CSE/EE 461 Lecture 21
Multicast and QoS

Tom Anderson

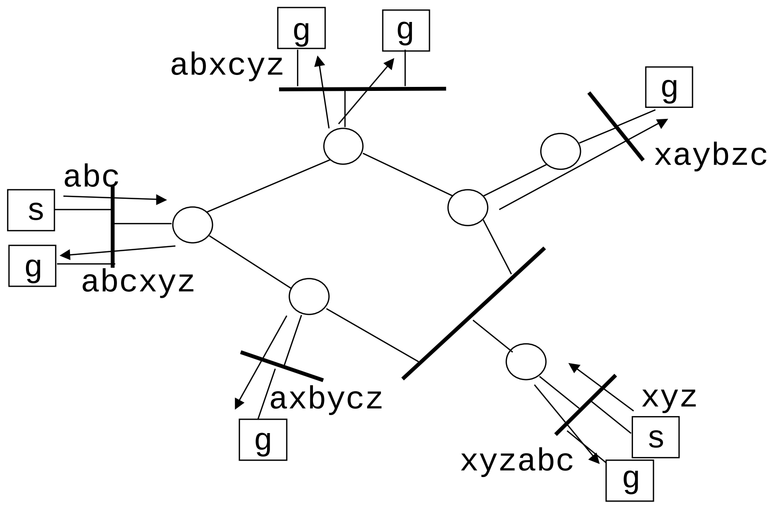
tom@cs.washington.edu

Peterson, Chapter 4.4, 6.5

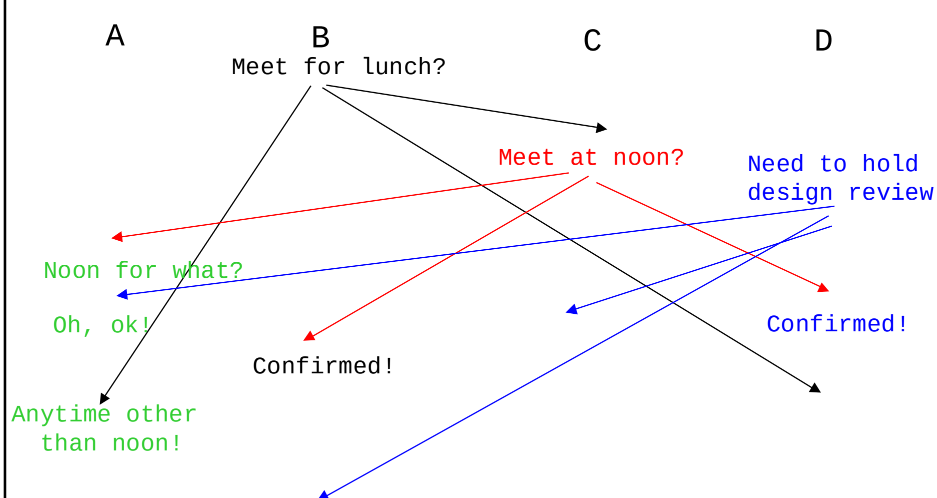
Multicast Packet Ordering

- Easy to order unicast packets => seq #s
- Easy to order multicast packets from a single source => seq #s
- What if multiple sources?
  - Packets can arrive in different order at different receivers
  - Is this bad?
  - If so, what can we do to fix it?

Multicast Ordering Example



Example: Email Groups



Multicast Total Ordering

- All packets are delivered in same order everywhere
- Single seq # for all packets to group
  - every source sends packets to arbiter
  - arbiter assigns sequence #
  - if arbiter fails, elect new one
  - receivers don't process packets out of order

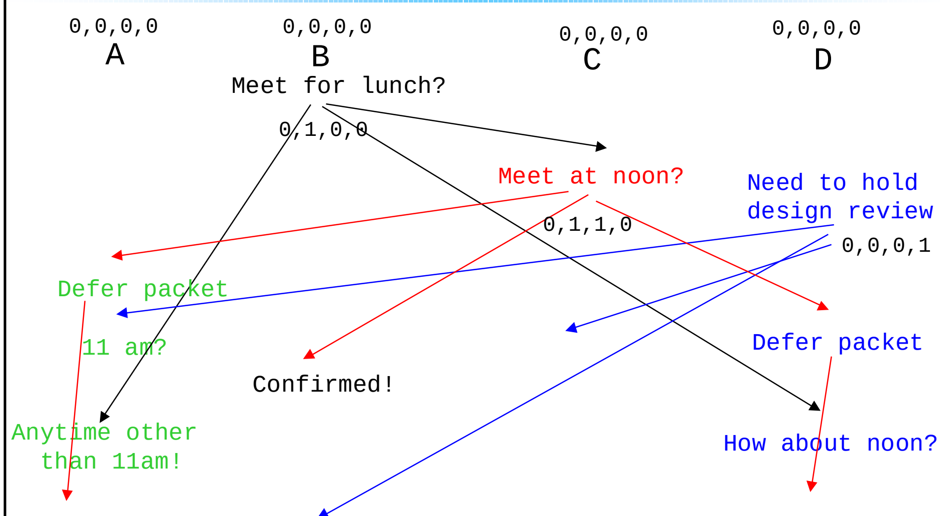
Multicast Causal Ordering

- Total ordering inefficient for subcasts
- Instead, causal ordering
  - packets are never delivered before packets that could have "caused" them
    - receiver must have gotten all the packets source has seen
  - packets that originate concurrently can be delivered in any order

Implementing Causal Ordering

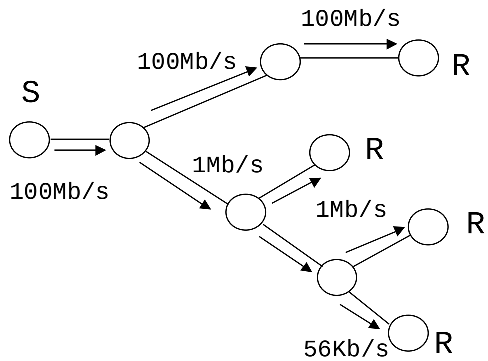
- Packets carry per-host sequence #
  - increment on each send
- Each host maintains a "version vector"
  - max seq #'s seen (in order) from each host
  - put version vector in each outgoing packet
- At receiver, delay packet until host vector > packet vector, for all sources

Causal Ordering Example



Multicast Congestion Control

- What if receivers have very different bandwidths?
- Send at max?
- Send at min?
- Send at avg?



Layered Multimedia

- Transmit signal at multiple granularities
  - 56Kb/s - voice only
  - 1Mb/s - choppy video
  - 100Mb/s - high quality video
- Layers can be
  - independent (redundant)
  - dependent (progressive refinement)

Receiver-Driven Layered Multicast

- Each layer a separate group
  - receiver subscribes to max group that will get through with minimal drops
- Dynamically adapt to available capacity
  - use packet losses as congestion signal
- Assume no special router support
  - packets dropped independently of layer

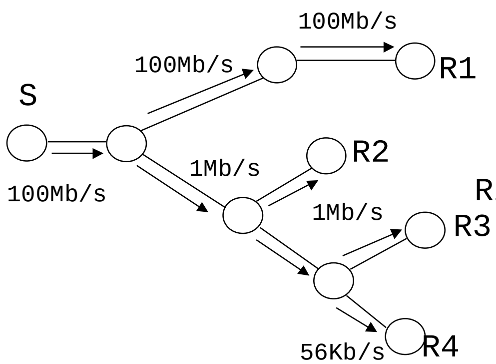
How does receiver know which layers to add?

- System dynamically adapts to available capacity
  - Use packet drops as congestion signal
  - No drops => try subscribing to higher layer
  - Drops => unsubscribe to layer
- Alternative: ask the user

RLM Join

- Periodically, receivers try subscribing to higher layer
- If enough capacity, no congestion, no drops => keep layer (& try next layer)
- If not enough capacity, congestion, drops => drop layer (& increase time to next retry)
- Coordination between receivers
  - use random delay and broadcast that join is in progress, so that others don't try at the same time
  - shared learning -> if neighbor join fails, wait longer to try yourself

RLM Join Example



R1 joins layer 1,
joins layer 2
joins layer 3

R2,R3 join layer 1,
join layer 2
fails at layer 3

R4 joins layer 1,
fails at layer 2

Drop Policies for Layered Multicast

- Priority
  - prioritize low bandwidth layers
    - drop packets for higher layers
    - ex: everyone still gets audio, even if video degrades
  - requires router support for priorities
- Uniform (e.g., drop tail, RED)
  - packets arriving at congested router are dropped regardless of their layer
- Which is better?

Intuition vs. Practice

- Intuition: priorities should be better
  - priority drops are less wasteful; always get something useful through
- However, with RLM, uniform has
  - better incentives to well-behaved users
    - if oversend, performance rapidly degrades
  - clearer congestion signal
    - allows shared learning

Multicast Summary

- Multicast needed for efficiency, group coordination
- Need to revisit all aspects of networking
  - Routing
  - Administration
  - Reliable delivery
  - Ordered delivery
  - Congestion control

Quality of Service

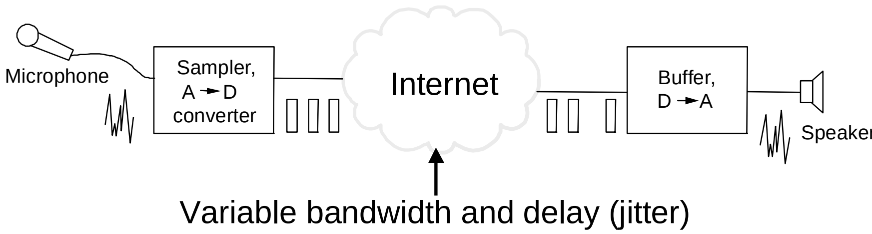
- What kinds of service do different applications need?
  - Web is built on top of “best-effort” service
  - Other applications may need more
    - Internet telephone service (voice over IP)
    - streaming audio/video
    - real-time games
    - remote controlled robotic surgery
- What mechanisms do we need to support these more demanding applications?
  - as with multicast, will need network to do more

IP Best Effort Service

- Our network model so far:
  - IP at routers: a shared, first come first serve (drop tail) queue
  - TCP at hosts: probes for available bandwidth, causing loss
- Router/host behavior determines the kind of service applications will receive
  - TCP *causes* loss, along with variable delay, variable bandwidth

An Audio Example

- Playback is a real-time service
  - audio must be received by a deadline to be useful

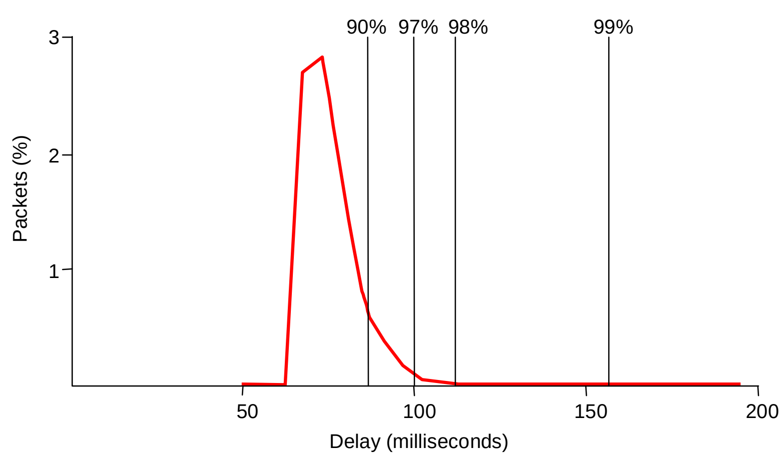


- Real-time applications need assurances from the network
  - What assurances does playback require?

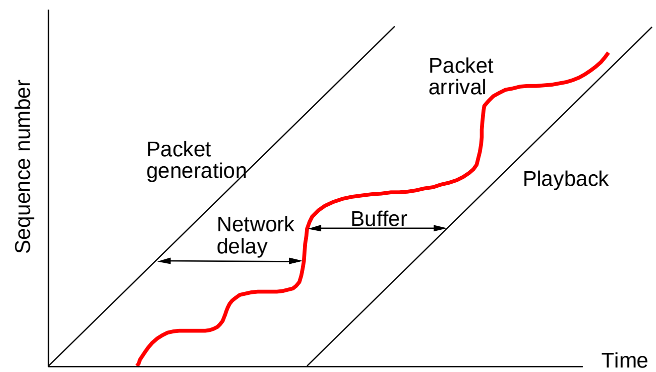
Network Support for Playback

- Bandwidth
  - There must be enough on average
  - But we can tolerate to short term fluctuations
- Delay
  - Ideally it would be fixed
  - But we can tolerate some variation (jitter)
- Loss
  - Ideally there would be none
  - But we can tolerate some losses

Example: Delay and Jitter

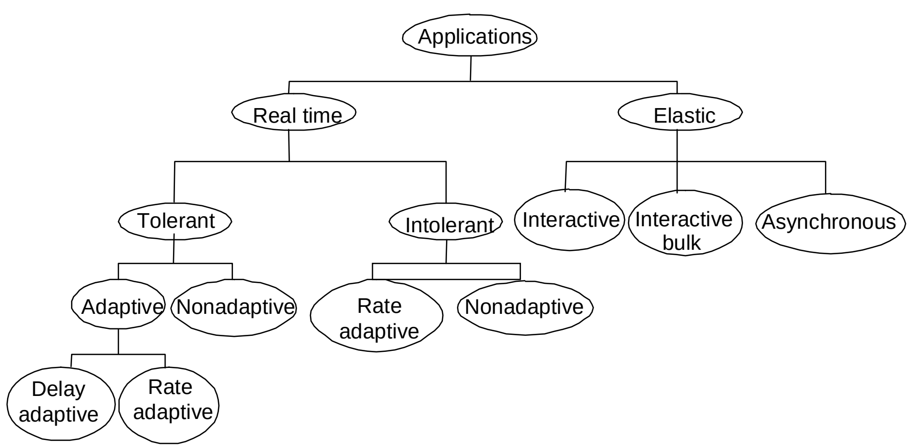


Tolerating Jitter with Buffering



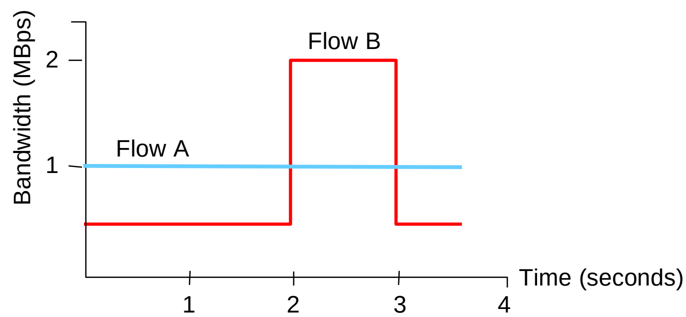
- Insert variable delay before playout to give time for late samples to arrive

Taxonomy of Applications



Specifying Bandwidth Needs

- Problem: Many applications have variable demands



- Same average bandwidth, but very different needs over time
  - how do we describe bandwidth to the network?

Token Buckets

- Simple model
  - reflects both average, variability over time

- Use tokens to send bits
- Avg bandwidth is R bps
- Maximum burst is B bits