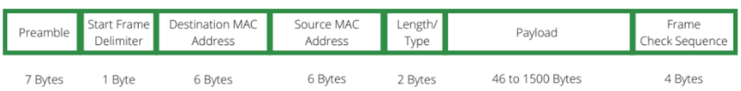
S/N	Jargon	Remarks	Formula
1.	**Bit Time**	A bit time is the time taken to transmit one bit R is the bandwidth of the link	$N\ Bit\ Time = \frac{N}{R}$
2.	**Packet Transmission Delay**	• Time needed to transmit L-bit packet into link • Host sending function	$D_{trans} = \frac{L(bits)}{R(bits/sec)}$
3.	**Propagation Delay**	• Time taken for signal to travel from S to R (based on distance & speed of signal)	$D_{prop} = \frac{d}{s}$ Distance over speed of signal **RTT ~ 2 x D_{prop} (assume no other D)
4.	**End-To-End Delay**	• Store and Forward (entire packet must arrive at router before transmission)	$D_{end-to-end} = 2 \times \frac{L}{R}$ *Assuming no other delays
5.	**Processing Delay**	• Time taken for devices to process incoming packets. • E.g check bits, determine output, ACK etc	
6.	**Queuing Delay**	• Time waiting in the queue for transmission. • Depends on congestion level of router	
7.	**Throughput**	The number of USEFUL bits that can be transmitted per unit time ** Depends on QNS. ½ or 1 RTT	$Throughput = \frac{total\ bits}{total\ time}$ $Throughput = \frac{L}{RTT + D_{trans}}\ OR\ \frac{L}{RTT/2 + D_{trans}}$
8.	**Utilization**	• Fraction of time sender is busy sending • Pipelining increases utilisation by window size N times! (100% use = 1) This is the formula for pipelined rdt protocols (using sliding window) - The denominator is (RTT + L/R) because thats the amount of time it takes for the sender to receive the ACK for the first pkt sent and hence send the next pkt (the window slides)	$U_{sender\_nopipe} = \frac{D_{trans}}{RTT + D_{trans}}$ $U_{sender\_pipe} = \frac{N \times D_{trans}}{RTT + D_{trans}}$ But there might be questions where there will be no mention of rdt protocols - so the denominator will not look like this - since there might not be any waiting for ACK involved
9.	**Bandwidth-Delay Product (BDP)**	Maximum amount of data that can be in transit in the network	$BDP = bandwidth \times RTT$
10.	**TCP window size**	• Window size determines the amount of data that can be sent before receiving an ACK. • MSS is influenced by MTU	$Optimal\ Window\ Size = \frac{BDP}{MSS}$ MSS → Max Segment Size
11.	**Minimum Frame Size** **	• Smallest amount of data that can be transmitted in a frame on a particular network • Collision happens but may not be detected by sending nodes. → No retransmission! • For example, Ethernet requires a minimum frame size of 64 bytes.  $L \geq D_{prop} \times Link\ Rate\ (Ignore\ Collision)$ $L \geq 2 \times D_{prop} \times Link\ Rate\ (Detect\ all\ Collisions)$	

12.	**Bellman Ford**	Single Source Shortest Path, O(VE)	$d_x(y) = min_v\{c(x,v) + d_v(y)\}$
13.	**Properties RSA**	$m = K_B^-(K_B^+(m))$	$K_B^-(K_B^+(m)) = m = K_B^+(K_B^-(m))$
14.	**Adaptively Estimate Packet Delay**	i.e Exponentially Weighted Moving Average (EWMA) • d_i → delay est. after i^{th} pkt • v_i → est. ave deriv after i^{th} pkt • r_i, t_i → time Rvc, timestamp • α, β → small constant	$d_i = (1-\alpha)d_{i-1} + \alpha(r_i - t_i)$ $v_i = (1-\beta)v_{i-1} + \beta\lvert r_i - t_i - d_i\rvert$
15.	**Playout-Time**	It consists of 1) Time needed to collect an audio sample and to prepare it for transmission. 2) Network delay, 3) Buffering time	$Playout\ Time = t_i + d_i + 4v_i$
16.	**Estimate Sample RTT**	Timeout Interval → EstimatedRTT + "Safety Margin" • Large variation in EstimatedRTT → larger safety margin • Estimate SampleRTT deviation from EstimatedRTT • α = 0.125 (usually)	$DevRTT = (1-\beta) \times DevRTT + \beta \times \lvert SampleRTT - EstimatedRTT\rvert$ $EstimatedRTT = (1-\alpha) \times EstimatedRTT + \alpha \times SampleRTT$
17.	**Timeout Interval**		$Timeout\ Interval = EstimatedRTT + 4 \times SampleRTT$
18.	**Compression Ratio**	Compression Ratio = Uncompressed Bitrate / Compressed Bitrate	
19.	**Uncompressed Data Rate (bps)**	Uncompressed Data Rate (bps) = Sampling Rate (samples per second) x Bit Depth (bits per sample) x Number of Channels	
20.	**Bit depth**	• Bit depth refers to the number of bits used to represent the color of each pixel in a digital image or the amplitude of each sample in digital audio. • **Given XXX levels, bit depth = $\log_2$(XXX)** • **Given YY-bit quantization, bit depth = YY** - "bit quantization" typically refers to the number of bits used to represent each sample in a digital signal.	

- RFC → Request for comments
- TCP segment can typically carry about 1460 bytes of app-data (MSS).

QNS	** Suppose the propagation delay between furthest nodes is d and link rate is r. What is the minimal frame size L to ensure collision will ALWAYS be detected ion CSMA/CD protocol?
ANS	• Usually → L = rd is the minimal frame size. • However, since we want to ensure that the collision is always detected, L = 2rd. Why? - In the worst case scenario, the collision will occur at t = dprop (assuming transmission starts at t = 0), and the transmitter will detect it at 2*D_{prop}, so the minimal frame size should be 2rd to allow for detection to happen in this case.

- IP address → 32 bit (2^5) or 4 bytes
- Port Number →16 bit (2^4) or 2 bytes
- Simple NAT table header → source IP #, source port #, dest IP #, dest port # → 12 bytes

TCP or UDP?

Protocol	Purposes
TCP	
HTTP (Hypertext Transfer Protocol)	Used for transferring hypertext documents on the web.
HTTPS (HTTP Secure)	Secure version of HTTP, often used for secure data transfer over the web.
FTP (File Transfer Protocol)	Used for transferring files between hosts on a network.
SMTP (Simple Mail Transfer Protocol)	Used for sending emails between email servers.
POP3 (Post Office Protocol version 3)	Used for retrieving emails from a mail server.
IMAP (Internet Message Access Protocol)	Allows an email client to access emails on a mail server.
Telnet	Provides a command-line interface to communicate with remote systems.
UDP	
DNS (Domain Name System)	Resolves domain names to IP addresses.
DHCP (Dynamic Host Configuration Protocol)	Used to dynamically assign IP addresses and network configuration to devices. • DHCP server port number: 67 • DHCP client port number: 68
SNMP (Simple Network Management Protocol)	Used for network management and monitoring.
TFTP (Trivial File Transfer Protocol)	Simple file transfer protocol often used for bootstrapping devices.
NTP (Network Time Protocol)	Synchronizes the clocks of computers on a network.

Commands and uses:

Commands	Purpose	Involved Protocol:
dig	Performs a DNS query	DNS
nslookup	Similar to`dig`, performs DNS query	DNS
telnet	Connects to a server on port	Telnet
ping	Uses ICMP to check reachability of host ** does NOT cause a DNS query to be issued	ICMP
traceroute (or tractrt)	Traces the route of the packets take to reach the host	ICMP (for probes)
Ifconfig/ dnscherker.org	Check MAC	

dig DNS:

Local DNS → Root DNS → TDL DNS → Authoritative DNS → IP Address

DNS	;; Received XXX bytes from XXX.XXX.XXX.XXX#YY(…) in ZZ ms
IP	Example.sg NNNNNN IN A (IP Address XXX.XXX.XXX.XXX)

- Routers by principle do not forward broadcast traffic
- ICMP is used by routers to send error messages.

Special Addresses	Present Use
0.0.0.0/8	Non-routable meta-address for special use
127.0.0.0/8	Loopback address. A datagram sent to an address within this block loops back inside the host. This is ordinarily implemented using only 127.0.0.1/32.
10.0.0.0/8 172.16.0.0/12 192.168.0.0/16	Private addresses, can be used without any coordination with IANA or an Internet registry.
255.255.255.255/32	Broadcast address. All hosts on the same subnet receive a datagram with such a destination address.

Network Layers and Constituents:

Application Layer	HTTP	TLS	DNS	DHCP
Transport Layer	TCP	UDP		
Network Layer	IP (v4, v5)	ICMP		
Link Layer	Ethernet	Wireless LAN		

1XX Informational	
100	Continue
101	Switching Protocols
102	Processing

2XX Success	
200	OK
201	Created
202	Accepted
203	Non-authoritative Information
204	No Content
205	Reset Content
206	Partial Content
207	Multi-Status
208	Already Reported
226	IM Used

3XX Redirectional	
300	Multiple Choices
301	Moved Permanently
302	Found
303	See Other
304	Not Modified
305	Use Proxy
307	Temporary Redirect
308	Permanent Redirect

4XX Client Error	
400	Bad Request
401	Unauthorized
402	Payment Required
403	Forbidden
404	Not Found
405	Method Not Allowed
406	Not Acceptable
407	Proxy Authentication Required
408	Request Timeout

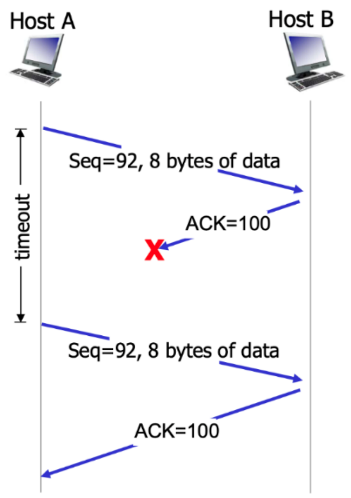
4XX Client Error Continued	
409	Conflict
410	Gone
411	Length Required
412	Precondition Failed
413	Payload Too Large
414	Request-URI Too Long
415	Unsupported Media Type
416	Requested Range Not Satisfiable
417	Expectation Failed
418	I'm a teapot
421	Misdirected Request
422	Unprocessable Entity
423	Locked
424	Failed Dependency
426	Upgrade Required
428	Precondition Required
429	Too Many Requests
431	Request Header Fields Too Large
444	Connection Closed Without Response
451	Unavailable For Legal Reasons
499	Client Closed Request

5XX Server Error	
500	Internal Server Error
501	Not Implemented
502	Bad Gateway
503	Service Unavailable
504	Gateway Timeout
505	HTTP Version Not Supported
506	Variant Also Negotiates
507	Insufficient Storage
508	Loop Detected
510	Not Extended
511	Network Authentication Required
599	Network Connect Timeout Error

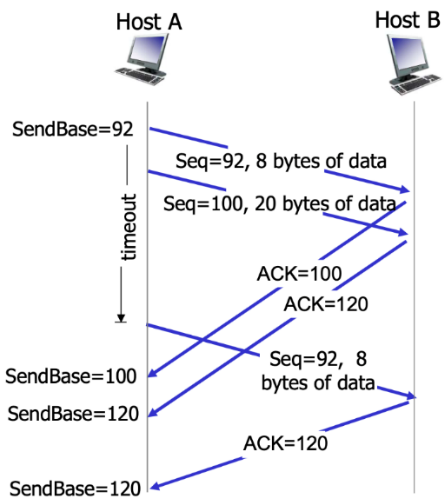
HTTP STATUS CODES

When a browser requests a service from a web server, an error may occur.
This is a list of HTTP status messages that might be returned.

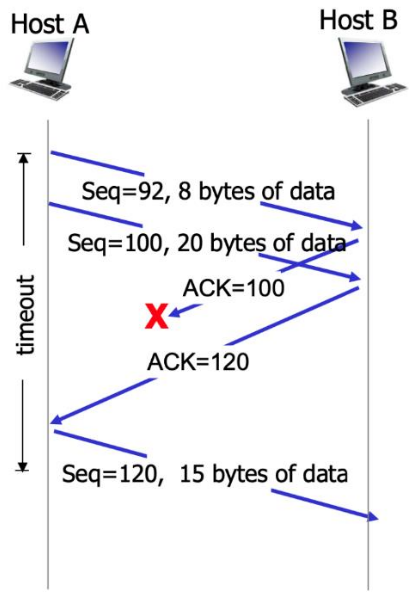
- App layer: Client-Server/ P2P
- Transport Layer: Process to Process
- IP: Host to Host
- Link: Physical Node over link (send data between N nodes via cable) [For CS2105]



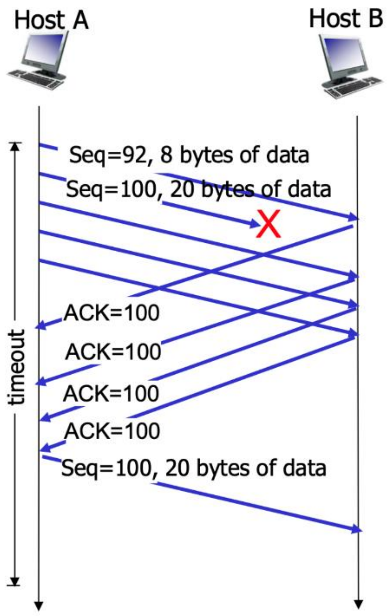
lost ACK scenario



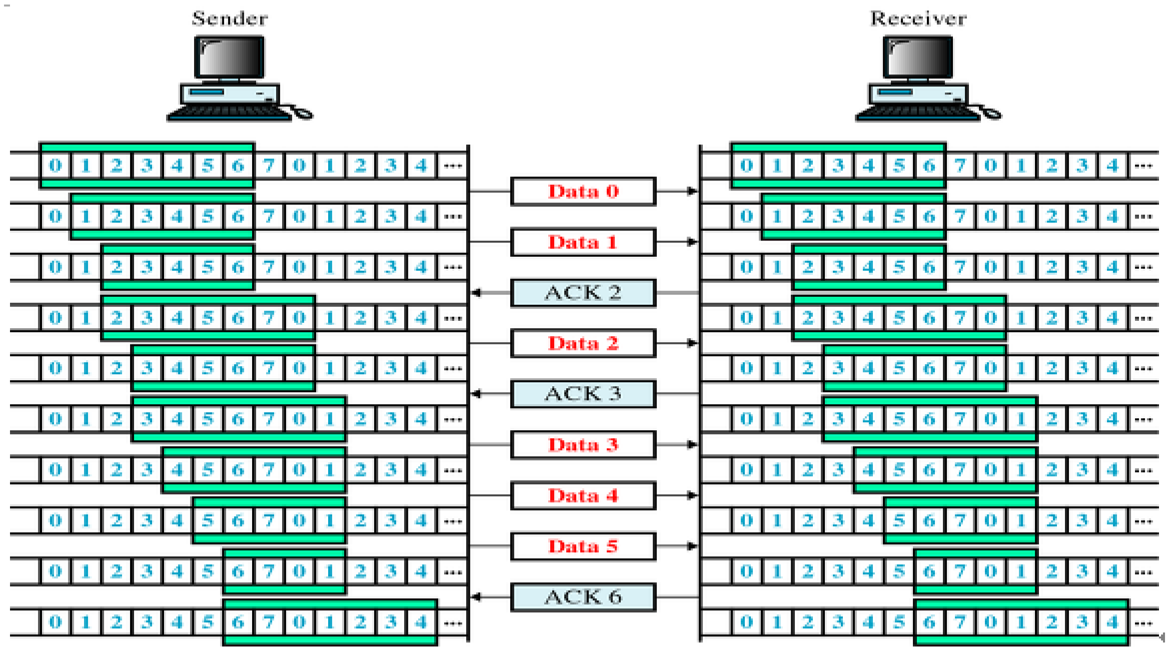
premature timeout



cumulative ACK



fast retransmit after sender receipt of triple duplicate ACK


Sender
Receiver
Data 0
Data 1
ACK 2
Data 2
ACK 3
Data 3
Data 4
Data 5
ACK 6