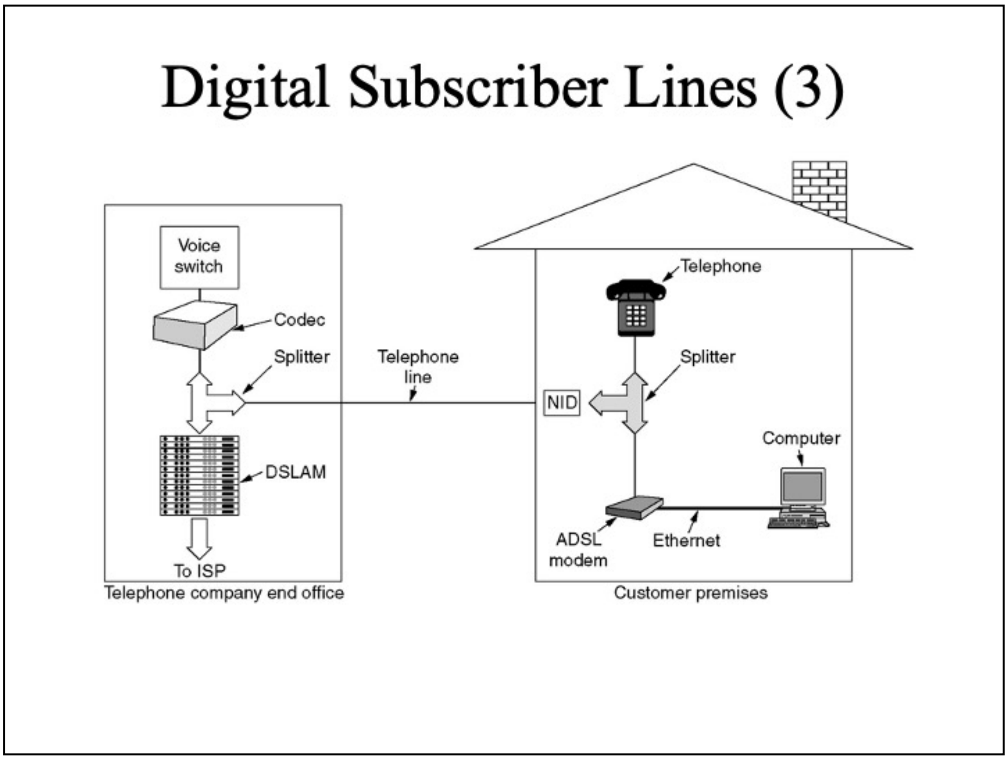
Digital Subscriber Lines (2)



Operation of ADSL using DMT (**Discrete Multi Tone**) modulation.

The available spectrum of the local loop (which is 1.1 MHz for a decent service) is divided into 256 channels of 4312.5 Hz each. Channel 0 is used for POTS (Plain Old Telephone Service). Channels 1 to 5 are not used, to make sure that there is enough guard between voice and data. The remaining 250 channels are used for data and stream control (one is used for upstream control and one is used for downstream control). It is up to the provider how many channels will be used for upstream and how many for downstream. A 50% 50% mix for upstream and downstream is possible, but usually the providers allocate more for downstream than for upstream (80% to20%, since most of the subscribers are using the line for Internet, so they pull pages down onto their system. This is why DSL service turned into ADSL service). A common split is 32 channels for upstream and the rest for downstream.

Within each channel, a modulation schema similar to V.34 is used, although the sampling rate is 4000 baud instead of 2400. The line quality in each channel is constantly monitored and the data rate adjusted continuously as needed, so different channels may have different data rates. The data is sent with QAM modulation with a maximum of 15 bits per symbol. With 224 downstream channels, the downstream data bit rate is about 13.4 Mbps. In practice, the signal to noise is never good enough to achieve this bit rate, but 8 Mbps is possible on short local loops, and the ADSL standard goes this far.



A telephone company technician must install a NID (NETWORK INTERFACE DEVICE) in the customer's house. Close or combined with the NID, a SPLITER is installed, which is an analog filter that separates the phone signal (0 to 4000Hz) from the data signal (over 26kHz).

The ADSL modem is actually a signal processor that acts as 250 QAM modems, operating in parallel at different frequencies available for each channel. Usually, the connection between the modem and the computer is Ethernet based.

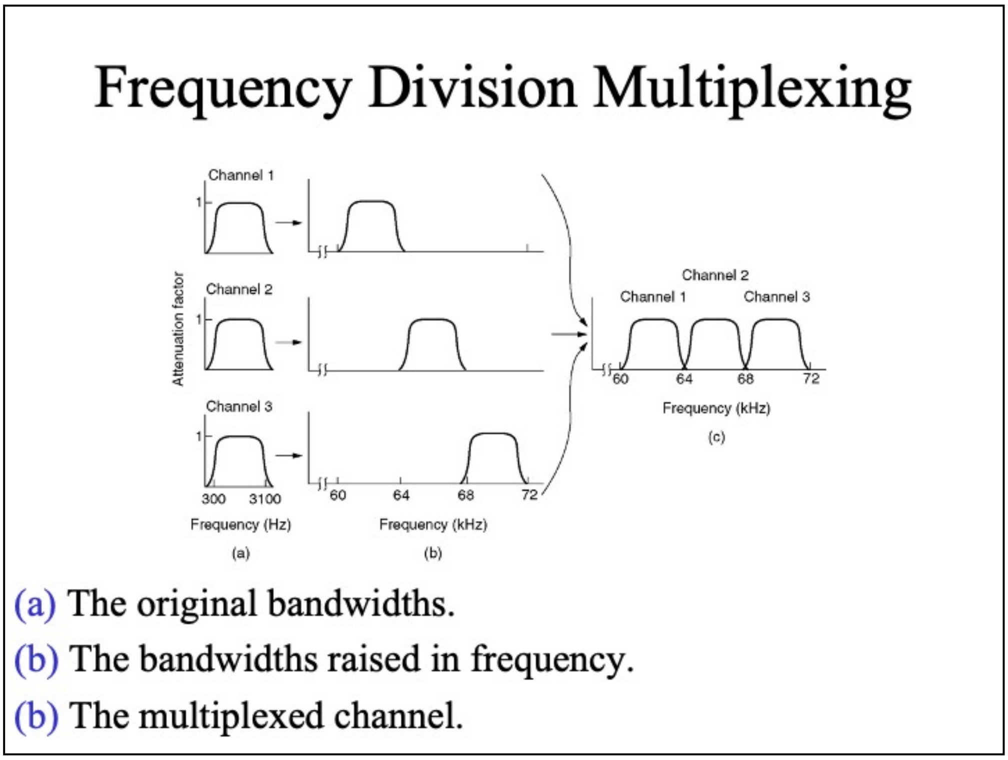
At the other end (End Office) a special DSLAM (DIGITAL SUBSCRIBER LINE ACCESS MULTIPLEXER) receives the data over 26 KHz (separated by the splitter) and recovers the bit stream from the data (same function as the ADSL modem at the customer end). The low frequency signal (POTS signal) is sent to a conventional switch.

Trunks and data Multiplexing

- FDM (Frequency Division Multiplexing)
  - Wavelength division multiplexing
- TDM (Time Division Multiplexing)

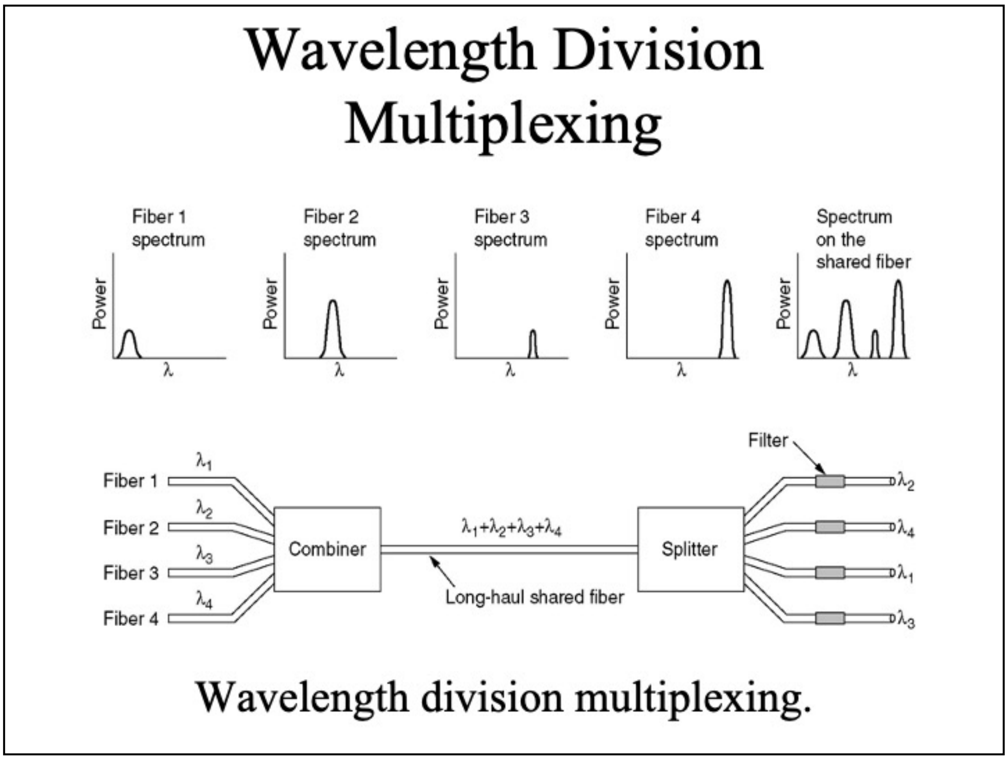
High bandwidth trunks are available between switching offices. The data collected from end loops needs to be multiplexed over those high bandwidth trunks. There are two main categories for data multiplexing: FDM and TDM.

In FDM the frequency spectrum is divided into frequency bands, with each user having exclusive possession of some band, while in TDM, the users take turn (in a round robin fashion), each one getting periodically the entire available bandwidth for a short period of time.

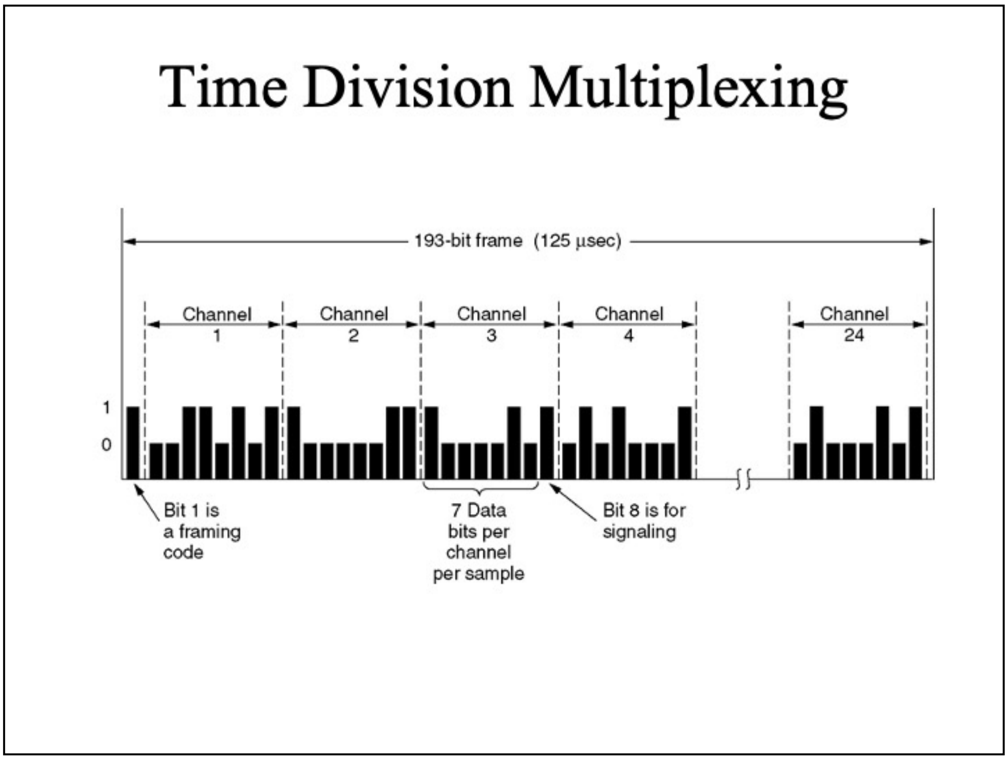


This slide presents how voice grade telephone channels are multiplexed using FDM. When many channels are multiplexed together, 4KHz bandwidth is allocated to each channel, to keep them separated. First, each channel is raised in frequency, each by a different amount. Then they can be combined, because no channel will occupy the same portion of the spectrum.

FDM schemas are to some degree standardized – a wide spread standard is 12 4KHz voice channels multiplexed into 60 to 108kHz bad. This is called a **GROUP**. Five groups can be multiplexed to form a **SUPERGROUP**. The next group is the **MASTERGROUP** which is made out of 5 supergroups.



It is a variation of the FDM at very high frequencies and it is used over the optical fiber trunks. Systems with 96 channels of 10Gbps each are available in production. 960 Gbps is enough to send 30 full time movies per second…



How multiple analog voice telephone channels are digitized and combined onto a single outgoing digital trunk.

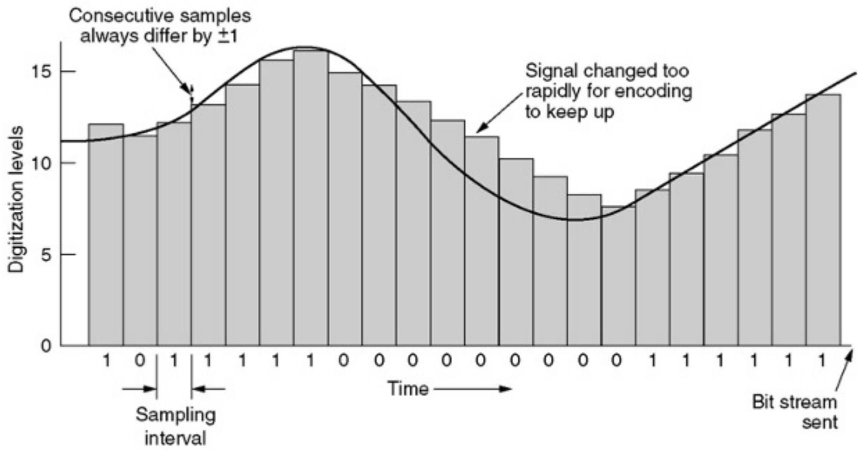
The analog data is digitized in the end office by a device called **CODEC**. The codec makes 8000 samples per second (125 us per sample, according to Nyquist theorem it is enough to sample at twice the bandwidth of the signal to capture all the information carried by the signal) and measures the amplitude of each sample with a precision of 8 bits. This process is called PCM (**PULSE CODE MODULATION**). PCM forms the heart of the modern telephone service. As a consequence, all time intervals within a phone system are multiples of 125 us.

Incompatible schemas are in use in US and Europe. Japan and US are using the T1 carrier (presented in this slide) while in Europe E1 carrier is used.

Each T1 frame accommodates 24 channels (sampled each 8000 times/second, 7 bit per sample), giving a gross data rate of 1.544 Mbps. An extra bit is used for framing control. This bit is changing value from 0 to 1 and from 1 to 0 every frame. The changing patter is 01010101… and it is used by the receiver to make sure it is not loosing synchronization. Analog customers can't generate this wave, since it corresponds to a sin wave at 4000 Hz (this frequency is not present in the analog voice channels).

E1 has 32 channels, 8 bit per channel packed into the basic 125 us frame. This yields to a gross data rate of 2.048 Mbps.

Time Division Multiplexing (2)

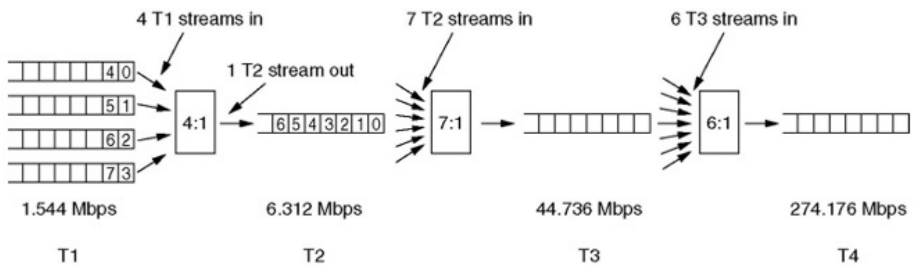


Once the voice channel has been digitized, it is tempting to use statistical techniques to obtain fewer bits per sample (therefore lower bandwidth required to carry the voice information). All the compaction methods are based on the fact that the analog voice signal is changing relatively **SLOWLY** compared to the sampling frequency, so much in the information in the 7^{th} and 8^{th} bit is redundant.

DPCM (DIFERENTIAL PULSE CODE MODULATION) is based on producing as output the difference between the current value and the previous one. Since jumps of +/- 16 in a scale from 0 to 128 are unlikely, only 5 bits are enough to represent the audio signal. If the signal does occasionally jump sharply, than a few samples are required to catch up with the new value. For speech, this kind of error can be ignored.

DELTA MODULATION is a variation of DPCM and requires the current value to differ only with + or -1 from the previous value. Under those conditions, only a bit can be sent to indicate if the next value is over or below the previous value.

Time Division Multiplexing (3)



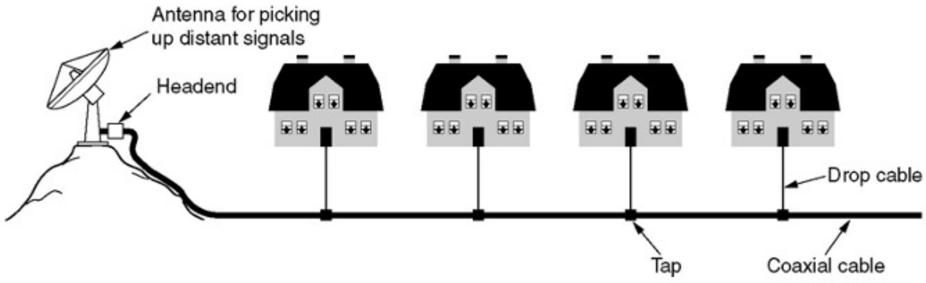
Multiplexing T1 streams into higher carriers.

Cable Television

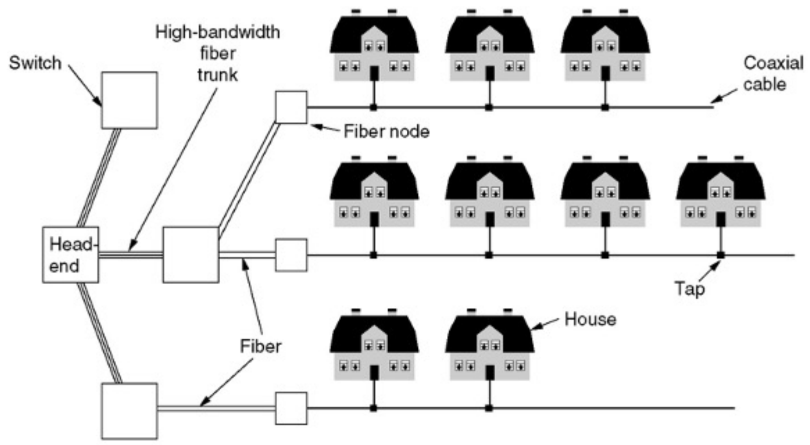
- Community Antenna Television
- Internet over Cable
- Spectrum Allocation
- Cable Modems

Community Antenna Television

An early cable television system.

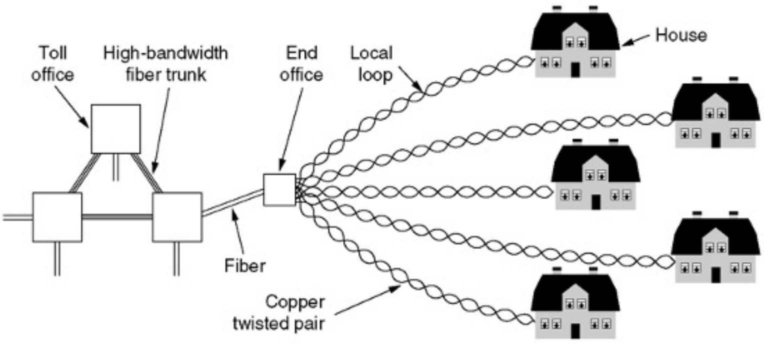


Internet over Cable



(a)

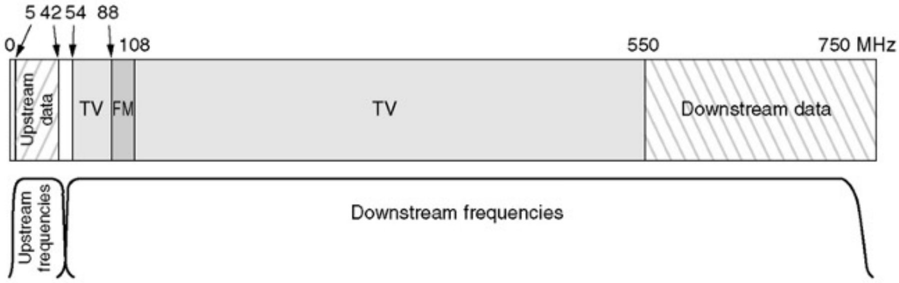
Internet over Cable (2)



(b)

Spectrum Allocation

Frequency allocation in a typical cable TV system used for Internet access



Cable Modems

Typical details of the upstream and downstream channels