UDP, TCP, IP multicast

Dan Williams

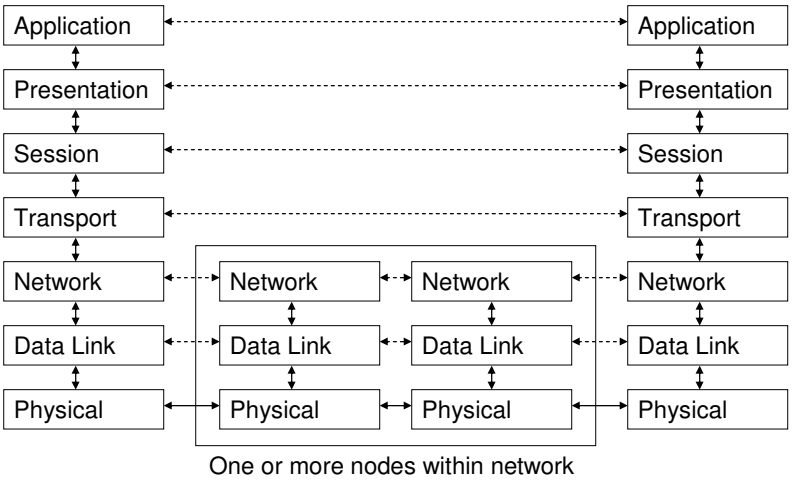
In this lecture

- UDP (user datagram protocol)
  - Unreliable, packet-based
- TCP (transmission control protocol)
  - Reliable, connection oriented, stream-based
- IP multicast

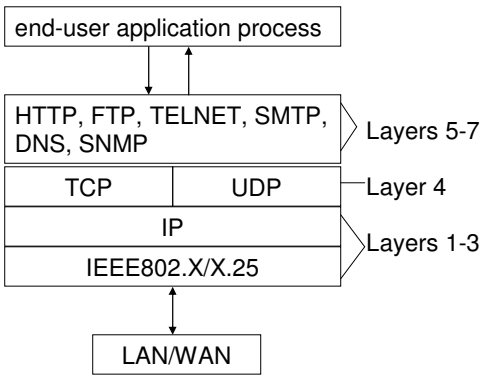
Process-to-Process

- We've talked about host-to-host packet delivery with IP
- Now we look at process-to-process communication
  - UDP (user datagram protocol)
  - TCP (transmission control protocol)
- Transport Layer Protocols
  - OSI model, TCP/IP model

ISO OSI Reference Model



TCP/IP Architecture



What we want from transport layer

- Guaranteed message delivery
- Same order message delivery
- Send arbitrarily large messages
- Allow receiver to apply flow control to sender
- Support multiple application processes on each host
- Network congestion control

Remember Network may

- Drop packets
- Reorder packets
- Deliver duplicate copies of a packet
- Limit packets to some finite size
- Deliver packets after arbitrarily long delay

Note: things sent in the network layer are normally called packets, whereas things sent in the transport layer are normally called messages.

UDP

- Simple – don't add much more than the best effort given by IP
- Demultiplex messages via port numbers
  - port number – number that kernel uses to deliver to appropriate application
  - For instance: HTTP is port 80, SMTP is port 25, Telnet is port 23, DNS is port 53, FTP is port 21
- Ensure correctness of message with checksum
- No connection established, datagram-based

What we get from UDP

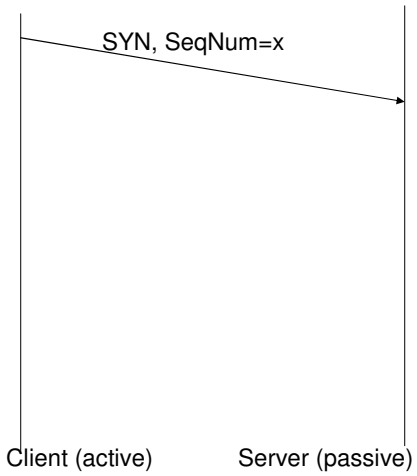
- NO guaranteed message delivery
- NO guaranteed same order message delivery
- NO ability to send arbitrarily large messages
- NO flow control
- Support multiple application processes on each host
- NO network congestion control

TCP

- Stream based protocol allows us to send arbitrarily long messages
- Uses port numbers like UDP to send to multiple processes on each host
- Full duplex protocol
  - Both sides are senders and receivers
- We need each packet to have a sequence number to ensure reliable/in-order delivery

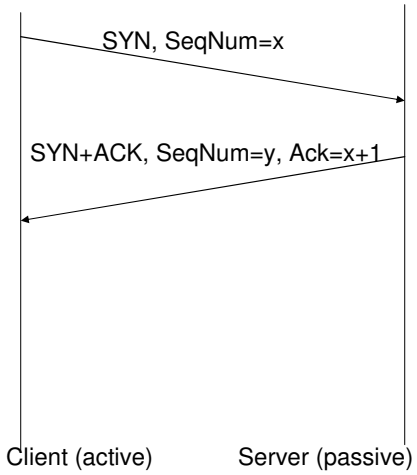
TCP connection establishment

- Three-way handshake
  - SYN sent with initial sequence number of client



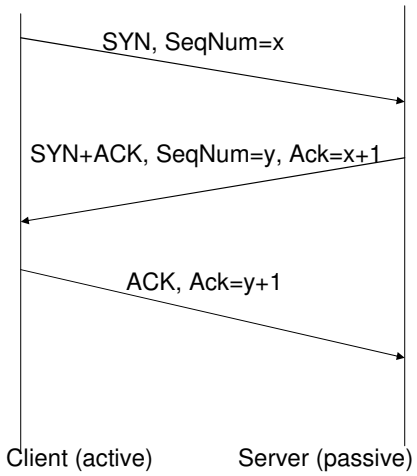
TCP connection establishment

- Three-way handshake
  - SYN sent with initial sequence number of client
  - Server ACKs client's sequence number and sends its own sequence number for SYN



TCP connection establishment

- Three-way handshake
  - SYN sent with initial sequence number of client
  - Server ACKs client's sequence number and sends its own sequence number for SYN
  - Client ACKs server's sequence number
  - Now we are synchronized and ready to communicate



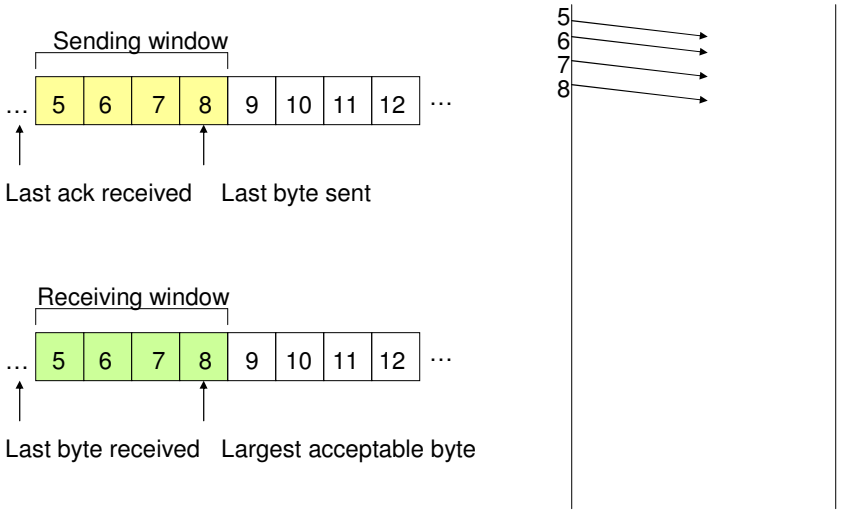
TCP connection termination

- Each side needs to independently close its own half of the connection
- There is a handshake for closing, but I won't describe it here

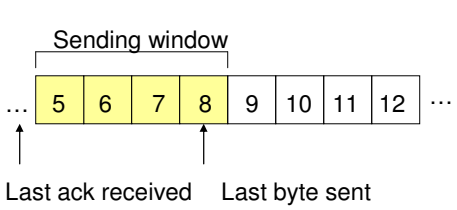
Sliding window gives us

- Guaranteed message delivery
- Same order message delivery
- Send arbitrarily large messages
- Allow receiver to apply flow control to sender
- Support multiple application processes on each host
- Network congestion control

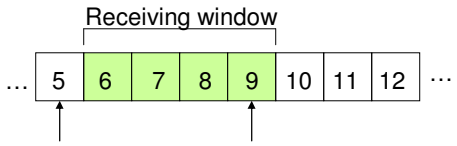
Sliding Window



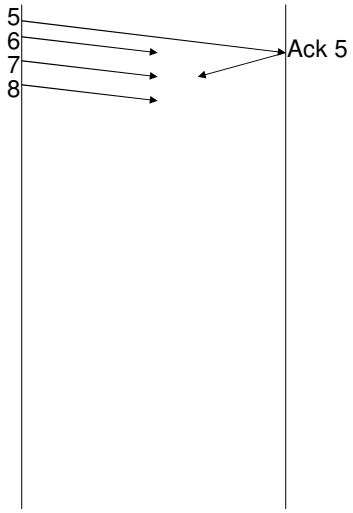
Sliding Window



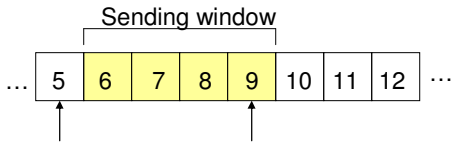
Last ack received Last byte sent



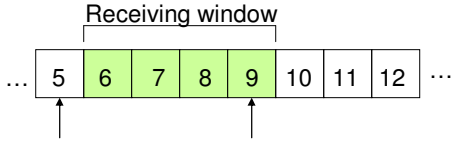
Last byte received Largest acceptable byte



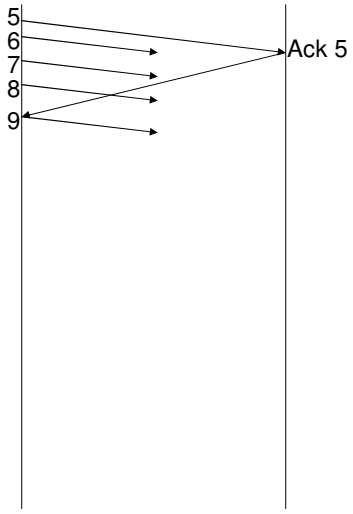
Sliding Window



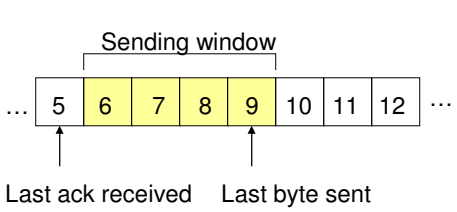
Last ack received Last byte sent

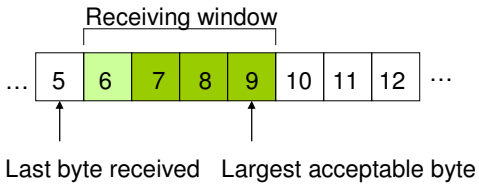


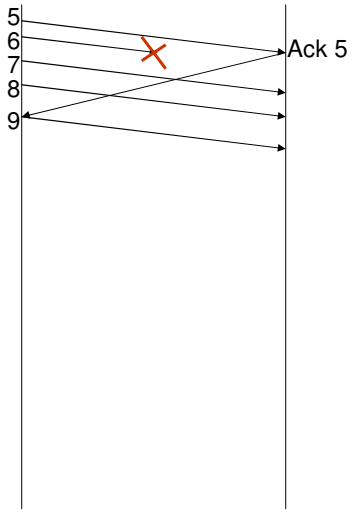
Last byte received Largest acceptable byte



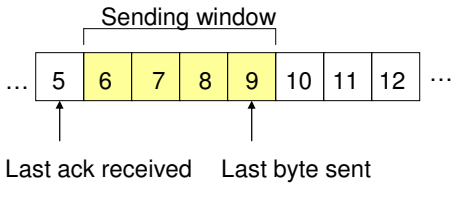
Sliding Window

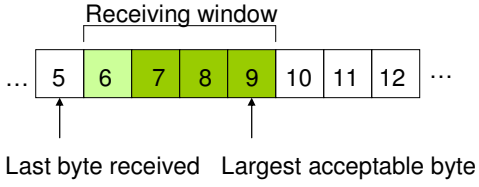


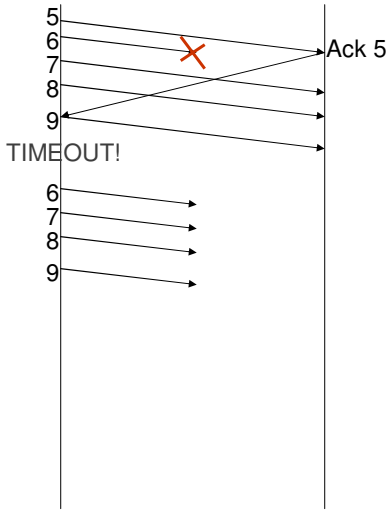




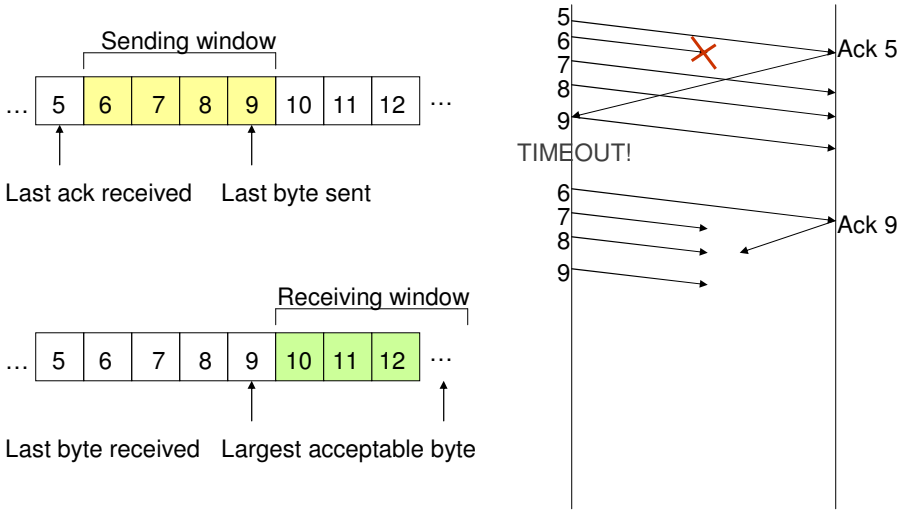
Sliding Window



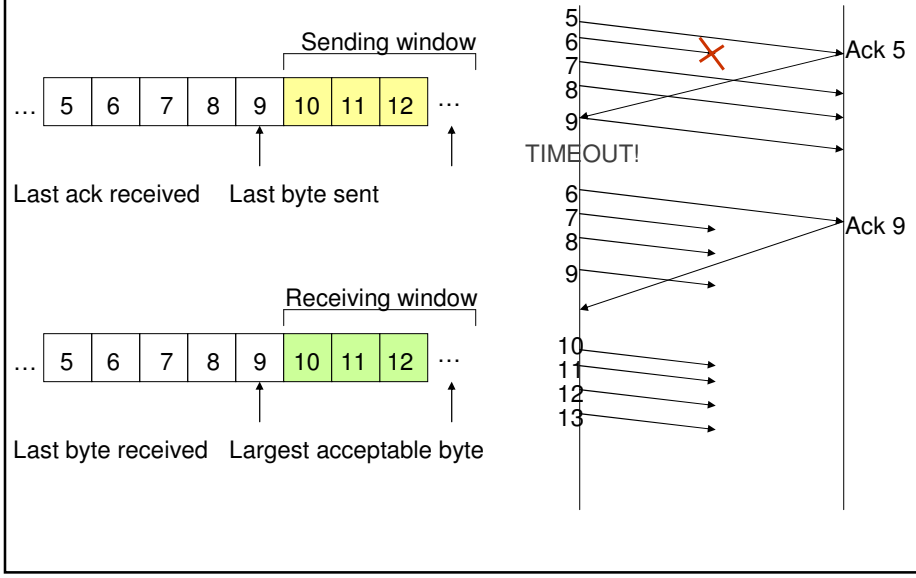




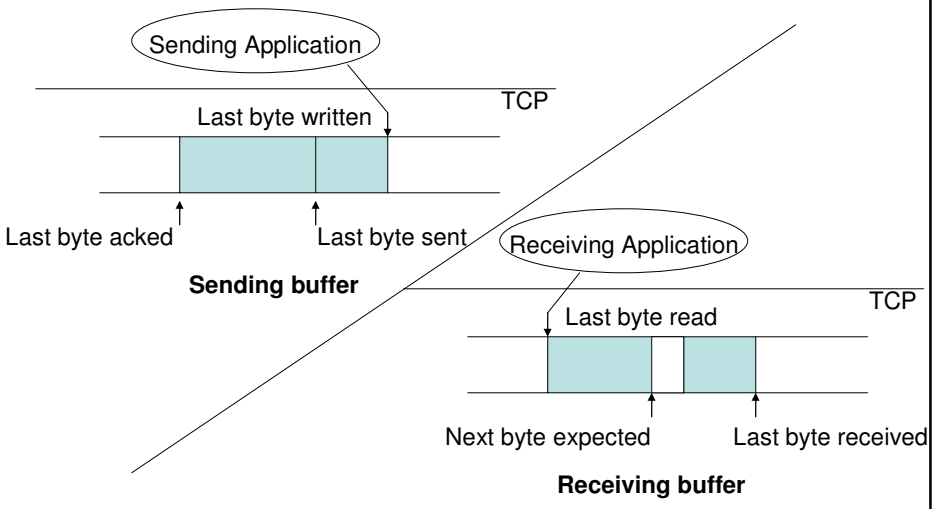
Sliding Window



Sliding Window



Sliding Window with TCP



Using sliding window for flow control

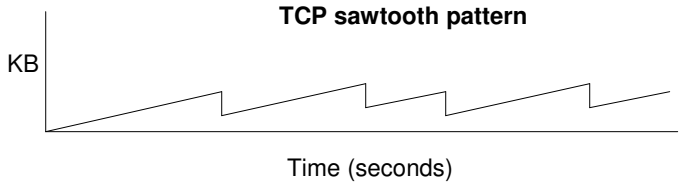
- Flow control:
  - make sure sender doesn't send more than receiver can handle
- Receive window can't slide if receiving process isn't ready to read from buffer
- Send advertised window size with ACK
  - Sender limits send window size based on advertised window size
- If send buffer fills up, TCP blocks sender

What about congestion control?

- Introduced 8 years after TCP/IP had become operational
- Internet suffering from congestion collapse
- Idea:
  - Determine how many packets can be safely transmitted
  - ACK is a signal that a packet is out of network and can safely add another
- How do you find out capacity of network?

Additive Increase Multiplicative Decrease

- Want to find point of congestion in network
- Maintain congestion window size
- TCP send window = min(congest, advert)
- Interpret dropped packets as congestion



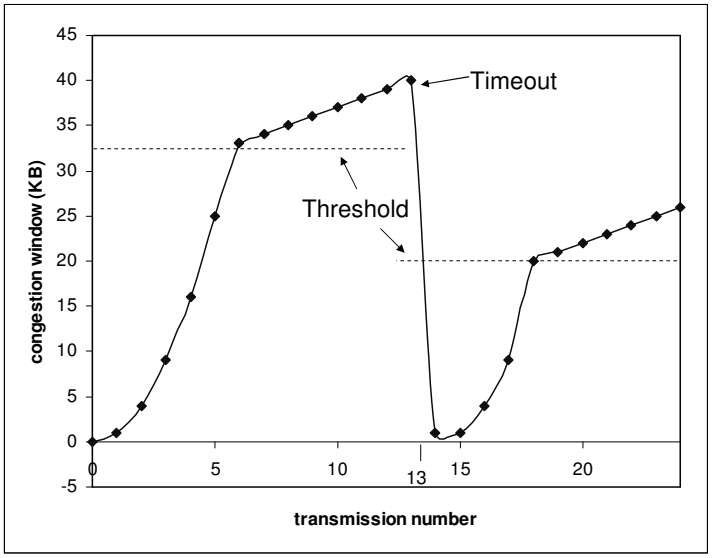
Why decrease aggressively increase conservatively?

- Too large window worse than too small
  - Too large: dropped packets retransmitted making congestion worse
  - Too small: can't send as many packets without receiving an ACK
- What about at the start of connection?
  - Takes too long to get a large window

Slow Start

- Double congestion window after every ACK until we get a loss or threshold is reached
- Threshold = half of the congestion window at which a packet was dropped
- Then do additive increase
- Called "slow start" because it starts slower than TCP without congestion control

Slow Start congestion window



Now we can see how RED works

- Recall RED (Random Early Detection)
  - Router monitors its own queue length
  - Router implicitly notifies sender of congestion by randomly dropping a packet
- TCP will note this as congestion and multiplicatively decrease congestion window

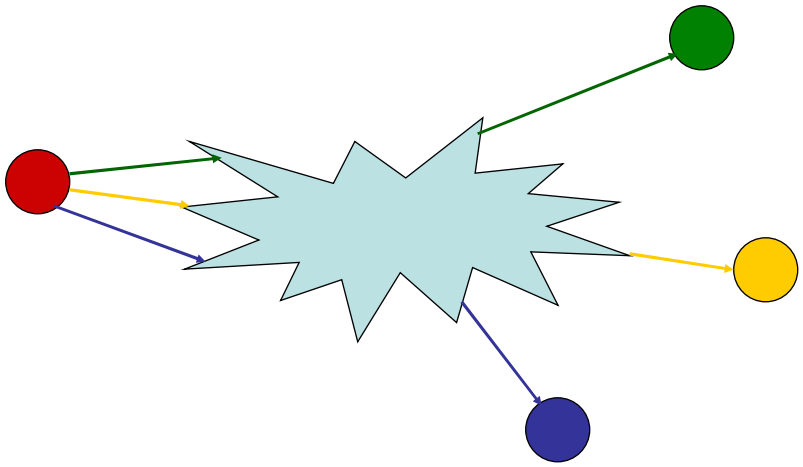
What we get from TCP

- Guaranteed message delivery
- Same order message delivery
- Send arbitrarily large messages
- Allow receiver to apply flow control to sender
- Support multiple application processes on each host
- Network congestion control

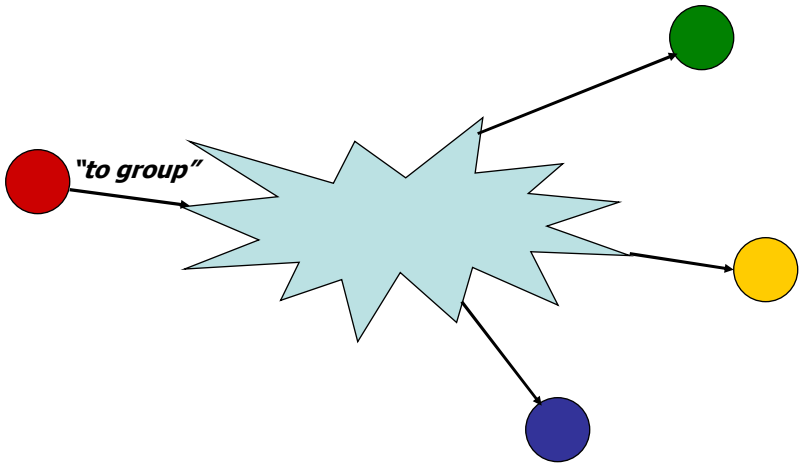
IP multicast

- Host may want to send a packet to multiple hosts
- Send to a group, not individual hosts
  - Reduces overhead for sender
  - Reduces bandwidth consumption in network
  - Reduces latency in receiver (all see packet at same time)

Unicast to multiple hosts



Multicast to multiple hosts



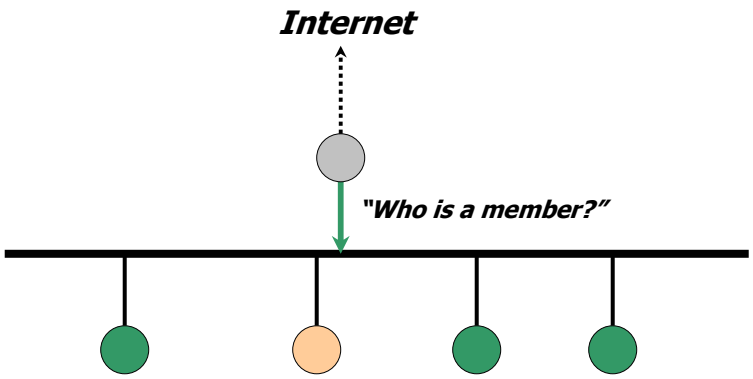
Multicast Group

- Idea:
  - Send to group that receivers may join
  - Group has specially assigned address
- Class D IP addresses for group
  - 224.0.0.0 to 239.255.255.255
- How does a host join a group?
  - IGMP (Internet Group Management Protocol)

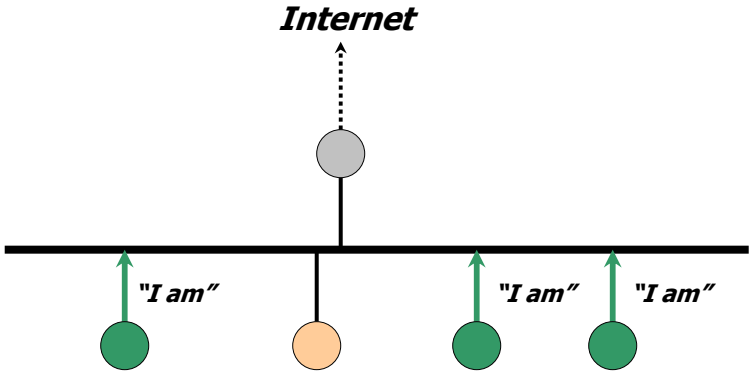
IGMP

- Detects if a multicast group has any members within a LAN
- *Query* and *report* messages
  - router sends query of group membership periodically
  - hosts report groups they're in

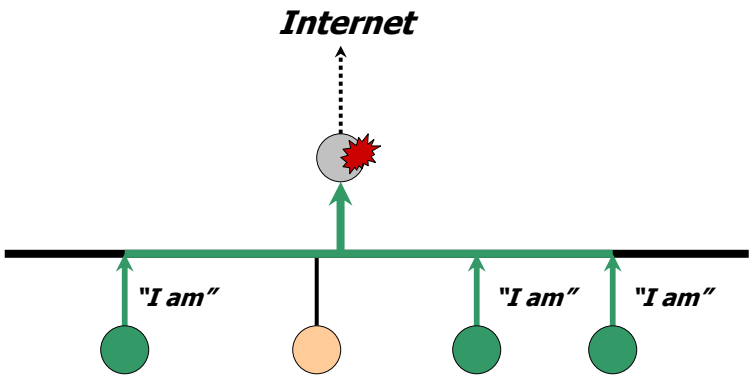
IGMP



IGMP



IGMP



Avoiding overloading

- Report packets may overload router
  - upon getting a query, each group member sets a timer
  - if it sees a report for its group before the timer expires, it suppresses its report
  - otherwise reports on expiration

Multicast over Ethernet

- Trivial
- Ethernet is a broadcast medium – every host hears every packet
- Simply need to receive packets sent to group address
- Want to extend this between LANs connected by routers

Naïve Approach

- We want to send multicasts everywhere where there are group members
  - use *flooding* to send multicast between routers
  - when we get to a LAN, use regular (Ethernet) multicast

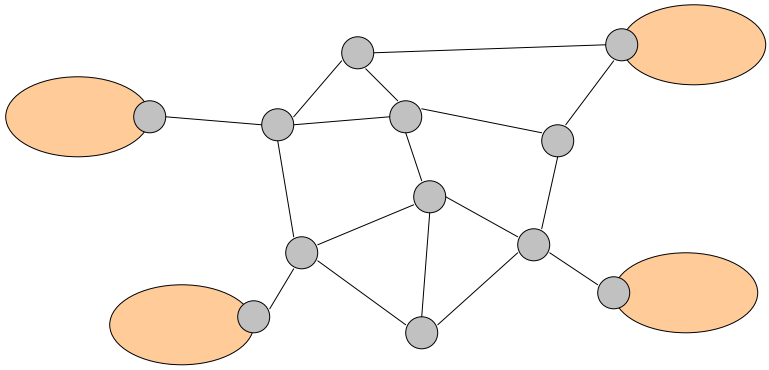
Flooding not a good solution

- Not scalable
- LANs which do not have any group members will still receive multicast
- Need to know which LANs want to receive multicast message
  - Implement in routers
  - Create spanning tree

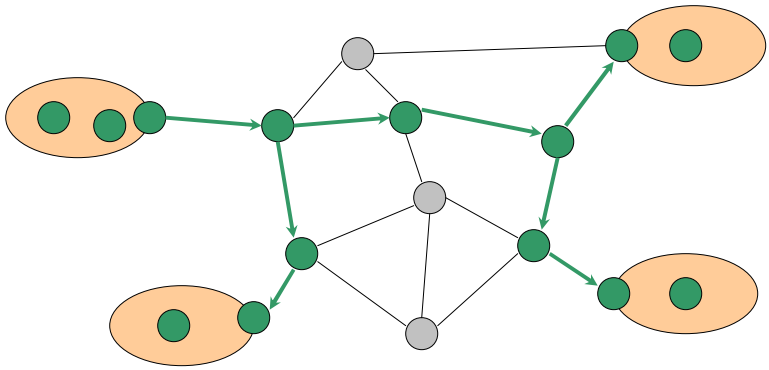
Multicast trees

- Shortest-path tree to all multicast members, rooted at sender
- But must be computed independently by each router
- And must be dynamically adjusted for joins and leaves

A multicast tree



A multicast tree



THE END