

## \* Assignment 3: Lower Network Layers

## • Question 1: Reliable Data Transfer

- we want to send data from one node to two other nodes using a simple broadcast signal
- Specifically we want to design a protocol for reliably sending data from host S to hosts R1 and R2 over this channel
- channel can disrupt or lose packets
  - ↳ packet sent by S received by R1 but not R2
- When there are collisions on the broadcast channel, you can assume receiving hosts will detect them as corrupt packets
- If data needs to be resent, you can ignore random backoffs etc.
- and assume colliding hosts will be able to read their data w/o interference

## • Design the protocol state machines for S and R (R1 and R2 = same protocols)

- use primitives vdt\_send vdt\_receive etc.
- don't consider pipelining
- RDT protocol with sequence numbers 0 and 1 → good starting point

- RDT protocol for single receiver
  - ↳ uses sequence numbers (0 and 1) to handle duplicates and acknowledgments
  - ↳ so sender sends a packet with a seq. num. waits for an ACK if it doesn't receive it within timeout, it retransmits
  - receiver sends ACKs for received packets; ignores duplicates

## • TWO receivers

- main challenge: ensure both R1 and R2 receive data correctly
- Since channel is a broadcast when S sends a packet both R1 and R2 can receive it but they might experience different outcomes (one gets it, the other doesn't; both receive corrupted versions)

- How does sender know when to move to the next packet?
  - Needs ACKs from both R1 and R2
  - How do receivers send ACKs w/o causing collisions?
  - Collisions can happen but when resending we can assume they eventually get through
  - So protocol could handle case where ACKs from R1 and R2 might collide
    - ↳ but we are ignoring random backoffs so sender can retransmit until it gets necessary ACKs

- Can receivers coordinate ACKs?
  - receivers a separate but have same protocol so maybe each receiver sends ACKs independently and sender must collect ACKs from both

- Senders State Machine needs to track whether both R1 and R2 have ACKed the current packet
  - ↳ maybe we need 2 separate state variables for each receiver

↳ For example, for the current sequence number S needs to know if R1 has ACKed and if R2 has ACKed  
Only when both have ACKed does S increment the sequence number and send the next data

## • Approach

1. Sender S sends packet with sequence number 'seq'
2. Both R1 and R2, upon receiving the packet correctly (checksum passes), send an ACK with 'seq'
3. Sender S must receive an ACK from R1 and R2 for the current 'seq' before moving to the next packet
4. If S doesn't receive ACKs from both within a timeout, it resends the packet
5. For handling sequence numbers, since we're using 0 and 1, after sending a packet with seq 0, S waits for ACK0 from R1 and R2. Once both are received, it sends the next packet with seq 1, and waits for ACK1 from both

- How does S track the ACKs?
  - channel is a broadcast, so when S sends a packet both R1 and R2 receive it (if not lost)
  - When S receives ACK, could be from R1 or R2
  - ACK packets must include identifier

## • Sender S

- S maintains 2 flags: acked\_R1 and acked\_R2 for current 'seq'
- When S sends packet, it starts a timer
- If S receives ACK(seq) from R1, set acked\_R1 = true
- Same for R2
- If both are true before timeout, send next packet
- If one or both = false when timeout occurs, S resends current packet, set flag and restart timer

- Receivers R1 and R2
  - Wait for packet from S
  - When packet is received correctly (no corruption), send ACK with own ID and 'seq'
  - If packet is corrupted (detected via checksum or collision), they ignore it
  - Check sequence number to avoid duplicates

- Receivers need to keep track of <sup>expected</sup> sequence number
  - So each receiver (R1 and R2) has their own state for the next expected 'seq'
  - received packet w/ seq = expected sequence number
    - ↳ deliver data to upper layer, send ACK with their ID and 'seq', increment expected 'seq'
  - ↳ IF # (ie duplicate) resend ACK for last expected 'seq'

## \* Sender S State Machine

1. States
  - Current sequence number (0 or 1)
  - Acked\_R1 (True / False)
  - Acked\_R2 (True / False)
  - Timer (Active / Inactive)

## 2. Transitions

- Initial State: seq = 0, Acked\_R1 = false, Acked\_R2 = false, timer = inactive
- Send Data
  - When data is available, send packet with current sequence via vdt\_send()
  - Start timer
- Receive ACK
  - If ACK is for current sequence and from R1: Set Acked\_R1 = true
  - If ACK is for current sequence and from R2: Set Acked\_R2 = true
  - If both Acked\_R1 and Acked\_R2 are True:
    - toggle current sequence (0 → 1)
    - set Acked\_R1 and Acked\_R2 to false
    - Stop Timer. If more data, send next packet
- Timeout
  - Resend packet with current sequence
  - Restart timer

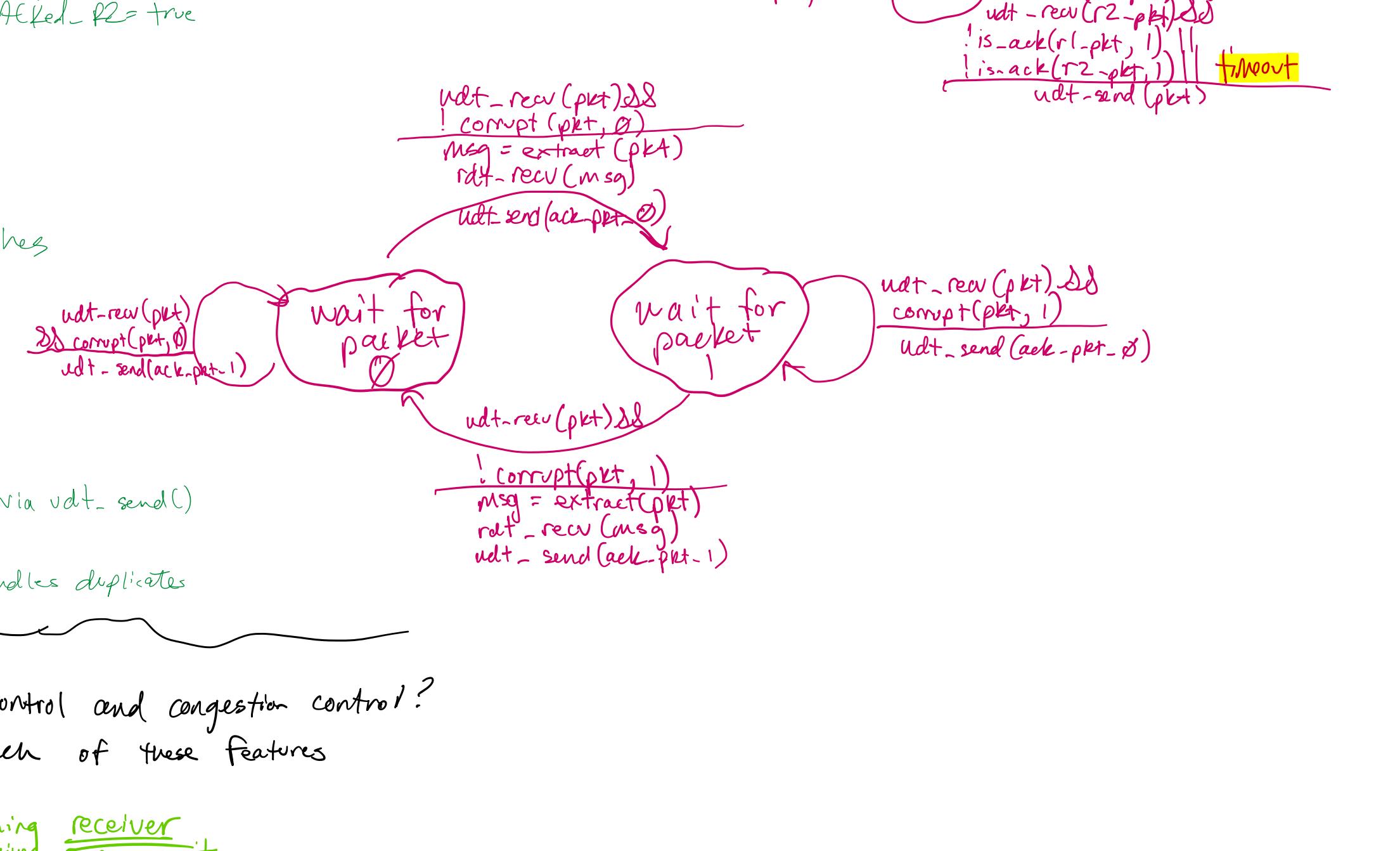
## \* Receiver R (R1 and R2) State Machines

## 1. States

- Expected Sequence Number (0 or 1)

## 2. Transitions

- Initial state: expected\_seq = 0
- Receive packet
  - If packet is corrupt: (ignore)
  - If packet is valid and sequence = expected\_seq:
    - deliver data to application via deliver\_data()
    - Send ACK with (receiver\_id, received\_seq) via vdt\_send()
    - toggle expected\_seq (0 → 1)
  - If packet is valid but seq ≠ expected\_seq:
    - send ACK with (receiver\_id, received\_seq) → handles duplicates

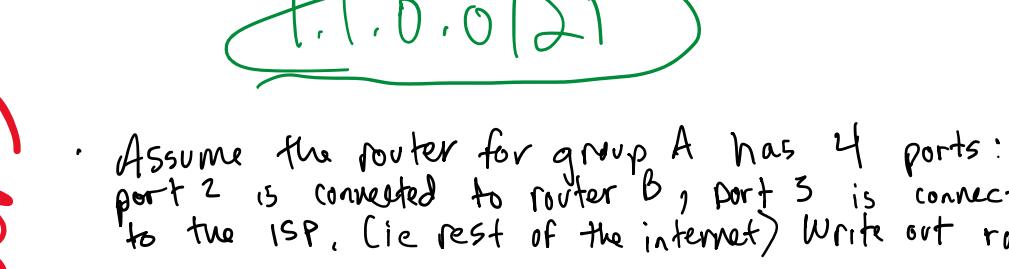


## \* What is the difference between flow control and congestion control?

Describe the way TCP implements each of these features

- Flow control [TCP] prevents sender from overwhelming receiver
  - ↳ by ensuring sender does not transmit data faster than receiver can process it
- Congestion control prevents sender from overwhelming network
  - ↳ by dynamically adjusting transmission rates to avoid congestion and packet loss
- Flow control TCP Implementation
  - Uses a receiver advertised window (rwnd) sent in TCP header fields
  - rwnd updated dynamically to indicate available buffer space
  - Sender limits data transmission to the size of rwnd
  - So it never exceeds receiver's capacity
  - If receiver's buffer fills (rwnd = 0) sender pauses transmission until receiver sends new rwnd > 0 via ACK
- Congestion Control TCP Implementation
  - Uses congestion window (cwnd) to limit amount of unacknowledged data in flight
  - Employs algorithms:
    - SLOW Start
    - Congestion Avoidance
    - fast Retransmit + Recovery
    - Timeout Handling
- TCP's combined Approach
  - sender's effective window is the minimum of rwnd and cwnd
  - ensures both receiver capacity and network conditions are respected
  - This dual mechanism allows TCP to balance efficiency with fairness in shared networks

## \* Company → 3 groups, each have subnet on the corporate network



What subnet prefix (the smallest one) describes the entire collection of addresses including the routers as identified in the network where they are linked?

$$1.1.1.0 \quad 0000001.0000001.0000001.0000000 \\ 0000001.0000001.0000001.0000010.0000001$$

$$8 + 8 + 5$$

$$1.1.0.0/21$$

Assume the router for group A has 4 ports: port 1 is connected to the group subnet (1.1.1.0/24)

port 2 is connected to router B (IP 1.1.4.0 on A's side)

port 3 is connected to router C (IP 1.1.5.0 on A's side)

port 4 is connected to ISP (rest of the internet)

Forwarding table based on destination IP addresses for each destination, router decides which interface to send the packet out

Group A's subnet (1.1.1.0/24) → send directly out Port 1

Group B's subnet (1.1.2.0/24) → Port 2 next hop 1.1.4.1

Group C's subnet (1.1.3.0/24) → Port 3 → router C's IP 1.1.5.1

Router A's directly connected networks

Port 1: 1.1.1.0/24 (direct)

Port 2: 1.1.4.0/30 (direct)

Port 3: 1.1.5.0/30 (direct)

Port 4: ISP connection (default route)

Other subnets

1.1.2.0/24 (Group B) reachable via Port 2 (next hop 1.1.4.1)

1.1.3.0/24 (Group C) reachable via Port 3 (next hop 1.1.5.1)

1.1.6.0/30 (B-C link) reachable via Port 2 (next hop 1.1.4.1)

1.1.6.0/30 (ISP Gateway) reachable via Port 4

1.1.6.0/30 (ISP Gateway) reachable via Port 4