EE 533 TELECOMMUNICATION NETWORKS

COURSE CONTENTS

Introduction to Telecommunication Networks, Backbone Bandwidth Requirements, Transmission Line
Virtual Circuits, Analog and Digital Communications
ASK, FSK, PSK, Multiplexing, PCM, TDM, FDM
Standards for Digital Data and Video,
Statistical Multiplexing, Intelligent multiplexing, Inverse Multiplexing, WDM
Transmission Medium
Microwave System Design,
Midterm
MMDS,LMDS
Wireless Local Loop Systems
Frequency Hopping and Spread Spectrum
WLAN Systems, Satellite
Optical Fiber, Laser Diodes,
Free Space Optics

REFERENCE BOOKS:

1. NAME : Communication Networks-Fundamental Concepts and Key Architectures
   AUTHORS : Alberto Leon-Garcia, Indra Widjaja
   PUBLISHER : McGraw-Hill
   ISBN : 0-07-123026-2
   EDITION : 2003 (International Edition)

2. NAME : Essential Guide to Telecommunications
   AUTHORS : Annabel Z. Dodd
   PUBLISHER : Prentice-Hall, Inc.
   ISBN : 0-13-064907-4
   EDITION : 2002 (Third Edition)

3. NAME : Communication Systems Engineering
   AUTHORS : John Proakis, Masoud Salehi
   PUBLISHER : Prentice-Hall, Inc.
   ISBN : 0-13-061793-8
   EDITION : 2002 (Second Edition)

4. NAME : Optical Fiber Communications
   AUTHOR : Gerd Keiser
   PUBLISHER : McGraw-Hill
   ISBN : 0-07-116468-5

EDITION : 2000

5. NAME : Data Communications and Networking
AUTHOR : Behrouz A. Forouzan
PUBLISHER : McGraw-Hill
ISBN : 0-201-63442-2
EDITION : 2001 (Second Edition)

6. NAME : Telecommunications Essentials
AUTHOR : Lillian Goleniewski
PUBLISHER : Addison-Wesley
ISBN : 0-201-76032-0
EDITION : 2002

7. NAME : Communication Sysytems
AUTHOR : Simon Haykin
PUBLISHER : John Wiley&Sons
ISBN : 0-471-17869-1
EDITION : 2001 (Fourth Edition)

8. NAME : Modern Digital and Analog Communication Systems
AUTHOR : B. P. Lathi
PUBLISHER : Oxford Univ. Press, Inc
ISBN : 0-19-511009-9
EDITION : 1998

9. NAME : Next generation intelligent optical networks
AUTHOR : Kartalopoulos, Stamatios
CODE : TK5103.59 K37
EDITION : 2008

10. NAME : Data communications and networks
AUTHOR : Miller, Dave
CODE : TK5105 M55
EDITION : 2006

11. NAME : Digital communications
AUTHOR : Proakis, John G
CODE : TK5103.7 P76
EDITION : 2008

12. NAME : Introduction to digital communications
AUTHOR : Pursley, Michael B.
CODE : TK5103.7 P87
EDITION : 2005

13. NAME : Advanced free-space optical communications
AUTHOR : Ross, Monte
CODE : TA1677 A38
EDITION : 2004

14. NAME : Advanced free-space optical communications
AUTHOR : Ross, Monte
CODE : TA1677 A38

EDITION : 2004

15. NAME : Mobile wireless communications
    AUTHOR : Schwartz, Mischa
    CODE : TK5103.2 S39
    EDITION : 2005

16. NAME : Communication systems: analysis and design
    AUTHOR : Stern, Harold P.E.
    CODE : TK5101 S74
    EDITION : 2004

17. NAME : Electronic communications systems and design
    AUTHOR : Tomasi, Wayne
    CODE : TK5101 T66
    EDITION : 2004

18. NAME : Principles of communication systems simulation
    AUTHOR : Tranter, William H.
    CODE : TK5102.5 P75
    EDITION : 2004

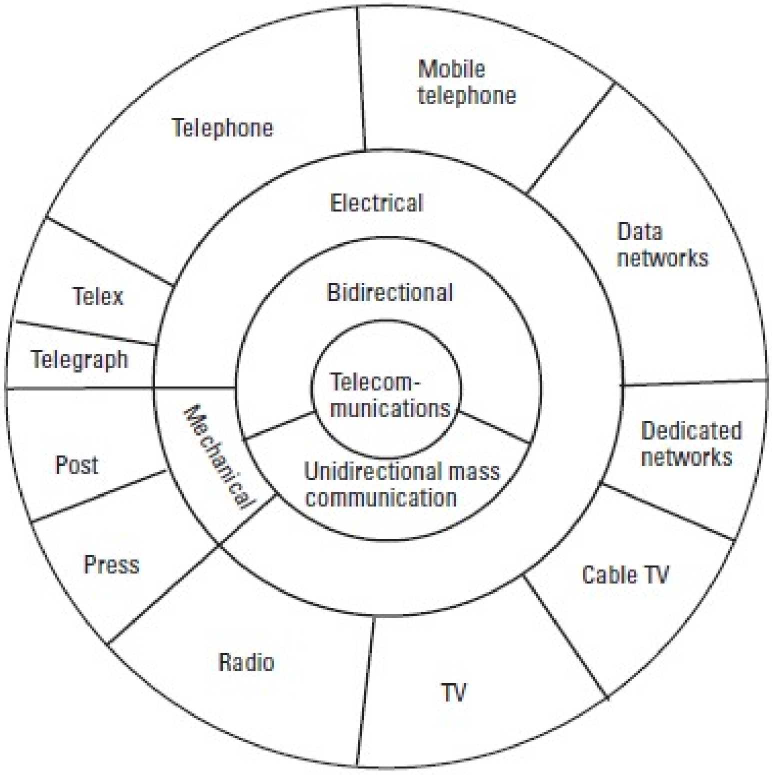
19. NAME : Introduction to Telecommunications Network Engineering
    AUTHOR : Tarmo Anttalainen
    CODE : TK5105 .A55
    EDITION : 2003

GRADING: 1 MID TERM EXAM (IN CLASS) : %45
1 FINAL EXAM (IN CLASS) : %55
TOTAL : %100

Introduction to Telecommunication Networks

What is Telecommunications?

Telecommunications can be defined as a technology concerned with communicating from a distance.

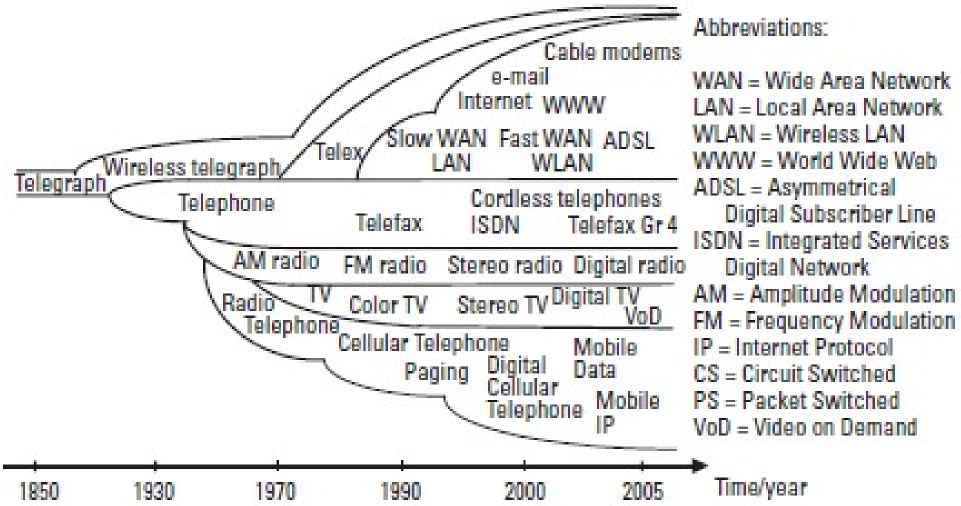


Telecommunications networks is the most complex system that the mankind designed. Only the telephone network has billions and billions of fixed and cellular telephones with universal access. When any of these telephones requests a call, the telephone network is able to establish a connection to any other telephone in the world.

Additionally, many other networks are interconnected with the telephone network.

Some services provided by telecommunications are telephone conversation, television, data transfer, banking, automatic teller machines, telebanking, aviation, booking of tickets, sales, wholesale and order handling, purchasing, credit card payments, booking, any internet transaction, etc.

Historical development of telecommunication systems



Standardization: Communication networks are designed to serve a wide variety of users who are using equipment from many different vendors. To design and build networks effectively, standards are necessary to achieve interoperability, compatibility and required performance. Some standards organisations are European Telecommunications Standards Institute (ETSI), The Conférence Européenne des Administrations des Postes et des Telecommunications or European Conference of Posts and Telecommunications Administrations (CEPT), The International Telecommunication Union (ITU), The Comité Consultatif International de Télégraphique et Téléphonique or International Telegraph and Telephone Consultative Committee (CCITT/ITU-T) is presently called ITU-T, where the "T" comes from telecommunications, IEEE ... etc.

Open standards are needed to enable the interconnection of systems, equipment, and networks from different manufacturers, vendors and operators.

Parties involved: Equipment manufacturers, network operators, academic experts, service users.

Basic Telecommunications Network

Basic purpose of telecommunication networks is to transmit user information in any form to another user of the network.

These users of public networks, for example, a telephone network, are called subscribers.

User information may take forms such as voice, video, image, text, data and combination (multi-media).

Subscribers may use different backbone or access network technologies to access the network, e.g. fixed or cellular telephone network, IP networks, ... etc.

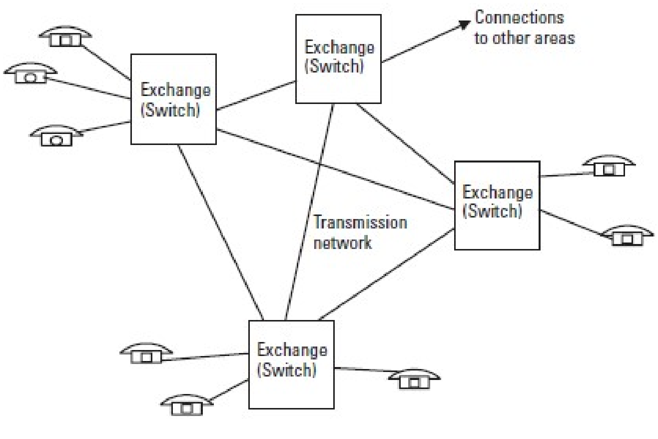
Transmission:

Transporting information between end points of a system or a network. Transmission systems use different media for information transfer from one point to another:

Twisted pair copper cables used in LANs and telephone subscriber lines, coaxial cable used in cable TV, optical fiber cables used high-data-rate transmission, radio waves used in cellular telephones, satellite, free-space optics used in access systems.

Switching:

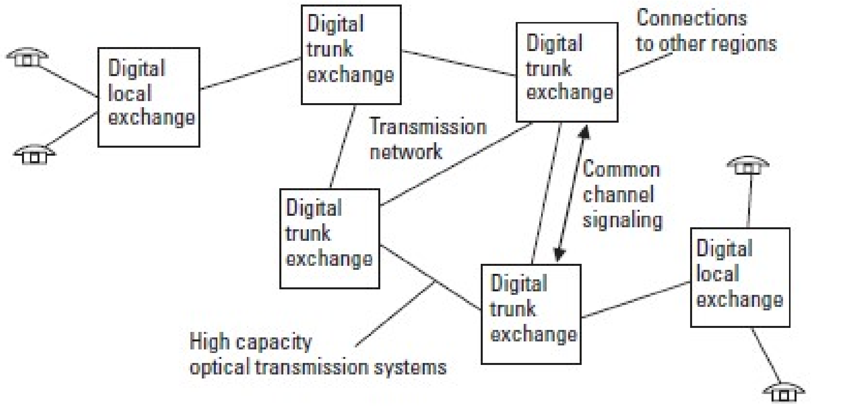
When there are large number of subscribers, the signals are switched from one wire to another. Then only a few cable connections are needed between exchanges since the number of simultaneously ongoing calls is much smaller than the number of telephones.



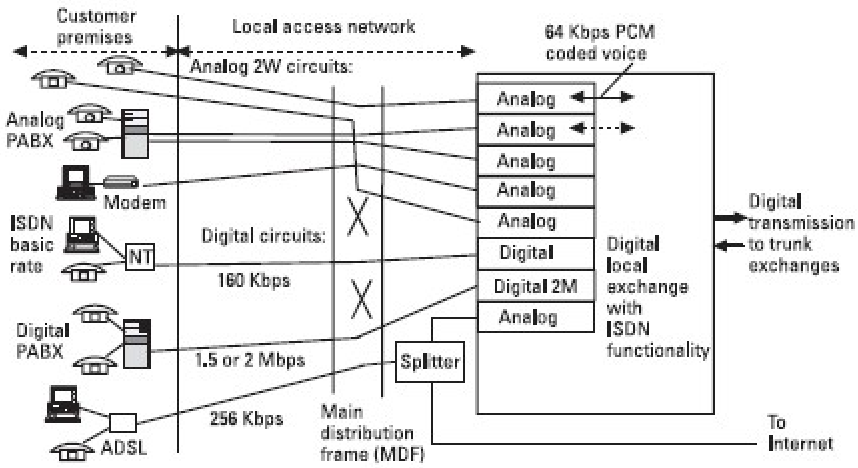
Signaling: These are non-information carrying signals, e.g., dialing digits, busy, starting communication, routing, ending communication.

Overall Telecommunication Network is composed of Backbone (Core) Networks and Access Networks.

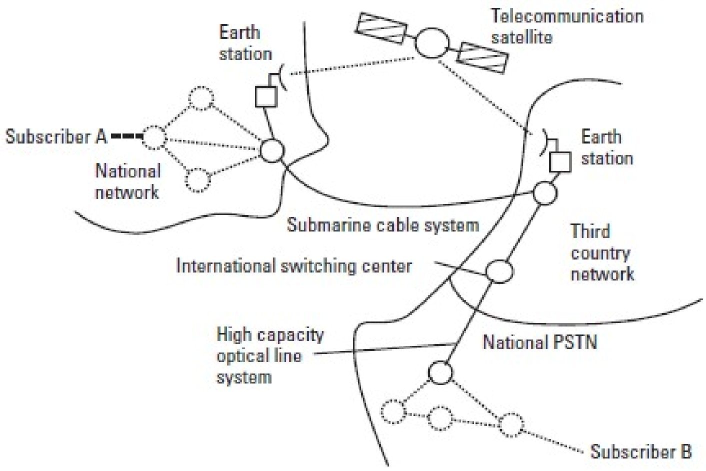
Backbone (Core) Networks: Highway in which very high data rate information flows.



Access Networks: Relatively smaller data rate information flow at the end points of the core network reaching the end users.



International Network:



Human Senses Added in Telecommunications

- Hearing and speaking computers
- Virtual touch known as haptics
  - The user feeling the weight of jewelry as if it is in the user's hand
  - The user feeling the fur of an animal
  - Virtual reality job training
  - Computer-aided design
  - Remote handling of hazardous materials
  - "Touch" museums

- Smell applications in computers
  - Aroma to trigger fear, excitement and other emotions

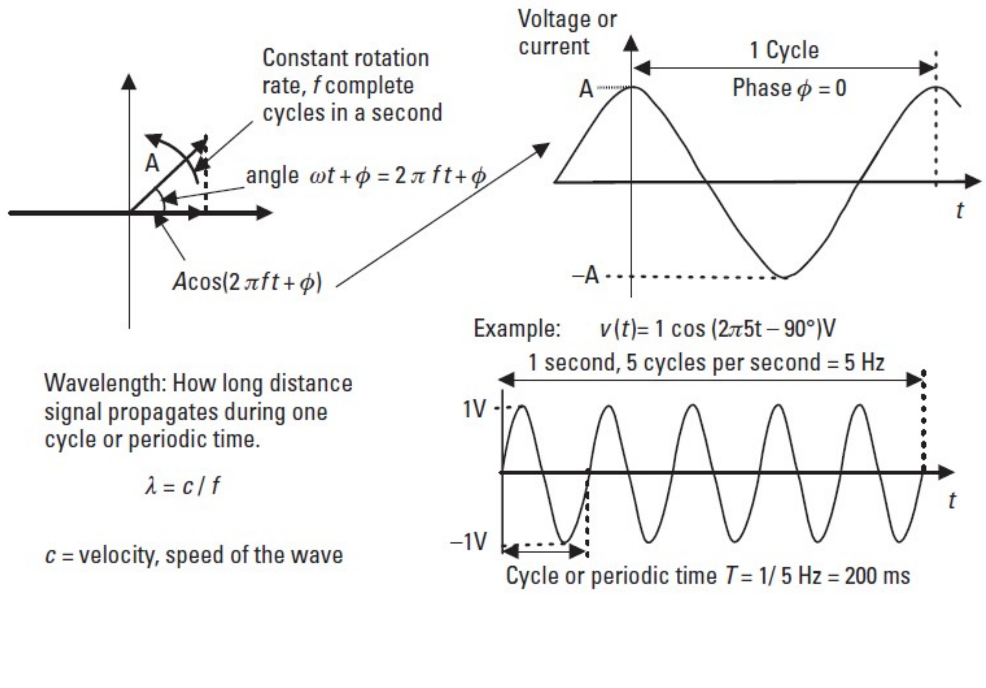
- e-commerce

- Seeing computers (equipped with camera)
  - Capturing and sending images
  - Displaying high-quality entertainment programming

- Wearable Computing – Dressed for success
  - Today's portable devices are approaching to wearables
  - Wearable computer with CPU, disc, RAM ..etc in the form of dress, wrist keyboard, headgear suspended in front of the eye, in the size of a stamp, providing full-color screen appearing as 15 inch monitor, shirt pocket video camera
- Intelligent home and office utilities – Becoming more intelligent, getting smaller, more powerfull
- Smart refrigerators, smart washing machines, smart ovens, smart furniture, smart T.V .... etc.
- Machine-to-machine communication
  - In 2030 it is estimated that 98% of the total communications traffic in the world will be machine-to-machine communication and only 2% will be human-to-human.

Frequency and Bandwidth

Telecommunications signal as a combination of many cosine or sine waves with different strengths and frequencies. The frequency refers to the number of cycles the wave oscillates in a second.



As an example of the concept of frequency, we hear the oscillation of air pressure as sound. We hear frequencies in the range of approximately 20 Hz to 15 kHz, where Hz (hertz) represents the number of cycles in a second.

The information that we transmit through a telecommunications network, whether it is analog or digital, is in the form of electrical voltage or current. The value of this voltage or current changes through time, and this alteration contains information.

The transmitted signal (the alteration of voltage or current) consists of multiple frequencies.

The range of frequencies that make up the signal is called the bandwidth of the signal.

The bandwidth is one of the most important characteristics of analog information and it is also the most important limiting factor for the data rate of digital information transfer.

- Narrowband: Small bandwidth carrying small amount of information,
- Wideband: Large bandwidth carrying large amount of information.

Wavelength: How long distance signal propagates during one cycle or periodic time.

e.g. the voice signal, which is the most common message in telecommunications network, does not look similar to a pure cosine wave but it contains many cosine waves with different frequencies, amplitudes, and phases combined together.

The range of frequencies that is needed for a good enough quality of voice, is defined to be the range from 100 to 3,400 Hz. This means that the bandwidth of the telephone channel through the network is 3,400 – 100 Hz = 3.3 kHz.

Human voice contains much higher frequencies, but this bandwidth is defined as a compromise between quality and cost

Bandwidth, together with noise, is the major factor that determines the information-carrying capacity of a telecommunications channel.

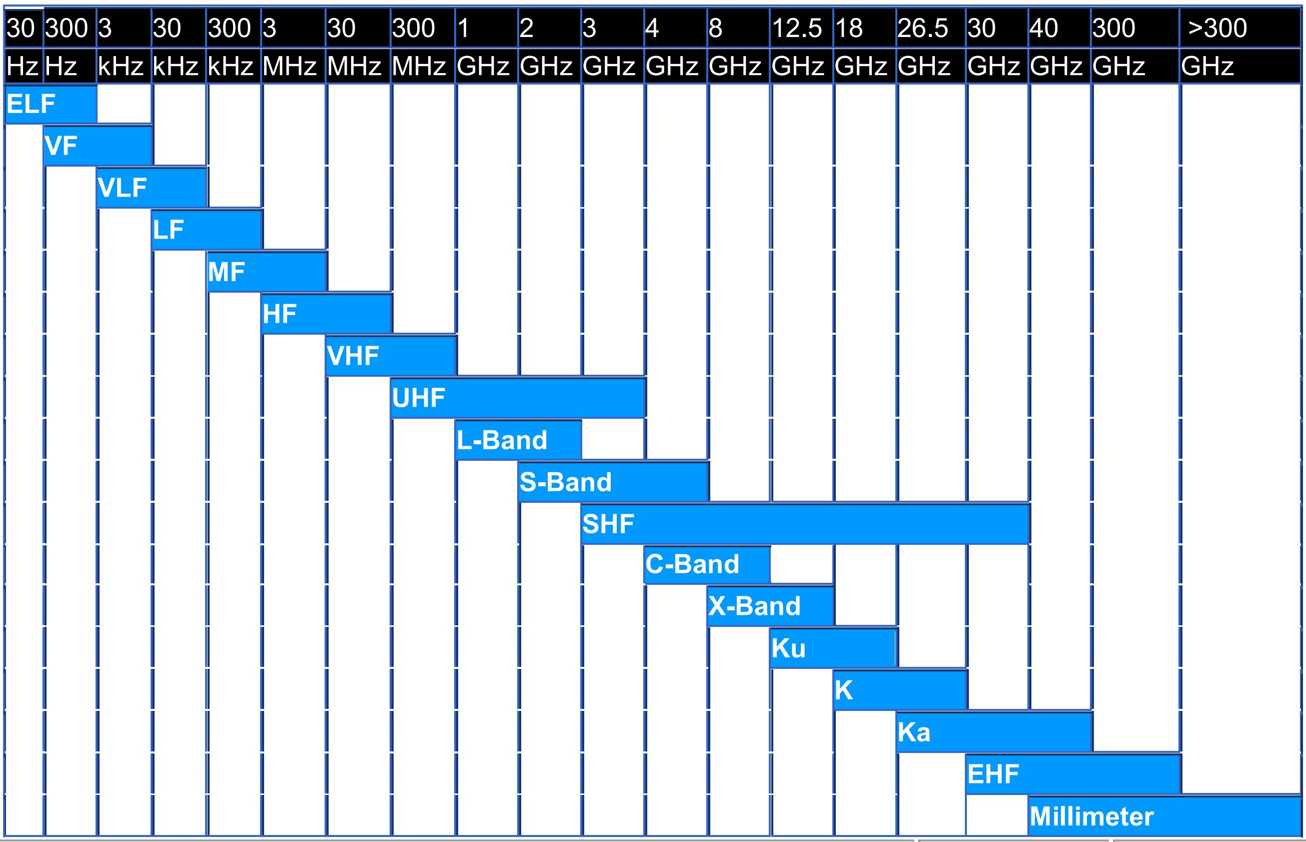
In analog signals, bandwidth is used and in digital signals, data rate is used where data rate is defined as the inverse of the duration of one digit.

(Sub)multiple	Prefix	Symbol	Name (US and Canada)
10^{24}	yotta	Y	Septillion
10^{21}	zetta	Z	Sextillion
10^{18}	exa	E	Quintillion
10^{15}	peta	P	Quadrillion
10^{12}	tera	T	Trillion
10^{9}	giga	G	Billion
10^{6}	mega	M	Million
10^{3}	kilo	k	Thousand
10^{2}	hecto	h	Hundred
10^{1}	deka or deca	da	Ten
10^{-1}	deci	d	Tenth

10^{-2}	centi	c	Hundredth
10^{-3}	milli	m	Thousandth
10^{-6}	micro	µ	Millionth
10^{-9}	nano	n	Billionth
10^{-12}	pico	p	Trillionth
10^{-15}	femto	f	Quadrillionth
10^{-18}	atto	a	Quintillionth
10^{-21}	zepto	z	Sextillionth
10^{-24}	yocto	y	Septillionth

Electromagnetic Spectrum

ELF = Extremely Low Frequency
VF = Voice Frequency
VLF = Very Low Frequency
LF = Low Frequency
MF = Medium Frequency
HF = High Frequency
VHF = Very High Frequency
UHF = Ultra High Frequency
SHF = Super High Frequency
EHF = Extremely High Frequency



	Wavelength	Frequency
Gamma rays	< 10 pm	>30.0 EHz
X-rays	< 10 nm	>30.0 PHz
Extreme UV	< 200 nm	>1.5 PHz

Near UV	< 380 nm	>789 THz
Visible	< 780 nm	>384 THz
Near IR	< 2.5 um	>120 THz
Mid IR	< 50 um	>6.00 THz
Far IR/submillimetre	< 1 mm	>300 GHz
Microwaves	< 100 mm	>3.0 GHz
Ultrahigh Frequency Radio (TV, Cellular Radio)	<1 m	>300 MHz
Very High Frequency Radio (TV, COAX), FM (88 - 108 MHz)	<10 m	>30 MHz
Shortwave Radio, (COAX from 1 MHz)	<180 m	>1.7 MHz
Medium Wave (AM) Radio, (Twisted Pair Up to 1 MHz)	<650 m	>650 kHz
Longwave Radio	<10 km	>30 kHz
Very Low Frequency Radio	>10 km	<30 kHz

The New Public Network

- End-to-end digitalization
  - Worldwide around 80% of the backbones are digitized
  - Local loop (Last mile between the subscriber and network) is currently around 93% analog. Great deal of effort needed to digitilize local loop. Without broadband access Internet and advanced applications can’t grow.
- Currently networks are electronic, in the future end-to-end optical or photonic networking is foreseen
- Intelligent programmable network: From anywhere in the world, any service or feature will be accessible irrespective of the connected network provider or network platform
- Very high bandwidth infrastructure
- Infrastructure offering multichannel service (one physical medium carrying multiple channels of different traffic)
- Low-latency network, i.e. networks with minimum delays. E.g. return time of 500 msec in satellite communications is disturbing. Today’s internet has up to 1 – 2 sec delay
- Agnostic network, i.e. should be able to follow multiprotocol. E.g. a box interfacing most used data protocols
- QoS (Quality of Service) guarantees: Meeting bandwidth, latency, loss requirements
- Encryption and security

Factors effecting the traffic

- Network capacity, signal and network bandwidth
- Tolerance for delays in the network (latency)
- Tolerance for the variations in the delay in the network (jitter)
- Tolerance for potential congestion, thus the loss of traffic in the network

Transmission Characteristics	Voice	Video	File Transfer	Interactive Media
Bandwith requirement	Low, fixed	Very high, fixed	High, variable	High, variable
Data loss tolerance	Tolerant	Tolerant	Nontolerant	Tolerant or nontolerant
Fixed delay tolerance	Low delay	Tolerant	Tolerant	Low delay
Variable delay tolerance	No	No	Tolerant	No
Peak information rate	Fixed	Fixed	High	Very high

<u>Backbone Bandwidth Requirements for Advanced Applications</u>

- Online virtual reality (e.g. life-size 3D holography; telepresence): 1 - 70 Petabits/sec (Peta=10^{15})
- Machine communications (e.g. smart utility communications, Web-agents, robots): 50 - 200 Petabits/sec
- High volume computing (e.g. 3-D computer aided design, weather forecasting): 50 - 200 Petabits/sec
- Residential applications after 100 Gbps broadband residential access is available: In the order of Exabits/sec (Exa=10^{18})

<u>Analog and Digital Transmission, Analog and Digital Modulation Schemes</u>

- Analog Transmission
  - Analog signal is continuous in amplitude and frequency
  - Natural voice and video signals are analog
  - Human voice is generated within the frequency range of 100 Hz – 10 KHz, i.e a bandwidth of 9.9 KHz.
  - Intelligible speech is in the frequency range of 250 Hz – 3.4 KHz. Total 4 KHz bandwidth is allotted for one analog voice transmission
  - In commercial TV broadcasting, bandwidth of the video signal is limited to 4.2 MHz. Allocated channel bandwidth for commercial TV is 6 MHz
  - Typical analog modulations are Amplitude Modulaion (AM) and Frequency Modulation (FM)
- Digital Transmission
  - Signal is represented by series of dicrete pulses (0's and 1's)
  - Computer output signals are digital
  - Bir rate determines the bandwidth. (e.g one voice channel carries 64 Kbps)
  - Very high speed data communications especially with fiber optic communications
  - Digital signal is much more easily reproduced as compared to analog signal. Regenerative repeaters does not only amplify the attenuated digital signal but also regenerates the degraded signal. Longer repeater distances can be used
  - Management of digital network is much more creative. Remote and/or central management, traffic statistics, performance measurements, management of different networks are possible through smart devices
  - Through encryption can be high security

Conversion between analog and digital signals

Some of the existing networks are neither all-analog nor all-digital, but a mixture of analog and digital. At the relevant points of such networks there is need for conversion from analog to digital or vice versa.

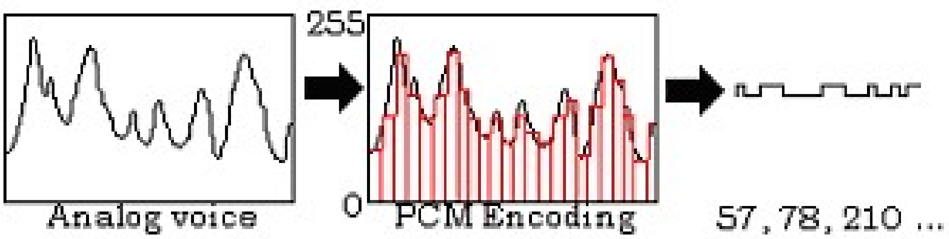
Analog-to-digital conversion for basic telephone voice signal

Steps involved in digitizing analog signals:

1. Sample the analog signal: The sampling rate must be twice the highest frequency (Nyquist rate) to produce playback that appears neither choppy nor too smooth.
2. Quantize the sample: Quantization consists of a scale made up of $2^8 = 256$ divisions
3. Encode the value into a binary expression: Encode the value into 8-bit digital form.
4. Compress the samples to reduce bandwidth, optional step. Compress the samples to reduce bandwidth.

Analog to Digital conversion

- Human voice is a continues signal in the range 0-4 KHz
- On the other hand digital communication is based on discrete bits (0 and 1)
- Thus, there is a need for converting the human voice into a stream of bits and vice versa
- Analog to digital conversion is done by sampling the sound wave and denoting the level of the wave by a number which is transmitted over the digital link
- Reverse process is done by creating a wave according to the received numbers
- According to Nyquist law, the minimum number of such wave samples needed for complete reconstruction of the wave is twice the number of the maximum frequency of that wave
- For voice signals, this yields: 2 x 4KHz = 8K Samples per second
- The most common method for denoting the level of the wave is called Pulse Code Modulation **(PCM)**
- In PCM, the level is divided into 256 levels (8 bits)
- Thus, if sampling is 8K times a second and each sample is in the range of 0-255, then per voice line 8K x 8 = 64K bits per second is obtained



Basic elements of Telecommunications

- Circuit: Physical path between two or more points, terminating on a port (an electrical or an optical interface)

  – Two-Wire Circuits: Has two electrical conductors. E.g. the ones used in analog local loop, i.e between the subscriber and the subscriber's first point of access into the network

  – Physical Four-Wire Circuits: Has two pairs of electrical conductors. e.g. connection between PSTN switches, leased lines, digital circuits

- Logical Four-Wire Circuits: Has two electrical conductors. Transmit and receive paths are formed by appling two frequencies

- Channel: Defines a logical conversation path. Channel is the frequency band, time slot or code over which a single traffic flows. Number of channels in a transmission line determines the number of simultaneous conversations.

- Line: A connection configured to support call generated by one user

- Trunk: A circuit configured to support call generated by a group of users. Trunk connects two switches.

- Transmission line is a specialized cable or other structure designed to conduct alternating current of radio frequency. Transmission lines are used for connecting radio transmitters and receivers with their antennas, distributing cable television signals, trunk lines routing calls between telephone switching centres, computer network connections and high speed computer data buses.

- Simplex: Signals can be passed in one direction only

- Half Duplex: Signals can be passed in either direction, but not in both simultaneously.

- Full Duplex: Signals can be passed in either direction in full speed, simultaneously. Full duplex operation on a two-wire line. This is accomplished by either FDM (frequency division multiplexing) in which the signals in the two directions occupy different frequency bands.

<u>Network Connection Types</u>

- Switched network connections: Dialup connection using a series of network switches

- Leased line network connections: Same two points are always connected, the transmission between these two always being on the same path.

- Dedicated network connections: Leased line connection where the end user may own the transmission equipments thus being exclusive to that user.

Circuit switching:

- Physical circuit is formed to establish the communication, and this physical circuit is released when the communication is over.
- Yields low latency (minimal delay) because the routing calculation of the path is made only once at the beginning of the call before the call is set up. After the set up is complete and the traffic starts to flow there are no more routing calculations to find the next hop.
- Bandwidth reserved for the circuit is not optimally utilized. In a real-time voice communication half of the time nothing is transmitted because of the breathing, pauses, etc during speech.
- Optimized for real-time voice traffic, thus guarantees Quality of Service (QoS). i.e., guarantees certain mximum delay, rate, tariff ..etc during the connection.

<u>Packet Switching</u>

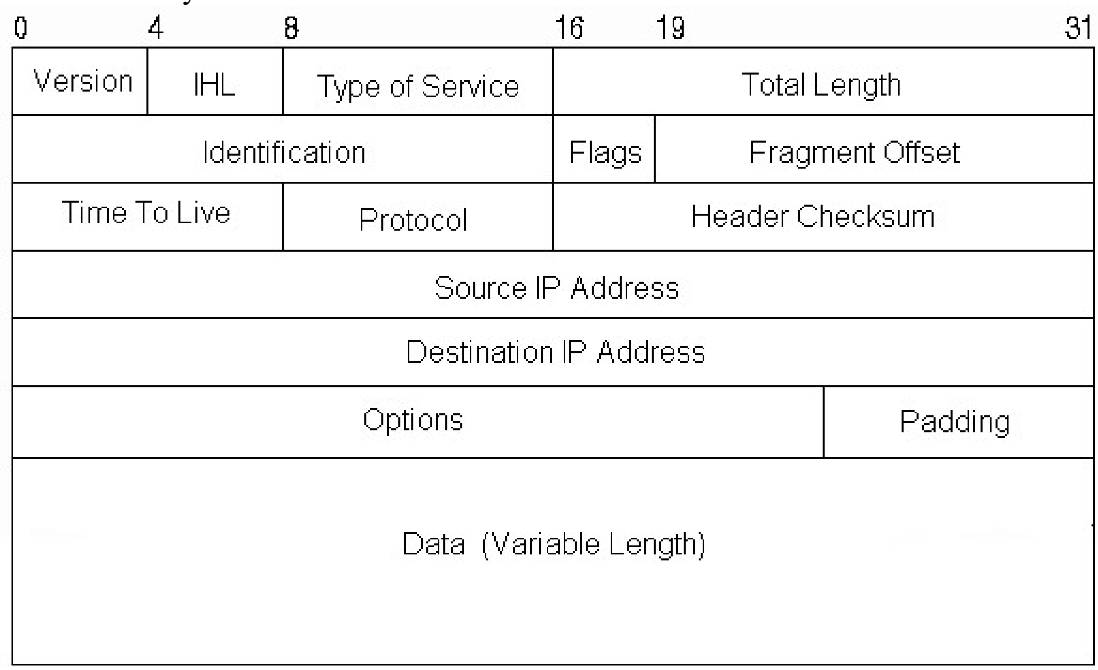
- A packet (or frame, block, cell or datagram)

  - Is a container carrying control and data bits.

- Control and data bits can each be in various sizes, i.e. can contain different number of bits
- Control bits (start, header, destination address, data sequence number, stop, ...etc) are used by the network nodes to route the packet under certain protocol (available bandwidth, existing noise, need for retransmission, latency considerations, ... etc).

e.g. **<u>IP Packet Structure (Ipv4)</u>**

- All IP packets are structured the same way:
  - An IP header and
  - Followed by a variable-length data field
- A summary of the contents of the internet header is as follows:



<u>Version</u> (4 bits): Indicates the format of the internet header. This document describes version 4.

<u>IHL (Internet Header Length)</u> (4 bits): Is the length of the internet header in 32 bit words, and thus points to the beginning of the data.

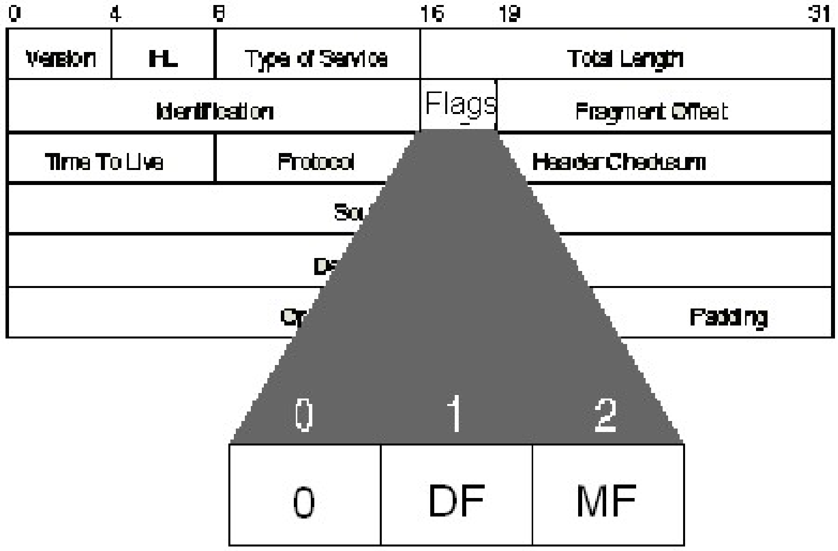
<u>Type of Service</u> (8 bits): Provides an indication of the abstract parameters of the quality of service desired. These parameters are to be used to guide the selection of the actual service parameters when transmitting a datagram through a particular network. Several networks offer service precedence, which somehow treats high precedence traffic as more important than other traffic (generally by accepting only traffic above a certain precedence at time of high load). The major choice is a three way tradeoff between low-delay, high-reliability, and high-throughput.

<u>Total Length</u> (16 bits): Is the length of the datagram, measured in bytes (octets), including internet header and data. This field allows the length of a datagram to be up to 65,535 bytes. Such long datagrams are impractical for most hosts and networks. All hosts must be prepared to accept datagrams of up to 576 bytes (whether they arrive whole or in fragments). It is recommended that hosts only send datagrams larger than 576 octets if they have assurance that the destination is prepared to accept the larger datagrams. The number 576 is selected to allow a reasonable sized data block to be transmitted in addition to the required header information. For example, this size allows a data block of 512 bytes plus 64 header bytes to fit in a datagram. The maximal internet header is 60 bytes, and a

typical internet header is 20 bytes, allowing a margin for headers of higher level protocols.

Identification (16 bits): An identifying value assigned by the sender to aid in assembling the fragments of a datagram.

Flags (3 bits): Various Control Flags.



Bit 0: reserved, must be zero

Bit 1: (DF) 0 = May Fragment, 1 = Don't Fragment.

Bit 2: (MF) 0 = Last Fragment, 1 = More Fragments.

Fragment Offset (13 bits): Indicates where in the datagram this fragment belongs. The fragment offset is measured in units of 8 bytes (64 bits). The first fragment has offset zero.

Time to Live (TTL) (8 bits): Indicates the maximum time the datagram is allowed to remain in the internet system. If this field contains the value zero, then the datagram must be destroyed. This field is modified in internet header processing. The time is measured in units of seconds, but since every module that processes a datagram must decrease the TTL (Time to Live) by at least one, even if it process the datagram in less than a second, the TTL must be thought of only as an upper bound on the time a datagram may exist. The intention is to cause undeliverable datagrams to be discarded, and to bound the maximum datagram lifetime.

Protocol (8 bits): Indicates the next level protocol used in the data portion of the internet datagram.

Header Checksum (16 bits): A checksum on the header only. Since some header fields change (e.g., time to live), this is recomputed and verified at each point that the internet header is processed.

Source Address (32 bits)

Destination Address (32 bits)

Options: Variable in length. May appear or not in datagrams. They must be implemented by all IP modules (host and gateways). What is optional is their transmission in any particular datagram, not their implementation. In some environments the security option may be required in all datagrams.

Padding: Variable in length. The internet header padding is used to ensure that the internet header ends on a 32 bit boundary. The padding is zero.

Data: Contains upper-layer information.

- Packets are stored-and-forwarded by packet switches up to the destination

- In packet switching, packets from many different sources are statistically multiplexed and sent to their destinations over virtual circuits

- Packet switches examine packet header and check destination against a routing table

- Same transmission lines are shared by multiple connections, i.e. packet switches or routers should do many more routing calculations

- Packets can be queued up at some nodes based on the availability of the virtual circuits

- Queuing causes latencies (delays)

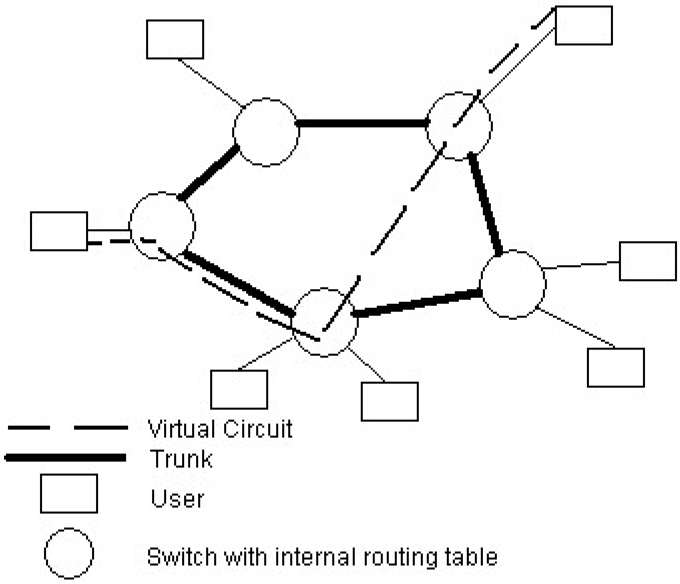
- Queues are realized through buffer storage

- If buffers are also full due to congestion, then packets (i.e information) can be lost

- In certain protocols (such as TCP) retransmission can be requested to replace the lost packets or packets received with unacceptable error performance

- In packet switching jitter occurs. Jitter means different delays among different two-switch transmissions, i.e., delay could be 30 msec between Switch 1 and 2 whereas 100 msec between Switch 2 and 3.

- Not strong QoS in connectionless packet switching



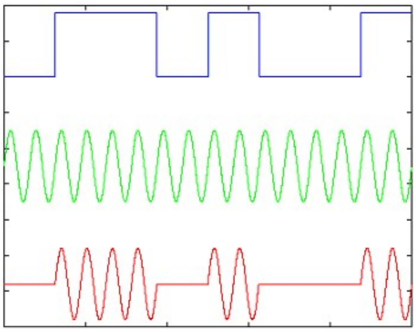
- PVC (Permanent Virtual Circuit): Virtual circuit available on a permanent basis. Manually configured by the network management system. Similar to leased line.

- SVC (Switched Virtual Circuits): Virtual circuit set up on demand. Provisioned dynamically by using signalling techniques. Must be reestablished each time data is to be sent and disappears after the data is sent. Similar to dialup connection in PSTN. Applicable when user is outside the physical location of the network (home, hotel etc.).

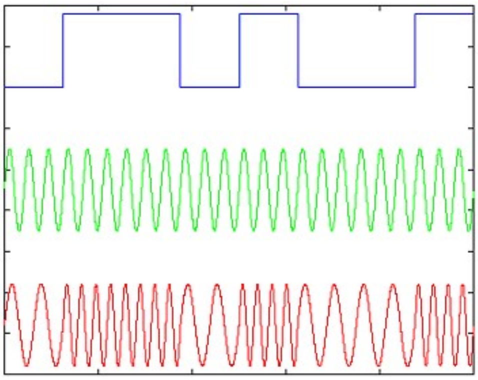
- Virtual Circuits: Series of logical connections between sending and receiving devices belonging to two hosts in a packet switching network. Connection is composed of a variety of different routes which are defined by table entries inside the switch. Connection is established after sending and receiving devices mutually agree on communication parameters like message size, path to be taken, error acknowledgements, flow control procedures, error control procedures.

Digital Modulation

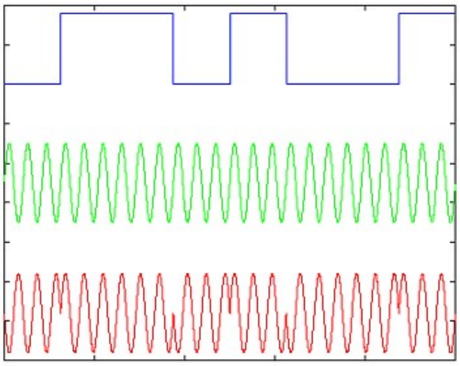
Amplitude Shift Keying (ASK)



Frequency Shift Keying (FSK).



Phase Shift Keying (PSK)



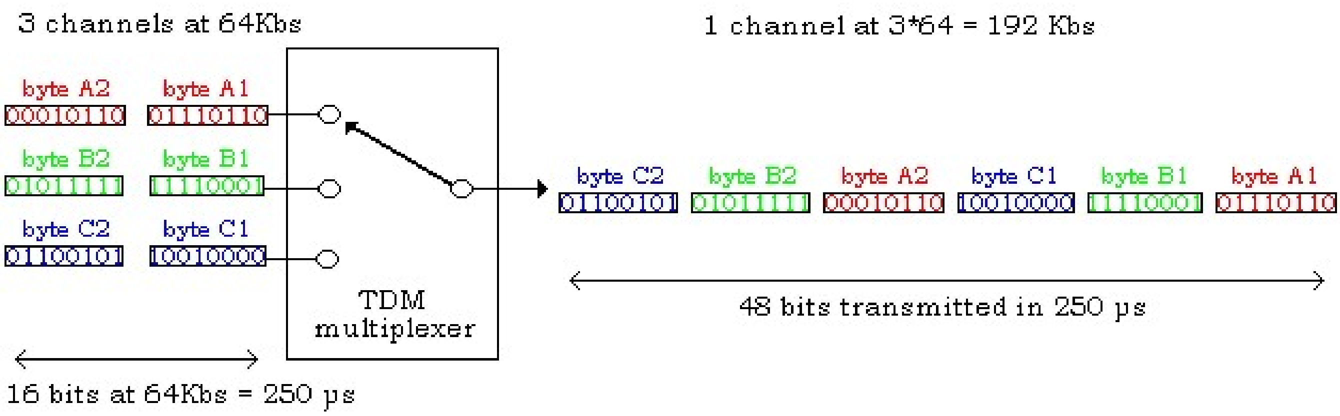
Multiplexing

- How to transfer data between two sites once there is a digital link between them?
- Multiplexing
  - There are two points to be settled:
    1. To be able to transmit more than just 64Kb/s
    2. The receiving end should know where in the bit stream is the beginning of a new 8 bit number.

- These two points are settled by multiplexing and the use of synchronization bits
- In order to transfer much more than a single channel between two sites, installing a separate line for every channel is clearly not a good solution
- Multiplexing is a way of sending many channels over a single line
- This is done by using Time Division Multiplexing (**TDM**)
- There are 32 channels, each with a rate of 64Kbs, that will be transferred to the other end
- The multiplexer takes from each of the 32 lines a single byte and sends them one after the other
- Then the multiplexer takes the next byte from every channel, and so on
- In order for the bytes not to be lost, the multiplexer must be able to send all the 32 x 8 bits from the 32 channels without the second byte of the first channel getting lost
- This implies that the output rate of the multiplexer should be at least 32 x 64Kbs or 2048 Kbs
- This method is called Time Division Multiplexing (TDM)
- In TDM, the multiplexer needs 1/8000 sec (i.e. 1/ (8K samples/sec)) for transferring a single byte of a single channel
- Then the multiplexer divides this between the 32 channels by increasing the rate so that each byte of a channel will take 1/(8000 * 32) sec to send

Example

To multiplex 3 channels of 64Kbs each:



- This method could be further used for increasing the number of channels from 32 channels to 4 x 32 channels and so on
- By each increase in the number of channels, bit rate of the line is increased accordingly
- After sending the 32 channels over a single line, then the question is how will the receiving end (the demultiplexer) know which bit belongs to which channel ?

- Synchronization

  - Special bits in the bit stream are used for synchronization

  - These bits tell the demultiplexer where a new 32 byte group starts so it will know how to divide the following bits between the channels

  - No synchronization is needed for distinguishing between each of the 32 channels

  - If we multiplex several 32 channels together, more synchronization bits are added for distinguishing between the different groups

- Digital data and digital Video

  - For transmitting digital data or digital video, no analog to digital conversion is needed.

  - Instead, the bit stream in directly inserted into the multiplexer

  - Video, which needs a much higher bit rate than 64Kbs is usually inserted directly into the second level multiplexer, thus allowing a bit rate of 1.5-2 Mbs

- Standards

  There are several standarts like

  - CEPT/E-Carrier mainly used in Europe and in Turkey

  - T-Carrier mainly used in USA and in some far-eastern countries.

  - J-Carrier used in Japan

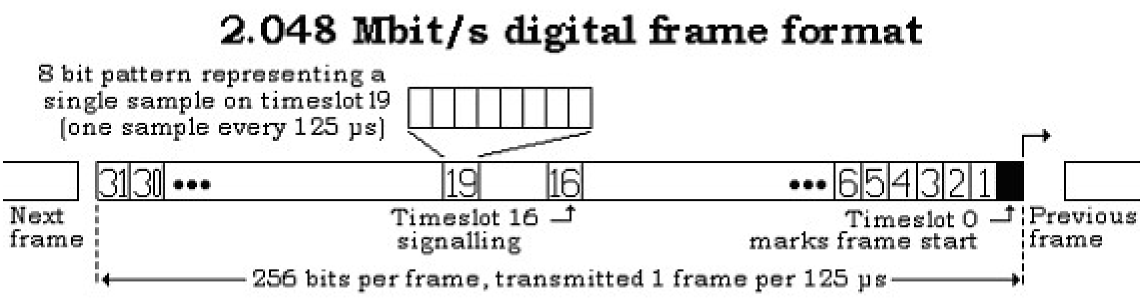
  Although all of the above standards start with a single channel rate being 64Kbs, those channels are still incompatible because of the different ways by which the voice was digitized

- CEPT/E1

  - The first hierarchy of E1 is composed of 32 channels totaling 32 x 64Kbs = 2048 Kbs

  - Two of the channels are not used for transmitting data but for frame synchronization and signaling

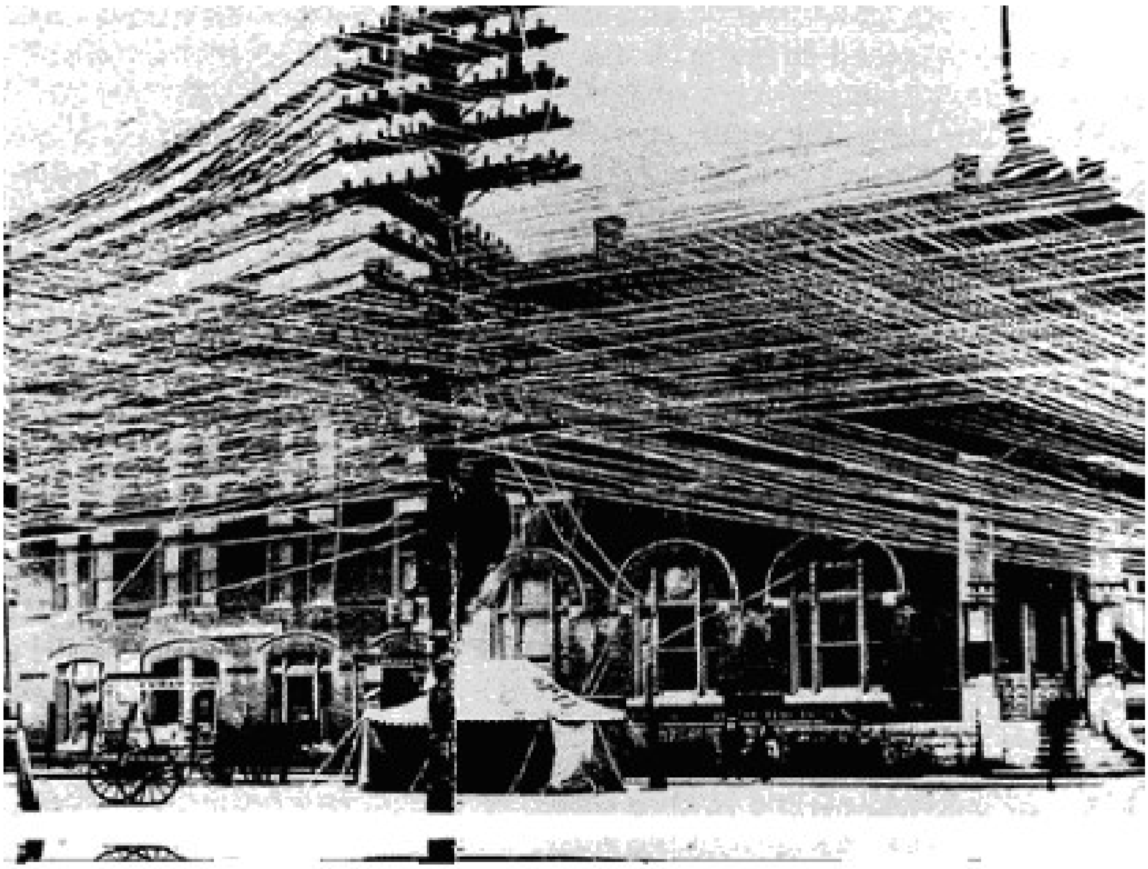
  -



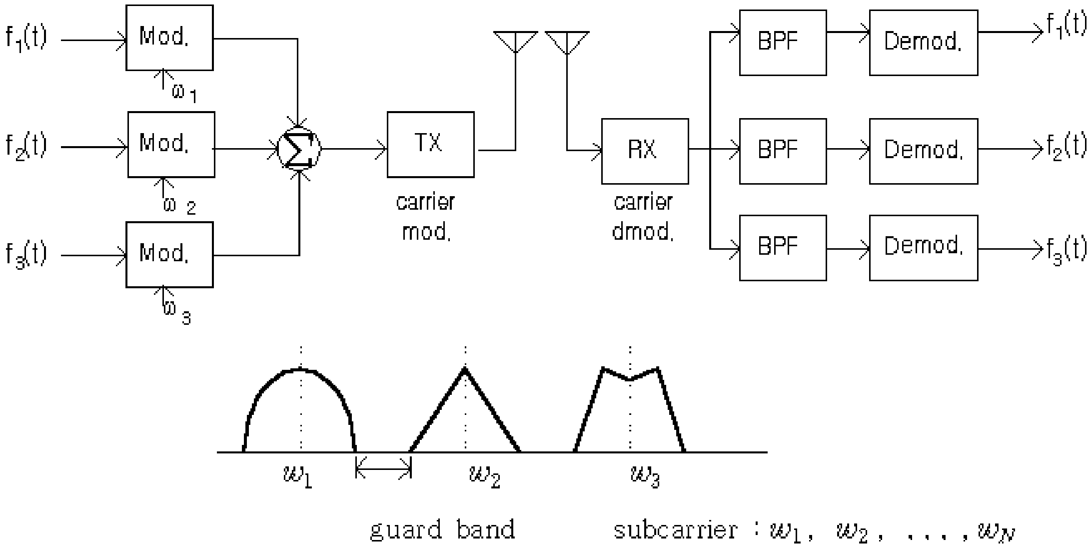
The hierarchies are presented in the following table:

	E-Carrier European (CEPT)	T-Carrier North American	J-Carrier Japanese
Level zero (Channel data rate)	64 kb/s	64 kb/s (DS0)	64 kb/s
First level (E-1, T-1, J-1)	2.048 Mb/s (30 user channels + 2 channels for synchronization and signalling)	1.544 Mb/s (DS1) (24 user channels + 8Kb/s for signalling)	1.544 Mb/s (24 user channels)
(Intermediate level, North American Hierarchy only)	-	3.162 Mb/s (DS1C) (48 Ch.)	-
Second level(E-2, T-2, J-2)	8.448 Mb/s (120 Ch.)	6.312 Mb/s (DS2) (96 Ch.)	6.312 Mb/s (96 Ch.)
Third level(E-3, T-3, J-3)	34.368 Mb/s (480 Ch.)	44.736 Mb/s (DS3) (672 Ch.)	32.064 Mb/s (480 Ch.)
Fourth level(E-4, T-4, J-4)	139.264 Mb/s (1920 Ch.)	274.176 Mb/s (DS4) (4032 Ch.)	97.728 Mb/s (1440 Ch.)
Fifth level(E-5, T-5, J-5)	565.148 Mb/s (7680 Ch.)	400.352 Mb/s (5760 Ch.)	397.200 Mb/s

Below is a photograph representing the situation before multiplexing is used.



- **Frequency Division Multiplexing**



Standards for Digital Data and Video (QoS)

- QoS provides *service differentiation* and *performance assurance* for Internet applications, i.e., provides a specification of how good the offered network services are.

- Service differentiation is a way for ISPs to obtain higher revenue.

- Internet is (slowly) evolving to support QoS.

QoS Parameters

- Service requirements are specified using QoS parameters:

  - End-to-end delay,
  - jitter,
  - packet rate,
  - burst,
  - throughput,
  - packet loss.

- Examples of QoS parameters:

  - Audio service (Sample rate of 8000 samples/sec, sample resolution of 8 bits/sample).
  - Network service (Throughput of 100 Mbps, connection setup time of 50ms).

Possible Audio QoS Parameters

- Application QoS:

  - Sample Size 8-bit Telephone voice quality. Sample Rate 8 KHz Intermediate delay 125 µs
  - 16-bit CD audio. 44.1 KHz Intermediate delay 22.7 µs.
  - Playback point ~100 to 150 ms, depending on the network delay

- Network QoS:

- End-to-end delay 0 to 150 ms Acceptable for most applications
- 150 to 400 ms, may impact some apps.
- > 400 ms, unacceptable
- Round-trip delay up to 800 ms, acceptable for conversation
- Packet loss $\leq 10^{-2}$ Telephone quality
- Bandwidth 16 Kbps Telephone speech
  32 Kbps Audio conference speech
  64 Kbps Near CD-quality audio
  128 Kbps CD-quality audio

Possible Video QoS Parameters:

- Application QoS:

  - Frame rate 30 fps (frames per second) NTSC format
  - 25 fps PAL format
  - 60 fps HDTC format
  - Frame width $\leq$ 720 pixels Video signal MPEG coded
  - Frame height $\leq$ 576 pixels Vertical size
  - Color resolution 8-bit or 16-bit/pixel Gray scale resolution of 256 or 65,536 colors
  - Aspect ratio 4:3 or 16:9 NTSC, PAL, TV format or HDTV format
  - Compression ratio 2:1 or 50:1 Lossy or lossless compression of HDTV

- Network QoS:

  - Bandwidth $\leq$ 1.86 Mbps MPEG encoded video
  - 64 Kbps to 2 Mbps H.261 encoded video
  - 1.544 Mbps to 2 Mbps H.120
  - 140 Mbps TV, PCM coding
  - > 1 Gbps HDTV uncompressed quality
  - ~ 500 Mbps HDTV lossless compression
  - 20 Mbps HDTV lossy compression
  - Bit error rate $\leq 10^{-6}$ Acceptable for conversation
  - Packet loss $\leq 10^{-2}$ Uncompressed video, $\leq 10^{-11}$ Compressed video
  - End-to-end delay ~ 250 ms Telephone speech

Components needed for QoS:

- Packet classification
- Isolation
- High resource utilization
- Admission control

Packet Classification

- Consider a phone application at 1 Mbps and an FTP application sharing a 1.5 Mbps link.

  - Bursts of FTP can congest the router and cause audio packets to be dropped.
  - Want to give priority to audio over FTP.

- Packet classification (marking) allows a router to distinguish among packets belonging to different classes of traffic.

Isolation

- Applications *misbehave* (audio sends packets at a rate higher than 1 Mbps assumed before). Need to provide protection (isolation) for one class from other classes.

- Need to regulate the rate at which a flow is allowed to inject packets into the network.

    - Policing Mechanism using leaky bucket.
    - Link-level packet scheduling.

Admission Control

- Cannot support traffic beyond link capacity.

- Need a Call Admission (admission control) process; application flow declares its needs, network may block call if it cannot satisfy the needs.

Scheduling Mechanisms

- Determines end-to-end delay => propagation delay + transmission delay + queuing delay

- How queued packets are selected for transmission is called Link Scheduling Discipline.

    - FIFO (First In First Out)
    - Priority Queuing
    - Round Robin
    - Weighted Fair Queing

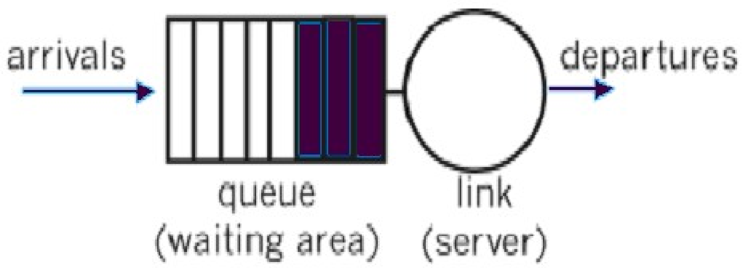
- Link scheduling discipline plays a crucial role in QoS:

FIFO (First In First Out)

- In-order of arrival to the queue; packets that arrive to a full buffer are either discarded (*tail drop*), or a discard policy is used to determine which packet to discard among the arrival and those already queued.

- Does not discriminate between different traffic sources (i.e., flows).

- Most widely used by today's Internet routers.



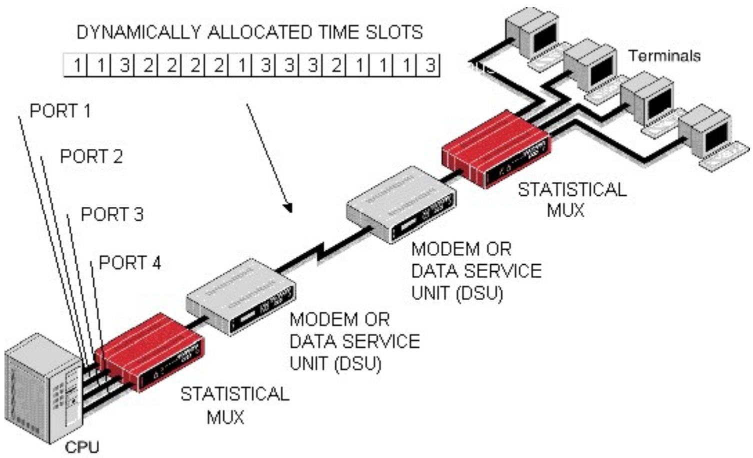
Priority Queuing

- Classes have different priorities; class may depend on explicit marking or other header info, e.g., IP source or destination, TCP port numbers, etc.

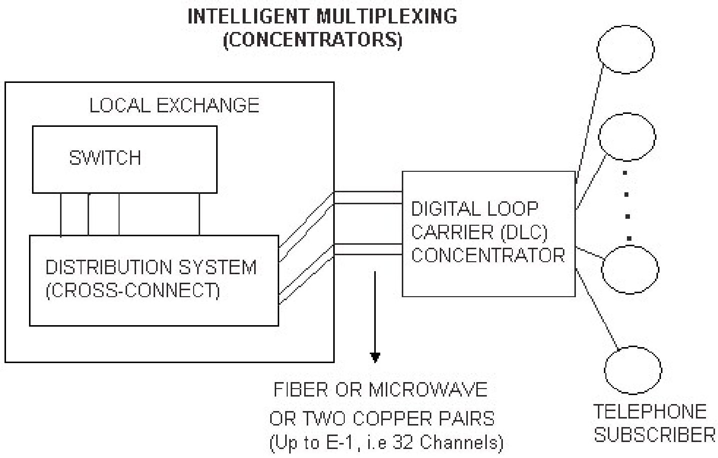
- Transmit a packet from the highest priority class with a non-empty queue.

Queue Management

- Controls packet loss.
- Packets get lost due to *damage* and *congestion*:
  - Loss due to damage is rare (<< 1%).
- Currently packets are dropped when queue is full using tail drop, drop front, random...
- Statistical Multiplexers
  - TDM is limited because the terminals cannot use each other's time slots
  - Statistical Time Division Multiplexers (STDM) dynamically allocate the time slots among the active terminals
  - In this way, bandwidth is used most efficiently, thus transmission is efficient
  - Is able to carry 2 – 5 times more traffic than TDM
  - Thus one can have more terminals than the available time slots
  - Statistical multiplexers are smarter and have more memory than other muxes
  - When all the time slots are busy, excess data goes into buffer
  - When buffer is full, additional access data gets lots
  - Thus traffic analysis must be made to ensure the performance parameters
  - Statistical multiplexers form the basis of packet switching technologies like X.25, IP, Frame Relay and ATM



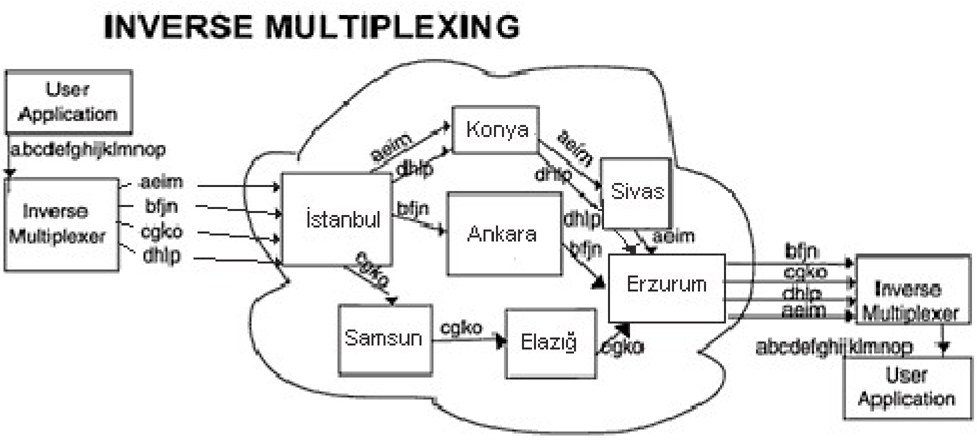
- Intelligent Multiplexers



- Intelligent Multiplexers are also known as **Concentrators**
- They concentrate (combine) large numbers of low speed lines to be carried over a high speed line to a further point in the network
- Digital Loop Carrier is an example for remote concentrator or remote terminal

- **Inverse Multiplexers**

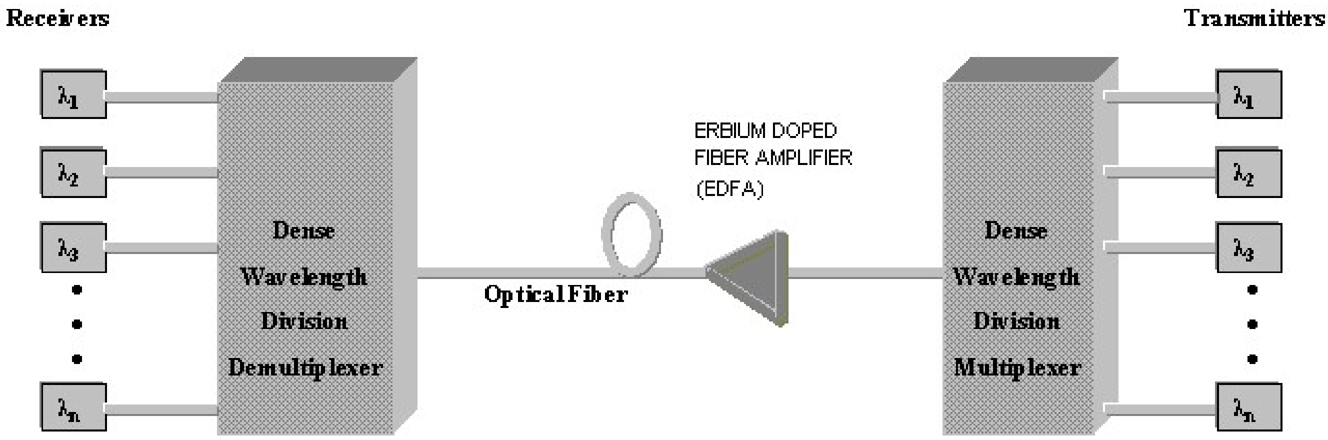
- Performs reverse operation as compared to multiplexers
- Thus, instead of combining many low-bit-rate streams to carry on a high bit-rate medium, inverse multiplexer:
    - Divides a high-speed serial data stream from a router or PC or other device into partial data streams of approximately 1.5 Mbps / 2Mbps each
    - Transmits these partial streams across separate T-1 / E-1 lines or N-ISDN or switched 64Kbps
    - Then recombines the partial streams into the exact original stream at the far end



    - Separate channels take diverse paths through the network and arrive at their destination at different times and not in order

- Inverse mux puts the packets back into proper order and adjusts the delays
- Via inverse multiplexing, high bandwidth applications which are not very frequently utilized (like videoconferencing once in a month) can be achieved without the need to pay a separate link for this use only

- **Wavelength Division Multiplexing/Dense WDM (WDM/DWDM)**



- Data from different sources are put together on an optical fiber, with each signal carried at the same time on its own separate light wavelength
- Using Wavelength Division Multiplexing (WDM) or Dense Wavelength Division Multiplexing (DWDM) more than 16 , realized up to 160 (and theoretically 15,000 channels pronounced) separate wavelengths or channels of data can be multiplexed into a lightstream transmitted on a single optical fiber
- Each channel carries a time division multiplexed (TDM) signal. In a system with each channel carrying 2.5 Gbps (30,720 telephone channels)
- Up to 160 x 2.5 Gbit/s (total of 4,915,000 telephone channels) is realizable by the same optical fiber
- If 15,000 channels can be realized, total of 460,800,000 telephone channels
- In dense WDM, wavelengths are closely spaced, commonly at intervals as small as 0.4 or 0.8 nm in the main telecommunications band near 1550 nm
- The International Telecommunications Union (ITU) has specified a grid of standard frequencies separated by increments of 100 GHz (approximately 0.8 nm), referenced to a frequency of 193.1 THz (corresponding to a wavelength of 1552.52 nm)
- These wavelengths are in the "conventional" or C band of the erbium-doped fiber amplifier (EDFA) at 1530 to 1570 nm
- Other bands of interest are the "long" or L band (approximately 1570 to 1610 nm) and the "short" or S band (approximately 1490 to 1530 nm)
- The importance of DWDM is for exploiting the enormous capacity of optical fiber to carry information

How a DWDM System Works

- Most DWDM systems support standard SONET/SDH short-reach optical interfaces to which any SONET/SDH compliant "client" device can attach

- Clients may be SONET/SDH terminals or add/drop multiplexers (ADMs), ATM switches, or IP routers

- Within the DWDM system a device called a transponder converts the SONET/SDH compliant optical signal from the client back to an electrical signal

- This electrical signal is then used to drive a WDM laser

- WDM laser is a very precise laser operating around the 1550-nm wavelength range

- Each transponder within the system converts its client's signal to a slightly different wavelength

- The wavelengths from all of the transponders in the system are then optically multiplexed onto a single fiber

- In the receive direction of the DWDM system, the reverse process takes place

- Individual wavelengths are filtered from the multiplexed fiber and fed to individual transponders, which convert the signal to electrical and drive a standard SONET/SDH interface to the client

Transmission Media

- It is the physical medium between the transmitter and receiver

- Transmission media make use of some form of electromagnetic energy, in the form of electricity, radio or light

- Types of transmission media are:

    - Twisted pair
    - Coaxial Cable
    - Microwave (Radio-Link)
    - Satellite
    - Optical Fiber

Twisted Pair Copper Cable

- Unshielded Twisted Pair (UTP)

    - Oldest and the most common transmission system

    - It comprises two thin copper wires, usually solid core, which are separately insulated and twisted around each other

    - Various categories (Cat 1 for voice only, Cat 2, ...,Cat 5 used in LAN applications operating over 100 MHz handling 100 Mbps over a range of 100 meters, ...Cat 6 expected to support 1 Gbps but only over short distances, Cat 7 expected to operate over 600 MHz)

- in UTP, as the conductor cross section increases, attenuation decreases. Attenuation is higher at higher frequencies

- Billions of miles of UTP installed, most especially in the local loop which is the circuit that connects the customer premises to the CO (Central Office) switch at the edge of the PSTN (Public Switched Telephone Network)

- UTP was originally installed for analog voice communications (4 KHz), but supports digital transmission as well (64 Kbps and higher bandwidth signals if properly deployed and conditioned

- Can support Digital Subscriber Line (DSL) and T-1/E-1

- UTP is also used in the LAN (Local Area Network) to connect terminals to hubs, switches and routers

- Inexpensive and easy to install and reconfigure

- Highly susceptible to EMI (Electromagnetic Interference), which is the source of crosstalk and other types of noise and signal distortion

- STP (Shielded Twisted Pair) and ScTP (Screened Twisted Pair)

  - STP has a metal foil, or shield, that surrounds each pair in a cable, sometimes with another shield surrounding all the pairs in a multi-pair cable

  - ScTP uses metal screen instead

  - Used to avoid crosstalk and EMI (Electromagnetic Interference)

  - Shields and screens block interference by absorbing it and conducting it to ground. İ.e. the foils and screens have to be spliced and there should be sound ground connection

  - STP and ScTP are a lot more expensive, and a lot more difficult to install

  - Developing high-speed LAN cabling standards for Cat 6 and Cat 7 are examples of this high-performance copper approach

Digital Subscriber Line (DSL) Technologies

- Digital applications of twisted pair in the local loop cover Narrowband ISDN and xDSL (HDSL, ADSL, IDSL, SDSL, RADSL, VDSL)

- Symetrical services provide the same rates both upstream and downstream

- Asymmetrical services have higher bit rate in downstream and lower bit rate upstream

- Some application parameters for DSL Technologies are tabulated below:

Abbreviation	Full Name	Downstream Rate (Rate from the Host to the Subscriber	Upstream Rate (Rate from the Subscriber to the Host)	Maximum Loop Length (i.e Subscriber Distance from the Digital Loop Carrier or from the Local Exchange)

N-ISDN, BRI	Narrowband Integrated Services Digital Network, Basic Rate Interface (Symetrical)	2B + D Channels (i.e.2 x 64+16 =144 Kbps) B (Bearer Channels for Telephone or Data), D (Delta Channel for signalling + low speed packet-switched data)	2B + D Channels (i.e.2 x 64+16 =144 Kbps)	5.5 km over single twisted pair
N-ISDN, PRI	Narrowband Integrated Services Digital Network, Primary Rate Interface (Symetrical)	30B + D Channels in Europian Standard (i.e.30 x 64+64 =2 Mbps)	30B + D Channels in Europian Standard (i.e.30 x 64+64 =2 Mbps)	3.5 km over two twisted pairs
HDSL	High-Bit-Rate Digital Subscriber Line (Symmetrical)	Up to 2 Mbps	Up to 2 Mbps	3.5 km
ADSL	Asymmetrical Digital Subscriber Line	Up to 6 - 8 Mbps	Up to 640 -840 Kbps	3.5 km
IDSL	ISDN DSL	128 Kbps (ISDN without voice service)	128 Kbps (ISDN without voice service)	5.5 km over single twisted pair
SDSL	Symmetrical Digital Subscriber Line	Up to 2.3 Mbps in multiples of 64 Kbps	Up to 2.3 Mbps in multiples of 64 Kbps	5.5 km over single twisted pair
RADSL	Rate Adaptive Digital Subscriber Line (Asymetrical)	Dynamically adapted data rate from 600 Kbps to 7 Mbps	Dynamically adapted data rate from 128 Kbps to 1 Mbps	5.5 km over single twisted pair
VDSL	Very-High-Bit-Rate Digital Subscriber Line	Up to 13 Mbps with 1.5 km Up to 52 Mbps with 300 m	Up to 1.5 - 2.3 Mbps	Over two twisted pairs 1.5 km for 13 Mbps 300 m for 52 Mbps

Coaxial Cable

- Formed by single thick solid core copper conductor surrounded by an insulator separating the center conductor from the outer shield of metal foil.

- That insulating material serves to separate the center conductor, over which the data is transmitted, from the shield

- Surrounding all of that often is a layer of metal mesh for protection, and then a cable sheath

- Thick center conductor supports high frequency signal

- Supports high frequency (1 GHz)

- Immune to Electromagnetic Interference (EMI).

- CATV systems traditionally uses coax to support signals as high as 500-750 MHz over considerable distances

- CATV signal is subdivided into frequency channels of 6 MHz for downstream TV transmission

- Interactive CATV systems also have channels of various widths for two-way data and even voice transmission

- Tradiditionally used in Ethernet and other LAN technologies, however today being replaced by data grade UTP

- Also used in Hybrid Fiber Coax (HFC) applications which uses fiber in the backbone and in the access network. From the access point (in the neighbourhood) to home, coax is used. HFC can support services like telephony, broadcast video and interactive services.

<u>Microwave (Radio Link)</u>

- Free-space systems

- Operates in the UHF (Ultra-High Frequency) up to the EHF (Extremely High Frequency) bands, which covers the range between 300 MHz and 300 GHz, current practice being mainly from 1 GHz up to 45 GHz

- Generally, point-to-point links

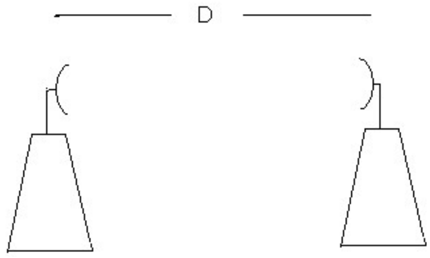
- Transmitter focuses (to overcome the spread) the radio beams over relatively long distances (around 50 km)

- Microwave signals being high frequency signals are severely impacted by atmospheric constituents like rain, fog, smog and haze between the transmit and receive antenna

- Line-of-sight is critical and dense physical objects like trees and mountains should be avoided

- Distance between the transmitting and receiving antenna towers (hop) decreases as the carrier frequency is increased. Hopping distance is around 70 km for up to 6 GHZ and around 8 km for 18GHz, 23 GHz, 45 GHz.

<u>Microwave System Design</u>



- <u>Free Space Loss</u> (FSL) in decibels (dB) is given by:

  FSL = 96.6 + 20 log D + 20 log F

  where F = frequency in GHz

  D = distance in miles

  e.g. **<u>LINK-1:</u>** 1-mile link at 5.825 GHz has a FSL of approximately

  FSL = 96.6 + 20log(1) + 20log(5.825) = 111.9 dB

  <u>LINK-2:</u> 1-mile link at 2.437 GHz has a FSL of approximately

  FSL = 96.6 + 20log(1) + 20log(2.437) = 104.3 dB

- Receiver Sensitivity Threshold

  The Receiver Sensitivity Threshold (R_x) defines the minimum signal strength required in order for a radio to successfully receive a signal

  A radio cannot receive or interpret a signal that is weaker than the receiver sensitivity threshold

  E.g. for LINK-1, Receiver Sensitivity Threshold is -77 dBm.

  for LINK-2, Receiver Sensitivity Threshold is -81 dBm

  where dBm is 10 log (received power/1 mwatt)

- Received Signal Level

  Received Signal Level (RSL) is the expected strength of a signal when it reaches the receiver. Receive Signal Level is defined as:

  $P_o - L_c t_x + G_a t_x - L_c r_x + G_a t_x - FSL = RSL$

  where P_o is the output power of the transmitter (in dBm)

  $L_c t_x$ is the cable loss between the transmitter and its antenna (in dB)

  $G_a t_x$ is the gain of the transmitter's antenna (in dBi)

  where dBi dB isotropic

  $L_c r_x$ is the cable loss between the receiver and its antenna (in dB)

  $G_a t_x$ is the gain of the receiver's antenna (in dBi)

  FSL is free space loss (in dB)

Example

Consider the 1-mile **LINK-1** in the above example where Free Space Loss (FSL) is 111.9 dB. Output power 1 dBm. For both transmitting and receiving antennas the gain is 26 dBi.

The RSL at the receiver is

1 dBm + 26 dBi + 26 dBi – 111.9 dB = -58.9 dBm

Example:

Consider the 1-mile **LINK-2** in the above example where Free Space Loss (FSL) is 104.3 dB dB. Output power +12 dBm. For both transmitting and receiving antennas the gain is 12 dBi. Both at the transmitter and at the receiver there is cabling with a loss of 1.5 dB.

The RSL at the receiver is

12 dBm - 1.5 dB + 12 dBi - 1.5 dB + 12 dBi - 104.3 dB = -71.3 dBm

Remark

RSL does not account for antenna alignment errors or path fading phenomena, such as multipath reflections, signal distortions, variable atmospheric conditions, and obstructions in the path.

- Link Feasibility Formula

To determine if a link is feasible, compare the calculated Receive Signal Level with the Receiver Sensitivity Threshold.

The link is theoretically feasible if $RSL \geq R_x$

If the Receive Signal Level ≥ Receiver Sensitivity Threshold, then the link may be feasible since the signal should be strong enough to be successfully interpreted by the receiver

In the above **LINK-1** Example, link is feasible since –58.9 dBm is greater than –77 dBm

In the above **LINK-2** Example, the link is feasible since -71.3 dBm is greater than -81 dBm.

- Fade Margin and Link Availability

Path fading weakens the radio signals

Fading occurs more frequently in flat, humid environments than in rough, dry places

Fade Margin = Unfaded Receive Signal Level - Receiver Sensitivity Threshold

Link must have sufficient Fade Margin to protect against path fading

Fade Margin is the link's insurance against unexpected system outages

Fade Margin is directly related to Link Availability, which is the percentage of time that the link is functional

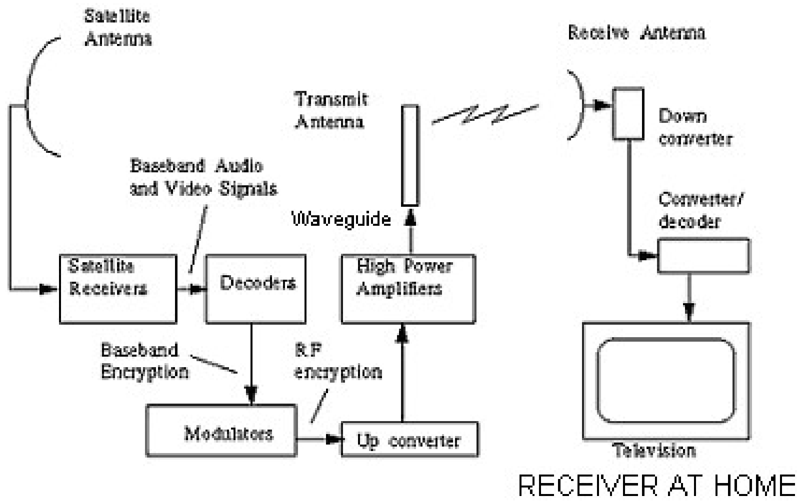
The percentage of time that the link is available increases as the Fade Margin increases

A link will experience fewer system outages with a greater Fade Margin

In the above **LINK-1** Example, Fade Margin is –58.9 – (-77) = 18.1 dB

In the above **LINK-2** Example, Fade Margin is –71.3 – (-81) = 9.7 dB

Multichannel Multipoint Distribution Services (MMDS)



MMDS is also known as cableless Cable-TV systems

TV Signals from satellite or other sources are received and retransmitted by microwave

Material to be delivered over MMDS are satellite, terrestrial and cable delivered programs, local baseband services.

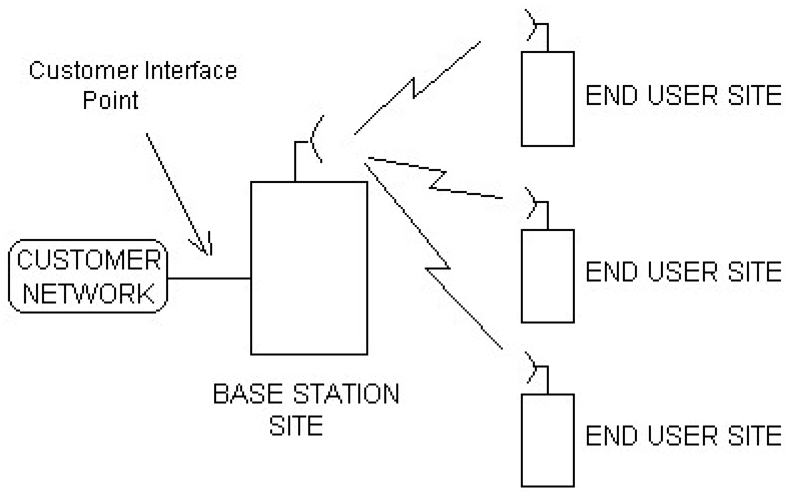
MMDS channels are transmitted from an omni-directional antenna (or doughnut pattern) which yields radiation equal in all directions in a chosen plane

Range is around 50 km

Only 200 MHz of spectrum (between 2.5 GHz and 2.7 GHz) is allocated for MMDS use

This means for TV signals with 6 MHz bandwidth, there are only 33 TV channels in MMDS

<u>Local Multipoint Distribution Services (LMDS)</u>



Deploying a fixed link for broadband network access to customers' premises is difficult and expensive

LMDS provides wireless broadband

LMDS consists of a radio transmitter which sends signals on a combination of channels to numerous receivers such as homes and businesses (i.e it is a point to multipoint system)

LMDS operates in various frequency bands, from 24GHz to 38GHz

Compared to MMDS which operates at lower frequencies (2.5 GHz) LMDS can have broader bandwidth but coverage is limited (around 5 km) and components are more expensive

Network coverage is increased by connecting the existing carrier network to a Base Transceiver Station (BTS) through a Customer Interface Point

This connection is extended, using high frequency radio transmission, to an antenna located at the customer's premises

i.e. LMDS provides wireless broadband connection between the carrier's network and its customers

LMDS applications

- LMDS provides digital two-way voice, high speed Internet access and data and video services
- LMDS offers the service providers and ISPs last mile connectivity between their fixed networks and customer sites
- LMDS connects LANs, intranets and PBXs of companies with distributed offices
- LMDS can provide 10 Mbps or faster connections which is attractive to customers who are using E1/T1 leased line connections between their LANs or to their ISP
- LMDS uses up to 622Mbps by allocating a large spectrum (100-112MHz) to a single subscriber or usually 10 Mbps for each subscriber in order to maximise the number of subscribers

LMDS link separation

Two ways of separating the uplink connection (from the subscriber to the base station) from the downlink connection (from the base station to the subscriber)

In Time Division Duplexing (TDD), the subscriber and the base station take turns talking to each other. At any time, both parties will use the entire spectrum allocated for that link

In Frequency Division Duplexing (FDD), the uplink and the downlink use different frequency bands separated by a large guard band (e.g. a separation of 1008MHz for the 24.5-26.5GHz band) to avoid interference

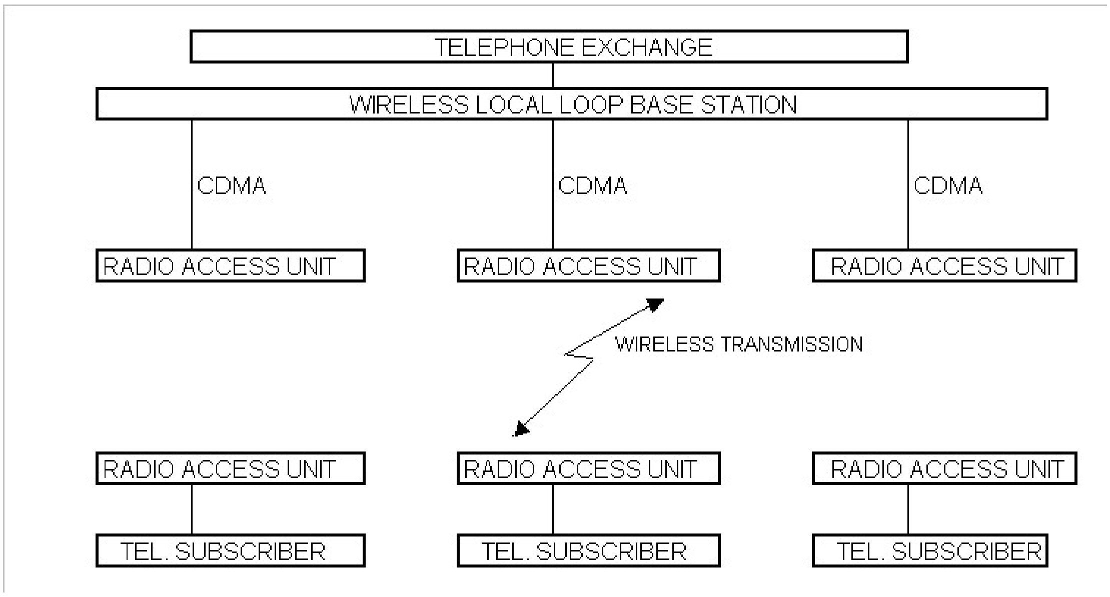
Since one base station needs to communicate with several sets of Consumer Premises Equipment (CPE), there is need to partition

- The uplink or the downlink frequency band (for the FDD case) among all the subscribers served by the base station
- The uplink or the downlink transmission duration (for the TDD case) among all the subscribers served by the base station

- In Frequency Division Multiple Access (FDMA), each CPE is allocated a small slice of the spectrum allocated to the uplink or downlink, and transmits simultaneously along with the other CPEs, i.e. different user transmissions are separated in frequency

- Time Division Multiple Access (TDMA) approach separates the transmissions to the various CPEs in time such that at any instance the base station communicates with only one CPE, i.e. different user transmissions are separated in time

Wireless Local Loop (WLL)



- Wireless Local Loop (WLL) system makes Public Switching Telephone Service possible in a wireless environment

- WLL can be based on CDMA and is connected directly to the telephone exchange

- Operates in wide range of frequency bands

- Covers an area of diameter bigger than 15 km

- Supports up to 56 kb/s modems or digital data rates of 64 kb/s or 128 kb/s

- Provides wireless Internet access

Wireless Local Area Networks (WLAN)

WLAN:

- Operates at 900 MHz or in the microwave range (2400 –2483.5 MHz, 5150-5250 MHz, 5470-5725 MHz)

- Data rates of 22Mbps , 54 Mbps

- Is an alternative to the traditional LANs based on twisted pair, coaxial cable, and optical fiber

- Is used for the same applications as wired or optical LAN

- Is more flexible because moving a wireless node is easier

- Is the best fit for portable computers
- Can be used in combination with cabled LANs

WLANs use three types of transmission techniques:

1. Spread Spectrum Technology

Currently the most widely used transmission technique for WLANs

In spread-spectrum more than essential bandwidth is used to achieve reliability and security

If a receiver is not tuned to the right frequency, a spread-spectrum signal looks like background noise

Two types of spread spectrum radio: frequency hopping and direct sequence

Direct-Sequence Spread Spectrum Technology (DS-SS)

Most wireless spread-spectrum LANs use DS-SS

Direct-sequence spread-spectrum (DS-SS) generates a redundant bit pattern for each bit to be transmitted

This bit pattern is called a code

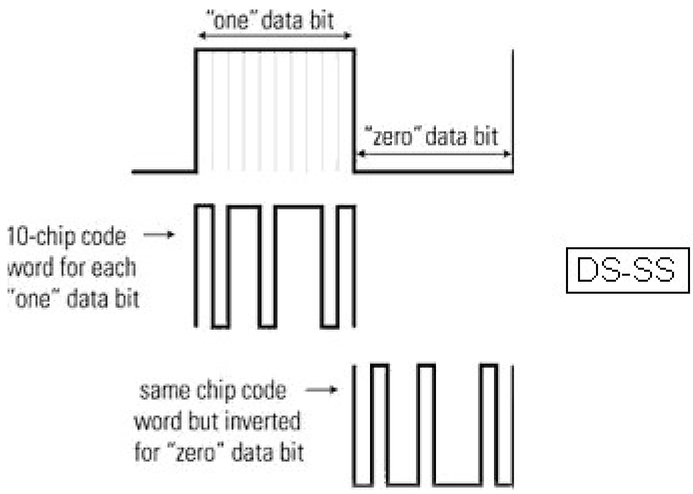
Each bit in this code is called a chip

Receiver should know the transmitter's spreading code to decipher data

This spreading code is what allows multiple direct sequence transmitters to operate in the same area without interference

Once the receiver has all of the data signal, it uses a correlator to remove the chips and bring the signal to its original length

To an unintended receiver, DS-SS appears as low-power wideband noise and is rejected (ignored) by most narrowband receivers



Frequency-Hopping Spread Spectrum (FH-SS)

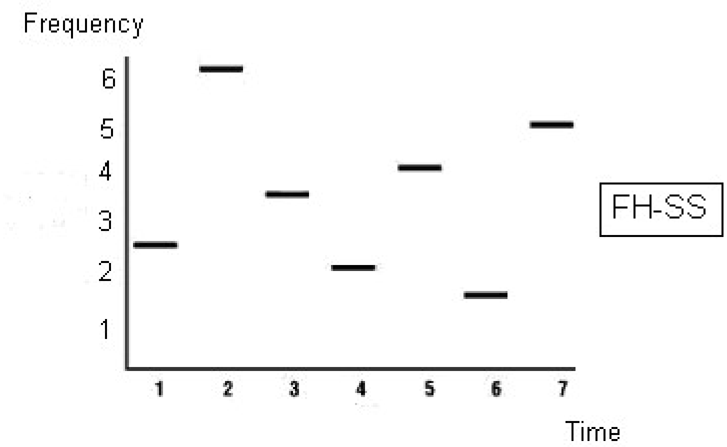
FH-SS uses a narrowband carrier that changes frequency in a code pattern known to both transmitter and receiver

A receiver, hopping between frequencies in synchronization with the transmitter, receives the message

The message can only be fully received if the series of frequencies are known

Since only the intended receiver knows the transmitter's hopping sequence, only that receiver can receive all the data

To an unintended receiver, FH-SS appears to be short-duration impulse noise.



Infrared WLAN (IR WLAN)

- Line-of-sight (LOS) or diffuse (or reflective) systems
- LOS are limited in range (a few meters)
- Diffuse IR WLAN does not require line-of-sight but their use is limited within a single room
- IR WLAN is high bandwidth
- Major disadvantage is that they can easily be obstructed

Comparison of WLAN Systems

Wireless LAN Transmission Techniques			
*	Spread Spectrum	Narrowband	Infrared
Frequency	902 - 928 MHz 2.4 -2.4385 GHz 5.725 - 5.825 GHz	18.825 - 19.205 GHz	3 x 10^{14} Hz
Maximum Coverage	30 - 250 meters	15 – 40 meters	10 m - 15 km
Line of sight	No	No	Yes/No
Transmit power	< 1 W	25 mW	1 – 800 mW
Maximum Rate	22 – 54 Mbps	22 – 54 Mbps	1.5 - 622 Mbps

Satellite Systems

- Microwave but not terrestrial

- In some cases satellites can operate in the same frequency range as terrestrial systems

- GEO (Geosynchronous Earth-Orbiting) satellites are positioned directly above the equator at altitudes of 35,786.1 km. GEOs maintain their positions relative to the Earth's surface. Orbital travel is in east-west direction.

- GEOs are used for communication and weather forecast

- LEO (Low Earth-Orbiting) satellites have altitudes of 320 - 800 - 1500 kilometres and mainly used in Remote Sensing applications.

- LEOs have polar orbits (north-south direction, descending from north-south, ascending from south-north), with orbital speed of LEO satellites are 27,359 kilometres per hour. They can circle Earth in about 90 minutes.

- MEOs (Middle Earth-Orbiting) are at at altitudes of 10.000 - 15.000 km.

- LEOs and MEOs do not maintain their relative positions.

- Satellites can transmit to, and receive from, a large area (foot print or coverage), thus advantageous in point-to-multipoint and broadcast applications.

- Thousands of satellites exist in space among which around 500 of them are communication based satellites (mainly GEOs)

- Similar to microwave systems, their performance varies with the weather condition.

- Propagation delay is quite important in satellite communications. 0.25 second delay from the transmitter to the receiver on earth, i.e 0.5 sec delay between the times when one communication point says ‘Hello’ and hears the response ‘Hello’ from the other communication point. For voice, videophone and some data applications (like games) this amount of delay is disturbing.

- Transponder in the satellite includes:

  - The receiving antenna to pick-up signals from the ground station

  - Broadband receiver

  - Input multiplexer and a frequency converter which is used to reroute the received signals through a high powered amplifier for downlink

- Telecommunication satellites receive signals from a ground station and send them down to another ground station located at a very long distance from the first (Relay action)

- In the case of a long distance phone call or data transmission, communication can be two-way

- In the case of television broadcasts, the ground station's uplink is then downlinked over a wide region

- Another application is in remote sensing where the satellite (equipped with cameras or various optical sensors). In this case the satellite only downlinks data which is sensed from Earth's surface Atmosphere.

Ground Station

- In the uplink or transmitting station, terrestrial data in the form of baseband signals is sent to the orbiting satellite by passing through:

  - Baseband processor
  - Up converter (frequency conversion)
  - High powered amplifier
  - Parabolic dish antenna

- In the downlink, or receiving station, operation is reversed as compared to uplink.

Applications of Satellite Communications

- Long distance telephone network among countries: International satellite consortium (Intelsat)

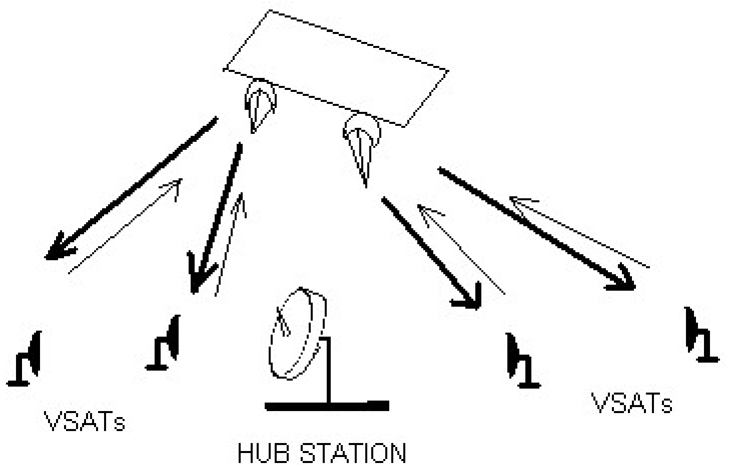
- Television Broadcasting (Analog and digital):

  - Direct free reception by home dishes (Free or scrambled channels)

  - Terrestrial distribution after the satellite reception at the ground station

- Automotive Navigation: Inmarsat applications as Global Positioning System GPS, Vehicle Tracking in a Fleet, Land Navigation as Maps in Cars, Maritime applications

- VSAT (Very Small Aperture Terminal) Networks



- Receive/transmit home/office small antenna aperture terminals connecting to a central hub via satellite

- Antenna dishe diameter around 0.6 - 3.8 meter

- Operates in the Ku-band (around 14 GHz uplink, 11 GHz downlink) and C-band (around 6 GHz uplink, 4 GHz downlink) frequencies

- Ku-band requires smaller antenna diameter than C-band
- Can have Bi-Directional Operation (uplink and downlink) or Receive-Only Operation (downlink)
- Multipoint network providing two-way data, voice and multi-media transmission
- Can provide internet downloads at up to 2 Mbps
- Star-network that connects one or more main sites to various remote sites
- Uses TDMS (Time Division Multiple Access) as the means to send data to each remote site
- A chart is given below for various satellite applications

	L-Band (390 - 1550 MHz)	800 MHz range	2 GHz range	C-Band (Uplink around 6 GHz, Downlink around 4 GHz)	Ku-Band (Uplink around 14 GHz, Downlink around 11 GHz)	Ka-Band (Uplink around 30 GHz, Downlink around 20 GHz)
GEO (Geosynchronous Earth-Orbiting) satellites above the equator at altitudes of 35,786.1 km altitude	VSAT			Commercial TV Broadcast, Digital Radio, VSAT	Up to 155 Mbps global area data communications, Most of the Current Commercial Analog/Digital TV Broadcast, Digital Radio, VSAT	Up to 155 Mbps data, Multimedia, Military Applications, High speed internet access, tele-education, tele-medicine, ATM based services including wide area networks and local area network interconnection, video-telephony, cheap videoconferencing, television services to aircraft, local television, rural telephony, satellite newsgathering, alternative to VSAT networks, remote monitoring
MEO (Middle Earth-Orbiting) satellites at altitudes of 10.000 - 15.000 km			Voice (Cellular)	Mobile	Up to 155 Mbps regional area data communications, Analog/Digital TV Broadcast, Digital Radio	
LEO (Low Earth-Orbiting) satellites have altitudes of 320 - 800 - 1500 kilometres		Little LEOs (Below 1 GHz) 2.4 - 300 Kbps. Messaging, paging, vehicle location	Big LEOs (Above 1 GHz) 2.4 - 9.6 Kbps. Voice (Cellular)	Mobile		Broadband LEOs 16 - 155 Mbps Data, Multimedia

Fiber Optics

Critical angle

According to Snell's Law:

$$n_1 \sin\theta_1 = n_2 \sin\theta_2$$

θ_1 and θ_2 are angle of incidences. The angle of incidence is measured with respect to the normal at the refractive boundary. n_2 is the refractive index of the less optically dense medium, and n_1 is the refractive index of the more optically dense medium.

*The **critical angle*** is the angle of incidence above which total internal reflection occurs. The critical angle θ_c is given by:

$$n_1 \sin\theta_1 = n_2 \sin(\pi/2) = n_2$$

i.e., $n_1 \sin\theta_c = n_2$

$$\theta_c = \arcsin(n_2 / n_1)$$

If the incident ray is precisely at the critical angle, the refracted ray is tangent to the boundary at the point of incidence.

If for example, visible light were traveling through glass (with an index of refraction of 1.50) into air (with an index of refraction of 1.00). The calculation would give the critical angle for light from acrylic into air, which is

$$\theta_c = \arcsin(1/1.5) = 41.8^0$$

Light incident on the border with an angle less than 41.8° would be partially transmitted, while light incident on the border at larger angles with respect to normal would be totally internally reflected.

- Any optical communications system can be studied in three main parts:

    1. Transmitter which converts information to light
    2. Medium (i.e. fiber optic cable or atmosphere) which transmits the light signal
    3. Receiver which converts the light signal into an electrical signal.

- Light Source is either a semiconductor Light Emitting Diode (LED) or a semiconductor Laser Diode

- LED or Laser Diode receives a modulated electrical signal and converts it into a light signal

- Light signal is coupled into the fiber optic cable

- Light sources emit light at wavelengths of 850, 1300 or 1550 nanometers

LED's

- Common and relatively inexpensive

- Usually low power, thus used in multimode applications for short distances

- Are used in low rate transmission because dispersion is high due to wide spectral widths (36 – 40 nanometers)

Laser diodes

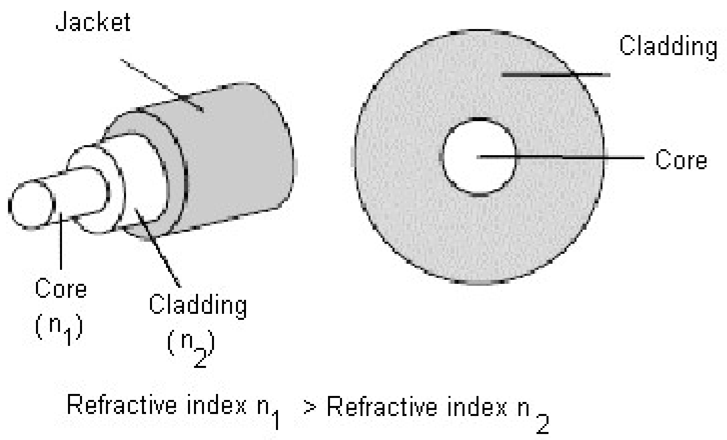
- Usually more expensive than LED's

- Can be high power, thus used in singlemode fibers for long haul communication links

- Are used in very high bit rate transmission (10 Gbps *160 in DWDM) because dispersion is very low due to narrow spectral widths (less than 1 nanometer)

Fiber Optic

- Fiber consists of an inner core, outer cladding and a protective buffer coating



- Core is the glass (SiO_2) area through which light travels and the information is carried

- Surrounding the core is the cladding which is also of glass but with a lower refractive index than the core

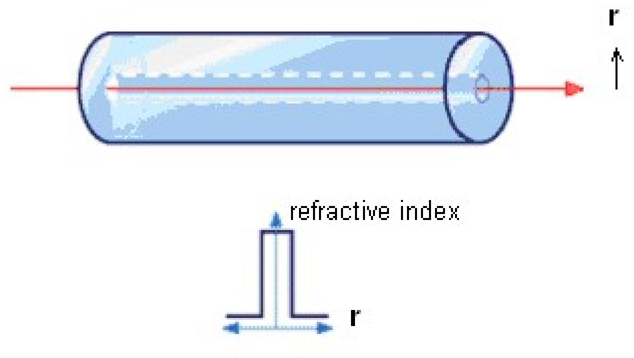
- The lower refractive index causes the light to be totally reflected in the core, thus staying in the core until the receiver

- To protect the fiber core and the cladding, several layers of plastic coatings (250 microns - 900 microns) are applied to preserve strength

- Fibers are classified as singlemode or multimode

Singlemode Fiber

- Core (9 micron diameter) is very small compared with cladding(125 micron diameter)

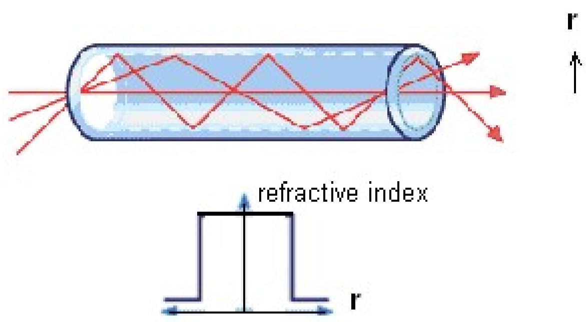


Singlemode Fiber

- Because of small core, light in the core travels in a straight line (i.e single mode)
- Has very high bandwidth
- Wavelength of 1310 nanometer is best for dispersion (pulse broadening)
- Wwavelegth of 1550 nanometer is best for attenuation
- For singlemode transmission, repeater distance required in practice is around 50 –100 km. In some systems 1Gbps is announced for repeaterless links of 20.000 km

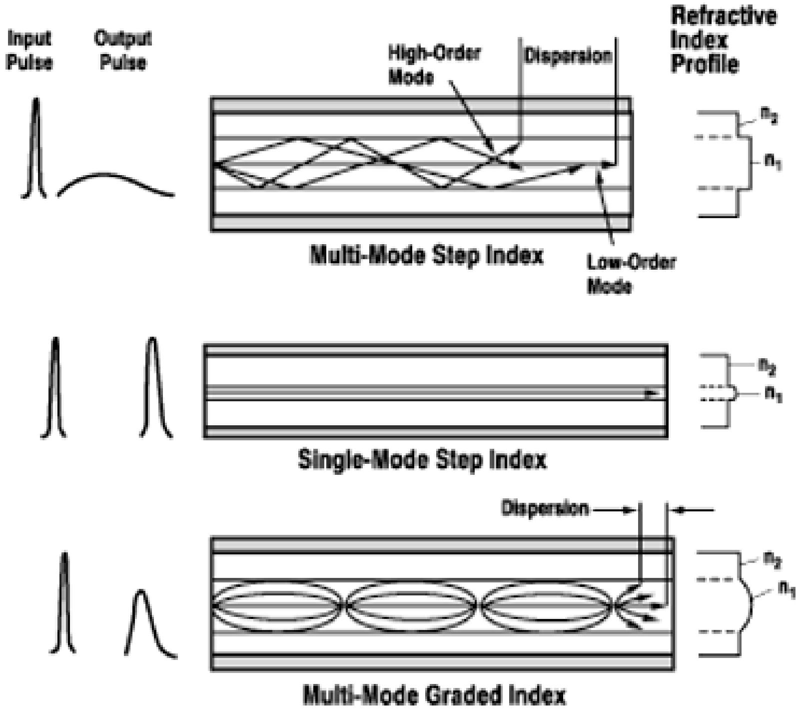
<u>Multimode Fiber</u>

- Has a much larger core (50/125, 62.5/125 and 100/140 micron)
- Used in LAN applications
- Since the core diameter is large, light travels in multiple paths, or as multimode
- Rates are relatively small, however can be up to 200 Mbs for distances less than 100 meters
- Manufactured as step index or graded index
- Step index has a slight step difference in-between the refractive index of the cladding as compared with the core. Each individual mode (or ray of light) takes a different path. When the signal reaches its destination, different light waves arrive at the receiver at different times

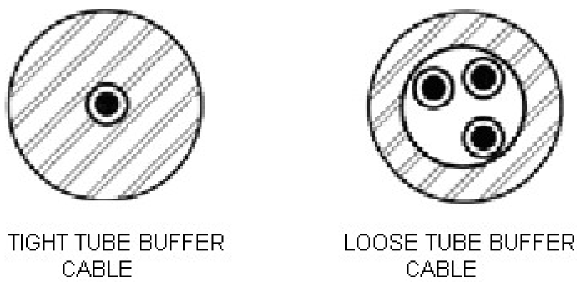


Step index Multimode Fiber

- To compensate for this problem, a graded index fiber is developed. Many layers of glass, each with a lower refractive index are applied to make the fiber core. Faster light rays traveling in the outer layers travel longer path than the slower light rays travelling in the inner layers. In this way, all light waves arrive at the receiver at the same time.



- Some cable types are shown below:



<u>Receivers</u>

- At the receiver, a semiconductor photo-diode converts the incoming light signal back into a modulated electrical signal which is then demodulated electrically

- Receiver wavelength must be the same as the transmitter

- System perfrormance is measured in Bit Error Rate (BER) for digital systems or Signal to Noise Ratio (SNR) for analog systems

- Sensitivity of the detector is the minimum power that must be received on an incoming signal.

- Saturation defines the maximum received power that can be accepted. If too much power is received, the result is a distortion of the signal, causing poor performance

Optical Fiber Communications Link Design

→ Will discuss point-to-point link

Procedure

→ Examine the components available for the particular application. The components to choose are:

1. LED or laser diode optical source
   - emission wavelength
   - spectral line width
   - optical output power
   - effective radiating area
   - emission pattern
   - modulation rate
   - life time
2. Multimode (step or graded index) or single-mode optical fiber
   - Core radius
   - Core refractive-index profile
   - Bandwidth (Dispersion limitation)
   - Attenuation
   - Numerical Aperture
3. Pin or avalanche photodiode
   - Responsivity
   - Operating wavelength
   - Speed (response time)
   - Sensitivity (NEP)

→ See how these component relate to the system performance criteria. System performance requirements are

1. The desired (or possible) transmission distance
2. The data rate or channel bandwidth
3. Bit error rate (BER)

→ For the given set of components and system performance

— Carry out a power budget analysis to

determine whether the link meets the attenuation requirements or if repeaters are needed.

— Carry out a system rise time analysis to verify that the overall system performance requirements are met.

— If performance requirements are not met, change appropriate components and repeat the same procedure.

— Try several designs to find the cheapest design.

Design Considerations

→ Choose the operating wavelength
- If path length is not long choose 0.8 – 0.9 μm range
- If " " " long choose 1.55 μm for low data rates
- " " " " " " 1.3 μm " high " ".

→ Choose the characteristics of two of the main building blocks (transmitter, fiber or receiver) and calculate the characteristics of the remaining third one. Check whether the system requirements are met.

Choosing Photodetector

— Determine the minimum optical power received by the photodetector to satisfy the required bit error rate (BER)

— Keep in mind that:

pin photodiode receiver is
→ simpler (cheaper)
→ more stable with temperature changes
→ smaller bias voltages.

avalanche photodiodes have → higher sensitivity (i.e. can detect lower optical power thus increasing the repeater spacing)

Choosing the Light source

— Parameters to be taken into consideration are: Data rate, transmission distance, cost, dispersion.

— Keep in mind that

LED → is cheaper
→ no feedback crkt. is needed for temperature stabilization
→ has longer life time

laser → has narrower spectral width
→ has higher coupling efficiency (can couple 10–15 dB more power than LED)
→ is more coherent
→ has higher modulation rate

LED at 0.8–0.9 μm	→	150 (Mb/s).km
LED at 1.3 μm	→	> 1.5 (Gb/s) km.
laser at 0.8–0.9	→	2.5 (Gb/s) km
laser " 1.3 μm	→	> 25 (Gb/s) km.

Choosing the Optical Fiber

— Depends on the type of the light source employed and on the maximum amount of dispersion that can be tolerated.

— Check the attenuation characteristics of the fiber and the cabling.

— Check the splice, connector and other losses.

— Keep in mind that

single mode fiber → can be used with laser
→ yields the maximum bit rate-distance product.
→ has more difficult splicing.

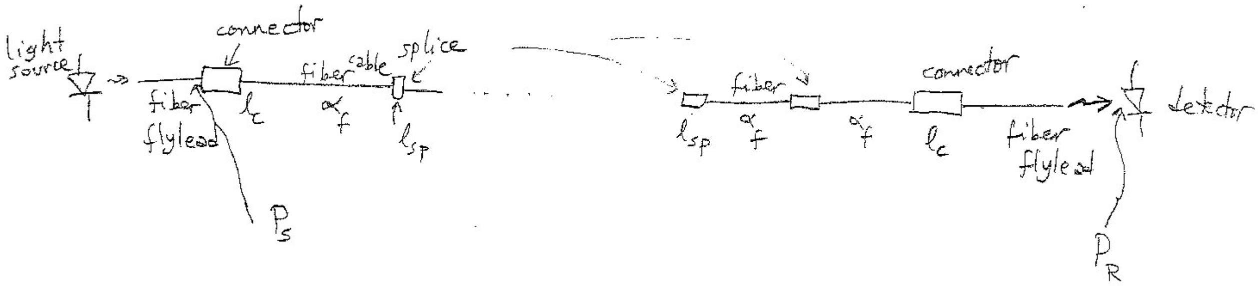
step index multimode fiber → can be used with LED and laser
→ cheaper
→ good for low bit rate-distance products.
→ easier splicing.

graded index fiber → can handle higher bit-rate-distance products as compared to step-index multimode fiber, yet much easier splicing as compared to single mode fiber.

Link Power Budget

→ Loss in an element $= 10 \log \dfrac{P_{out}}{P_{in}}$ (dB)

where P_{out} = output optical power of that specific element
P_{in} = input " " " " " "



$$P_T = P_S - P_R$$

P_T: total optical power loss

P_S: optical power at the end of the fiber flylead attached to source

P_R: receiver sensitivity

$$P_T = 2\,l_c + \alpha_f L + \text{system margin}$$

l_c: connector loss

α_f: fiber attenuation (dB/km)

L: transmission distance

system margin: ~6 – 8 dB due to component aging, temp. fluctuation ... etc.

Rise Time Budget

→ Performed to determine the dispersion limitation of the fiber link.

$$t_{sys} = \left(t_{tx}^2 + t_{mat}^2 + t_{mod}^2 + t_{rx}^2 \right)^{1/2}$$

t_{sys}: total rise time of the link

t_{tx} is the rise time of the transmitter

t_{mat} is the material dispersion rise time of the fiber

t_{mod} is the modal dispersion rise time

t_{rx} is the receiver rise time.

→ t_{tx} results from the rise time of the source and its drive circuitry.

→ t_{tx} can be estimated from Eq. 4-28 of Keiser's book.

→ $t_{mat} = D_{mat} \, \sigma_\lambda \, L$

D_{mat}: material dispersion factor in (n sec/nm.km) (can be found from Fig 3-13 of Keiser)

σ_λ: spectral width of the optical source

L: transmission distance

→ $t_{mod} = \dfrac{0.44}{B_M} = \dfrac{0.44 \, L^q}{B_0}$

q = 0.5 for steady state modal equilibrium
= 1 for little mode mixing
= 0.7 practical value

Here $B_M = \dfrac{B_0}{L^q}$

empirical formula

B_M: bandwidth in a link of length L

B_0: Bandwidth in 1 km. of cable

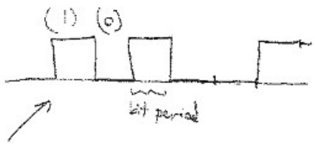
→ $t_{mod} = \dfrac{440 \, L^q}{B_0}$ (L in km)

t_{mod} in nsec, B_0 in (MHz)

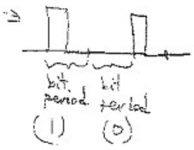
→ t_{rx} result from the photodetector rise time and the 3-dB electric bandwidth of the receiver front end. (B_{rx})

→ $t_{rx} = \dfrac{350}{B_{rx}}$ ← (in MHz)

t_{rx} in (n sec.), empirical formula



⟹ t_{sys} should be <u>less</u> than { 70 % of an NRZ (non-return to zero) bit period
35% of an RZ (return-to-zero) bit period



→ i.e for NRZ if

{ $t_{sys} < \dfrac{0.7}{\text{data rate}}$ ⟹ design is O.K

if $t_{sys} > \dfrac{0.7}{\text{data rate}}$ ⟹ try a new design

for RZ, replace 0.7 by 0.35

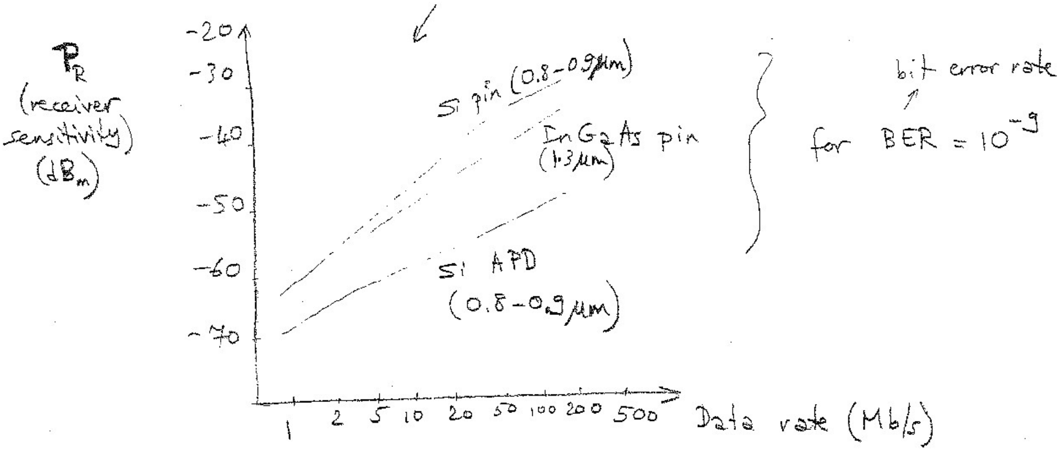
→ Check the cost of the overall system and try other designs to find a cheaper one.

An example

Perform the power budget and the rise time budget of an optical fiber link with 20 Mb/s data rate (NRZ format) and 10^{-9} bit error rate. Determine the maximum repeaterless transmission distance for the design.

→ Pick a silicon pin photodiode at 0.85 μm.

From Fig. 8-3 of Keiser



For Si pin photodiode at 0.85 μm, $P_R = -42\ dB_m$ for 20 Mb/s data rate

→ Pick a GaAlAs LED that can couple 50 μW ($-13\ dB_m$) average optical power into a fiber flylead with 50 μm core diameter. i.e. $P_S = -13\ dB_m$

$$\therefore P_T = P_S - P_r = -13\ dB_m - (-42\ dB_m) = 29\ dB_m$$

→ Assume 1 dB loss at the fiber flylead to cable connector and a 6 dB system margin.

$$\therefore 29\ dB = 2\,(1\ dB) + \alpha_f L + 6\ dB$$

$$\alpha_f L = 21\ dB$$

→ If $\alpha_f = 3.5\ dB/km$ for the fiber you pick, then

$$L = \frac{21}{3.5} = 6\ km. \Rightarrow$$ This is the repeaterless transmission distance

For graphical representation see Fig. 8-4 of Keiser's book.

For the rise time budget analysis

→ Assume LED and its drive circuitry has $t_{tx} = 15\,ns$

→ From Fig. 3-13 of Keiser / or (from Eq 3-27) $D_{mat} \cong 87.5\,ps/nm.km$ at $\lambda = 0.85\,\mu m$

Typical spectral width (σ_λ) of LED = 40 nm

$$\therefore t_{mat} = D_{mat}\, \sigma_\lambda\, \underset{\substack{\uparrow \\ \text{transmission} \\ \text{distance}}}{L} = 87.5\ ps/nm.km \times 40\ nm \times 6\ km$$

$$t_{mat} = 21\ nsec.$$

→ If the fiber selected has 400 MHz. km bandwidth distance product

i.e if $B_o = 400\,MHz$

and $q = 0.7$ for practical values

$$t_{mod} = \frac{440\, L^{q} \leftarrow \text{in km}}{\underset{\substack{\uparrow \\ \text{in MHz}}}{B_o}} = \frac{440 \times 6^{(0.7)}}{400} \cong 3.9\ nsec.$$

— For further information on mode coupling effect i.e the choice of q read sec. 3-5 of Keiser.

Actually for the first 0.1 – 0.55 km of fiber length $q \cong 1$, $B_M \propto L$ ← no mode mixing

" " fiber lengths >> 0.55 km $q \cong 0.5$ ⇒ mode mixing $B_M \propto \sqrt{L}$

Reasonable estimate for q is 0.7

→ Assuming the 3dB electric bandwidth of the receiver is

$$25\ MHz = \frac{1}{2\pi R_T C_T}$$

$$t_{rx} = \frac{350}{25} = 14\ ns$$

$$\therefore t_{sys} = \left[(15\,ns)^2 + (21\,ns)^2 + (3.9\,ns)^2 + (14\,ns)^2\right]^{1/2}$$

$$\cong 29.6\ ns$$

→ $$\frac{0.7}{\text{data rate}} = \frac{0.7}{20\ Mb/s} = 35\ ns$$

$$t_{sys} = 29.6\ ns < 35\ ns$$

∴ the design is O.K based on rise time budget

→ Check the cost of the overall system and try other designs

Free Space Optics (FSO)

- FSO is a wireless optical transmission in the atmosphere
- Current RF bandwidth is limited to 622 Mbps and does not provide economical solution for service providers looking to extend to optical networks
- In USA, only 5 percent of the buildings are connected to fiber-optic infrastructure (backbone) but 75 percent are within one mile of fiber
- As bandwidth demands increase and businesses require high-speed LANs, FSO becomes one of the most attractive solutions
- Two infrared wavelengths, around 1550 nm (194 THz) and around 800 nm (375 THz)
- Each FSO unit uses mainly high power laser sources (sometimes LED) and a lens that transmits light through the atmosphere to another lens receiving the information
- Receiving lens connects to a high-sensitivity receiver via optical fiber
- Optical pulse modulation
- Line-of-sight (LOS)
- Broadband (100 Mbps, 155 Mbps, 622 Mbps and up to 2.5 Gigabit capacities
- Even DWDM is also tried
- 1.5 –2 km
- Full duplex (bi-directional) communication
- Some disturbances facing FSO:
    - Fog: Major effect to FSO. Rain and snow have little effect. Fog is vapor composed of water droplets, which are only a few hundred microns in diameter modifying light characteristics or completely stopping light through absorption, scattering and reflection. Solution is to shorten the FSO link distances and to add network redundancies
    - Absorption: (Molecular and Aerosol Absorption). Light is converted into heat. Occurs mainly due to water molecules present in the atmosphere. Solution is to use of appropriate power, based on atmospheric conditions, and use of spatial diversity (multiple beams within an FSO unit)

- Scattering: Occurs when the light beam collides with the scatterer of size d. In scattering, unlike absorption, there is no loss of energy, only a directional redistribution of energy occurs that may have significant reduction in beam intensity for longer distances.

  - For $d < \lambda$ (wavelength), Rayleigh scattering (i.e molecular scattering). Rayleigh scattering is inversely proportional to λ^4.
  - For d is comparable with λ, Mie scattering (i.e aerosol scattering). More directive
  - For $d >> \lambda$, Non-selective scattering

- Physical obstructions**:** Flying birds can temporarily (for a short time) block a single beam and transmissions are easily and automatically resumed. Solution is to use multi-beam systems (spatial diversity)

- Building sway/seismic activity: Movement of buildings can disturb receiver and transmitter alignment. Solution is to use divergent beam or make tracking

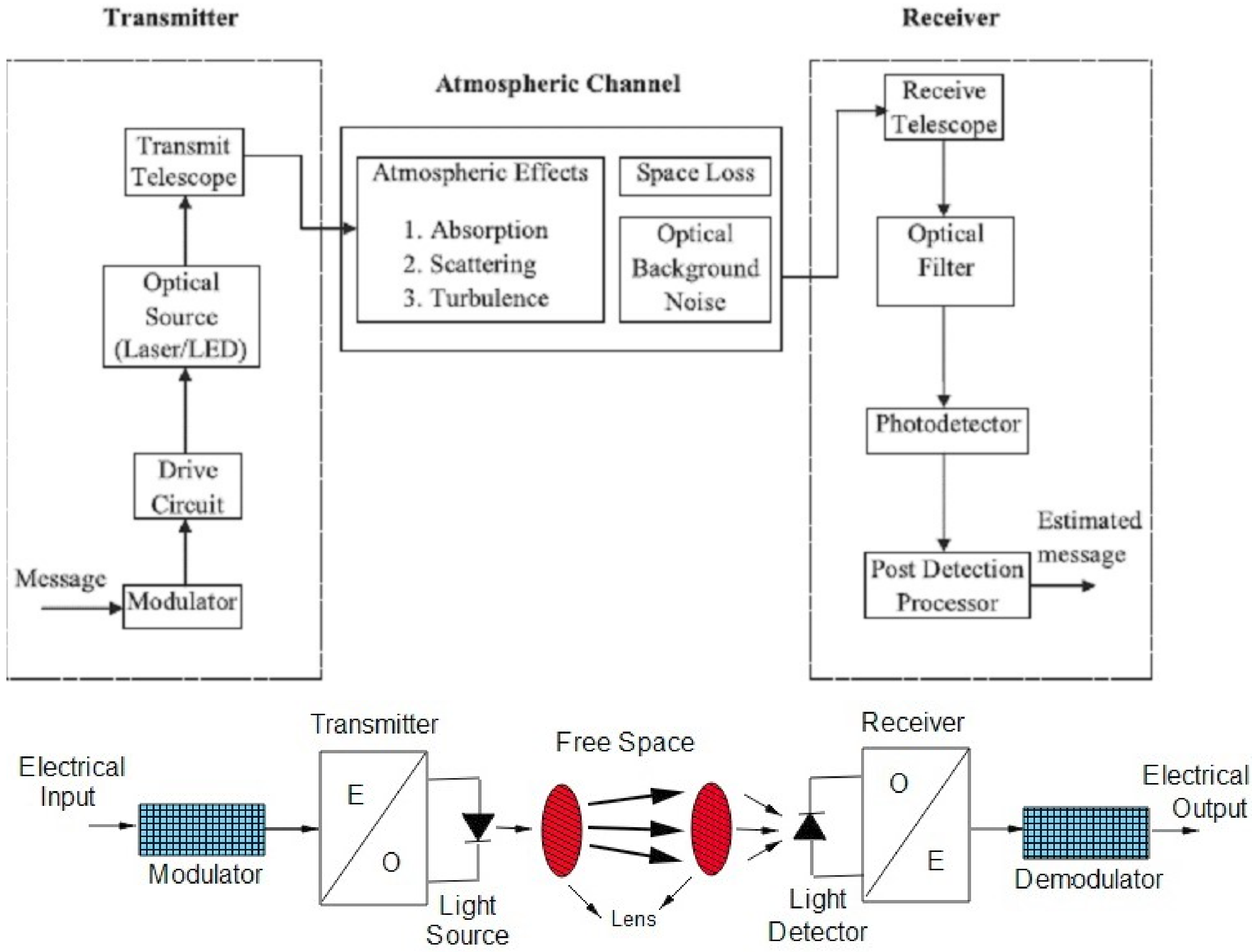
- Scintillation: Heated air rising from the earth or man-made devices such as heating ducts creates temperature variations among different air parsels known as turbulence. This can cause fluctuations in signal amplitude which leads to "image dancing" at the FSO receiver end. Remedy is to use multi-beam system

  - Beam Wander
  - Beam Spreading

- Safety: Human exposure to laser beams and high voltages within the laser systems and their power supplies

Based on their transmission range, WOC (Wireless Optical Communication) can be classified into five broad categories:

(i) Ultrashort-range WOC – used in chip-to-chip communication or all optical lab-on-a-chip system.
(ii) Short-rangeWOC – used in wireless body area networks (WBANs) or wireless personal area networks (WPANs).
(iii) Medium-range WOC – used in indoor IR or visible light communication (VLC) for wireless local area networks (WLANs) and inter-vehicular and vehicle-to-infrastructure communications.
(iv) Long-range WOC – used in terrestrial communication between two buildings or metro area extensions.
(v) Ultra-long-range WOC – used in ground-to-satellite/satellite-to-ground or inter-satellite link or deep space missions.



Block diagram of WOC

FSO applications:

• Telecommunication and computer networking
• Point-to-point LOS links
• Temporary network installation for events or other purpose as disaster recovery
• For communications between spacecraft, including elements of satellite constellation
• Security applications
• Military application: (its potential for low electromagnetic emanation when transferring sensitive data for air forces)
• Metro network extensions: carriers can deploy FSO to extend existing metropolitan area fiber rings, to connect new networks, and, in their core infrastructure, to complete SONET rings.
• Enterprise connectivity: the ease with which FSO links can be installed makes them a natural for interconnecting local area network segments that are housed in buildings separated by public streets or other right-of-way property.
• Fiber backup: FSO may also be deployed in redundant links to backup fiber in place of a second fiber link.
• Backhaul: FSO can be used to carry cellular telephone traffic from antenna towers back to facilities wired into the public switched telephone network.
• Service acceleration: FSO can be also used to provide instant service to fiber-optic customers while their fiber infrastructure is being laid.
• Last-Mile access: In today’s cities, more than 95% of the buildings do not have access to the fiber optic infrastructure due to the development of communication systems after the metropolitan areas. FSO technology seems a promising solution to the connection of endusers to the service providers or to other

existing networks. Moreover, FSO provides highspeed connection up to Gbps, which is far more beyond the alternative systems.

FSO Advantages:

- Long distance up to 8 km.
- High bit rates speed rates: the high bandwidth capability of the fiber optic of 2.5 Gbps to 10 Gbps achieved with dense wavelength division multiplexing (DWDM). Modern systems can handle up to 160 signals and can thus expand a basic 10 Gbit/s system over a signal fiber pair to over 1.6 Tbit/s.
- Immunity from electromagnetic interference: secure cannot be detected with RF meter or spectrum analyzer, very narrow and directional beams
- Invisible and eye safe, no health hazards so even a butterfly can fly unscathed through a beam
- Low bit error rates (BER)
- Absence of side lobes
- Deployment of FSO systems quickly and easily
- Low maintenance (Practical)
- Lower costs as compared to fiber networks (FSO costs are as low as 1/5 of fiber network costs).
- License-free long-range operation (in contrast with radio communication)

FSO disadvantages:

For terrestrial applications, the principal limiting factors are beam dispersion, atmospheric absorption, rain, fog, snow, interference from background light sources (including the sun), shadowing, pointing stability in wind, and pollution.

Atmospheric effects:

Transmitted power of the emitted signal is highly affected by scattering, absorption and turbulence.

Attenuation is the result of absorption and scattering by molecules and particles (aerosols) suspended in the atmosphere.
Distortion, on the other hand, is caused by atmospheric turbulence due to random index of refraction fluctuations.

Attenuation affects the mean value of the received signal in an optical link whereas distortion results in variation of the signal around the mean.

Aerosols:

Aerosols are particles suspended in the atmosphere with different concentrations. Each aerosol cause absorption and scattering.

They have diverse nature, shape, and size. Aerosols can vary in distribution, constituents, and concentration. As a result, the interaction between aerosols and light can have a large dynamic, in terms of wavelength range of interest and magnitude of the atmospheric scattering itself. Some aerosols are rain, smoke, fog, snow, desert dust particles, human-made industrial particulates, maritime droplets.

Aerosol scattering are explained by Mie scattering theory because the sizes of aerosols are comparable to or larger than the wavelength of the optical communications.

Transmitted optical beams in free space are attenuated most by the fog and haze droplets mainly due to dominance of Mie scattering and absorption effects in the wavelength band of interest in FSO (0.5 μ m – 2 μ m).

The Mie scattering coefficient is

$$\beta_{a(scat)} = \alpha_a N_a \text{ in 1/km}$$

where α_a is the Mie scattering cross-section in km^2 and N_a is the number density of air particles in 1/km^3.

An aerosol's concentration, composition and dimension distribution vary temporally and spatially varying, so it is difficult to predict attenuation by aerosols. Although their concentration is closely related to the optical visibility, there is no single particle dimension distribution for a given visibility. Due to the fact that the visibility is an easily obtainable parameter,either from airport or weather data, the scattering coefficient $\beta_{a(scat)}$ is expressed as

$$\beta_{a(scat)} = \left(\frac{3.91}{V}\right)\left(\frac{0.55}{\lambda}\right)^i$$

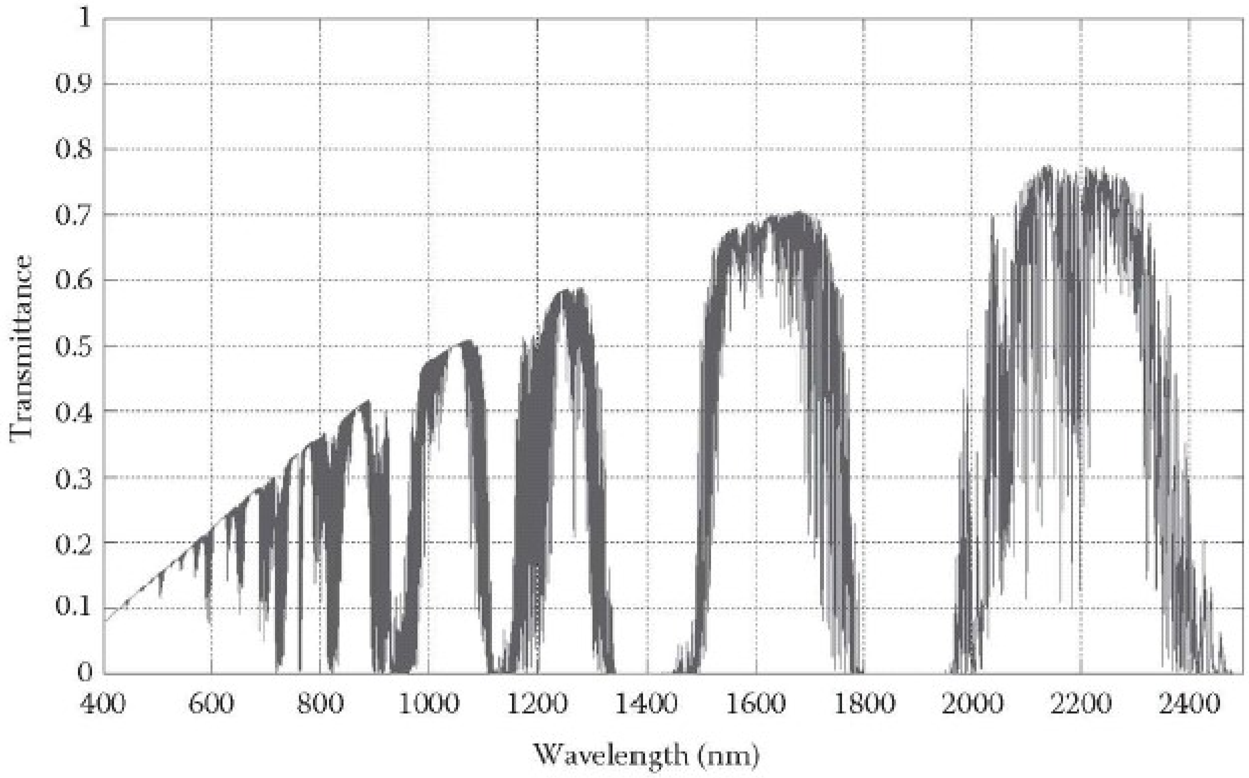
where V is the visibility (Visual Range) in km, λ is the incident laser beam wavelength μm, i is the size distribution of the scattering particles which typically varies from 0.7 to 1.6 corresponding to visibility conditions from poor to excellent.

Also the absorption coefficient of the aerosol $\beta_{a(absorp)}$ is found and the atmospheric transmittance due to an aerosol is found to be

$$\tau_{aerosol} = \exp\left(-\beta_{T(aerosol)}L\right)$$

where the total attenuation due to an aerosol is $\beta_{T(aerosols)} = \beta_{a(scat)} + \beta_{a(absorp)}$ and L is the link distance, i.e., the distance between from the transmitter and the receiver.

Thus, the total attenuation due to aerosols is found by evaluating the scattering and absorption coefficients of each aerosol present in the FSO link and by adding them to find $\beta_{T(aerosols)}$ and $\tau_{aerosols} = \exp\left(-\beta_{T(aerosols)}L\right)$.



Atmospheric transmittance window with absorption contribution.

<u>Molecules</u>:

There are more than 40 different molecules in the atmosphere, e.g., nitrogen, hydrogen, carbon dioxide, ... etc. Each of these molecules cause scattering and absorption.

Rayleigh (molecular) scattering refers to scattering by molecular and atmospheric gases of sizes much less than the incident light wavelength. The Rayleigh scattering coefficient is given by

$\beta_{m(scat)} = \alpha_m N_m$ in 1/km

where α_m is the Rayleigh scattering cross-section in km^2,
N_m is the number density of air molecules in 1 /km^3.

Rayleigh scattering cross section is inversely proportional to fourth power of the wavelength of incident beam (λ^{-4}) as

$$\alpha_{m(scat)} = \frac{8\pi^3\left(n^2-1\right)^2}{3N^2\lambda^4} \text{ in km}^2$$

where n is the index of refraction, λ is the incident light wavelength in m, N is the volumetric density of the molecules in 1 /km^3.

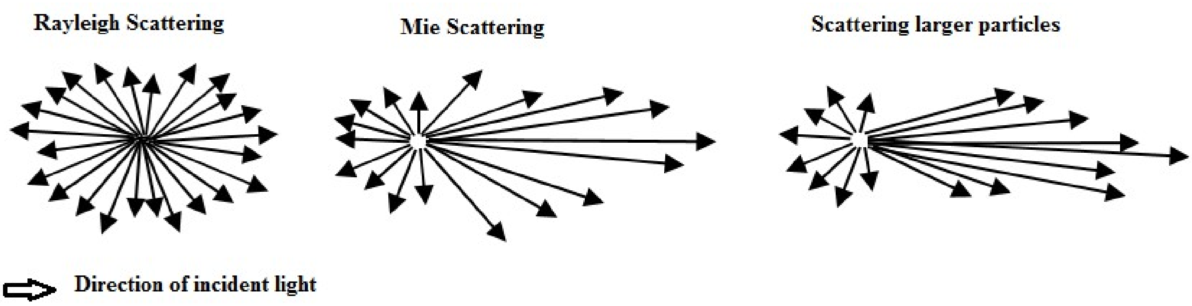
The result is that Rayleigh scattering is negligible in the infrared waveband because Rayleigh scattering is primarily significant in the ultraviolet to visible wave range.

Also the absorption coefficient of the molecule $\beta_{m(absorp)}$ is found and the atmospheric transmittance attenuation due to a molecule is found to be

$$\tau_{molecule} = \exp\left(-\beta_{T(molecule)} L\right)$$

where the total attenuation due to a molecule is $\beta_{T(molecule)} = \beta_{m(scat)} + \beta_{m(absorp)}$ and L is the link distance, i.e., the distance between from the transmitter and the receiver.

Thus, the total attenuation due to molecules is found by evaluating the scattering and absorption coefficients of each molecule present in the FSO link and by adding them to find $\beta_{T(molecules)}$ and $\tau_{molecules} = \exp\left(-\beta_{T(molecules)} L\right)$.



Turbulence

Clear air turbulence phenomena affect the propagation of optical beam by both spatial and temporal random fluctuations of refractive index due to temperature, pressure, and wind variations along the optical propagation path.

Atmospheric turbulence primary causes phase shifts of the propagating optical signals resulting in distortions in the wave front.

These distortions, referred to as optical aberrations, also cause intensity distortions, referred to as scintillation.

Moisture, aerosols, temperature and pressure changes produce refractive index variations in the air by causing random variations in density.

These variations are referred to as eddies and have a lens effect on light passing through them.

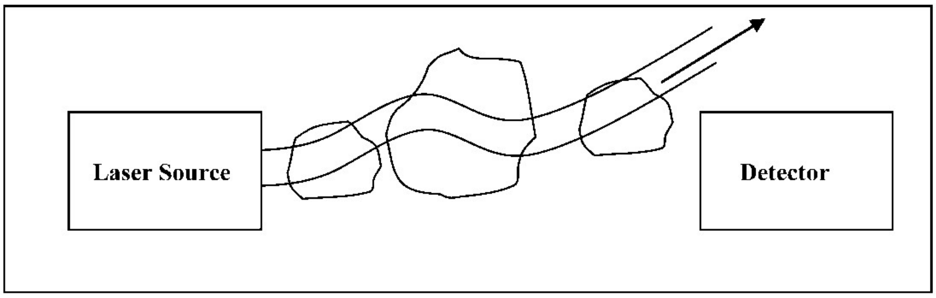
When the light beam wave passes through these eddies, parts of it are refracted randomly causing a distorted wave front with the combined effects of variation of intensity across the wave front and warping of the isophase surface.

As the result, turbulence causes scattering $\tau_{turbulence} = \exp\left(-\beta_{t(scat)} L\right)$

i.e., the overall total attenuation due to aerosols, molecules and turbulence is found by $\beta_{T(overall)} = \beta_{a(scat)} + \beta_{a(absorp)} + \beta_{m(scat)} + \beta_{m(absorp)} + \beta_{t(scat)}$ and $\tau_{overall} = \exp\left(-\beta_{T(overall)} L\right)$.

Other effects of turbulence:

- Beam wander



- Beam spread: Beam size at the receiver is increased further on top of free space diffraction.
- Intensity fluctuations: Intensity at the receiver fluctuates in time and space.

2. Propagation of Light in FSO

Free-Space Propagation of Gaussian-Beam Waves

The mathematical description of a propagating wave involves a field.

Basically, a field $u(\mathbf{R},t)$ is a function of space $\mathbf{R} = (x, y, z)$ and time t that satisfies a partial differential equation.

In the case of electromagnetic radiation, the field may be a transverse electromagnetic (TEM) wave, whereas for acoustic waves the field may represent a pressure wave.

The governing equation in most cases is the wave equation

$$\nabla^2 u = \frac{1}{c^2}\frac{\partial^2 u}{\partial t^2}$$

where $c = 3\times10^8$ m/s is the speed of the propagating wave which is light and ∇^2 is the Laplacian operator defined in rectangular coordinates by

$$\nabla^2 u = \frac{\partial^2 u}{\partial x^2} + \frac{\partial^2 u}{\partial y^2} + \frac{\partial^2 u}{\partial z^2}.$$

If we assume that time variations in the field are sinusoidal (i.e., a monochromatic wave), then we look for solutions of the form $u(\mathbf{R},t) = U_0(\mathbf{R})e^{-i\omega t}$ where ω is the angular frequency and $U_0(\mathbf{R})$ is the complex amplitude of the wave which is the spatial field.

The substitution of this solution form into the wave equation leads to the time-independent reduced wave equation (or Helmholtz equation)

$\nabla^2 U_0 + k^2 U_0 = 0$ where k is the optical wave number related to the optical wavelength λ by $k = \omega / c = 2\pi / \lambda$.

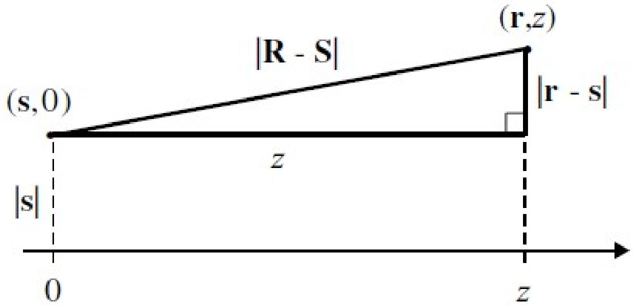
For optical wave propagation, Helmholtz equation can be further reduced to what is called the paraxial wave equation.

Let us assume the beam originates in the plane at z=0 and propagates along the positive z-axis. If we also assume the free-space optical field at any point along the propagation path remains rotationally symmetric, then it can be expressed as a function of $r = \sqrt{x^2 + y^2}$ and z.

Thus, the reduced wave equation in cylindrical coordinates can be written as

$$\frac{1}{r}\frac{\partial}{\partial r}\left(r\frac{\partial U_0}{\partial r}\right)+\frac{\partial^2 U_0}{\partial r^2}+k^2U_0=0$$

Paraxial approximation can be made when the propagation distance of an optical wave along the *z*-axis is much greater than the transverse spreading of the wave.



If **R** = (**r**, *z*) and **S** = (**s**, *z*) denote two points in space with **r** and *s* transverse to the propagation axis, then the distance between such points is

$$\left|\mathbf{R}-\mathbf{S}\right|=\left(z^2+\left|\mathbf{r}-\mathbf{s}\right|^2\right)^{1/2}=z\left(1+\frac{\left|\mathbf{r}-\mathbf{s}\right|^2}{z^2}\right)^{1/2}$$

If the transverse distance is much smaller than the longitudinal propagation distance between the points, then the second factor can be expanded in a binomial series to obtain

$\left|\mathbf{R}-\mathbf{S}\right|=z\left(1+\frac{\left|\mathbf{r}-\mathbf{s}\right|^2+...}{2z^2}\right)=z+\frac{\left|\mathbf{r}-\mathbf{s}\right|^2}{2z}+...,\ \left|\mathbf{r}-\mathbf{s}\right|<<z$ which is known as the paraxial approximation.

The complex amplitude at propagation distance *z* from the source is given by Huygens-Fresnel integral as

$$U(\mathbf{r},z)=-2ik\int_{-\infty}^{\infty}\int_{-\infty}^{\infty}G(\mathbf{s},\mathbf{r};z)U_0(\mathbf{s},0)d^2s$$

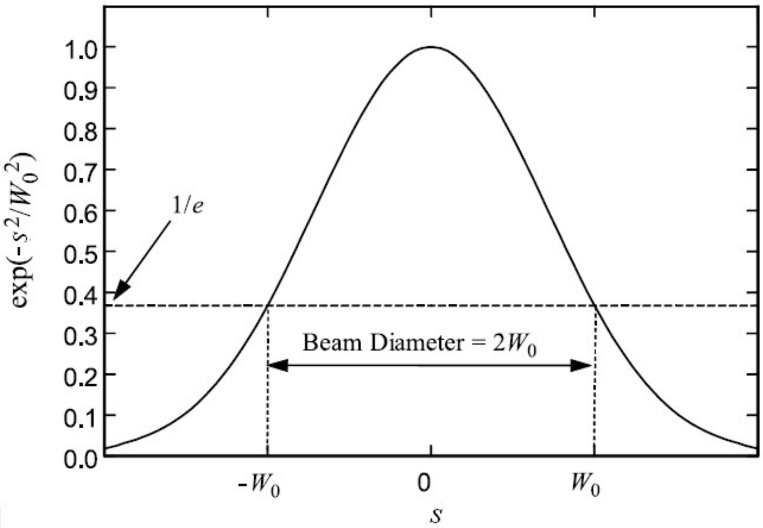
where $U_0(\mathbf{s},0)$ is the optical wave at the source plane and $G(\mathbf{s},\mathbf{r};z)$ is the free-space Green's function which can be expressed under the paraxial approximation as

$$G(\mathbf{s},\mathbf{r};z)=\frac{e^{ik}\left|\mathbf{R}-\mathbf{S}\right|}{4\pi\left|\mathbf{R}-\mathbf{S}\right|}\cong\frac{1}{4\pi z}\exp\left(ikz+\frac{ik}{2z}\left|\mathbf{r}-\mathbf{s}\right|^2\right)$$

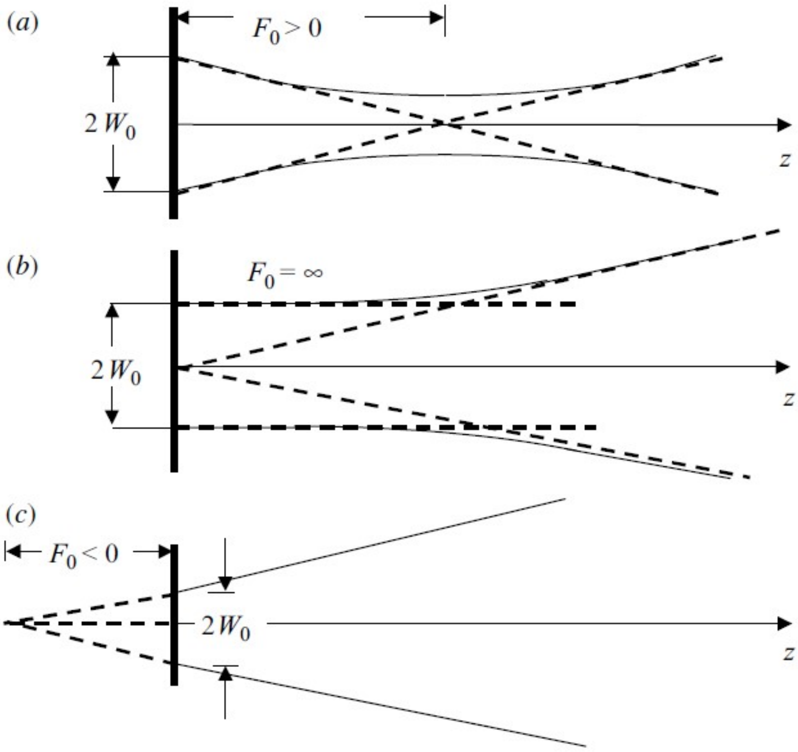
The complex amplitude of the Gaussian-beam wave at the source plane *z*=0 is

$$U_0(\mathbf{s},0)=A\exp\left(-\frac{1}{2}\alpha_0ks^2\right)=A\exp\left[\frac{ik}{2z}(i\alpha_0z)s^2\right]=A\exp\left[-\frac{s^2}{W_0^2}-\frac{ik}{2F_0}s^2\right]$$

where A is the amplitude at the origin, $s=\sqrt{x^2+y^2}$ is radial distance from the beam center line and $\alpha_0=\frac{2}{kW_0^2}+i\frac{1}{F_0}$. It is assumed that the transmitting aperture is located in the plane *z*=0 and the amplitude distribution in this plane is Gaussian with effective beam radius (spot size) W_0 in meters, which denotes the radius at which the field amplitude falls to 1/e of that on the beam axis as shown below for A=1



Additionally, the phase front is taken to be parabolic with radius of curvature F_0 in meters. The particular cases $F_0 = \infty, F_0 > 0, F_0 < 0$ correspond to collimated, convergent, and divergent beam forms, respectively



(a) Convergent beam, (b) collimated beam, and (c) divergent beam.

Thus, for the Gaussian beam, Huygens-Fresnel integral becomes

$$U(\mathbf{r},z) = -2ik \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} \frac{1}{4\pi z} \exp\left(ikz + \frac{ik}{2z}|\mathbf{r}-\mathbf{s}|^2 \right) A \exp\left[\frac{ik}{2z}(i\alpha_0 z)s^2 \right] d^2 s$$

Changing the integration to polar coordinated where $d^2 s = s d\theta ds$ and rearranging

$$U(\mathbf{r},z) = -\frac{Aik}{2\pi z}\exp\left(ikz + \frac{ik}{2z}r^2\right)\int_0^\infty\int_0^{2\pi}\exp\left(-\frac{ik}{z}rs\cos\theta\right)\exp\left[\frac{ik}{2z}(1+i\alpha_0 z)s^2\right]sd\theta d^2 s$$

Performing the integrations, the electric field at the receiver is found as

$$U(\mathbf{r},z) = \frac{A}{1+i\alpha_0 z}\exp\left[ikz + \frac{ik}{2z}\left(\frac{i\alpha_0 z}{1+i\alpha_0 z}\right)r^2\right]$$

The intensity at the receiver is $I(\mathbf{r},z) = U(\mathbf{r},z)U^*(\mathbf{r},z)$

3. FSO Link Design

The ability for an optical link to deliver the signal power to the receiver is governed by the link equation

$$P_R = P_T\left(\eta_T\eta_A\frac{4\pi A_T}{\lambda^2}\right)L_{TP}L_{atm}L_{pol}L_{RP}\left(\frac{A_R}{4\pi z^2}\right)\eta_R$$

where

P_R is the total signal power at the input to the receiver. For the uplink, this is defined at the input to the optical detector. For the downlink, the receive signal power is defined at the input to the receive optical detector,
P_T is the transmit optical power at the transmit interface,
η_T is the transmit optics efficiency,
η_A is the aperture illumination efficiency of the transmitter lens,
λ is the wavelength,
A_T is the aperture area,
L_{TP} is the transmitter pointing loss, defined as the ratio of power radiated in the direction of receiver to the peak radiated power. If the transmitter is directly pointed at the receiver, the pointing loss is 0 dB,
L_{atm} is the fractional loss due to absorption of the medium (e.g., earth atmosphere),
L_{pol} is the fractional signal loss due to mismatch of the transmitting and receiving polarization patterns,
L_{RP} is the receiver pointing loss, defined as the ratio of receiving lens gain in the direction of the transmittting lens to the peak receiving lens gain,
A_R is the receive aperture area,
z is the link distance,
the term $\left(\frac{A_R}{4\pi z^2}\right)$ is the fraction of power that is collected by the receiving aperture if the transmitter is an isotropic radiator.
η_R is the receiving optics collecting efficiency, defined as the fraction of optical power at the receiving aperture that is collected within the field of view of the receive detector.

Thus, the receive signal power can be improved by the following:

1) Increasing the transmit power. The most straightforward method of improving the receive signal power is to increase the power at the transmitter since the receive power scales linearly with the transmit power. However, increasing the transmit power also increases the overall system power consumption

which, for a deep-space mission, is typically at a premium. Furthermore, the increased power consumption can lead to thermal management issues (increased radiator size and hence mass) for the host spacecraft, as well as reliability concerns.
2) Increasing the transmit aperture. This effectively reduces the transmit beamwidth and hence improves the power delivery efficiency. However, the pointing and tracking of the narrow downlink becomes increasingly more difficult with a narrower downlink. Furthermore, the aperture size is highly correlated with the mass of the transmit terminal and hence cannot be increased indefinitely.
3) Reducing the operating wavelength. Reducing the operating wavelength reduces the diffraction loss of the signal (i.e., reduces the transmit beamwidth). However, the wavelength selection is strongly constrained by the available laser technology, as well as considerations on the receiver sensitivity and detector technology. Furthermore, the transmittance of the atmosphere also depends on the wavelength, as well as the amount of sky background irradiance.
4) Increasing the receiver aperture area. Since the receive signal power scales linearly with the receive aperture area, increasing the receiver aperture area is a relatively simple way to improve the system performance. However, for daytime operations of a receiver inside the Earth's atmosphere, the amount of background noise collected also increases with increasing receiver aperture, and the effective performance improvement does not always scale linearly with increasing aperture area.
5) Reduced pointing loss. Reducing the pointing loss improves the overall signal energy and also reduces the point-induced signal power fluctuation.
6) Improving the overall efficiency, including transmit and receive optical loss, and polarization mismatch losses. This generally requires attention to the optical design. Of particular attention is the transmit optics design. The transmit aperture illumination efficiency, η_A, depends on the phase and intensity distribution over the aperture. For the general case of a transmit aperture being illuminated by a Gaussian beam, the aperture illumination efficiency can be written as:

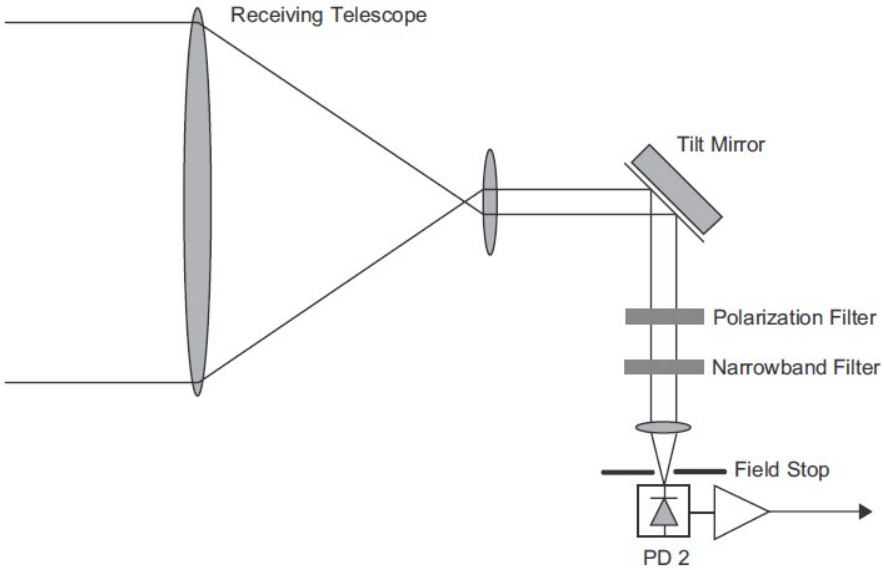
$$\eta_A = \frac{2S}{\alpha^2}\left[\exp\left(-\alpha^2\gamma^2\right) - \exp\left(-\alpha^2\right)\right]^2$$

where α is the ratio between the aperture diameter and the Gaussian beam ($1\ /\ e^2$) diameter of the transmit signal, γ is the obscuration ratio (darkening ratio) and S is the Strehl ratio, which is defined as the intensity at the center of the aberrated system to that of an ideal optical system.

Optical-Receiver Sensitivity

In addition to the effective delivery of the signal to the detector, the performance of the optical link also depends on the receiver sensitivity (measured in terms of received photons per bit). Because of the high cost associated with increasing the transmit power and system aperture, improving the receiver sensitivity is an important factor

In a direct-detection receiver, the received optical intensity is detected without extensive front-end optical processing.



The incident signal is collected by the receive telescope. A polarization filter followed by a narrowband filter, and a field stop effectively reduces the amount of background noise incident onto the detector.

The capacity of a direct detention optical channel in the presence of background can be written as:

$$C = (\log_2 e)\frac{\lambda_s}{M}\left[\left(1+\frac{1}{\rho}\right)\ln(1+\rho)-\left(1+\frac{M}{\rho}\right)\ln\left(1+\frac{\rho}{M}\right)\right]$$

<u>Photon Detection Sensitivity</u>

Improving the photon detection efficiency is an obvious method of improving the channel performance. For a direct-detection receiver, this is generally accomplished by using detectors with internal amplifications, such as avalanche photodiodes (APDs) and photomultiplier tubes (PMTs).

<u>Modulation Format</u>

One practical modulation format to achieve high peak-to-average-power ratio is the M-ary pulse-position modulation (PPM). In an M-ary PPM modulation scheme, each channel symbol period is divided into M time slots, and the information is conveyed through the channel by the time window in which the signal pulse is present.

<u>Link Availability</u>

The communications link budget or the DCT is a useful tool in estimatingthe physical layer link performance (e.g., the link bit error rate). An operational communications link, on the other hand, must also address the issue of link availability. Overall link availability is aimed at >90 percent depending on the link type. In some links, 99.999 % availability may be required.

<u>Beam Pointing and Tracking</u>

Due to the narrow transmit beamwidth, accurate pointing acquisition and tracking are critical to the deep-space laser communications system implementation.

<u>Typical design parameters considered in an FSO link design:</u>

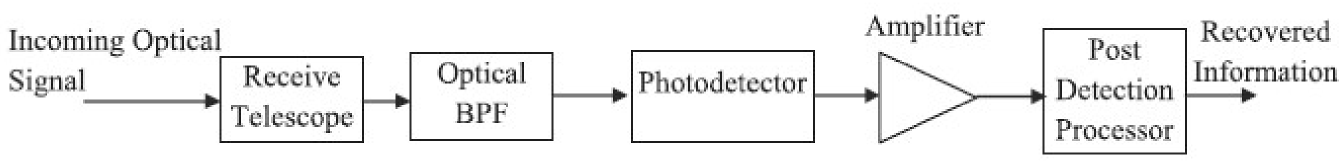
Link Budget	Parameters

Received signal power	Operating wavelength
	Link distance
	Transmit power
	Transmit aperture area
	Transmit optics efficiency
	Transmit Strehl ratio
	Transmit pointing loss
	Polarization mismatch loss
	Receiver aperture area
	Receive optics efficiency
	Receiver detector field of view
	Receiver pointing loss
	Atmospheric attenuation loss
	Scintillation-induced loss

Received background power	Receive aperture area
	Receive optics efficiency
	Detector field of view
	Receive optical bandwidth
	Background spectral irradiance
	Receive optics scattering behavior
	Detector dark count

Receiver sensitivity	Detector quantum efficiency
	Detector noise characteristics • Dark count rate or • Detector Excess and thermal noise
	Modulation format
	Coding scheme

<u>Direct Detection System</u>



Block diagram of a direct detection scheme

In direct detection technique, the received optical signal is passed through optical band-pass filter to restrict the background radiation.

It is then allowed to fall on the photodetector which produces the output electrical signal proportional to the instantaneous intensity of the received optical signal.

It may be regarded as linear intensity to current convertor or quadratic (square law) converter of optical electric field to detector current.

The photodetector is followed by an electrical low-pass filter (LPF) with bandwidth sufficient enough to pass the information signal.

The signal-to-noise ratio (SNR) of direct detection receiver can be obtained by using noise models for a particular detector, i.e., PIN or avalanche photodetector (APD).

With the received power as given above by

$$P_R = P_T\left(\eta_T\eta_A\frac{4\pi A_T}{\lambda^2}\right)L_{TP}L_{atm}L_{pol}L_{RP}\left(\frac{A_R}{4\pi z^2}\right)\eta_R$$

and detector noise sources, the SNR expressions are obtained for PIN photodetector to be

$$SNR = \frac{\left(R_0P_R\right)^2}{2qB\left(R_0P_R + R_0P_B + I_d\right) + 4K_BTB/R_L}$$

where B is the receiver bandwidth, I_d is the dark current, K_B=1.3807 x 10 $^{-23}$ joules per kelvin ($J{\cdot}K^{-1}$) is the Boltzmann's constant, T is the absolute temperature, R_L is the equivalent load resistance, P_B is the background noise power and R_0 in mA/watt is the detector responsivity given by

$$R_0 = \frac{\eta q}{h\nu}$$

where η is the detector quantum efficiency, $q = 1.602\times10^{-19}$ Coulomb is the electronic charge, $h = 6.623\times10^{-34}$ Joule.sec is the Planck's constant, ν the operating frequency.

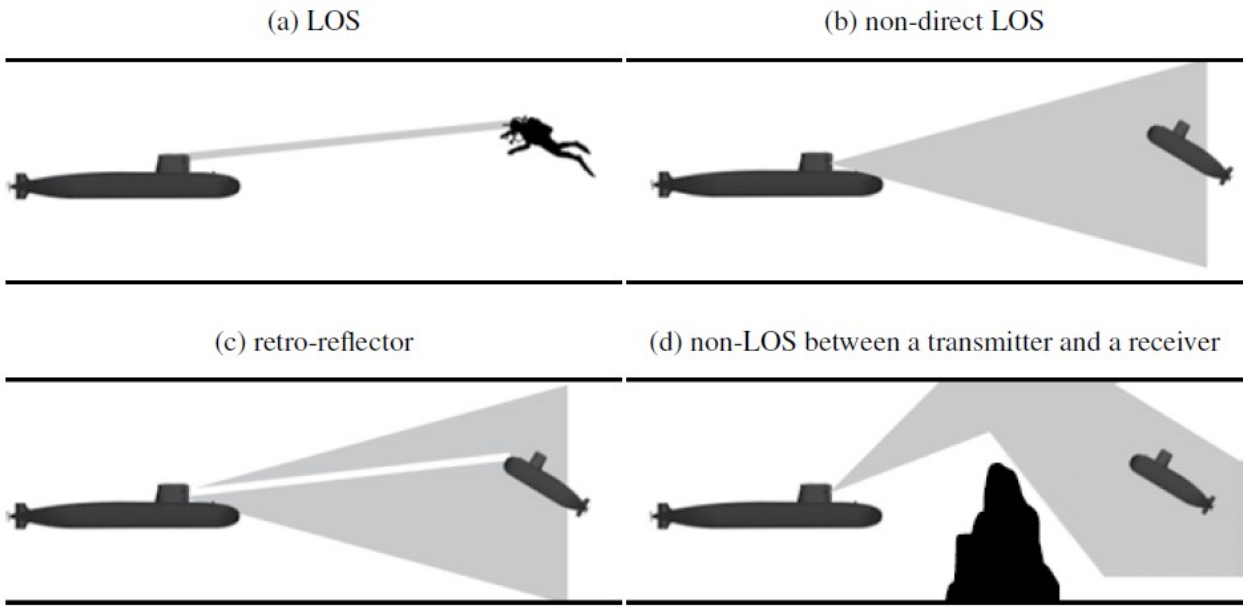
When APD is used, the dark current and shot noise are increased by the multiplication process; however, the thermal noise remains unaffected. Therefore, if the photocurrent is increased by a factor of M avalanche multiplication factor, then the total shot noise is also increased by the same factor. For the surface dark current I_{ds}=0, the direct detection SNR for APD photodetector is

$$SNR = \frac{\left(MR_0P_R\right)^2}{2qB\left(R_0P_R + R_0P_B + I_{db}\right)M^2F + 4K_BTB/R_L}$$

where F is the excess noise factor arising due to random nature of multiplication factor, I_{db} is the bulk dark current.

Since the photodetector response is insensitive to the frequency, phase, or polarization of the carrier, this type of receiver is useful only for intensity-modulated signals.

4. Optical Wireless Communication in Underwater Medium



Different underwater optical wireless link configuration.

Optical wireless communications are a relatively new technology providing many serious advantages, such as the very high rates of data transmission, secure links, very small and light.

Optical waves in the visible spectrum (400–700 nm) present an alternative way to provide broadband communications in the water. They propagate faster in water (300,000,000 m/s) than the acoustic ones (340 m/s in air, 1500 m/s in water), which is about 200,000 times faster than sound travels through water. That is the main reason why they have gained a considerable interest during the last years to serve as a broadband (10–100 Mbps), safe (non-interceptable) and reliable complement to acoustic underwater communications systems.

In general, optical signals are highly absorbed in water, and this is one of the main disadvantages; the other one is the optical scattering by all the particles existing inside the sea. However, seawater shows a decreased absorption in the blue/green region of the visible spectrum. Thus, using suitable wavelengths, for instance in the blue/green region, high speed connections can be attained according to the type of water (400–500 nm for clear to 300–700 nm for turbid water conditions). Minimum attenuation is centered near 0.460 μm in clear waters and shifts to higher values for dirty waters approaching 0.540 μm for coastal waters.

The power received $P(z)$, given initial power P_0, propagating through a medium of thickness z is estimated by the Beer's Law given by

$$P(z) = P_0 e^{-c(\lambda)z}$$

where $c(\lambda)$ in m^{-1} is the extinction coefficient expressing the total attenuation occurred by the propagation through the water.

The total attenuation can be described as the sum of absorption and scattering. Thus,

$$c(\lambda) = \alpha(\lambda) + \beta(\lambda)$$

where $\alpha(\lambda)$ is the absorption coefficient, $\beta(\lambda)$ is the scattering coefficient and the product *cz* is the attenuation length, and it contributes on the reduction of the received power by a factor of exp(-1), or ~63%.

Beer's Law provides a limited applicability as it describes only the attenuation due to absorption and single scattering events. In reality, however, many cases of multiple scattering may occur. Also it presumes that the source and receiver are in exact alignment with each other, and it can be applied only in Line-of-Sight (LOS) communication scenarios. Moreover, Beer's Law ignores temporal dispersion.

More accurate expressions have to take the link geometry into account. For instance, assuming that the transmitter and receiver are positioned in a non LOS configuration, the received power dependent on time *t*, lateral displacement from the beam axis *r*, and range *z* is

$$P_R(t,r,z) = P_T(t) D_T L_w(t,r,z) D_R$$

where $P_T(t)$ is the transmitted power, D_T is the aperture and divergence of the optical source, and D_R is the photoreceiver aperture and field of view. The channel loss term, $L_w(t,r,z)$, characterizes the spatial and temporal characteristics of light propagation in seawater.

<u>UOWC propagation</u>

Underwater medium characteristics

Underwater medium contains almost 80 different elements, dissolved or suspended in pure water, with different concentrations. Some of them are

• Various dissolved salts such as NaCl, MgCl2, etc, which absorb light at specific wavelengths and induce scattering effects.
• Minerals like sand, metal oxides, which contribute to both absorption and scattering.
• Colored dissolved organic matters such as fulvic and humic acids which affect absorption, mainly in blue and ultraviolet wavelengths.
• Organic matters such as viruses, bacteria, and organic detritus which add backscattering, especially in the blue spectral range.
• Phytoplankton with chlorophyll-A which strongly absorbs in the blue-red region and scatters green light.

Since chlorophyll absorbs the blue and red wavelengths and the particles strongly contribute to the scattering coefficient, we can use its concentration C (in mg/m^3) as the free parameter to calculate the absorption and scattering coefficients.

The exact type of water plays a significant role in the estimation of the amount of chlorophyll concentration and consequently the amount of absorption and scattering for a specific geographic location. A classification system for the clarity of water types based on their spectral optical attenuation depth was proposed by Jerlov.

The four major water types are

• Pure deep ocean waters cobalt blue where the absorption is high and the scattering coefficient is low.

• Clear sea waters with higher scattering due to many dissolved particles.

• Near coasts ocean waters with absorption and scattering due to planktonic matters, detritus and mineral components.

• Harbor murky waters, which are quite constraining for optical propagation due to dissolved and in-suspension matters.

Absorption

The absorption coefficient, $\alpha(\lambda)$ is the ratio of the absorbed energy from an incident power per unit distance due to various dissolved particles such as phytoplankton, detritus, etc.

$$\alpha(\lambda) = \alpha_w(\lambda) + \alpha_c^0(\lambda)\left(C_c / C_c^0\right)^{0.602} + \alpha_f^0 C_f e^{-k_f\lambda} + \alpha_h^0 C_h e^{-k_h\lambda}$$

where $\alpha_w(\lambda)$ is the absorption by the pure water in m^{-1}, λ is the wavelength in nm, $\alpha_c^0(\lambda)$ is the absorption coefficient of chlorophyll in m^{-1}, C_c is the total concentration of chlorophyll per cubic meter (C_c^0 = 1mg/m^3), α_f^0 = 35.959 m^2/mg is the absorption coefficient of fulvic acid, k_f = 0.0189 nm^{-1}, C_f = 0.0189 nm^{-1}, α_h^0 = 18.828 m^2/mg is the absorption coefficient of humic acid and k_h = 0.01105 nm^{-1}. The concentrations C_f and C_h are expressed as

$$C_f = 1.74098 C_c e^{0.12327\left(C_c / C_c^0\right)}$$

$$C_h = 0.19334 C_c e^{0.12343\left(C_c / C_c^0\right)}$$

Scattering

Scattering coefficient, $\beta(\lambda)$ is the ratio of energy scattered from an incident power per unit distance. It is the sum of backward scattering, $\beta_b(\lambda)$ and forward scattering coefficient, $\beta_f(\lambda)$.

Scattering is caused by small and large particles. Small particles are the particles with refractive index equal to 1.15, whereas large particles have a refractive index of 1.03. The scattering and backscattering coefficients are

$$\beta(\lambda) = \beta_w(\lambda) + \beta_s^0(\lambda) C_s + \beta_l^0(\lambda) C_l$$

$$\beta_B(\lambda) = 0.5\beta_w(\lambda) + 0.039\beta_s^0(\lambda) C_s + 6.4\times 10^{-4}\beta_l^0(\lambda) C_l$$

For small and large particulate matter

$$\beta_s^0(\lambda) = 1.151302\left(\frac{400}{\lambda}\right)^{1.7}$$

$$\beta_l^0(\lambda) = 0.341074\left(\frac{400}{\lambda}\right)^{0.3}$$

and the concentrations are

$$C_s = 0.01739 C_c e^{0.11631\left(C_c / C_c^0\right)}$$

$$C_l = 0.76284 C_c e^{0.03092\left(C_c / C_c^0\right)}$$

Oceanic turbulence

Optical wireless communications are greatly affected by optical turbulence, which refers to random fluctuations of the refraction index.

In the case of underwater systems, these fluctuations are mainly caused by variations in temperature and salinity of the oceanic water.

An important parameter for the description of oceanic turbulence is the scintillation index, which expresses the variance of the wave intensity.

Link budget

Empirical path loss models are effective enough to estimate the received optical power for underwater communications under LOS conditions.

$$P_R = P_T \eta_t \eta_r e^{-\frac{c(\lambda)R}{\cos\theta}} \frac{A_R \cos\theta}{2\pi R^2 \left(1-\cos\theta_0\right)}$$

where P_T is the transmitted power, η_t and η_r are the optical efficiencies of the Tx and Rx correspondingly, $c(\lambda)$ is the extinction coefficient, R is the perpendicular distance between the Tx plane and the Rx plane, θ_0 is the Tx beam divergence angle, θ is the angle between the perpendicular to the Rx plane and the Tx-Rx trajectory, and A_R is the receiver aperture area.

Table of Comparison between various transmission media is shown below:

Medium Type	Frequency of Operation	Maximum Bit Rate	Performance as Bit Error Rate (BER)	Distance Between Repeaters	Security	Cost
Twisted Pair	1MHz-100MHz-1GHz	2Mbps-100Mbps-1Gbps	10^{-5}	2 km - 100 m	Poor	Low-Moderate
Coaxial	1 GHz	565 Mbps	10^{-7} - 10^{-9}	2-3 km	Good	Moderate
Microwave	300 MHz - 40 GHz	622 Mbps	10^{-9}	30-70 km	Poor	Moderate
Satellite	390 MHz - 30 GHz	155 Mbps	10^{-9}	800-1500-36000 km	Poor	Moderate-High
Fiber	750 -194 THz	2.5 -10 Gbps -150 Tbps	10^{-11} - 10^{-13}	50 -100 - 6000 km	Good	Moderate-High

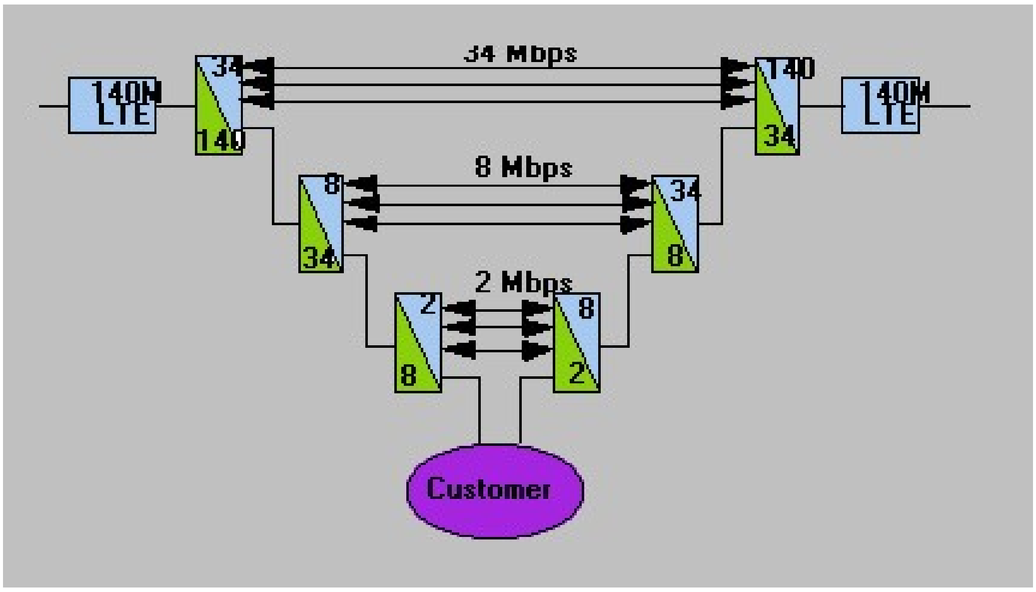
Transport Network Infrastructure, PDH, SDH/SONET

PDH (Plesiochronous Digital Hierarchy)

- PSTN backbone is based on PDH (Plesiochronous Digital Hierarchy) including E-carrier, T-carrier, J-carrier

CEPT Signal Level	Bit Rate	No. of E-0 Channels (Excluding Signalling Channels)	No. of E-0 Channels (Including Signalling Channels	No. of E-1 Lines	No. of E-2 Lines	No. of E-3 Lines
CEPT-0 (E-0)	64 Kb/s	1	-	-	-	-
CEPT-1 (E-1)	2.048 Mb/s	30	32	1	-	-
CEPT-2 (E-2))	8.448 Mb/s	120	128	4	1	-
CEPT-3 (E-3)	34.368 Mb/s	480	512	16	4	1
CEPT-4 (E-4)	139.264 Mb/s	1920	2048	64	16	4
CEPT-5 (E-5))	565.148 Mb/s	7680	8192	256	64	16

- In PDH, each network element (i.e. each exchange, multiplexer, cross-connect, repeater, ..etc) gets its clocking pulse from different clocking sources. These clocks are then synchronized. (Plesiochronous means close to clock)
- 30 DS-0 (Digital signal-0 level, i.e. 64 Kbps) channels plus one framing channel and one signaling channel make up a single E-1 frame, also known as a CEPT-1
- 16 E-1 frames make a single G.703 frame.
- In PDH, in order to access a single 2 Mbit\s line in a 140 Mbit\s system, the 140 Mbit\s channel must be completely demultiplexed to its 64 constituent 2 Mbit\s lines via 34 and 8 Mbit\s.
- Once the required 2 Mbit\s line has been identified and extracted, the channels must then be multiplexed back up to 140 Mbit\s.
- This problem with the "drop and insert" of channels does not make for very flexible connection patterns or rapid provisioning of services.
- Also the "multiplexer mountains" required are very expensive
- Drop-Insert mechanism in PDH is shown below:



<u>SDH/SONET</u> (Synchronous Digital Hierarchy / Synchronous Optical Network)

- SDH and SONET refer to a group of fiber-optic transmission rates that can transport digital signals with different capacities.

- SDH has provided transmission networks with a vendor-independent and sophisticated signal structure that has a rich feature set.

- As digital networks increased in complexity in the early 1980s, demand from network operators and their customers grew for features based on high-order multiplexing through a hierarchy of increasing bit rates up to 140 Mbps or 565 Mbps

- PDH has high costs of transmission bandwidth

- SDH brings economical use of bandwidth, better performance monitoring and greater network flexibilityswitched broadband services.

- SDH brings the following advantages to network providers:

1. High transmission rates: Up to 10 Gbit/s. SDH is therefore the most suitable technology for backbones, i.e. super highways in telecommunications networks.

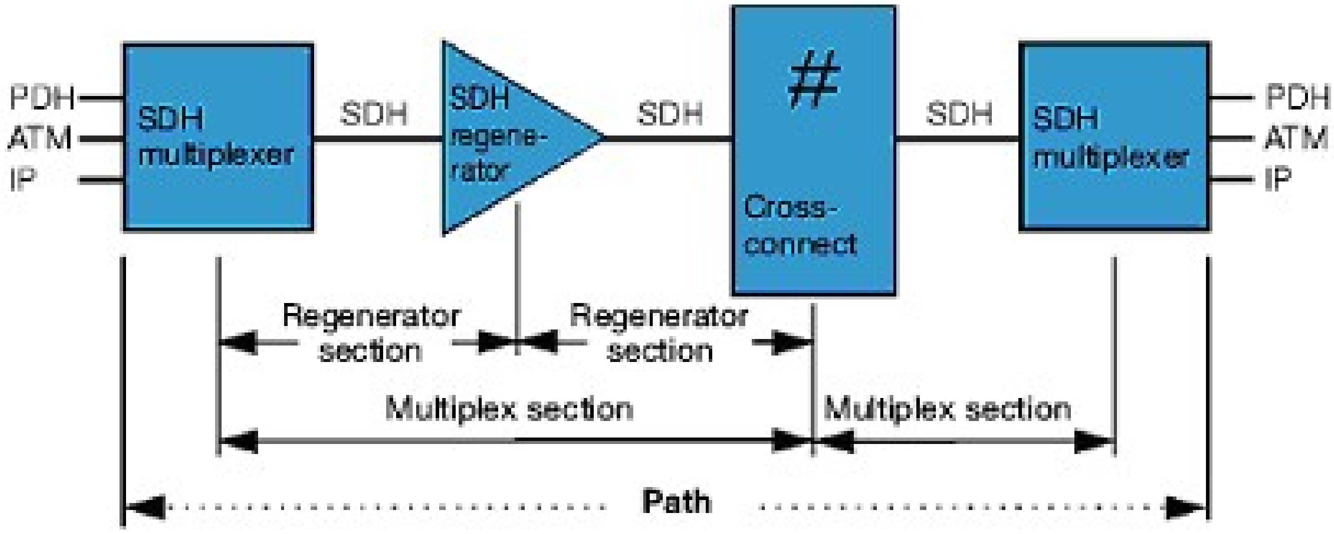
2. Simplified Add / Drop function: Compared with the older PDH system, it is much easier in SDH to extract and insert low-bit rate channels from or into the high-speed bit streams. It is not necessary to demultiplex and then remultiplex the plesiochronous structure.

3. Future-proof platform for new services: Can serve ranging from POTS, ISDN, mobile radio, data communications (LAN, WAN, etc.), video on demand, digital video broadcasting via ATM

- Driving forces behind the future telecommunications are:

  - Ever growing demand for more bandwidth (such as STM-64, i.e 10 Gbps)
  - Better quality of service and reliability
  - Lower costs

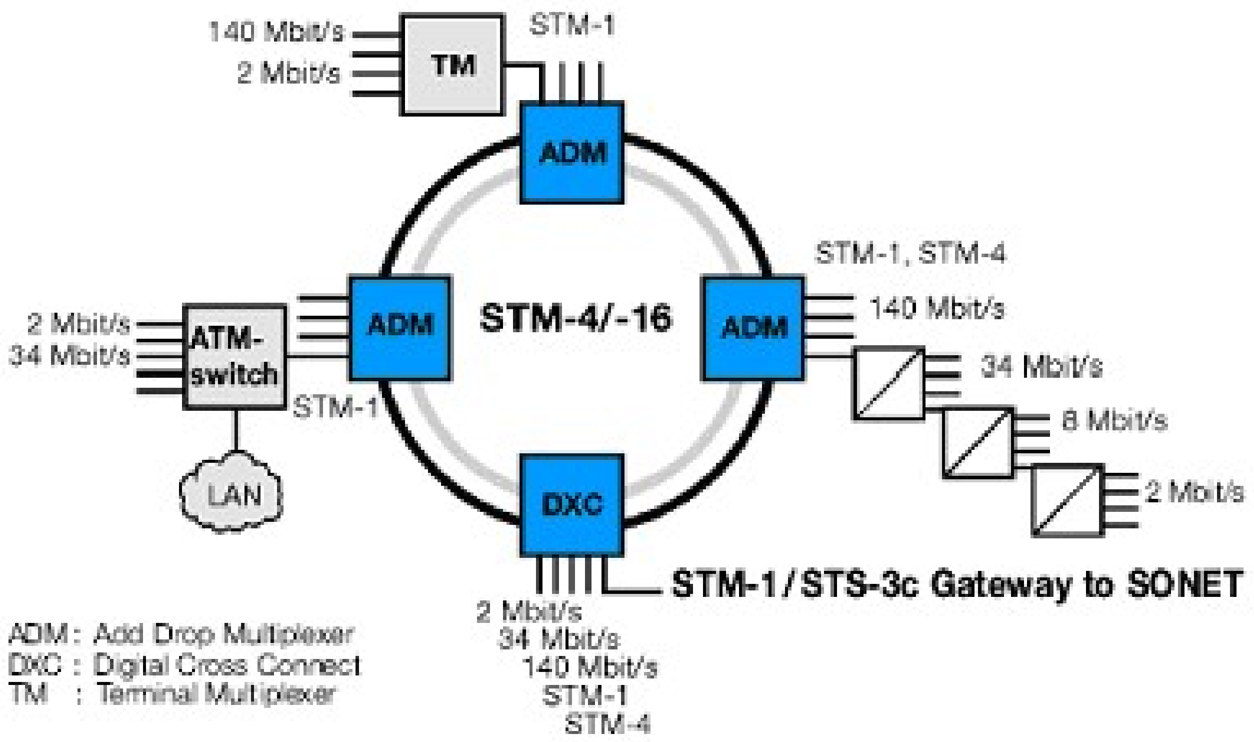
<u>SDH in terms of layer model:</u>



Path section designations

- SDH networks are subdivided into various layers
- The lowest layer is the physical layer, which represents the transmission medium. This is usually fiber or radio-link or satellite link
- Regenerator section is the path between regenerators
- Part of the overhead (RSOH, Regenerator Section Overhead) is available for the signaling required within Regenerator layer
- Remainder of the overhead (MSOH, Multiplex Section Overhead) is used for the needs of the multiplex section.
- Multiplex section is the part of the SDH link between multiplexers
- The carriers (VC, virtual containers) are available as payload at the two ends of Multiplex section
- Two VC layers represent a part of the mapping process
- Mapping is the procedure whereby the tributary signals, such as PDH and ATM signals are packed into the SDH transport modules
- VC-4 mapping is used for 140 Mbit/s or ATM signals and VC-12 mapping is used for 2 Mbit/s signals
- Uppermost layer represents applications of the SDH transport network.

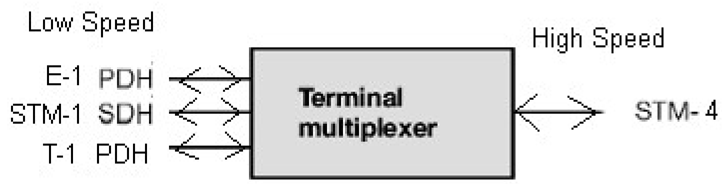
<u>Components of a synchronous network</u>



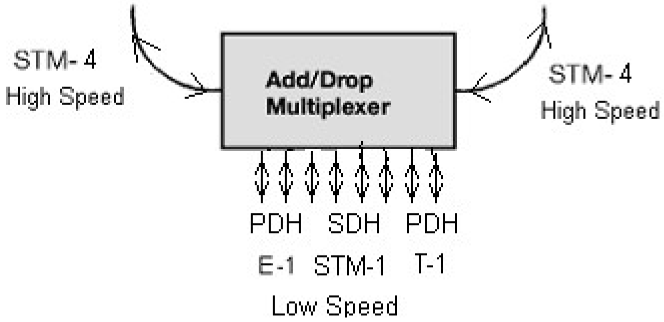
- Above Figure is a schematic diagram of a SDH ring structure with various tributaries
- Mixture of different applications is typical of the data transported by SDH
- Synchronous networks must be able to transmit plesiochronous signals and at the same time be capable of handling future services such as ATM
- Current SDH networks are basically made up of four different types of network elements in the topology (i.e. ring or mesh structure)

1. Regenerators regenerate the clock and amplitude relationships of the incoming data signals that have been attenuated and distorted by dispersion. Regenerators derive their clock signals from the incoming data stream. Messages are received by extracting various 64 kbit/s channels (e.g. service channels E1) in the RSOH (regenerator section overhead). Messages can also be output using these channels.

2. Terminal multiplexers are used to combine plesiochronous and synchronous input signals into higher bit rate STM-N signals.



3. Add/drop multiplexers (ADM): Plesiochronous and lower bit rate synchronous signals can be extracted from or inserted into high speed SDH bit streams by means of ADMs. This feature makes it possible to set up ring structures, which have the advantage that automatic back-up path switching is possible using elements in the ring in the event of a fault.



4. Digital crossconnects (DXC or DCS) direct and manage traffic from a multiplicity of sources at different speeds world-wide, continuously.

DXC allows:

- Mapping of PDH tributary signals into virtual containers

- Switching of various containers up to and including VC-4.

e.g. DCS converts between T1 and E1 data and signaling, it cross connects:

- Several fractional E-1 channels to single E-1

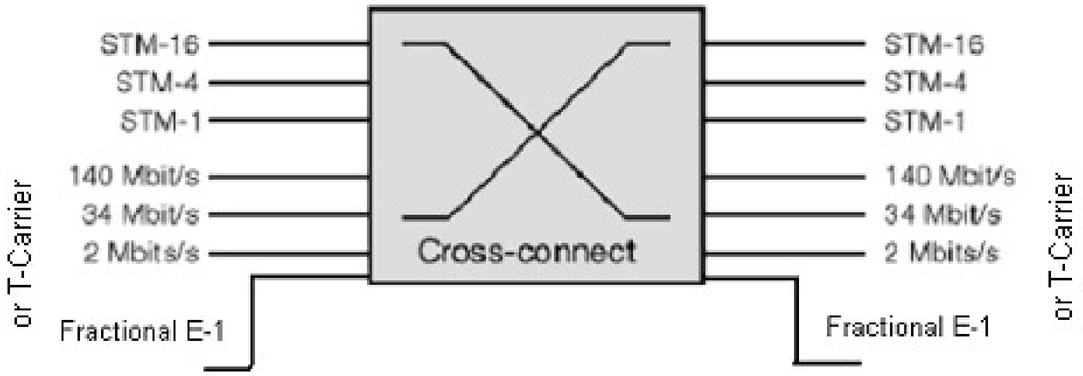
- Several fractional T-1 channels to single T-1

- DS-0s, "n" x 64Kbps consecutive data channels and fractional E-1 channels to full E-1 channels

- A 63 E-1 DS-0, using either input streams from the three 21-E1 or the input stream from the single STM-1

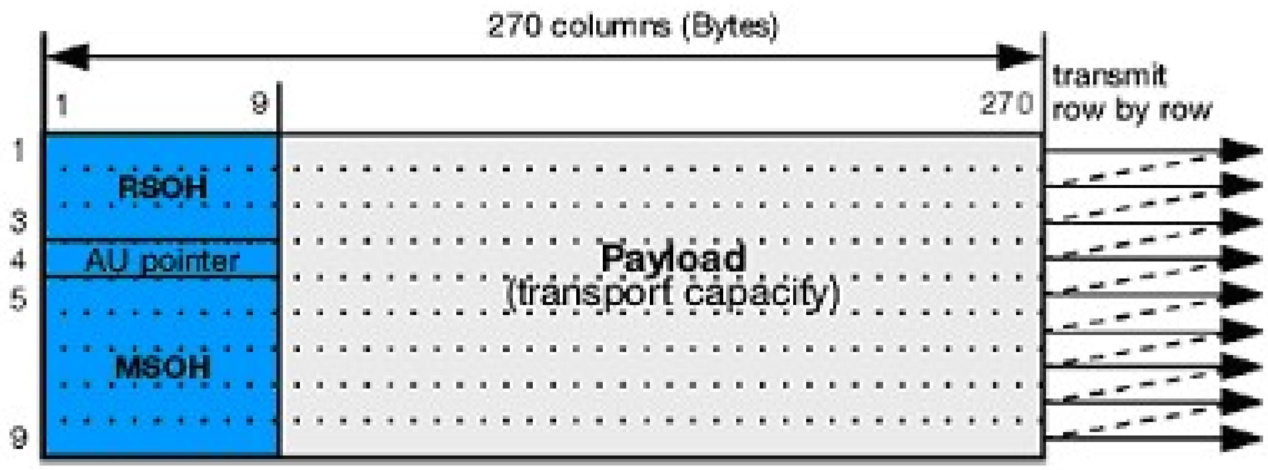
- Multiplex to combine 63 E-1 from the low speed inputs into a single STM-1/OC-3 stream. Similarly, 84 T-1 combined into an STM-1/OC-3 stream.

- As many as four OC-192 (10 Gbps) streams or 16 OC-48 (2.5 Gbps) streams,



The STM-1 frame format

- The frame with a bit rate of 155.52 Mbit/s is defined as the first level of Synchronous Transport Module (STM-1)



STM-1 frame

- STM-1 frame is made up from a byte matrix of 9 rows and 270 columns
- STM-1 frame carries 9 rows X 270 columns = 2430 bytes (one byte is 8 bits)
- One STM-1 frame is transmitted every 0.000125 seconds (1/8000th of a second)
- Thus rate of STM-1 is (2430 bytes x 8 bits / byte) = 19440 bits in 0.000125 seconds. İ.e 19440 bits x (1/0.000125 sec) = 155.52 Mbps
- Transmission is row by row, starting with the byte in the upper left corner and ending with the byte in the lower right corner, i.e in line order just like the order of reading text in a book.
- First 9 bytes in each of the 9 rows (=81 bytes) are called the **Transport Overhead** (except row 4); Regenerator Section Overhead (RSOH) and Multiplex Section Overhead (MSOH).
- Section overhead allows control information to be passed between adjacent synchronous network elements.

- Columns 10 through 12 are reserved for **Path Overhead (POH)**.

- Remaining columns (13 through 270) comprise the **Synchronous Payload Envelope (SPE)** for an actual data rate of 148.608 Mbps (258 columns x 9 rows x 8 bits/byte x 8000 frames/sec = 148.608 Mbps).

- There is no restriction on STM payloads. Some of the payloads are:

  - PDH frames

  - Broadband ISDN ATM cells

  - Narrowband ISDN data

  - LAN packets

  .

- Each byte in the payload represents a 64 kbit/s channel (like in TDM)

- Individual payloads are loaded into the SDH frame by placing them in **Virtual Containers (VCs)** within the frame.

- Any **Virtual Container** consists of the payload information placed within it plus a one column field called **Path Overhead (POH).** İ.e. path overhead (POH) plus a container forms a virtual container

- The POH stays with the container contents from the point where the VC is entered into the STM frame to the point where the payload is extracted and delivered to the end user.

- POH enables the SDH system to identify a VC (indicating the type of container) as it passes through various stages and also to monitor quality.

- As the term virtual container suggests, the bytes of the VC are not loaded contigously into an STM frame <u>but are distributed throughout the frame</u>.

- In this manner, payload area is used efficiently, i.e. individual VCs can be written into and read out of the payload area in a periodic manner

- A virtual container may also start in one frame and finish in the next frame.

- POH format and size depend on the container type. Two different POH types:

- SDH is used everywhere except in the USA, Canada and Japan.

- SONET is the American equivalent of SDH. SONET base bit rate is 51.84 Mbit/s and is designated STS-1 (synchronous transport signal).

- If this bit rate is transmitted over an optical cable system, the signal is designated OC-1 (optical container).

- Levels in SDH/SONET hierarchy are:

SONET signals		Bit rates	Equivalent SDH signal
STS-1	OC-1	51.84 Mbit/s	STM-0
STS-3	OC-3	155.52 Mbit/s	STM-1
STS-9*	OC-9*	466.56 Mbit/s	
STS-12	OC-12	622.08 Mbit/s	STM-4
STS-18*	OC-18*	933.12 Mbit/s	
STS-36*	OC-36*	1244.16 Mbit/s	
STS-48*	OC-48	2488.32 Mbit/s	STM-16
STS-192*	OC-192	9953.28 Mbit/s	STM-64

(* These hierarchy levels are not normally used and are mentioned only for the sake of completeness)

Circuit Switching

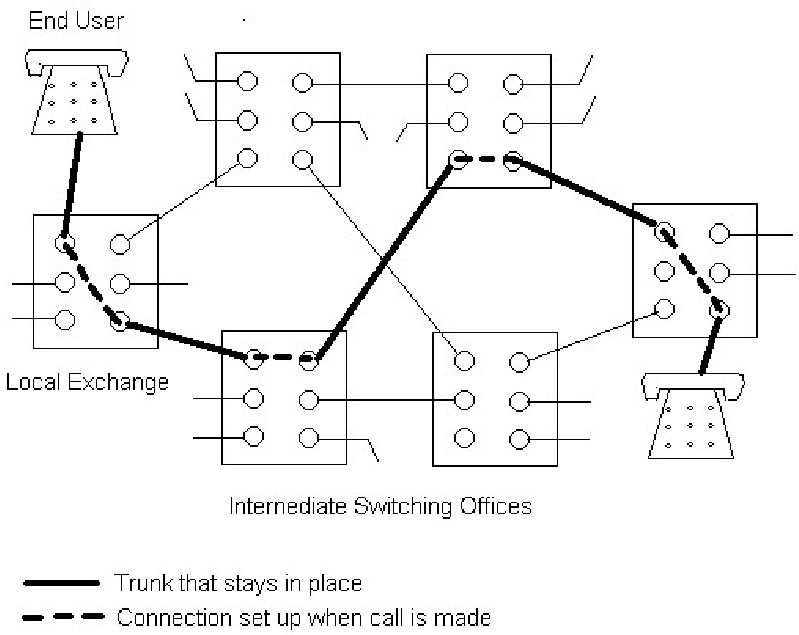
- Forms the basis of the classical telephone networks

- In the circuit switching:

  - When requested by the end user (for example when the user dials up the phone), a circuit is formed between the calling and the called party

  - A fixed share of the network resources for that connection are reserved for this specific communication during the full duration of conversation. İ.e no other call can use those resources until the communication ends. This means that the capacity provisioned on that specific path can only be used by this call, no one else can share or use the capacity available on that path

  - When the conversation is over, connection is released, i.e the circuit is disconnected.



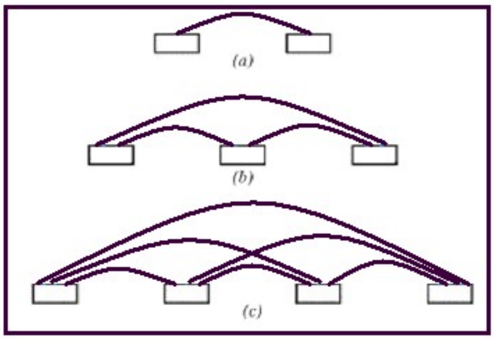
Local Area Networking (LAN)

Direct point-to-point communication

- Computers connected by communication channels that each connect exactly two computers
- Forms *mesh* or *point-to-point* network
- Allows flexibility in communication hardware, packet formats, etc.
- Provides security and privacy because communication channel is not shared

Connections in a point-to-point network

- Number of wires grows as square of number of computers



- For n computers: Connections = $(n^2 - n) / 2$
- Adding a new computer requires (n – 1) new connections

Reducing the number of communication channels

- LANs developed in late 1960s and early 1970s
- Key idea in LAN development is to reduce number of connections by *sharing* connections among many computers
- Computers take turns - TDM
- Synchronizing the use

Growth of LAN technologies

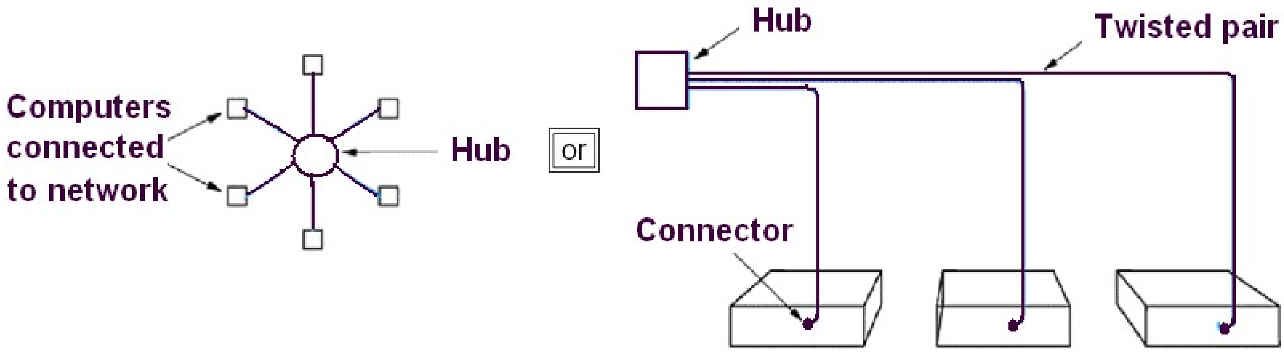
- LAN technologies reduce cost by reducing number of connections
- Attached computers compete for use of shared connection
- Local communication is almost exclusively by LAN

LAN topologies

Networks may be classified by shape: 3 most popular topologies:
- Star
- Ring
- Bus

Star topology

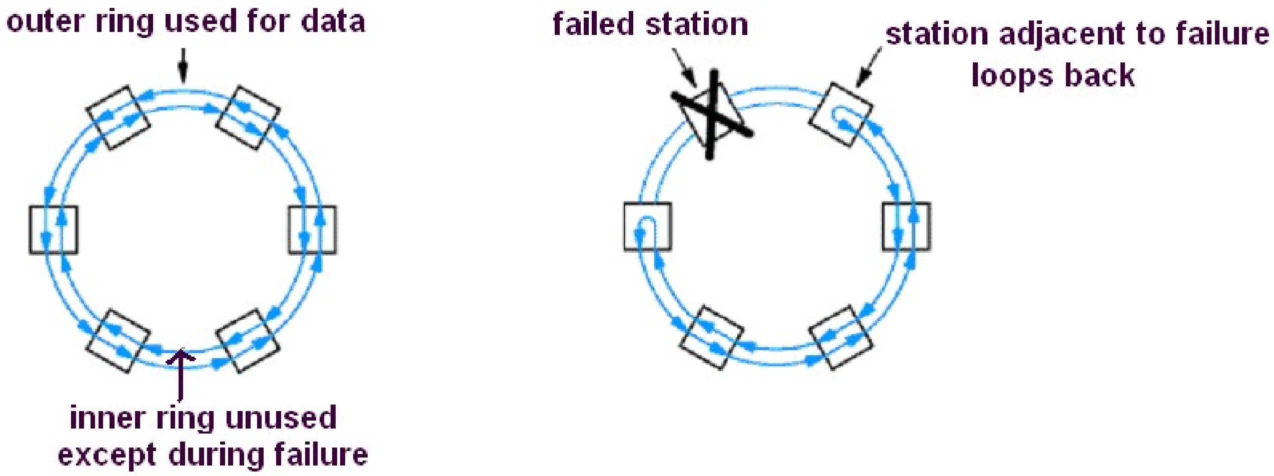
All computers attach to a central point:



Center of star is sometimes called a *hub*

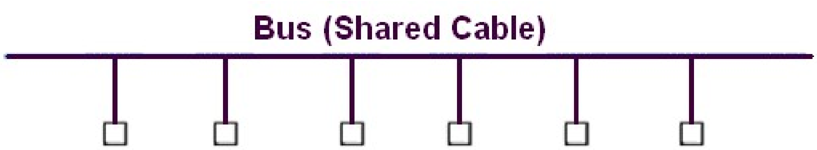
Ring topology

- Computers connected in a closed loop
- First passes data to second, second passes data to third, and so on
- In practice, there is a short connector cable from the computer to the ring
- Ring connections may run past offices with connector cable to socket in the office



Bus topology

- Single cable connects all computers
- Each computer has connector to shared cable
- Computers must synchronize and allow only one computer to transmit at a time



Why multiple topologies?

Each has advantages and disadvantages:
- Ring topology eases synchronization but may be disabled if any cable is cut
- Star is easier to manage but requires more cables

- Bus requires fewer cables but may be disable if cable is cut

Ethernet

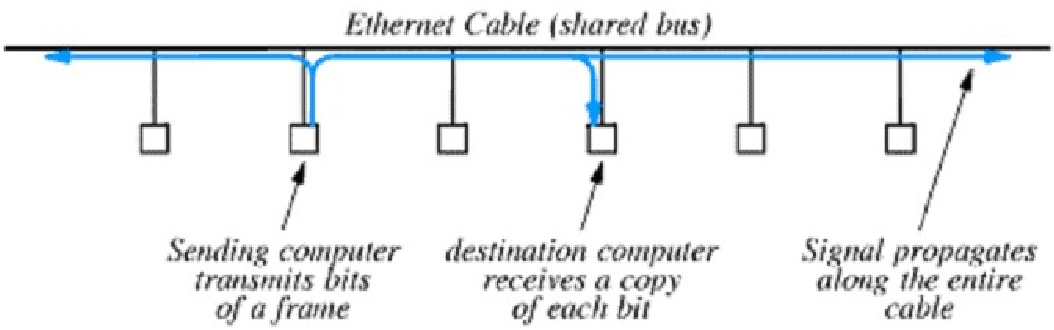
- Widely used LAN technology
- Standard is managed by IEEE - defines formats, voltages, cable lengths, ...
- Uses bus topology
- Multiple computers connect to a single coax cable - the *ether*
- One Ethernet cable is sometimes called a *segment*
- Limited to 500 meters in length
- Minimum separation between connections is 3 meters

Ethernet speeds

- Originally 3Mbps
- Previous standard is 10Mbps
- *Fast Ethernet* operates at 100Mbps
- Super fast at 10 Gbps.

Ethernet operation

- One computer transmits at a time
- Signal is a modulated carrier which propagates from transmitter in both directions along length of segment



CSMA (Carrier Sense with Multiple Access)

- No central control managing when computers transmit on ether
- Ethernet employs CSMA to coordinate transmission among multiple attached computers
- Carrier sense - computers want to transmit tests ether for carrier before transmitting
- Multiple access - multiple computers are attached and any computer can be transmitter

Collision detection - CD

- Even with CSMA, two computers may transmit simultaneously
- Both check ether at same time, find it idle, and begin transmitting
- Window for transmission depends on speed of propagation in ether
- Signals from two computers will interfere with each other
- Overlapping frames is called a *collision*
- No harm to hardware
- Data from both frames is garbled

Ethernet CD (Collision Detection)

- Ethernet interfaces include hardware to detect transmission
- Garbled signal is interpreted as a collision
- After collision is detected, computer stops transmitting
- So, Ethernet uses CSMA/CD to coordinate transmissions

Recovery from collision

- Computer that detects a collision sends special signal to force all other interfaces to detect collision
- Computer then waits for ether to be idle before transmitting
- If both computers wait same length of time, frames will collide again
- Standard specifies maximum delay, and both computers choose random delay less than maximum
- After waiting, computers use carrier sense to avoid subsequent collision
- Computer with shorter delay will go first

Exponential backoff

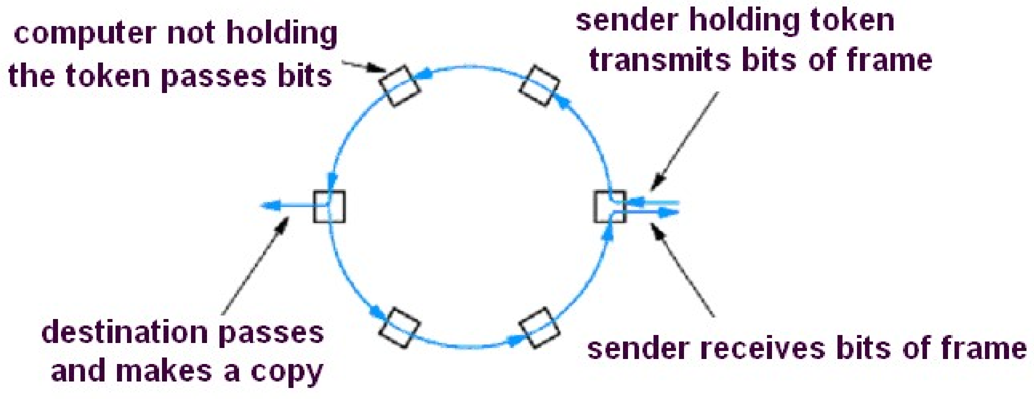
- Even with random delays, collisions may occur, especially likely with busy segments
- Computers double delay with each subsequent collision
- Reduces likelihood of sequence of collisions

Token ring

- Many LAN technologies that use ring topology use *token passing* for synchronized access to the ring
- Ring itself is treated as a single, shared communication medium
- Bits pass from transmitter, past other computers and are copied by destination
- Hardware must be designed to pass token even if attached computer is powered down

Using the token

- When a computer wants to transmit, it waits for the *token*
- After transmission, computer transmits token on ring
- Next computer ready to transmit receives token and then transmits
- Because there is only one token, only one computer will transmit at a time
- Token is short, reserved frame that cannot appear in data
- Hardware must regenerate token if lost
- Token gives computer permission to send one frame
- If no computer is ready to transmit, token circulates around ring



FDDI

- Fiber Distributed Data Interconnect (FDDI) is another ring technology
- Uses fiber optics between stations
- Transmits data at 100Mbps
- Uses pairs of fibers to form two concentric rings

FDDI and reliability

- FDDI uses *counter-rotating* rings in which data flows in opposite directions
- In case of fiber or station failure, remaining stations *loop back* and reroute data through spare ring
- All stations automatically configure loop back by monitoring data ring

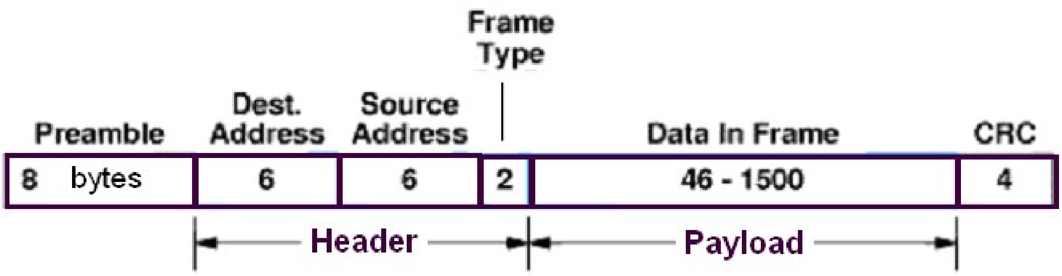
Headers and frame formats

- LAN technology standards define frame format for each technology
- All contemporary standards use the following general format:

- Frame header has address and other identifying information
- Information typically in *fields* with fixed size and location
- Data area may vary in size

Example frame format

Ethernet frame format:



Field	Purpose
Preamble	Receiver synchronization
Destination address	Identifies intended receiver
Source address	Hardware address of sender
Frame type	Type of data carried in frame
Data	Frame payload
CRC	32-bit CRC code

Ethernet fields

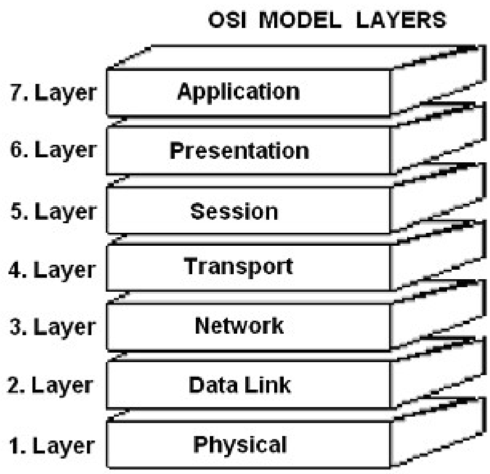
- Preamble and CRC often not shown
- Destination address of all 1s is the broadcast address (message is to all the computers)

- Special values are reserved for frame type field:

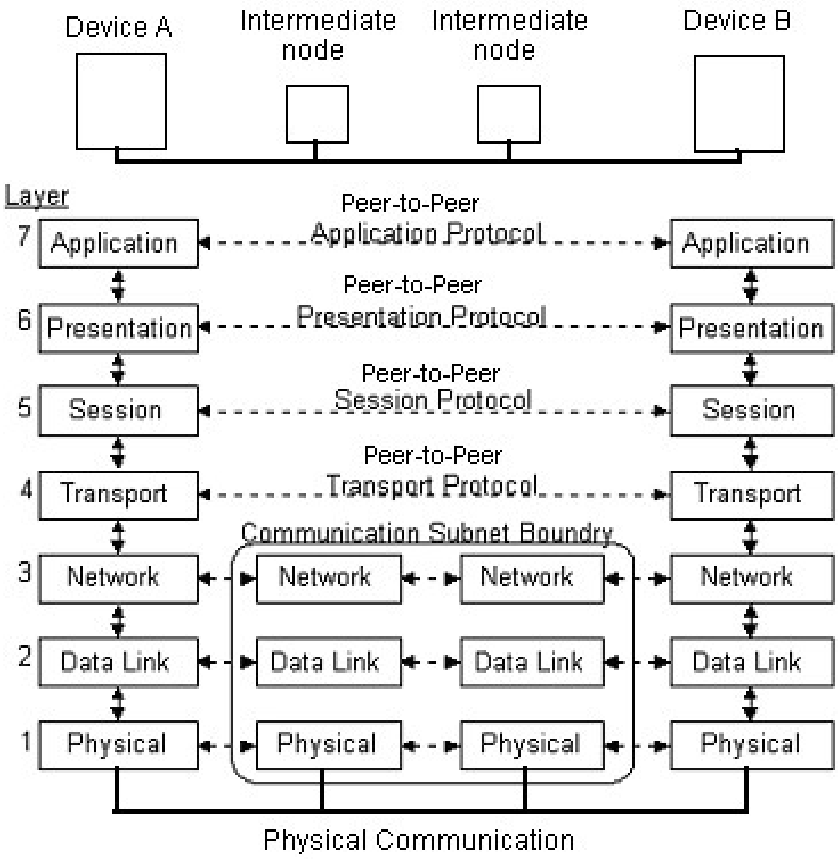
Value	Meaning
0000-05DC	Reserved for use with IEEE 802.3
0800	Internet IP Version 4
0805	CCITT X.25
0900	Ungermann-Bass Corporation network debugger
0BAD	Banyan Systems Corporation VINES
1000-100F	Berkeley UNIX Trailer encapsulation
6004	Digital Equipment Corporation LAT
6559	Frame Relay
8005	Hewlett Packard Corporation network probe
8008	AT&T Corporation
8014	Silicon Graphics Corporation network games
8035	Internet Reverse ARP
8038	Digital Equipment Corporation LANBridge
805C	Stanford University V Kernel
809B	Apple Computer Corporation AppleTalk
80C4-80C5	Banyan Systems Corporation
80D5	IBM Corporation SNA
80FF-8103	Wellfleet Communications
8137-8138	Novell Corporation IPX
818D	Motorola Corporation
FFFF	Reserved

10Base-T *network topology* is a bus; *wiring topology* is a star
Token ring *network topology* is a ring; *wiring topology* is a star

<u>OSI 7-Layer Model</u>



- OSI consists of 7 separate but related layers
- Each of the layers defines a segment of process of moving information across a network
- Below Figure shows the layers involved when a message is sent from Device A to Device B

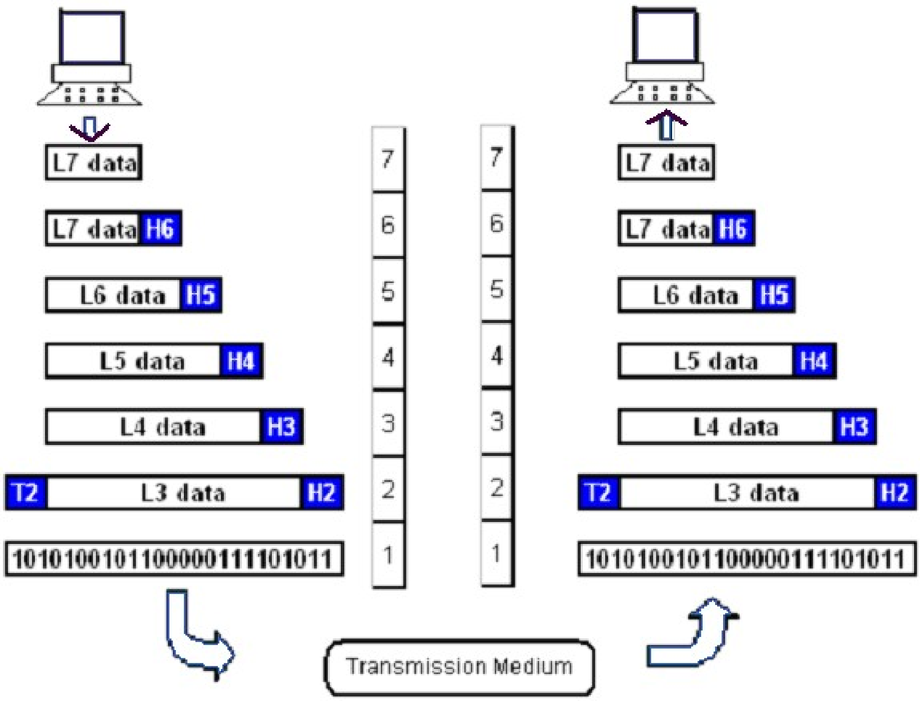


- As the message travels from A to B it may pass through many intermediate nodes.
- These intermediate nodes usually involve only the first 3 layers of the OSI model.
- Each layer defines a family of functions distinct from those of the other layers.
- By defining and localizing functionality in this manner, comprehensive and flexible architecture is designed.
- OSI model provides transparency between otherwise incompatible systems.

Peer-to-Peer Process

- Within a single device, each layer uses
  - The services provided by the layer just below it
  - Provides services for the layer just above it
- Between devices, layer x on one machine communicates with layer x on another device.
- This communication is made possible by using the peer-to-peer protocols (i.e.rules and conventions) appropriate to that layer x.
- Processes on each device that communicate at a given layer are called peer-to-peer processes.
- At the physical layer (layer 1), communication is direct, i.e. device A sends a bit stream to machine B.

- At the higher layers, communication must move down through the layers on device A, over to device B, and then back up through the layers.
- Each layer in the sending device:
  - Adds its own information to the message it receives from the layer just above it
  - Passes the whole package to the layer just below it
- This information is added in the form of headers or trailers.
  - Header is the control data appended to the beginning of the data parcel. Headers are added at layers 6, 5, 4, 3 and 2.
  - Trailer is the control data appended to the beginning of the data parcel. Trailer is added at layer 2.
- At layer 1 the entire package is converted to a form that can be transferred to the receiving device
- At the receiving device, the message is unwrapped layer by layer, each process receiving and removing data meant for it. E.g:
  - Layer 2 removes the data meant for it then passes the rest to layer 3,
  - Layer 3 removes the data meant for it then passes the rest to layer 4,
  - And so on...
- Overall view of OSI layers are shown in the below Figure:



- Process starts at layer 7 and moves from layer to layer in descending sequential
- L7 data means the data unit at layer 7, ...

- At each layer a header is added to the data unit.

- At layer 2, a trailer is also added.

- When formatted data unit passes through the physical layer (layer 1), data is converted to electromagnetic signal and transported along the transmission medium.

- When electromagnetic wave reaches its destination, signal passes into layer 1 and transformed back into digital form.

- Data units move up the OSI layers.

- As each block of data reaches the next higher layer, the headers and and trailers attached to it at the corresponding sending layers are removed and actions appropriate to that layer are taken.

- When the data unit reaches layer 7, message is again in a form appropriate to the application and is made available to the recipient.

Basic Functions of Layers

- **Physical Layer (Layer 1)**

  - Connects the entity to the transmission media

  - Describes the physical properties of the various communications media, as well as the electrical properties and interpretation of the exchanged signals.

  - E.g. this layer defines the size of Ethernet coaxial cable, the type of BNC connector used, and the termination method.

- **Data Link Layer (Layer 2)**

  - Responsible for node-to-node delivery.

  - Provides error control between adjacent nodes.

  - Headers and trailers added at this layer include physical addresses of the most recent node and the next intended node.

  - Regulates the amount of data that can be transmitted at one time in order not to overwhelm the receiver.

  - E.g. this layer defines the framing, addressing and checksumming of Ethernet packets.

- **Network Layer (Layer 3)**

  - Responsible for the routing and delivery of packets across multiple network links from source to destination.

  - Describes how a series of exchanges over various data links can deliver data between any two nodes in a network.

- E.g. this layer defines the addressing and routing structure of the Internet

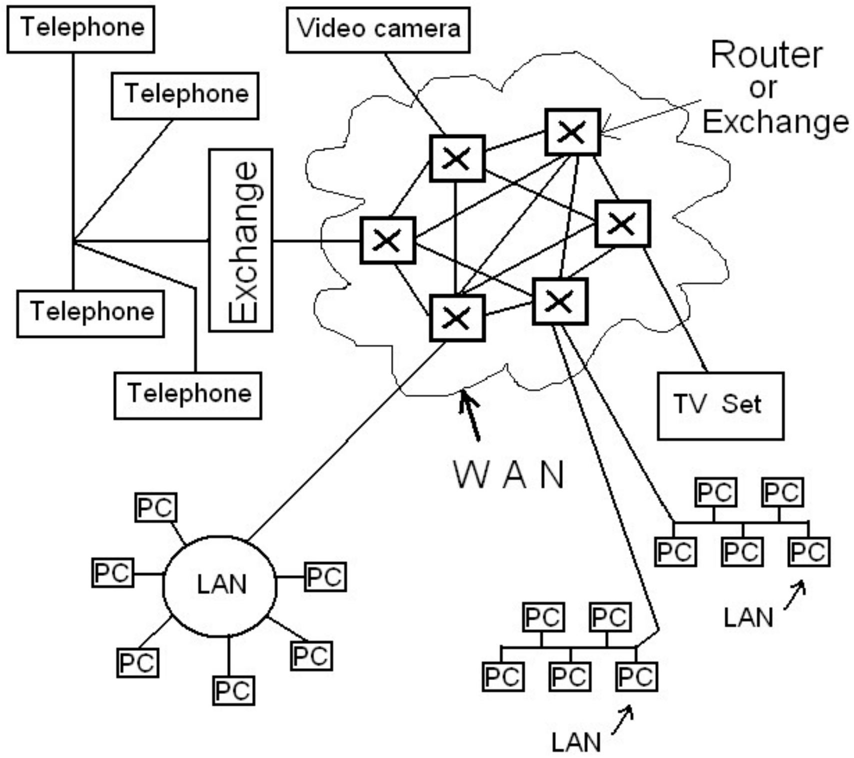
- **Transport Layer (Layer 4)**
  - Provides end to end communication control.
  - Describes the quality and nature of the data delivery. Divides the message into transmittable segments and marks with a sequence number in order to correctly reassemble the message at the destination and to identify and replace packets lost in transmission
  - E.g. this layer defines if and how retransmissions will be used to ensure data delivery.

- **Session Layer (Layer 5)**
  - Establishes, maintains and synchronizes the interaction between communicating devices.
  - Achieves dialog control, decides who sends and when
  - Ensures that the information exchange is completed appropriately before the session closes.

- **Presentation Layer (Layer 6)**
  - Makes necessary translation of different control codes to ensure interoperability among communicating devices.
  - Also responsible from encryption and decryption of data
  - Describes the syntax of data being transferred.
  - E.g. this layer describes how floating point numbers can be exchanged between hosts with different math formats.

- **Application Layer (Layer 7)**
  - Enables the user (human or software) to access the network.
  - Provides different services to the applications (such as file access, transfer and management, mail forwarding and storage.
  - Provides distributed database sources and access for global information about various services.

Wide Area Networking (WAN)



- Local Area Network (LAN) may support a majority of communication and resource-sharing needs within an enterprise or campus setting

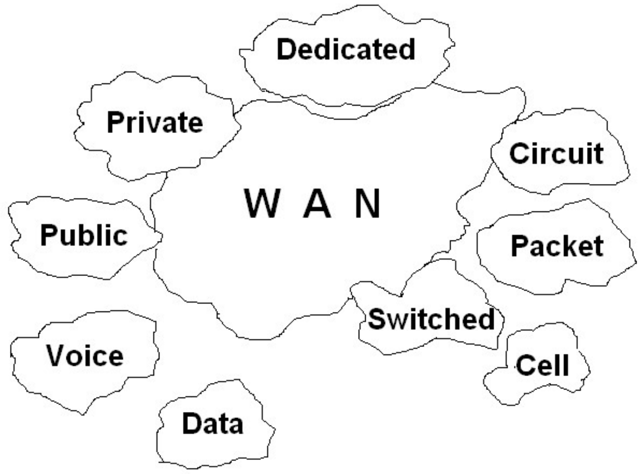
- Wide Area Network (WAN) connectivity allows individuals and organizations to take further advantage of internetworking services such as the Internet, e-commerce, and videoconferencing.

- By definition, a Wide Area Network (WAN) is a government-regulated public network or privately owned network that crosses into the public network environment. It doesn't matter whether the area being bridged is across the country or across the street.

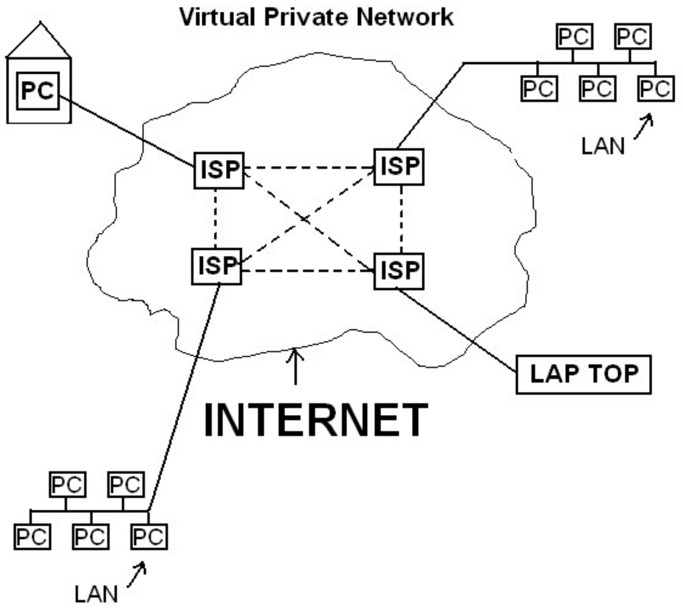
- If the geographical separation crosses over a public network, a WAN is required to make the connection.

- WAN is typically used to connect two or more local area networks (LANs).

- For the attached user, WAN is as a virtual network cloud.



- Example: A private non-lease WAN option is a **Virtual Private Network (VPN)**, which connects distributed LAN locations across the Internet. VPN is shown below.

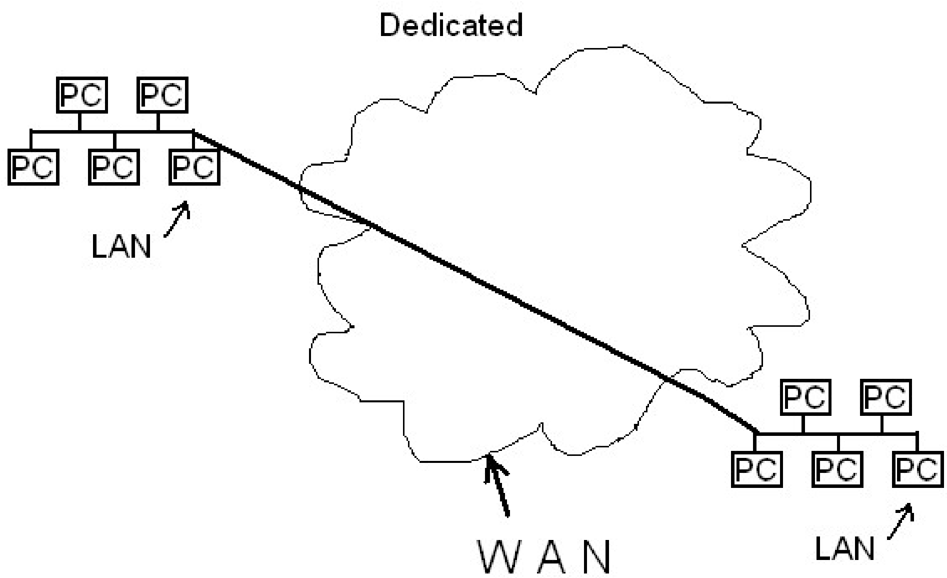


- A public WAN designed for voice is the **Public Switched Telephone Network (PSTN)**.

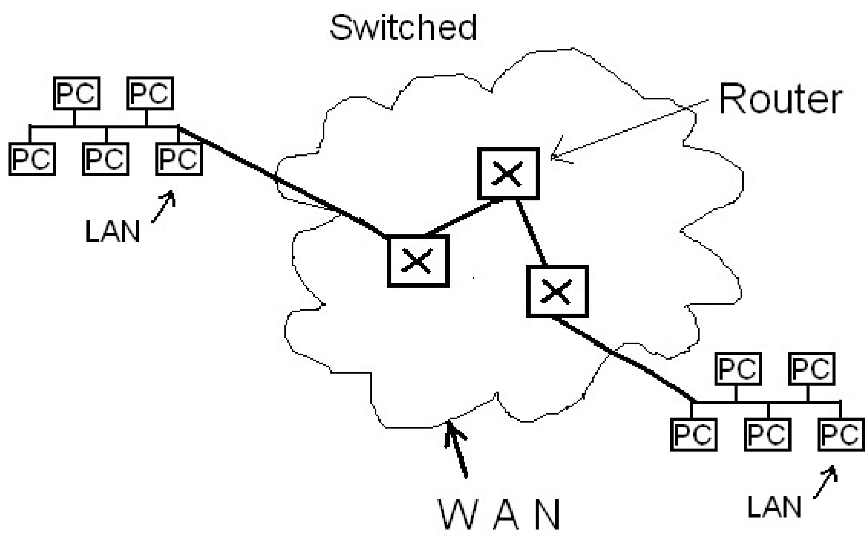
- Internet is the largest public WAN designed for data.

<u>WAN Connection Types</u>

- **Dedicated** connections are links that is reserved for a single telecommunications purpose and available to the user at all times

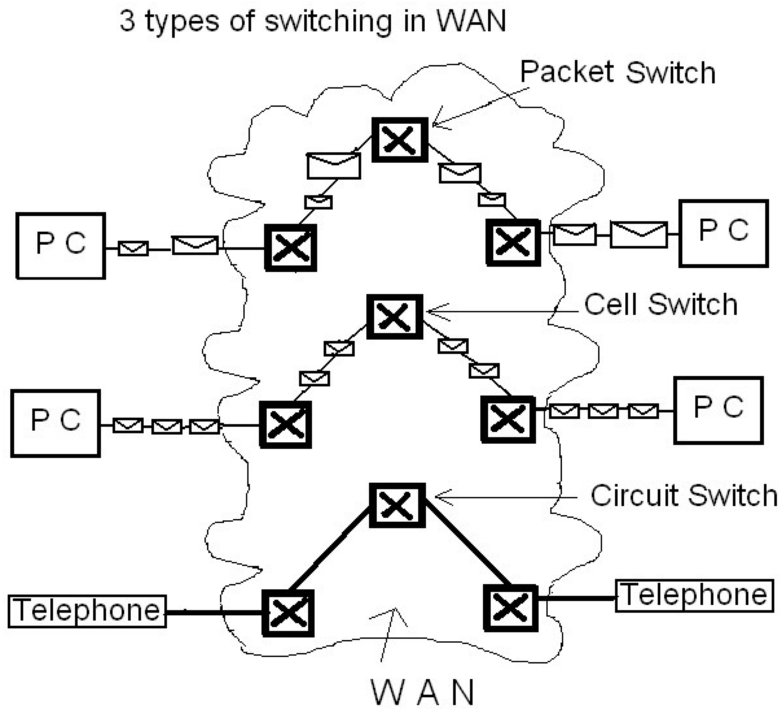


- **Switched** connections are general purpose links that are available on demand and are usually paid for on a per usage basis. Switched connections are commonly found in the PSTN, as well as ISDN, Frame Relay, and ATM networks.



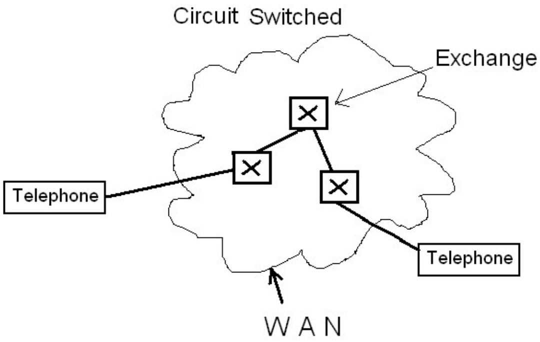
- **Hybrid** connections in which a leased line is needed to make the connection between the customer location and the service provider's Point of Presence (POP). E.g.. X.25 networks are often accessed using a dedicated connection, and then a switched connection is used

<u>Types of switched networks</u>.



- There are 3 types of WAN switching services:

  1. **Circuit-switched** networks create a dedicated circuit, or channel, which is used for the duration of the transmission.



- Circuit-switched networks were originally designed for the transmission of analog voice.

- Circuit-switched networks are **connection-oriented** and call setup is required prior to the exchange of information.

- This temporary point-to-point connection is known either as a *circuit* or *channel*.

- The call path remains constant and **bandwidth is dedicated** throughout the duration of the call.

- Unused bandwidth is not recovered.

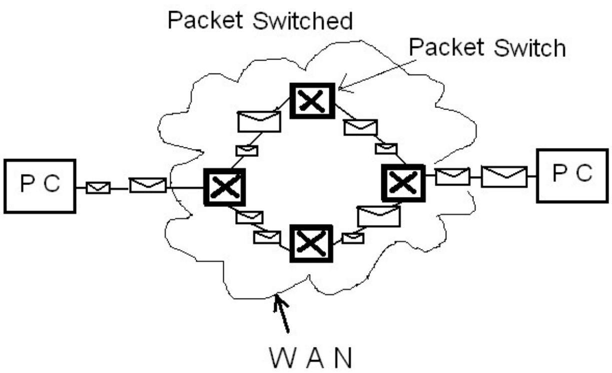
- Traffic is transmitted with minimal delay

- There is **no error recovery** because circuit-based switches maintain only small buffers.

- Circuit-switched network operators typically charge customers for the duration of the connection, which includes any "transmitted silence."

- Examples of a circuit-switched WAN networks are PSTN, ISDN and cellular networks

2. **Packet-switched** networks separate messages into variable-length segments and transmits them individually across dynamically created connections.



- Packet-switching originally developed for sending data over analog circuits, which are subject to errors and noise.

- Examples include internet, X.25 and Frame Relay

- Packet-switched networks require no call setup because each packet contains a destination address that is used to route each packet through the network.

- Dynamically routed connection through the network is known as either a <u>virtual circuit or virtual channel.</u>

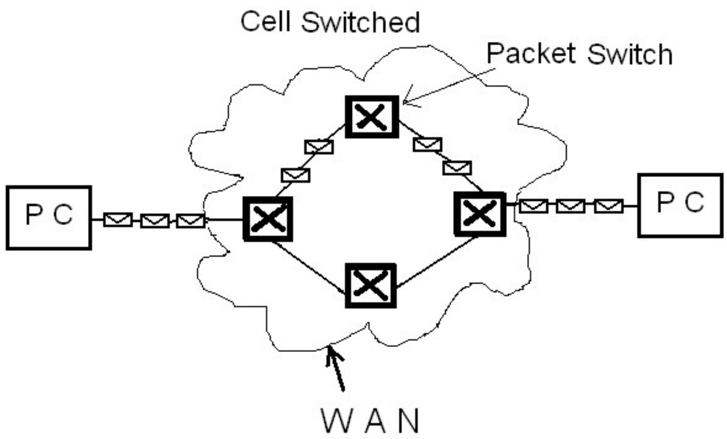
- Dynamic routing results in **flexible use of bandwidth** and network resources.

- Packet switches use a **store-and-forward** technique to carry voice and data through the network.

- Temporary storage of switched packets allows for **error correction** and prioritization.

- Packet-switched network operators charge based on the actual number of packets sent, which means you only pay for data that is transmitted.

3. **Cell-switched** networks separate messages into fixed-length cells and transmits them individually across routed connections that are either dynamically or permanently created.



- ATM networks employ cell-switching, which combines the guaranteed bandwidth of a circuit-switched network with the efficient bandwidth-sharing and prioritization capabilities of a packet-switched network.

- Cell-switched networks are **connection-oriented**, because the receiving end must reply before transmission begins.

- Dynamically routed connections are called Switched Virtual Circuits (SVC).

- Statically routed connections are called Permanent Virtual Circuits (PVCs).

- Logical routing allows for **flexible use of bandwidth** and network resources.

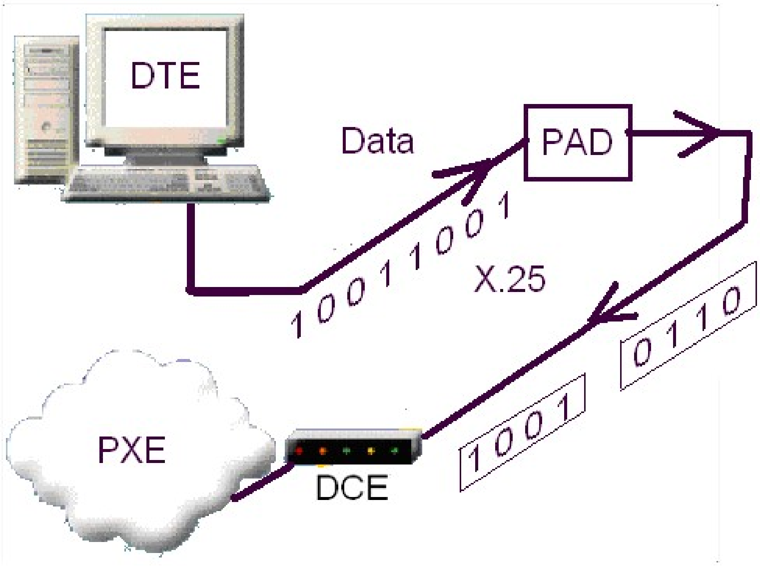
- Logical circuits allow cell-switched networks to guarantee **Quality of Service (QoS)**.

- Cell switches use a **store-and-forward** technique to carry voice and data through the network.

- The temporary storage of switched cells allows for **error detection,** as well as prioritization.

- Cell-switched network operators charge based on the actual number of cells sent, which means you only pay for data that is transmitted.

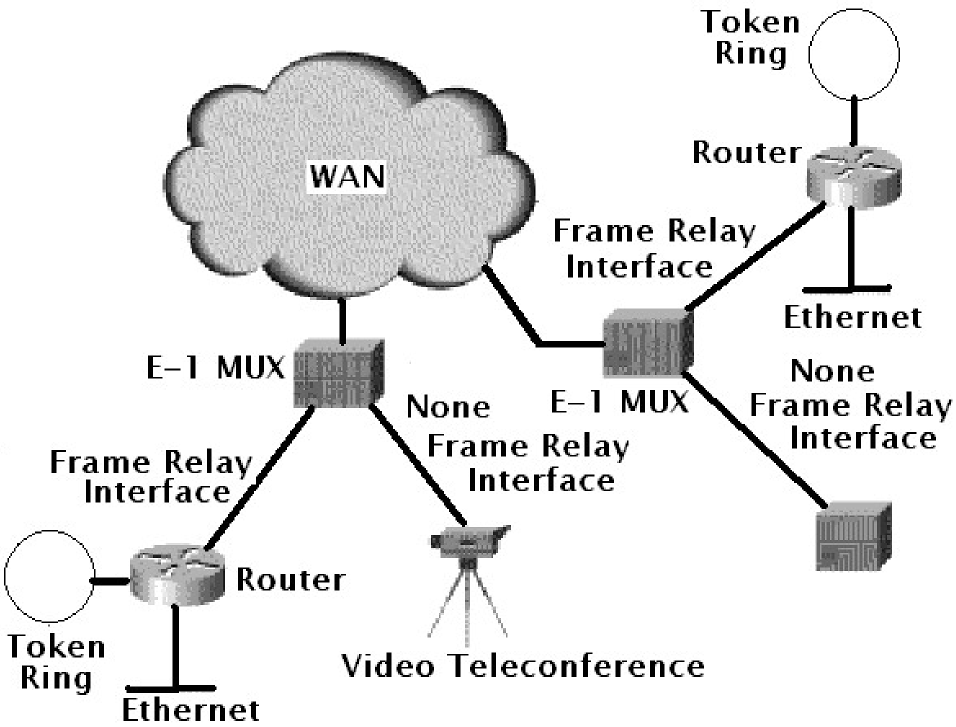
X.25



- **X.25** is a connection-oriented packet switched technology
- Uses the standard protocol of the **Packet Switch Exchange (PXE)**.
- X.25 networks consist of:
  - DTE (Data Terminating Equipment, i.e. terminal)
  - DCE (Data Communications Equipment, i.e. modem)
  - **Packet Assembler / Disassembler (PAD)** that supports packet assembly for outgoing data, packet disassembly for incoming packets
  - Buffering.
- Error checking is performed at each node.
- X.25 is used for data only, not capable of transmitting real-time voice and video.
- No QoS guarantee.
- Used for low-speed (2400 - 19200 bit/sec, 19200 - 64 kbit /sec, 64 kbit - 128 kbit/sec and above) electronic transactions, including database verifications for credit cards and automatic teller machines.
- X.25 has small packet size, generally 128 bytes or 256 bytes long.

Frame Relay

- Frame relay is the second generation (after X.25) of packet switching.



- Frame relay assumes digital infrastructure exists and few errors will result from network noise.

- Therefore entire error detection and correction process is removed from Frame Relay network.

- Error control is done entirely at the endpoints.

- Thus Frame Relay is much faster (mainly up to 2 Mbps but can also operate at 34 Mbps) as compared to X.25.

- Possible to carry voice and video also, however it is not particularly designed for this purpose.

- Frame Relay packet sizes are large and variable (up to 4,096 bytes long), i.e a 100 bytes packet can be followed by a 4,000 bytes packet going through a network node.

- Thus, delay and jitter prediction are not easy, thus QoS is not guaranteed in general

- Main application of Frame Relay is LAN internetworking because provides bandwidth flexibility and cost advantage.

<u>Frame Relay Service Features</u>

Frame Relay:

- Transfers frames of data between two user devices (router) over a permanent virtual circuit (PVC) or switched virtual circuit (SVC).

- Multiplexes / demultiplexes different user data streams within the same access channel through data-link layer addressing.

- Each user data stream within the physical access channel is called a data-link connection (DLC).

- To identify different DLCs within the same channel, each DLC is given a local address called the data link connection Identifier (DLCI).

- There can be connection to different places using PVC/SVC's with different DLCI number within the same physical channel and all frames belonging to a particular connection are transfered over the channel belonging same DLCI number sequentialy.

- Frame Relay service has two main traffic components:

  1. **CIR (Committed Information Rate)**

     CIR is the rate (in bit/s) that the network agrees to transfer information over a virtual circuit under typical conditions. A virtual circuit can be either a permanent virtual circuit (PVC) or switched virtual circuit (SVC).

     CIR applies to the rate of data entering the network.

     The Commited Burst (B_c) is the maximum amount of data (in bits) that a network agrees to transfer under normal conditions over a measurement interval.

     Data may be in the form of one frame or many frames.

     Measurement Interval (T_c) is the time over which rates and burst sizes are measured.

     In general, the duration of T_c is proportional to the burstiness of traffic.

     $CIR = B_c / T_c$

     CIR can be more than 20 different rates starting from 8 Kbps up to 2.048 Mbps.

  2. **EIR (Excess Information Rate)**.

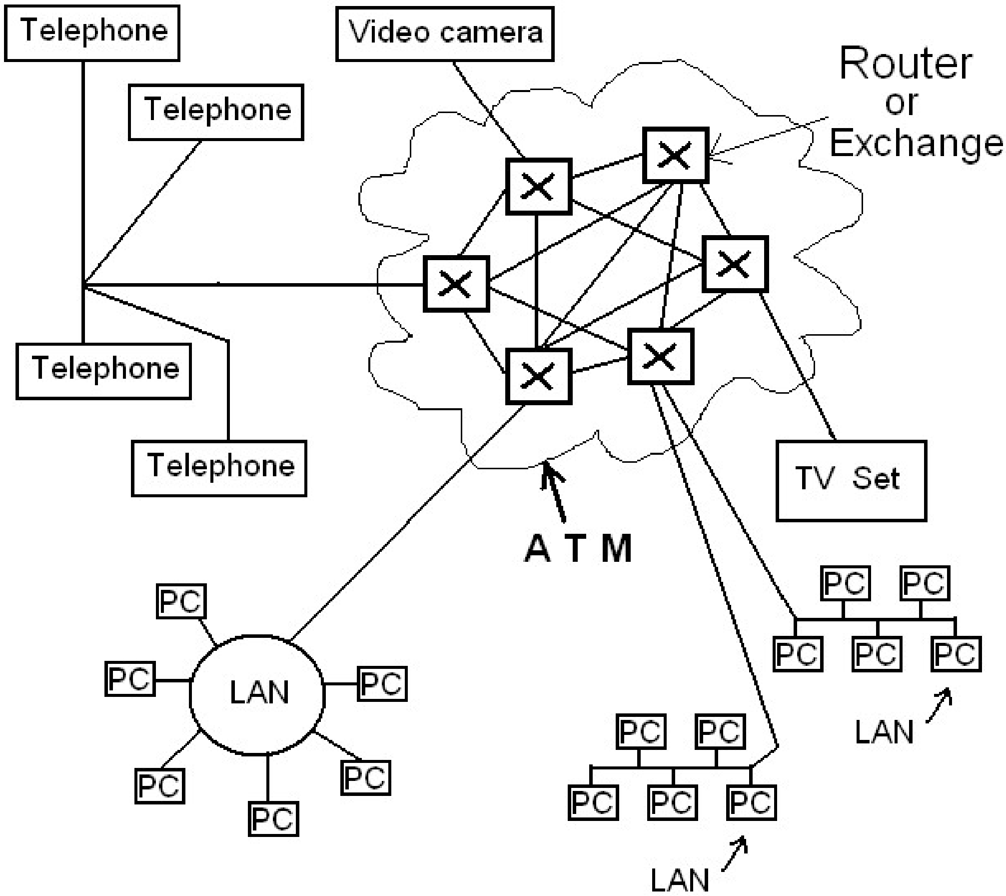
     EIR is the sustainable rate of information in excess of CIR, that the network will deliver if there is available bandwidth.

     $EIR = B_e / T_c$

     Total of information rate is CIR+EIR.

     EIR can be more than 20 different rates starting from 8 Kbps and 1.536 Mbps

<u>ATM (Asynchronous Transfer Mode)</u>



- **Asynchronous Transfer Mode (ATM)** is a high-speed, connection-oriented switching technology that uses fixed-length 53-bytes cells to transmit voice, video, and data traffic simultaneously and reliably.

- ATM networks are dominant within the core of the WAN network, but also occupy prominent positions along the edge.

- Quality of Service (QoS) features allow WAN providers to optimize use of bandwidth and to easily adjust to the delay and loss requirements of specific applications.

- ATM relies on cell-switching technology.

- ATM cells having a fixed length of 53 bytes allows very fast switching.

- ATM creates pathways between end nodes called virtual circuits which are identified by the VPI/VCI (Virtual Path Identifier/ Virtual Channel Identifier) values.

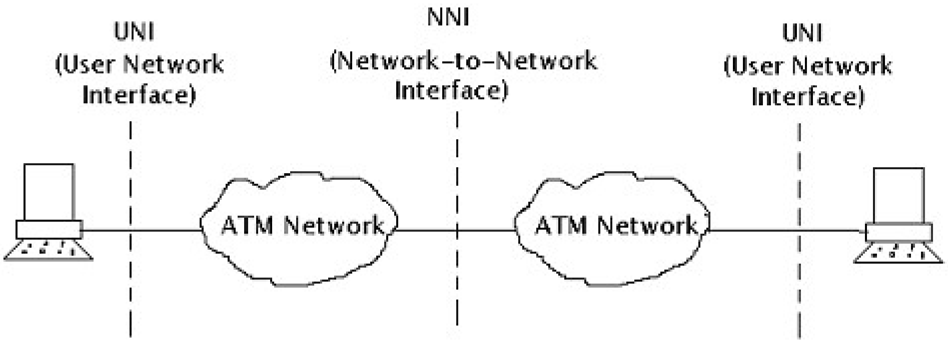
ATM Cell Structure

- ATM cell has fixed size of total 53 bytes length (byte being 8 bits).

- First 5 bytes forms the header

- The remaining 48 bytes comprise the payload of the cell whose format depends on the AAL type of the cell.

- ATM Interfaces are shown below:



- ATM Cell Structures for UNI and NNI Cells are given below:

ATM UNI (User Network Interface)

Byte	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	GFC (Generic Flow Control				VPI (Virtual Path Identifier)			
2	VPI (Virtual Path Identifier)				VCI (Virtual Channel Identifier)			
3	VCI (Virtual Channel Identifier)							
4	VCI (Virtual Channel Identifier)				PTI (Payload Type)			CLP
5	HEC (Header Error Control)							
6 - 53	Payload, i.e Data							

ATM NNI (Network-to-Network Interface)

Byte	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	VPI (Virtual Path Identifier)							
2	VPI (Virtual Path Identifier)				VCI (Virtual Channel Identifier)			
3	VCI (Virtual Channel Identifier)							
4	VCI (Virtual Channel Identifier)				PTI (Payload Type)			CLP
5	HEC (Header Error Control)							
6 - 53	Payload, i.e Data							

CLP is Cell - Loss Priority

- GFC (Generic Flow Control) prevents overload conditions and control traffic flow.
- VPI (Virtual Path Identifier) identifies virtual paths.
- VPI and VCI together indicate the routing information within the ATM cell.
- PTI (Payload Type Identifier) distinguishes between user cells and non-user cells, identifies the payload type carried in the cell and identifies control procedures.
- CLP (Cell Loss Priority) indicates a cells loss of priority. This bit is set to one when a cell can be discarded due to congestion; if a switch experiences congestion, it will drop cells with this bit set. This results in giving priority to certain types of cells carrying certain types of traffic, such as video in congested networks.
- HEC (Header Error Check) is used for detection and correction of 1-bit errors in the cell header, detection of multi-bit-errors in the header.
- Below chart gives the class of services provided by ATM

Class of Service	Applications Supported	Priority
CBR	Real Time Voice and Video	High
Rt-VBR	Compressed Video, Packetized Voice	High
Nrt-VBR	Interactive Data	High
ABR	ATM-oriented Interactive Data	Low
UBR	Bursty Data	Low

Quality of Service (QoS)

- QoS is the ability of an ATM network to define performance levels for user information streams.

- ATM networks specify modes of service that ensure optimum performance for traffic such as real-time voice and video.

- QoS has become a major issue on the Internet as well as in enterprise networks, because voice and video are increasingly traveling over IP-based data networks.

- The following service categories define the traffic attributes required for general types of ATM connections:

Constant Bit Rate (CBR) supports applications that require continuous bandwidth and low delay, such as voice and uncompressed video.

Variable Bit Rate (VBR) supports applications that are less dependent on time, such as packetized voice, compressed video, and data. Like CBR, it provides a guaranteed amount of bandwidth, but at a lower cost.

Real-time Variable Bit Rate (rt-VBR) supports applications that requires end-to-end synchronization, such as compressed video and packetized voice.

Non-real-time Variable Bit Rate (nrt-VBR) supports interactive data applications that are less sensitive to timing, but still require a reliable supply of bandwidth.

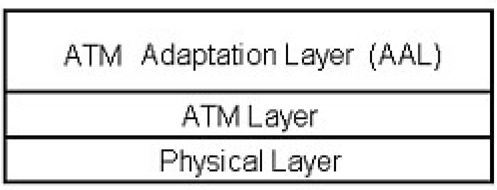
Available Bit Rate (ABR) is designed for ATM-oriented applications that have the ability to adjust the transmission rate during times of network congestion, making use of available bandwidth without violating existing CBR and VBR contracts.

Unspecified Bit Rate (UBR) provides no guaranteed amount of bandwidth or limits on delay, and is therefore best suited for bursty traffic such as batch data. UBR cells may be dropped in order to satisfy existing CBR and VBR contracts.

- More than 20 different rates between 1 Mbps and 622 Mbps can be offered for various service types

- In ATM core networks rate can be 10 Gbps or more.

ATM Layers



- Application Adaptation Layer (AAL) allows existing networks (such as packet networks) to connect to ATM facilities.

- AAL protocols:

- Accept transmissions from upper layer services (e.g. packet data)
- Map them into fixed size ATM cells.

- These transmissions can be:
  - Any type (voice, data, video)
  - Variable or fixed rates.
- At the receiver, this process is reversed, i.e. segments are reassembled into their original formats and passed to ther receiving service.
- AAL layer reformats data from other protocols, acting like a gateway in internetworking.

<u>Data Types</u>

- Each AAL layer supports the requirements of different type of application.
- Four types of data streams are considered in defining AAL categories:

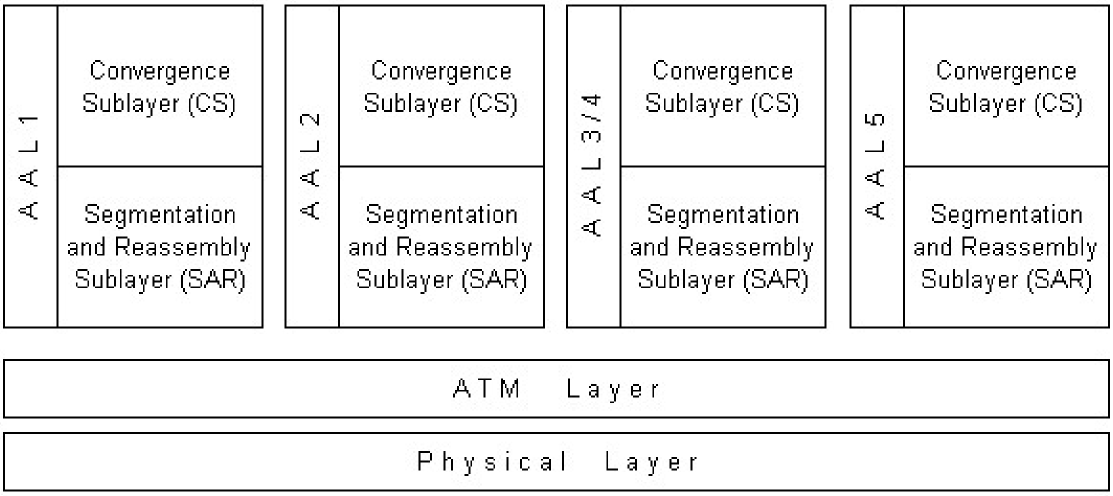
1. Constant bit rate data:
   - Refers to applications that generate and consume bits at a constant rate.
   - Transmission delays must be minimal and transmission must simulate real time
   - Examples are real time voice (telephone calls) and real time video (television)
2. Variable bit rate data:
   - Refers to applications that generate and consume bits at a variable rates.
   - Bit rate varies from section to section of transmission but within established parameters.
   - Examples are compressed voice, data and video
3. Connection oriented packet data:
   - Refers to conventional packet applications (such as X.25 and TCP protocol of TCP/IP) that use virtual circuits.
4. Connectionless packet data:
   - Refers to applications that use datagram approach to routing (such as IP protocol in TCP/IP).

<u>ATM Adaptation Layers (AAL)</u>

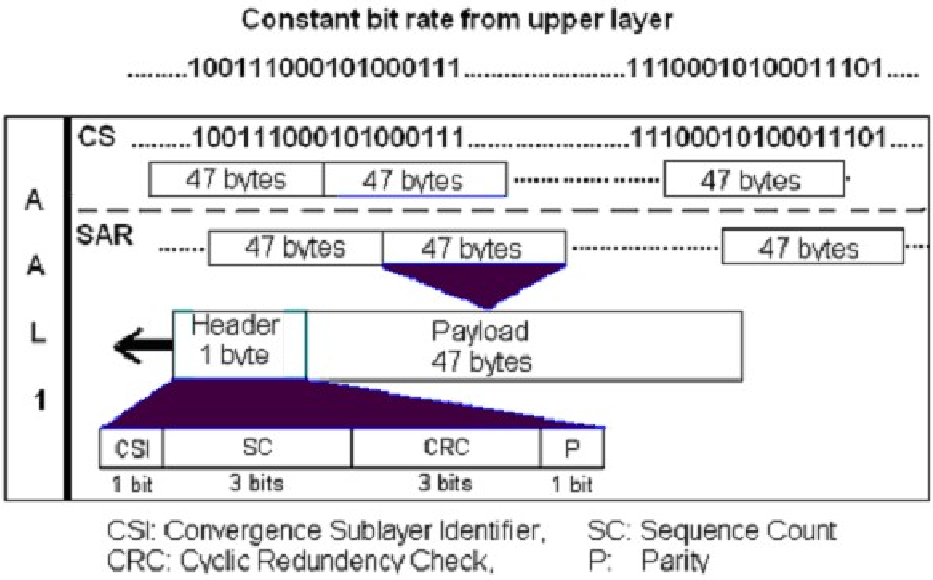
- There are several AAL categories: AAL1, AAL2, AAL3/4, AAL5.
- Each AAL category has two sublayers:

1. Convergence Sublayer (CS)
2. Segmentation and Reassembly Sublayer (SAR)

- Duties of CS and SAR vary for different AAL.



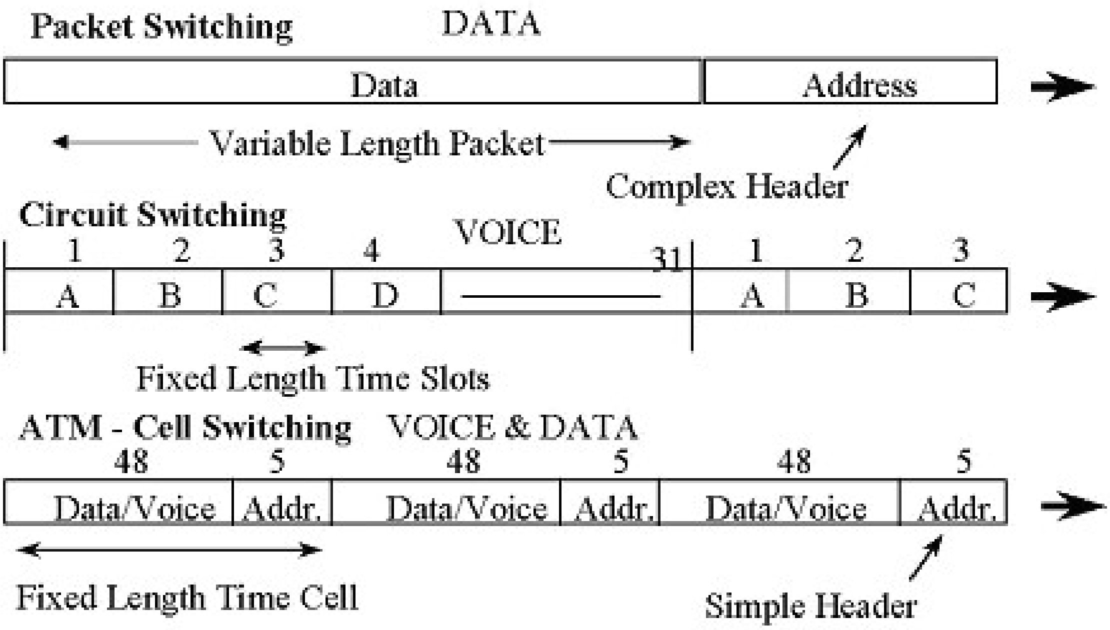
- **AAL1**: The structure of the AAL1 is given below:



- AAL1 supports CBR (constant bit rate) such as real time voice and real time video
- AAL1 allows ATM to connect existing digital telephone networks such as E-1.
- In AAL1, Convergence Sublayer (CS) divides the bit stream into 47-byte segments passes them to the SAR sublayer below.
- Segmentation and Reassembly (SAR) layer accepts a 47-byte from CS and adds a one byte header.
- The result is a 48-byte data unit that is passed to the ATM layer where it is encapsulated in an ATM cell of 53 bytes.

- **AAL2**:
  - Provides bandwidth-efficient transmission of low-rate, short and variable packets in delay sensitive applications
  - Supports VBR and CBR.
  - Also provides for variable payload within cells and across cells.
- **AAL3/4**:
  - Supports connection-oriented and connectionless data services.
- **AAL5 (Sometimes referred to as SEAL (simple and easy adaptation layer))**:
  - Simplified version of AAL3/4.
  - Provides point-to-point and point-to-multipoint (ATM layer) connections.

ATM Switching (Cell Switching) as compared with packet and circuit switching



Virtual Channels (VC) and Virtual Paths (VP) in ATM

- The VP and VC allow flexibility in the management of the resources in the network, by simplifying the routing and the resource allocation methods.
- It is possible for the network to lump many VCs together and then treat them as a single entity, rather than may be hundreds.
- Establishment of an end-to-end connection requires a series of links from source to destination.
- The series of virtual channel links is called a virtual channel connection, VCC.

- The virtual channel is identified in each cell by the virtual channel identifier, VCI, which is part of the cell header.

- Within a particular VC link, the VCI has a particular value, but will change with the aid of lookup tables in nodes from link to link within the VCC.

- A VP is a bundle of VC links

- All the VC links in the bundle have the same endpoints, so that a VC link is equivalent to a VP connection.

- A VP Identifier (VPI) identifies a group of VC links that share the same VP Connection (VPC).

- VP links are concatenated to form a VPC

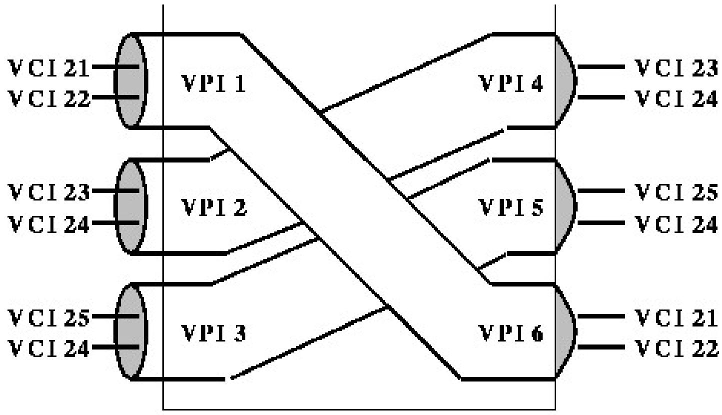
- A VPC endpoint is where the VCI changes, originates or terminates.

- When there is a VC switch, there first must be a termination of the VPCs that support the VC links that are going to be switched.

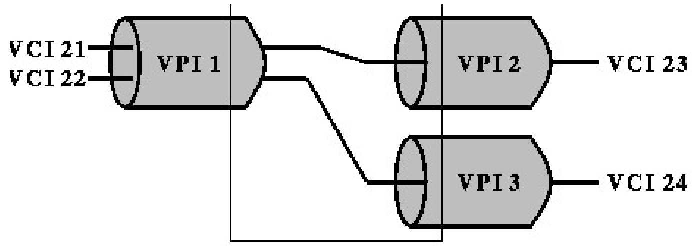
- Cell sequence is preserved in a VP and also in each VC link within a VPC.

- In a VP switch, the VC links that share a VPC must remain the same after the switch as before

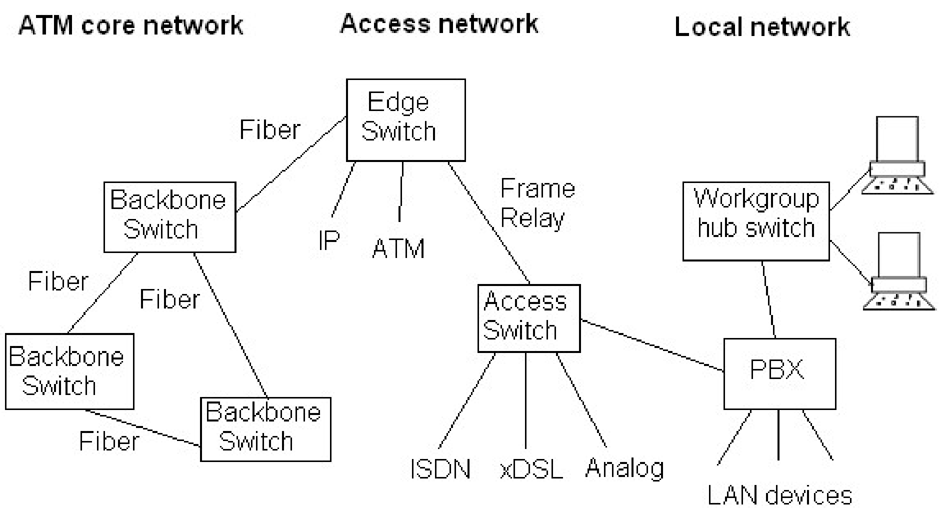
- This is seen in the below Figure for VP switch:



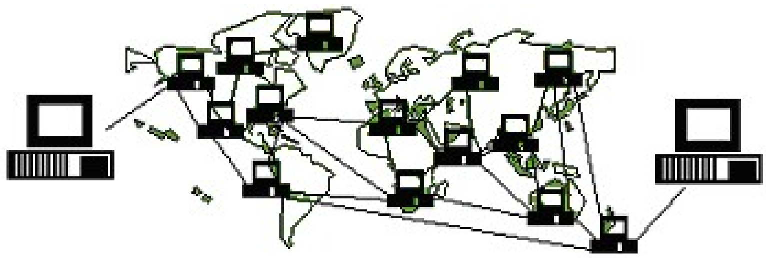
- In a VC switch all the VPs involved in the switching must be terminated and then originated again as seen in the below Figure for VC switch.



- Below Figure gives a general ATM infrastructure:



Internet Infrastructure



A piece of data (eg. a Web page) when it is transferred over the Internet:

- Is broken up into a whole lot of pieces (called packets).
- A header is added to each packet that explains where it came from, where it should end up and how it fits in with the rest of the packets.
- Each packet is sent from computer to computer until it finds its way to its destination.
- Each computer along the way decides where next to send the packet. This could depend on things like how busy the other computers are when the packet was received.
- The packets may not all take the same route.
- At the destination, the packets are examined. If there are any packets missing or damaged, a message is sent asking for those packets to be resent. This continues until all the packets have been received.
- The packets are reassembled into their original form.
- Each computer connected up to the Internet has software called TCP/IP (Transmission Control Protocol/Internet Protocol) which is responsible for receiving, sending and checking packets.

<u>IP Addressing</u>

- IP addressing scheme is integral to the process of routing IP datagrams through an internetwork.

- Each host on a TCP/IP network is assigned a unique 32-bit logical address that is divided into two main parts:

    - The network number which identifies a network and must be assigned by the Internet Network Information Center (InterNIC) if the network is to be part of the Internet. An Internet Service Provider (ISP) can obtain blocks of network addresses from the InterNIC and can itself assign address space as necessary.

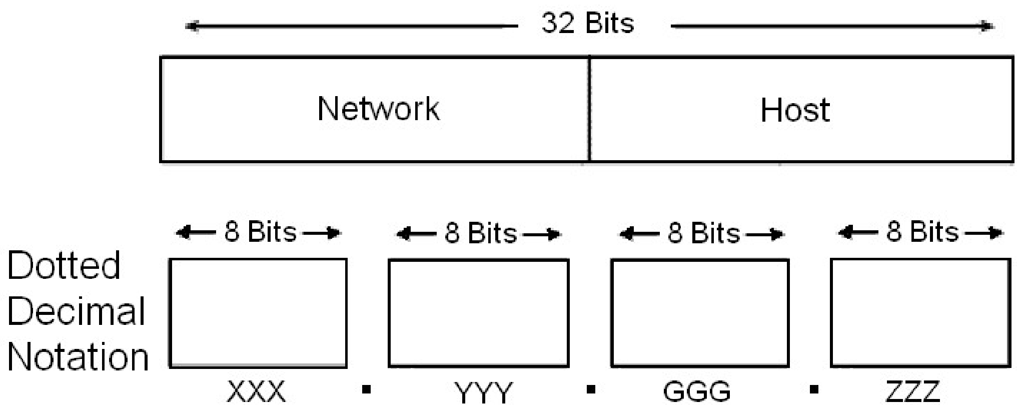
    - The host number which identifies a host on a network and is assigned by the local network administrator.

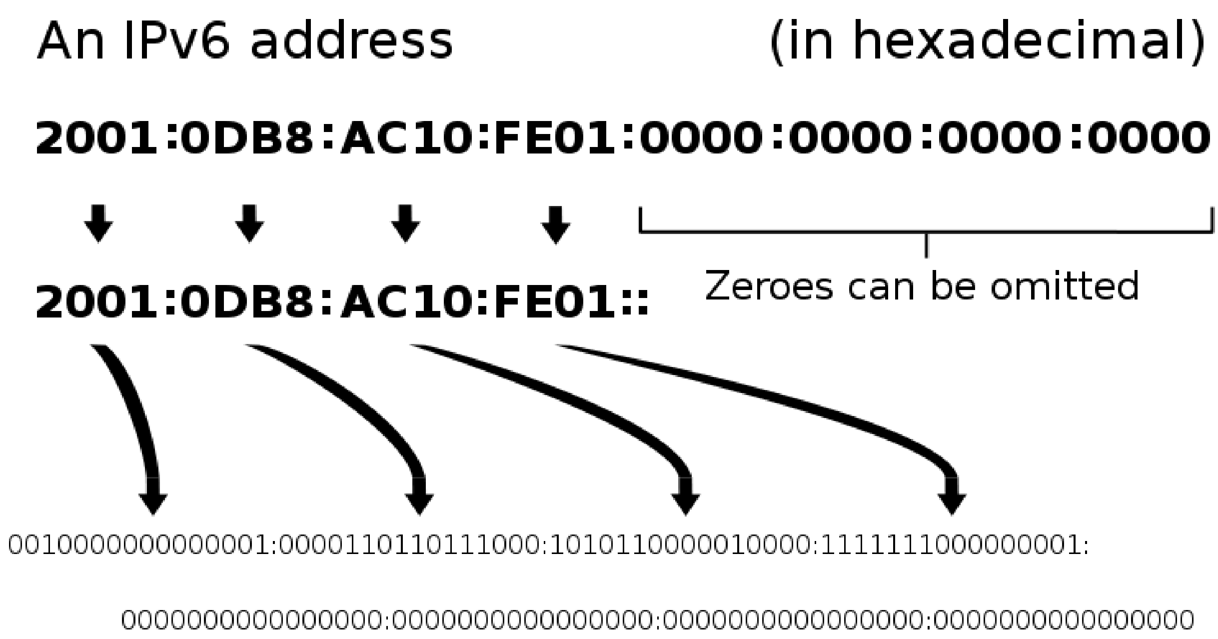
<u>IP Address Format</u>

- 32-bit IP address is grouped eight bits at a time, separated by dots, and represented in decimal format (known as *dotted decimal notation*).

- Each bit in the byte (octet) has a binary weight (128, 64, 32, 16, 8, 4, 2, 1). The minimum value for an octet is 0, and the maximum value for an octet is 255.

- Basic format of an IP address is shown below:





Protocols

- Protocol is "a formal description of message formats and the rules two or more machines must follow to exchange those messages."

- Protocols usually exist in two forms:

  - First, they exist in a textual form for humans to understand.

  - Second, they exist as programming code for computers to understand.

  - Both forms should ultimately specify the precise interpretation of every bit of every message exchanged across a network.
- Protocols exist at every point where logical program flow crosses between hosts. In other words, we need protocols every time we want

  - to do something on another computer
  - to print on a network printer
  - to download a file we need protocols
  - to save our work on disk, we *don't* need protocols - unless the disk is on a network file server.

- Usually multiple protocols will be in use simultaneously.

- Computers usually do several things at once, and often for several people at once. Therefore, most protocols support multitasking.

- One operation can involve several protocols. For example, consider the NFS (Network File System) protocol. A write to a file is done with an NFS operation, that uses another protocol (RPC) to perform a function call on a remote host, that uses another protocol (UDP) to deliver a datagram to a port on a remote host, that uses another protocol to delivery a datagram on an Ethernet, and so on. Along the way we may need to lookup host names (using the DNS protocol), convert data to a network standard form (using the XDR protocol), find a routing path to the host (using one or many of numerous protocols).

Protocol Layering

- Protocols are normally structured in layers, to simplify design and programming.

- *Protocol layering* is a common technique to simplify networking designs by dividing them into functional layers, and assigning protocols to perform each layer's task.

- For example, it is common to separate the functions of data delivery and connection management into separate layers, and therefore separate protocols.

- Thus, one protocol is designed to perform data delivery, and another protocol, layered above the first, performs connection management. The data delivery protocol is fairly simple and knows nothing of connection management. The connection management protocol is also fairly simple, since it doesn't need to concern itself with data delivery.

- Protocol layering produces simple protocols, each with a few well-defined tasks.

- These protocols can then be assembled into a useful whole.

- Individual protocols can also be removed or replaced as needed for particular applications.

- The most important layered protocol designs are the Internet's original DoD model, and the OSI Seven Layer Model. Internet represents a fusion of both models.

Encapsulation

- Layered protocol models rely on encapsulation, which allows one protocol to be used for relaying another's messages.

- Encapsulation, refers to the practice of enclosing data using one protocol within messages of another protocol.

- To make use of encapsulation, the encapsulating protocol must be open-ended, allowing for arbitrary data to placed in its messages. Another protocol can then be used to define the format of that data.

Encapsulation Example

- Consider an Internet host that requests a hypertext page over a dialup serial connection. The following scenario is likely:

  - First, the HyperText Transfer Protocol (HTTP) is used to construct a message requesting the page. The message, the exact format of which is unimportant at this time, is represented as follows:

HTTP

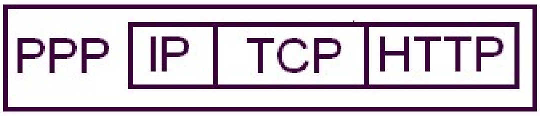
  - Next, the Transmission Control Protocol (TCP) is used to provide the connection management and reliable delivery that HTTP requires, but does not provide itself. TCP defines a message header format, which can be followed by arbitrary data. So, a TCP message is constructed by attaching a TCP header to the HTTP message, as follows:

TCP	HTTP

  - Now TCP does not provide any facilities for actually relaying a message from one machine to another in order to reach its destination. This feature is provided by the Internet Protocol (IP), which defines its own message header format. An IP message is constructed by attaching an IP header to the combined TCP/HTTP message:

IP	TCP	HTTP

  - Finally, although IP can direct messages between machines, it can not actually transmit the message from one machine to the next. This function is dependent on the actual communications hardware. In this example, we're using a dialup modem connection, so it's likely that the first step in transmitting the message will involve the Point-to-Point Protocol (PPP):



- Note that PPP encapsulation is drawn a little differently, by enclosing the entire message, not just attaching a header. This is because PPP may modify the message if it includes bytes that can't be transmitted across the link. The receiving PPP reverses these changes, and the message emerges. The point to remember is that the **encapsulating protocol can do anything it wants to the message - expand it, encrypt it, compress it - so long as the original message is extracted at the other end.**

Standards

- Protocols must be consistent to be effective. Therefore, *standards* are agreed upon and published.

- Standardized protocols provide a common meeting ground for software designers. Without standards, it is unlikely that an IBM computer could transfer files from a Macintosh, or print to a NetWare server, or login to a Sun. The technical literature of the Internet consists primarily of standard protocols that define how software and hardware from wildly divergent sources can interact on the net.

- IETF, the Internet Engineering Task Force, is one of the chief organizations in standards.

- ISO, the International Standards Organization, issues the OSI standards.

- IEEE, the Institute of Electrical and Electronic Engineers, issues key LAN standards such as Ethernet and Token-Ring.

- ANSI, the American National Standards Institute, issues FDDI.

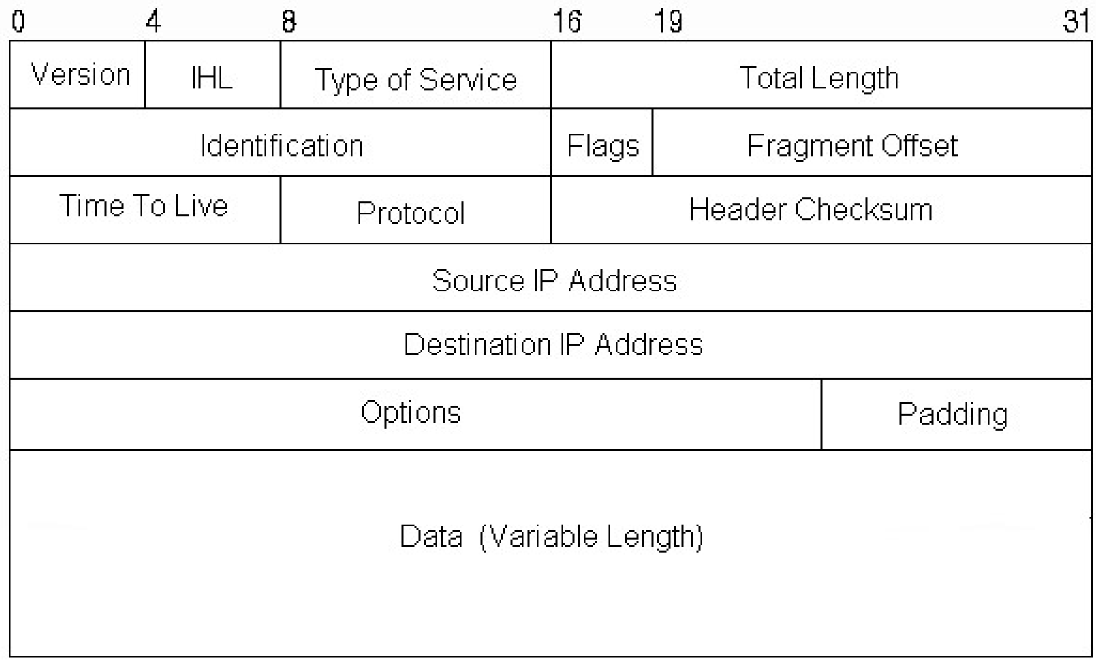
IP Packet Structure (Ipv4)

- All IP packets are structured the same way:

  - An IP header and

  - Followed by a variable-length data field

- A summary of the contents of the internet header is as follows:



Version (4 bits): Indicates the format of the internet header. This document describes version 4.

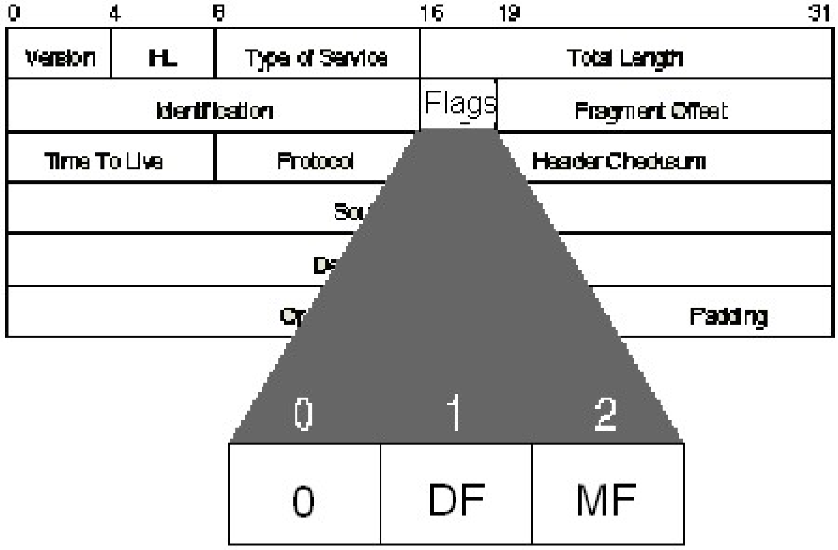
IHL (Internet Header Length) (4 bits): Is the length of the internet header in 32 bit words, and thus points to the beginning of the data.

Type of Service (8 bits): Provides an indication of the abstract parameters of the quality of service desired. These parameters are to be used to guide the selection of the actual service parameters when transmitting a datagram through a particular network. Several networks offer service precedence, which somehow treats high precedence traffic as more important than other traffic (generally by accepting only traffic above a certain precedence at time of high load). The major choice is a three way tradeoff between low-delay, high-reliability, and high-throughput.

Total Length (16 bits): Is the length of the datagram, measured in bytes (octets), including internet header and data. This field allows the length of a datagram to be up to 65,535 bytes. Such long datagrams are impractical for most hosts and networks. All hosts must be prepared to accept datagrams of up to 576 bytes (whether they arrive whole or in fragments). It is recommended that hosts only send datagrams larger than 576 octets if they have assurance that the destination is prepared to accept the larger datagrams. The number 576 is selected to allow a reasonable sized data block to be transmitted in addition to the required header information. For example, this size allows a data block of 512 bytes plus 64 header bytes to fit in a datagram. The maximal internet header is 60 bytes, and a typical internet header is 20 bytes, allowing a margin for headers of higher level protocols.

Identification (16 bits): An identifying value assigned by the sender to aid in assembling the fragments of a datagram.

Flags (3 bits): Various Control Flags.



Bit 0: reserved, must be zero

Bit 1: (DF) 0 = May Fragment, 1 = Don't Fragment.

Bit 2: (MF) 0 = Last Fragment, 1 = More Fragments.

Fragment Offset (13 bits): Indicates where in the datagram this fragment belongs. The fragment offset is measured in units of 8 bytes (64 bits). The first fragment has offset zero.

Time to Live (TTL) (8 bits): Indicates the maximum time the datagram is allowed to remain in the internet system. If this field contains the value zero, then the datagram must be destroyed. This field is modified in internet header processing. The time is measured in units of seconds, but since every module that processes a datagram must decrease the TTL (Time to Live) by at least one, even if it process the datagram in less than a second, the TTL must be thought of only as an upper bound on the time a datagram may exist. The intention is to cause undeliverable datagrams to be discarded, and to bound the maximum datagram lifetime.

Protocol (8 bits): Indicates the next level protocol used in the data portion of the internet datagram.

Header Checksum (16 bits): A checksum on the header only. Since some header fields change (e.g., time to live), this is recomputed and verified at each point that the internet header is processed.

Source Address (32 bits)

Destination Address (32 bits)

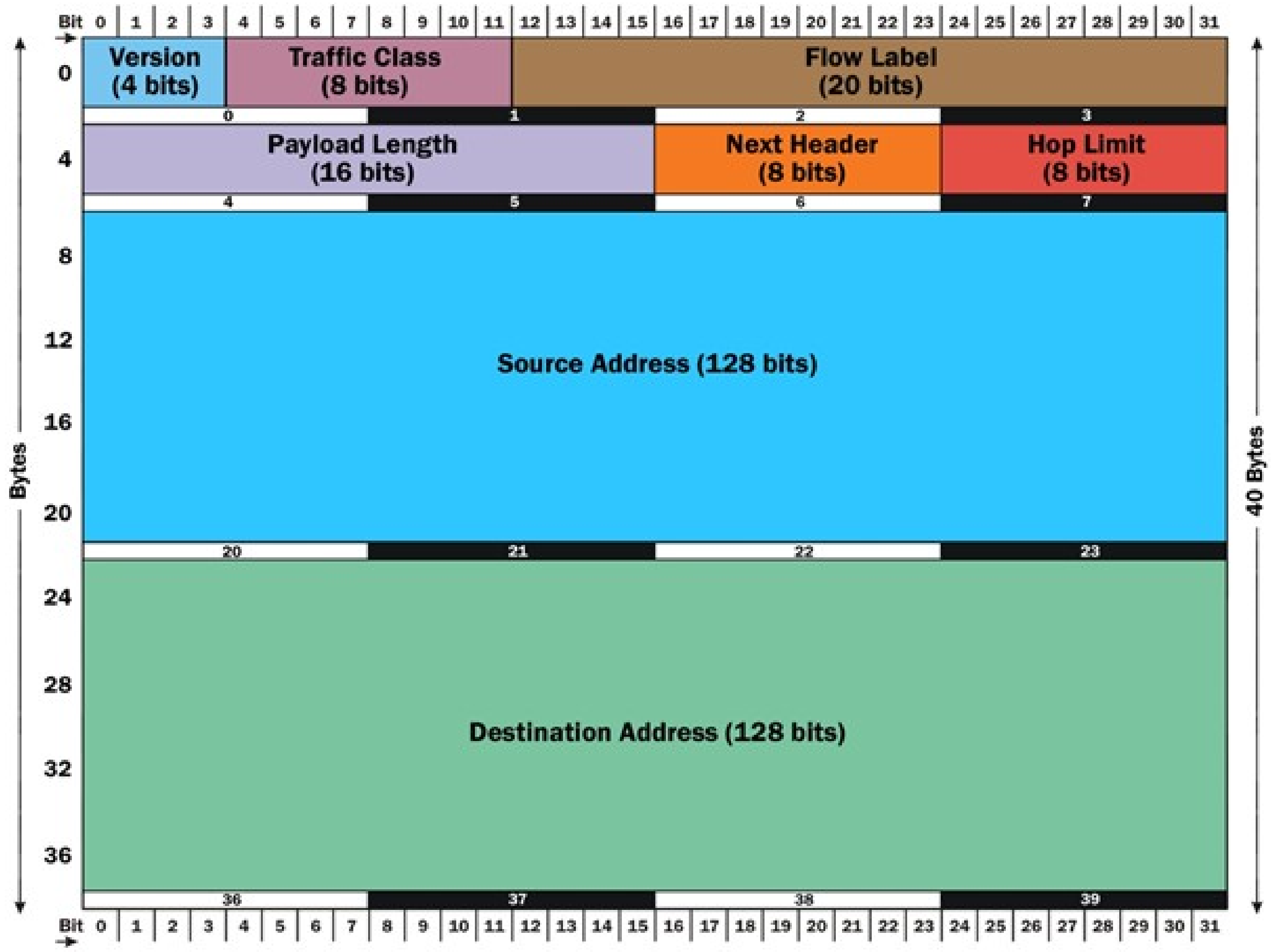
Options: Variable in length. May appear or not in datagrams. They must be implemented by all IP modules (host and gateways). What is optional is their transmission in any particular datagram, not their implementation. In some environments the security option may be required in all datagrams.

Padding: Variable in length. The internet header padding is used to ensure that the internet header ends on a 32 bit boundary. The padding is zero.

Data: Contains upper-layer information.

IPv6 PACKET HEADER

The IPv6 Packet Header is found at the start of every IPv6 Packet. It is always 40 bytes in length, Every bit of it is accounted for. It is twice the size of the (which results in 20 bytes of additional overhead in every IPv6 packet, compared to IPv4), yet has fewer fields. This is due to the far larger (4X) Source Address and Destination Address fields. It not only has fewer fields, it is actually much simpler. The complexity is moved off into Packet Header Extensions.



The **Version** field (4 bits) contains the value 6 in all IPv6 packets (imagine that!). In comparison, the Version field in all IPv4 packets contains the value 4. This field allows IPv4 and IPv6 traffic to be mixed in a single network.

The **Traffic Class** field (8 bits) is available for use by originating nodes and/or forwarding routers to identify and distinguish between different classes or priorities of IPv6 traffic, in a manner identical to that of IPv4 “Type of Service”.

The **Flow Label** field (20 bits) is something new in IPv6. It can be used to tag up to 2^{20} (1,048,576) distinct traffic flows, for purposes such as fine grained bandwidth management (QoS). Its use is still experimental. Hosts or routers that do not support this function should set it to zero when originating a packet, or ignore it when receiving a packet. A specific traffic flow is identified by a 3-tuple which includes the Source Address, Destination Address and a Flow Control number. As with Differentiated Service, the Flow Label field is just a request for prioritization - the actual prioritization is done in routers in the path. Unfortunately most current routers do not process the Flow Label field, so at this time, QoS in IPv6 is identical to that in IPv4. Once routers process the Flow Label information, IPv6 QoS will be significantly better than that in IPv4.

The **Payload Length** field (16 bits) is the length of the IPv6 packet payload (data field) in bytes, not counting the standard packet header (as it is in IPv4 Total Length). However, the Payload Length DOES include the size of any extension headers, which don’t even exist in IPv4. You can think of packet extension headers as being the first part of the data field (payload) of the IPv6 packet. Since the Payload Length field is 16 bits, the data field can be up to 65,535 bytes long. A new Hop-by-Hop extension header is defined in RFC 2675, "IP Jumbograms", August 1999. If this extension header is present, it overrides the Payload Length field with a 32 bit value. This allows the payload length to be up to 4 gigabytes.

The **Next Header** field (8 bits) indicates the type of header immediately following the basic IPv6 packet header. It uses some of the same values as the IPv4 Protocol field but there are some new values possible in IPv6 Packet Headers.

If the Next Header field in the basic packet header contains the code for TCP (6), UDP (17) or SCTP (132), then the transport layer header (TCP, UDP or SCTP) begins immediately after the basic packet header, followed by the data. If the Next Header field contains the value for ICMPv6 (58), then the ICMPv6 header begins immediately after the basic packet header, and may be followed with data. Otherwise one or more IPv6 extension headers will be found between the basic packet header and the transport or ICMPv6 header, which may be followed by data. Since each extension header has another Next Header field (and a Header Length field), this constitutes a linked list of headers before the transport or ICMPv6 header, which is followed by the data.

The **Hop Limit** field (8 bits) serves the same purpose as the Time To Live field in the IPv4 Packet Header. It is used to prevent packets from being circling around indefinitely on a network. Every time a packet crosses a switch or router, the hop count is decremented by one. If the hop count reaches zero, the packet is dropped, and the node that drops the packet sends an ICMPv6 “time exceeded” message to the packet sender. This mechanism is used to implement the traceroute command.

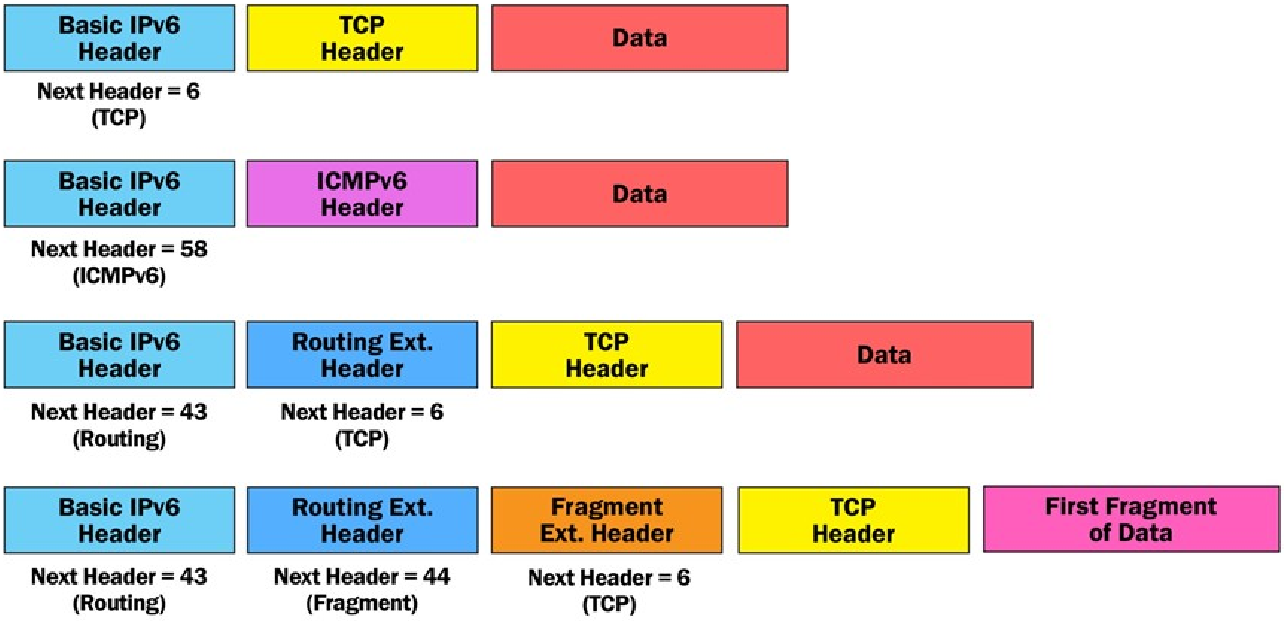
The **Source Address** field (128 bits) contains the IPv6 address of the packet sender. This can be any unicast IPv6 address (link local, global or ULA). It cannot be a multicast address. In some cases (if the node does not yet have any unicast address), the unspecified address (::) may be used.

The **Destination Address** field (128 bits) contains the IPv6 address of the packet recipient. This can be a unicast IPv6 address (link local, global or ULA). It can also be a multicast IPv6 address of any scope. It cannot be the unspecified address.

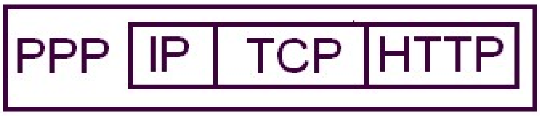
Summary of changes from the IPv4 Packet Header to IPv6 Packet Header:

- IPv4 IHL field was discarded (IPv6 basic header is fixed length of 40 bytes, no need for IHL)
- IPv4 Type of Service field was renamed the IPv6 Traffic Class field (same functionality)
- IPv4 Total Length field became the IPv6 Payload Length field (it no longer includes the length of the basic packet header)
- IPv4 Fragmentation fields moved to the IPv6 Fragment Extension Header
- IPv4 Time To Live field renamed as the IPv6 Hop Limit field (same functionality)
- IPv4 Protocol field renamed as the IPv6 Next Header field (same functionality, slightly different list of possible values)
- IPv4 Header Checksum discarded
- 32 bit IPv4 Source Address grew to 128 bit IPv6 Source Address, but otherwise same function
- 32 bit IPv4 Destination Address few to 128 bit IPv6 Destination Address, but otherwise same function
- New 20-bit Flow Label field added to the basic IPv6 packet header
- Several new Packet Extension headers already defined, more can easily be specified in the future

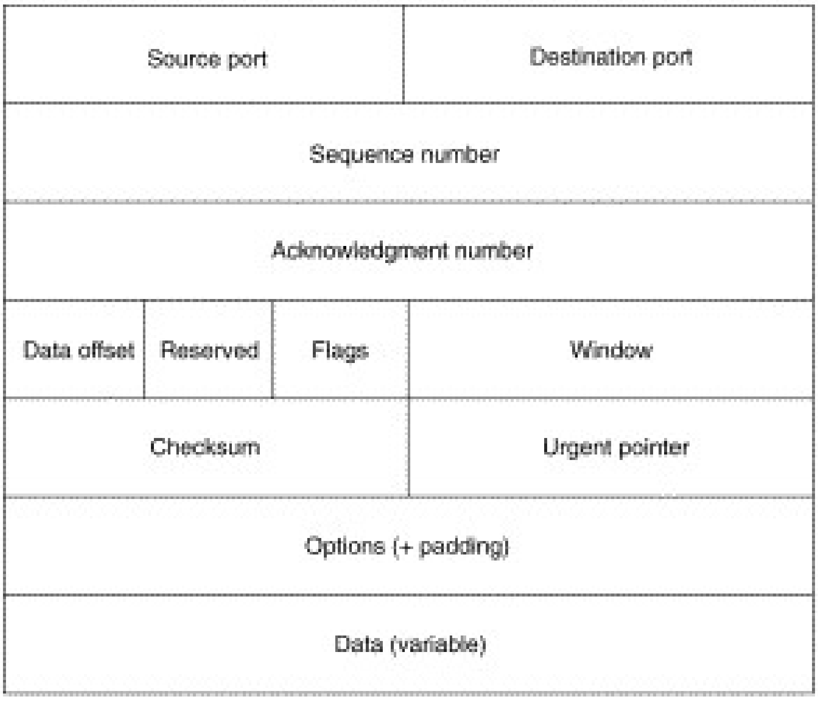
Various typical IPv6 packet header chains:



TCP (Transmission Control Protocol) Protocol



- TCP uses IP to implement data streams.
- TCP Packet Field Descriptions are as follows:



Session Management

- Network-layer protocols of the Internet are datagram-oriented and unreliable.

- It is the responsibility of the Transport and Session layer protocols to enhance the quality of service to that desired by a particular application. Known as the protocols of the *Host-to-Host Layer* or *Session Management.*

- These protocols function as an intermediary between the application and network layers.

- There are 3 major Internet session management protocols:

  - **UDP** (User Datagram Protocol) provides almost no additional functionality over IP. It performs fast, unreliable, datagram delivery. UDP Field Descriptions are as follows:

    ← 32 Bits →

Source Port	Destination Port
Length	Checksum

  - **TCP** (Transmission Control Protocol) provides reliable delivery for applications such as file transfers and remote logins. TCP takes steps to insure reliable data transfer, resending if needed due to network overloads or malfunctions.

  - **RPC** (Remote Procedure Call) is designed for programs to make subroutine calls on other systems.

VIRTUAL PRIVATE NETWORKS (VPN)

- VPN is a private network that uses a public network (usually the Internet) to connect remote sites or users together.

- Instead of using a dedicated, real-world connection such as leased line, a VPN uses "virtual" connections routed through the Internet from the Institution's private network to the remote site or employee.

- VPNs typically include a number of security features including encryption, authentication, and tunneling.

- Tunnel is the portion of the connection in which the data is encapsulated.

- A tunnel is created and the data is sent through the tunnel with encryption.

- If no encryption is involved this is not a VPN connection because the private data is sent across a shared or public network in an unencrypted and easily readable form.

- Tunneling protocols:

- Communication standards used to manage tunnels and encapsulate private data. (Data that is tunneled must also be encrypted to be a VPN connection.)

- PPTP (Point-to-Point Tunneling Protocol) - (PPTP) encapsulates Point-to-Point Protocol (PPP) frames into IP datagrams for transmission over an IP-based internetwork, such as the Internet or a private intranet.

- PPTP uses a TCP connection known as the PPTP control connection to create, maintain, and terminate the tunnel and a modified version of Generic Routing Encapsulation (GRE) to encapsulate PPP frames as tunneled data.

- Payloads of the encapsulated PPP frames can be encrypted or compressed or both.

- PPTP data tunneling is performed through multiple levels of encapsulation.

- Below Figure shows the resulting structure of PPTP tunneled data.

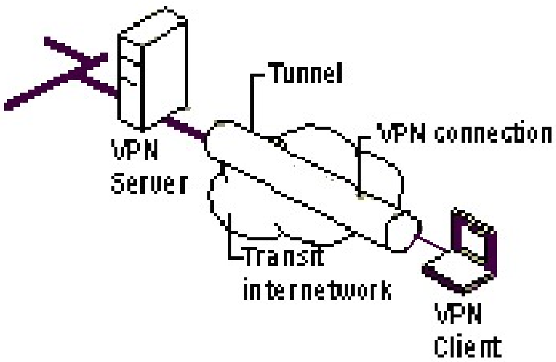
Data-link Header	IP Header	GRE Header	PPP Header	Encrypted PPP Payload (IP Datagram, IPX Datagram)	Data-link Trailer

- The initial PPP payload is encrypted and encapsulated with a PPP header to create a PPP frame.

- PPP frame is then encapsulated with a modified GRE header.

- GRE was designed to provide a simple, general purpose mechanism for encapsulating data sent over IP internetworks.

- GRE is a client protocol of IP using IP protocol 47.



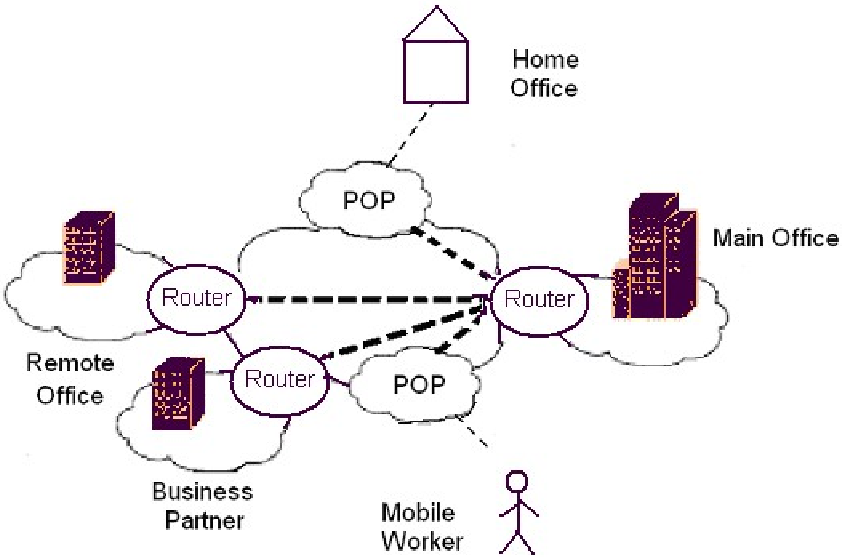
- Layer Two Tunneling Protocol (L2TP)

- L2TP encapsulates PPP frames to be sent over IP, X.25, Frame Relay, or ATM networks.

- When sent over an IP internetwork, L2TP frames are encapsulated as User Datagram Protocol (UDP) messages.

- L2TP can be used as a tunneling protocol over the Internet or over private intranets

- Windows 2000 includes the PPTP and L2TP tunneling protocols.



- There are two common VPN types:

1. **Remote-Access VPN** (or **Virtual Private Dial-up Network** (**VPDN**))

   - This is a user-to-LAN connection used by an institution that has employees who need to connect to the private network from various remote locations.

   - To set up a large remote-access VPN, the Institution outsources to an **enterprise service provider** (**ESP**).

   - ESP sets up a **network access server** (**NAS**) and provides the remote users with desktop client software for their computers.

   - Telecommuters can then dial a toll-free number to reach the NAS and use their VPN client software to access the corporate network.

   - Remote-access VPN permits secure, encrypted connections between the Institution's private network and remote users through a third-party service provider.

2. **Site-to-site VPN**

   - Multiple fixed sites are connected over a public network such as the Internet.

   - Site-to-site VPNs can be:

     - **Intranet-based** - If a company has one or more remote locations that they wish to join in a single private network, they can create an intranet VPN to connect LAN to LAN.

     - **Extranet-based** - When a company has a close relationship with another company (for example, a partner, supplier or customer), they can build an extranet VPN that connects LAN to LAN, and that allows all of the various companies to work in a shared environment.

Advantages of VPN

- Extends geographic connectivity
- Improves security
- Reduces operational costs versus traditional WAN
- Reduces transit time and transportation costs for remote users
- Provides global networking opportunities
- Provides broadband networking compatibility

Features needed in a VPN

- Security

  - Firewalls - Provides a strong barrier between the private network and the Internet. Firewalls can be set to restrict the number of open ports, what type of packets are passed through and which protocols are allowed through.

  - Encryption – Data sent by one computer is encoded in ciphered form so that only the computer which has the counter decryption key can resolve the data. It appears as a garbled data for the other computers.:

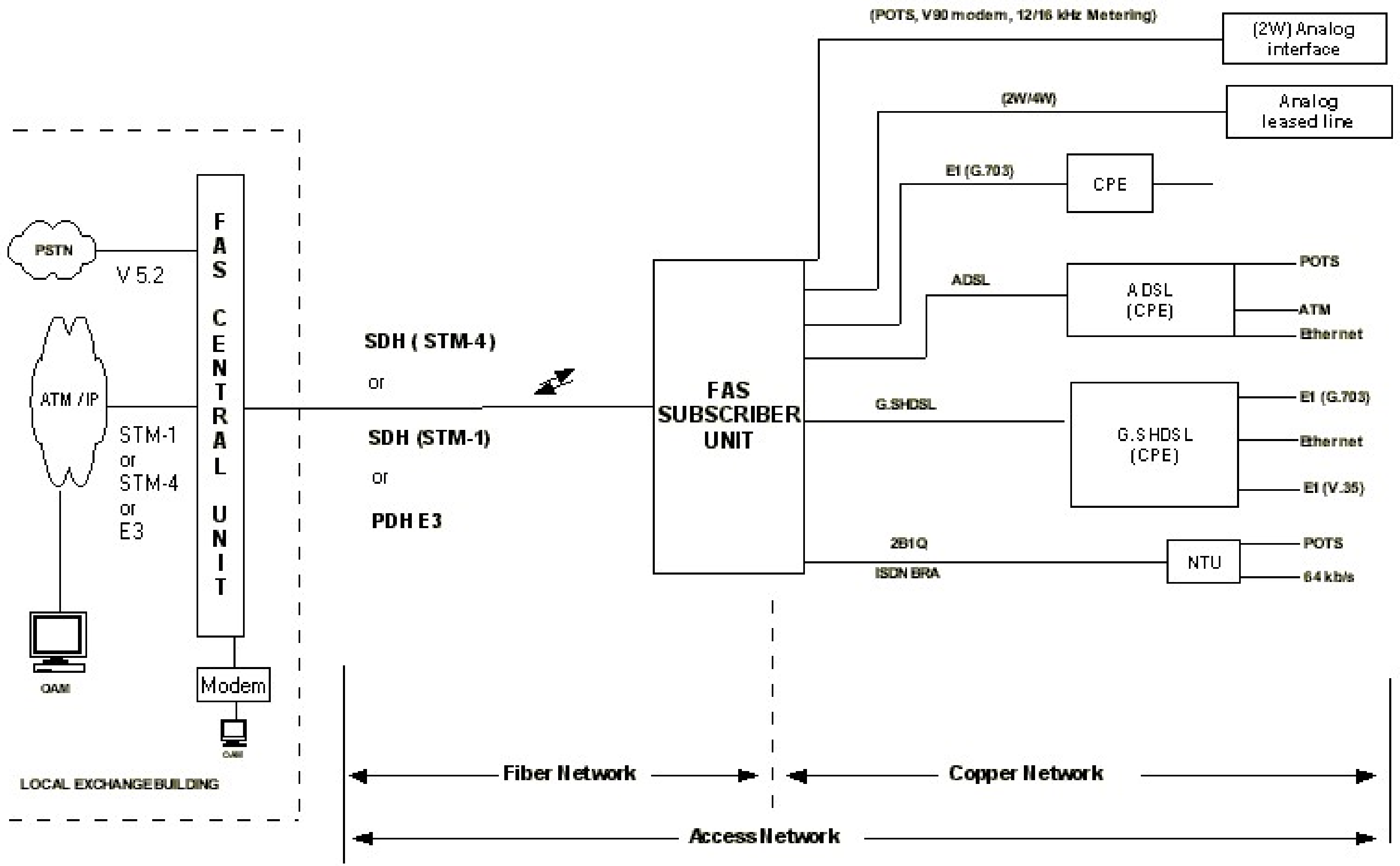
- Reliability

- Scalability - Unlike with leased lines, where the cost increases in proportion to the distances involved, the geographic locations of each office matter little in the creation of a VPN.

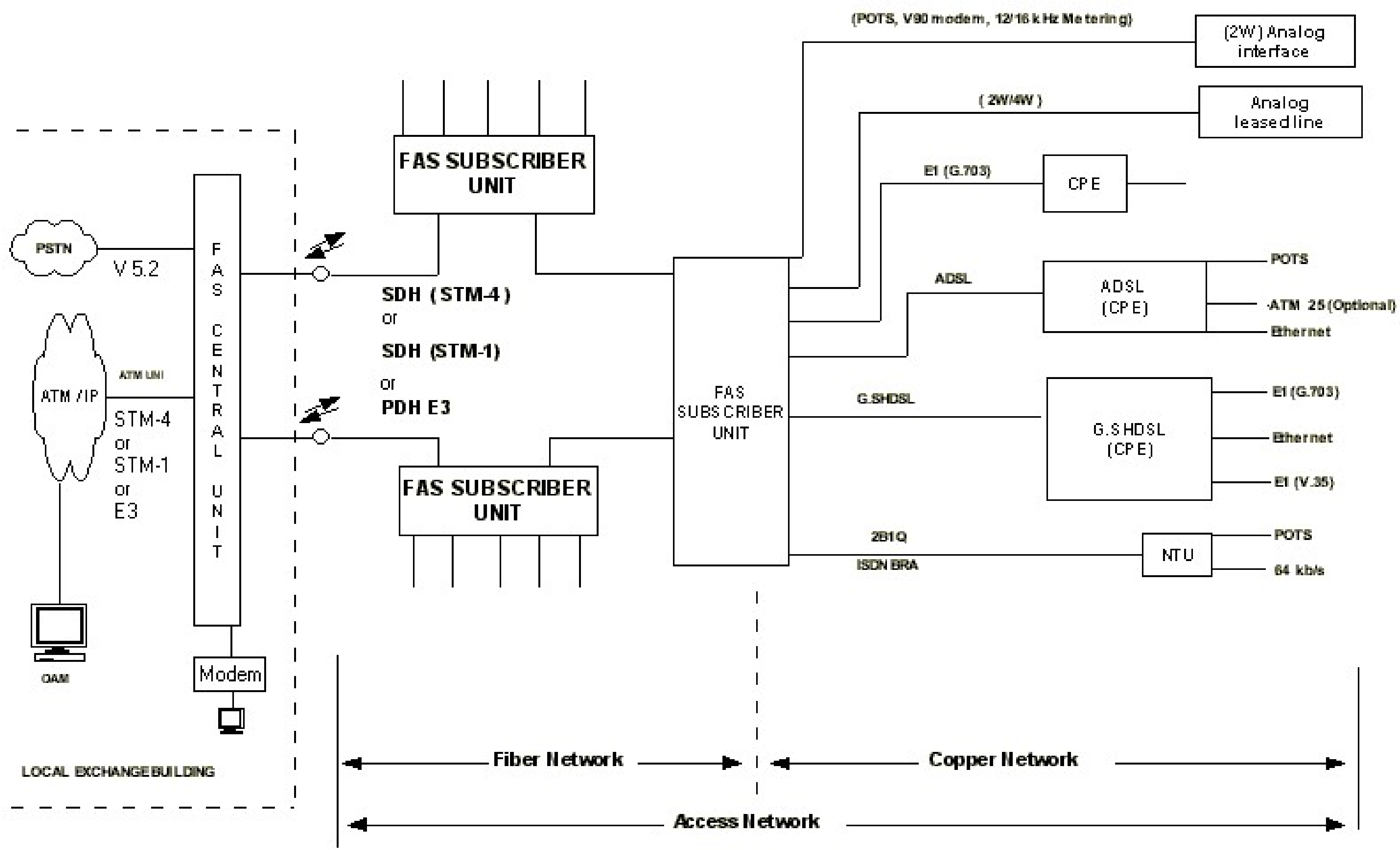
- Network management

Fiber Access Systems

BLOCK DIAGRAM OF FIBER ACCESS SYSTEM (FAS)
(Point - to - Point Access)



BLOCK DIAGRAM OF FIBER ACCESS SYSTEM (FAS)
(Ring Topology)



QoS

- QoS provides *service differentiation* and *performance assurance* for Internet applications, i.e., provides a specification of how good the offered network services are.

- Service differentiation is a way for ISPs to obtain higher revenue.

- Internet is (slowly) evolving to support QoS.

QoS Parameters

- Service requirements are specified using QoS parameters:

  - End-to-end delay,
  - jitter,
  - packet rate,
  - burst,
  - throughput,
  - packet loss.

- Examples of QoS parameters:

  - Audio service (Sample rate of 8000 samples/sec, sample resolution of 8 bits/sample).
  - Network service (Throughput of 100 Mbps, connection setup time of 50ms).

Possible Audio QoS Parameters

- Application QoS:

  - Sample Size 8-bit Telephone voice quality. Sample Rate 8 KHz Intermediate delay 125 μs
  - 16-bit CD audio. 44.1 KHz Intermediate delay 22.7 μs.
  - Playback point ~100 to 150 ms, depending on the network delay

- Network QoS:

  - End-to-end delay 0 to 150 ms Acceptable for most applications
  - 150 to 400 ms, may impact some apps.
  - > 400 ms, unacceptable
  - Round-trip delay up to 800 ms, acceptable for conversation
  - Packet loss ≤ 10^{-2} Telephone quality
  - Bandwidth 16 Kbps Telephone speech
32 Kbps Audio conference speech
64 Kbps Near CD-quality audio
128 Kbps CD-quality audio

Possible Video QoS Parameters:

- Application QoS:

  - Frame rate 30 fps (frames per second) NTSC format
  - 25 fps PAL format
  - 60 fps HDTC format
  - Frame width ≤ 720 pixels Video signal MPEG coded

- Frame height ≤ 576 pixels Vertical size
- Color resolution 8-bit or 16-bit/pixel Gray scale resolution of 256 or 65,536 colors
- Aspect ratio 4:3 or 16:9 NTSC, PAL, TV format or HDTV format
- Compression ratio 2:1 or 50:1 Lossy or lossless compression of HDTV

- Network QoS:

  - Bandwidth ≤ 1.86 Mbps MPEG encoded video
  - 64 Kbps to 2 Mbps H.261 encoded video
  - 1.544 Mbps to 2 Mbps H.120
  - 140 Mbps TV, PCM coding
  - > 1 Gbps HDTV uncompressed quality
  - ~ 500 Mbps HDTV lossless compression
  - 20 Mbps HDTV lossy compression
  - Bit error rate ≤ 10^{-6} Acceptable for conversation
  - Packet loss ≤ 10^{-2} Uncompressed video, ≤ 10^{-11} Compressed video
  - End-to-end delay ~ 250 ms Telephone speech

Components needed for QoS:

- Packet classification
- Isolation
- High resource utilization
- Admission control

Packet Classification

- Consider a phone application at 1 Mbps and an FTP application sharing a 1.5 Mbps link.

  - Bursts of FTP can congest the router and cause audio packets to be dropped.
  - Want to give priority to audio over FTP.

- Packet classification (marking) allows a router to distinguish among packets belonging to different classes of traffic.

Isolation

- Applications *misbehave* (audio sends packets at a rate higher than 1 Mbps assumed before). Need to provide protection (isolation) for one class from other classes.

- Need to regulate the rate at which a flow is allowed to inject packets into the network.

  - Policing Mechanism using leaky bucket.
  - Link-level packet scheduling.

Admission Control

- Cannot support traffic beyond link capacity.

- Need a Call Admission (admission control) process; application flow declares its needs, network may block call if it cannot satisfy the needs.

Scheduling Mechanisms

- Determines end-to-end delay => propagation delay + transmission delay + queuing delay

- How queued packets are selected for transmission is called Link Scheduling Discipline.

  - FIFO (First In First Out)
  - Priority Queuing
  - Round Robin
  - Weighted Fair Queing

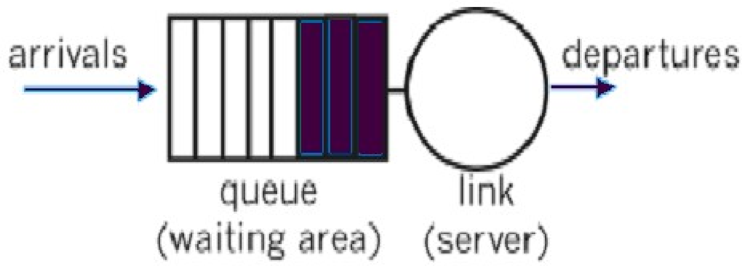
- Link scheduling discipline plays a crucial role in QoS:

FIFO (First In First Out)

- In-order of arrival to the queue; packets that arrive to a full buffer are either discarded (*tail drop*), or a discard policy is used to determine which packet to discard among the arrival and those already queued.

- Does not discriminate between different traffic sources (i.e., flows).

- Most widely used by today's Internet routers.



Priority Queuing

- Classes have different priorities; class may depend on explicit marking or other header info, e.g., IP source or destination, TCP port numbers, etc.

  - Transmit a packet from the highest priority class with a non-empty queue.

Queue Management

- Controls packet loss.

- Packets get lost due to *damage* and *congestion*:

  - Loss due to damage is rare (<< 1%).

- Currently packets are dropped when queue is full using tail drop, drop front, random...

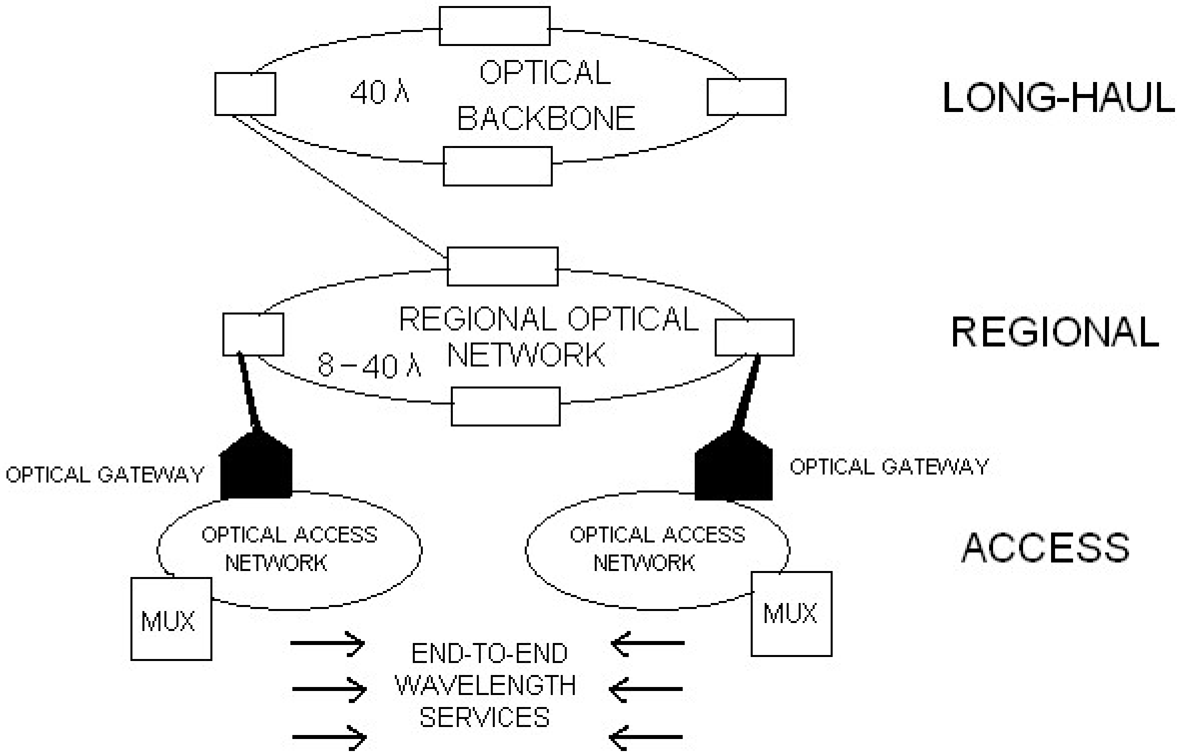
All Optical Networks

- High-capacity telecommunications networks.

- Based on all optical components.

- All the network is to be designed with all optical elements, thus bandwidth will not be a limiting factor since opto electronic conversions will not be needed throughout the network.

End-to-End Wavelength Services



- Optical networks began with wavelength division multiplexing (WDM), providing additional capacity on existing fibers.

- Like SDH/SONET, defined network elements and architectures provide the basis of the optical network.

- However, unlike SDH/SONET, rather than using a defined bit-rate and frame structure as its basic building block, the optical network will be based on wavelengths.

Optical-Network Drivers

- Fiber Capacity

    - First it was fiber limited. İ.e. more capacity between two sites meant the installation of more fibers.

    - Then more time division multiplexed (TDM) signals are placed in the same fiber, i.e. the bandwith handling capability of the fibers were increased. (both through fiber manufacturing and semiconductor laser modulation techniques supporting high rates of 40 Gbps)

    - Wavelength Division Multiplexing (WDM) is introduced providing many virtual fibers on a single physical fiber.

- Dense Wavelength Division Multiplexing (DWDM) further increased drastically the information rate carrying capability of fibers (in the order of hundreds of Terabits per second.

- Restoration Capability

  - As fiber capacity is increased, a fiber cut can have massive implications.

  - In current electrical architectures, each network element usually performs its own restoration.

  - For a WDM system with many channels on a single fiber, a fiber cut would initiate multiple failures, causing many independent systems to fail.

  - Optical networks can perform protection switching faster and more economically, because the back up in big rates.

  - Additionally, the optical layer can provide restoration in networks that currently do not have a protection scheme.

- Reduced Cost

  - In systems using only DWDM (i.e without optical Add-Drop Multiplexers, each location that demultiplexes signals will need an electrical network element for each channel, even if no traffic is dropping at that site.

  - By implementing an optical network, only those wavelengths that add or drop traffic at a site need corresponding electrical nodes. Other channels can simply pass through optically, which provides tremendous cost savings in equipment and network management.

  - In addition, performing space and wavelength routing of traffic avoids the high cost of electronic cross-connects.

- Wavelength Services

  - In optical networks, service providers are able to resell bandwidth rather than fiber.

  - By maximizing capacity available on a fiber, service providers can improve revenue by selling wavelengths, regardless of the data rate required.

  - To customers, this service provides the same bandwidth as a dedicated fiber.

<u>Optical Technologies</u>

- Broadband WDM

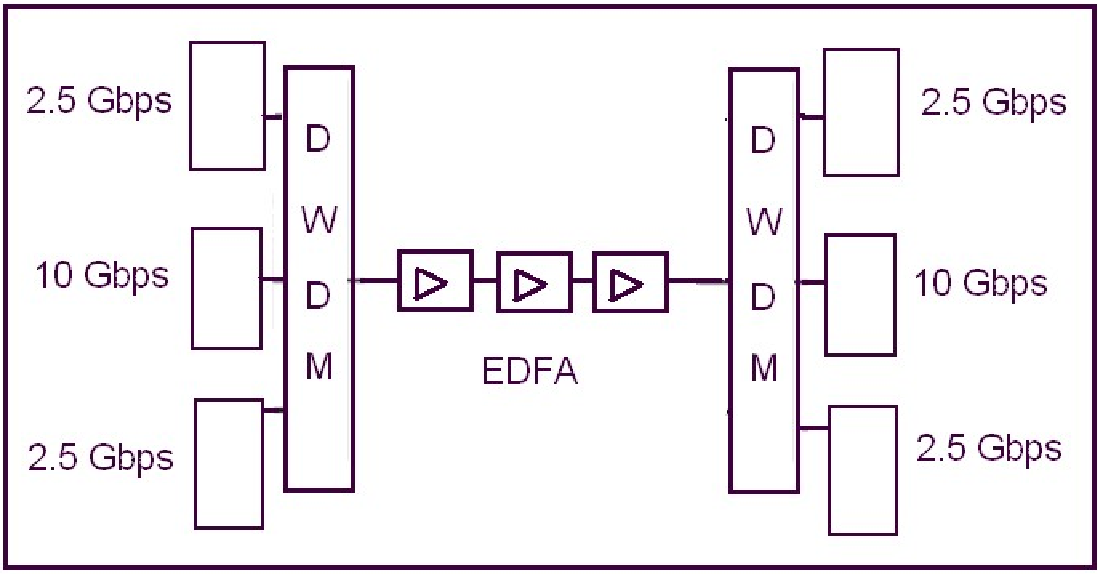
- Optical Amplifiers

  - Erbium-Doped Fiber Amplifier (EDFA). By doping a small strand of fiber with a rare earth metal, such as erbium, optical signals could be amplified without converting the signal back to an electrical state.

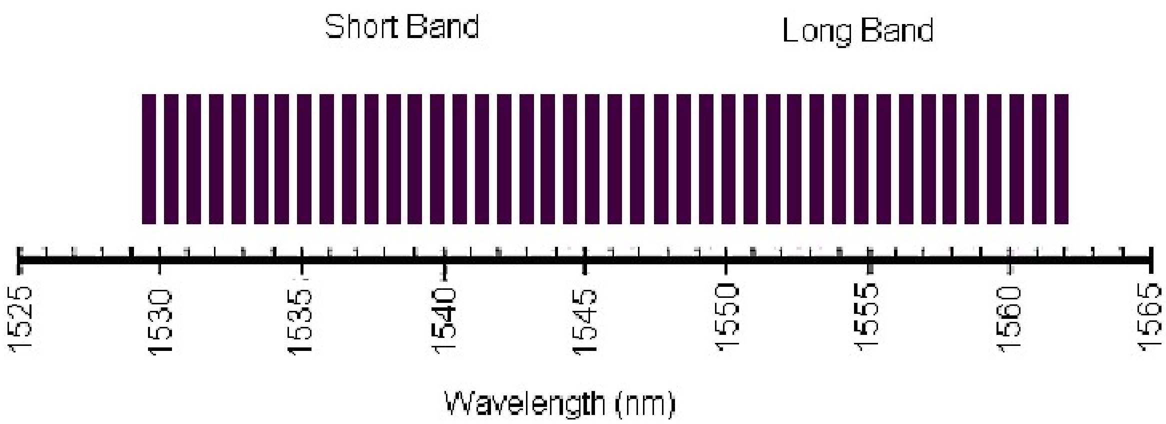
- EDFA operating at 1550 nm is used at each 50 – 100 km and replaces electronic regenerators.

- EDFA enables data rates of 10 Gbps or higher. With the electronic conversion the rate was limited by 2.5 Gbps.

• Dense Wavelength Division Multiplexer (DWDM).

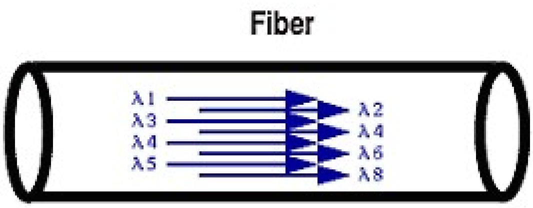


- ITU Channel Spacing is shown below:

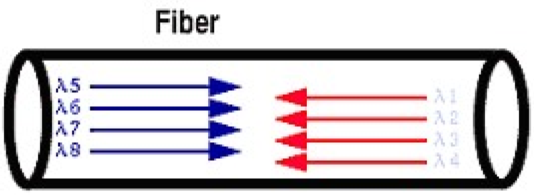


- Two basic types of DWDM:

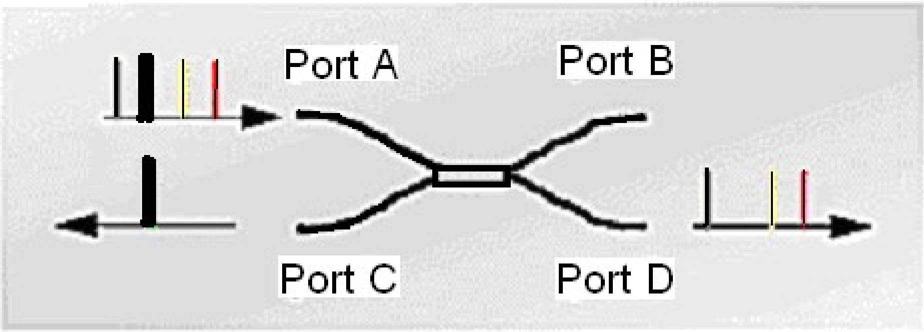
i. Unidirectional: All the wavelengths travel in the same direction on the fiber



ii. Bidirectional: Signals are split into separate bands, with both bands traveling in different directions.

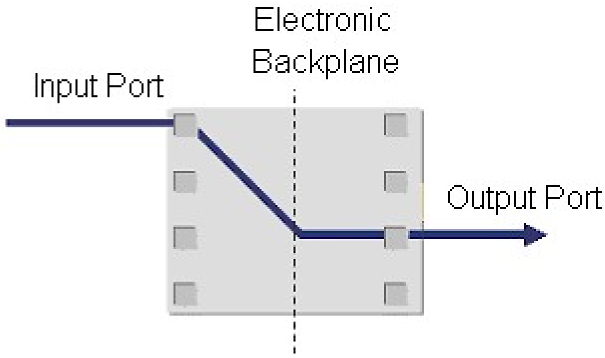


- Narrowband Lasers
  - Advanced lasers have extremely narrow source spectral bandwidths (<< 1 nm), very narrow wavelength spacings.
  - Long-haul applications use externally modulated lasers, while shorter applications can use integrated laser technologies.
- Fiber Bragg Gratings
  - It is a small section of fiber modified to create periodic changes in the index of refraction.
  - Depending on the space between the changes, a certain frequency of light - the Bragg resonance wavelength - is reflected back, while all other wavelengths pass through.



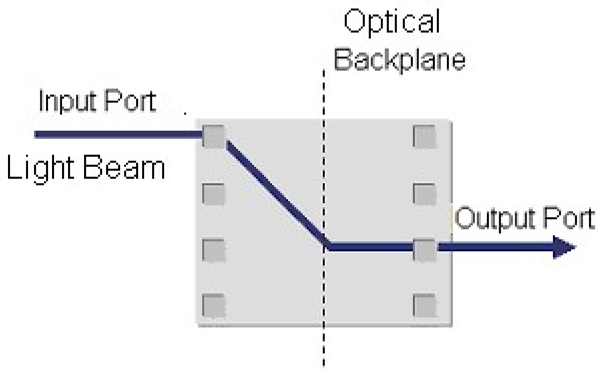
  - Fiber Bragg gratings are used in OADM (Optical Add/Drop Multiplexers) and in signal filtering.
- Thin Film Substrates
  - By coating a thin glass or polymer substrate with a thin interference film of dielectric material, the substrate can be made to pass through only a specific wavelength and reflect all others.
  - By integrating several of these components, optical network devices such as multiplexers, demultiplexers and add/drop devices are designed.
- Optical Switches (Sometimes referred to as Optical Cross Connects or Wavelength Routers)
  - Switch takes traffic in electrical form from an input port or connection and directs it again in electrical form over a backplane, to an output port.
  - Electronic switches direct variable-length packets, fixed-length cells, and synchronous timeslots from an input port to an output port.

- An electronic space switching is shown below:

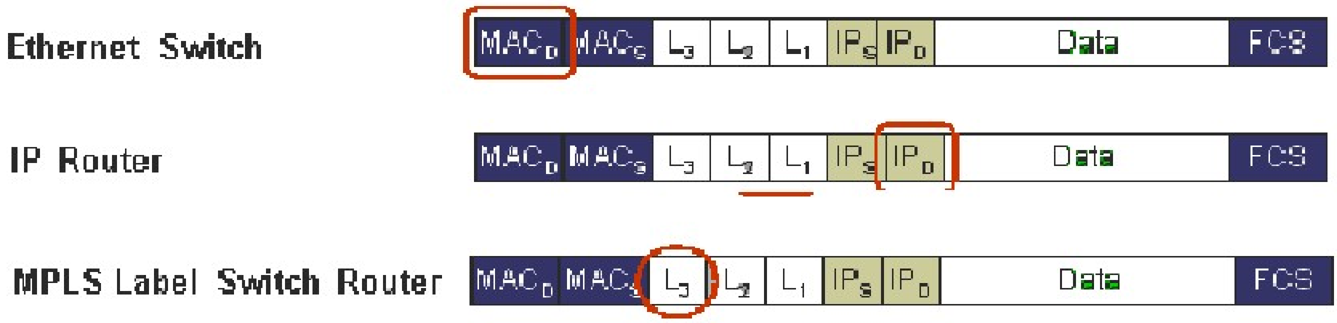


- An optical switch works with light. It directs a light beam of a single wavelength or of a range of wavelengths from an input port to an output port.

- An optical space switching is shown below:



- A switch needs some kind of information to make the switching decision. In electronic switches, this information is carried inside packets.



- An Ethernet, or MAC Layer switch, reads the destination MAC (media access control) (MAC_D) address on the frame and makes its forwarding decision based on this information.

- An IP switch, or router, uses the destination IP address (IP_D) to make its decision.

- In an MPLS (multiprotocol label switching) Label Switch Router, once Label Switched Paths have been established in the network, the outermost label is used to make a forwarding decision.

- The criterion of the optical switch for making a forwarding decision is carried in the so called digital wrapper around each input wavelength of the light.

- Wrapper is equivalent to packet header which carries information such as what type of traffic is in the wavelength, where the traffic is headed, ... etc.

- As the wavelength moves around the network, the nodes read the wrapper and get the information for originating and terminating details, whether it carries an IP or ATM or another protocol signal, commands such as error correction and whether the wavelength needs to be rerouted.

Types of Optical Switches:

- MEMS (Micro Electro Mechanical System) Switches:

  - Light in one fiber is just redirected to move to a different fiber by using microscopic (with diameters of a human hair) moveable (moveable in three dimensions) mirrors (several hundred mirrors placed together on mirror arrays in an area of a few centimeters square).

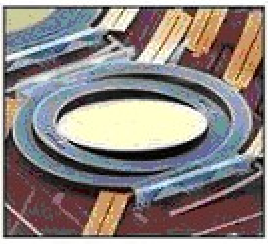
  - Light from an input fiber is aimed at a mirror, which is directed to move the light to another mirror on a facing array.

  - Light beams themselves tell the mirror (through digital wrappers) what bend to make in order to route the light appropriately.

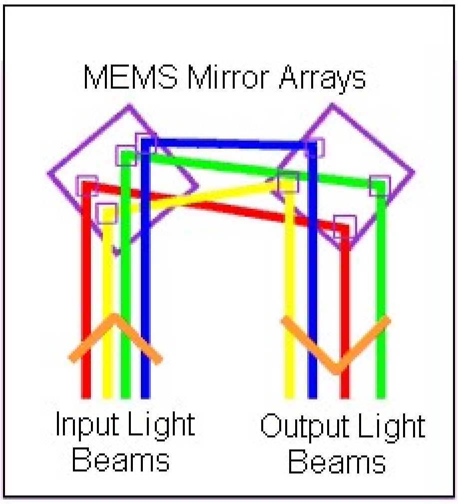
  - This mirror then reflects the light down towards the desired output optical fiber.

  - There exists designs of 1,024 x 1,024 wavelengths (if each can carry 40 Gbps it corresponds to a capacity of 40 Gbps x 1,024 = 40.96 Tbps) in an area of around 25 cm x 15 cm.

  - Picture of a MEMS mirror and MEMS mirror array deflection mechanism are shown below:



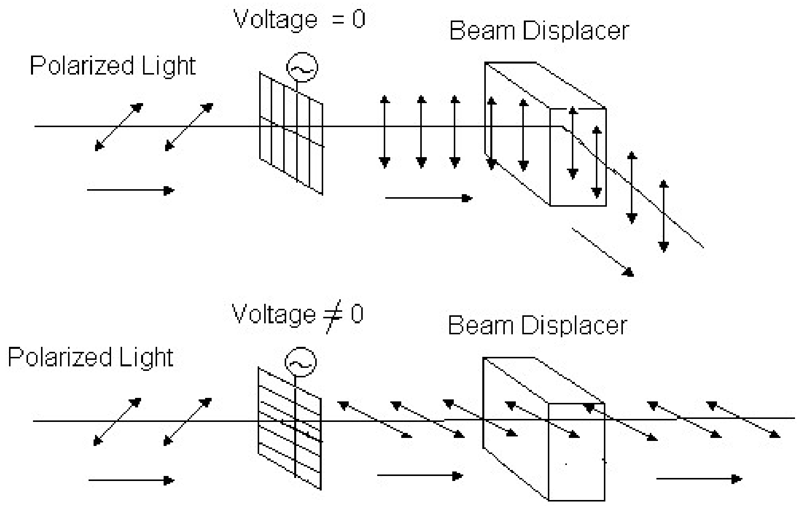
MEMS Mirror



- Buble Switches:

  - Use heat to create small bubles in fluid channels which then reflect and direct light

- Thermo-optical Switches:
  - Light passing through glass is heated up or cooled down by using electrical coils.
  - Heat alters the refractive index of the glass, bending the light to enter one fiber or another.
- Liquid Crystal (LCD) Switches:
  - Use liquid to bend light



Liquid Crystal Cell

- Tunable Lasers:
  - Radiate light at different wavelengths.
  - Can switch from one wavelength to another very quickly.
- Wavelength Switching:
  - Single wavelength enters the switch
  - A “wavelength” selection is made by using prisms, filters or gratings.
  - Based on the wavelength selected, the light is switched to a known output port.
- Optical Burst Switching:
  - Disadvantage of lambda switching is that, once a wavelength has been assigned, it is used exclusively by its “owner.”
  - If 100 percent of its capacity is not in use for 100 percent of the time, then there is an inefficiency in the network.
  - A solution to this is to allocate the wavelength for the duration of the data burst being sent giving rise to optical burst switching.

- However, the amount of time used in setting up and tearing down connections is very large compared to the amount of time the wavelength is “occupied” by the data burst.

- In ATM, X.25, ISDN, a multi-way handshaking process is used to ensure that the channel really is established before data is sent but these signalling techniques could not be applied to optical burst switching because they need too long time.

- For this reason, a simplex signalling mechanism is used in optical burst switching and there is no explicit confirmation that the connection is in place before the data burst is sent. (Unreliable signalling).

- <u>Optical Packet Switching (OPS)</u>:

    - OPS is the optical equivalent of an electronic packet switch, reading the embedded label and making a switching decision using this information.

    - OPS devices could operate in connectionless environment (e.g. using destination IP addresses) and also in connection-oriented mode by using GMPLS protocols to signal a path setup, and then embedded labels to allow switching to take place.

    - It is necessary to read the header bits at high speeds.

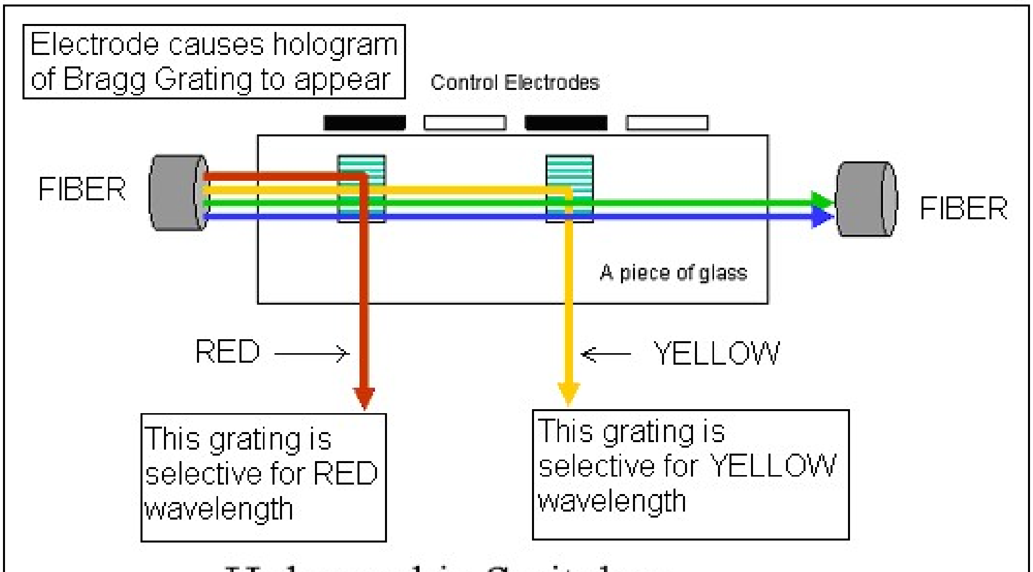
    - In the case of the Keys to Optical Packet Switching (KEOPS) project, KEOPS addressing headers are transmitted at a lower bit rate than the actual data payload.

- <u>Holographic Switching</u>:

    - Creates a wavelength-specific reflective grating, but does this dynamically.

    - The grating structure in these devices is written as a hologram into a piece of glass.

    - The holograms are “invisible” until they are energized by a set of control electrodes.



Holographic Switches

<u>Wireless</u>

- Points effecting the performance of wireless:

- Path Loss: The ratio of the transmitted power to the received power (measured in dB)

- Multipath: Artifact (noise) of reflections and echoes. Multipath can create secondory, tertiary, .. signals together with the primary signal

- Fading - As the mobile stations move within a cell, multipath signals can rapidly add constructively or destructively based on their instanteneous amplitude and phases, yielding a total signal varying a lot in magnitude.

  This is known as Rayleigh fading (in the absence of direct path, i.e line of sight)

  Multipath delays can be predicted on statistical basis and systems are designed accordingly.

- Interference and Noise - Byproducts of molecules and aerosols in the air or currents in the electronics used or some other anom alies.

  Error correction techniques are used to settle interference and noise.

- Antenna design, position and orientation.

• <u>Spectrum Reuse</u>

  - Wireless frequency spectrum is limited.

  - All the wireless users has to share this limited spectrum.

  - Efficient use of the spectrum is necessary.

  - For the efficient use of the spectrum:

    i. Apply space division: Split the service area into smaller coverage areas, cells in order to reuse frequencies across the cells.

    ii. Apply multiple access techniques to allow the sharing of the spectrum by multiple users.

    iii. After differentiating the space and combining the multiple conversations onto one channel, then apply spread spectrum, duplexing and compression to use the bandwidth even more efficiently.

• <u>Space Division</u>

  - Cell site is the basic building block of the cellular system

  - Several cell sites aligned in a strategic configuration are known as a cellular system

  - Each cell site is a low power transmitter/receiver that generates a specific calling coverage area

- This coverage area, known as a "cell", can be anywhere from one mile to twenty miles in diameter, depending on the location of the cell, type of terrain, and transmission power

- Traditional cellular network depends on cells organized as honeycomb configuration composed of seven cells as shown below:

- Frequencies can be reused as long as they are not in adjacent cells.

- E.g. if 700 channels are available in the honeycomb configuration, then each of the cells can use 100 of those channels.

- Those 100 channels can be reused in the next honeycomb configuration as long as those channels are not adjacent to one another between cells.

- Each cell has its own BTS (Base Transceiver Station)

- Around 30 BTS units are connected to a BSC (Base Station Controller).

- All the BSCs are connected to an OMC (Operation Maintenance Center.

<u>Type of Cells</u>

- Satellite Cells
- Macrocells
  - Around 15 km in diameter
  - Fast moving users
  - Base Station Power is relatively high, around 10 watts or more
- Microcells
  - Around 1 km in diameter
  - Slowly moving users
  - Lower power

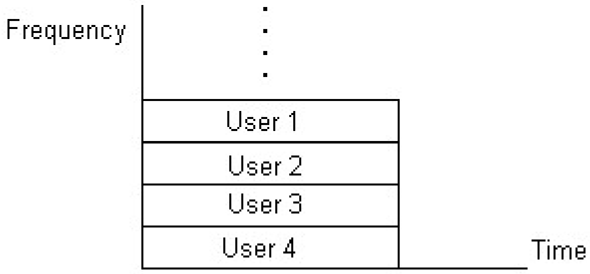
- Better frequency reuse

- Picocells
  - Around 100 meters in diameter
  - Stationary or very slowly moving users
  - Base Station Power is 10 mwatts or less.
  - Even better frequency reuse.

Multiple Access Techniques

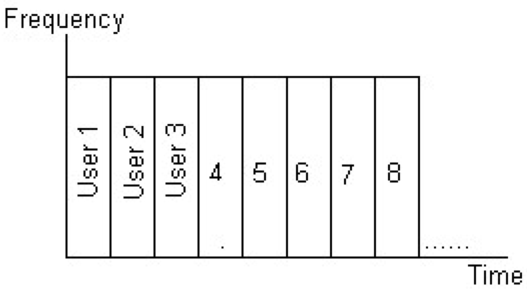
FDMA (Frequency Division Multiple Access)

- Divide assigned bandwidth into several channels or slots
- Each user gets one frequency slot assigned that is used at will
- FDMA could be compared to AM or FM broadcasting radio where each station has a frequency assigned
- FDMA demands good filtering.



TDMA (Time Division Multiple Access)

- Divide each channel into time slots; several calls per channel
- Frequency band is not partitioned but users are allowed to use it only in predefined intervals of time, one at a time
- Thus, TDMA demands synchronization among the users
- GSM (Global System for Mobile Communications) uses TDMA



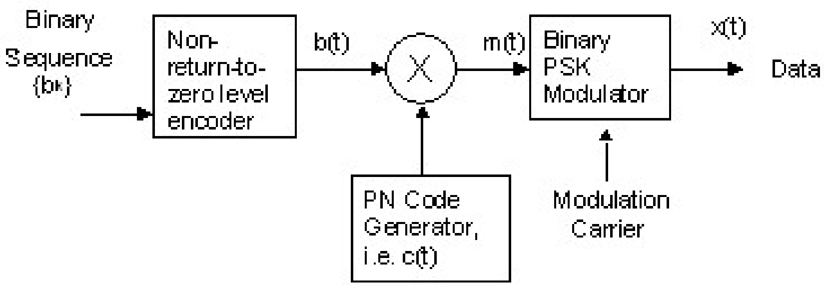
CDMA (Code Division Multiple Access)

- Data is spread over a range of bandwidth wider than actually needed by the information content
- Mixes the signal with Pseudorandom Code (PN) and spreads the signal over a broad frequency range
- Spread receivers recognize the signal, acquire and despread it to obtain original signal
- Increased capacity (10 x analog)
- Basically two kinds of CDMA:

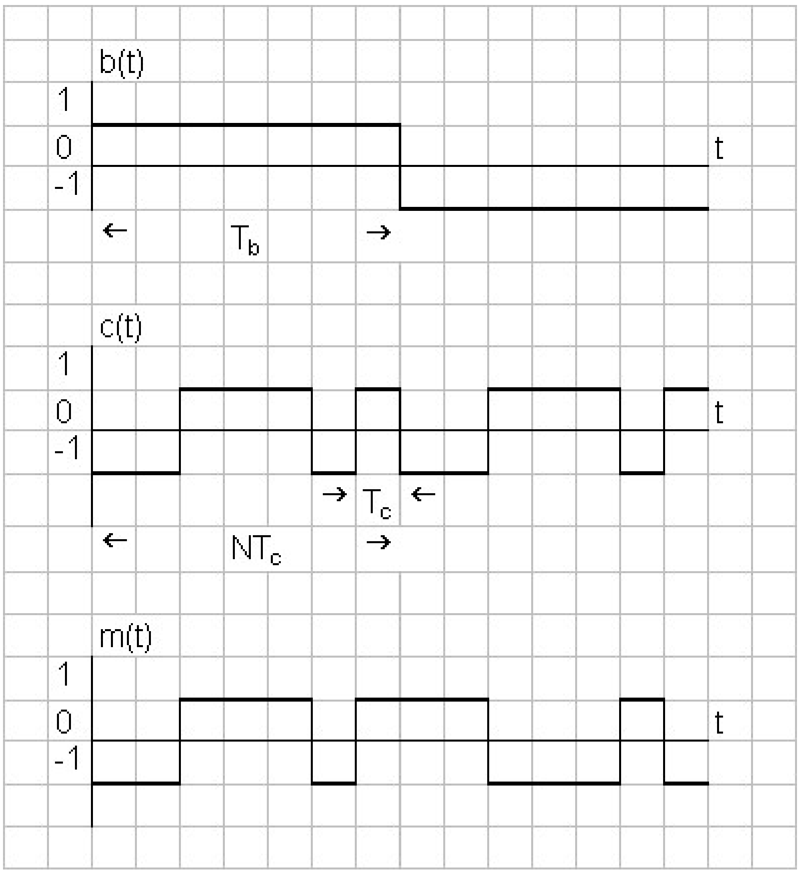
  1) DS-CDMA (Direct Sequence CDMA)

  2) FH-CDMA (Frequency Hopping Spread Spectrum)

- DS-SS (Direct Sequence Spread Spectrum) System Transmitter Block Diagram is shown below:



DS/BPSK Transmitter Block Diagram



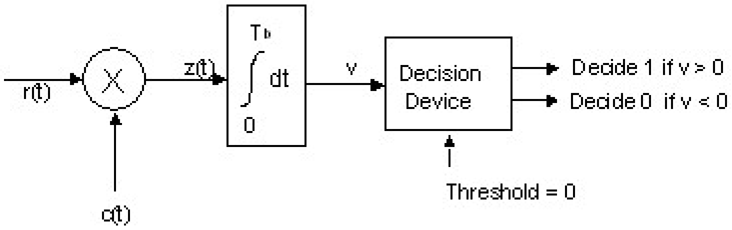
T_b is the period of one data bit

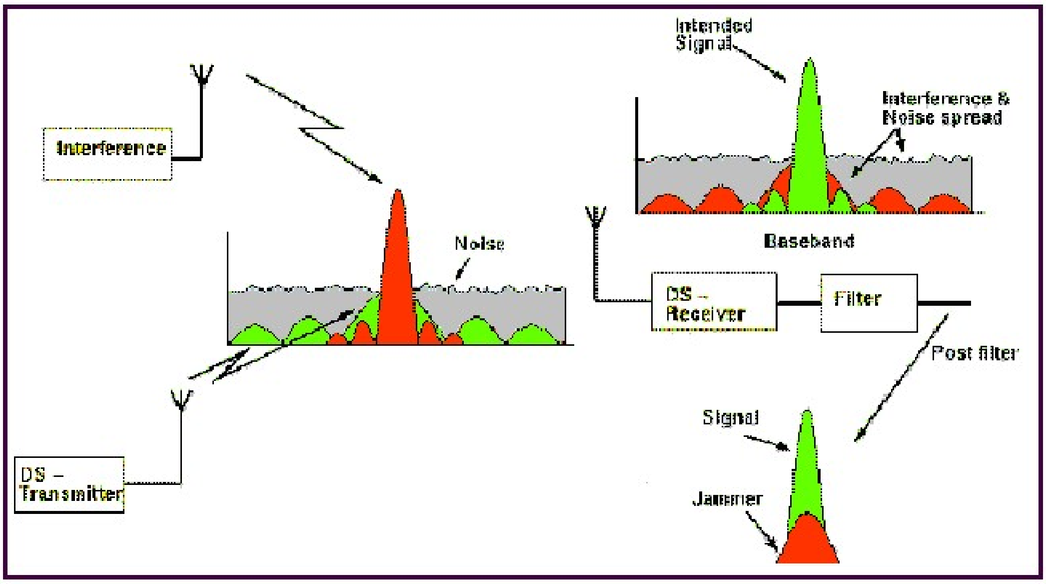
T_c is the period of one chip

Chip rate, $1/T_c$, is used to characterize a spread spectrum transmission system

Processing Gain or the **Spreading Factor** is defined as the ratio of the information bit duration over the chip duration. $\mathbf{G_p = SF = T_b / T_c}$

- DS-SS (Direct Sequence Spread Spectrum) System Receiver Block Diagram is shown below:





<u>Basic points in Spread Spectrum:</u>

- Use of spreading code (with pseudorandom property) at the transmitter produces a wideband transmitted signal

- Wideband transmitted signal appears as noise to a receiver which has no knowledge about the spreading code

- For a given information data rate, the longer the period of the spreading code, the closer is the transmitted signal to be a true random binary waveform

- Thus the longer the period of the spreading code, harder it is to detect

- The information signal is only recovered when the spreading signals at the transmitter and the receiver match and they are synchronized

- Price paid for spread spectrum are increased transmission bandwidth, system complexity and processing delay

3G Technology Evolution