HELSINKI UNIVERSITY OF TECHNOLOGY
Networking Laboratory

Introduction

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

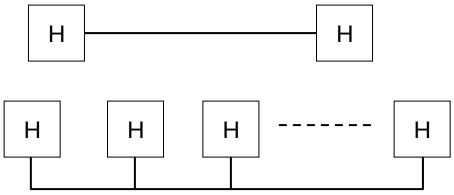
- Typically communications networks optimized for some service
  - telephone network
  - television/radio broadcast network
  - user terminals are special purpose devices
- Modern computer networks are more general:
  - terminals are general purpose PCs/workstations
  - networks able to carry essentially 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.
  - Attention: The term “link” may have different meanings
- Network nodes: PCs, servers, special purpose hardware
  - Internet terminology
    - hosts, end-systems: PCs and servers running network applications
    - routers (gatway): store and forward packets through the network
- (Physical) 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
- “Internet” Link: any single network
  - ISDN, Ethernet, WLAN, ATM, …

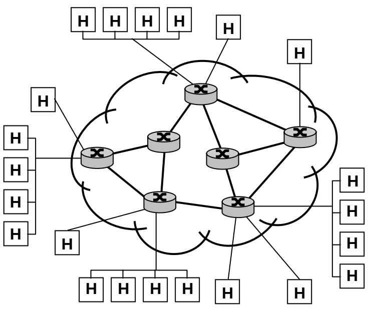


Building larger networks

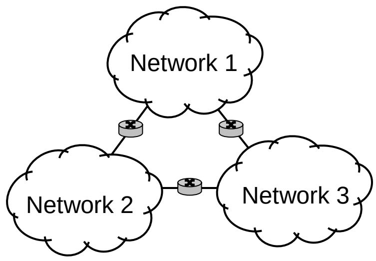
- Large networks cannot be built based on fully meshed point-to-point connectivity

  ⇒ use routers (switches) to interconnect hosts to each other

- Nodes connected together through switches to form connected networks

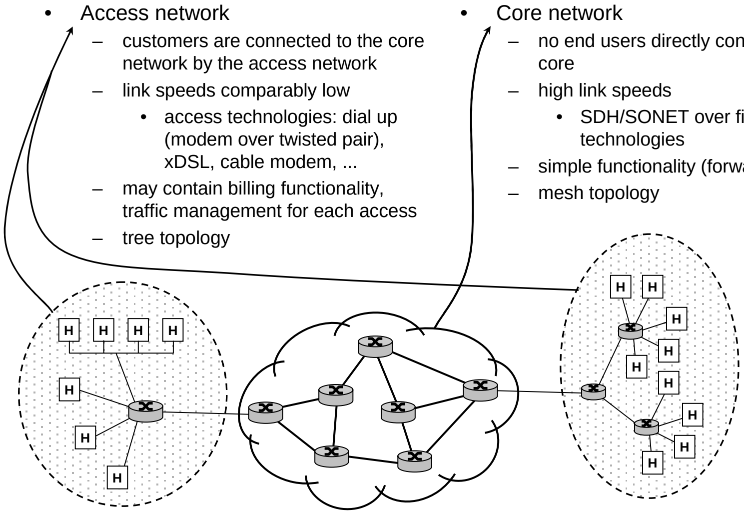


- Networks interconnected by routers (“gateways”) to form bigger entities



Network edge vs. core network in the Internet

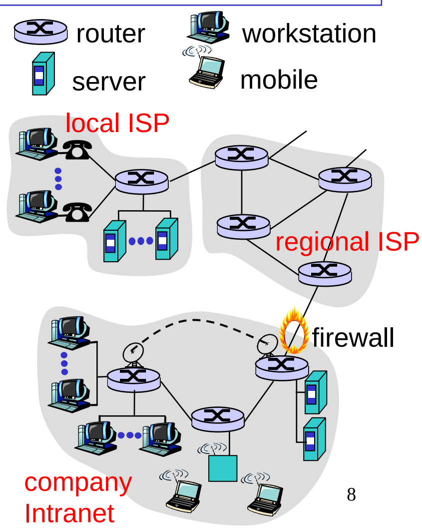
- Access network
  - customers are connected to the core network by the access network
  - link speeds comparably low
    - access technologies: dial up (modem over twisted pair), xDSL, cable modem, ...
  - may contain billing functionality, traffic management for each access
  - tree topology
- Core 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[1]

[1]Jim Kurose, lecture notes for the course MPSCI 591E Computer Networking, http://www-net.cs.umass.edu/cs591/

- 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

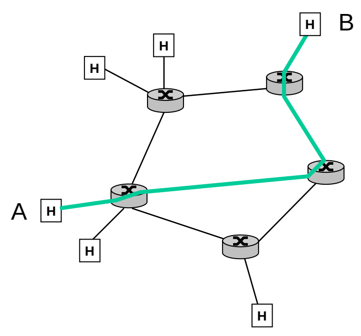
- Easy to build and manage a network supporting 100 users, but what if the number of users is 100 million ...
- A system allowing unlimited growth in size is said to **scale**.
  - Scalability a very desirable property for networking technologies
- Scalability of networks is often influenced very much 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
  - “Emulated” in other networks
    - [[ATM technology attempts to mimic circuit switching]]
      - fast packet switching with fixed length packets (cells): ATM
      - integration of different traffic types (voice, data, video) $\Rightarrow$ multiservice networks
- **Packet switching**
  - data networks
  - two possibilities
    - **connectionless**: e.g. Internet (IP), SS7 (MTP)
    - (**connection oriented**: e.g. X.25, Frame Relay, ATM

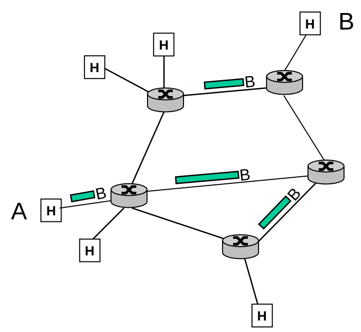
Circuit switching

- **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**
  - End-to-end bit pipe
- 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
  - **routing** 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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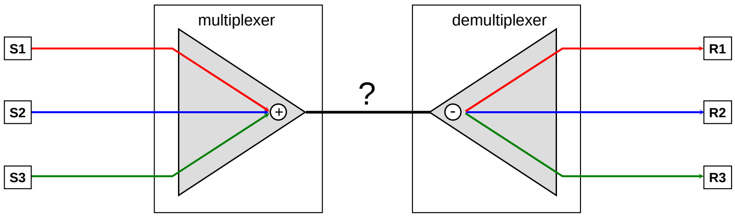
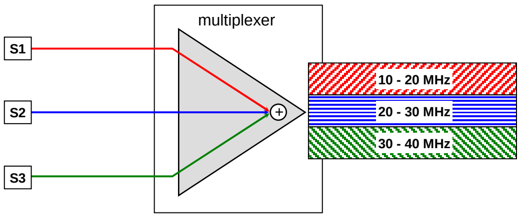
Multiplexing

- 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



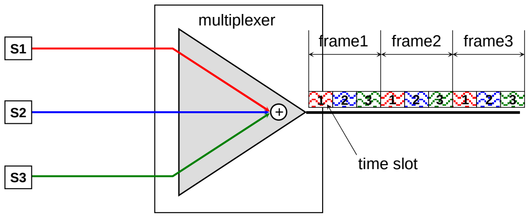
Frequency Division Multiplexing (FDM)

- 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



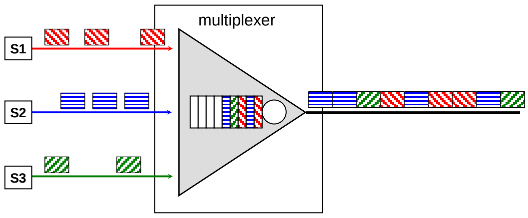
Time Division Multiplexing (TDM)

- 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 $C \le R_1 + \ldots + R_n$
- However, the loss probability can be decreased by enlarging the buffer
  - with an "infinite" buffer enough that C exceeds **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 **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) ⇒ QoS enabled networks

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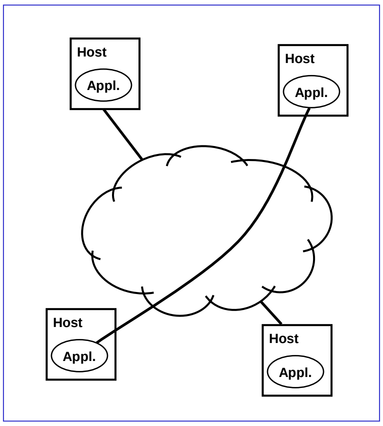
Communication needs of applications

- 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
  - but don't over-engineer
- Network (+ protocols) provide logical channels between application peers



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
    - 1980's: 2 Mbps, 1990's: 155 Mbps, 2000: 1 Gbps
  - 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

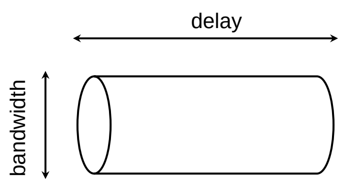
- Latency = delay
  - How long it takes a message to travel from one end of the nw to another
  - Measured in units of time, e.g., latency across US continent 24 ms
  - RTT (round trip time): time it takes a message to reach its destination and come back to the sender
- Components: propagation delay, transmission delay, queuing delay

$$\text{Latency} = \text{Propagation} + \text{Transmit} + \text{Queue}$$
$$\text{Propagation} = \text{Distance/SpeedOfLight}$$
$$\text{Transmit} = \text{Size/Bandwidth}$$

  - 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 feed-back 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 Mbps
  - The two-way propagation delay is
    - 2*1500/300,000 s = 0.01 s
  - Thus, the product of RTT and C is
    - 0.01*100,000,000 bits = 1,000,000 bits = 1 Mbit

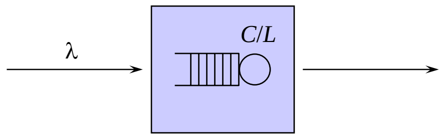


Delay x bandwidth product in high speed networks

- Assume RTT = 100 ms, we aim to transmit a file of size 1 MB
  - 1 Mbps network: time to transmit = 80 x 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 x 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
    - Which, actually, boils down to dealing with delay

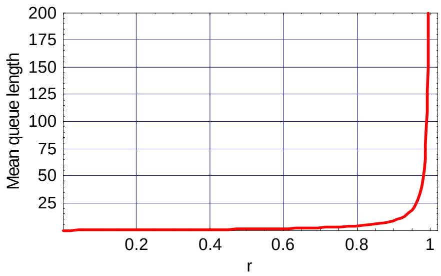
Performance of a statistical multiplexer (1)

- 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 λ and let L denote the average packet length
  - packet service rate μ will be $\mu = C/L$
  - let $\rho = \lambda / \mu$, stability requirement: packet arrival rate $\lambda < \mu \Rightarrow \rho < 1$



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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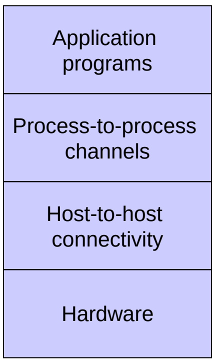
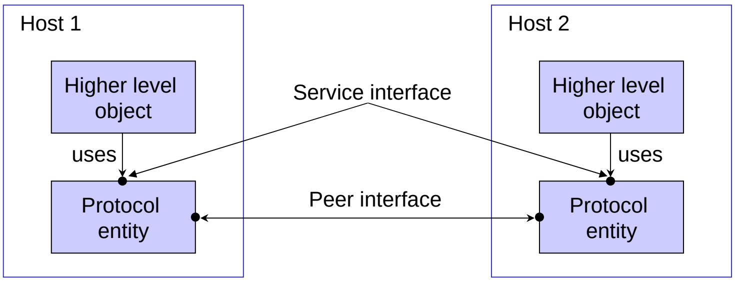
Layered architectures

- 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)



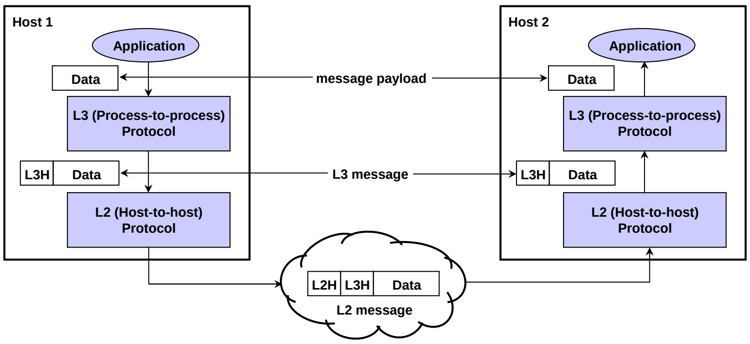
Protocols

- 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 entities existing on other hosts



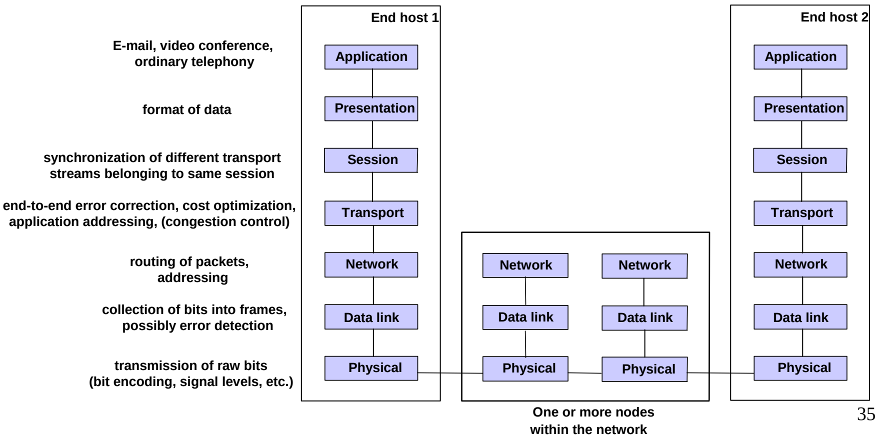
Encapsulation

- 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



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

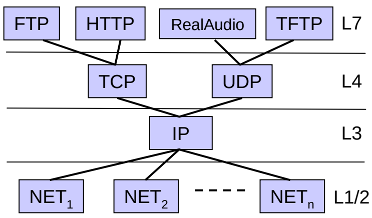


35

S-38.2188 - Computer Networks – 2005-2006, Period 3

Internet architecture

- Internet architecture has only 4 layers
  - L7: range of application protocols (FTP, …)
  - L4: TCP (reliable byte transfer, congestion control!) and UDP (unreliable datagram delivery) provide logical channels to applications
  - L3: IP protocol interconnects multiple networks into a single logical network
  - L1/2: wide variety of network protocols
  - “hour glass” shape



36

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Brief Internet history (1)

[1]Jim Kurose, lecture notes for the course MPSCI 591E Computer Networking, http://www-net.cs.umass.edu/cs591/

1961-1972: Early packet-switching principles

- 1961: Kleinrock - queuing theory shows effectiveness of packet-switching
- 1964: Baran - packet-switching in military nets;
  Donald Davies, Roger Scantlebury
- 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)

[1]Jim Kurose, lecture notes for the course MPSCI 591E Computer Networking, http://www-net.cs.umass.edu/cs591/

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
  - ACM Turing Award 2004
- late70's: proprietary architectures: DECnet, SNA, XNA
- late 70's: switching fixed length packets (ATM precursor)
- 1979: ARPAnet has 200 nodes

Brief Internet history (3)

[1]Jim Kurose, lecture notes for the course MPSCI 591E Computer Networking, http://www-net.cs.umass.edu/cs591/

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)

[1]Jim Kurose, lecture notes for the course MPSCI 591E Computer Networking, http://www-net.cs.umass.edu/cs591/

1990's: commercialization, the WWW

- Early 1990's: ARPAnet decommissioned
- 1991: NSF lifts restrictions on commercial use of NSFnet (decommissioned, 1995)
- 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