Revision notes - CS2105

Ma Hongqiang

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Contents

1 Introduction

1.1 What is the Internet?

Definition 1.1 (Internet)**.**
The *Internet* is a network of connected computing devices known as **hosts** or **end systems**. Hosts run **network applications**.

Examples of hosts include PC, laptops, servers and smartphones.
Examples of network applications include Web, Pokemon Go, Skype.

1.2 Network Edge

Hosts access the Internet through access network. Hosts connect to the access network over different physical media.

- **Guided media**: signals propagate in solid media, e.g. fiber
- **Unguided media**: signals propagate freely, e.g., Wi-Fi, cellular

1.3 Network Core

Network core consists of a mesh of interconnected routers. The data transmit through net via either **circuit switching** or **packet switching**.

Definition 1.2 (Circuit Switching)**.**
In **circuit switching**, end-end resources allocated to and reserved for "call" between source and destination.

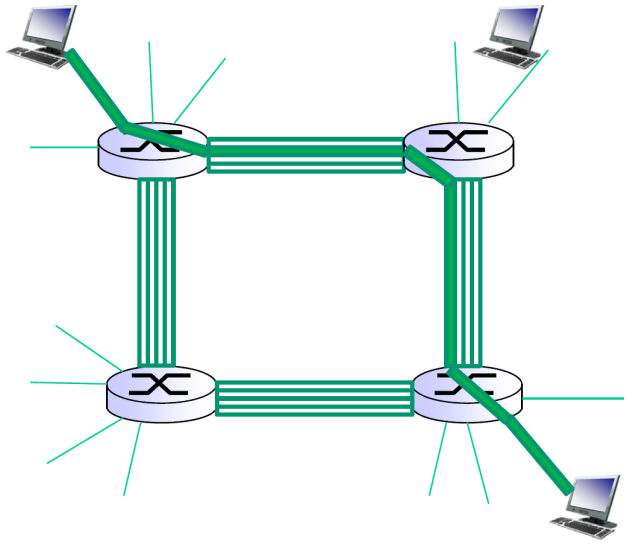


Figure 1: In above diagram, each link has four circuits. A call gets 2nd circuit in top link and 1st circuit in right link.

Some properties of **circuit switching** include

- call setup required
- circuit-like (guaranteed) performance
- circuit segment idle if not used by call (no sharing)
- *commonly used in traditional telephone networks*
- divide link bandwidth into "pieces"
  - frequency division
  - time division

Definition 1.3 (Packet Switching)**.**
In **packet switching**, host sending function by

- breaking application message into smaller chunks known as **packets**, of length L bits; and
- transmitting packets onto the link at **trasmission rate**[1] R bits/sec.

Definition 1.4 (Packet Trasmission Delay)**.**
Packet transmission delay, d_{trans} is defined as time nedded to transmit L-bit packet onto link.

$$d_{\text{trans}} = \frac{L}{R}$$

Packet switching adopts **store-and-forward** approach when a *router* is encountered.

- Packets are passed from one *router* to the next, across links on path from source to destination.
- **Store and forward**: entire packet must arrive at a router before it can be transmitted on the next link.

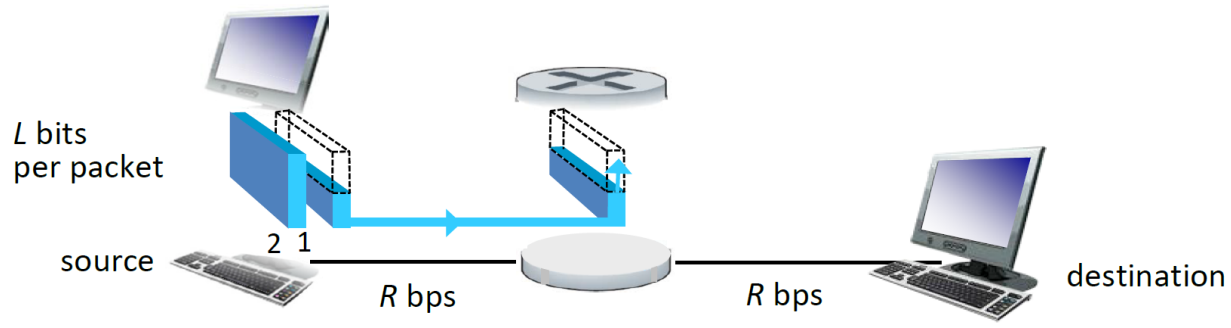


Figure 2: End to end $d_{\text{trans}} = 2 \times \frac{L}{R}$, assuming no other delays

[1]link transmission rate is also known as **link capacity** or **link bandwidth**

Definition 1.5 (Routing and Addressing)**.**
Routers determines source-destination route taken by packets by routing algorithms.
Addressing is achieved by each packet carrying source and destinaton information.

Some properties of packet switching include

- The Internet **is** a packet switching network.
- User A, B, ...'s packets *share* network resources
- Resources are used on demand.
- Excessive congestion is possible.[2]

Definition 1.6 (Internet Structure: Network of Networks)**.**

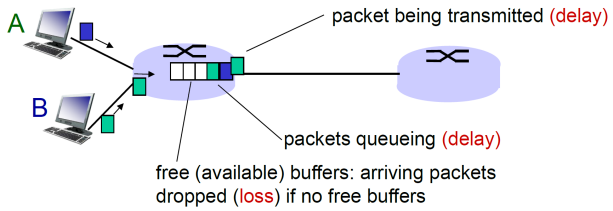
- Hosts connect to Internet via access **ISPs**(Internet Service Providers)
- Access ISPs in turn must be interconnected.
- Resulting network of networks is very complex.
  Evolution was driven by *economics* and *national policies*.
- Therefore, the Internet is a "network-of-networks", organised into autonomous systems(AS), each is owned by an organisation.

1.4 Delay, Loss and Throughput in Networks

Recall, the algorithm of sneding a packet in a packet switching network is

1. Sender transmit a packet onto the link as a sequence of bits.
2. Bits are propagated to the next node(e.g. a router) on the link.
3. Router stores, processes and forwards the packet to the next link.
4. Steps 2 and 3 repeat till the packet arrives at the receiver.

Delay and Loss could occur as packets need to queue in router buffers, waiting for turn to be send out one by one.



[2]These properties suggest bandwidth division into "pieces", dedicated allocation and resource reservation are not available.

- Queue(aka **buffer**) of a router has finite capacity.
- Packet arriving to full queue will be dropped(aka lost).
- Lost packet may be retransmitted by previous node, by source host, or not at all.

Definition 1.7 (Four Sources of End-to-end Packet Delay)**.**
End-to-end packet delay is the time taken for a packet to travel from source to destination.
It consists of:

- transmission delay
- propagation delay
- processing delay
- queueing delay

d_{proc}: nodal processing delay	d_{queue}: queueing delay	d_{trans}: transmission delay	d_{prop}: propagation delay
• check bit errors • determine output link • typically $<$ msec	• time waiting in the queue for transmission • dependes on congestion level of router	• L: packet length • R: link bandwidth • $d_{\text{trans}} = \frac{L}{R}$	• d: length of physical link • s: propagation speed in medium ($\sim 2 \times 10^8$m/sec) • $d_{\text{prop}} = \frac{d}{s}$

Definition 1.8 (Throughput)**.**
Throughput is defined as how many bits can be transmitted per unit time.

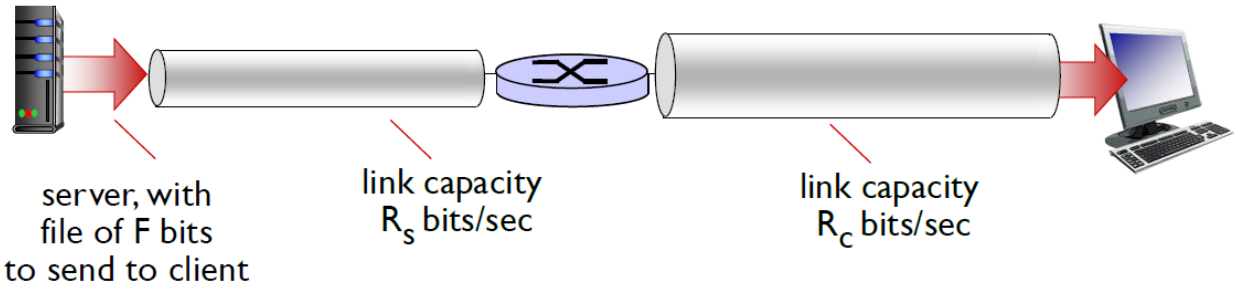


Figure 3: The throughput is $\frac{F}{\left(\frac{F}{R_s}+\frac{F}{R_c}\right)} = \frac{R_s R_c}{R_s+R_c}$

The difference between throughput and link capacity is that

- Throughput is measured for end-to-end communication.
- Link capacity(bandwidth) is meant for a specific link.

1.5 Protocol Layers and Service Models

The Internet supports various network applications. Network applications exchange messages and communicate among peers according to **protocols**.

Definition 1.9 (Protocol)**.**
Protocols define *format* and *order* of messages exchanged and the *actions* taken after messages are sent or received.

Protocols in the Internet are logically organised into "layers" according to their purposes.

- Each layer provides a service.
- Simple interfaces between layers
- Hide details from each other

Definition 1.10 (Internet Protocol Stack)**.**
The Internet consists of five layers.

- **Application**: supporting network applications, e.g. FTP, SMTP, HTTP
- **Transport**: process-to-process data transfer, e.g. TCP, UDP
- **Network**: routing of datagrams from source to destination, e.g. IP
- **Link**: data transfer between neighbouring network elements,e.g. Ethernet, 802.11
- **Physical**: bits "on the wire"

2 Application Layer

2.1 Principle of Network Applications

Application architectures refer to possible structures of network applications, which include:

- Client-server
- Peer-to-peer(P2P)
- Hybrid of client-server and P2P

Definition 2.1 (Client-Server Architecture)**.**
In **client-server** architecture,
Server:

- Waits for incoming requests
- Provides requested service to client

Client:

- Initiates contact with server("speak first")
- Typically requests service from server
- For Web, client is usually implemented in browser

Definition 2.2 (P2P Architecture)**.**
In **P2P** Architecture,

- *No* always-on server
- Arbitrary end systems directly communicate.
- Peers request service from other peers, provide service in return to other peers.

P2P Architecture is highly scalable but difficult to manage.

An example of Hybrid of Client-Server and P2P architecture is the instant messaging, where

- Chatting between two users is **P2P**
- Presence detection/location is **centralised**

Usually for applications, some considerations of *transport* service include

- Data integrity
- Throughput
- Timing

- Security

Definition 2.3 (Two Internet Transport Protocols)**.**
Two Internet Transport Protocols are `TCP` and `UDP`. The main difference is that `TCP` is reliable while `UDP` is not. In detail,
`TCP` service offers:

- **reliable transport** between sending and receiving pocess
- **flow control**: sender won't overwhelm receiver
- **congestion control**: throttle sender when network is overloaded

However, `TCP` does *not* provide: timing, minimum throughput guarantee, security.
In contrast, `UDP` service provide **unreliable** data transfer between sending and receiving process. It does *not* provide: reliability, flow control, congestion control, timing, throughput guarantee or security.

For **application layer protocol** which is over transport layer protocol, instead it defines

- Types of messages exchanged: e.g., request, response
- Message syntax: what field in messages and how fields are delineated
- Message semantics: meaning of information in fields
- Rules: for when and how applications send and respond to messages
- Open protocols: defined in RFCs; allow for interoperability; e.g., `HTTP`, `SMTP`
- Proprietary protocols, e.g., Skype

2.2 Web and `HTTP`

Definition 2.4 (Web Page)**.**
A **Web page** typically consists of:

- base `HTML` file, and
- several referenced *objects*
  An object can be HTML file, JPEG image etc.
  Each object is addressable by a **URL**,e.g.,

$$\underbrace{\texttt{www.comp.nus.edu.sg}}_{\text{host name}}\underbrace{\texttt{/ cs2105/img/doge.jpg}}_{\text{path name}}$$

Definition 2.5 (`HTTP`)**.**
`HTTP` refers to **Hypertext transfer protocol**.
`HTTP`

- is Web's application layer protocol
- usees client/server model
  - **client**: usually is browser that requests, receives and displays Web objects
  - **server**: Web server sends objects in response to requests

Note that `HTTP` uses `TCP` as transport service.

2.2.1 Non-persistent `HTTP`

In **non-persistent** `HTTP/1.0`,

1. Client initiates `TCP` connection to server.
2. Server accepts `TCP` connection request from client.
3. `HTTP` messages are exchanged between browser(`HTTP` client) and Web server(`HTTP` server) over `TCP` connection.
4. `TCP` connection closed.
5. `HTTP` client receives response message containing HTML file, displays HTML. Parsing HTML file, client notices referenced objects.
6. Step 1∼5 repeated for the referenced objects.

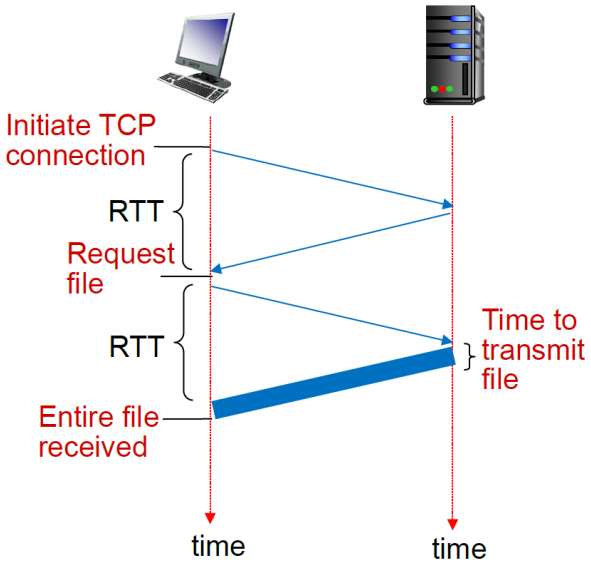
Definition 2.6 (Round Trip Time(RTT))**.**
Round trip time(RTT) refers to the time for a packet to travel from client to server and go back.

Clearly, the `HTTP` **response time** for one object transfer equals the sum of:

- one RTT to establish `TCP` connection
- one RTT for `HTTP` request and the first few bytes of `HTTP` response to return
- file transimission time

As a result, the **non-persistent** `HTTP` response time =

$$2 \times \text{RTT} + \text{file trasmission time}$$



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2.2.2 Persistent HTTP

The problems of non-persistent HTTP include:

- requires 2 RTTs per object
- OS overhead for *each* TCP connection
- browsers often open parallel TCP connections to fetch referenced objects

Persistent HTTP/1.1 solves these issues by:

- server leaves connection open after sending response
- subsequent HTTP messages between same client/server sent over the same TCP connection
- client sends requests as soon as it encounters a referenced object(*persistent with pipelining*)
- as little as one RTT for all the referenced objects

Example HTTP request message usually entails both **request** and **response**.
A HTTP **request** message looks like:

```
GET /~cs2105/demo.html HTTP/1.1
Host: www.comp.nus.edu.sg
User-Agent: Mozilla/5.0
Connection: close
\r\n
```

The 1st line is the request line. Subsequent 3 lines are header lines. Last extra blank line indicates end of header lines.
A HTTP **response** message looks like:

```
HTTP/1.1 200 OK
Date: Thu, 15 Jan 2015 13:02:41 GMT
Server: Apache/2.4.6 (Unix)
Content-Type: text/html
\r\n
data data data data ...
```

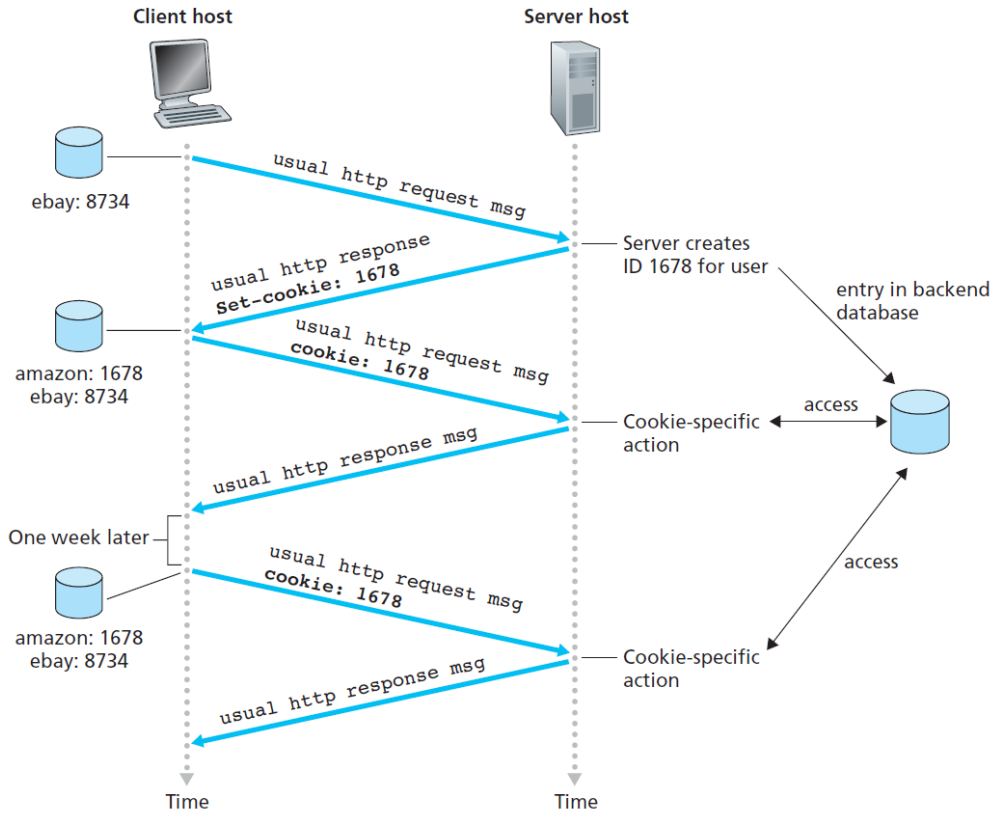
The 1st line is the protocol status line. Subsequent 3 lines are header lines. After the extra blank line is the data of the requested HTML file.

2.2.3 Cookies

HTTP is desisned to be **stateless**, in a sense that server maintains *no* information about past client requests.
However, sometimes it is good to maintain states(history) at server/client over multiple transactions.

Definition 2.7 (Cookies)**.**
Cookies allow sites to keep track of users.
Cookie technology has four components:

- A cookie header line in the HTTP response message
- A cookie header line in the HTTP request message
- A cookie file kept on the user's end system and managed by the user's browser
- A back-end database at the Web site



2.2.4 Conditional GET

The goal of conditional GET is to **not** send object if client cache has **up-to-date** cached version.

In the client cache, date of cached copy in `HTTP` request is specified by `If-modified-since: <date>`.
In the server response, no object will be contained in the response if cached copy is up-to-date, and return `HTTP/1.0 304 Not Modified`.

2.3 Domain Name System

There are two ways to identify a host:

1. **Hostname**, e.g., www.comp.nus.edu.sg
2. `IP` **address**, e.g., 137.132.80.57

Definition 2.8 (DNS)**.**
Domain Name System(DNS) translates between hostname and `IP` address.

A client must carry out a DNS query to determine the `IP` address corresponding to the server name (e.g., www.comp.nus.edu.sg) prior to the connection.

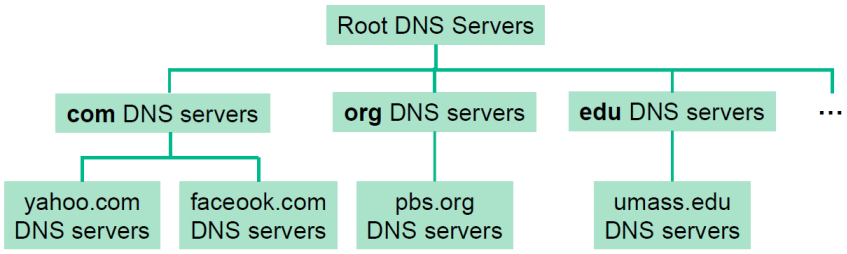
Definition 2.9 (DNS Resource Records)**.**
DNS resource records are mappings between host names and `IP` addresses and others. The format of resource records is

```
(name, value, type, TTL)
```

There are 4 `type` if resource records.

1. `type` = A: `(hostname, IP address, A, TTL)`
2. `type` = CNAME: `(caconical name, real name, CNAME, TTL)`
3. `type` = NS: `(domain name, hostname of authoritative name server for this domain, NS, TTL)`
4. `type` = MX: `(domain name, mail server, MX, TTL)`

DNS stores resource records in distributed databases implemented in hierarchy of many name servers. For example, if a client wants `IP` address for www.facebook.com:



- Client queries root server to find .com DNS server

- Client queries .com DNS server to get facebook.com DNS server
- Client queries facebook.com DNS server to get `IP` address for www.facebook.com

Here, **root servers** answer requests for records in the root zone by returning a list of the authoritative name servers for the appropriate top-level domain(TLD).
Top level domain(TLD) servers us responsible for com,org,net.edu and all top-level country domains and returns a list of authoritative servers.
Authoritative servers are organisation's own DNS servers, providing authoritative hostname to `IP` mappings for organisation's named hosts(e.g., web, mail). They can be maintained by organisation or service provider.
Local DNS server is also called *default name server*. It does not strictly belong to the hierarchy; each ISP has one local DNS server. When host makes a DNS query, query is sent to its local DNS server and

- Retrieve name-to-address translation from local cache;
- if answer is not found *locally*, acts as proxy and forwards query into hierarchy if answer is not found locally
- Once it learns mapping, it caches mapping. Cache entries will expire after Time to Live(TTL) countdown.

Note that DNS runs over `UDP` for speed. `UDP` is faster since compared to `TCP`, it avoids one RTT induced by `TCP` handshake.

3 Socket Programming

3.1 Processes

Definition 3.1 (Process)**.**
Applications runs in hosts as **processes**.

- Within the *same* host, two processes communicate using **inter-process communication** (defined by OS).
- Processes in *different* hosts communicate by exchanging **messages** (according to protocols).

In Client/Server model, **server process** waits to be contacted; **client process** initiates the communication.

Theorem 3.1 (Addressing processes)**.**
A process is identified by (**IP address**, **port number**), where **port number** is a 16-bit integer.

3.2 Sockets

Definition 3.2 (Socket)**.**
Socket is the software interface between app processes and transport layer protocols.

- Process sends/receives messages to/from its socket.
- Programming-wise: a set of **API** calls

Refer to Lecture Notes for programming details.

4 Transport Layer Protocol

4.1 Transport-layer Services

Internet transport layer protocols mainly are:

- `TCP` : connection-oriented and reliable
- `UDP` : connection-less and unreliable

Transport layer protocols run in hosts.

- Sender side:
  - breaks app message into **segments**
  - passes them to network layer
- Receiver side:
  - reassembles segments into message
  - passes it to application layer

4.2 `UDP` : User Datagram Protocol

`UDP` adds very little service on top of `IP`:

- Connectionless multiplexing/de-multiplexing
- Checksum

`UDP` transmission is **unreliable**. Therefore, to achieve reliable transmission over `UDP` , application layer protocol should implement error detection and recovery mechanism.
`UDP` has advantage over `TCP` in

- No connection establishment (No handshake delay)
- Simple: no connection state at sender and receiver
- Small header size
- No congestion control

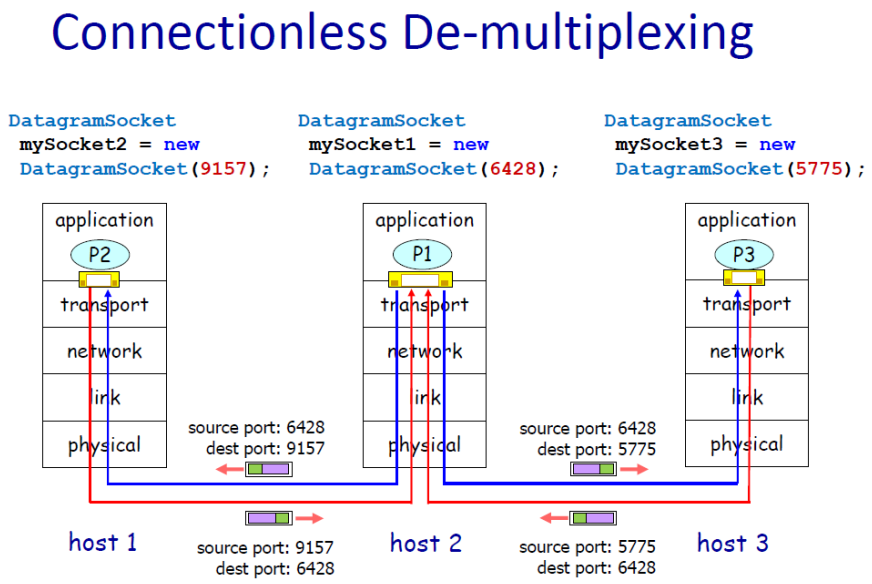
4.2.1 Connectionless De-multiplexing

Connectionless de-multiplexing is achieved by information on port numbers.

- `UDP` sender:
  - creates a socket with **local port number**.

- When creating a datagram to send to UDP socket, sender must specify **destination IP address** and **port number**.

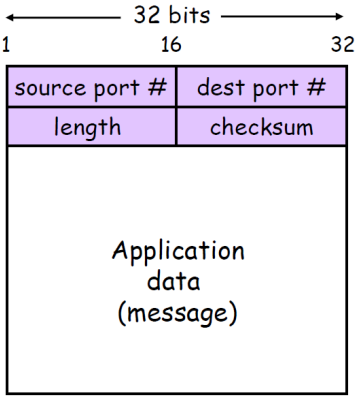
- When UDP receiver receives a UDP segment:
  - Checks **destination port number** in segment.
  - Directs UDP segment to the socket with that port number.
  - IP datagram from different sources with the **same destination port number** will be directed to the same UDP socket at destination.



4.2.2 UDP Header

Definition 4.1 (UDP Header)**.**
UDP **header** consists of 64 bits:

- bit 1 $\sim$ 16: source port number (0 $\sim$ 65535)
- bit 17 $\sim$ 32: destination port number (0 $\sim$ 65535)
- bit 33 $\sim$ 48: length of UDP segment in **bytes**, including header
- bit 49 $\sim$ 64: checksum



4.2.3 `UDP` Checksum

`UDP` Checksum is used to detect errors in transmitted segment.

- Sender
  1. Compute checksum value
  2. Put checksum value into `UDP` checksum field
- Receiver
  1. Computer checksum of received segment
  2. Check if computed checksum equals checksum field value
     - NO – error detected
     - YES – no error with respect to the checksum check

Theorem 4.1 (Checksum Computation)**.**
CHecksum is computed accordingly

1. Treat `UDP` segment as a sequence of **16-bit** integers.
2. Apply binary addition on every 16-bit integer. If the last integer is of length less than 16 bit, add zeroes to the right of last integer until 16 bit is reached.
3. Carry (if any) from the most significant bit will be added to the result
4. Compute 1's complement to get `UDP` checksum.

4.3 Principle of Reliable Data Transfer 3.0

In this section, we assume the underlying channel

- *may* flip bits in packets
- *may* lose packets
- *may* incur arbitrarily long packet delay
- *will NOT* reorder packets

Reliable data transfer involves a sender and a receiver.

- The sender sends out **segments**;
- the receiver replies **ACK** message.

A **sequence ID** is assigned by sender to each packet. **Acknowledgements** contains the sequence ID of the last successfully received packet.

4.3.1 Behavior of Sender

In reliable data transfer, sender maintains a **timer**. The timer will start when a segment is sent and stops by receiving ACK with sequence ID of this segment. Until then, another packet with a different sequence ID will be sent, with the whole process again. Otherwise, the timer will timeout, in which case this same packet is sent again.
All incoming packets(ACK) will be ignored if the sender waits to send, in which case there is no timer running.

4.3.2 Behaviour of the receiver

The data transfer is initiated by sender sending the first segment. On receiving a packet, *corrupted* copy will trigger an ACK of last correctly received copy. Receiver will also check the sequence ID for duplicate segment for duplicated copy. However, ACK of that copy's sequence ID will be sent regardless of whether it is a duplicate or not; the only difference is that duplicated copy will be discarded. If it is not corrupted and not a duplicate packet, it will be delivered to application layer.

4.3.3 Bit Error

Bit error occurs when some bits of the segment are flipped.
Bit error can occur in

- **segment**, in which case
    1. receiver detects bit error using checksum
    2. receiver ignores packet
    3. receives send ACK with sequence ID of last successfully transmitted segment
- ACK message, in which case
    1. sender will ignore the corrupted ACK

4.3.4 Packet Loss

Packet loss occurs in

- Segment, in which case receiver will do nothing;
- ACK message, in which case sender will do nothing.

4.3.5 Arbitrarily Long Packet Delay

Arbitrarily long packet delay can occur in

- Segment, in which case receiver reply ACK accordingly once received;
- ACK message, in which case sender will only act according to timer.

4.3.6 Finite State Machine Diagram of Reliable Transfer Protocol

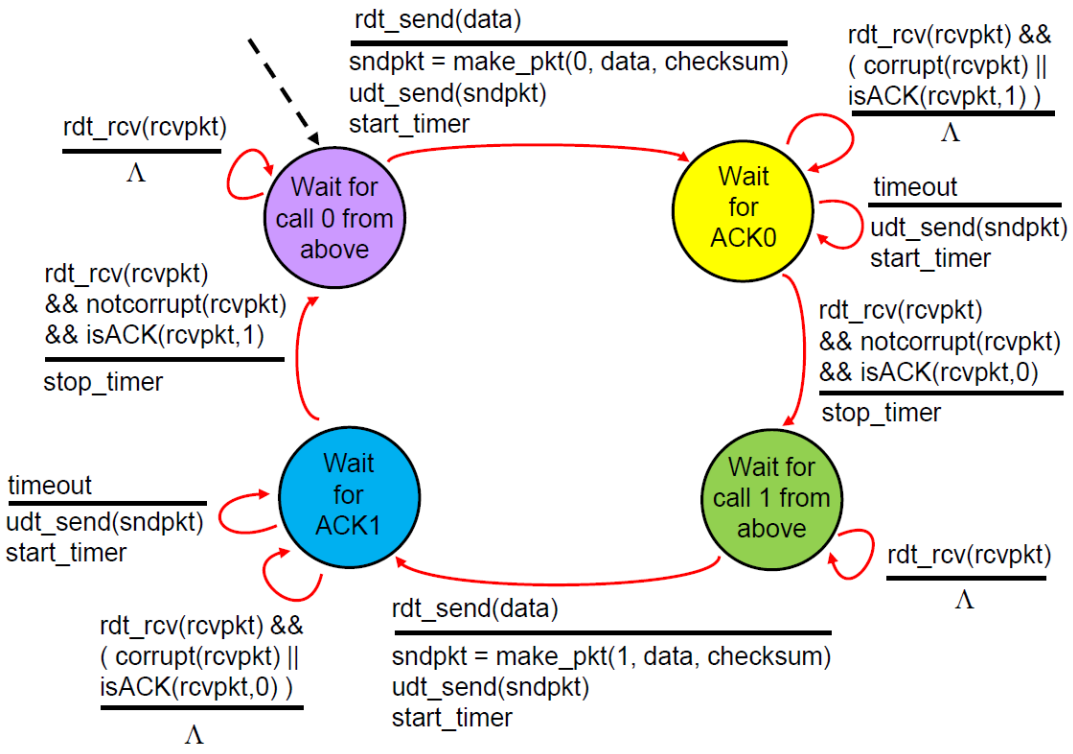


Figure 4: Reliable Data Transfer Sender Finite State Machine

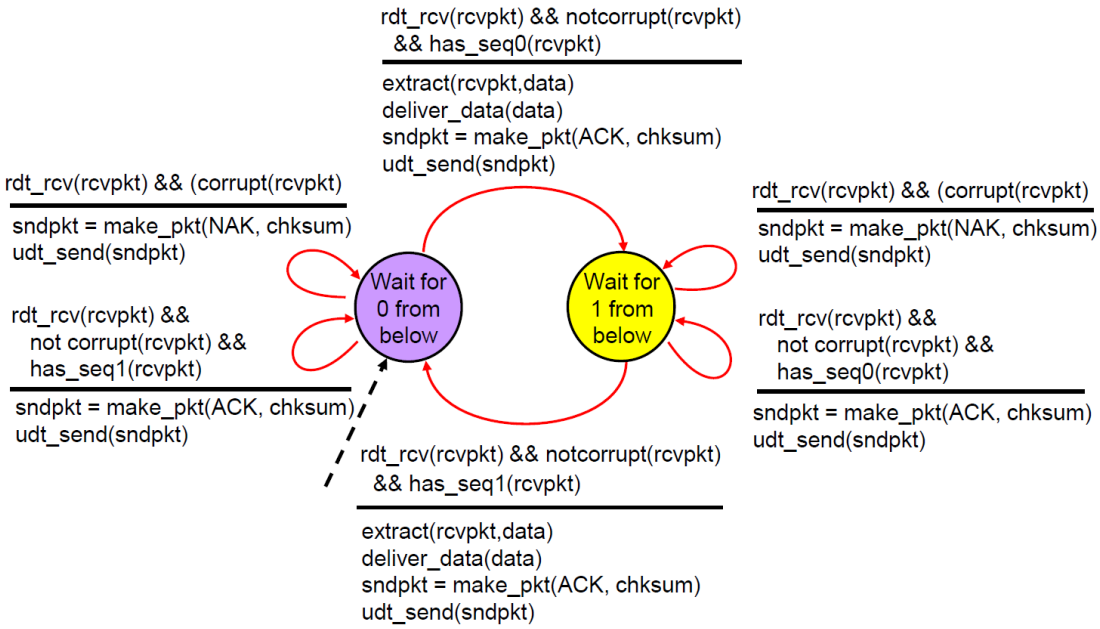


Figure 5: Reliable Data Transfer Receiver Finite State Machine

4.3.7 Performance of rdt 3.0

Definition 4.2 (Utilisation)**.**
Utilisation is defined as the fraction of time sender is busy sending.

rdt 3.0 stinks in performance, given its **stop-and-wait** nature. **Pipelining** can be used to increase utilisation, which allows multiple "in-flight", yet-to-be-acknowledged packets.
Two generic form of pipelined protocols are

- Go-Back-N
- Selective Repeat

4.4 Go-back-N

4.4.1 Sender Behaviour

- Sender maintains a window size of N, i.e., there can be N packets in pipeline, where the *first* is unACKed.
- There will be a timer for the first packet in the window.
- Packets in the window will be *cumulatively* ACKed.
- If the sender receive ACK of a packet with sequence ID k, all packet no later than sequence ID k will be ACKed; the window will slide until the next packet of sequence ID $k+1$.[3] Sender will send the packets newly entered in the window.
- Else, the first packet's timer will timeout by receiving no ACK with sequence ID in the window. The sender will retransmit all the packets in the window.

4.4.2 Receiver Behaviour

- Same as rdt 3.0, corrupted packets will be discarded.
- Acknowledge packet in order, by maintaining an `expectedSeqNum`. Any packet of different sequence ID from `expectedSeqNum` will be discarded.
- In all cases, acknowledgement is **cumulative**.
  When the expected packet is received, ACK with sequence ID `expectedSeqNum` will be sent and `expectedSeqNum++`.
  When other packet is received, ACK with sequence ID `expectedSeqNum-1` will be sent.
  As such, ACK k means all packets up to k are received.

The benefit of cumulative ACK is that lost ACK packet will not cause retransmission as long as one ACK packet after this lost packet is received by sender before timeout. Therefore, timeout value shall be delicately chosen.

However, Go-back-N will waste network resources in a sense corrupt-free packet will be discarded merely for being out-of-order.

[3] This is due to cumulative acknowledgement.

4.5 Selective Repeat

Selective repeat solves the problem by **buffering** the out-of-order packets.

4.5.1 Sender Behaviour

- Sender maintains a window size of N. Window will be maintained such that the first packet of the window is unACKed.
- There will be a timer for *every* packet in the window.
- Packets in the window will be *individually* ACKed.
- If one timer timeout, *only* the timed-out packet will be resent.
- Upon sliding of window due to acknowledgement of first packet, packets newly entered the window will be sent.

4.5.2 Receiver Behaviour

- Same as rdt 3.0, corrupted packets will be discarded.
- Acknowledge packet individually.
- Maintain a **receiver window** of equal length N to the sender window. The first packet in receiver window is not received.
- All the packets received *correctly*, which either are inside the receiver window, or outside the receiver window but within just one window length N before the first UnACKed packet, will fire an individual ACK message.
- If packets correctly received is in-order, all consecutive packets buffered in the receiver window will be delivered to application layer. The receiver window will slide to maintain the "first UnACKed" property.
- If packets correctly received is out-of-order, it will be buffered.
- All the packets *corrupted*, or out of the specified range before will not be buffered, and will not fire ACK.

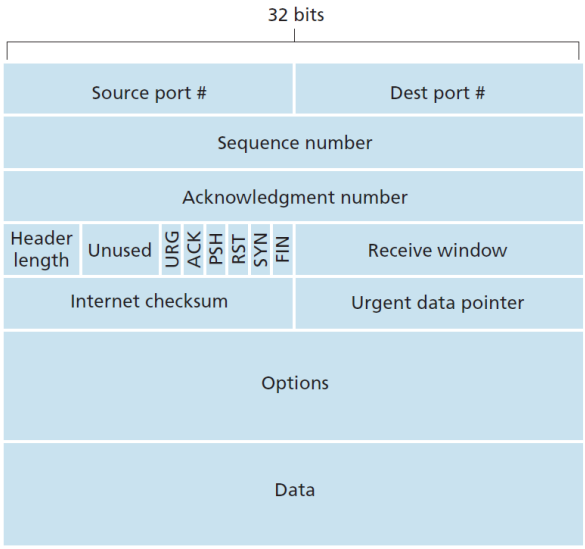
5 TCP : Transport Control Protocol

TCP is **connection-oriented** in a sense handshaking is required before sending application data.
TCP supports a **reliable, in-order** byte stream: Application passes data to TCP and TCP forms packets in view of **MSS**(maximum segment size).[4]
TCP also offers **flow control** and **congestion control**.

5.1 Connection-oriented De-mux

De-multiplexing in TCP is achieved by driecting a segment to the appropriate socket.
A TCP connection socket is identified by 4-tuple: `(srcIPAddr, srcPort, destIPAddr, destPort)`.

5.2 TCP Header



5.2.1 Sequence Number

TCP sequence number refers to the "byte number" of the first byte of data transferred in the segment in the file. Note the sequence number is 0-indexed.
The first sequence number used in handshake is randomly selected and this randomly selected number will be used as offset for all subsequent sequence number.

5.2.2 TCP ACK Number

TCP Acknowledgement number records the sequence number of the *next* byte of data *expected* by receiver.

[4]MSS excludes the length of TCP header.

TCP implements **cumulative** acknowledgement, in a sense TCP ACKs up to the first missing byte in the stream.

However, TCP spec doesn't say how receiver should handle out-of-order segments - it's up to implementer.

The A bit in the header:

- $A = 0$ denotes a data packet
- $A = 1$ denotes a ACK packet

There is only 1 timer running on the sender side.

5.2.3 TCP Sender Behaviour

- Event: `data` received from application above
  1. Create TCP segment with sequence number `NextSeqNum`
  2. Start timer if it is not running
  3. Send the packet
  4. Update sequence number used by next segment:`NextSeqNum = NextSeqNum + length(data)`
  5. Send until sender window is full.
- Event: Timer timeout
  1. Retransmit not-yet-acknowledged segment with *smallest* sequence number[5]
  2. Start timer
- Event: ACK received, with ACK field value `y`
  1. If `y>send_base`, shift sender window: `send_base=y`.
  2. If there are currently any not-yet-acknowledged segment, start timer.

5.2.4 TCP Receiver Behaviour

- Event: Arrival of in-order segment with sequence number `seqNum`
  - if all data up to `seqNum` are already ACKed, wait up to 500ms for next segment. Send ACK if no next segment arrival.
  - if one other segment is awaiting to send ACK, send single cumulative ACK, to ACK both in-order segment

[5]Only the oldest unACKed packet is retransmitted

- Event: Arrival of out-of-order segment with higher than expected `seqNum`(gap formed and detected)
  - Immediately send **duplicate** ACK, indicating `seqNum` of next expected segment[6]
- Event: Arrival of segment that partially or completely fills gap
  - Immediately send ACK, provided that the segment starts at lower end of gap.

5.2.5 TCP Timeout Value

TCP computes and keeps updating timeout interval based on estimated RTT.

$$\begin{aligned}
\text{Estimated RTT} &= (1-\alpha) \times \text{ Estimated RTT } + \alpha \times \text{ Sample RTT} \\
&\quad \text{typical value of } \alpha = 0.125 \\
\text{DevRTT} &= (1-\beta) \times \text{ DevRTT} + \beta \times |\text{SampleRTT} - \text{EstimatedRTT}| \\
&\quad \text{typical value of } \beta = 0.25 \\
\text{TimeoutInterval} &= \text{EstimatedRTT } + 4 \times \text{ DevRTT}
\end{aligned}$$

5.2.6 TCP Fast Retransmission

TCP adopts **fast retransmission** if sender receives 3 duplicate ACKs for the same data[7], it supposes that segment after the last ACKed segment is lost. Thus it will resend segment before timer expires[8].

5.2.7 Establishing Connection

Before exchanging data, TCP sender and receiver "shake hands".

1. Client chooses initial sequence number x, and send TCP SYN msg, by setting $S = 1$.
2. Server chooses initial sequence number y, and send TCP SYNACK msg, by setting $A = 1$,$S = 1$.
3. Client sends ACK msg by setting $A = 1$. In this message, it can contain client-to-server data.

5.2.8 Closing Connection

Client and server will close their side of connection by

1. Client sends FIN msg by setting $F = 1$. After this segment sent, client can no longer send application data, but can receive data.

[6] i.e. ACK the last correctly received packet
[7] 1+3 = 4 ACKs of the same data in total
[8] Retransmission will not reset the timer

2. Server send ACKFIN msg by setting $A = 1$, $F = 1$. Server can no longer send application data after this segment.

3. Client sends ACK mssg by setting $A = 1$.