General requirements

- So, you want to build a network ...
  - First you need to know the requirements the network must satisfy
  - The requirements vary depending on who you ask (different views)

- Requirements from different views:
  - Application programmer: service specific needs, e.g., packets sent should not get lost and should arrive in the same order
  - Network designer: cost effective design, efficient and fair usage of network resources
  - Network provider: easy management, reliable, fault isolation
  - Users expect services: e-mail, tele- and videoconferencing, e-commerce, video-on-demand, ...
Computer network characteristics

• Typically communications networks optimized for some service
  – telephone network
  – television/radio broadcast network
  – user terminals special purpose devices
• Modern computer networks are more general:
  – terminals general purpose PCs/workstations
  – networks able to carry essentially any kind of data
  – support many different applications
• Topics in this lecture
  – How computer networks provide connectivity (Requirement 1)
  – How efficient resource sharing is achieved (Requirement 2)
  – How applications “talk” to each other (Requirement 3)
  – How network performance affects the system (Requirement 4)
  – Requirements are reflected in network architectures
  – Basically, we get a “snap shot” of the issues covered in this course

Outline

• Achieving connectivity
• Methods for resource sharing
• Enabling application level communication
• Performance issues
• Network architecture
• Historical background
Basic building blocks

- A network, in principle, consists of nodes and links connecting the nodes.
- Network nodes: PCs, servers, special purpose hardware
  - Internet terminology
    - hosts, end-systems: PCs and servers running network applications
    - routers (switches): store and forward packets through the network
- Links: optical fiber, coaxial cable, twisted pair copper, radio, etc.
  - point-to-point
    - hosts directly connected
  - multiple access (LANs, etc.)
    - hosts share the common transmission medium

Building larger networks

- Large networks can not be built based on point-to-point connectivity
  ⇒ use routers (switches) to interconnect hosts to each other
- Nodes connected together through switches to form connected networks
- Networks connected together through gateway routers to form bigger entities
Network edge vs. core network in the Internet

- **Access network**
  - customers are connected to the core network by the access network
  - link speeds comparably low
    - access technologies: dial up (modem over twisted pair), xDSL, cable modem, ...
  - may contain billing functionality, traffic management for each access
  - tree topology

- **Core network**
  - no end users directly connected to the core
  - high link speeds
    - SDH/SONET over fiber based technologies
  - simple functionality (forwards packets)
  - mesh topology

Internet

- Consists of millions of hosts (end systems) connected by links and routers
- Hosts exchange messages by using protocols offering e.g.
  - reliable transfer
  - packet sequence integrity
- Routers forward data
  - based on best effort service
  - no guarantees on loss or timeliness
- “Network of networks”
  - loosely hierarchical
  - public Internet vs. private Intranets
  - Internet access provided by ISPs (Internet Service Providers)
Issues of scale

- Easy to build and manage a network supporting 100 users, but what if the number of users is 100 million ...
- A system allowing unlimited growth in size is said to **scale**.
  - Scalability a very desirable property for networking technologies
- Scalability of networks is often influenced very much by
  - the nature of the guarantees regarding service quality
  - the amount of information that the network has about the users
- One reason for the success of Internet technology is its scalability
  - The networking paradigm is based on best-effort service (no guarantees are made about the service quality) and the network is connectionless
  - The nodes of the network do not store any state information of the users/connections
  - New nodes and users can be added to the network (almost) without any complexity increases
  - Only the routing is affected by the increase in the number of nodes (route computation complexity grows with the number of nodes)

Switching modes

- **Circuit switching**
  - telephone networks
  - mobile telephone networks
  - ATM technology is also based on circuit switching
    - fast packet switching with fixed length packets (cells): ATM
    - integration of different traffic types (voice, data, video)
      ⇒ multiservice networks
- **Packet switching**
  - data networks
  - two possibilities
    - **connectionless**: e.g. Internet (IP), SS7 (MTP)
    - **connection oriented**: e.g. X.25, Frame Relay
Circuit switching

- **Connection oriented:**
  - connections **set up** end-to-end before information transfer
  - resources **reserved** for the whole duration of connection
- Information transfer as a **continuous stream**
- Before information transfer
  - delay (to set up the connection)
- During information transfer
  - no overhead
  - no extra delays

(Connectionless) packet switching

- **Connectionless:**
  - no connection set-up
  - no resource reservation
- Information transfer by using **discrete packets**
  - varying length
  - global address (of the destination)
- Before information transfer
  - no delays
- During information transfer
  - overhead (header bytes)
  - packet processing delays
  - queuing delays (since packets compete for shared resources)
  - routers "store-and-forward"
Addressing and routing

- Hosts need to distinguish each other when wishing to communicate
- Each host is assigned a unique byte-string known as **address**
- When a sender communicates with some destination B, in a packet switched network
  - the address of the destination (B) is attached to each packet, and
  - each router determines how to forward the packet based on the destination address
  - **rout ing** is the systematic process of determining where a packet is sent (which output port) based on the destination address
- Different addressing and routing scenarios
  - **unicast**: between a single sender and destination pair
  - **broadcast**: from a single user to all other users (e.g. network control messages)
  - **multicast**: from a single user to a subset of all users (e.g. distribution of files)

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Multiplexing

- Multiplexing
  - mechanism for achieving resource sharing, i.e., sharing of link bandwidth
- Problem:
  - How can the link bandwidth be shared among \( n \) different senders
- 1st approach: partition the bandwidth strictly for all users
  - FDM and TDM

Frequency Division Multiplexing (FDM)

- FDM
  - oldest multiplexing technique
  - used e.g. in analogue circuit switched systems
  - fixed portion (frequency band) of the link bandwidth reserved for each channel
- FDM multiplexer is lossless
  - input: \( n \) 1-channel physical connections
  - output: 1 \( n \)-channel physical connection
Time Division Multiplexing (TDM)

- **TDM**
  - used in digital circuit switched systems and digital transmission systems
  - information conveyed on a link transferred in *frames* of fixed length
  - fixed portion (time slot) of each frame reserved for each channel
- **TDM multiplexer is lossless**
  - input: $n$ 1-channel physical connections
  - output: 1 $n$-channel physical connection

Statistical multiplexing (1)

- **FDM and TDM are inefficient**
  - If a sender has no data to transmit, the bandwidth allocated to the sender can not be used by others ⇒ **statistical multiplexing**
- **In statistical multiplexing**
  - basic transmission unit is called a packet
  - physical link is shared over time (cf. TDM) but **on-demand** (per each packet)
  - simultaneous packet arrivals are buffered (contention)
    - as a result, packets from multiple senders are *interleaved* at the output
  - buffer space is finite, thus buffer overflow is possible (congestion)
Statistical multiplexing (2)

- Statistical multiplexer is (typically) lossy
  - input: \( n \) physical connections with link speeds \( R_i \) \( (i = 1, \ldots, n) \)
  - output: 1 physical connection with link speed \( C \leq R_1 + \ldots + R_n \)

- However, the loss probability can be decreased by enlarging the buffer
  - with an “infinite” buffer enough that \( C \) exceeds \textit{average aggregated} input rate
  - possible to dimension the size of the buffer such that a given loss probability is achieved (under some assumptions regarding the traffic)

- Statistical multiplexer and QoS (Quality of Service)
  - determining which packet to transmit from the buffer is called \textit{scheduling}
    - FIFO: packets are served in the arrival order
    - Round robin: each connection (class) has own queue and they are served cyclically according to some weights
      - Many more exist...
  - by using different scheduling mechanisms, some connections can be given “preferential” treatment (e.g., weighted round-robin) \( \Rightarrow \) QoS enabled networks
Communication needs of applications

• Applications (processes) running on hosts need to communicate
  – different applications have different needs
• Typical application considerations
  – reliability?
  – packet sequence order?
  – security?
• Network design challenge
  – identify the set of common services that the applications need
  – hide the complexity of the network without imposing too many constraints on the applications
• Network provides “logical channels”
  – IPC = Inter Process Communication
  – fills in the “logical gap”

Application requirement classification

1 Client/server applications (request/reply applications)
  – client process makes a request and the server process replies
  – strict requirements on packet loss (no loss), may have security requirements
  – Examples: file transfer (FTP), file systems (NFS), HTML documents on the web, digital libraries

2 Streaming applications
  – sender generates a continuous stream of packets
    • the stream can correspond to, e.g., digitized audio or video
    • applications have relatively tight requirements on the timeliness of packet delivery, but they can tolerate packet loss to some degree
      • videoconferencing has tighter demands than video on-demand
      • Security? Conferencing may require, e.g., encrypted transmission…

• Question 1: Are only 2 categories enough?
• Question 2: Where is the functionality of each service implemented?
Reliable transfer - what can go wrong?

- Reliable transfer: one of the most important service properties
  - “network hides certain failures to make the network seem more reliable”
- Error types
  - **Bit errors**: bit or burst of bits is corrupted
    - Error correction detection may be able to fix the problem
  - **Packet errors**: complete packet is lost
    - Due to unrecoverable bit errors, congestion (most likely reason), software errors (misplaced packets, relatively rare)
    - Problem: Not easy to distinguish between packets that are excessively late (due to e.g. severe overload) and actually lost packets.
  - **Node/link failures**:
    - A physical link is damaged/cut, router crashes …
    - Can cause massive service disruptions
    - In Internet routing protocols can recover from link failures
    - Problem: Not easy to determine if a router is e.g. completely down or just congested.

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Performance measures (1): bandwidth

- Bandwidth = throughput
  - nof bits that can be transmitted over the network in a given time
  - unit: bits per second (bps), e.g. 10 Mbps (cf. MB = megabytes = 8 Mb)
- Link bandwidth and end-to-end bandwidth
  - bandwidth of a physical link has a deterministic value, e.g. 155 Mbps
  - link bandwidths are constantly improving: link bandwidths in the backbone
  - end-to-end the received bandwidth of an application depends on
    - other traffic in the network (congestion)
    - application limitations (CPU speed of the computer)
    - protocol overhead (each bit sent by the application is “wrapped” in possibly several “envelopes” until the bit is transmitted on a physical link)

Performance measures (2): latency

- Latency = delay
  - How long it takes a message to travel from one end of the network to another
  - Measured in units of time, e.g., latency across US continent 24 ms
  - RTT (round trip time): time it takes a message to reach its destination and come back to the sender
- Components: propagation delay, transmission delay, queuing delay
  - Speed of Light: 2.3 x 10^8 m/s in cable, 2.0 x 10^8 m/s in fiber
- Applications can be either bandwidth or latency bound
  - Telnet sessions are latency bound but large FTP transfers are bw bound
Delay x bandwidth product

- The product of RTT and bandwidth determines
  - the amount of information transmitted by the user before any feedback from the destination can be received
- In broadband wide-area-networks (WAN) this product can be very large
  - the sender can overload the receiver
  - if the sender does not “fill in the pipe”, the network utilization may be low
- Example:
  - Assume that
    - distance is 1500 km
    - transmission rate \( C = 100 \text{ Mbps} \)
  - The two-way propagation delay is
    - \( 2 \times \frac{1500}{300,000} \text{ s} = 0.01 \text{ s} \)
  - Thus, the product of RTT and C is
    - \( 0.01 \times 100,000,000 \text{ bits} = 1,000,000 \text{ bits} = 1 \text{ Mbit} \)

Delay x bandwidth product in high speed networks

- Assume RTT = 100 ms, we aim to transmit a file of size 1 MB
  - 1 Mbps network: time to transmit = \( 80 \times \text{RTT} \)
    - 80 pipes full of data (stream of data to send)
    - clearly, the network design problem would be to increase the bandwidth
  - 1 Gbps network: time to transmit = \( 0.08 \times \text{RTT} \)
    - only 8 % of the pipe is filled (the file has become a single “packet”)
    - now, the latency dominates the network design
- Thus, coping with the delay seems like the main design issue in future high speed networks
- Applications have other performance requirements than delay and bandwidth
  - Applications may have an upper bound on required bandwidth
  - Real time applications have requirements on delay variation (jitter) caused by queuing in the network routers
Performance of a statistical multiplexer (1)

- Internet is based on the use of statistical multiplexing
  - the output port of a router operates as a statistical multiplexer
- A statistical multiplexer can be modeled as a waiting system (= queue)
- Traffic consists of packets
  - each packet is transmitted with the full link speed $C$
  - packets arrive at a rate $\lambda$ and let $L$ denote the average packet length
  - packet service rate $\mu$ will be $\mu = C/L$
  - let $\rho = \lambda / \mu$, stability requirement: packet arrival rate $\lambda < \mu \Rightarrow \rho < 1$

![Diagram of performance of a statistical multiplexer](image)

Performance of a statistical multiplexer (2)

- Assume Poisson packet arrivals with exponentially distributed sizes
  - M/M/1 queuing system
- Load vs. mean queue length
  - mean queue length (and delay) rises sharply as load approaches 1
- Reasonable to design the network s.t. load < 0.9
  - link utilization always < 100%
  - congestion control needed
- The results are qualitatively the same regardless of the assumptions of the traffic
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Layered architectures

- A computer network must provide for a large number of hosts
  - cost effective, fair, robust and high performance connectivity, and
  - it must be easily able to accommodate new network technologies
- Network architecture
  - to guide the design and implementation of networks
  - abstractions used to hide complexities
- In networks, abstractions lead to layered designs
  - services offered at higher layers are implemented in terms of services provided by lower layers
  - often multiple abstractions (services) are provided to serve the varying requirements of above layers (multiplexing of upper layer protocols)
- Benefits of layering
  - decomposes the implementation problem into manageable components
  - modular design (adding new functionality may only affect one layer)
Protocols

- Each layer implemented by a **protocol**
  - protocols offer communication services to higher level objects
- A protocol offers two interfaces:
  - **Service interface**: offered to higher level objects on the same host
  - **Peer interface**: offered to peer protocol objects existing on other hosts

![Protocol diagram](image)

Encapsulation

- At the sender side, each lower layer protocol adds a header (L3H, L2H) thus encapsulating the upper layer packet
  - simple transformations (compression, encryption) of the packet possible
- At the receiver side, each layer removes the corresponding header and forwards the packet to the higher layer protocol entity

![Encapsulation diagram](image)
**OSI (Open Systems Interconnect) architecture**

- The “classic” 7-layer reference model (late 70’s)
  - protocols following the model defined in conjunction with ISO and ITU-T

  - E-mail, video conference, ordinary telephony
  - format of data
  - synchronization of different transport streams belonging to same session
  - end-to-end error correction, congestion control
  - routing of packets, addressing
  - collection of bits into frames, possibly error detection
  - transmission of raw bits (bit encoding, signal levels, etc.)

**Internet architecture**

- Internet architecture has only 4 layers
  - L4: range of application protocols (FTP, …)
  - L3: TCP (reliable byte transfer) and UDP (unreliable datagram delivery)
    - provide logical channels to applications
  - L2: IP protocol interconnects multiple networks into a single logical network
  - L1: wide variety of network protocols
  - “hour glass” shape
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Brief Internet history (1)

1961-1972: Early packet-switching principles

• 1961: Kleinrock - queuing theory shows effectiveness of packet-switching
• 1964: Baran - packet-switching in military nets
• 1967: ARPAnet conceived by Advanced Research Projects Agency
• 1969: first ARPAnet node operational
• 1972:
  – ARPAnet demonstrated publicly
  – NCP (Network Control Protocol) first host-host protocol
  – first e-mail program
  – ARPAnet has 15 nodes

Brief Internet history (2)

1972-1980: Internetworking, new and proprietary nets

- 1970: ALOHAnet satellite network in Hawaii
- 1973: Metcalfe’s Ph.D. thesis proposes Ethernet
- 1974: Cerf and Kahn - architecture for interconnecting networks
  - still determine largely the development of today’s Internet
- late 70’s: proprietary architectures: DECnet, SNA, XNA
- late 70’s: switching fixed length packets (ATM precursor)
- 1979: ARPAnet has 200 nodes

1972-1980: Internetworking, new and proprietary nets

1980-1990: new protocols, a proliferation of networks

- 1983: deployment of TCP/IP
- 1982: smtp e-mail protocol defined
- 1983: DNS defined for name-to-IP-address translation
- 1985: ftp protocol defined
- 1988: TCP congestion control
- new national networks: Csnet, BITnet, NSFnet, Minitel
- 100,000 hosts connected to confederation of networks
Brief Internet history (4)

1990’s: commercialization, the WWW

- Early 1990’s: ARPAnet decommissioned
- early 1990s: WWW
  - hypertext [Bush 1945, Nelson 1960’s]
  - HTML, http: Berners-Lee
  - 1994: Mosaic, later Netscape
  - late 1990’s: commercialization of the WWW

Late 1990’s:
- estimated 50 million computers on Internet
- estimated 100 million+ users
- backbone links running at 1 Gbps