QoS (Quality of Service)
Chapter 6
Multimedia Networking

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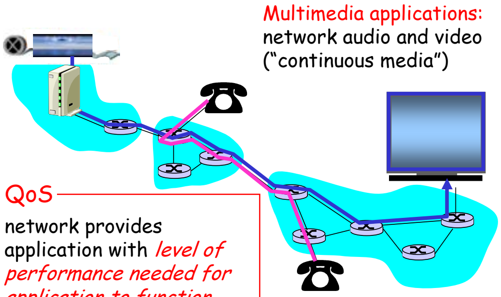
Thanks and enjoy! JFK / KWR

Computer Networking: A Top Down Approach Featuring the Internet,
2nd edition.
Jim Kurose, Keith Ross
Addison-Wesley, July 2002.

Multimedia, Quality of Service: What is it?

Multimedia applications: network audio and video ("continuous media")



QoS

network provides application with *level of performance needed for application to function.*

MM Networking Applications

Classes of MM applications:

1) Streaming stored audio and video

2) Streaming live audio and video

3) Real-time interactive audio and video

Fundamental characteristics:

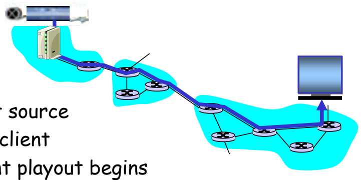
❑ Typically **delay sensitive**
  - ❍ end-to-end delay
  - ❍ delay jitter

❑ But **loss tolerant**: infrequent losses cause minor glitches

❑ Antithesis of data, which are loss intolerant but delay tolerant.

Jitter is the variability of packet delays within the same packet stream

Streaming Stored Multimedia



Streaming:

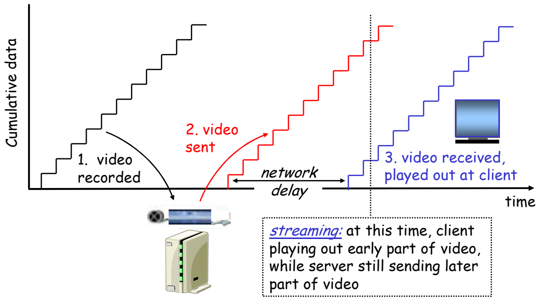
❑ media stored at source

❑ transmitted to client

❑ streaming: client playout begins *before* all data has arrived

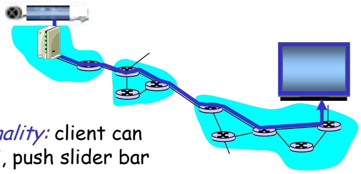
  ❑ timing constraint for still-to-be transmitted data: in time for playout

Streaming Stored Multimedia: What is it?



streaming: at this time, client playing out early part of video, while server still sending later part of video

Streaming Stored Multimedia: Interactivity



❑ *VCR-like functionality:* client can pause, rewind, FF, push slider bar
  - ❍ 10 sec initial delay OK
  - ❍ 1-2 sec until command effect OK
  - ❍ RTSP often used (more later)

❑ timing constraint for still-to-be transmitted data: in time for playout

Streaming Live Multimedia

Examples:

- Internet radio talk show
- Live sporting event

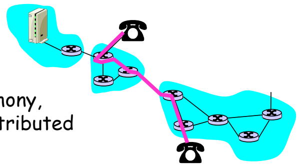
Streaming

- playback buffer
- playback can lag tens of seconds after transmission
- still have timing constraint

Interactivity

- fast forward impossible
- rewind, pause possible!

Interactive, Real-Time Multimedia



- applications: IP telephony, video conference, distributed interactive worlds
- end-end delay requirements:
  - audio: < 150 msec good, < 400 msec OK
    - includes application-level (packetization) and network delays
    - higher delays noticeable, impair interactivity
- session initialization
  - how does callee advertise its IP address, port number, encoding algorithms?

Multimedia Over Today's Internet

TCP/UDP/IP: "best-effort service"

- *no* guarantees on delay, loss

? ? ? ? ? ? But you said multimedia apps requires ? QoS and level of performance to be ? ? effective! ? ?

Today's Internet multimedia applications use application-level techniques to mitigate (as best possible) effects of delay, loss

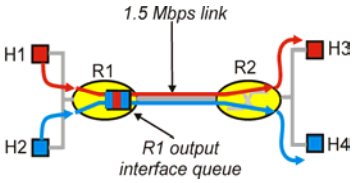
Improving QOS in IP Networks

Thus far: "making the best of best effort"

Future: next generation Internet with QoS guarantees

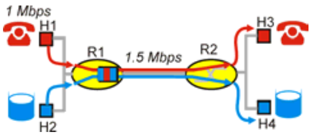
- RSVP: signaling for resource reservations
- Differentiated Services: differential guarantees
- Integrated Services: firm guarantees

- simple model for sharing and congestion studies:



Principles for QOS Guarantees

- Example: 1MbpsI P phone, FTP share 1.5 Mbps link.
  - bursts of FTP can congest router, cause audio loss
  - want to give priority to audio over FTP

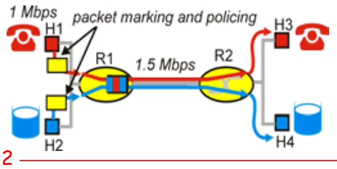


Principle 1

packet marking needed for router to distinguish between different classes; and new router policy to treat packets accordingly

Principles for QOS Guarantees (more)

- what if applications misbehave (audio sends higher than declared rate)
  - policing: force source adherence to bandwidth allocations
- marking and policing at network edge:
  - similar to ATM UNI (User Network Interface)

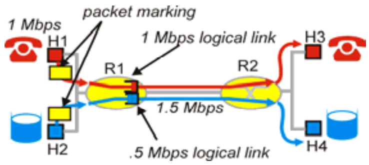


Principle 2

provide protection (*isolation*) for one class from others

Principles for QOS Guarantees (more)

- Allocating *fixed* (non-sharable) bandwidth to flow: *inefficient* use of bandwidth if flows doesn't use its allocation

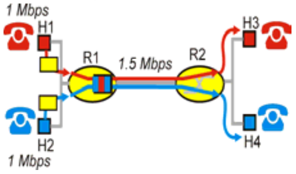


Principle 3

While providing isolation, it is desirable to use resources as efficiently as possible

Principles for QOS Guarantees (more)

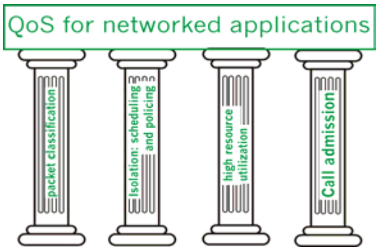
- *Basic fact of life:* can not support traffic demands beyond link capacity



Principle 4

Call Admission: flow declares its needs, network may block call (e.g., busy signal) if it cannot meet needs

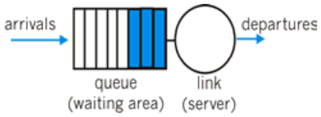
Summary of QoS Principles


QoS for networked applications
packet classification
Isolation: scheduling and policing
high resource utilization
Call admission

Let's next look at mechanisms for achieving this ….

Scheduling And Policing Mechanisms

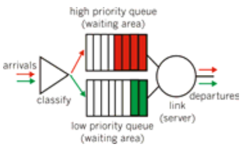
- scheduling: choose next packet to send on link
- FIFO (first in first out) scheduling: send in order of arrival to queue
  - real-world example?
  - discard policy: if packet arrives to full queue: who to discard?
    - Tail drop: drop arriving packet
    - priority: drop/remove on priority basis
    - random: drop/remove randomly

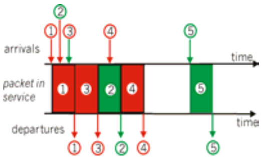


Scheduling Policies: more

Priority scheduling: transmit highest priority queued packet

- multiple *classes*, with different priorities
  - class may depend on marking or other header info, e.g. IP source/dest, port numbers, etc..
  - Real world example?

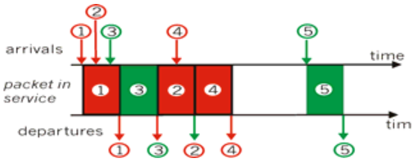




Scheduling Policies: still more

round robin scheduling:

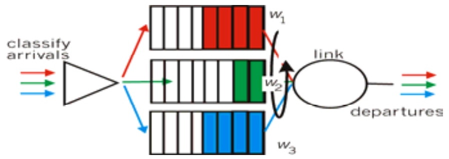
- multiple classes
- cyclically scan class queues, serving one from each class (if available)
- real world example?



Scheduling Policies: still more

Weighted Fair Queuing:

- generalized Round Robin
- each class gets weighted amount of service in each cycle
- real-world example?



Policing Mechanisms

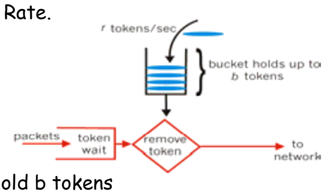
Goal: limit traffic to not exceed declared parameters

Three common-used criteria:

- *(Long term) Average Rate:* how many pkts can be sent per unit time (in the long run)
  - crucial question: what is the interval length: 100 packets per sec or 6000 packets per min have same average!
- *Peak Rate:* e.g., 6000 pkts per min. (ppm) avg.; 1500 ppm peak rate
- *(Max.) Burst Size:* max. number of pkts sent consecutively (with no intervening idle)

Policing Mechanisms

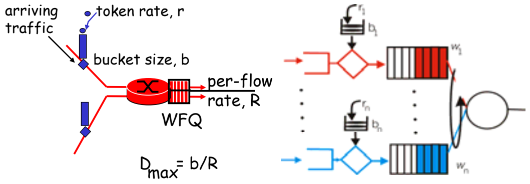
Token Bucket: limit input to specified Burst Size and Average Rate.



- bucket can hold b tokens
- tokens generated at rate *r token/sec* unless bucket full
- *over interval of length t: number of packets admitted less than or equal to (r t + b).*

Policing Mechanisms (more)

- token bucket, WFQ combine to provide guaranteed upper bound on delay, i.e., *QoS guarantee*!



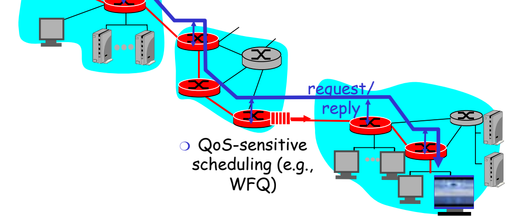
IETF Integrated Services

- architecture for providing QOS guarantees in IP networks for individual application sessions
- resource reservation: routers maintain state info (a la VC) of allocated resources, QoS req's
- admit/deny new call setup requests:

Question: can newly arriving flow be admitted with performance guarantees while not violated QoS guarantees made to already admitted flows?

Intserv: QoS guarantee scenario

- Resource reservation
  - call setup, signaling (RSVP)
  - traffic, QoS declaration
  - per-element admission control



- QoS-sensitive scheduling (e.g., WFQ)

Call Admission

Arriving session must :

- declare its QOS requirement
  - R-spec: defines the QOS being requested
- characterize traffic it will send into network
  - T-spec: defines traffic characteristics
- signaling protocol: needed to carry R-spec and T-spec to routers (where reservation is required)
  - RSVP

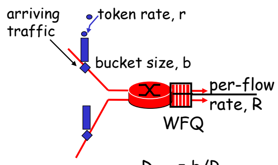
Intserv QoS: Service models [rfc2211, rfc 2212]

Guaranteed service:

- worst case traffic arrival: leaky-bucket-policed source
- simple (mathematically provable) *bound* on delay [Parekh 1992, Cruz 1988]

Controlled load service:

- "a quality of service closely approximating the QoS that same flow would receive from an unloaded network element."



$D_{max} = b/R$

IETF Differentiated Services

Concerns with Intserv:

- Scalability: signaling, maintaining per-flow router state difficult with large number of flows
- Flexible Service Models: Intserv has only two classes. Also want "qualitative" service classes
  - "behaves like a wire"
  - relative service distinction: Platinum, Gold, Silver

Diffserv approach:

- simple functions in network core, relatively complex functions at edge routers (or hosts)
- Do't define define service classes, provide functional components to build service classes

Diffserv Architecture

Edge router:

- per-flow traffic management

- marks packets as in-profile and out-profile

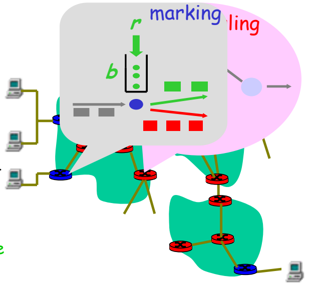
Core router:

- per class traffic management

- buffering and scheduling based on marking at edge

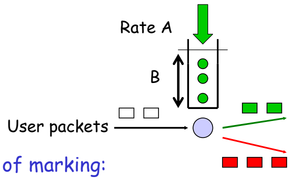
- preference given to in-profile packets

- Assured Forwarding



Edge-router Packet Marking

- profile: pre-negotiated rate A, bucket size B
- packet marking at edge based on per-flow profile

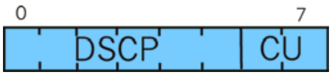


Possible usage of marking:

- class-based marking: packets of different classes marked differently
- intra-class marking: conforming portion of flow marked differently than non-conforming one

Classification and Conditioning

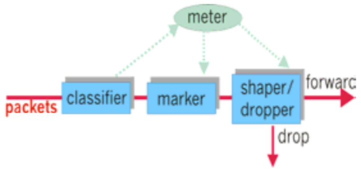
- Packet is marked in the Type of Service (TOS) in IPv4, and Traffic Class in IPv6
- 6 bits used for Differentiated Service Code Point (DSCP) and determine PHB that the packet will receive
- 2 bits are currently unused



Classification and Conditioning

may be desirable to limit traffic injection rate of some class:

- user declares traffic profile (eg, rate, burst size)
- traffic metered, shaped if non-conforming



MPLS (MultiProtocol Label Switching)

- Defined in RFC 3031
- Different forwarding method
- Similar to VCs (ATM, FR, etc.)
- Called also label switching, tag switching
- New message header (a.k.a tag):
  - 20 bits label
  - 3 bits QoS
  - 1 bit S (stacking)
  - 8 bits TTL
- Supports aggregation of tags called FEC (forwarding equivalence class)

MPLS – VC construction

- Data driven approach: VC is initialized on demand in a recursive way
  - Possible problem of loops
- Control driven approach: when router boots it creates labels for the destination in its routing tables

Signaling in the Internet

connectionless (stateless) forwarding by IP routers + best effort service = no network signaling protocols in initial IP design

- New requirement: reserve resources along end-to-end path (end system, routers) for QoS for multimedia applications
- RSVP: Resource Reservation Protocol [RFC 2205]
  - " … allow users to communicate requirements to network in robust and efficient way." i.e., signaling !
- earlier Internet Signaling protocol: ST-II [RFC 1819]

RSVP Design Goals

1. accommodate heterogeneous receivers (different bandwidth along paths)
2. accommodate different applications with different resource requirements
3. make multicast a first class service, with adaptation to multicast group membership
4. leverage existing multicast/unicast routing, with adaptation to changes in underlying unicast, multicast routes
5. control protocol overhead to grow (at worst) linear in # receivers
6. modular design for heterogeneous underlying technologies

RSVP: does not…

- specify how resources are to be reserved
  - rather: a mechanism for communicating needs
- determine routes packets will take
  - that's the job of routing protocols
  - signaling decoupled from routing
- interact with forwarding of packets
  - separation of control (signaling) and data (forwarding) planes

RSVP: overview of operation

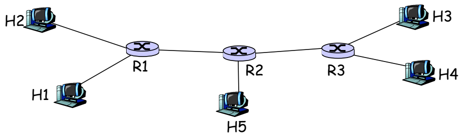
- senders, receiver join a multicast group
  - done outside of RSVP
  - senders need not join group
- sender-to-network signaling
  - *path message:* make sender presence known to routers
  - path teardown: delete sender's path state from routers
- receiver-to-network signaling
  - *reservation message:* reserve resources from sender(s) to receiver
  - reservation teardown: remove receiver reservations
- network-to-end-system signaling
  - path error
  - reservation error

Path msgs: RSVP *sender-to-network* signaling

- path message contents:
  - *address:* unicast destination, or multicast group
  - *flowspec:* bandwidth requirements spec.
  - *filter flag:* if yes, record identities of upstream senders (to allow packets filtering by source)
  - *previous hop:* upstream router/host ID
  - *refresh time:* time until this info times out
- path message: communicates sender info, and reverse-path-to-sender routing info
  - later upstream forwarding of receiver reservations

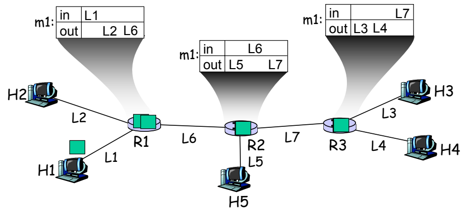
RSVP: simple audio conference

- H1, H2, H3, H4, H5 both senders and receivers
- multicast group m1
- no filtering: packets from any sender forwarded
- audio rate: *b*
- only one multicast routing tree possible



RSVP: building up path state

- H1, …, H5 all send path messages on *m1:*
  (address=*m1*, Tspec=*b*, filter-spec=no-filter,refresh=100)
- Suppose H1 sends first path message



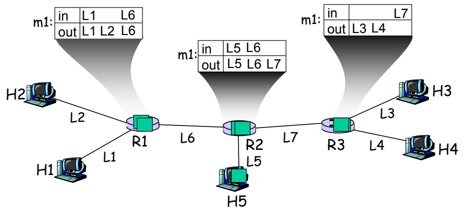
m1:	
in	L1
out	L2 L6

m1:	
in	L6
out	L5 L7

m1:	
in	L7
out	L3 L4

RSVP: building up path state

- next, H5 sends path message, creating more state in routers



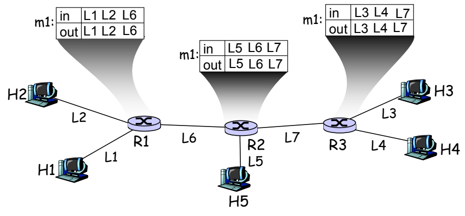
m1:	
in	L1 L6
out	L1 L2 L6

m1:	
in	L5 L6
out	L5 L6 L7

m1:	
in	L7
out	L3 L4

RSVP: building up path state

- H2, H3, H5 send path msgs, completing path state tables



m1:	
in	L1 L2 L6
out	L1 L2 L6

m1:	
in	L5 L6 L7
out	L5 L6 L7

m1:	
in	L3 L4 L7
out	L3 L4 L7

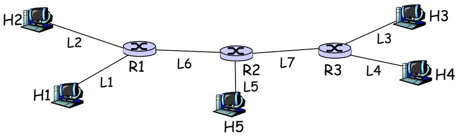
reservation msgs: *receiver-to-network* signaling

- reservation message contents:
  - *desired bandwidth:*
  - *filter type:*
    - no filter: any packets address to multicast group can use reservation
    - fixed filter: only packets from specific set of senders can use reservation
    - dynamic filter: senders who's packets can be forwarded across link will change (by receiver choice) over time.
  - *filter spec*
- reservations flow upstream from receiver-to-senders, reserving resources, creating additional, *receiver-related* state at routers

RSVP: *receiver* reservation example 1

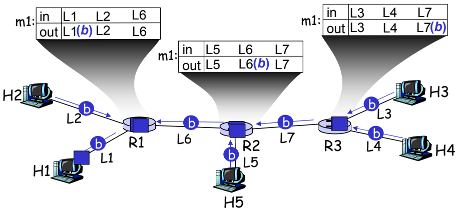
H1 wants to receive audio from all other senders

- H1 reservation msg flows uptree to sources
- H1 only reserves enough bandwidth for 1 audio stream
- reservation is of type "no filter" – any sender can use reserved bandwidth



RSVP: *receiver* reservation example 1

- H1 reservation msgs flows uptree to sources
- routers, hosts reserve bandwidth b needed on downstream links towards H1



m1:

in	L1	L2	L6
out	L1(*b*)	L2	L6

m1:

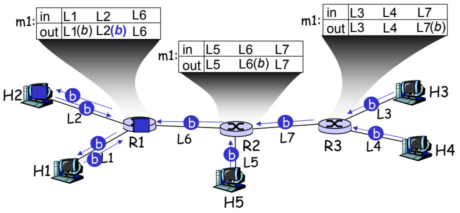
in	L5	L6	L7
out	L5	L6(*b*)	L7

m1:

in	L3	L4	L7
out	L3	L4	L7(*b*)

RSVP: *receiver* reservation example 1 (more)

- next, H2 makes no-filter reservation for bandwidth *b*
- H2 forwards to R1, R1 forwards to H1 and R2 (?)
- R2 takes no action, since *b* already reserved on L6



m1:

in	L1	L2	L6
out	L1(*b*)	L2(*b*)	L6

m1:

in	L5	L6	L7
out	L5	L6(*b*)	L7

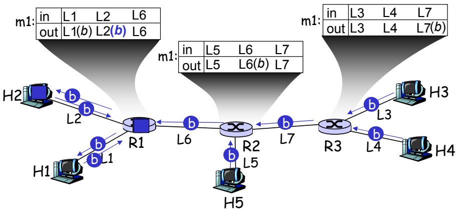
m1:

in	L3	L4	L7
out	L3	L4	L7(*b*)

RSVP: *receiver* reservation: issues

What if multiple senders (e.g., H3, H4, H5) over link (e.g., L6)?

- arbitrary interleaving of packets
- L6 flow policed by leaky bucket: if H3+H4+H5 sending rate exceeds b, packet loss will occur



m1:

in	L1	L2	L6
out	L1(*b*)	L2(*b*)	L6

m1:

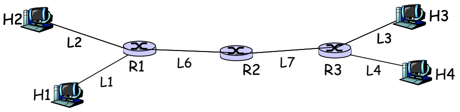
in	L5	L6	L7
out	L5	L6(*b*)	L7

m1:

in	L3	L4	L7
out	L3	L4	L7(*b*)

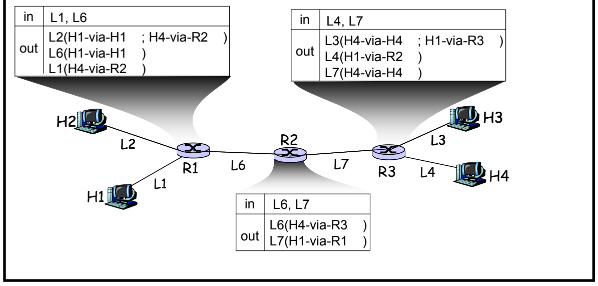
RSVP: example 2

- H1, H4 are only senders
  - send *path messages* as before, indicating filtered reservation
  - Routers store upstream senders for each upstream link
- H2 will want to receive from H4 (only)



RSVP: example 2

- H1, H4 are only senders
  - send *path messages* as before, indicating filtered reservation



in	L1, L6

out	L2(H1-via-H1 ; H4-via-R2)
	L6(H1-via-H1)
	L1(H4-via-R2)

in	L4, L7

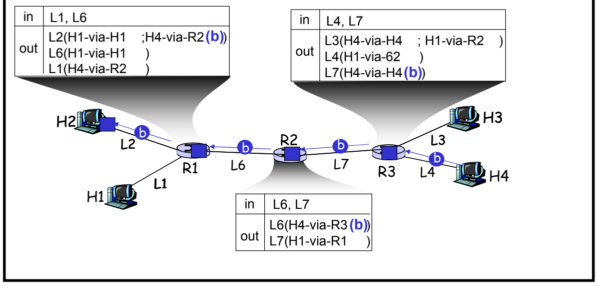
out	L3(H4-via-H4 ; H1-via-R3)
	L4(H1-via-R2)
	L7(H4-via-H4)

in	L6, L7

out	L6(H4-via-R3)
	L7(H1-via-R1)

RSVP: example 2

- receiver H2 sends reservation message for source H4 at bandwidth *b*
  - propagated upstream towards H4, reserving *b*



in	L1, L6

out	L2(H1-via-H1 ;H4-via-R2 (b))
	L6(H1-via-H1)
	L1(H4-via-R2)

in	L4, L7

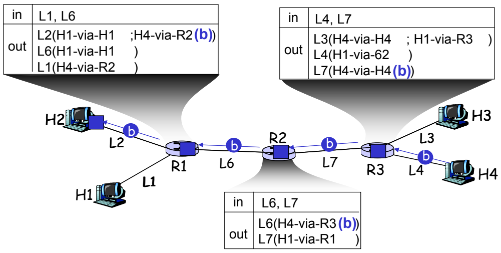
out	L3(H4-via-H4 ; H1-via-R2)
	L4(H1-via-62)
	L7(H4-via-H4 (b))

in	L6, L7

out	L6(H4-via-R3 (b))
	L7(H1-via-R1)

RSVP: *soft-state*

- senders periodically resend path msgs to refresh (maintain) state
- receivers periodically resend resv msgs to refresh (maintain) state
- path and resv msgs have TTL field, specifying refresh interval



in	L1, L6

out	L2(H1-via-H1 ;H4-via-R2 (b))
	L6(H1-via-H1)
	L1(H4-via-R2)

in	L4, L7

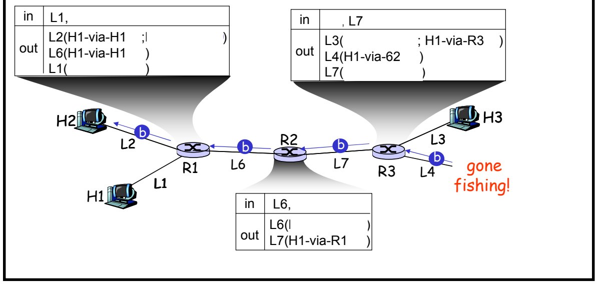
out	L3(H4-via-H4 ; H1-via-R3)
	L4(H1-via-62)
	L7(H4-via-H4 (b))

in	L6, L7

out	L6(H4-via-R3 (b))
	L7(H1-via-R1)

RSVP: *soft-state*

- suppose H4 (sender) leaves without performing teardown
- eventually state in routers will timeout and disappear!



in	L1,

out	L2(H1-via-H1 ;I)
	L6(H1-via-H1)
	L1( )

in	, L7

out	L3(; H1-via-R3)
	L4(H1-via-62)
	L7( )

in	L6,

out	L6(I)
	L7(H1-via-R1)

The many uses of reservation/path refresh

- recover from an earlier lost refresh message
  - expected time until refresh received must be longer than timeout interval! (short timer interval desired)
- Handle receiver/sender that goes away without teardown
  - Sender/receiver state will timeout and disappear
- Reservation refreshes will cause new reservations to be made to a receiver from a sender who has joined since receivers last reservation refresh
  - E.g., in previous example, H1 is only receiver, H3 only sender. Path/reservation messages complete, data flows
  - H4 joins as sender, nothing happens until H3 refreshes reservation, causing R3 to forward reservation to H4, which allocates bandwidth

RSVP: reflections

- multicast as a “first class” service
- receiver-oriented reservations
- use of soft-state