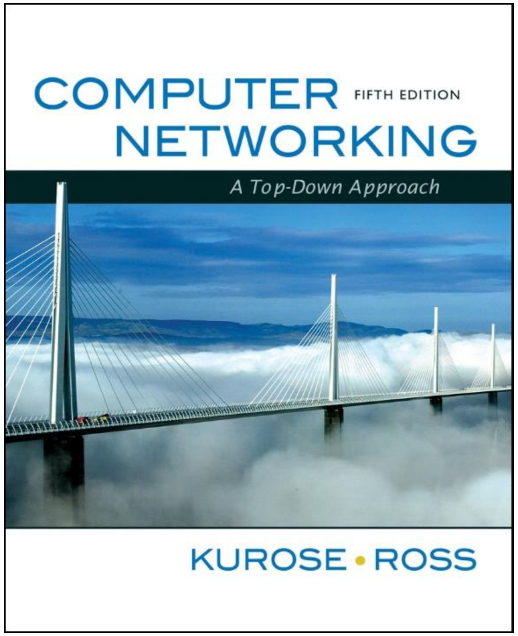
# Chapter 7
# Multimedia Networking

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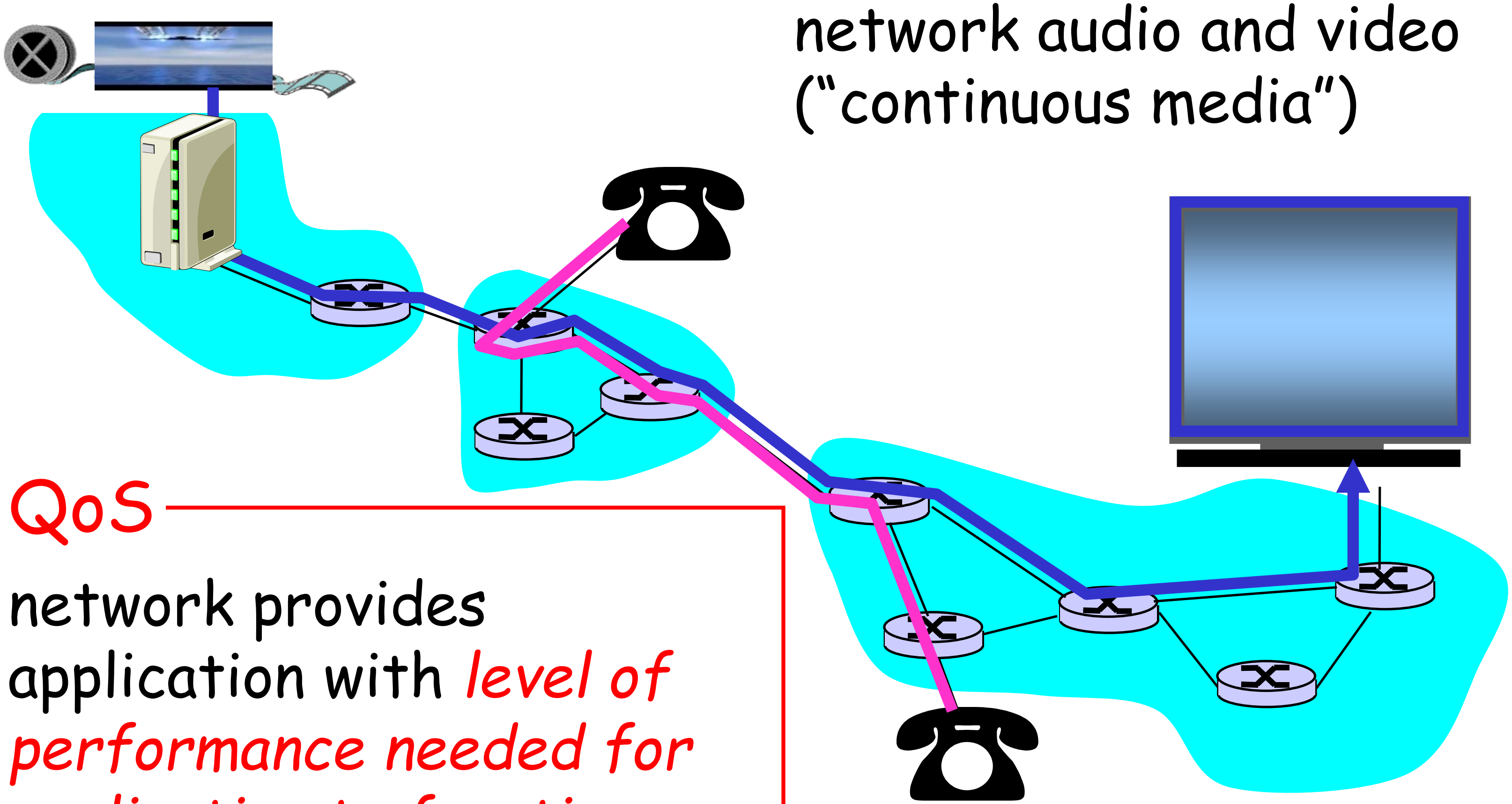
*Computer Networking: A Top Down Approach*
5th edition.
Jim Kurose, Keith Ross
Addison-Wesley, April 2009.

# Multimedia and 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.*

# Chapter 7: goals

## Principles

- classify multimedia applications
- identify network services applications need
- making the best of best effort service
- 尽全力而为

## Protocols and Architectures

- specific protocols for best-effort
- mechanisms for providing QoS
- architectures for QoS

# Chapter 7 outline

7.1 multimedia networking applications

7.2 streaming stored audio and video

7.3 making the best out of best effort service

7.4 protocols for real-time interactive applications
RTP,RTCP,SIP

7.5 providing multiple classes of service

7.6 providing QoS guarantees

# MM Networking Applications

**Classes of MM applications:**

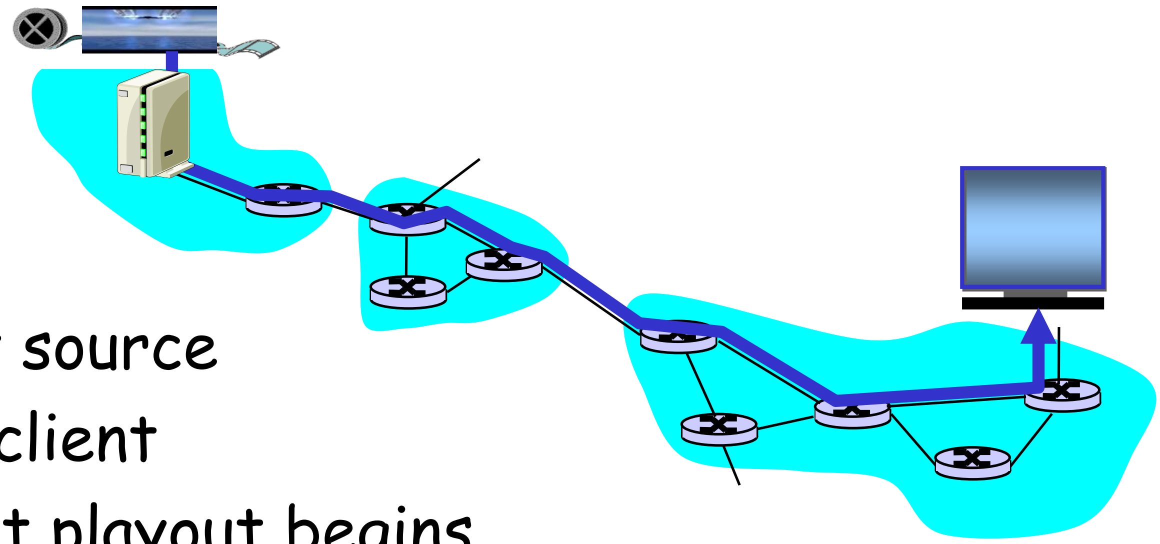
1) stored streaming
2) live streaming
3) interactive, real-time

**Fundamental characteristics:**

- typically **delay sensitive**
  - end-to-end delay
  - delay jitter
- **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

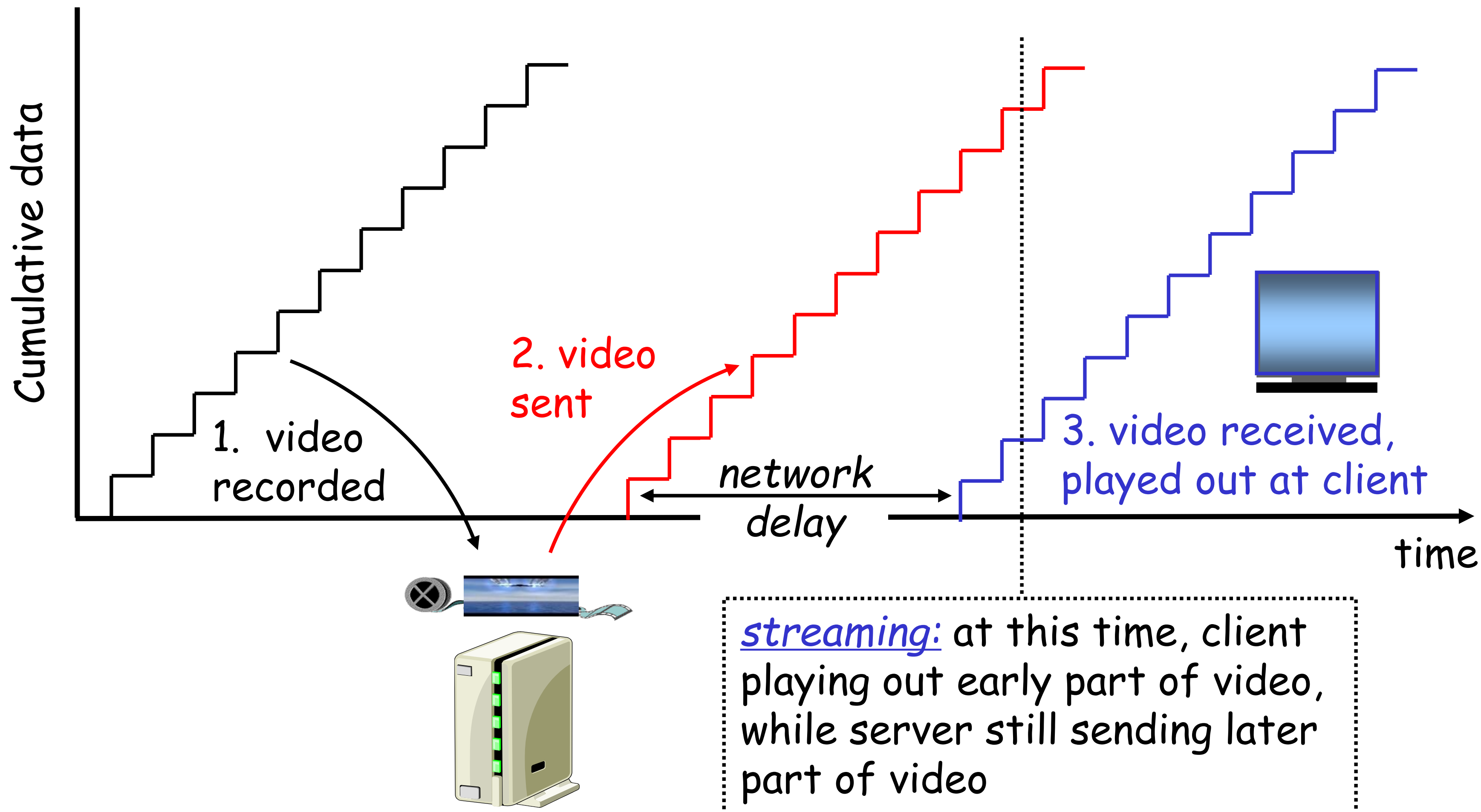
# Streaming Stored Multimedia



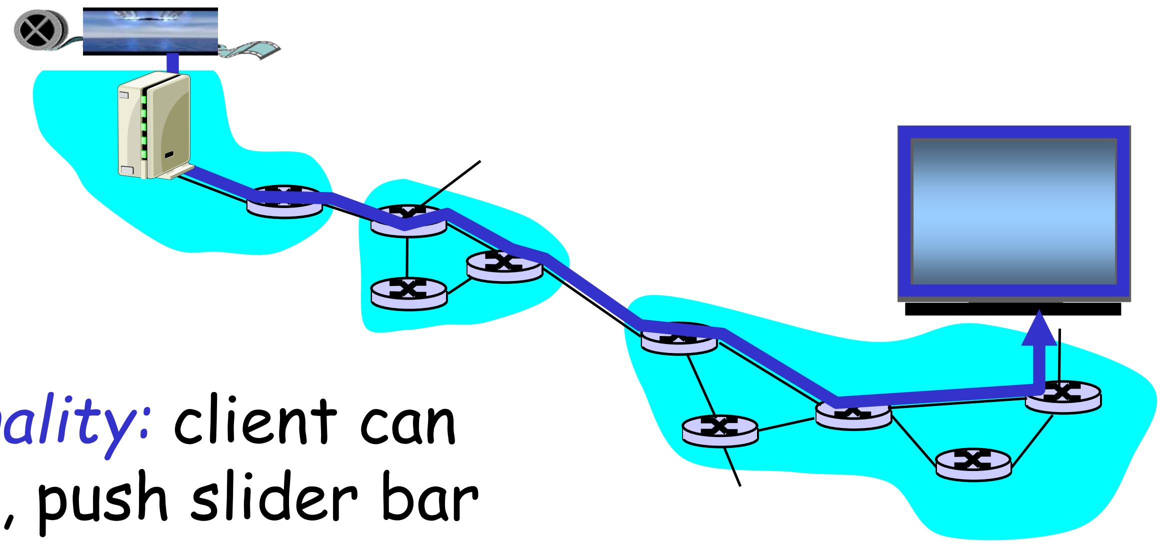
Stored streaming:

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

# Streaming *Live* Multimedia

Examples:

- Internet radio talk show
- live sporting event

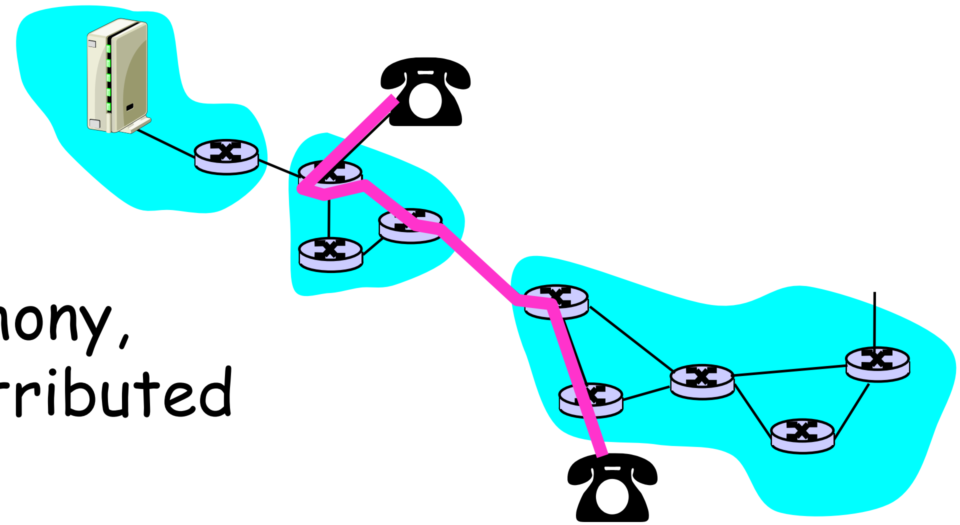
Streaming (as with streaming *stored* multimedia)

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

Interactivity

- fast forward impossible
- rewind, pause possible!

# Real-Time Interactive Multimedia

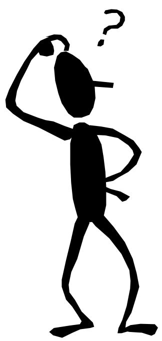


- applications: IP telephony, video conference, distributed interactive worlds
- end-end delay requirements:
  - audio: < 150 msec good, < 400 msec OK
    - includes application-level (packetization) and network delays
    - higher delays noticeable, impair interactivity
- session initialization
  - 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

# How should the Internet evolve to better support multimedia?

**Integrated services philosophy:**

- fundamental changes in Internet so that apps can reserve end-to-end bandwidth
- requires new, complex software in hosts & routers

**Laissez-faire**

- no major changes
- more bandwidth when needed
- content distribution, application-layer multicast
  - application layer

**Differentiated services philosophy:**

- fewer changes to Internet infrastructure, yet provide 1st and 2nd class service

What's your opinion?

# A few words about audio compression

- analog signal sampled at constant rate
  - telephone: 8,000 samples/sec
  - CD music: 44,100 samples/sec
- each sample quantized, i.e., rounded
  - e.g., $2^8$=256 possible quantized values
- each quantized value represented by bits
  - 8 bits for 256 values

- example: 8,000 samples/sec, 256 quantized values --> 64,000 bps
- receiver converts bits back to analog signal:
  - some quality reduction

## Example rates

- CD: 1.411 Mbps
- MP3: 96, 128, 160 kbps
- Internet telephony: 5.3 kbps and up

# A few words about video compression

- video: sequence of images displayed at constant rate
  - e.g. 24 images/sec
- digital image: array of pixels
  - each pixel represented by bits
- redundancy
  - spatial (within image)
  - temporal (from one image to next)

Examples:

- MPEG 1 (CD-ROM) 1.5 Mbps
- MPEG2 (DVD) 3-6 Mbps
- MPEG4 (often used in Internet, < 1 Mbps)

Research:

- layered (scalable) video
  - adapt layers to available bandwidth

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Loading...
00:00 / 04:51


广告剩余 13 秒
静音
跳过广告
香辣鸡翅
详细了解


优酷
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# Streaming Stored Multimedia

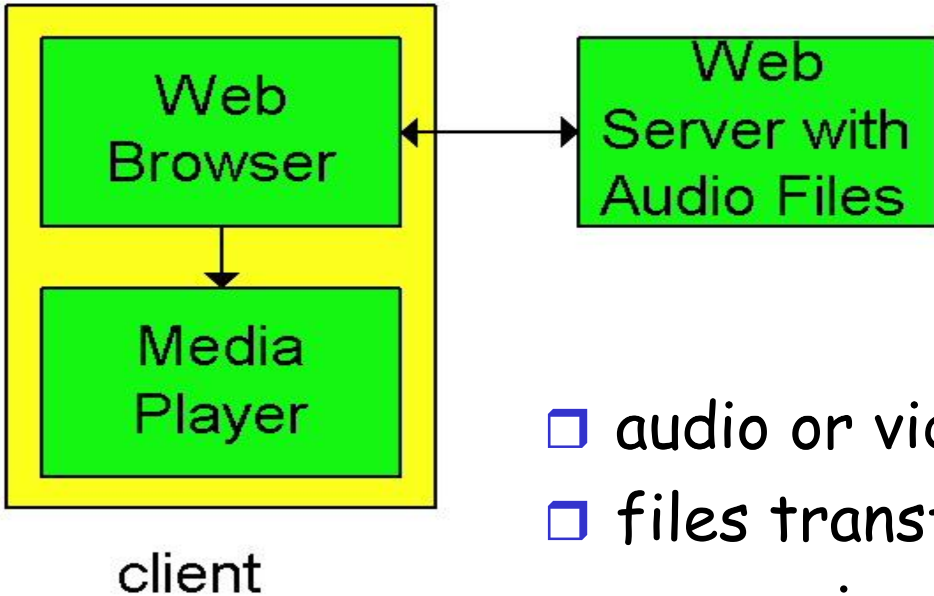
application-level streaming techniques for making the best out of best effort service:

- client-side buffering
- use of UDP versus TCP
- multiple encodings of multimedia

**Media Player**

- jitter removal
- decompression
- error concealment
- graphical user interface w/ controls for interactivity

# Internet multimedia: simplest approach

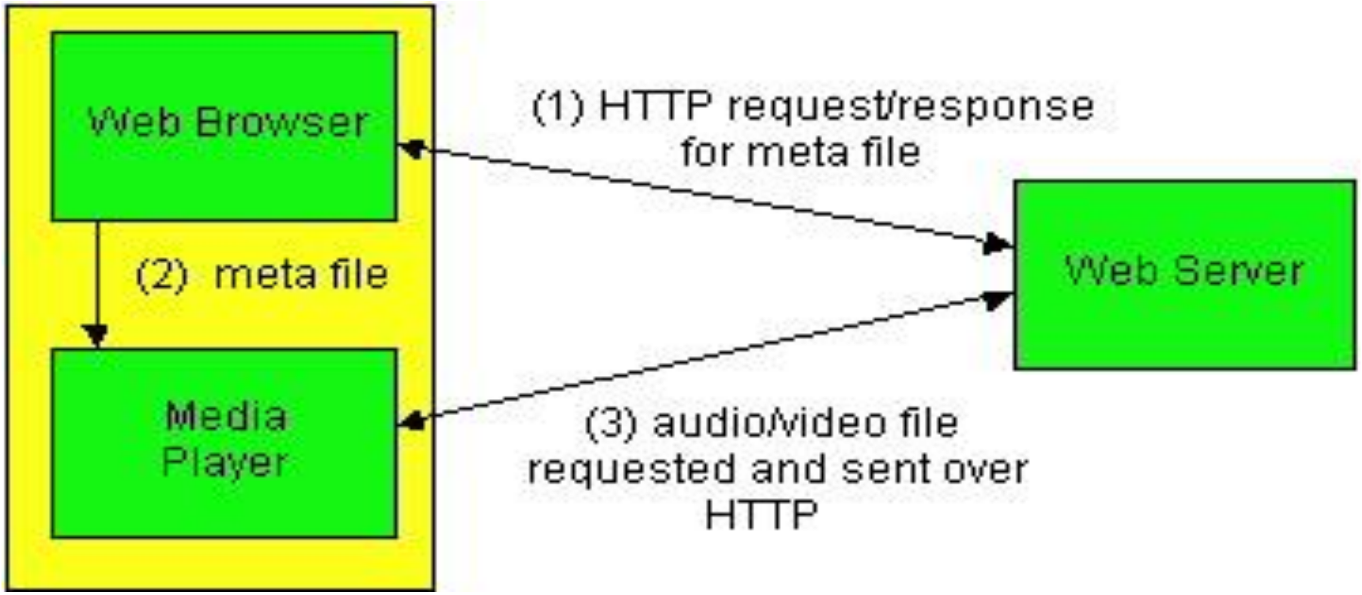



- audio or video stored in file
- files transferred as HTTP object
  - received in entirety at client
  - then passed to player

audio, video not streamed:

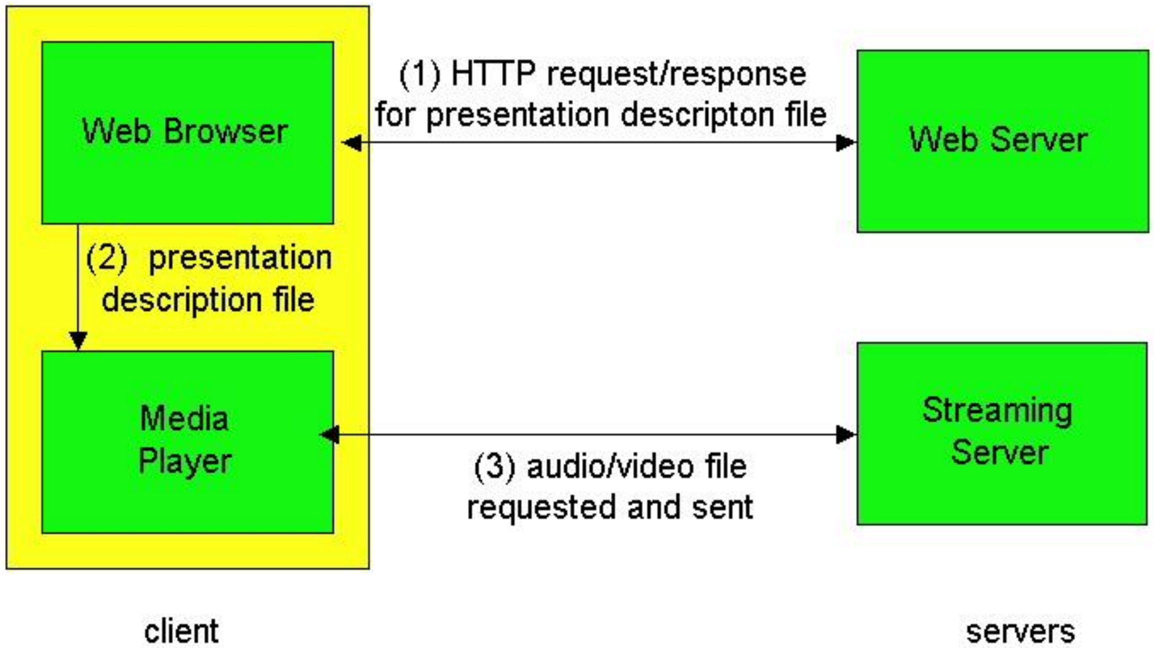
- no, "pipelining," long delays until playout!

# Internet multimedia: streaming approach



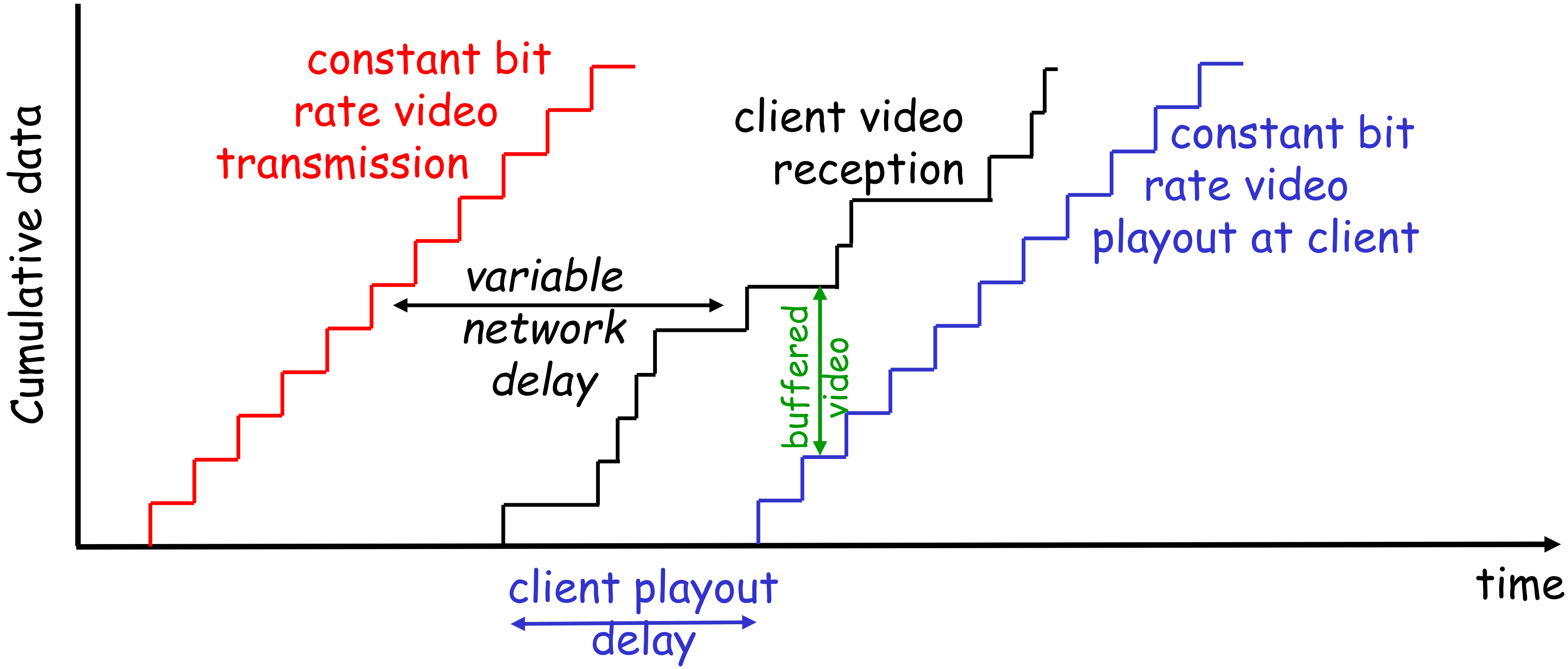

- browser GETs **metafile**
- browser launches player, passing metafile
- player contacts server
- server streams audio/video to player

# Streaming from a streaming server



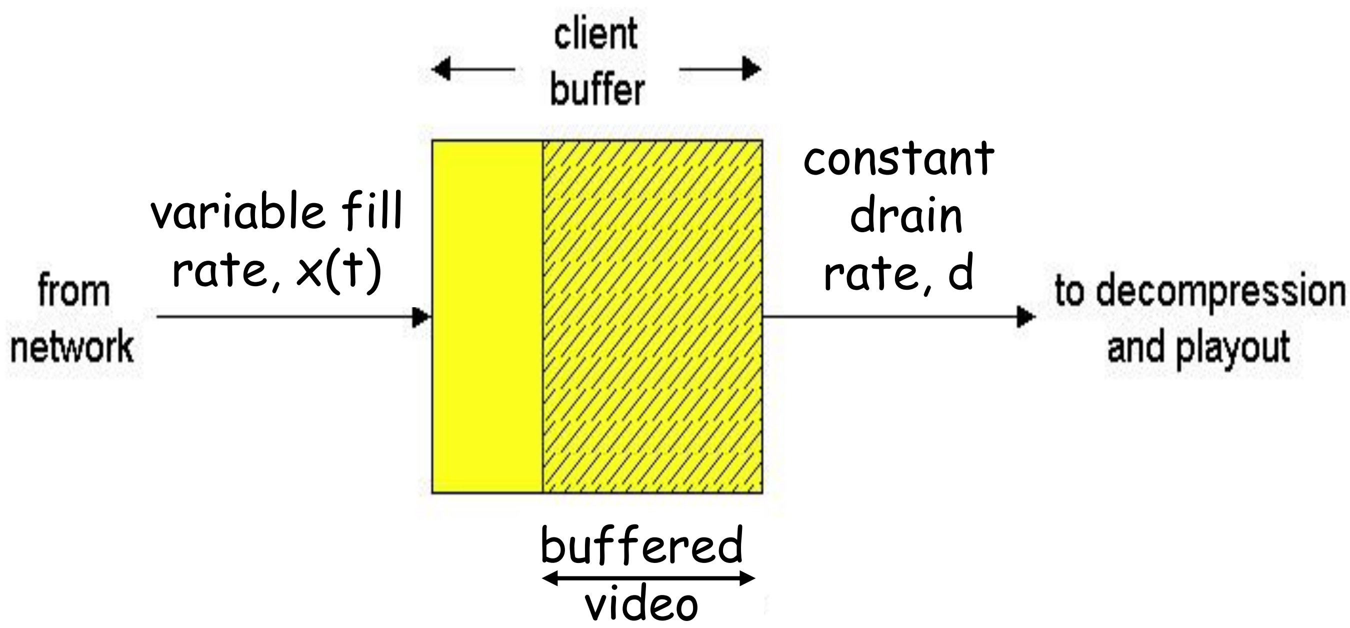

- allows for non-HTTP protocol between server, media player
- UDP or TCP for step (3), more shortly

# Streaming Multimedia: Client Buffering




- client-side buffering, playout delay compensate for network-added delay, delay jitter

# Streaming Multimedia: Client Buffering




- client-side buffering, playout delay compensate for network-added delay, delay jitter

# Streaming Multimedia: UDP or TCP?

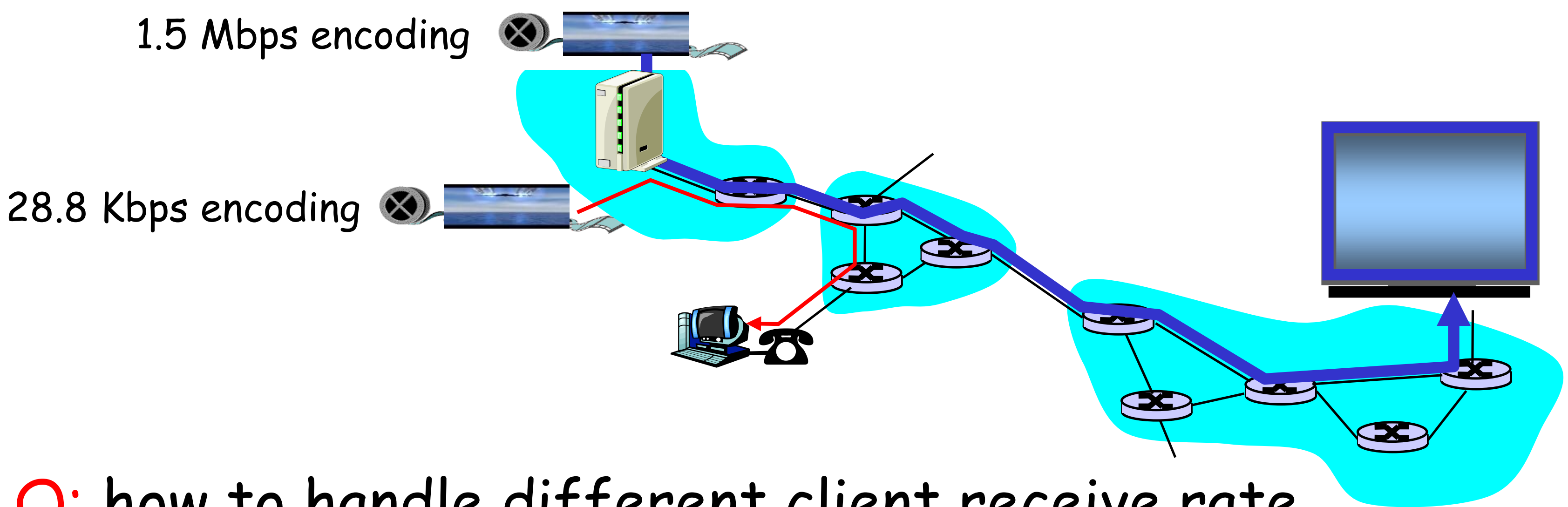
## UDP

- server sends at rate appropriate for client (oblivious to network congestion !)
  - often send rate = encoding rate = constant rate
  - then, fill rate = constant rate - packet loss
- short playout delay (2-5 seconds) to remove network jitter
- error recover: time permitting

## TCP

- send at maximum possible rate under TCP
- fill rate fluctuates due to TCP congestion control
- larger playout delay: smooth TCP delivery rate
- HTTP/TCP passes more easily through firewalls

# Streaming Multimedia: client rate(s)




Q: how to handle different client receive rate capabilities?

- 28.8 Kbps dialup
- 100 Mbps Ethernet

A: server stores, transmits multiple copies of video, encoded at different rates

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# Real-time interactive applications

- PC-2-PC phone
  - Skype
- PC-2-phone
  - Dialpad
  - Net2phone
  - Skype
- videoconference with webcams
  - Skype
  - Polycom

Going to now look at a PC-2-PC Internet phone example in detail

# Interactive Multimedia: Internet Phone

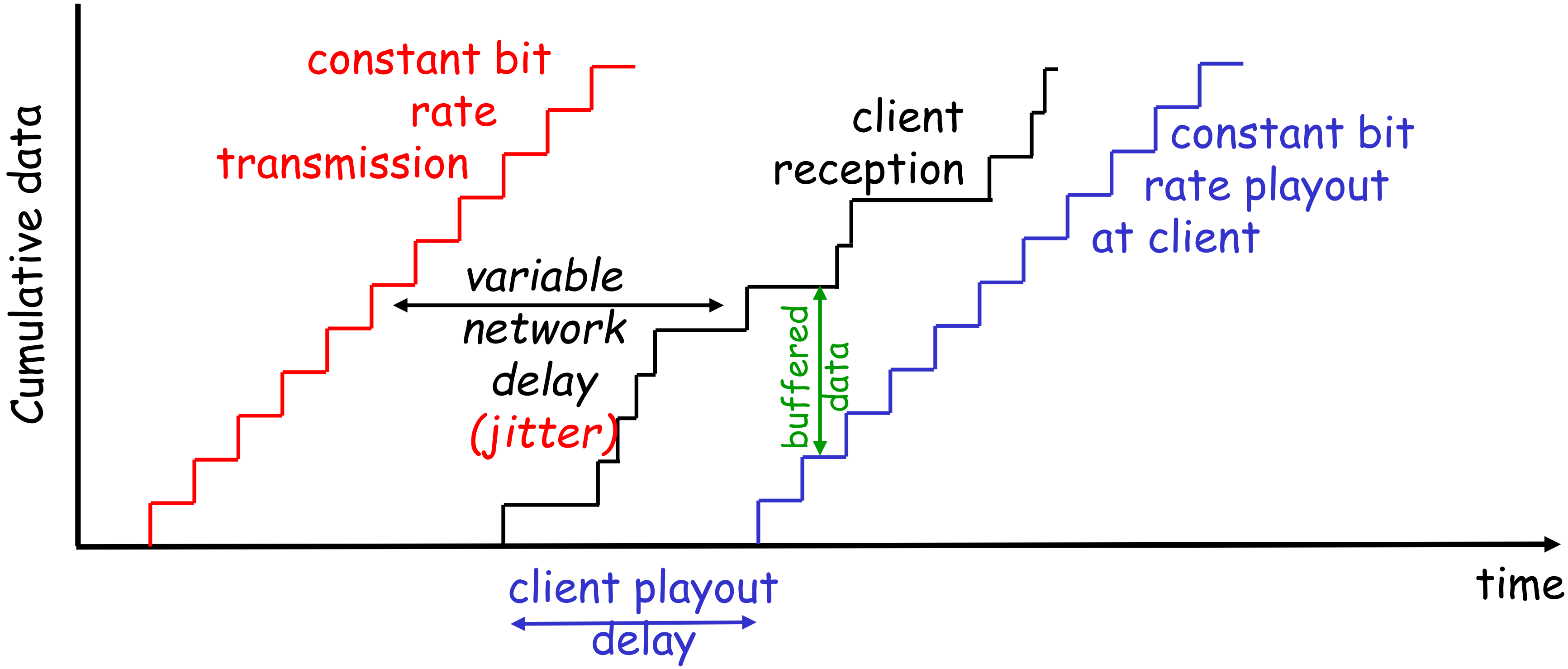
Introduce Internet Phone by way of an example

- speaker's audio: alternating talk spurts, silent periods.
  - 64 kbps during talk spurt
  - pkts generated only during talk spurts
  - 20 msec chunks at 8 Kbytes/sec: 160 bytes data
- application-layer header added to each chunk.
- chunk+header encapsulated into UDP segment.
- application sends UDP segment into socket every 20 msec during talkspurt

# Internet Phone: Packet Loss and Delay

- network loss: IP datagram lost due to network congestion (router buffer overflow)
- delay loss: IP datagram arrives too late for playout at receiver
  - delays: processing, queueing in network; end-system (sender, receiver) delays
  - typical maximum tolerable delay: 400 ms
- loss tolerance: depending on voice encoding, losses concealed, packet loss rates between 1% and 10% can be tolerated.

# Delay Jitter



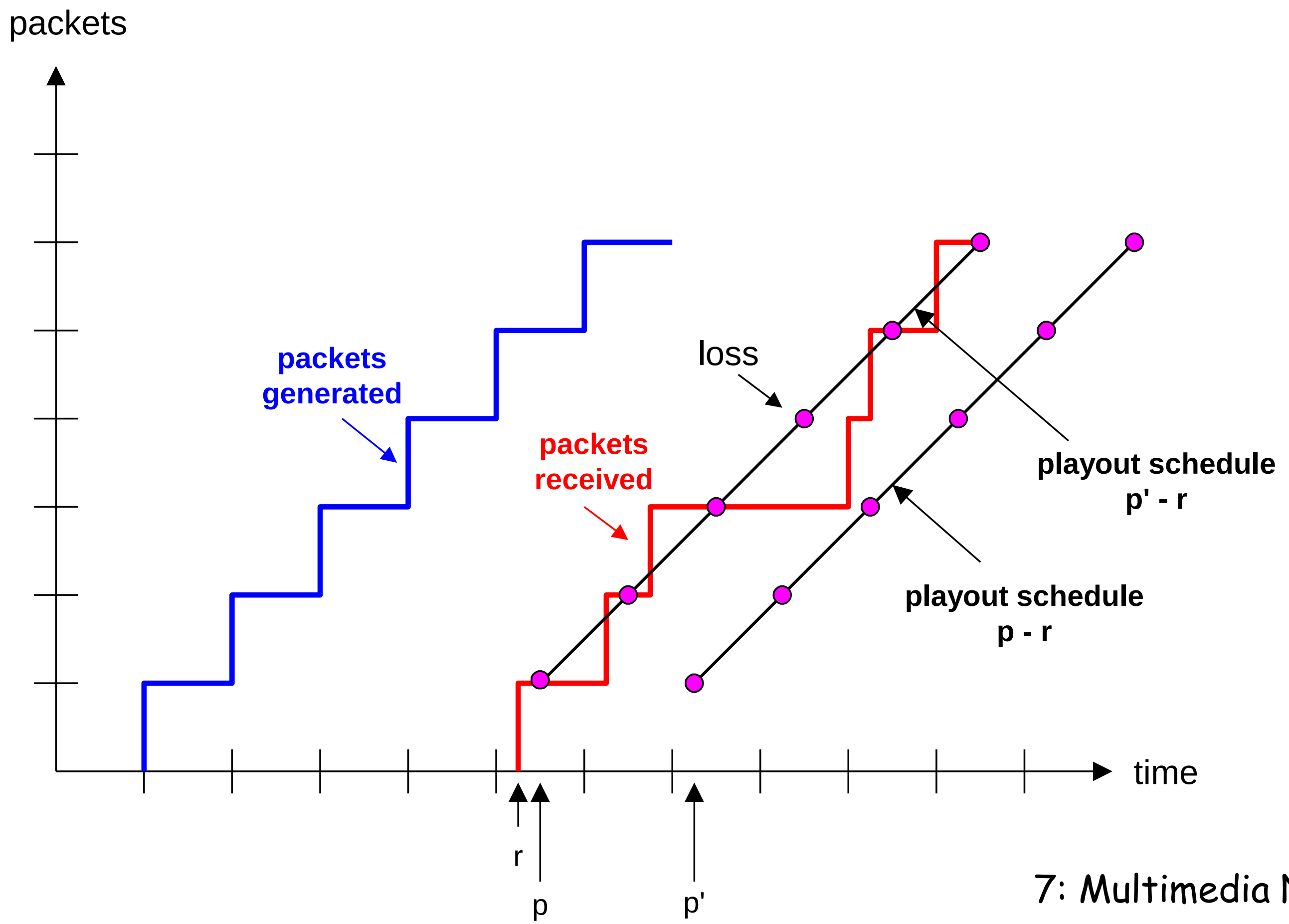

- consider end-to-end delays of two consecutive packets: difference can be more or less than 20 msec (transmission time difference)

# Internet Phone: Fixed Playout Delay

- receiver attempts to playout each chunk exactly q msecs after chunk was generated.
  - chunk has time stamp t: play out chunk at t+q .
  - chunk arrives after t+q: data arrives too late for playout, data "lost"
- tradeoff in choosing q:
  - *large q:* less packet loss
  - *small q:* better interactive experience

# Fixed Playout Delay

- sender generates packets every 20 msec during talk spurt.
- first packet received at time r
- first playout schedule: begins at p
- second playout schedule: begins at p'



# Adaptive Playout Delay (1)

- Goal: minimize playout delay, keeping late loss rate low
- Approach: adaptive playout delay adjustment:
  - estimate network delay, adjust playout delay at beginning of each talk spurt.
  - silent periods compressed and elongated.
  - chunks still played out every 20 msec during talk spurt.

$t_i$ = timestamp of the ith packet

$r_i$ = the time packet i is received by receiver

$p_i$ = the time packet i is played at receiver

$r_i - t_i$ = network delay for ith packet

$d_i$ = estimate of average network delay after receiving ith packet

dynamic estimate of average delay at receiver:

$$d_i = (1-u)d_{i-1} + u(r_i - t_i)$$

where $u$ is a fixed constant (e.g., $u$ = .01).

# Adaptive playout delay (2)

- also useful to estimate average deviation of delay, $v_i$ :

$$v_i = (1-u)v_{i-1} + u\,|\,r_i - t_i - d_i\,|$$

- estimates $d_i$ , $v_i$ calculated for every received packet (but used only at start of talk spurt
- for first packet in talk spurt, playout time is:

$$p_i = t_i + d_i + Kv_i$$

where K is positive constant

- remaining packets in talkspurt are played out periodically

# Adaptive Playout (3)

Q: How does receiver determine whether packet is first in a talkspurt?

- if no loss, receiver looks at successive timestamps.
  - difference of successive stamps > 20 msec -->talk spurt begins.
- with loss possible, receiver must look at both time stamps and sequence numbers.
  - difference of successive stamps > 20 msec and sequence numbers without gaps --> talk spurt begins.

# Recovery from packet loss (1)

**Forward Error Correction (FEC): simple scheme**

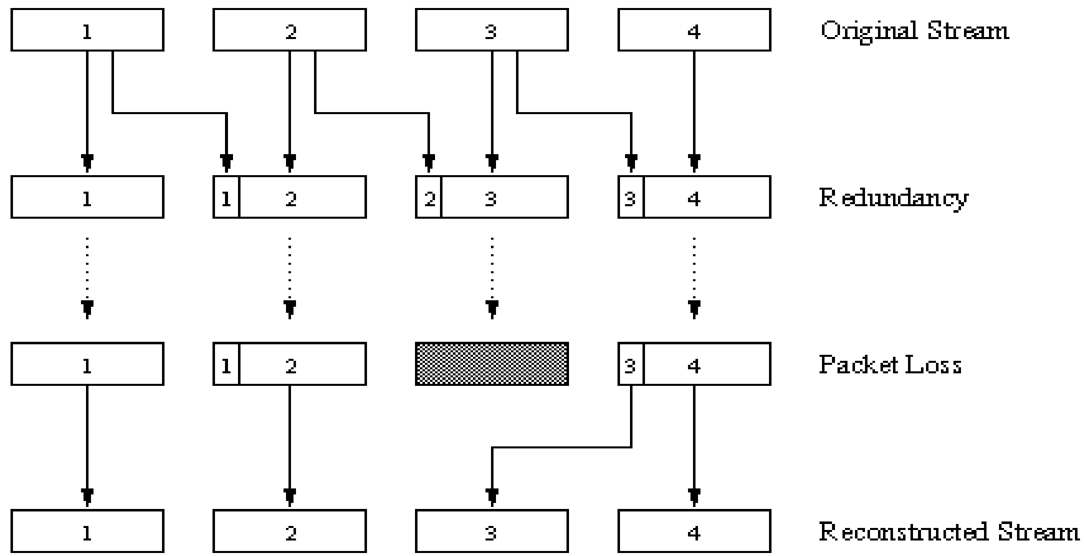
- for every group of *n* chunks create redundant chunk by exclusive OR-ing *n* original chunks
- send out *n+1* chunks, increasing bandwidth by factor *1/n.*
- can reconstruct original *n* chunks if at most one lost chunk from *n+1* chunks

- playout delay: enough time to receive all n+1 packets
- tradeoff:
  - increase n, less bandwidth waste
  - increase n, longer playout delay
  - increase n, higher probability that 2 or more chunks will be lost

# Recovery from packet loss (2)

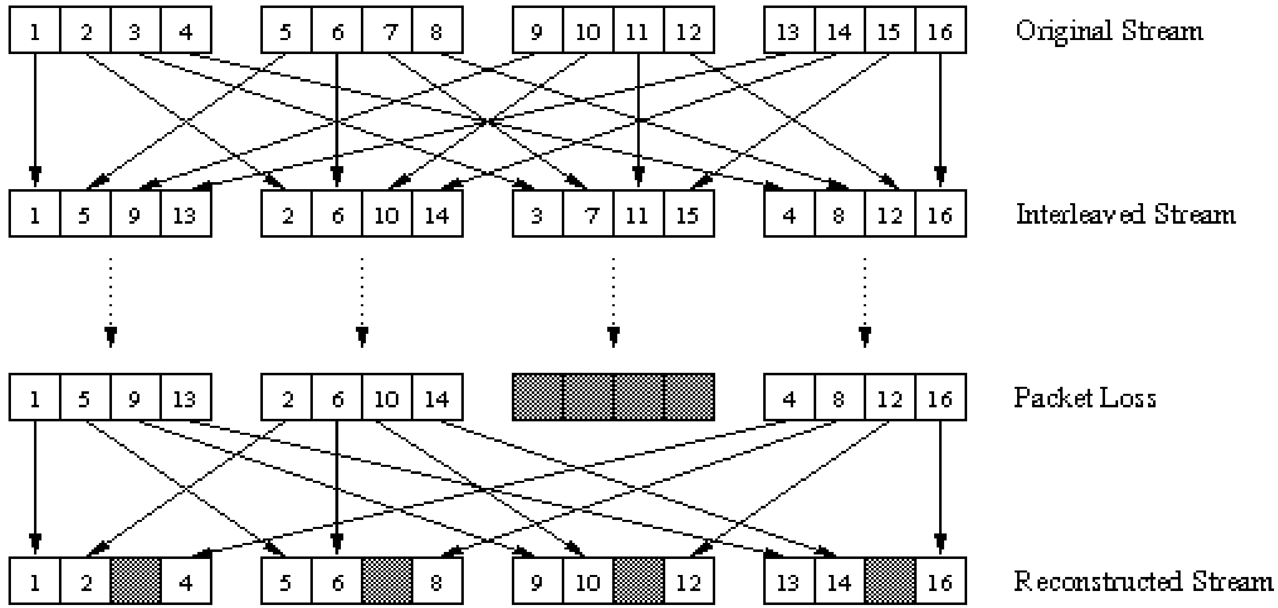
**2nd FEC scheme**

- "piggyback lower quality stream"
- send lower resolution audio stream as redundant information
- e.g., nominal stream PCM at 64 kbps and redundant stream GSM at 13 kbps.




- whenever there is non-consecutive loss, receiver can conceal the loss.
- can also append (n-1)st and (n-2)nd low-bit rate chunk

# Recovery from packet loss (3)



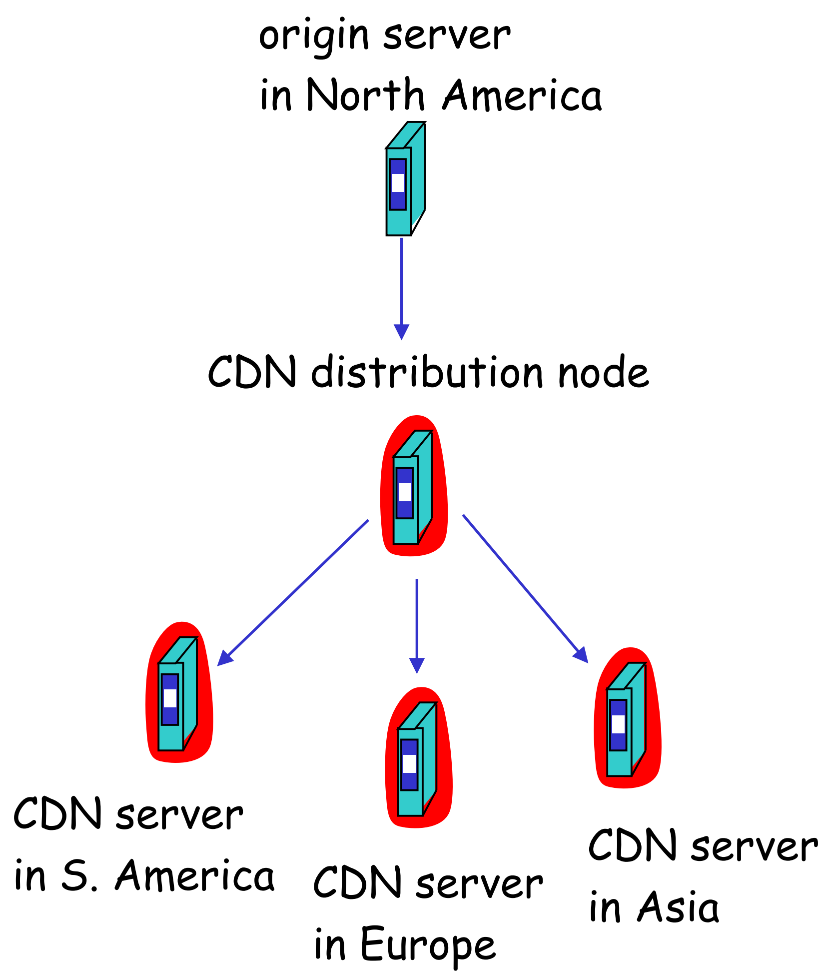

## Interleaving

- chunks divided into smaller units
- for example, four 5 msec units per chunk
- packet contains small units from different chunks
- if packet lost, still have most of every chunk
- no redundancy overhead, but increases playout delay

# Content distribution networks (CDNs)

## Content replication

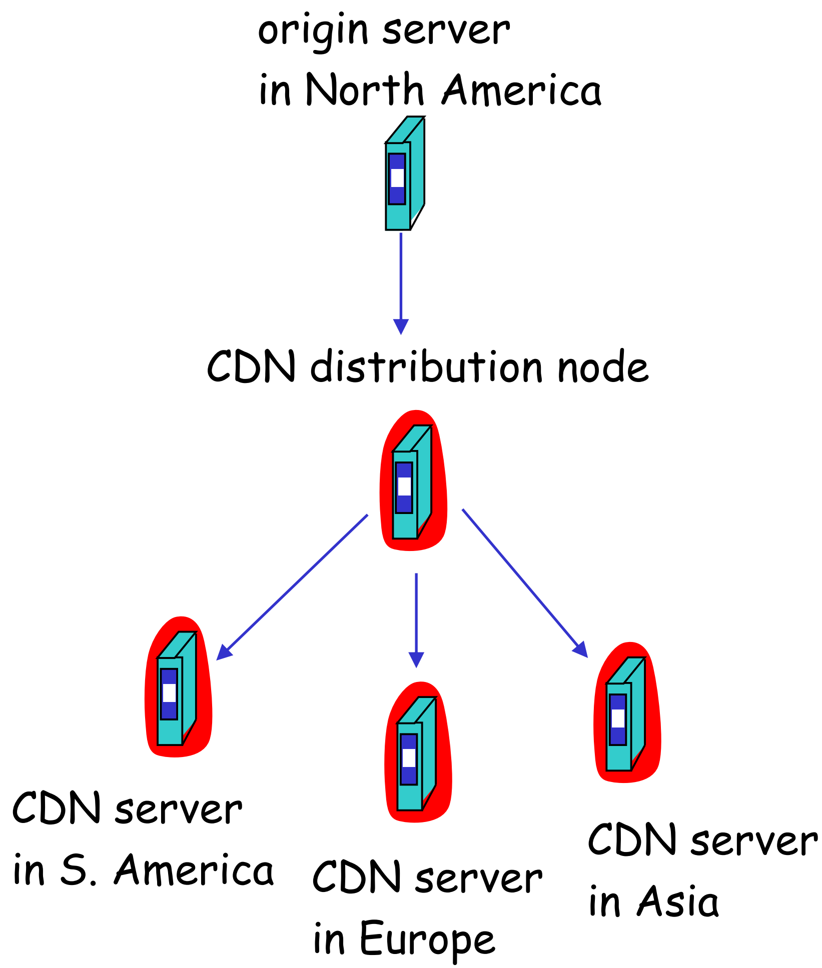
- challenging to stream large files (e.g., video) from single origin server in real time
- *solution:* replicate content at hundreds of servers throughout Internet
  - content downloaded to CDN servers ahead of time
  - *placing content "close" to user avoids impairments (loss, delay) of sending content over long paths*
  - CDN server typically in edge/access network



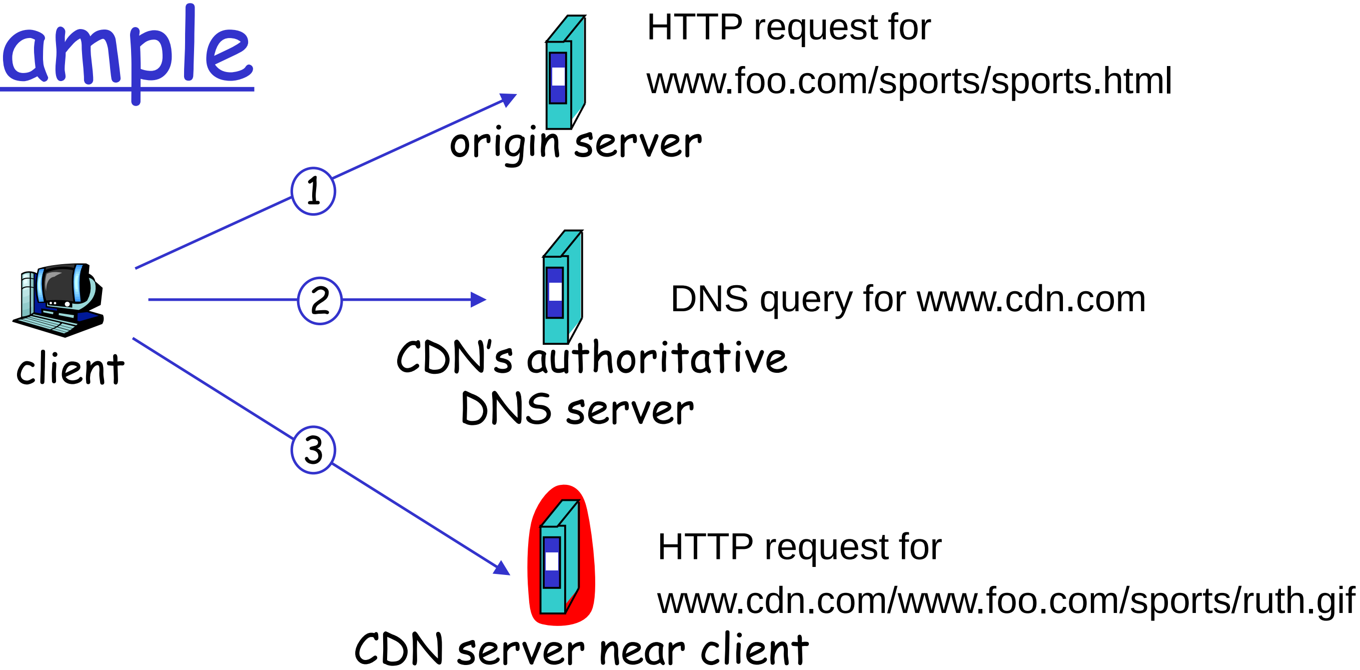
# Content distribution networks (CDNs)

## Content replication

- CDN (e.g., Akamai) customer is the content provider (e.g., CNN)
- CDN replicates customers' content in CDN servers.
- when provider updates content, CDN updates servers



# CDN example




## origin server (www.foo.com)

- distributes HTML
- replaces:

  http://www.foo.com/sports.ruth.gif

  with

  http://www.cdn.com/www.foo.com/sports/ruth.gif

## CDN company (cdn.com)

- distributes gif files
- uses its authoritative DNS server to route redirect requests


东北电信客户
东北网通客户
东北电信节点
华北网通客户
华北电信客户
华北网通节点
东北网通节点
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另有:
美国节点
香港节点
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# More about CDNs

routing requests

- CDN creates a “map”, indicating distances from leaf ISPs and CDN nodes
- when query arrives at authoritative DNS server:
  - server determines ISP from which query originates
  - uses “map” to determine best CDN server
- CDN nodes create application-layer overlay network

# Summary: Internet Multimedia: bag of tricks

- use UDP to avoid TCP congestion control (delays) for time-sensitive traffic
- client-side adaptive playout delay: to compensate for delay
- server side matches stream bandwidth to available client-to-server path bandwidth
  - chose among pre-encoded stream rates
  - dynamic server encoding rate
- error recovery (on top of UDP)
  - FEC, interleaving, error concealment
  - retransmissions, time permitting
- CDN: bring content closer to clients

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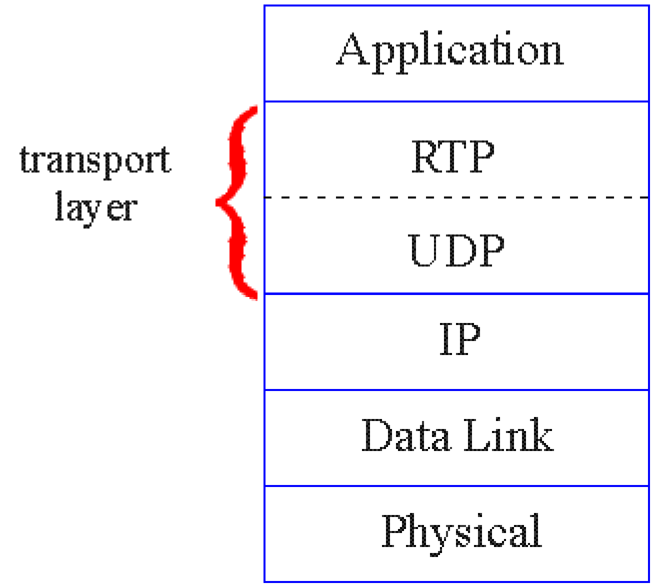
# Real-Time Protocol (RTP)

- RTP specifies packet structure for packets carrying audio, video data
- RFC 3550
- RTP packet provides
  - payload type identification
  - packet sequence numbering
  - time stamping
- RTP runs in end systems
- RTP packets encapsulated in UDP segments
- interoperability: if two Internet phone applications run RTP, then they may be able to work together

# RTP runs on top of UDP

RTP libraries provide transport-layer interface that extends UDP:

- port numbers, IP addresses
- payload type identification
- packet sequence numbering
- time-stamping



# RTP Example

- consider sending 64 kbps PCM-encoded voice over RTP.
- application collects encoded data in chunks, e.g., every 20 msec = 160 bytes in a chunk.
- audio chunk + RTP header form RTP packet, which is encapsulated in UDP segment
- RTP header indicates type of audio encoding in each packet
  - sender can change encoding during conference.
- RTP header also contains sequence numbers, timestamps.

# RTP and QoS

- RTP does **not** provide any mechanism to ensure timely data delivery or other QoS guarantees.
- RTP encapsulation is only seen at end systems (not) by intermediate routers.
  - routers providing best-effort service, making no special effort to ensure that RTP packets arrive at destination in timely matter.

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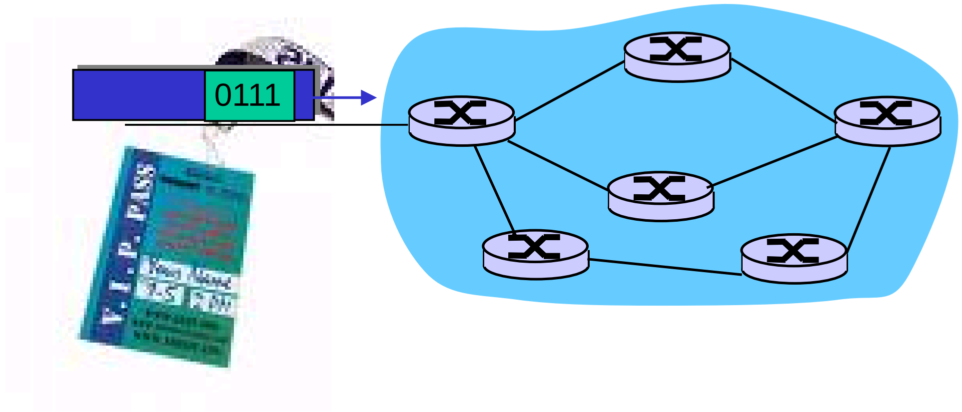
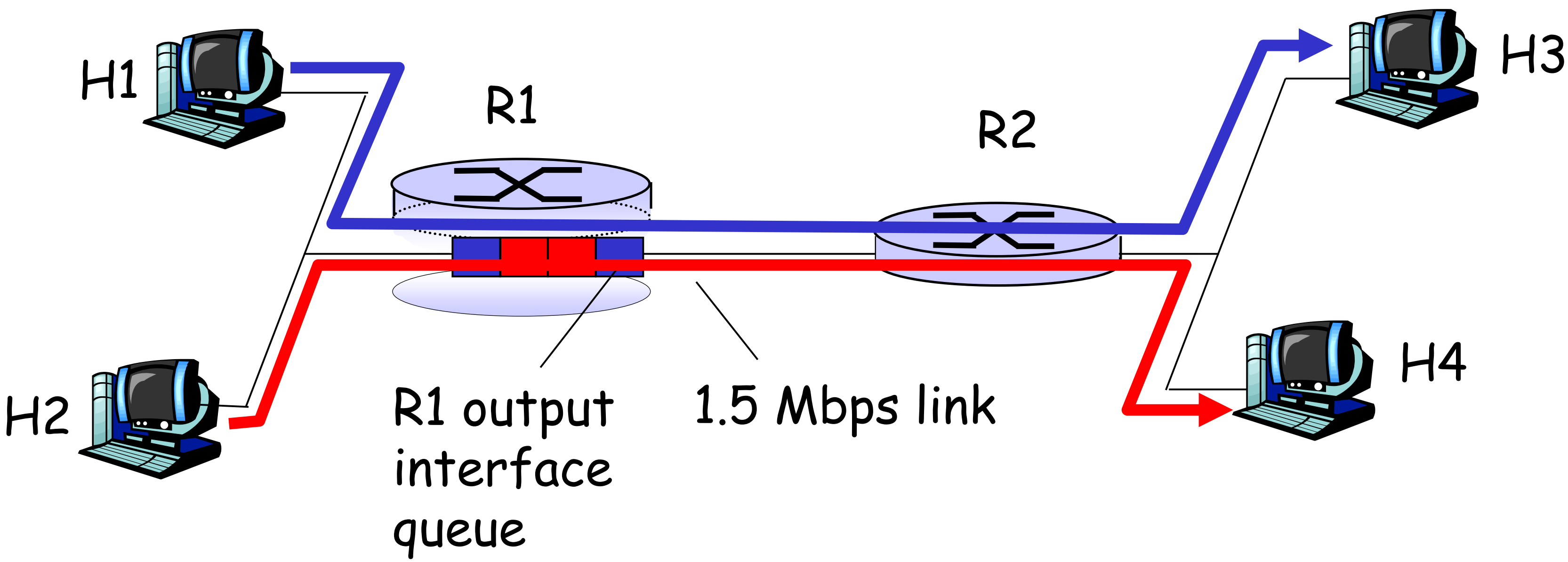
# Providing Multiple Classes of Service

- thus far: making the best of best effort service
  - one-size fits all service model
- alternative: multiple classes of service
  - partition traffic into classes
  - network treats different classes of traffic differently (analogy: VIP service vs regular service)
- granularity: differential service among multiple classes, not among individual connections
- history: ToS bits

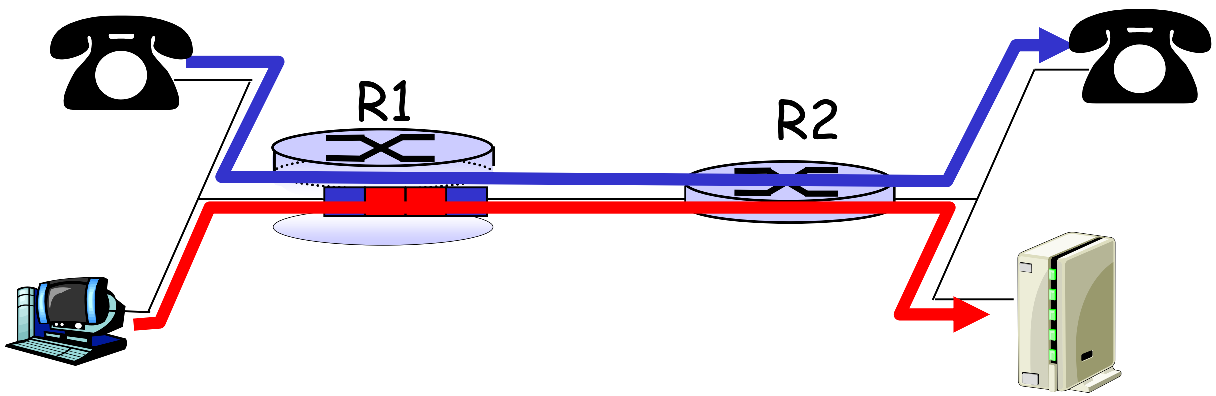


# Multiple classes of service: scenario



# Scenario 1: mixed FTP and audio

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

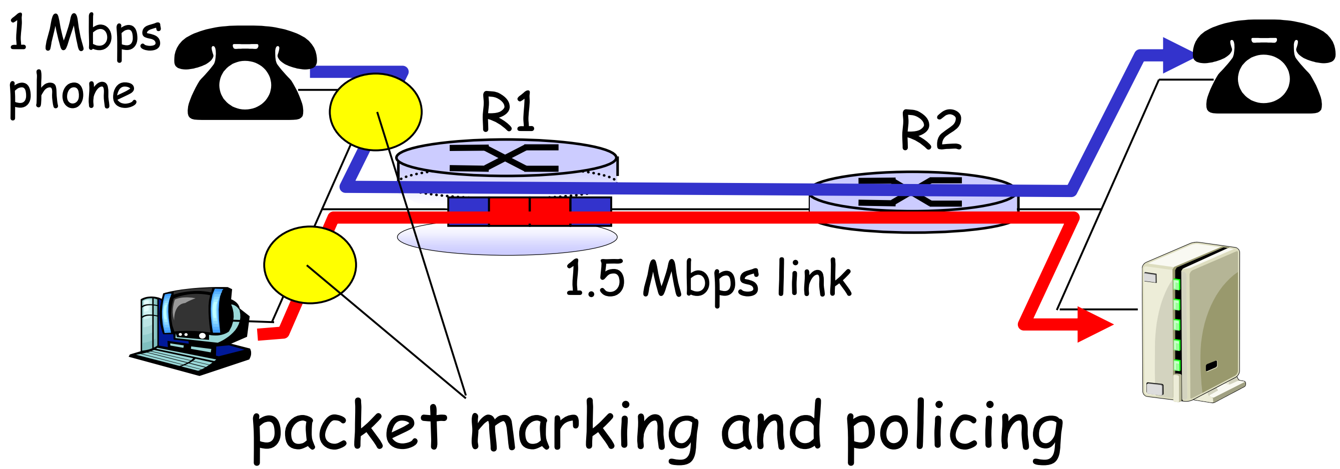



**Principle 1**

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)

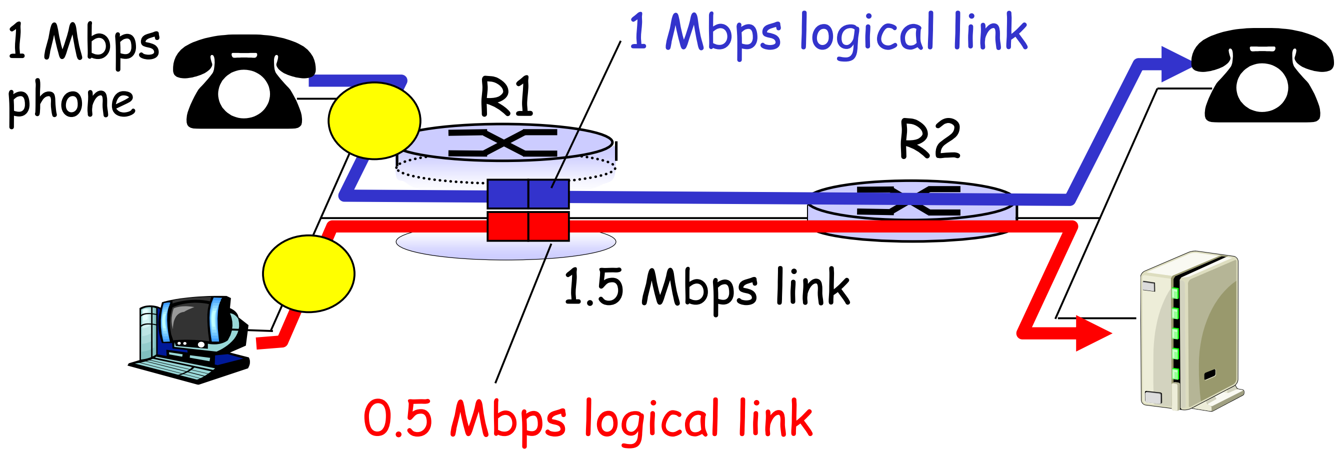



**Principle 2**

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

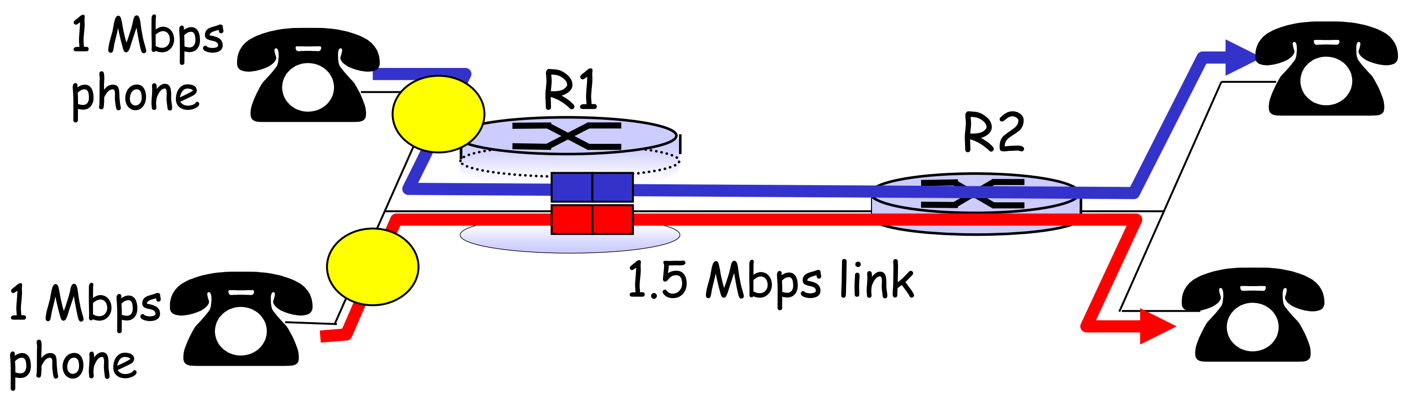



**Principle 3**

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

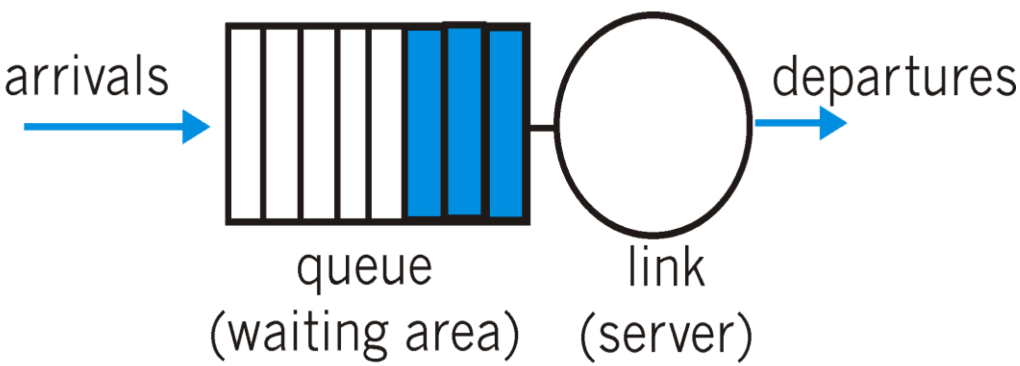



**Principle 4**

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

# Scheduling And Policing Mechanisms

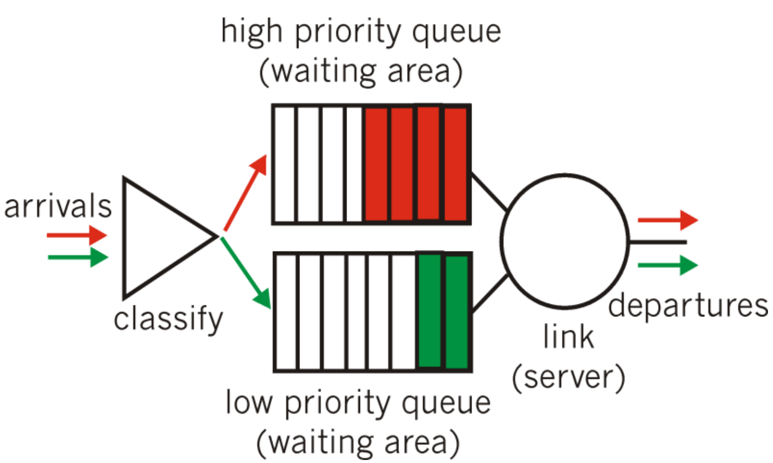
- **scheduling:** choose next packet to send on link
- **FIFO (first in first out) scheduling:** send in order of arrival to queue
  - real-world example?
  - **discard policy:** if packet arrives to full queue: who to discard?
    - Tail drop: drop arriving packet
    - priority: drop/remove on priority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

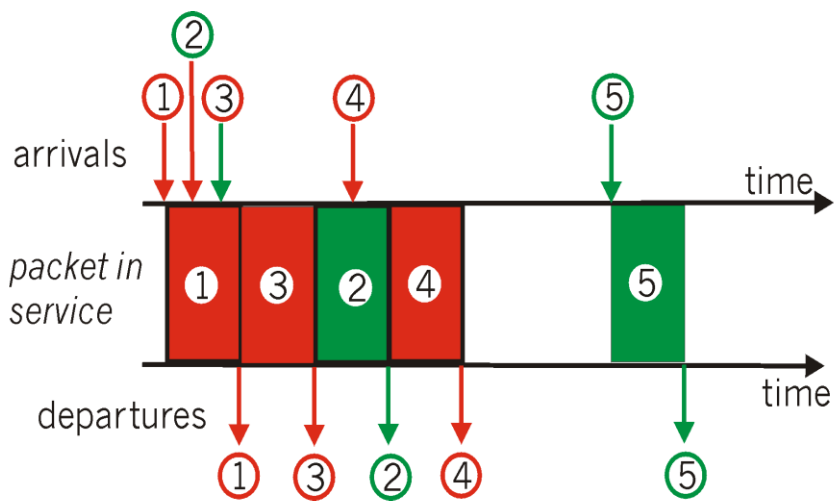


# Scheduling Policies: more

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?

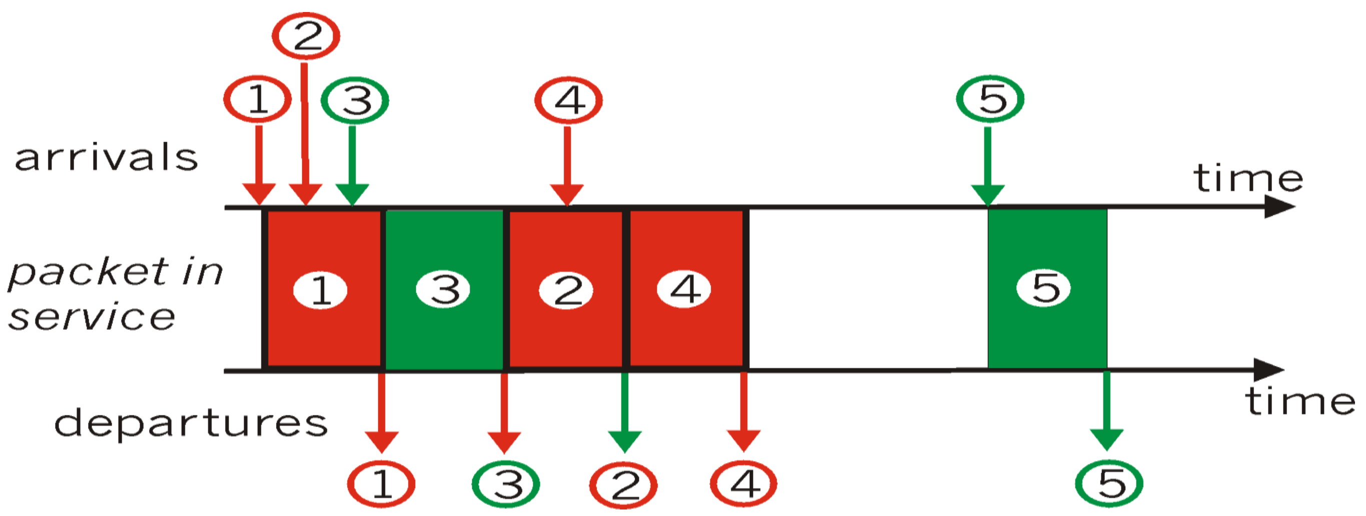


# Scheduling Policies: still more

round robin scheduling:

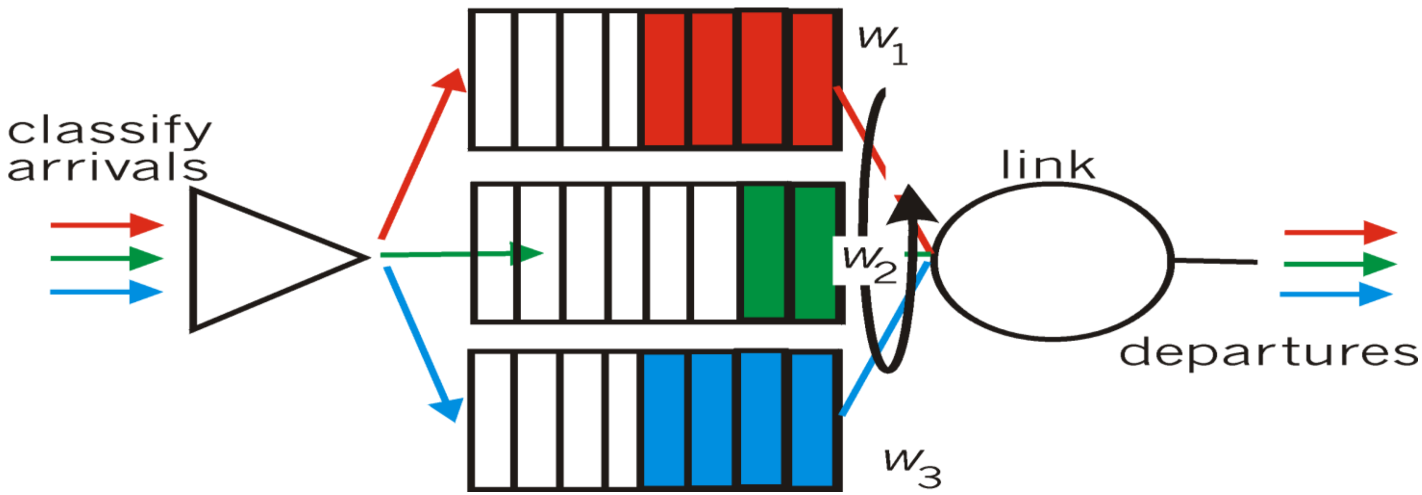
- multiple classes
- cyclically scan class queues, serving one from each class (if available)
- real world example?



# 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

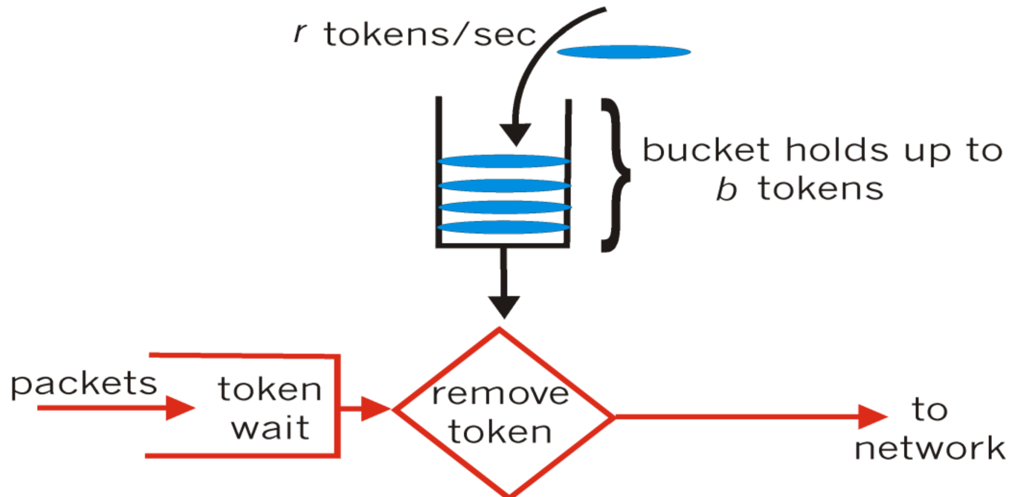
**Goal:** limit traffic to not exceed declared parameters

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

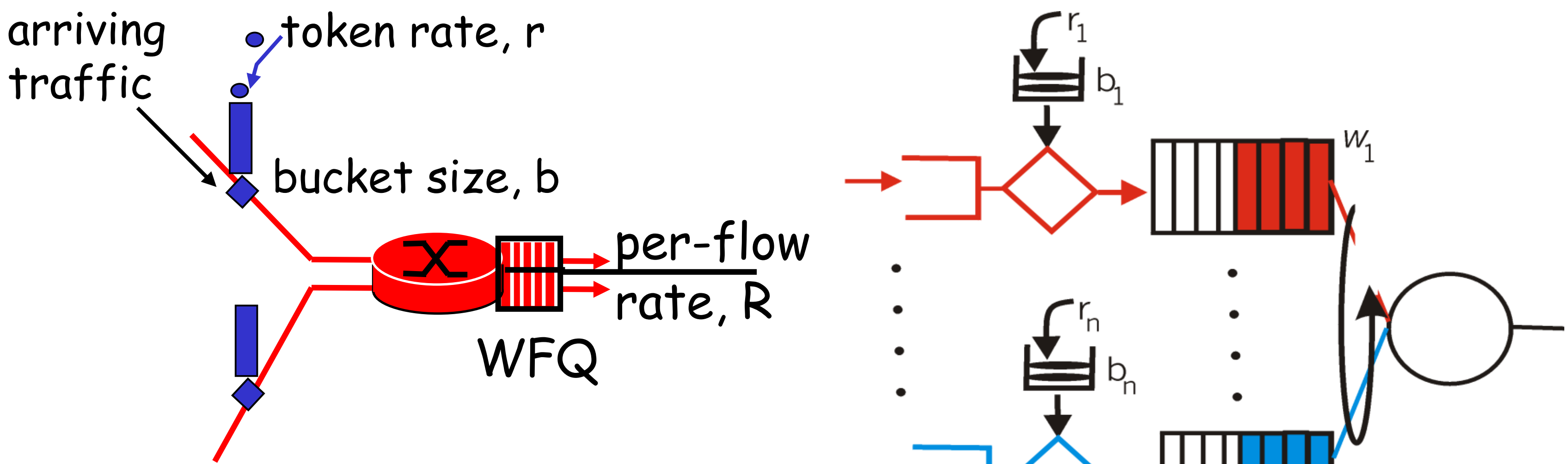
**Token Bucket:** limit input to specified Burst Size and Average Rate.




- 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*!




$D_{max} = b/R$

# IETF Differentiated Services

- want "qualitative" service classes
  - "behaves like a wire"
  - relative service distinction: Platinum, Gold, Silver
- *scalability:* simple functions in network core, relatively complex functions at edge routers (or hosts)
  - signaling, maintaining per-flow router state difficult with large number of flows
- don't define define service classes, provide functional components to build service classes

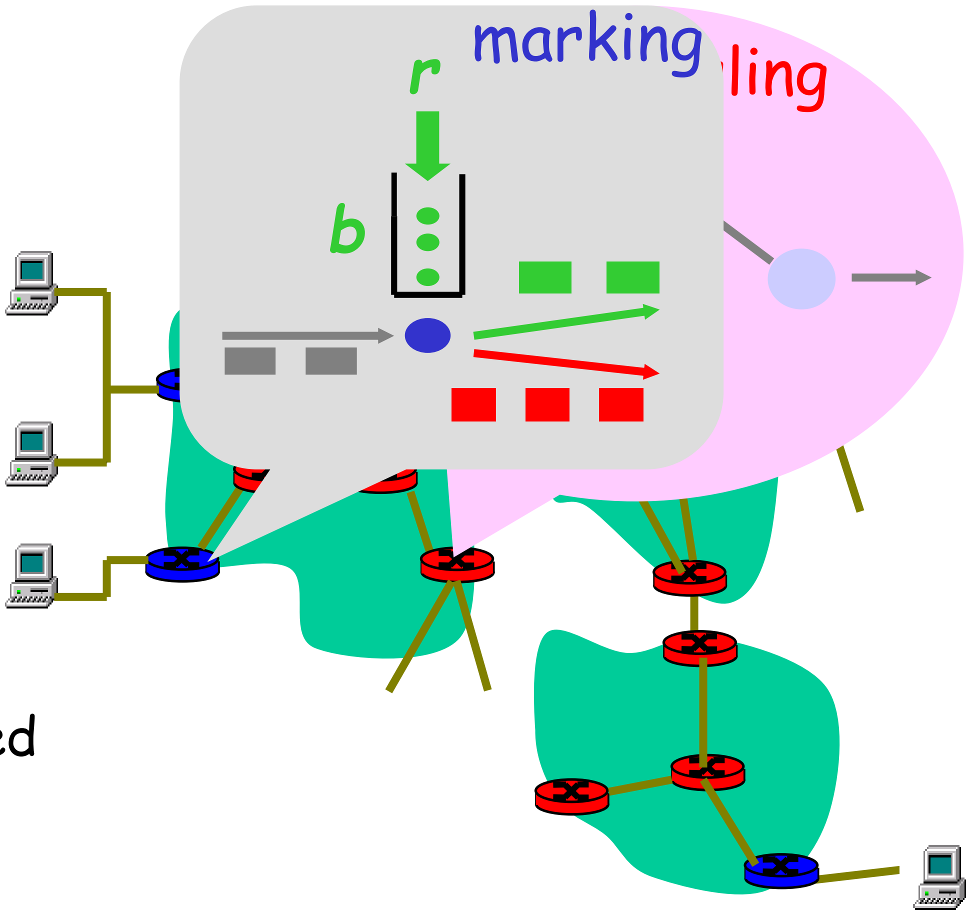
# Diffserv Architecture

## Edge router:

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

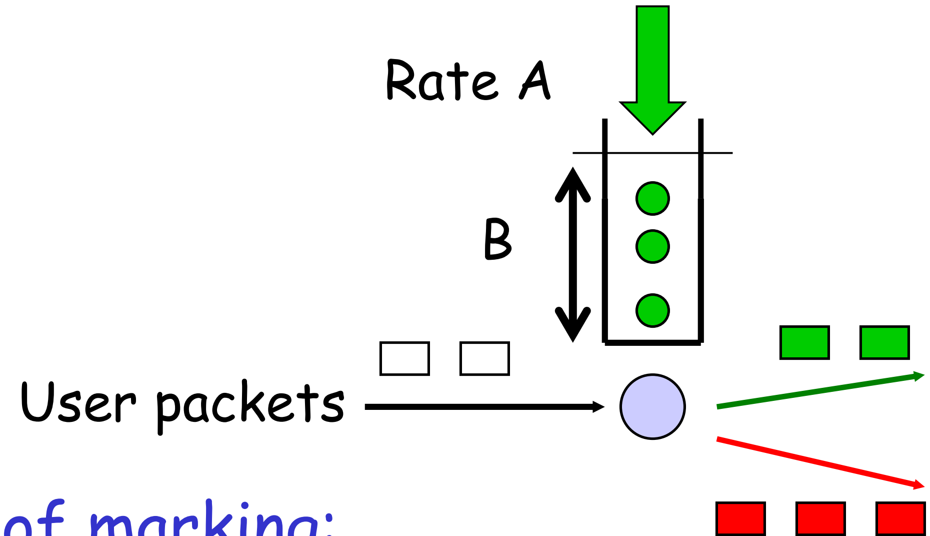
## Core router:

- per class traffic management
- buffering and scheduling based on marking at edge
- preference given to in-profile packets



# 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

# 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?

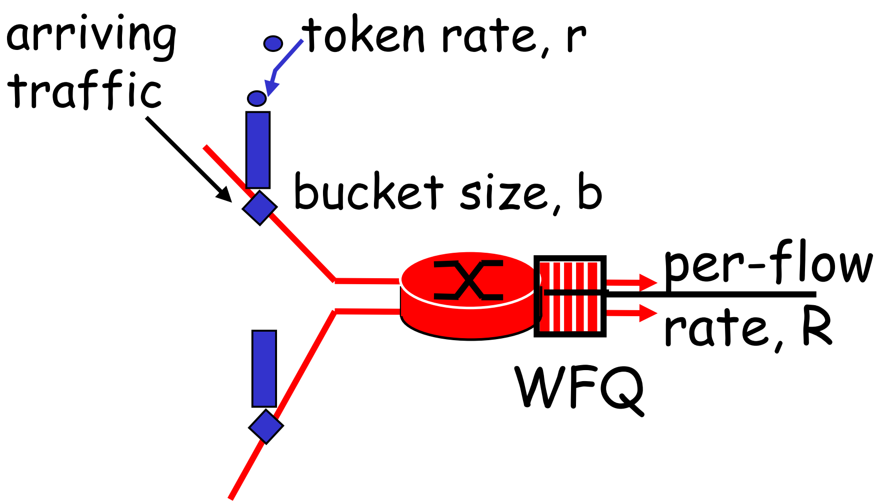
# Intserv QoS: Service models [rfc2211, rfc 2212]

## Guaranteed service:

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

## Controlled load service:

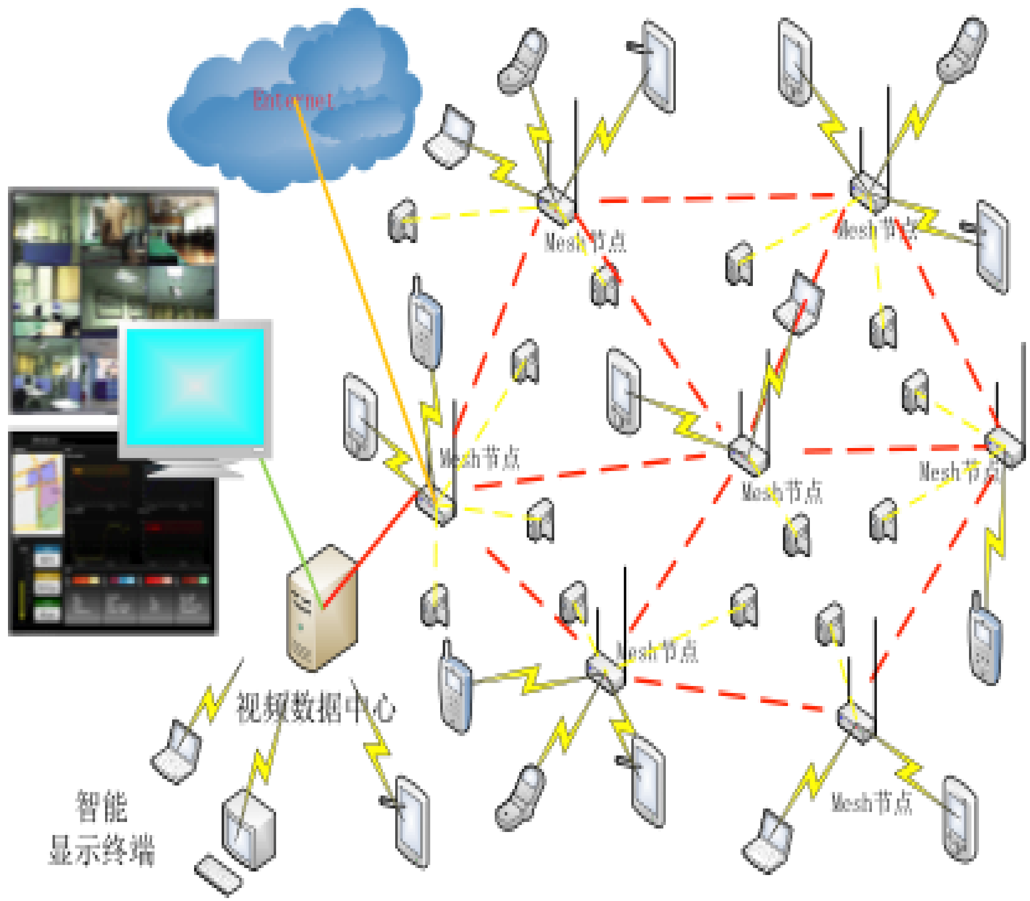
- "a quality of service closely approximating the QoS that same flow would receive from an unloaded network element."




$$D_{max} = b/R$$

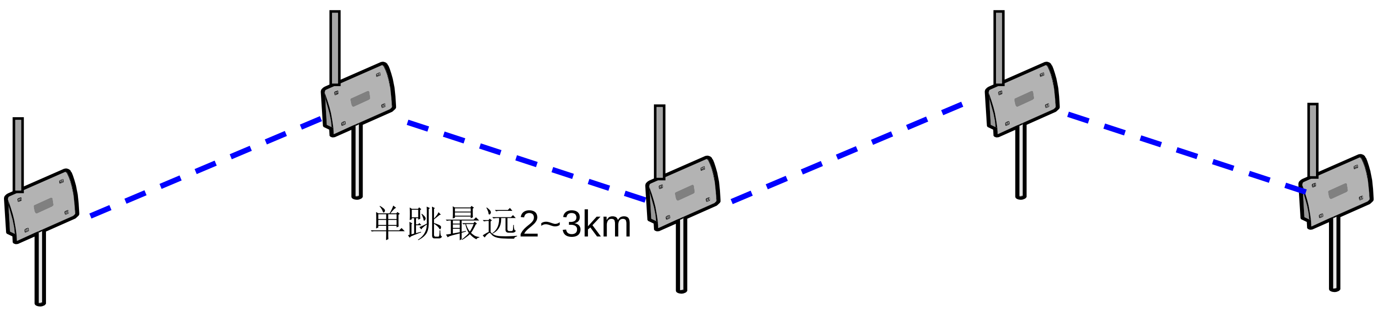
# 案例：利用无线Mesh网络传输视频

- 多跳自组织
  - 自愈合
  - 自组织
- 高带宽
  - 单跳最大150Mbps
- 多用途
  - 语音、视频通话
  - 视频监控、抓拍系统
  - 网络覆盖
- 方便部署
  - 多种供电
  - 加电自动入网



# 高带宽无线MESH网络—优势

- **优势1：多跳组网，通信距离长**



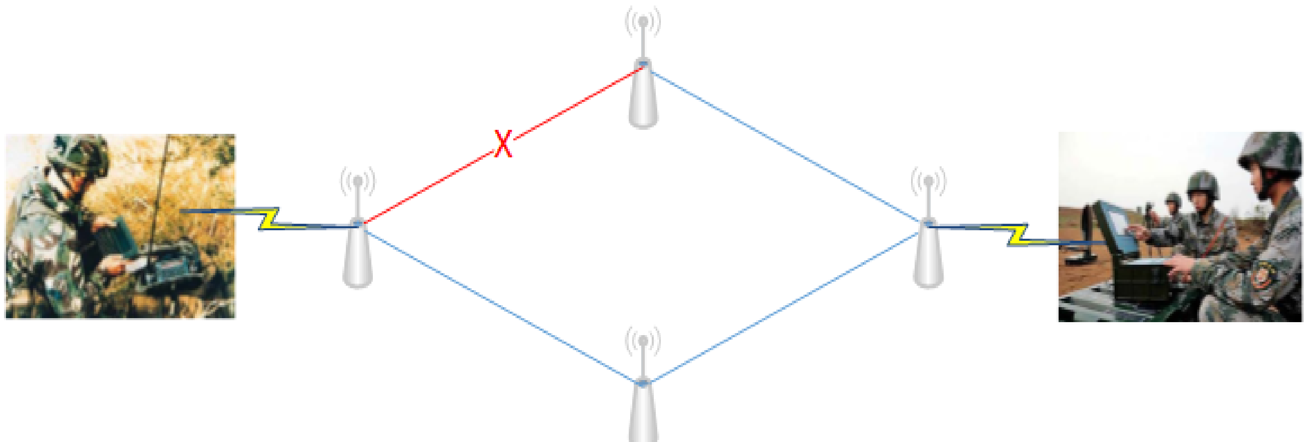

以多跳传输的方式与数十公里外的节点通信

# 高带宽无线MESH网络—优势

- **优势2：带宽高**，支持多路视频传输
  - 最高传输速率：**150Mbps**

# 高带宽无线MESH网络—优势

- **优势3：网络自愈合**



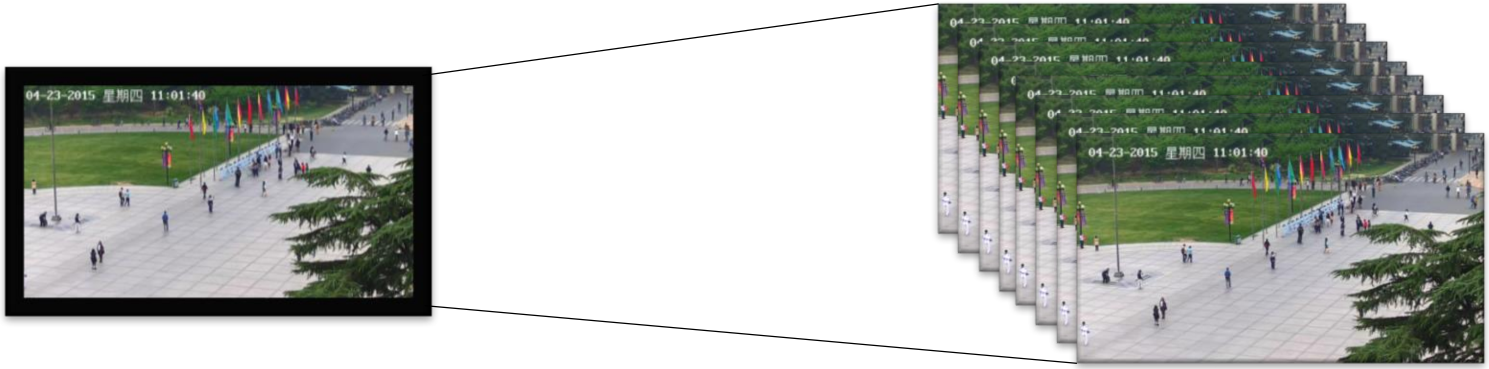

**当一条线路断路时，该骨干网络会自动寻找其他最优路径，保证数据传输的通畅**

# 高带宽无线MESH网络—优势

- **优势4：** 网络自动搭建，**部署、维护简单**
  - 第一步：固定节点
  - 第二步：通电启动
  - 第三步：完成！

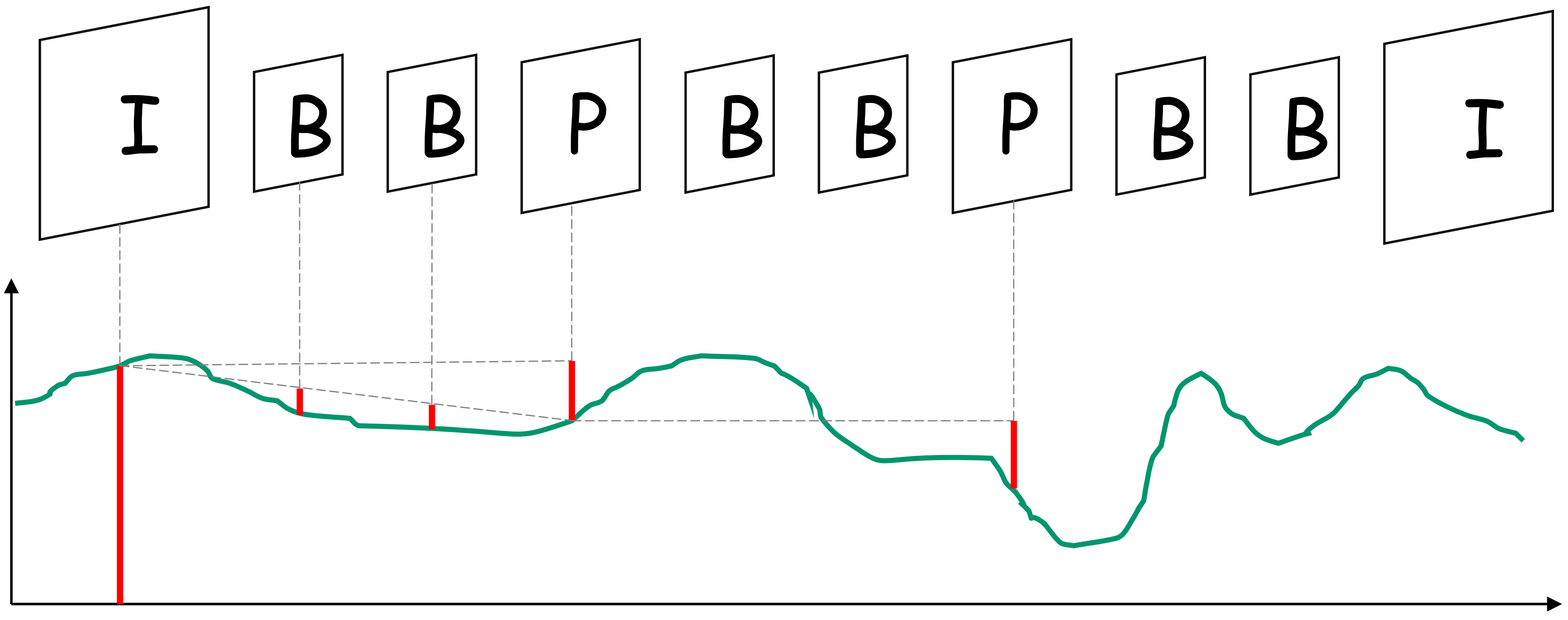
# H.264

- **H.264** is a video compression format that is currently one of the most commonly used formats.
- Video -> Frames -> Groups of Frames



# GOP of H.264

- Three Types of Frames: I, P and B

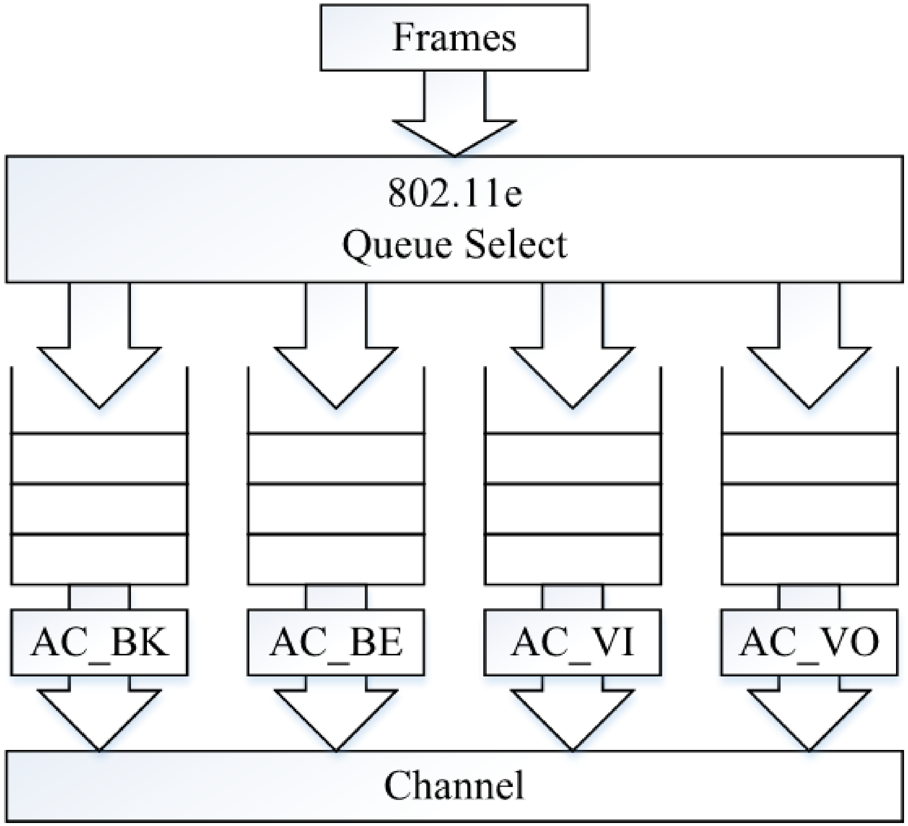


# Importance of H.264 Frames

- Considering Video Disruption Caused by Frame Loss

# 802.11e EDCA

- 802.11 -> DCF
  - 1 Queue
  - Best Effort
- 802.11e -> EDCA
  - 4 Queue
  - Different Priorities



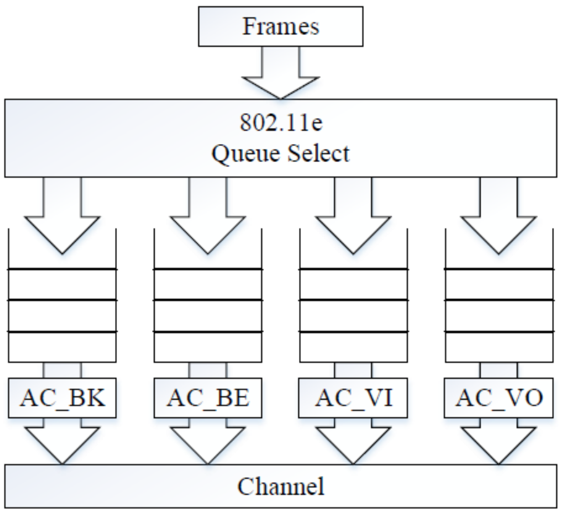
# Mapping

- Higher Importance -> Higher Priority

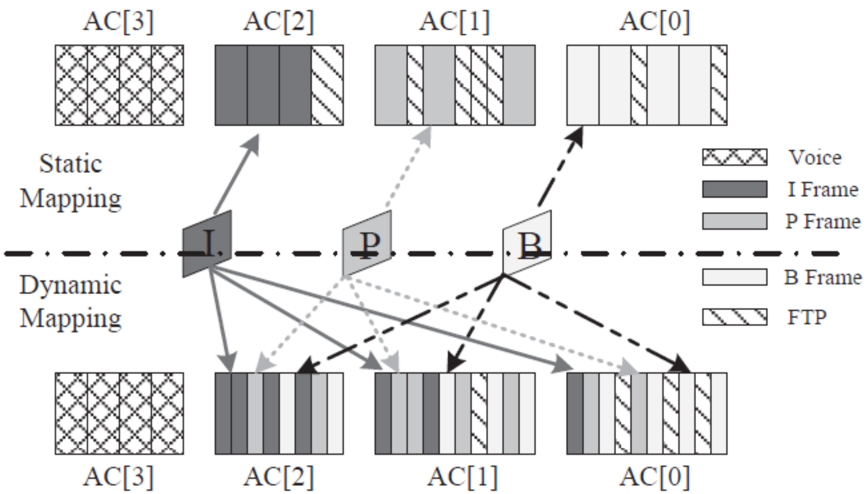
| | | | |
|---|---|---|---|
| I | I | B | B |
| I | P | P | B |
| P | P | B | B |
| I | P | B | B |

IPIIPIPPPBPBBPBB…

# 工作原理

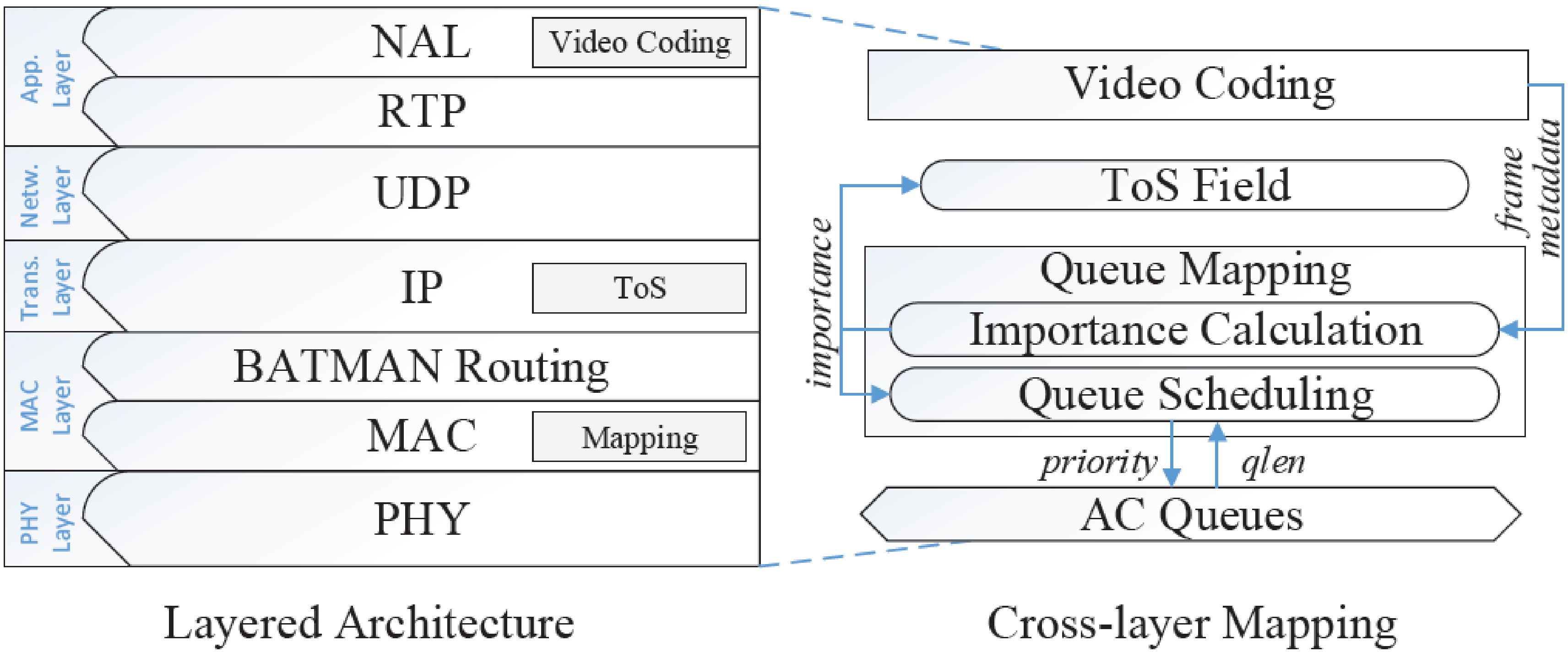



多优先级传输



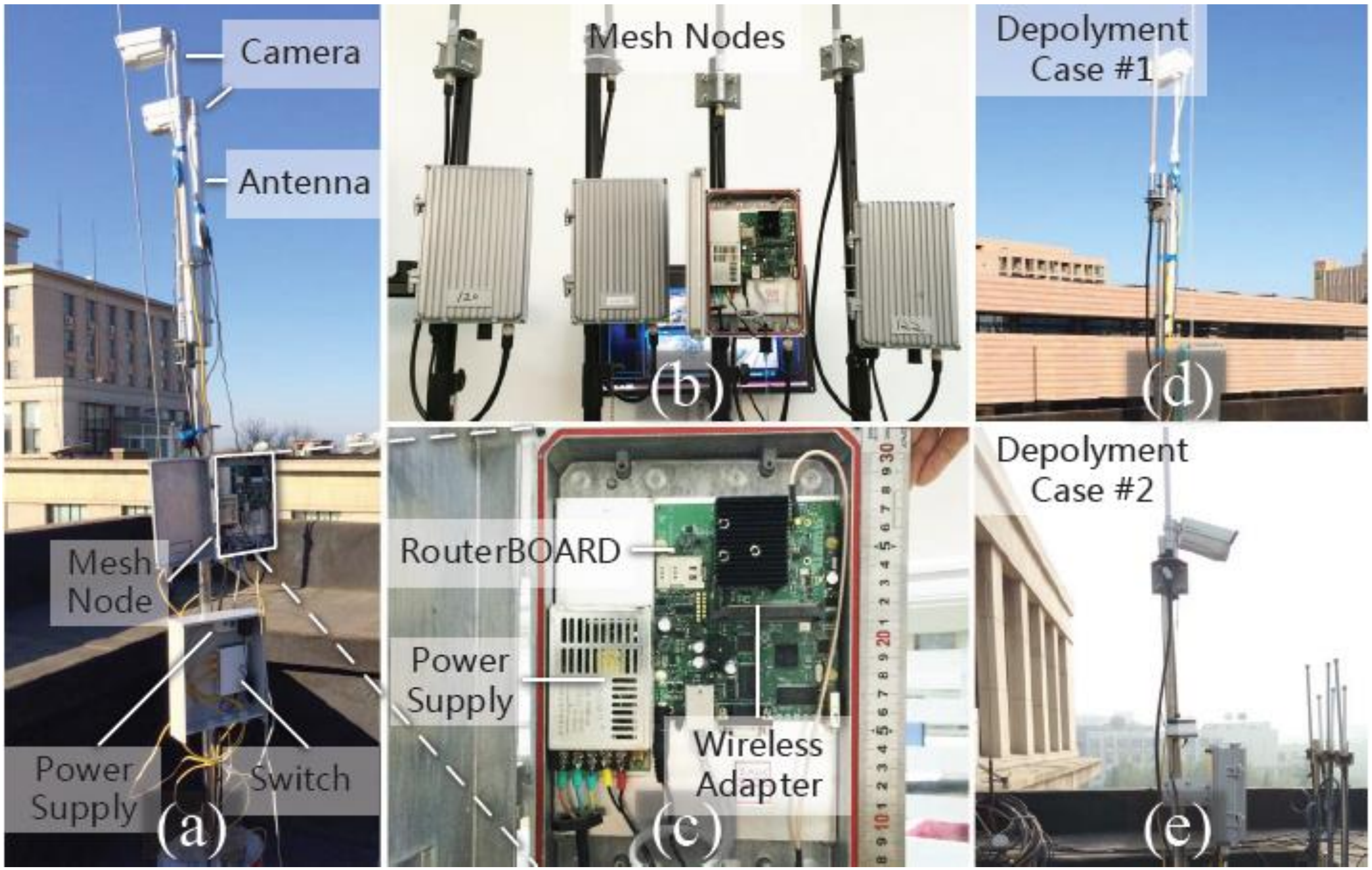

动态数据映射

# 实现方式



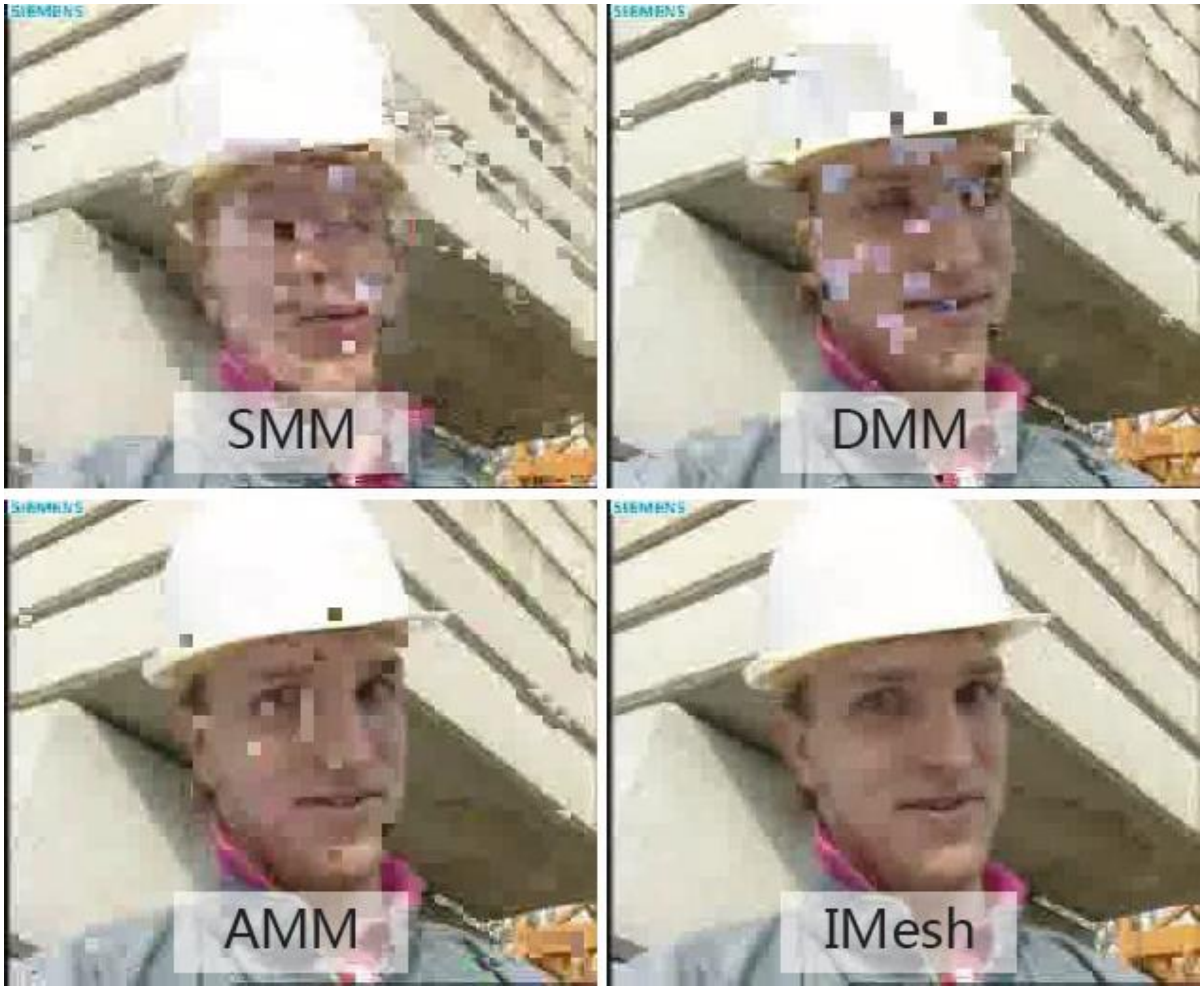

(b) Cross-layer design

# 部署


Camera
Antenna
Mesh Node
Power Supply
Switch
(a)
Mesh Nodes
(b)
RouterBOARD
Power Supply
Wireless Adapter
(c)
Depolyment Case #1
(d)
Depolyment Case #2
(e)

# 部署

# 效果



# 技术特点

多优先级传输

- 在无线自组织网络中实现多优先级队列，不同权重的数据包可以通过不同优先级的队列传输，协议实现与802.11e协议兼容，具有高效、鲁棒等特性

动态数据映射

- 实时探测网络传输质量并针对网络状况动态调整数据包映射策略，确保权重更大的数据包具有更高的链路占用概率，充分利用无线信道容量

跨层协议设计

- 针对主流视频编码协议（例如H.264）设计高效的数据链路层与网络层协议，统一优化多层协议的运行机制和参数，保障多媒体应用在极端恶劣的网络条件下流畅运行。

# Chapter 7: Summary

## Principles

- classify multimedia applications
- identify network services applications need
- making the best of best effort service

## Protocols and Architectures

- specific protocols for best-effort
- mechanisms for providing QoS
- architectures for QoS
  - multiple classes of service
  - QoS guarantees, admission control