5 measured RTT (106, 120, 140, 90, 115) Compute Estimated RTT of those values & the estimated RTT before = 100 m/s
EstimatedRTT = (1-α) * EstimatedRTT + α • SampleRTT α = 0.125 Then compute DevRTT after each sample is obtained
β = 0.25 & the dev RTT was 5 m/s before those 5 numbers. DevRTT = (1-β) * DevRTT + β • |SampleRTT - EstimatedRTT| Then compute
Tcp timeout interval after each sample is obtained. TimeoutInterval = EstimatedRTT + 4 * DevRTT

Tcp Sequence Number Acks

Client Server

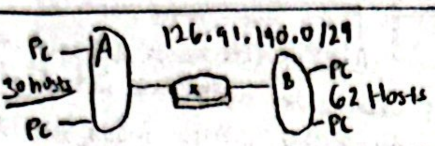
Tcp Seq # 1 Len = 669 → Tcp Ack # 670

Tcp Seq # 670 Len = 1460 → Tcp Ack # 2130

Tcp Seq = 2130 Len = 1460

CIDR IP Addressing a.b.c.d/x
x = num bits in subnet portion of Address. Example 200.23.16.0/23
Subnet is 23 bits long 32-9
host part is 9 bits long

126.91.140.0/24



The subnets are 24 high order bits 2⁸

Assign subnet Addresses to A & B so the ammount of Address space is minimal & leaving the largest possible contiguous address avilble for assignment if a new subnet were added. 1. How many Hosts in this Address space subnet = 24 so 8 left over so $2^8 = 256 - 2 = 254$ -2 one for First & Last Address

2. What is the subnet Address of subnet A 30 hosts +2 1 For first & last. Explanation we Need support for 32 Addresses, $2^5 = 32$ so 32 bit ip address -5 = 27 So subnet A 126.91.140.0/27

3. What is the broadcast address of subnet A? Broadcast Address in range last Address

126.91.140.31

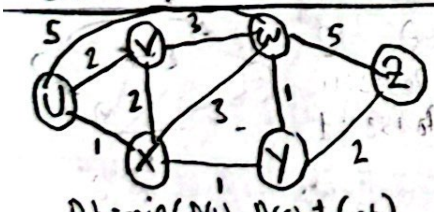
4 What is the first host address in subnet A? 126.91.140.0 is the subnet so First
First host address is 126.91.140.1

5. Last host address in subnet A? is one less then broadcast Address

6. Subnet Address of subnet B? SubnetA ends at 126.91.140.31 so B Starts at
* 126.91.140.132 so in cidr Notation 126.91.140.32/26 - has 64 Address because $2^6 = 64$

7. Broadcast Address for B BroadcastAddress is Last Address in range so broadcast Address = Subnet Address of b + broadcast address
126.91.140.32+63 = 126.91.140.32.95

8. First Host Address of subnet B Ex 1 more than subnet Address of B so 126.91.140.33*
9. Last Host Adress of Subnet B E 1 less than broadcast Address 126.91.140.32.94

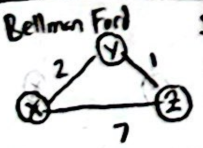


Db = min(D(b), D(a) + Cab)

Find shortest path
rember path it took to get there when picking path
Dijkstra's Algo

	N'		D(v) P(v)	D(w) P(w)	D(x) P(x)	D(y) P(y)	D(z) P(z)
0	[illegible]		2,u	5,u	1,u	∞	∞
1	[illegible]		2,u	4x		2x	∞
2	[illegible]	1	2,u	3,y		-	4y
3	[illegible]	2y		3y			4y
4	[illegible]						4y
5	[illegible]						

Bellman Ford
Start at x x=0
doesn't Know Next hop till it gets there



x table

	x	y	z
x	0	2	7
y	∞	∞	∞
z	∞	∞	∞

	x	y	z
x	0	2	3
y	2	0	1
z	7	1	0

	x	y	z
x	0	2	3
y	2	0	1
z	3	1	0

y table

	x	y	z
x	∞	∞	∞
y	2	0	1
z	∞	∞	∞

	x	y	z
x	0	2	7
y	2	0	1
z	7	1	0

	x	y	z
x	0	2	3
y	2	0	1
z	3	1	0

z table

	x	y	z
x	∞	∞	∞
y	∞	∞	∞
z	7	1	0

	x	y	z
x	0	2	7
y	2	0	1
z	7	1	0

	x	y	z
x	0	2	3
y	2	0	1
z	3	1	0

Tcp - Connection oriented reliable, 3 way handshake
Syn initizes connection
Syn Ack Acknowledges connection request
Ack confirms connection request
Window based Flow & Congestion Control
Flow control - uses a sliding window to manage transmitted data
Congestion control - adjust window based on Network traffic

UDP - None of what Tcp has to offer - but is fast & simple. used in streaming multimedia apps, DNS SNMP, HTTP/3
add reliablity at app layer, add congestion control at app layer

Transport Layer services -
Relicbele data transfer - in order & No errors
Flow control - so sender doesn't overwhelm slow server
Congestion control - avoids Network congestion Controls rate at which of data entering network
Multiplexing - handle data from multiple sockets
Demultiplexing - use header info to deliver data to correct socket

application
Transport udp/Tcp
Network Ip
Link
Physical

Udp Segment header
32 bit

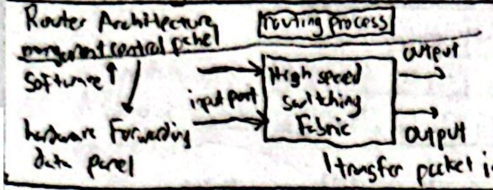
Source port	dest port
Length	Checksum
App data	

Tcp Flow control - reciever controls Sender so dont overflow buffer.
rwnd - recieve window free buffer space, RcvBuffer - buffered data usually 4096 bytes
Sender Limits unacked data to received rwnd so wont overflow buffer
Congestion is infered from loss & delay
Network assisted - routers provide direct feedback to sending receiving hosts
Tcp, Ecn, ATm, Decbit protocols
AIMD - Senders increase sending rate until packet loss then decreases sending rate

Additive increase - increase by 1 until loss. Multiplicative Decrease cuts sending rate in half at loss. Triple Ack is Loss Reno Tcp cut to 1mss when timeout. optimize congested Flow rates and is stable. ssthresh on Loss, ssthresh is set to 1/2 cwnd congestion window
double double then · Slow start

Data plane - transfer packets from input to output port using address Local - Control Plane - Network wide Logic. How datagram is routed among routers along end to end path from source to dest port. Using algos in routers, or SDN Software defined networking

Router Architecture



Memory Buffering - arrival rates exceeds transmission rate
Round Robin - queued by class
Weighted Queing - generalized round robin

Inside a port

line termination	Link layer	Lookup Forwarding

Physical Layer · queuing

Ip Forwarding techniques - Longest prefix match Ipv4 32 bit 4.3 bill Ipv6 128 bit address
Ipv6 difference. 128 dest & source Addy & No checksum

Control plane - path selection using Algos. control plane routes where traffic goes, Data plane makes packets move. control plane use data table to forward traffic
Distance Vector - router advertises distance to all destinations it knows like bellman Ford
Link state - router builds map of connection to Network, Dijkstras alg
OSPF Link state, RIP distance vector, BGP Go to path vector protocol

BGP - determines best path between autonomous systems - large collections of Ip address under control of one or several network operators that present a common routing policy
Route Aggregation - Reduces the number of routes by advertising a single route to cover multiple destinations

32 bits · length of header in bytes · max Length 64 K

version number	Ver	head len	type of service	Length datagram Length
Tcp or Udp	time to Live		upper Layer	Checksum
	Source Ip			
	dest ip			
	options			
	Payload data			

Link Layer - Error detection using Crc Cyclic Redundancy check. Framing makes Network Layer datagrams into Frames. Each Frame contains Mac address
Parity checking - Even parity No error 10101|1 add one to make even 10110|0 error would need a 1 to make even works up and down
cyclic Redundancy check CRC more powerful. TDMA access to channel in rounds. each station gets a time slot, unused slots go idle
FDMA - channel spectrum devided into frequency bands
Slotted Aloha - pros can transmit at full rate simple cons collision wasting slots, idle slots, decting collision may take longer then time to transit. Clock sync. efficiency N nodes with many frames to send with problity p
Prob that the node has success in slot $p(1-p)^{N-1}$ · many Nodes limit of $Np^*(1-p)^{N-1}$
Prob that any Node can succed $Np(1-p)^{N-1}$ · max efficiency $= 1/e$ .37
max efficiency find p^* $Np(1-p)^{N-1}$ · at best channel used for useful transmission is 37% time

Pure Aloha - when frame first arrives transmit. prob any node has success is $p(1-p)^{2(N-1)}$ max efficiency find p^* $p(1-p)^{2(N-1)}$
Pure Aloha efficiency 18% $1/(2e)$ when N goes to ∞

3 Active Nodes constant size L bits 2 or more frames collide all nodes know use Slotted Aloha & Pure

Prob given Node has success $p(1-p)^{N-1}$
Prob any Node has success $Np(1-p)^{N-1}$
max efficiency $= Np(1-p)^{N-1}$
many Nodes take limit of $Np^*(1-p)^{N-1}$
Slotted at best 37% max efficiency

Pure Aloha
any node has success $= p(1-p)^{2(N-1)}$
max efficiency p^* $p(1-p)^{2(N-1)}$

Probablity of transmission $p = 0.36$ what is max efficiency
Slotted Aloha $- Np(1-p)^{N-1}$ $3 \cdot .36 \cdot (1-0.36)^{(3-1)} = .44$ or 44%
Pure Aloha $Np(1-p)^{2(N-1)}$ $= 3 \cdot .36 \cdot (1-0.36)^{2(3-1)} = 0.18$ or 18%

N - number of Nodes
p - probablity