



**Electronics and
Computer Science**

University of Southampton



CM214-COMP2008

Data Communications and Networks

Protocols Over IP

Karl R. Wilcox

krw@ecs.soton.ac.uk



Objectives



- To look at higher level protocols that make use of the “lowest common denominator” packet delivery of IP
 - UDP
 - TCP
- (Peterson & Davie, Section 5.1, 5.2)



Review



- In the last lecture we looked at IP
- “Best Effort” packet delivery
 - No guarantee of delivery, uniqueness or ordering
- Datagram (connectionless) protocol
 - All packets are independent



User Datagram Protocol



- Very simple demultiplexer
 - Allows more than one process on each host to use network connections
- Introduces the concept of “ports”
 - Up to 64K different ports per host
- Adds an additional (UDP) header containing:
 - Source port number
 - Destination port number
 - Length of data and checksum



UDP Characteristics



- Adds only demultiplexing
- Packet delivery is still:
 - Best efforts
 - No guaranteed delivery, order or uniqueness
- The Application must sort out any problems
 - E.g. Early FTP (File Transfer Protocol) programs use UDP for data transfer
 - Retransmission etc. was handled by the application program



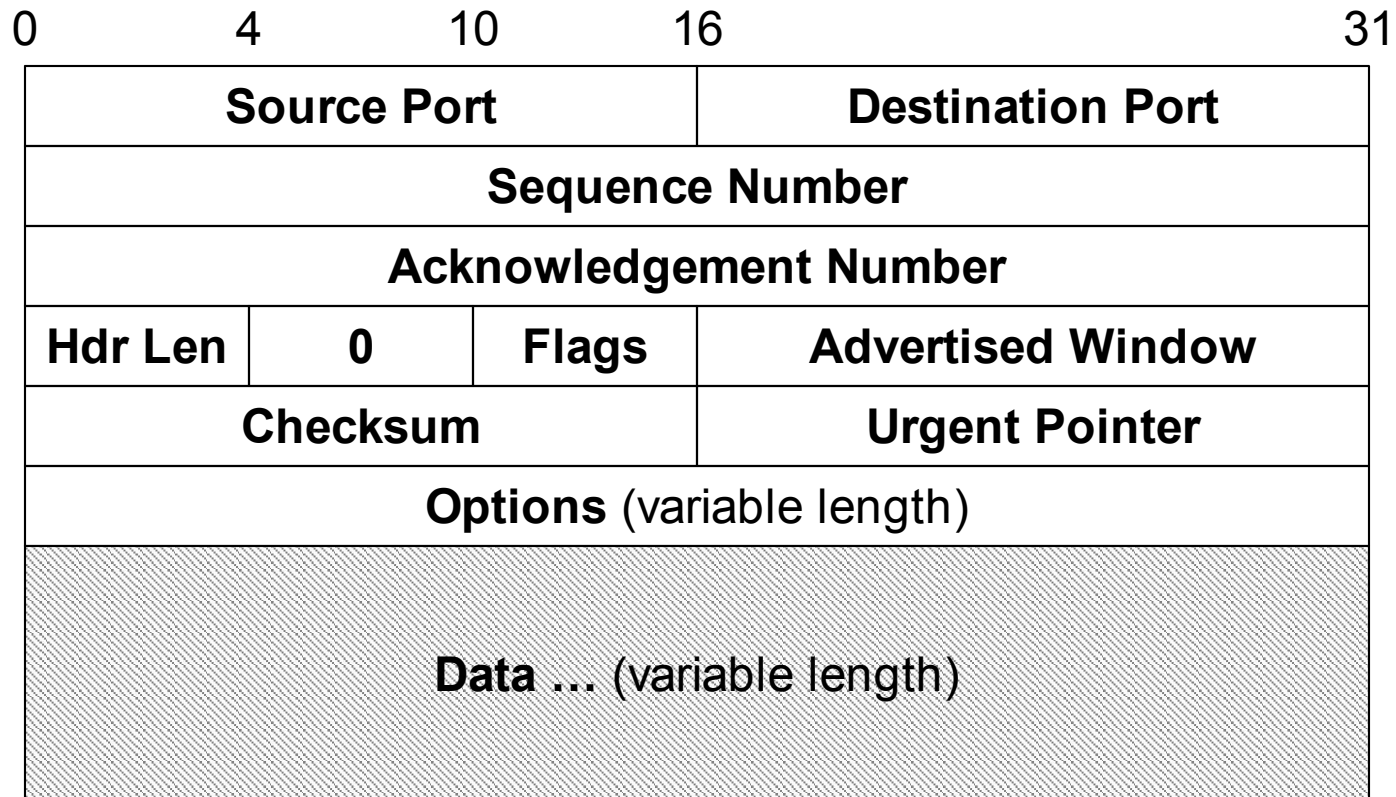
Transmission Control Protocol



- A Transport Protocol offering:
 - A connection oriented service
 - A byte-stream interface
 - Reliable connections (guaranteed delivery)
 - Flow control
- When run over IP, known as TCP/IP



TCP Header

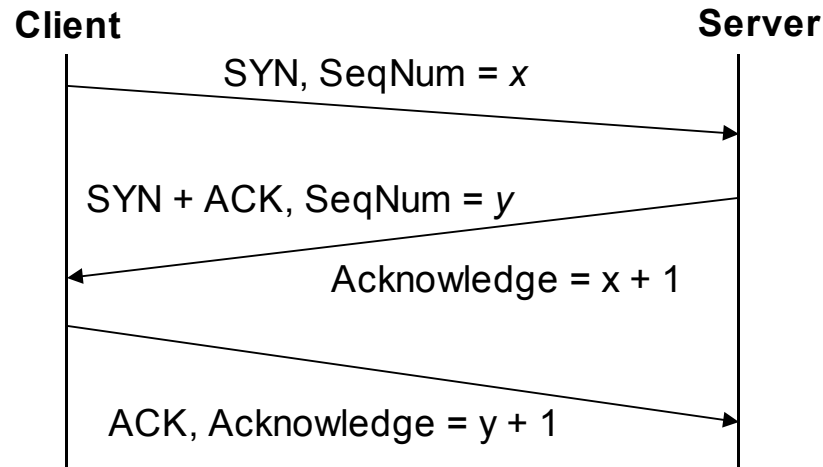




TCP Connections



- TCP is connection oriented
 - There must be an explicit set up phase to create an end-to-end connection
 - Uses a three way handshake





Full Duplex Connection



- TCP provides Full Duplex connections
- Both ends of the connection can simultaneously send & receive
 - Hence separate sequence numbers for each direction
- Acknowledgements are “piggy-backed” on sent packets
 - Or placed on an empty “dummy” packet



TCP Setup / Closedown



- In reality the handshaking is more complex, due to:
 - Possible loss of packets in the network
 - Different response to loss of each handshake
 - There is also an explicit close down phase
 - Either side can initiate close down
 - Different sequence of events depending on which side closed first and whether the other side continues to send packets



Byte Stream Interface



- Applications do not have to be concerned with packet sizes
 - They can write as few or as many bytes, as often as required
 - They can read as few or as many bytes are available, as often as required
- The Transport layer must provide input and output buffering



When to Send Packets



- A packet will be sent when
 - The minimum segment size (MSS) amount of data is available
 - Usually set to the maximum packet size of the local network minus the header
 - A timeout occurs
 - The application calls the “push” operation
 - Also forces acknowledgement from receiver
 - Therefore can be used to mark boundaries



Reliable Connections



- TCP provides reliable delivery
 - Check-summed
 - in order sent
 - No missing data
- Uses a “sliding window” algorithm
 - 32 bit sequence number of bytes
 - Receiver acknowledges every packet
 - Using ACK flag in header



Flow Control



- Sliding window sized is NOT fixed
- Return packets contain “advertised window” field (in bytes)
 - Sender should not have more unacknowledged bytes than advertised window size
 - Advertised window may be zero
 - Sender responsible for retrying
 - Implements end-to-end flow control



Retransmission



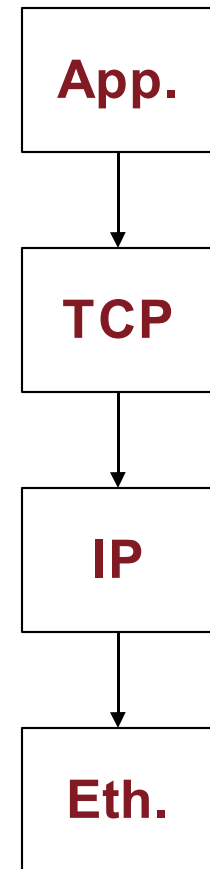
- TCP sets a time out period before assuming a packet is lost (i.e. not acknowledged)
 - Timeout is adaptive, dependent on estimated round trip time (RTT)
- After timeout, packet is retransmitted
- Several RTT estimation algorithms
 - To take account of congestion, lost packets



TCP Applications



- Telnet
 - Uses “push” function to flush packets quickly
- HTTP
- SMTP
- POP3
- Routing protocols
- Client / Server programs in general





Summary



- TBD