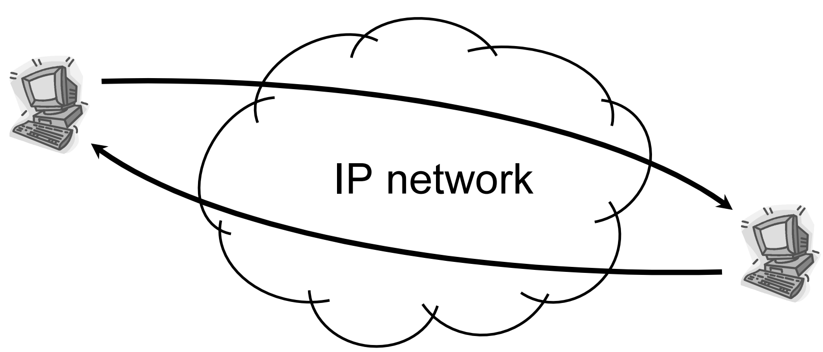
Internet transport protocols

Problem

- IP can be used to connect together heterogenous networks
  - IP network offers only best effort packet delivery (with no guarantees)
- Applications need end-to-end (process-to-process) communication
  - ”logical channels” through the network
  - ⇒ **transport layer** protocols on top of **IP layer**



Outline

- Overview of end-to-end protocols
- UDP
- TCP
  - Connection establishment / termination
  - Sliding window and flow control
  - Adaptive timeout + TCP extensions
- SCTP
- About Remote Procedure Calls (RPC)

Requirements of an end-to-end protocol (1)

- An end-to-end transport protocol is shaped
  - from above, by application requirements, and
  - from below, by limited capabilities of the network layer
- Common application requirements on transport protocols
  - guarantee message delivery
  - deliver messages in the same order they are sent
  - deliver at most one copy of each message
  - support arbitrarily large messages
  - support synchronization
  - allow the receiver to flow control the sender
  - support multiple application processes on each host
- Note(!), security is not in list above
  - implemented above transport layer

Requirements of an end-to-end protocol (2)

- Best-effort networks (Internet) have limited capabilities and can
  - drop messages
  - re-order messages
  - deliver duplicate copies of a given message
  - limit messages to some finite size
  - deliver messages after an arbitrarily long delay

- Challenge:
  - to develop protocols that use best-effort network, but can provide high (sufficient) level of service

Internet transport protocols

- Traditional IP transport protocols
  - UDP: simple multiplexing/demultiplexing
  - TCP: reliable byte stream
  - covered in this course

- New/emerging IP transport protocols
  - SCTP: reliable message transport protocol
    - briefly covered at the end of lecture
  - DCP: (proposed) transport protocol for streaming media
    - “TCP-friendly” congestion control but without retransmissions

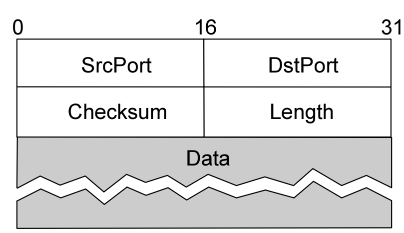
Outline

- Overview of end-to-end protocols
- UDP
- TCP
  - Connection establishment / termination
  - Sliding window and flow control
  - Adaptive timeout + TCP extensions
- SCTP
- About Remote Procedure Calls (RPC)

Simple demultiplexor (UDP)

- UDP = User Datagram Protocol
  - used in real time services, question-reply protocols
  - traditionally not much UDP traffic in Internet
  - increasing amount of real time services/applications ⇒ more UDP traffic
- Basic features
  - unreliable and unordered datagram service
  - only adds multiplexing to best-effort
  - greedy: no flow or congestion control
- Endpoints identified by ports
  - servers have well-known ports
  - ex. DNS uses port 53 etc.
  - port only 16 bits (hostwide)
  - server's full address (IP addr, port nr)
- Optional checksum
  - pseudo header + UDP header + data

UDP header format

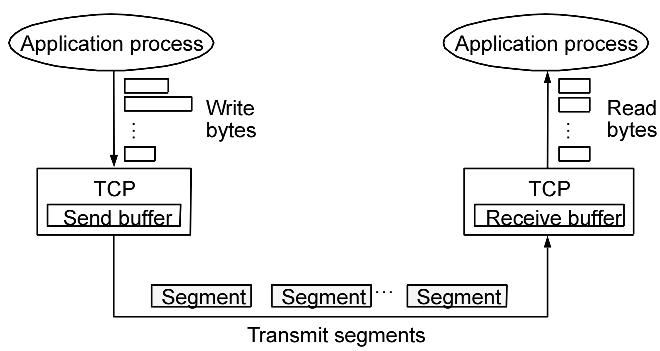


Outline

- Overview of end-to-end protocols
- UDP
- TCP
  - Connection establishment / termination
  - Sliding window and flow control
  - Adaptive timeout + TCP extensions
- SCTP
- About Remote Procedure Calls (RPC)

TCP overview

- Connection-oriented
- Byte-stream
  - application writes bytes
  - TCP sends segments
  - application reads bytes
- Full duplex (pair of byte streams)



- Traffic management
  - Flow control:
    - keep sender from overrunning receiver
  - Congestion control:
    - keep sender from overrunning network
- Protocols that use TCP
  - majority of Internet traffic still generated by TCP
  - Telnet, FTP, Simple Mail Transfer Protocol (SMTP), POP (reading of e-mails), IMAP (reading of e-mails), HTTP, X Window System (X11, decentralized window system)

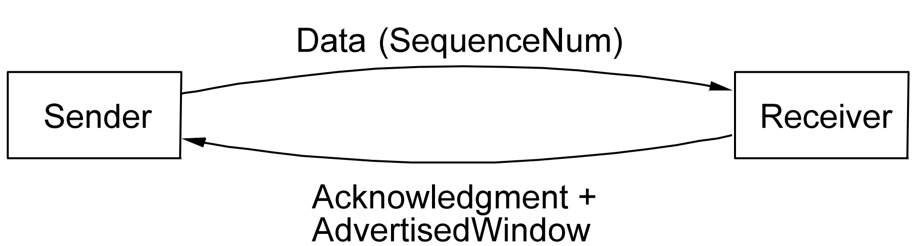
End-to-end issues

- TCP implements a sliding window protocol
  – similar to the one covered in lecture 2 for point-to-point links

- Issues that complicate design of an end-to-end sliding window protocol
  1. Connects many different hosts/applications
     - needs explicit connection establishment and termination
  2. Different RTTs
     - needs adaptive timeout mechanism (variations in RTTs)
  3. Long delays in network (packets reordered)
     - each TCP packet has **Maximum Segment Lifetime** (MSL, 120 s)
     - need to be prepared for arrival of very old packets
  4. Different/varying capacity at destination (delay x bandwidth)
     - accommodate very different node capacities
     - for a given node, amount of resources (buffer space) available changes with number of simulteneous TCP connections
  5. Different/varying network capacity (TCP has no link info)
     - need to be prepared for network congestion

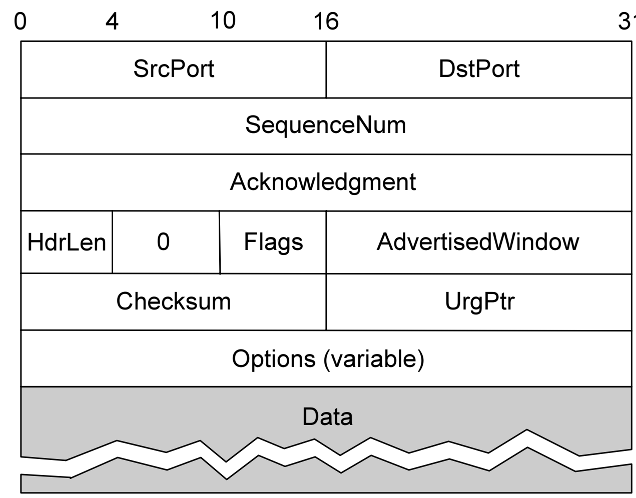
Sending data

- TCP byte oriented
  – sender application writes and receiver application reads bytes
- … but still TCP sends segments
  – MSS = Maximum Segment Size
- 3 mechanisms to trigger segment transmission
  – send segment once MSS bytes received
  – sender invoked operation (push)
  – timer that periodically fires
- Segment transmission:
  – data has sequence number, receiver acknowledges data and includes info on current buffer space (AdvertisedWindow)



Segment format

- Unique identification
  - <SrcPort, SrcIPAddr, DstPort, DstIPAddr>
- For sliding window algorithm
  - SequenceNum: number of first byte carried
  - Acknowledgment & AdvertisedWindow: info on flow on reverse direction
- 6-bit Flags: control info btw. TCP peers
  - Syn, Fin: establish & terminate TCP connection
  - Ack: if Acknowledgment is valid
  - Urg: urgent data
  - Push: sender invoked push operation
  - Reset: receiver is confused
- Checksum:
  - TCP header, TCP data, pseudoheader



Outline

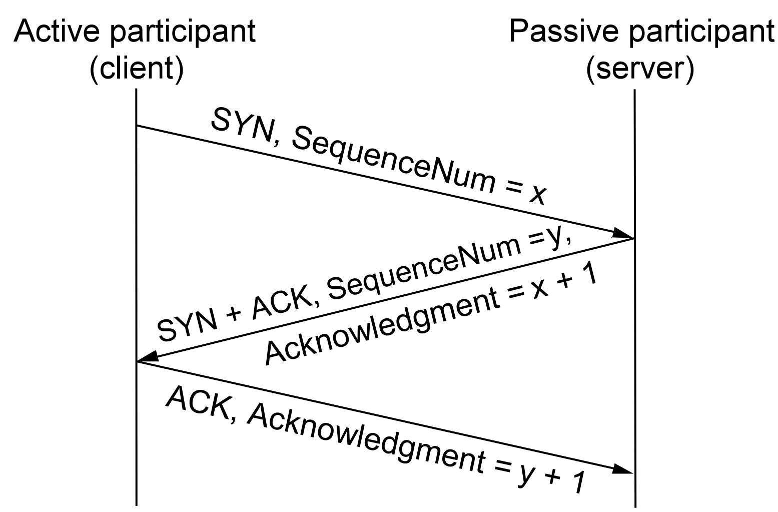
- Overview of end-to-end protocols
- UDP
- TCP
  - Connection establishment / termination
  - Sliding window and flow control
  - Adaptive timeout + TCP extensions
- SCTP
- About Remote Procedure Calls (RPC)

Connection establishment

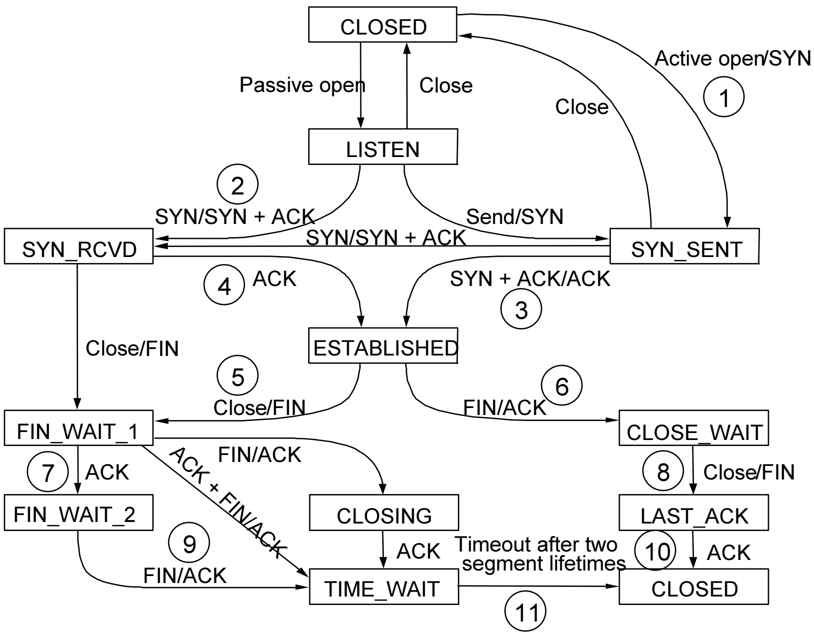
- Connection setup:
  - done before any actual data is transmitted
  - asymmetric, active open (caller/client) & passive open (callee/server)
- Connection teardown:
  - symmetric (each side closes independently)
- 3-way handshake
  - algorithm for establishing a connection and connection tear down
  - idea during establishment: to agree on starting sequence numbers
- Why not fixed starting numbers?
  - TCP specification: random initial sequence numbers
  - protection against two incarnations of the same connection
    - "incarnation" = same connection from same source <IP addr, port> pair

Timeline for 3-way handshake (establishment)

1. Client sends SYN and own initial seq. number x
2. Server responds with SYN+ACK, where own initial seq. number = y, and next expected byte (acknowledgment) = x+1
3. Client responds with ACK, where next expected byte (acknowledgement) = y+1



State transitions (not all, no timeouts)



- Format: event/action
- Connection establishment (above ”Established”)
  - asymmetric
- Connection tear down (below ”Established”)
  - symmetric (both sides tear down independently)

- Client initiated establish (C=client, S=server)
  - Server is in state LISTEN
  - (C,1), (S,2), (C,3), (S,4)
- Server initiated tear down
  - (S,5), (C,6), (S,7), (C,8), (S,9), (C,10), (S,11)
  - transition 11 takes 240 s!
  - server does not know if client received last ACK ⇒ must wait max possible time for a retransmitted FIN from client

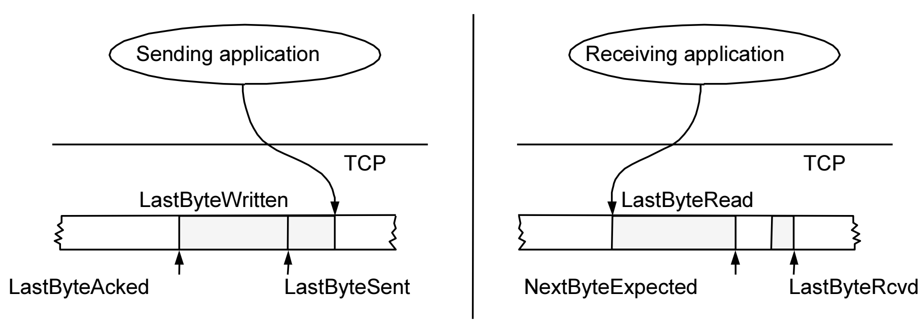
Outline

- Overview of end-to-end protocols
- UDP
- TCP
  - Connection establishment / termination
  - Sliding window and flow control
  - Adaptive timeout + TCP extensions
- SCTP
- About Remote Procedure Calls (RPC)

TCP sliding window

- Purposes:
  - guarantees reliable delivery of data
  - ensures that data is delivered in order
  - enforces flow control between sender and receiver
- Receiver advertises a window size to sender
  - idea: prevent sender from overrunning receiver's buffer
- Both sides have (finite) buffers and 3 pointers

Sliding window buffers and pointers



- Sending side
  - LastByteAcked ≤ LastByteSent
  - LastByteSent ≤ LastByteWritten
  - buffer bytes between LastByteAcked and LastByteWritten
- Receiving side
  - LastByteRead < NextByteExpected
  - NextByteExpected ≤ LastByteRcvd +1
  - buffer bytes between LastByteRead and LastByteRcvd

Flow control

- Variables
  - Send buffer size: MaxSendBuffer
  - Receive buffer size: MaxRcvBuffer
- Receiving side
  - LastByteRcvd - LastByteRead ≤ MaxRcvBuffer
  - AdvertisedWindow = MaxRcvBuffer - (LastByteRcvd - LastByteRead)
  - AdvertisedWindow = max amount of buffer space left
  - always send ACK in response to arriving data segment (even when receiving out of order segments ⇒ seq.number does not change)
- Sending side
  - LastByteSent - LastByteAcked ≤ AdvertisedWindow
  - EffectiveWindow = AdvertisedWindow - (LastByteSent - LastByteAcked)
    - amount of data that can be sent
  - LastByteWritten - LastByteAcked ≤ MaxSendBuffer
  - block sender if (LastByteWritten - LastByteAcked) + y > MaxSendBuffer
- Persist when AdvertisedWindow = 0
  - send periodically probe segments

Protection against wrap around

- 32-bit SequenceNum may wrap around
  - during connection's lifetime (not during one RTT)
  - during one connection, segments as old as MSL may arrive (120 s)

- fast network ⇒ sequence numbers consumed fast
  - 2^{32} ⇒ 4GB may be sent before wrap around

Bandwidth	Time Until Wrap Around
T1 (1.5 Mbps)	6.4 hours
Ethernet (10 Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
FDDI (100 Mbps)	6 minutes
STM-1 (155 Mbps)	4 minutes
STM-4 (622 Mbps)	55 seconds
STM-8 (1.2 Gbps)	28 seconds

- Protection against wrap around implemented by TCP options

Keeping the pipe full

- 16-bit AdvertisedWindow
  - maximum window size = 64 KB
- Delay x bandwidth worth of data can be transmitted, RTT of 100 ms assumed below

Bandwidth	Delay x Bandwidth Product
T1 (1.5 Mbps)	18KB
Ethernet (10 Mbps)	122KB
T3 (45 Mbps)	549KB
FDDI (100 Mbps)	1.2MB
STS-3 (155 Mbps)	1.8MB
STS-12 (622 Mbps)	7.4MB
STS-24 (1.2 Gbps)	14.8MB

- Increased maximum window size through TCP options

Outline

- Overview of end-to-end protocols
- UDP
- TCP
  - Connection establishment / termination
  - Sliding window and flow control
  - Adaptive timeout + TCP extensions
- SCTP
- About Remote Procedure Calls (RPC)

Adaptive retransmission

- Reliable delivery of data ⇒ retransmission needed
- ACK not received ⇒ timeout
- Timeout depends on RTT
  - in Internet RTT is a random variable (depends on random queuing delays and also routes may change)
- Choosing the right timeout (i.e., RTT):
  - RTT too short: too many retransmissions
  - RTT too long: unnecessary waiting
  - many algorithms to determine RTT

Adaptive retransmission (original algorithm)

- Measure SampleRTT for each segment/ACK pair
  - Note! for retransmissions SampleRTT includes also retransmission time
- Compute weighted average of RTT

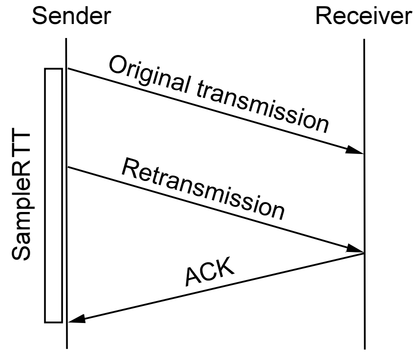
$$EstRTT = a \times EstRTT + (1-a) \times SampleRTT$$

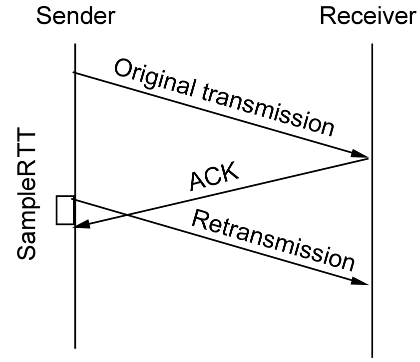
  - where a between 0.8 and 0.9

- Set timeout based on EstRTT

$$TimeOut = 2 \times EstRTT$$

Karn/Partridge algorithm





- Problem:
  - impossible to know if ACK came from original packet or retransmission (see above)
  - timeout implies congestion, retransmissions increase congestion ⇒ reaction to timeout should be conservative
- Solution:
  - don't sample RTT when retransmitting!
  - double timeout after each retransmission (⇒ exponential backoff)

Jacobson/ Karels algorithm

- Problem with earlier algorithms:
  - RTT is a random variable, moreover it has a variance
  - If variance is small, EstimatedRTT more reliable
  - If variance is large, timeout should not be "too much" based on EstimatedRTT
- New Calculations for EstRTT (i.e., average RTT)

Diff = SampleRTT - EstRTT
EstRTT = EstRTT + (d x Diff)
Dev = Dev + d x (|Diff| - Dev)

  - where d is a factor between 0 and 1
- Idea: consider variance when setting timeout value

TimeOut = m x EstRTT + f x Dev

  - where m = 1 and f = 4
- Notes
  - algorithm only as good as granularity of clock (500ms on Unix)
  - accurate timeout mechanism important to congestion control (later)

TCP extensions

- Implemented as header options
  - store timestamp (32 bit) in outgoing segments (to improve RTT measurement accuracy by measuring it from the packet)
  - extend sequence space with 32-bit timestamp (protection against sequence number wrap around)
  - shift (scale) advertised window (larger window sizes)
  - use of Options negotiated during TCP connection establishment
- Nagle algorithm (RFC896)
  - built in most TCP implementations
  - sender holds a partial segment's worth of data (even if PUSHed) until either a full segment accumulates or the most recent outstanding ACK arrives (= all data has been received correctly by receiver)
    - small packet problem: Telnet can generate 1B + 40B packets (4000% overhead)
- Delayed ACKs (RFC813)
  - to limit generation of small ACK packets
  - receiver waits until there is data to transmit on reverse path and "piggybacks" ACKs on TCP data segment
    - timer guards that receiver does not wait "too long"
  - ACKs must still be generated at least after receiving 2*MSS bytes (every 2 segments)

Outline

- Overview of end-to-end protocols
- UDP
- TCP
  - Connection establishment / termination
  - Sliding window and flow control
  - Adaptive timeout + TCP extensions
- SCTP
- About Remote Procedure Calls (RPC)

Stream Control Transmission Protocol (SCTP)

- Suitable transport protocol for signaling traffic (SS7) over IP
  - may be used for traditional Internet services such as those based on HTTP and SIP
  - sometimes called the “Next Generation TCP”
  - Internet standard RFC2960
- Service model
  - unicast transport protocol
  - provides r**eliable, message-oriented** data delivery
    - SCTP sends “complete” messages whereas TCP is byte oriented and does not preserve any structure in the byte stream
    - reliable = lost/corrupted messages are retransmitted
- Other features
  - functionality is TCP-like with modifications due to message oriented principle
    - TCP-like sliding window, flow control and congestion control
    - uses selective acknowledgements (SACKs) to report out of sequence data
  - multistreaming: data stream can be partitioned into multiple streams, each controlled independently
  - multihoming: for redundancy, an SCTP endpoint can be associated with multiple IP addresses

Outline

- Overview of end-to-end protocols
- UDP
- TCP
  - Connection establishment / termination
  - Sliding window and flow control
  - Adaptive timeout + TCP extensions
- SCTP
- About Remote Procedure Calls (RPC)

About remote procedure calls

- Request/reply paradigm, but UDP message going in one direction and UDP message back is not enough
  - messages can be lost or reordered
  - use of TCP overkill
- Protocol family Remote Procedure Call (RPC)
  - RPC more complex than local procedure call:
    - network between calling process and called process is complex
    - computer at each end may have different architectures and data representation
  - semantics:
    - at-most-once = for every request message, only one copy of message is delivered to the server
    - zero-or-more = remote procedure call invoked zero or more times
  - examples: SunRPC, DCE RPC
- Internet society view
  - RPC carried on top of UDP ⇒ RPC is not a transport protocol according to Internet architecture

Some remarks

- Getting a transport protocol right is hard, changing circumstances make it harder
- In Internet, transport protocols have a crucial role
  - IP network is unreliable
  - IP design principle has usually been that end systems implement all intelligent functions ⇒ transport layer is (first) end-to-end layer
- Protocols can and do change
  - for example TCP timers and congestion control
  - networks do change
  - How to change a protocol in the Internet?
    - Must always consider interoperability…
  - How to supply the level of service required by applications (that also changes all the time)
    - New requirements by real time applications ⇒ DCP
    - Other emerging requirements…