TCP/IP Emin Gun Sirer

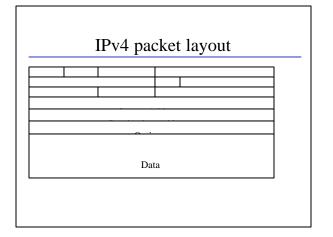
ΙP

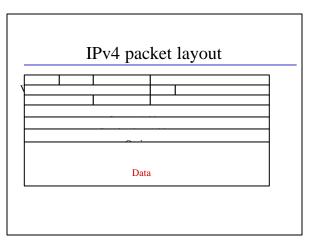
- Internetworking protocol
 Network layer
- Common packet format for the Internet
 Specifies what packets look like
 Fragments long packets into shorter packets
 Reassembles fragments into original shape
- Some parts are fundamental, and some are

 - arbitrary

 IPv4 is what most people use

 IPv6 clears up some of the messy parts, but is not yet in wide use





IP Fragmentation

- · Networks have different maximum packet sizes
 - Big packets are sometimes desirable less overhead
 - Huge packets are not desirable reduced response time for others
- Higher level protocols (e.g. TCP or UDP) could figure out the max transfer unit and chop data into smaller packets
 - The endpoints do not necessarily know what the MTU is on the path
 - The route can change underneath
- Consequently, IP transparently fragments and reassembles packets

IP Fragmentation Mechanics

- IP divides a long datagram into N smaller datagrams
- · Copies the header
- · Assigns a Fragment ID to each part
- Sets the More Fragments bit
- Receiving end puts the fragments together based on the new IP headers
- Throws out fragments after a certain amount of time if they have not be reassembled

IP Options

- Source Routing: The source specifies the set of hosts that the packet should traverse
- Record Route: If this option appears in a packet, every router along a path attaches its own IP address to the packet
- Timestamp: Every router along the route attaches a timestamp to the packet
- Security: Packets are marked with user info, and the security classification of the person on whose behalf they travel on the network
 - Most of these options pose security holes and are generally not implemented

UDP

- Unreliable Datagram Protocol
- IP goes from host to host
- We need a way to get datagrams from one application to another
- How do we identify applications on the hosts?
 - Assign port numbers
 - E.g. port 13 belongs to the time service

UDP Packet Layout

	Version	Ш	TOS		Total Length
IP	Identification			Flage	Fragment Offset
	TTI		Protocol	Header Checksum	
	Source Address				
	Destination Address				
UDP	Source Port			Destination Port	
	Length			Checksum	
	Data				

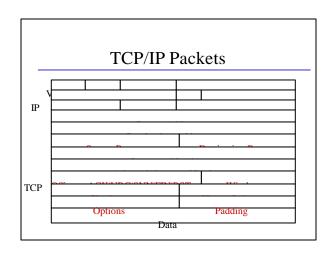
• UDP adds Ports, Data Length and Data checksum

UDP

- · UDP is unreliable
 - A UDP packet may get dropped at any time
 - It may get duplicated
 - A series of UDP packets may get reordered
- Applications need to deal with reordering, duplicate suppression, reliable delivery
 - Some apps can ignore these effects and still function
- Unreliable datagrams are the bare-bones network service
 - · Good to build on, esp for multimedia applications

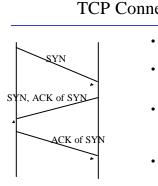
TCP

- Transmission Control Protocol
 - · Reliable, ordered communication
- Enough applications demand reliable ordered delivery that they should not have to implement their own protocol
- A standard, adaptive protocol that delivers goodenough performance and deals well with congestion
- All web traffic travels over TCP/IP



TCP Packets

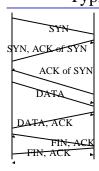
- Each packet carries a unique ID
 - The initial number is chosen randomly
 - · The ID is incremented by the data length
- Each packet carries an acknowledgement
 - · Can acknowledge a set of packets by ack'ing the latest one received
- Reliable transport is implemented using these identifiers



TCP Connections

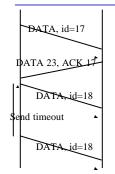
- TCP is connection oriented
- A connection is initiated with a threeway handshake
- · Three-way handshake ensures against duplicate SYN packets
- Takes 3 packets, 1.5 RTT

Typical TCP Usage



- Three round-trips to set up a connection, send a data packet, receive a response, tear down connection
- FINs work (mostly) like SYNs to tear down connection
 - · Need to wait after a FIN for straggling packets

Reliable transport



- TCP keeps a copy of all sent, but unacknowledged packets
- If acknowledgement does not arrive within a "send timeout" period, packet is resent
- Send timeout adjusts to the round-trip delay

TCP timeouts

- What is a good timeout period?
 - Want to improve throughput without unnecessary transmissions

NewAverageRTT = (1 - ?) OldAverageRTT + ? LatestRTT NewAverageDev = (1 - ?) OldAverageDev + ? LatestDev where LatestRTT = (ack_receive_time - send_time), LatestDev = |LatestRTT - AverageRTT|, ? = 1/8, typically. Timeout = AverageRTT + 4*AverageDev

• Timeout is thus a function of RTT and deviation

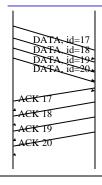
TCP Windows





• Multiple outstanding packets can increase throughput

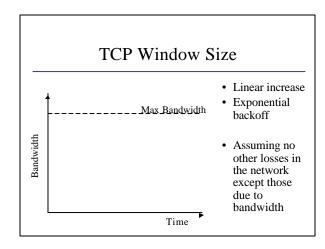
TCP Windows

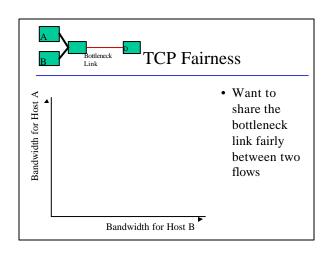


- Can have more than one packet in transit
- Especially over fat pipes, e.g. satellite connection
- Need to keep track of all packets within the window
- Need to adjust window size

TCP Congestion Control

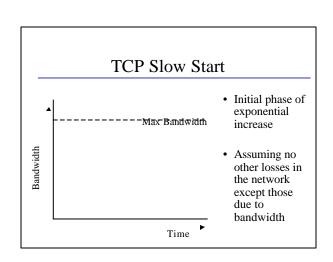
- TCP Increases its window size as long as no packets are dropped
- It halves the window size when a packet drop
 - A packet drop is evident from the acknowledgements
- Therefore, it will slowly build up to the max bandwidth, and hover around the max
 - It doesn't achieve the max possible though
 - Instead, it shares the bandwidth well with other TCP connections
- This linear-increase, exponential backoff in the face of congestion is termed *TCP-friendliness*





TCP Slow Start

- Linear increase takes a long time to build up a window size that matches the link bandwidth*delay
- · Most file transactions are not long enough
- Consequently, TCP can spend a lot of time with small windows, never getting the chance to reach a sufficiently large window size
- Fix: Allow TCP to build up to a large window size initially by doubling the window size until first loss



TCP Summary

- Reliable ordered message delivery
 - Connection oriented, 3-way handshake
- Transmission window for better throughput
 - Timeouts based on link parameters
- Congestion control
 - Linear increase, exponential backoff
- Fast adaptation
 - Exponential increase in the initial phase