Terminology, Classification, and Performance metrics

Definitions- Bandwidth, Data Rate, and Channel Capacity

Bandwidth

- Bandwidth of the transmitted signal as constrained by the transmitter and transmission medium, expressed in Hertz

Data Rate

- The rate in bits per second (bps) at which data can be communicated

Channel Capacity

- The maximum rate at which data can be transmitted over a communication path or channel under certain conditions such as SNR (signal-to-noise ratio)

In the context of "Computer Networks" the terms bandwidth, data rate, and capacity are sometimes used interchangeably (bps).

Notation: 1 Kbps bandwidth = 10^3 bits per second

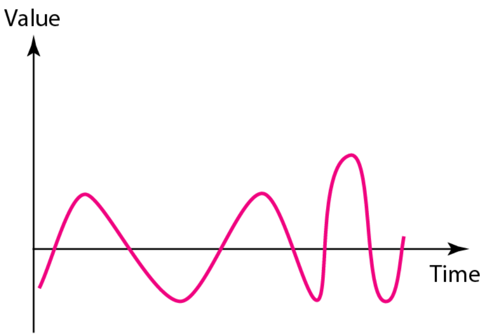
Networks, Applications, and **Information types**

- Networks are driven by the applications
- Therefore, networks must be designed to support the requirements imposed by the information types associated with the applications
- Information: text, data, speech, audio, image, video

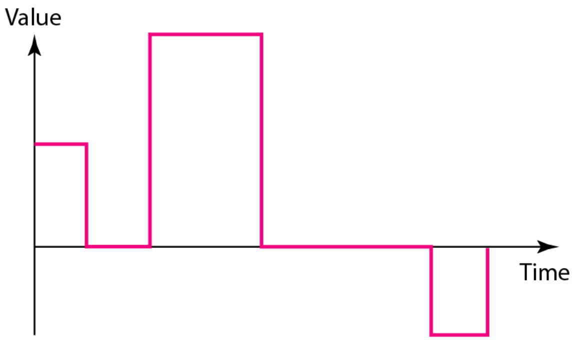
Information

- Information in its original form can be analog or digital

- Analog data: continuous and take continuous values.
  - E.g., voice, music



- Digital data have discrete states and take discrete values.
  - E.g., a data file in a computer



Digital representation of information

- We try to represent all information in digital form (binary representation)
  - Analog data: converted into digital data
- Information type: block and stream

Block-oriented Information

- Information that occurs in a single block
  - Text message
  - Data file
  - JPEG image

- Size = Bits / block
  or bytes/block
  - 1 Kbyte = 2^{10} bytes
  - 1 Mbyte = 2^{20} bytes
  - 1 Gbyte = 2^{30} bytes

Stream-oriented Information

- Information that is produced/transmitted *continuously* (and play them while receiving the rest)

- *Typically **multimedia information** such as audio (music, voice) and video*
  1) Streaming 'stored' audio and video (e.g., pre-recorded videos such as movies, YouTube,..)

  2) Streaming 'live' audio and video (e.g., live radio/tv programs such as cricket match, news)

  3) Conversational voice- and video- over IP (Real-time interactive audio/video) (e.g., Skype, Google Talk)

- Bit rate = bits / second
  - 1 kbps = 10^3 bps
  - 1 Mbps = 10^6 bps
  - 1 Gbps = 10^9 bps
  - 1 Tbps = 10^{12} bps

Block oriented information

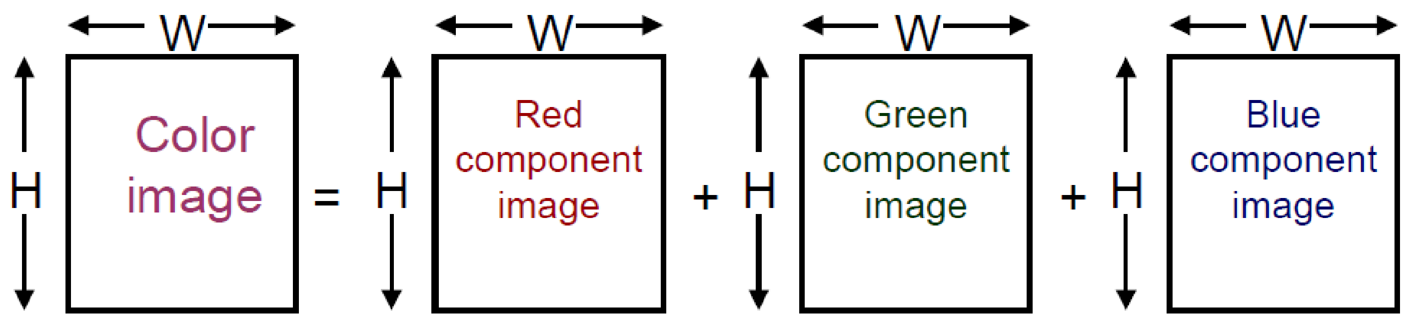
Information type	Data compression technique	Format	Uncompressed	Compressed (compression ratio)	Applications
Text files	Compress, zip, and variations	ASCII	Kbytes to Mbytes	(2–6)	Disk storage, modem transmission
Scanned black-and-white documents	CCITT Group 3 facsimile standard	A4 page @ 200 × 200 pixels/inch and options	256 Kbytes	15–54 Kbytes (1-D) 5–35 Kbytes (2-D) (5–50)	Facsimile transmission, document storage
Color images	JPEG	8-×-10 inch photo scanned @ 400 pixels/inch	38.4 Mbytes	1.2–8 Mbytes (5–30)	Image storage or transmission

Data compression algorithms

- Represent the information using fewer bits
- Lossless: original information recovered exactly
  E.g. zip, compress, GIF
- Lossy: recover information approximately
  - JPEG
  - Tradeoff: # bits vs. quality
- Compression Ratio: #bits (original file) / #bits (compressed file)

Color Image

(Pixel: a single dot in a digitized image)



Total bits = 3 × H × W pixels × B bits/pixel = 3HWB bits

Example: 8×10 inch picture at 400 × 400 pixels per inch^2

400 × 400 × 8 × 10 = 12.8 million pixels

8 bits/pixel/color

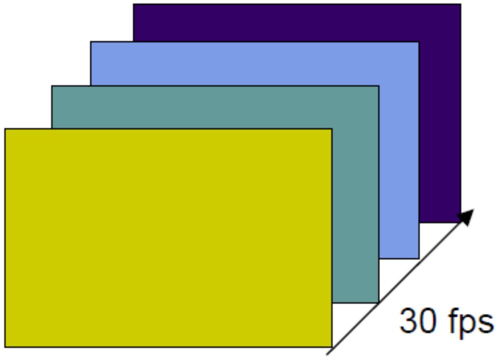
12.8 megapixels × 3 bytes/pixel = 38.4 megabytes

Stream oriented information

Information type	Compression technique	Format	Uncompressed	Compressed	Applications
Voice	PCM	4 kHz voice	64 kbps	64 kbps	Digital telephony
Voice	ADPCM (+ silence detection)	4 kHz voice	64 kbps	32 kbps (16 kbps)	Digital telephony, voice mail
Voice	Residual-excited linear prediction	4 kHz voice	64 kbps	8–16 kbps	Digital cellular telephony
Audio	MPEG audio MP3 compression	16–24 kHz audio	512–748 kbps	32–384 kbps	MPEG audio
Video	H.261 coding	176 × 144 or 352 × 288 frames @ 10–30 frames/second	2–36.5 Mbps	64 kbps–1.544 Mbps	Video conferencing
Video	MPEG-2	720 × 480 frames @ 30 frames/second	249 Mbps	2–6 Mbps	Full-motion broadcast video
Video	MPEG-2	1920 × 1080 frames @ 30 frames/second	1.6 Gbps	19–38 Mbps	High-definition television
	MPEG-4				
	HEVC/H.265				

Video Signal

- Sequence of picture frames
  - Each picture digitized & compressed
- Frame repetition rate
  - 10-30-60 frames/second depending on quality
- Frame resolution
  - Small frames for videoconferencing
  - Standard frames for conventional broadcast TV
  - HDTV frames



Rate = M bits/pixel **x** (WxH) pixels/frame **x** F frames/second

Transmission of Stream Information

- Constant bit-rate (CBR)
  - Signals such as digitized telephone voice produce a steady stream: e.g. 64 kbps
  - Network must support steady transfer of signal, e.g. 64 kbps circuit
  - E.g., MPEG1 standard compressed video (1.15 Mbps, 3 Mbps), Audio (8 Kbps, 1.3 Mbps for CD quality)

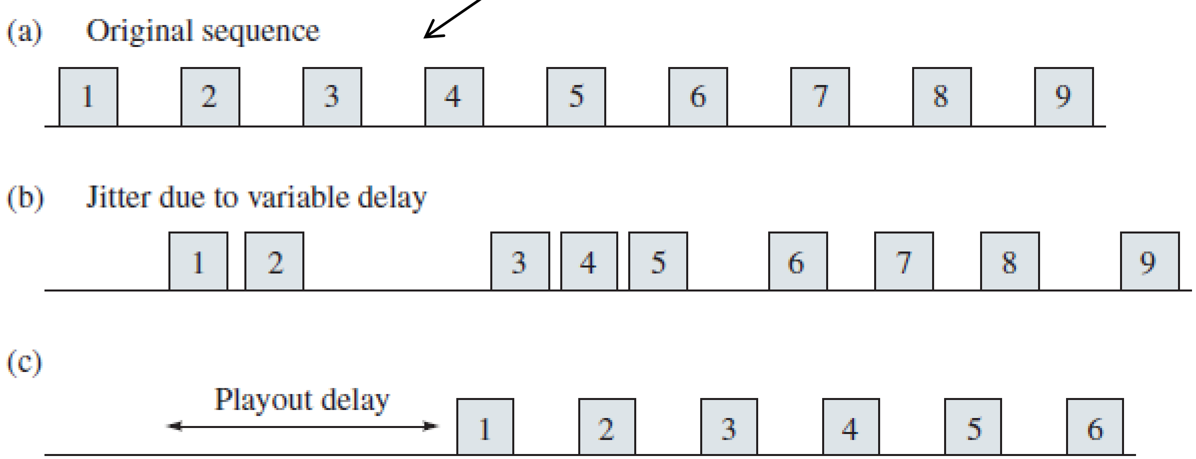
- Variable bit-rate (VBR)
  - Signals such as digitized video produce a stream that varies in bit rate, e.g. according to motion and detail in a scene
    - E.g., Audio (lossy compression): MP3, Opus, WMA, Vorbis, and AAC audio files: can be encoded in VBR (optional)
    - Audio (lossless compression): FLAC, Apple Lossless Audio Codec (ALAC)
    - Video (lossy compression): MPEG-2 video, MPEG-4 Part 2 video (Xvid, DivX, etc.), MPEG-4 Part 10/H.264 video or AVC, Theora, Dirac and other video compression formats
  - Network must support variable transfer rate of signal, e.g. packet switching

Network requirements of different information types

- What are the requirements when transmitting digital information
  - Network must be capable of transferring the required *volume of information* (block or stream)
  - *Accuracy* (lost some bits or errors?) and *timeliness* (takes too long to get?)
    - Loss and transmission errors: Is information delivered without loss and errors? If loss or errors, delivered signal quality acceptable or not?
      - Data files: cannot tolerate any errors/loss
      - Digitized voice: can tolerate some errors/loss
    - Delay: Is information delivered in timely fashion?
    - Jitter (delay variation): Is information delivered in sufficiently smooth fashion?

E.g., a streaming application (e.g., video streaming):

A stream of information can be viewed as a sequence of blocks of information

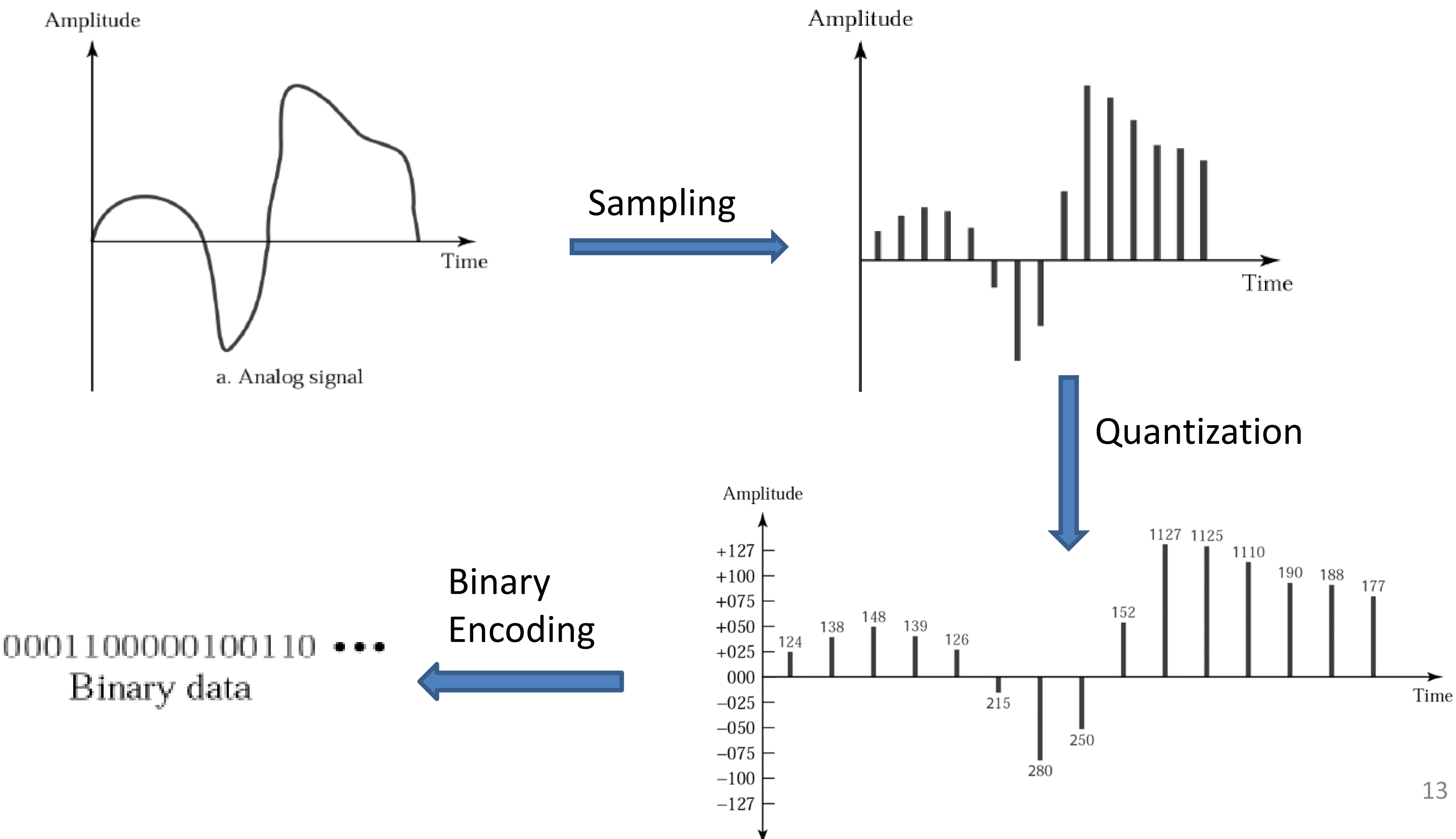


When playing back the information at the receiver, a steady stream of information blocks must be fed

E.g., Requirements for voice-over-IP?

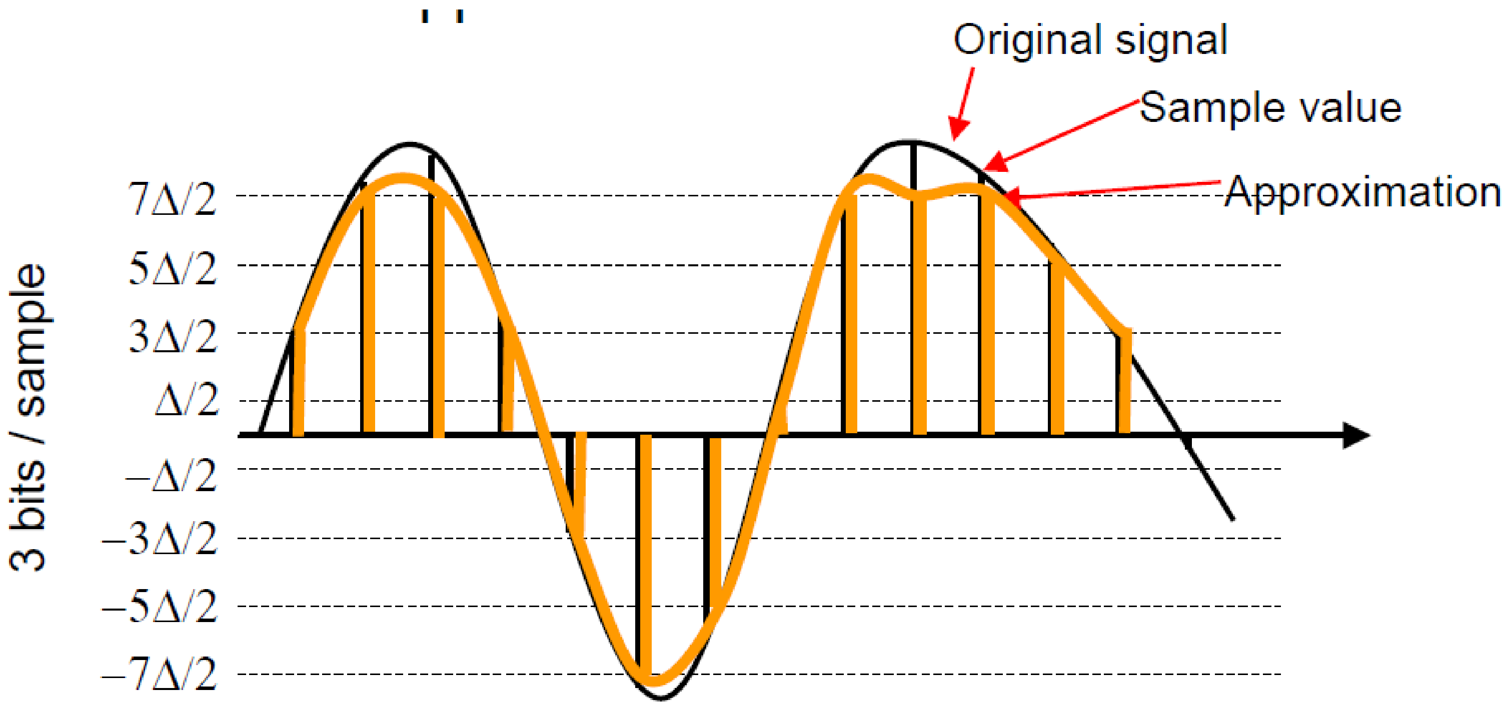
Digitization of Analog Signal

- (We try to represent all information in digital form) How analog information is converted into digital



Digitization of Analog Signal…

- Example: We are going to sample every T Sec and we are going to have 8 levels for each sample



R_s = Bit rate = # bits/sample x # samples/second

Digitization of Analog Signal…

- Representation accuracy
  - Higher accuracy

  → More samples: smaller spacing between approximation values

  → more bits per sample

- Bandwidth W_s Hertz of the information: how fast the signal changes
  - Higher bandwidth → Need to take more frequent samples
  - Minimum sampling rate = 2 x W_s (samples/sec)

- Example: Voice & Audio

Telephone voice (PCM)

- W_s = 4 kHz → 8000 samples/sec
- Need to sample every T=1/8000=125 micro sec
- 8 bits/sample
- R_s=8 x 8000 = 64 kbps

- Cellular phones use more powerful compression algorithms: 8-12 kbps

CD Audio

- W_s = 22 kHertz → 44000 samples/sec
- 16 bits/sample
- R_s=16 x 44000= 704 kbps per audio channel
- MP3 uses more powerful compression algorithms

Network Services

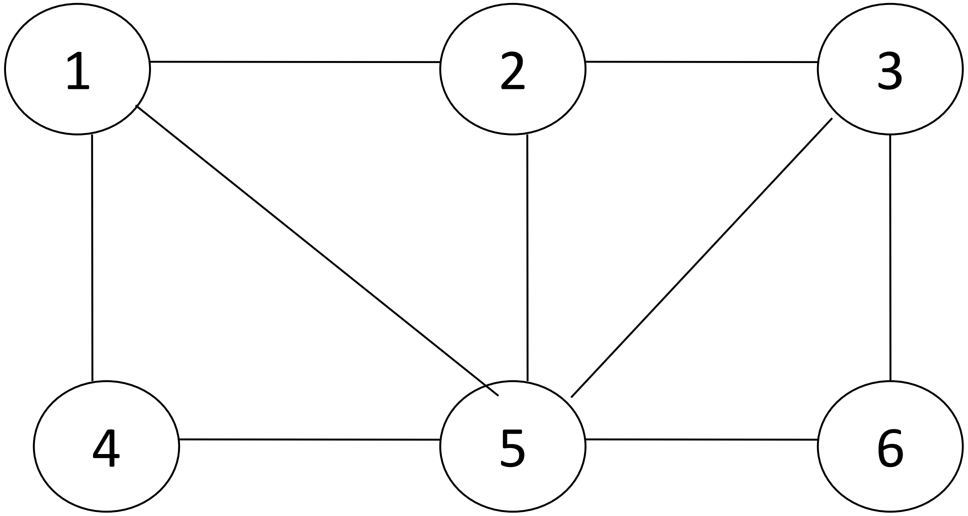
Connection-oriented service

- Three phases: connection set up, message transmission and connection release
- All packets in a message follows the same route
- Packets are received in the order they were sent
- Every node maintains connection state (id) information
- Small header
- Supports both CBR and VBR traffic
- Better support for Quality-of-Service (QoS). QoS: delay, jitter, packet/cell loss rate, bandwidth
- Connection setup/release overhead
- Bandwidth utilization may not be efficient
- E.g.,
  - ATM (Asynchronous Transfer Mode Networks)
  - SONET (Synchronous Optical Networks)
  - WDM (wavelength-division multiplexing optical networks)
  - ‘Connection-oriented’ Ethernet (used in carrier grade networks). This is different from the typical Ethernet which is connectionless!
  - TCP

Connectionless service (Datagram service)

- Packets in a message need not follow the same route
- Packets may be received out of order
- No connection state (id) information is maintained
- large header (every packet carries source, destination information…)
- No QoS guarantee (poor QoS support)
- No Connection setup/release overhead
- Bandwidth utilization is efficient
- E.g.,
  - IP
  - UDP
  - ‘Typical’ Ethernet

Connection-oriented and Connectionless Routing



Source: 1   destination: 6

Connection-oriented: 1-5-3-6 (all packets)

Connectionless: 1-4-5-6 (pkt1) 1-5-3-6 (pkt2) 1-5-6 (pkt3)

Computer networks: classification

- The network edge and the network core:
  - The network edge: End-systems/hosts and Access networks.
    - Access networks: DSL, Cable, FTTH, (old Dial-Up), Satellite, Ethernet, WiFi, Wide-Area Wireless Access: 3G, LTE, LTE-Advanced,…
  - The network core: ISPs and Internet backbones

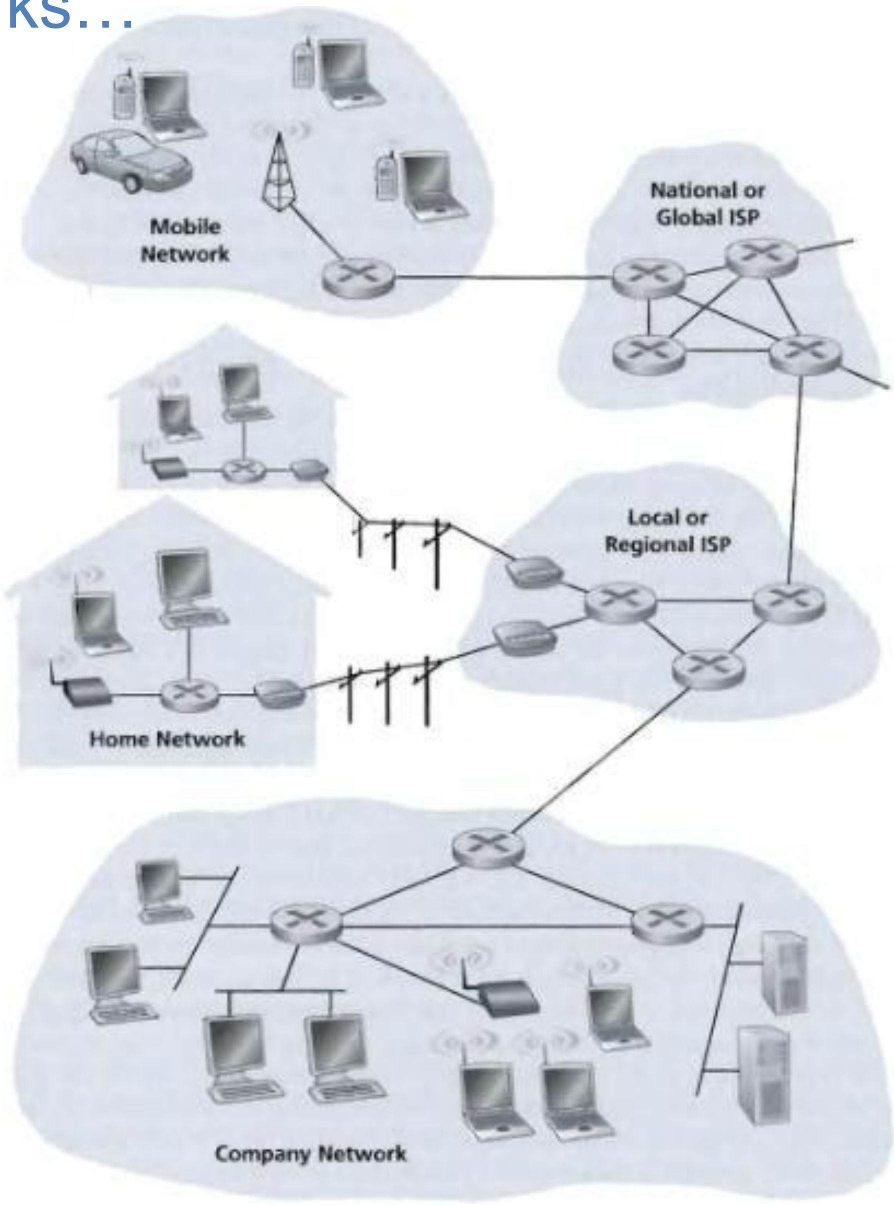
Other classifications:

» LANs, MANs, and WANs
  - Local area networks (LANs)
  - Metropolitan area networks (MANs)
  - Wide area networks (WANs)

» Switched networks (or point-to-point networks) and broadcast networks

Classification of networks…

- The network edge and the network core



Classification of networks…

The network edge:
End systems/Hosts

How applications interact?

- end systems (hosts):
  - run application programs
  - e.g. Web, email
  - at "edge of network"
- client/server model
  - client host requests, receives service from always-on server
  - e.g. Web browser/server; email client/server
- peer-peer model:
  - minimal (or no) use of dedicated servers
  - e.g. Skype, BitTorrent

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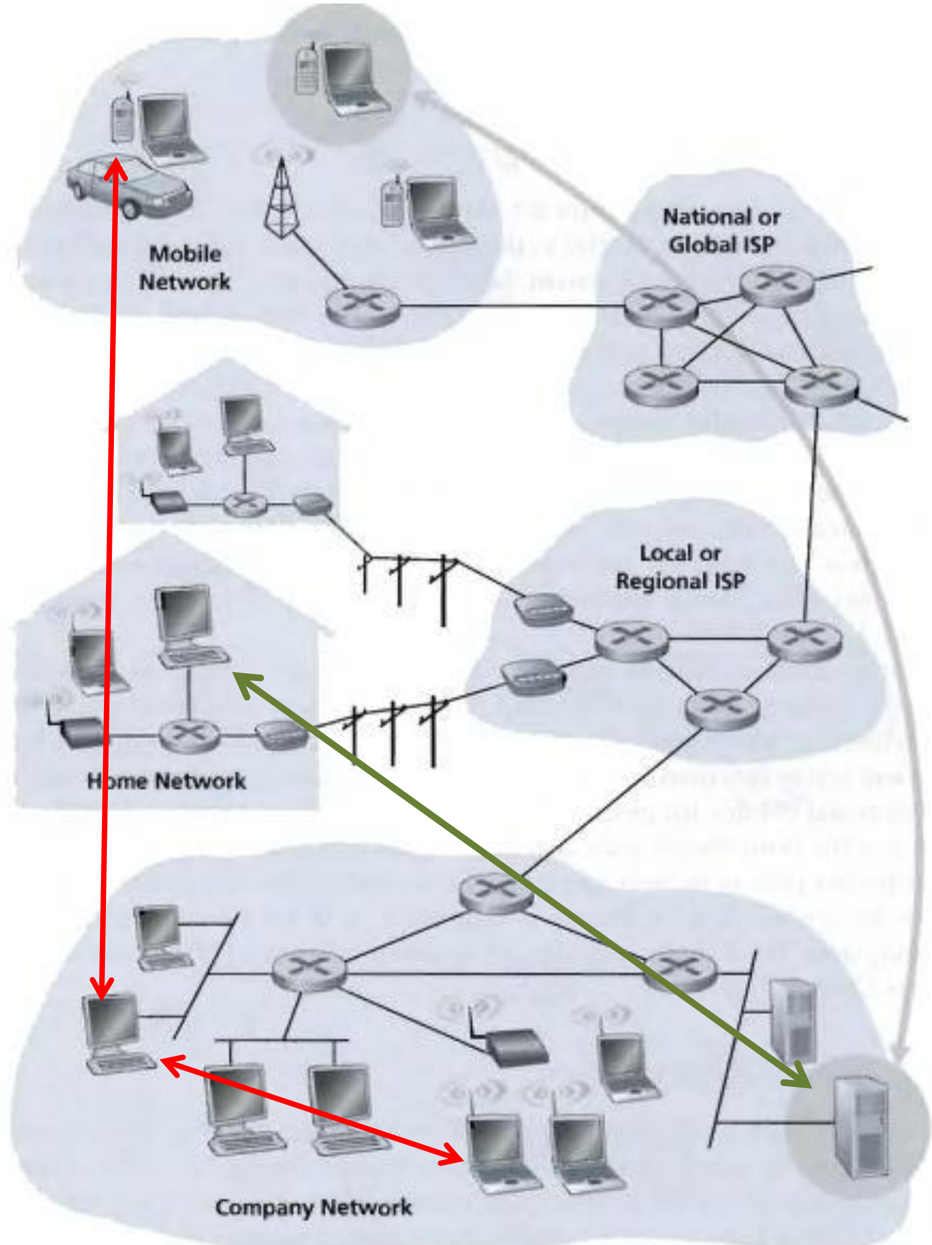
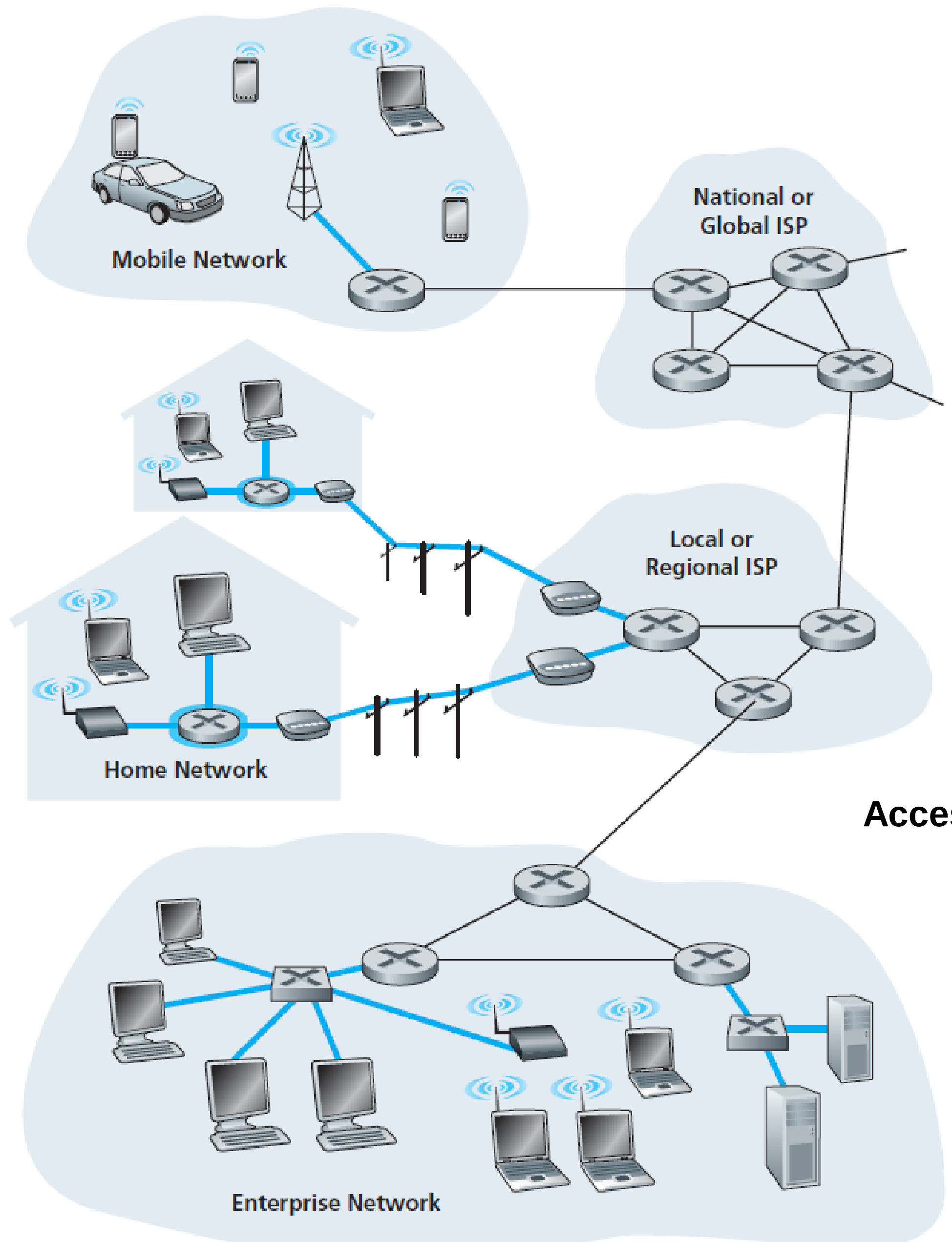


Figure 1.3 ♦ End-system interaction

Classification of networks…

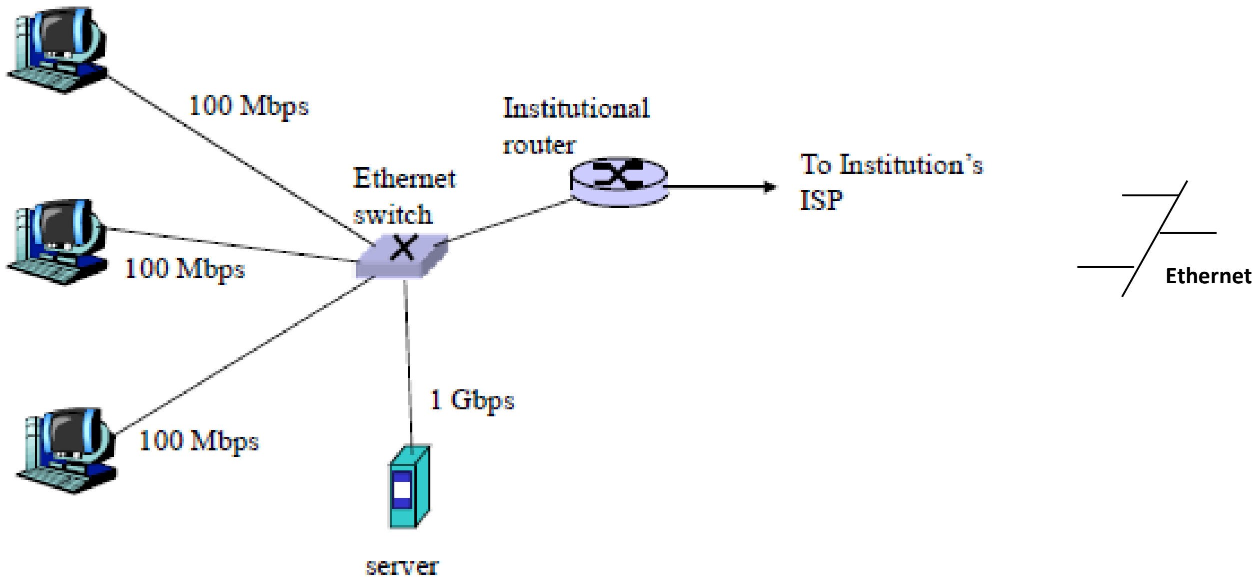
The network edge:
Access networks

How to connect end systems to the edge router?



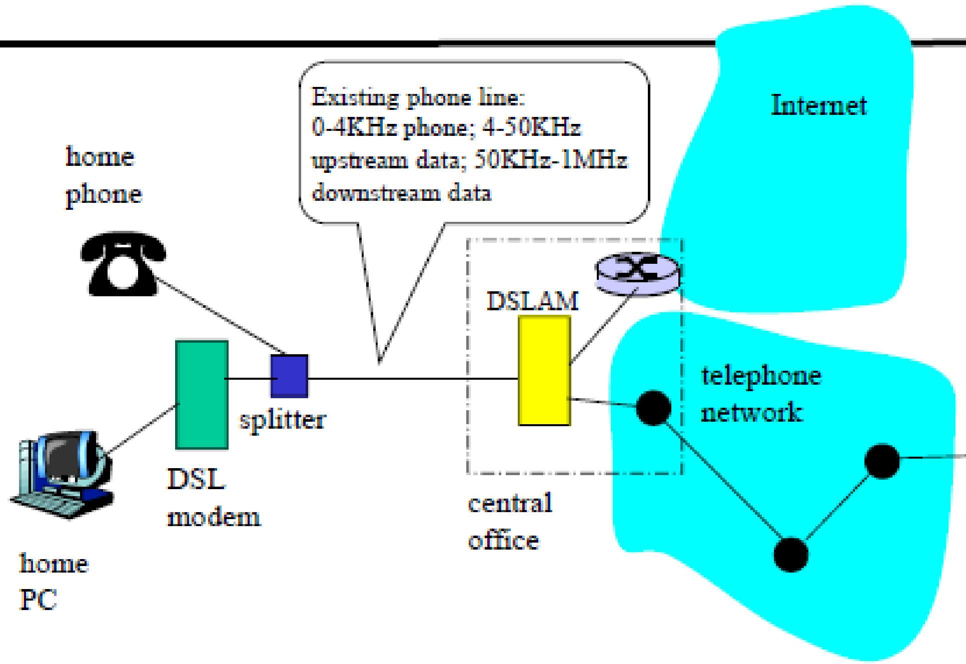
Access networks

Ethernet Internet access



- Typically used in companies, universities, etc
- 10 Mbs, 100Mbps, 1Gbps, 10Gbps Ethernet
- Today, end systems typically connect into Ethernet switch

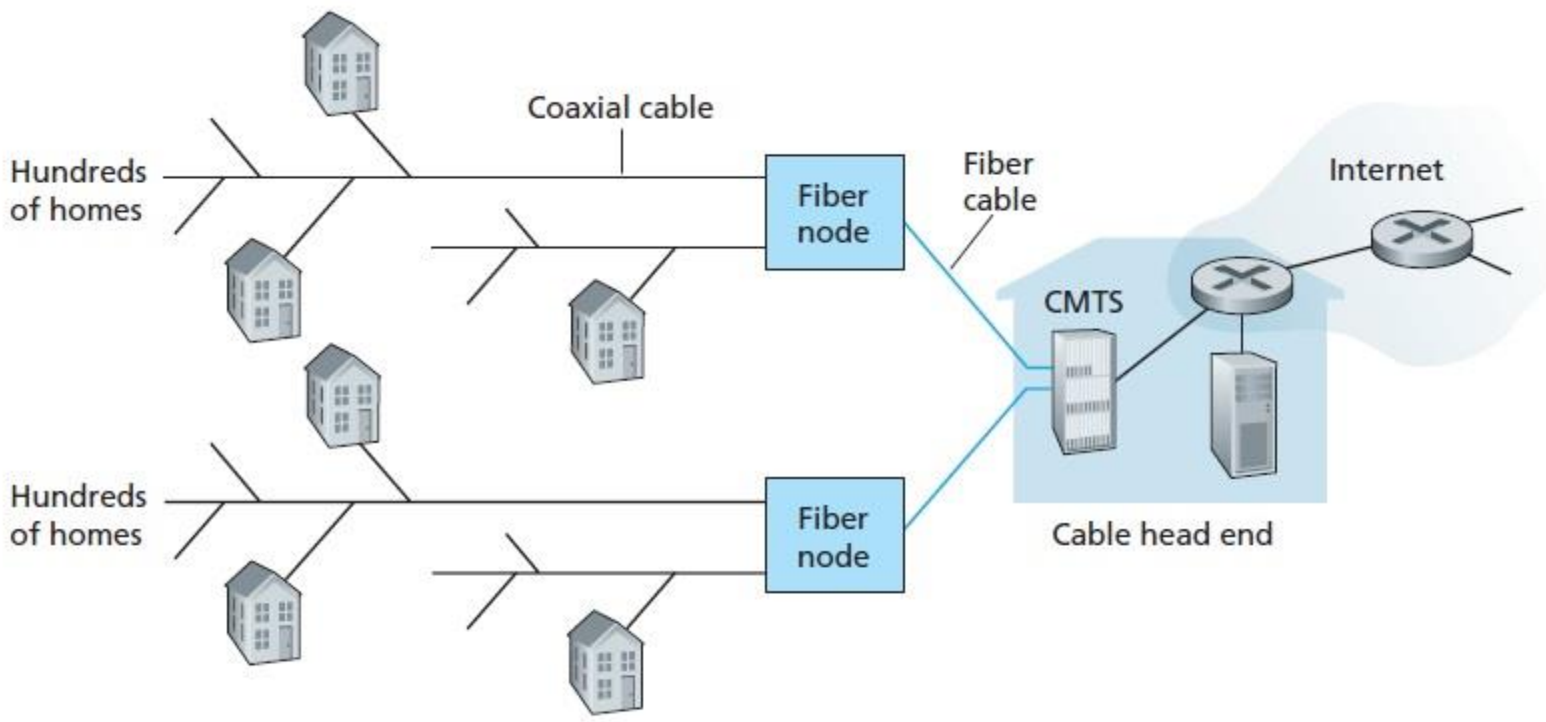
Digital Subscriber Line (DSL)



Customer's DSL modem uses the telephone line (twisted pair copper wire) to exchange data with a digital subscriber line access multiplexer (DSLAM) @ central office (CO)

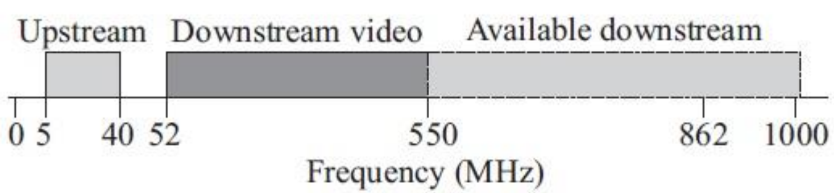
- Also uses existing telephone infrastruture
- up to 1 Mbps upstream
- up to 8 Mbps downstream or up to 16 Mbps
- dedicated physical line to telephone central office

Cable Internet Access



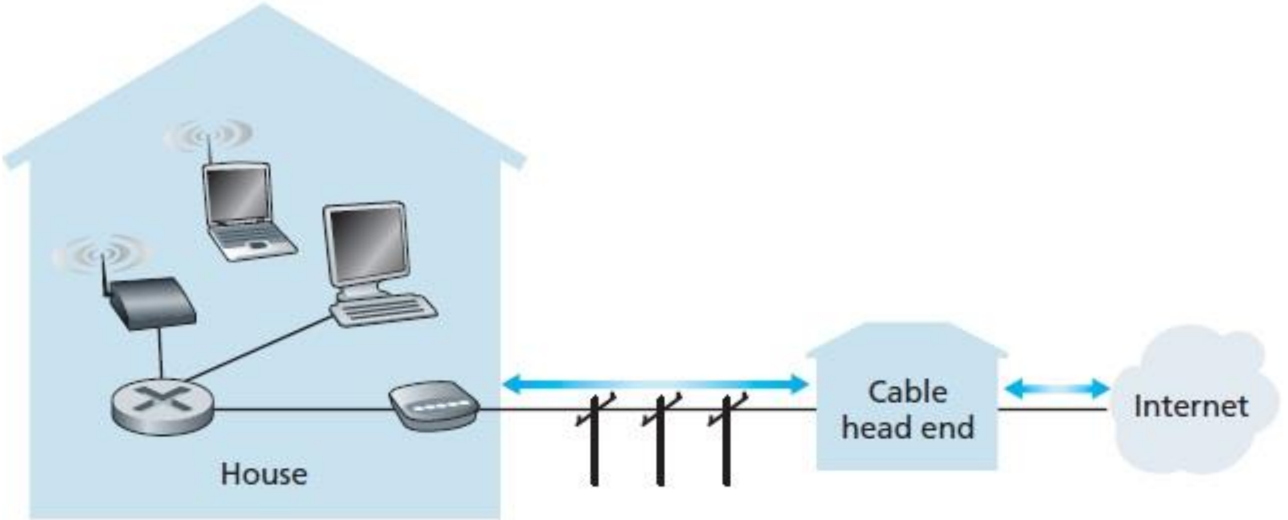
A hybrid fiber coax (HFC) network

Cable internet access requires special modems: cable modems
CMTS: cable modem termination system



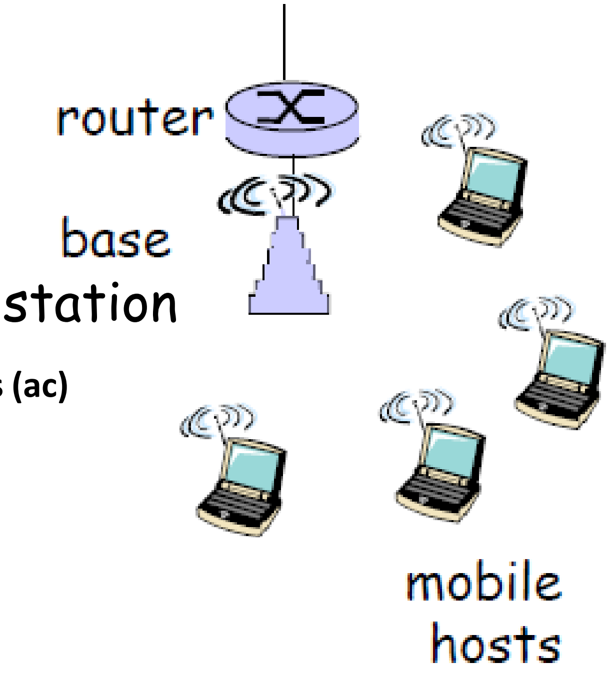
Bandwidth allocation in an enhanced HFC network

A typical home network



Wireless Access Networks

- shared *wireless* access network connects end system to router
  - via base station aka "access point"
- wireless LANs:
  - WiFi: IEEE 802.11 a/b/g/n/ac
    up to 54Mbps(g) 600Mbps (n) several Gbps (ac)
- wider-area wireless access
  - provided by telco operator
  - Cellular Internet Access
  - WiMAX

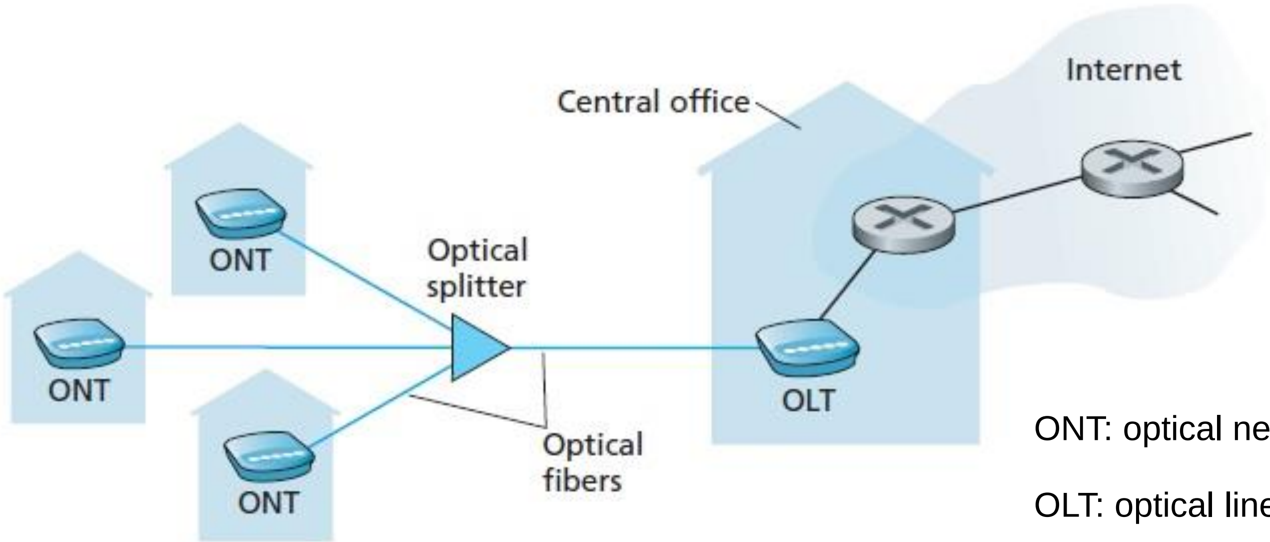


Access networks

FTTx, x = Home (FTTH)
= Building (FTTB)
= Premises (FTTP)
= Curb (FTTC)
= ….

Fiber to the home (FTTH)

- FTTH concept — provide an optical fiber path from the CO directly to the home
- An optical distribution network architecture for FTTH: Passive Optical Networks (PONs)



ONT: optical network terminator

OLT: optical line terminator

FTTH using the PON distribution architecture

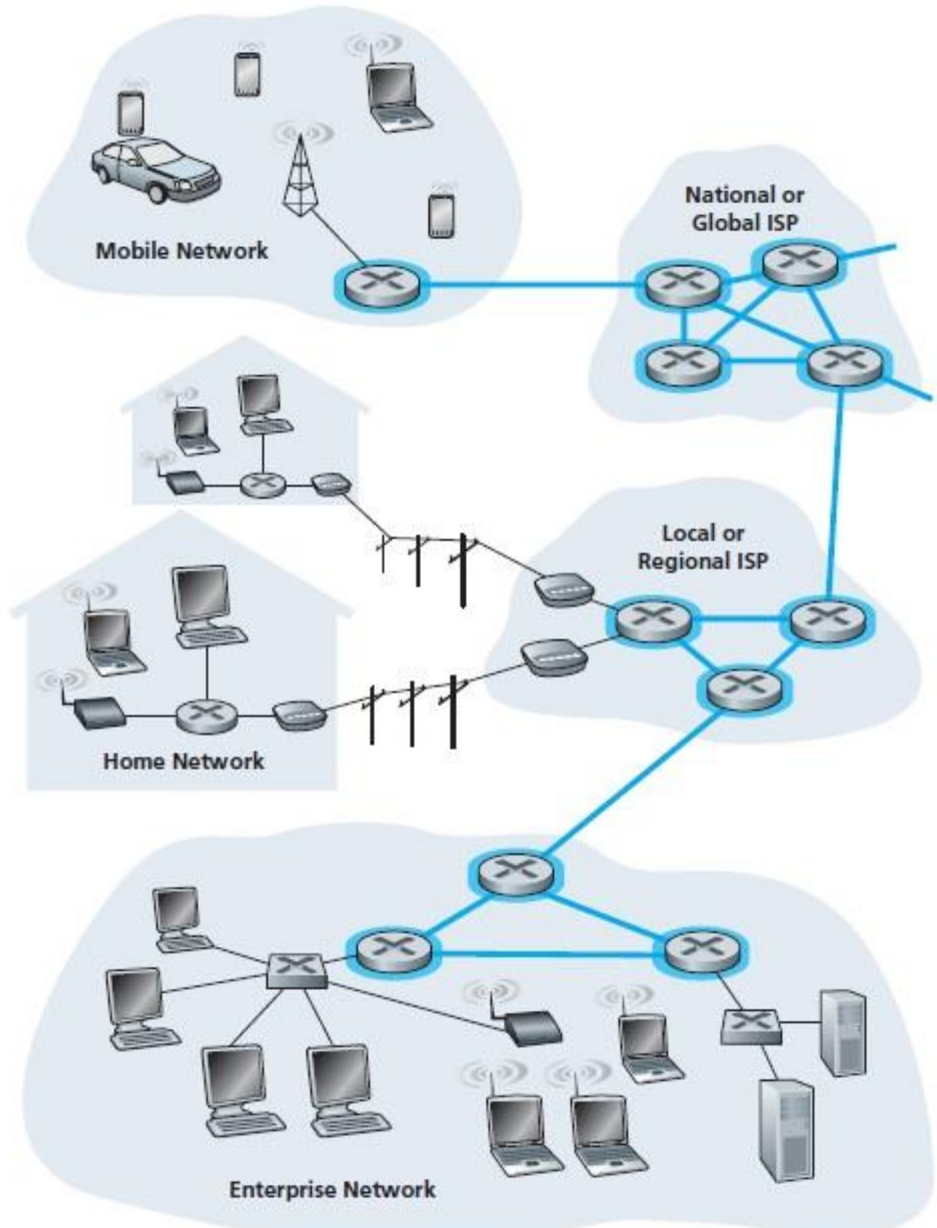
- Different forms of PON:
  - TPON (PON for Telephony) (the most common type), APON, BPON, GPON, 10G-EPON
  - WDM PON
  - WRPON

Classification of networks…

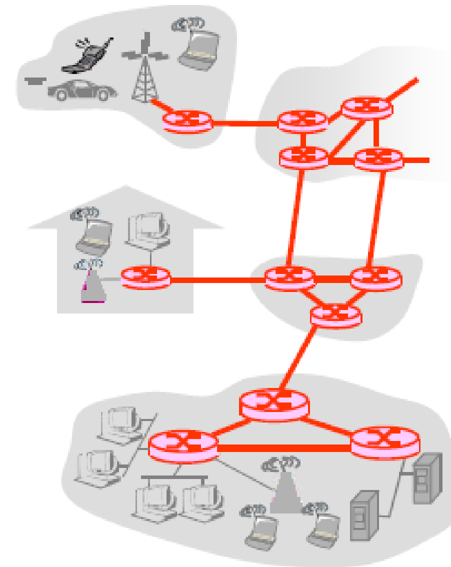
The network core:

The mesh of packet switches and links that interconnect end systems

How do we move data in the core?



Switching Techniques

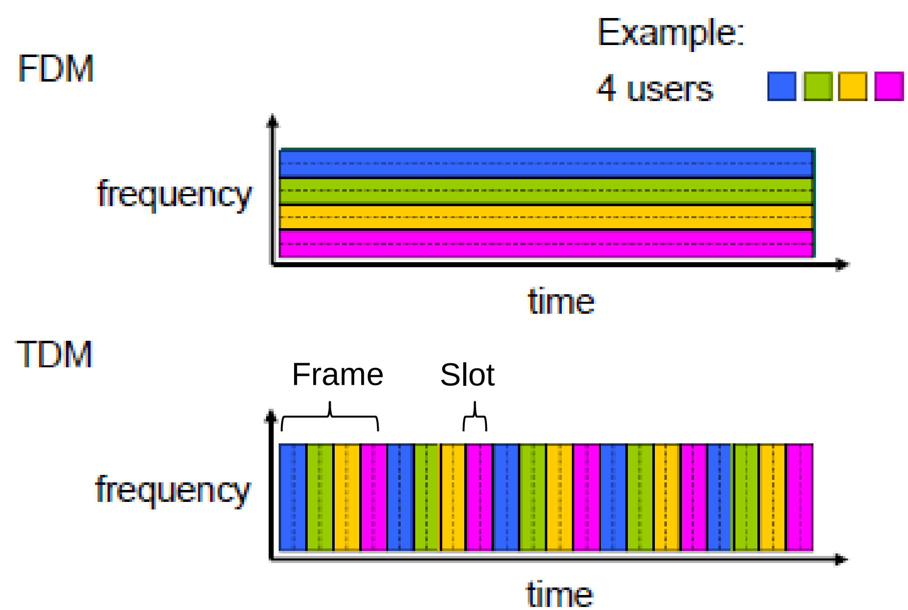


Circuit switching

- Link bandwidth (capacity) is divided into fixed-size circuits eg. telephone network, SONET, WDM
- 3 distinct phases: circuit setup, data transmission, and circuit release phases
- Connection-oriented service
- An application uses a fixed path to be used by all data. Data arrive in sequence
- Fixed bandwidth (BW) circuits, low BW utilization
- Guaranteed service

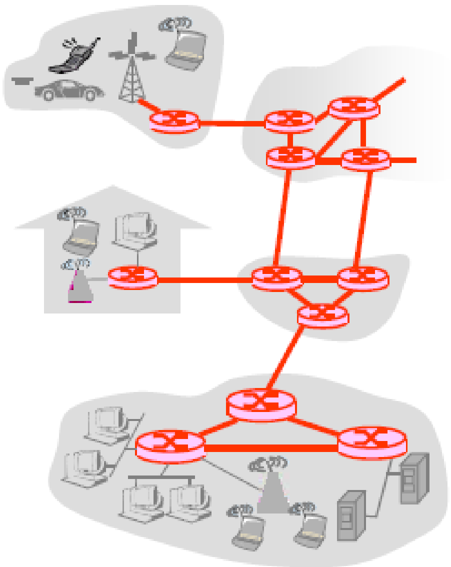
Dividing link bandwidth into 'pieces':

- Frequency division (Frequency division multiplexing- FDM)
- Time division (Time division multiplexing- TDM)



With FDM, each circuit continuously gets a fraction of the bandwidth. With TDM, each circuit gets all of the bandwidth periodically during brief intervals of time (that is, during slots)

Switching Techniques…



Network Core: Packet Switching

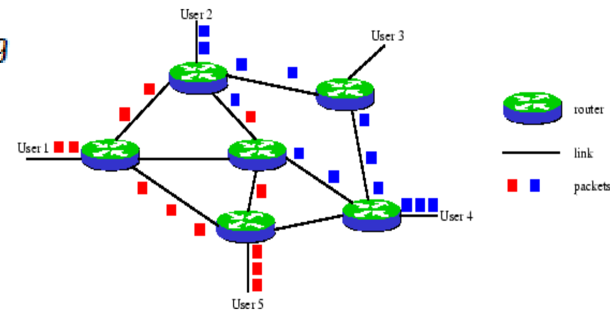
each end-end data stream divided into *packets*

- user A, B packets *share* network resources
- each packet uses full link bandwidth
- resources used *as needed*

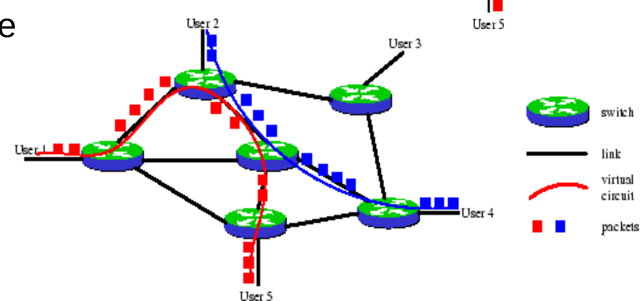
Bandwidth division into "pieces"
Dedicated allocation
Resource reservation

resource contention:

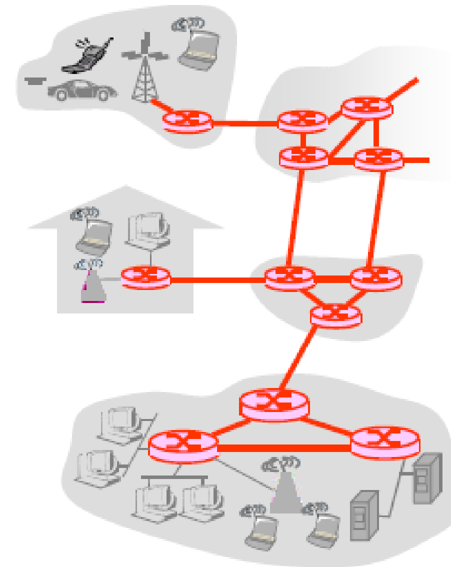
- aggregate resource demand can exceed amount available
- congestion: packets queue, wait for link use
- store and forward: packets move one hop at a time
  - Node receives complete packet before forwarding



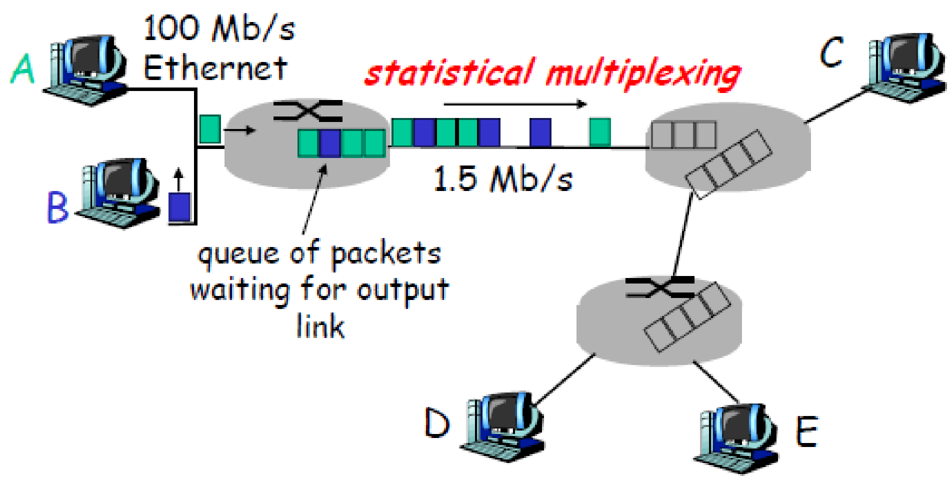
- Could be **connection-less service (IP networks)** (Different packets of the same message may traverse different routes) or connection oriented service (ATM networks)



Switching Techniques…

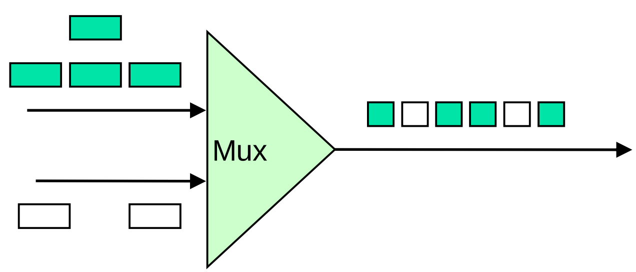


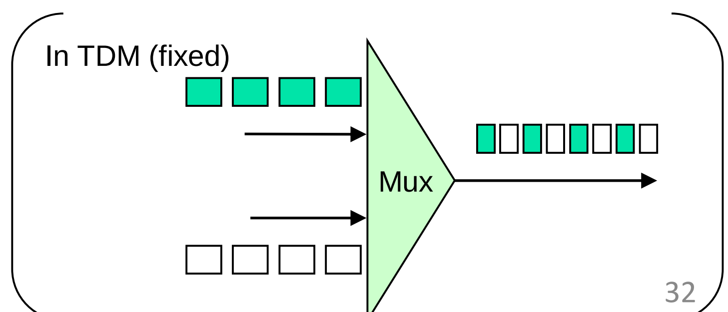
Packet Switching: Statistical Multiplexing



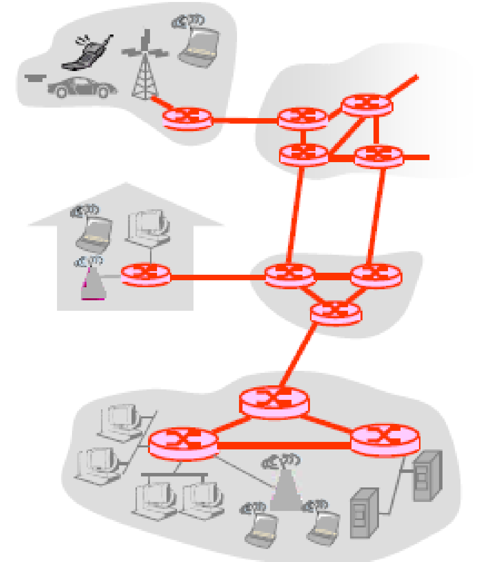
Sequence of A & B packets does not have fixed pattern, bandwidth shared on demand ➔ *statistical multiplexing.*

TDM: each host gets same slot in revolving TDM frame.

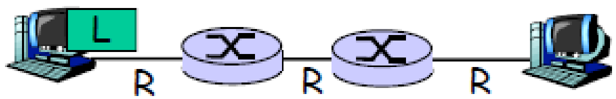




Switching Techniques…



Packet-switching: store-and-forward



- takes L/R seconds to transmit (push out) packet of L bits on to link at R bps
- *store and forward:* entire packet must arrive at router before it can be transmitted on next link
- delay* = 3L/R (assuming zero propagation delay)

Example:

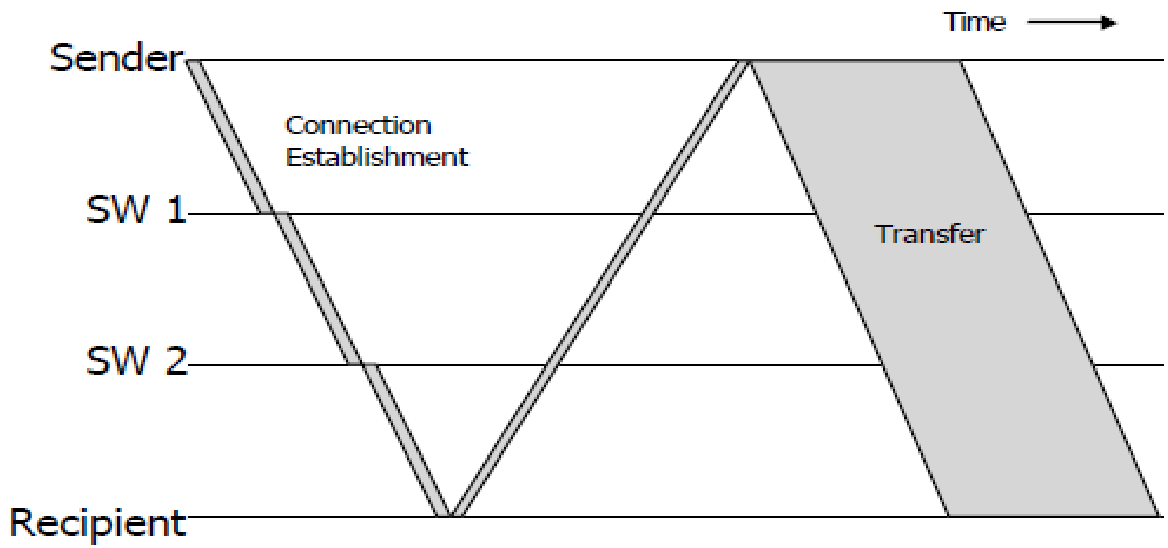
- L = 7.5 Mbits
- R = 1.5 Mbps
- transmission delay = 15 sec

Most packet switches use store-and-forward transmission.
Routers/nodes need to receive, store, and *process* the entire packet before forwarding.

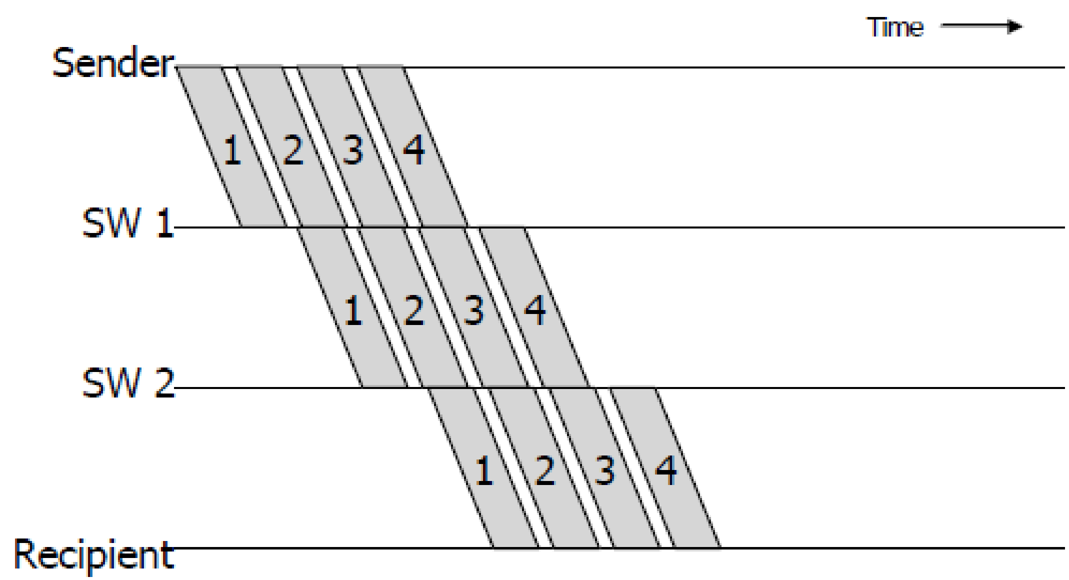
* the amount of time that elapses from when the source begins to send the packet until the destination has received the entire packet

- Transmission with two intermediate nodes

Circuit switching



Packet Switching



Direction of Information Flow

- **Simplex** communication: communication that occurs in one direction only
  - E.g., broadcast radio and television, Radio controlled models

- ***Half-duplex*** communication: communication in both directions, but only one direction at a time (not simultaneously).
  - E.g, walkie-talkies (two-way radio with half-duplex), old Ethernet networks, Wireless LANs (WiFi)(IEEE 802.11)

- ***Full-duplex*** communication: communication in both directions simultaneously.
  - Full-duplex channels can be constructed either as a pair of simplex links or using one channel designed to permit bidirectional simultaneous transmissions. E.g.,
  - Ethernet (today): Switched Ethernet networks typically employ either twisted pair or fiber optic cabling, with separate conductors/fiber-strands for sending and receiving data
  - Routers connected with bi-directional links
  - In optical networks, nodes are connected with bi-directional links (a separate fiber for each direction)
  - Most cellular systems (cellular phones – two way radio with full-duplex), WiMAX

    (two different radio frequencies or channels to carry the two directions of conversation simultaneously-Frequency Division Duplexing) (other method- Time Division Duplexing)
  - ADSL

What is a protocol?

Some known protocols:

Postal Mail Protocol

- Sender’s address at the left -upper corner
- Recipient ’s address at the center
- Stamps at the right -upper corner

Walkie-Talkie Protocol

- Listen
- Hear “Over!” or nothing
- Say a sentence and “Over!” while holding down the button
- Release the button

Protocols in computer communication networks

- TCP, IP, HTTP, FTP, Ethernet, CDMA, ATM, SONET, PPP, ICMP, SNMP, SMTP, ARP, RARP, DHCP, RTP, UDP, HDLC, IGMP, RIP, OSFP, LCP, NHRP, RSVP,…

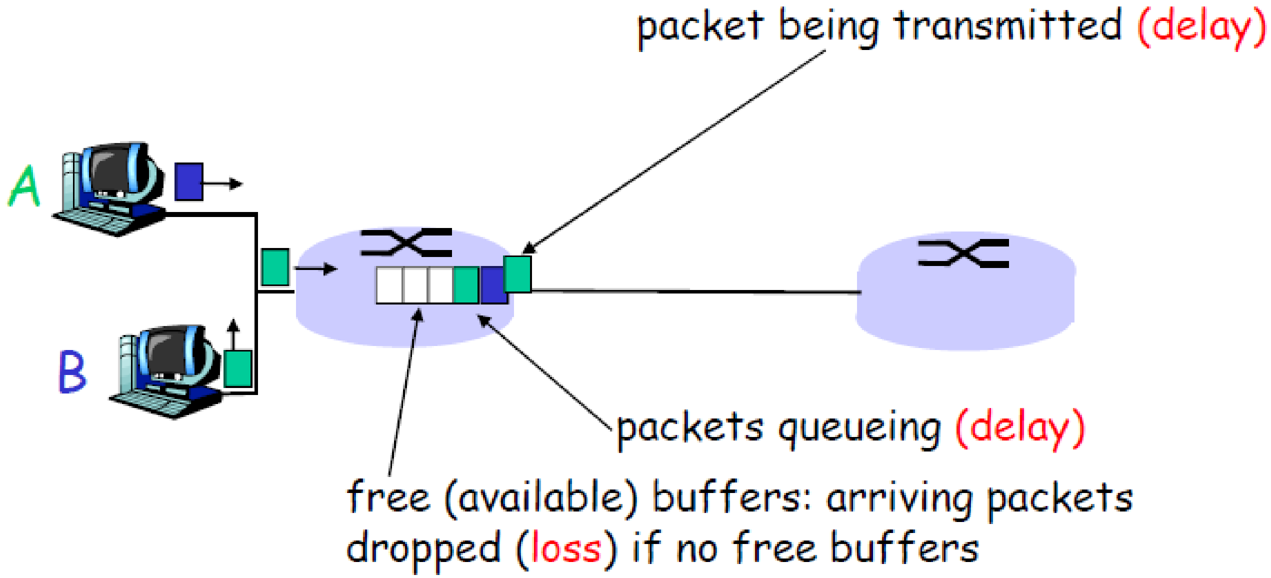
Protocols define format, order of messages sent and received among network entities, and actions taken on message transmission and receipt

Performance measurement and associated issues: delay loss and throughput

How do loss and delay occur?

packets *queue* in router buffers

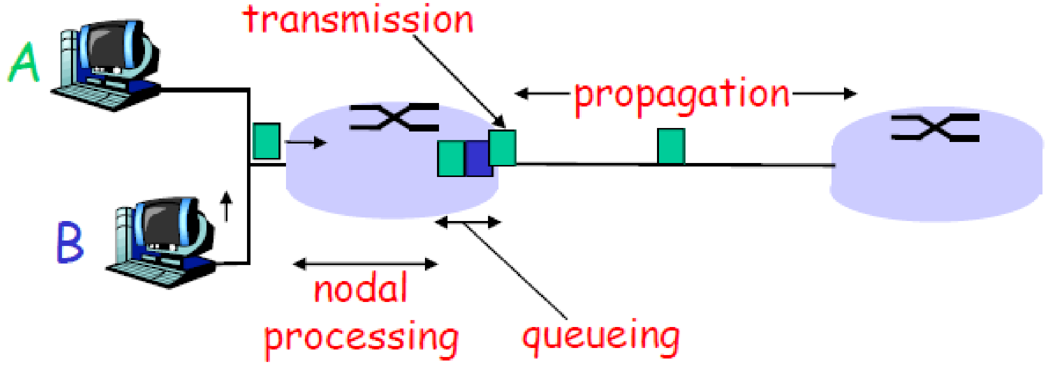
- packet arrival rate to link exceeds output link capacity
- packets queue, wait for turn



Performance measurement and associated issues: delay loss and throughput

Four sources of packet delay

- 1. nodal processing:
  - check bit errors
  - determine output link
- 2. queueing
  - time waiting at output link for transmission
  - depends on congestion level of router



Performance measurement and associated issues: delay loss and throughput

Delay in packet-switched networks

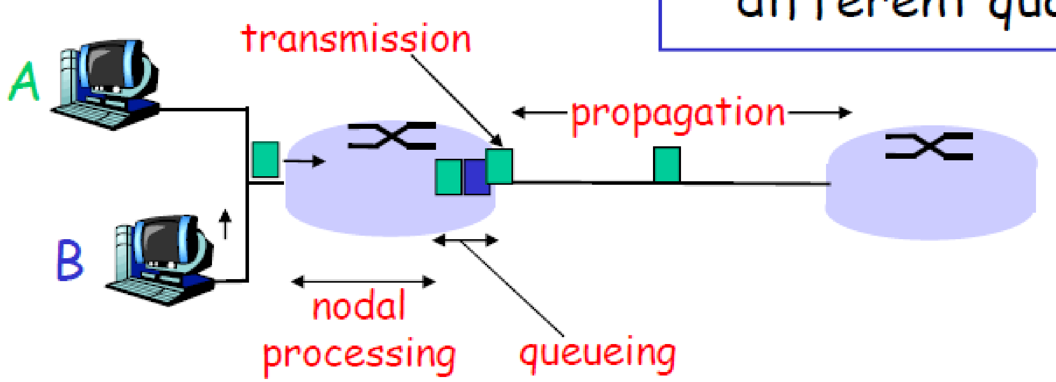
3. Transmission delay:

- R=link bandwidth (bps)
- L=packet length (bits)
- time to send bits into link = L/R

4. Propagation delay:

- d = length of physical link
- s = propagation speed in medium (~2×10^8 m/sec)
- propagation delay = d/s

Note: s and R are *very* different quantities!



Performance measurement and associated issues: delay loss and throughput

Nodal delay

$$d_{\text{nodal}} = d_{\text{proc}} + d_{\text{queue}} + d_{\text{trans}} + d_{\text{prop}}$$

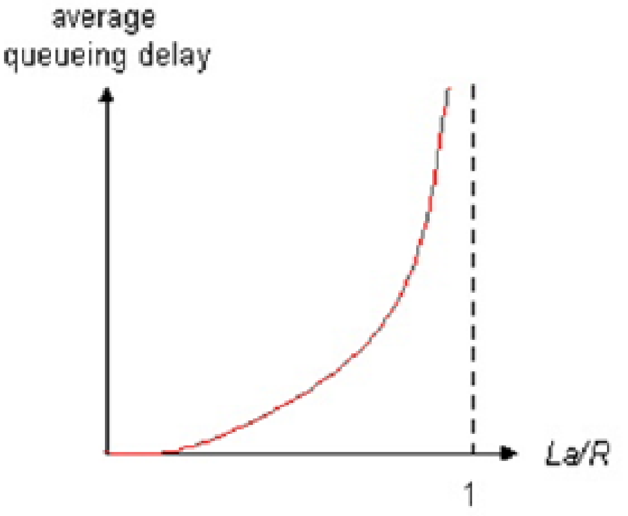
- d_{proc} = processing delay
  - typically a few microsecs or less
- d_{queue} = queuing delay
  - depends on congestion
- d_{trans} = transmission delay
  - = L/R, significant for low-speed links
- d_{prop} = propagation delay
  - a few microsecs to hundreds of msecs

Performance measurement and associated issues: delay loss and throughput

Queueing delay (revisited)

- R=link bandwidth (bps)
- L=packet length (bits)
- a=average packet arrival rate

traffic intensity = La/R

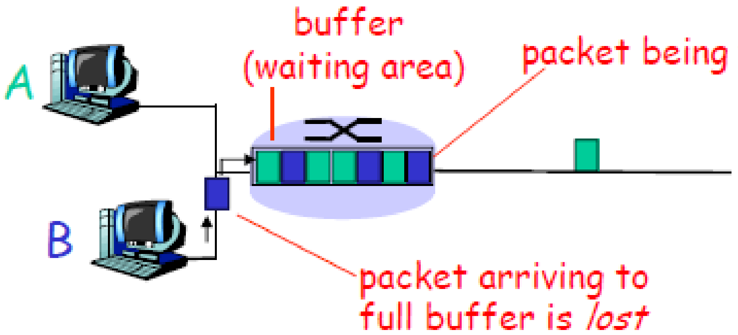


- La/R ~ 0: average queueing delay small
- La/R -> 1: delays become large
- La/R > 1: more "work" arriving than can be serviced, average delay infinite!

Performance measurement and associated issues: delay loss and throughput

Packet loss

- queue (aka buffer) preceding link in buffer has finite capacity
- packet arriving to full queue dropped (aka lost)
- lost packet may be retransmitted by previous node, by source end system, or not at all



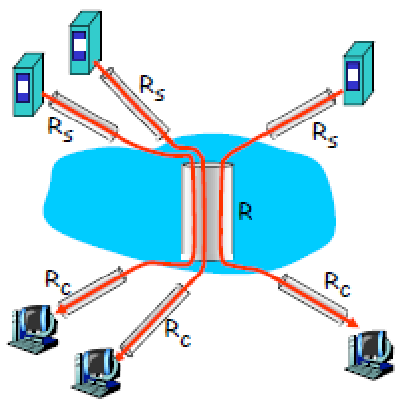
Performance measurement and associated issues: delay loss and throughput

Throughput or Effective throughput

- Number of bits transferred per unit time (measured quantity)
- Ratio between message size and message transfer time
- Notation: bps, packets per second
  message size 1 KB = 2^{10} bytes

Throughput: Internet scenario

- per-connection end-end throughput: $\min(R_c, R_s, R/10)$
- in practice: R_c or R_s is often bottleneck



10 connections (fairly) share backbone bottleneck link R bits/sec