# Chapter 6
# Multimedia Networking

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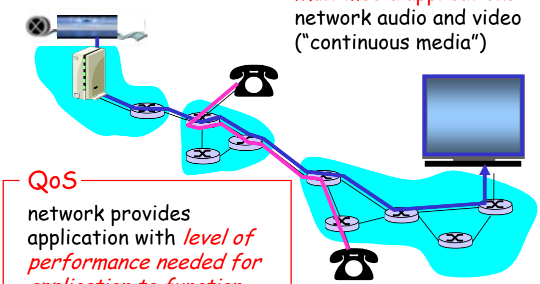
*Computer Networking: A Top Down Approach Featuring the Internet,*
2nd edition.
Jim Kurose, Keith Ross
Addison-Wesley, July 2002.

## 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.*

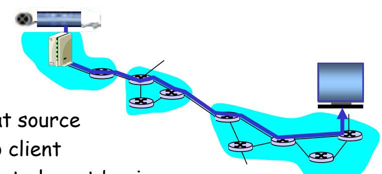
## MM Networking Applications

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.

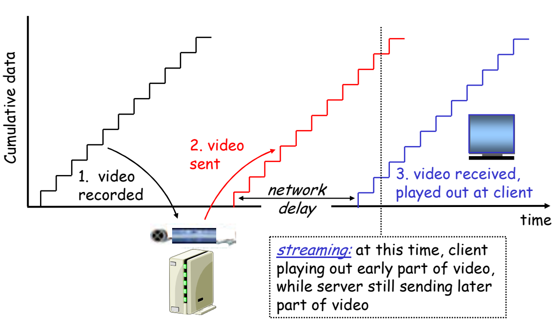
**Jitter** is the variability of packet delays within the same packet stream

## Streaming Stored Multimedia

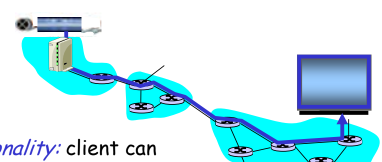


Streaming:
- media stored at source
- transmitted to client
- streaming: client playout begins *before* all data has arrived
  - timing constraint for still-to-be transmitted data: in time for playout

## Streaming Stored Multimedia: What is it?




## Streaming 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

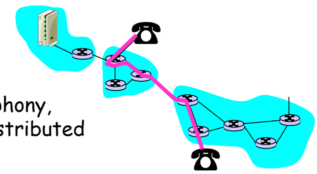
## Streaming Live Multimedia

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! ? ? ?

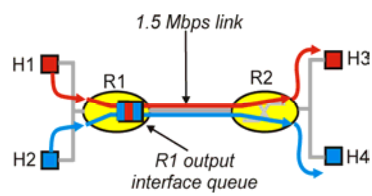
Today's Internet multimedia applications use application-level techniques to mitigate (as best possible) effects of delay, loss

## Improving QOS in IP Networks

Thus far: "making the best of best effort"

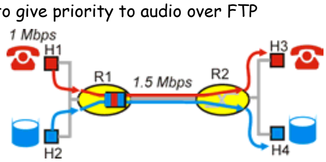
Future: next generation Internet with QoS guarantees
- RSVP: signaling for resource reservations
- Differentiated Services: differential guarantees
- Integrated Services: firm guarantees

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



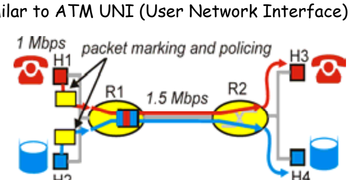

Principle 1

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)

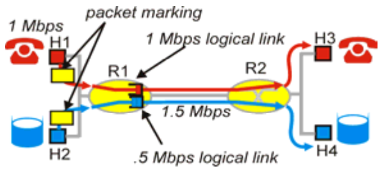



Principle 2

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

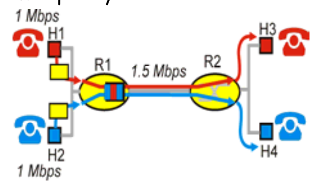



**Principle 3**

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

## Principles for QOS Guarantees (more)

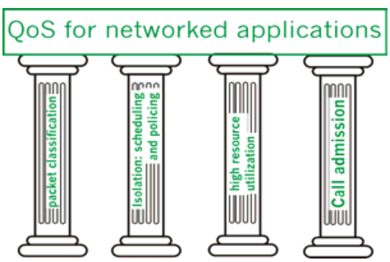
*Basic fact of life:* can not support traffic demands beyond link capacity




**Principle 4**

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

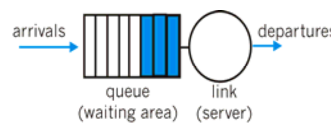
## Summary of QoS Principles




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

## Scheduling And Policing Mechanisms

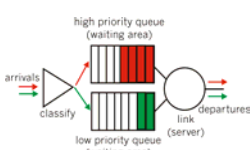
- scheduling: choose next packet to send on link
- FIFO (first in first out) scheduling: send in order of arrival to queue
  - real-world example?
  - discard policy: if packet arrives to full queue: who to discard?
    - Tail drop: drop arriving packet
    - priority: drop/remove on priority basis
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

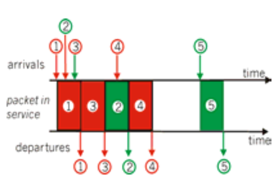



## Scheduling Policies: more

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?

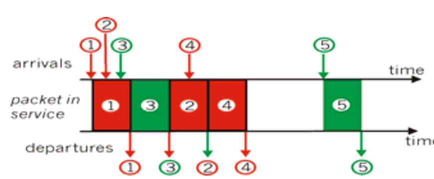



## Scheduling Policies: still more

round robin scheduling:

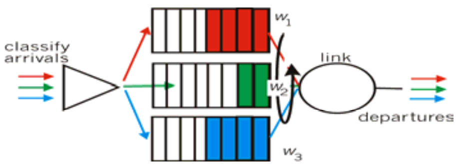
- multiple classes
- cyclically scan class queues, serving one from each class (if available)
- real world example?



## 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

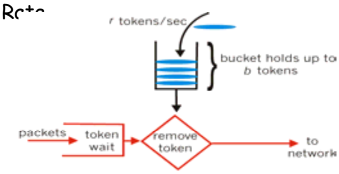
Goal: limit traffic to not exceed declared parameters

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

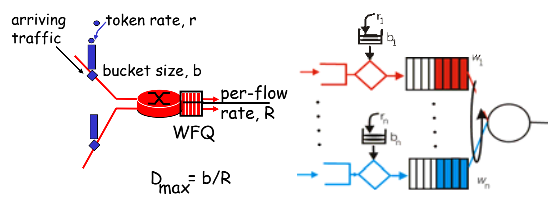
Token Bucket: limit input to specified Burst Size and Average Rate.




- 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*!



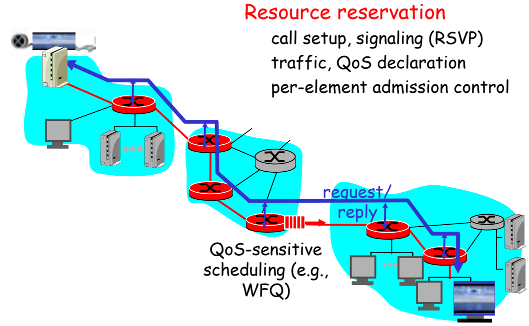

## IETF Integrated Services

- 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



## 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

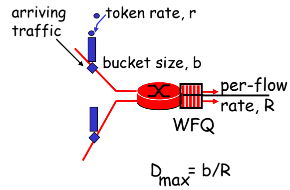
## Intserv QoS: Service models [rfc2211, rfc 2212]

**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."




## 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

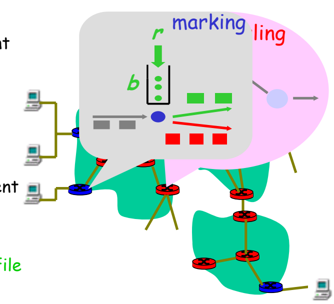
## Diffserv Architecture

Edge router:

- per-flow traffic management
- marks packets as in-profile and out-profile

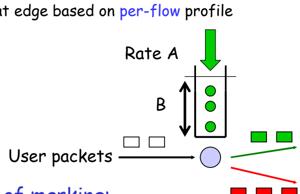
Core router:

- 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

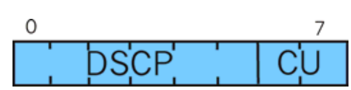



Possible usage of marking:

- 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