Multimedia

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Multimedia Applications

- **Multimedia requirements**
- **Streaming**
- **Phone over IP**
- **Recovering from Jitter and Loss**
- **RTP**
- **Diff-serv, Int-serv, RSVP**

Application Classes

- **Typically sensitive to delay, but can tolerate packet loss (would cause minor glitches that can be concealed)**

- **Data contains audio and video content ("continuous media"), three classes of applications:**
  - **Streaming**
  - **Unidirectional Real-Time**
  - **Interactive Real-Time**

Application Classes (more)

- Streaming
  - **Clients request audio/video files from servers and pipeline reception over the network and display**
  - **Interactive: user can control operation (similar to VCR: pause, resume, fast forward, rewind, etc.)**
  - **Delay: from client request until display start can be 1 to 10 seconds**
  - **Example: RealAudio/RealVideo**

Application Classes (more)

- Unidirectional Real-Time:
  - **similar to existing TV and radio stations, but delivery on the network**
  - **Non-interactive, just listen/view**
  - Example, online course broadcast

- Interactive Real-Time **:**
  - **Phone conversation or video conference**
  - **More stringent delay requirement than Streaming and Unidirectional because of interactive real-time nature**
  - **Video: < 150 msec acceptable**
  - **Audio: < 150 msec good, <400 msec acceptable**

Challenges

- **TCP/UDP/IP suite provides best-effort, no guarantees on expectation or variance of packet delay**

- **Streaming applications delay of 5 to 10 seconds is typical and has been acceptable, but performance deteriorates if links are congested (transoceanic)**

- **Real-Time Interactive requirements on delay and its jitter have been satisfied by over-provisioning (providing plenty of bandwidth), what will happen when the load increases?...**

Challenges (more)

- **Most router implementations use only First-Come-First-Serve (FCFS) packet processing and transmission scheduling**
- **To mitigate impact of "best-effort" protocols, we can:**
  - **Use UDP to avoid TCP and its slow-start phase…**
  - **Buffer content at client and control playback to remedy jitter**
  - **Adapt compression level to available bandwidth**

Solution Approaches in IP Networks

- **Just add more bandwidth and enhance caching capabilities (over-provisioning)!**
- **Two Camps**
  - **Need major change of the protocols (Integrated Services):**
    - **Incorporate resource reservation (bandwidth, processing, buffering), and new scheduling policies**
    - **Set up service level agreements with applications, monitor and enforce the agreements, charge accordingly**
  - **Need moderate changes ("Differentiated Services"):**
    - **Use two traffic classes for all packets and differentiate service accordingly**
    - **Charge based on class of packets**
    - **Network capacity is provided to ensure first class packets incur no significant delay at routers**

Streaming

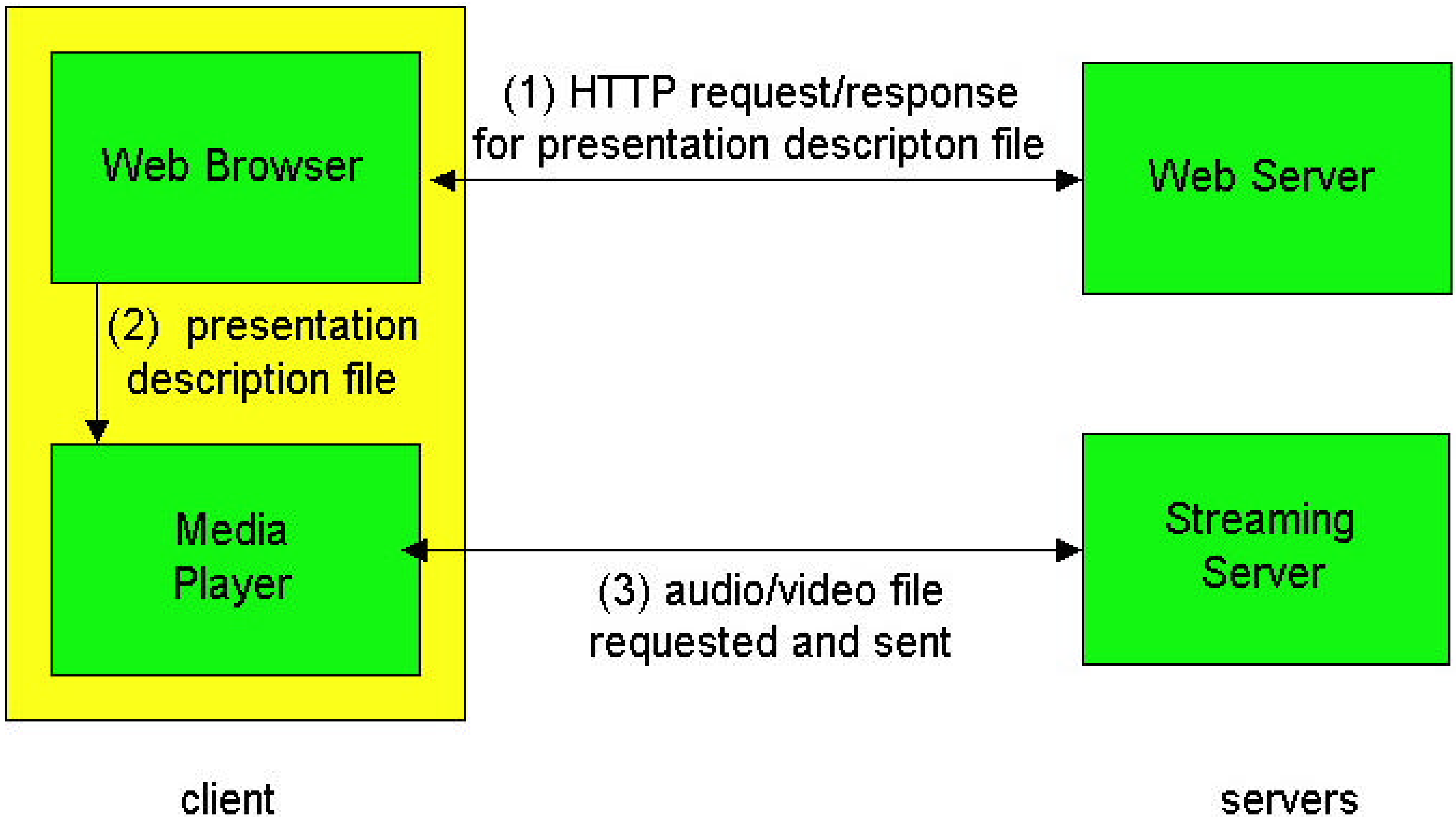
- **Important and growing application due to reduction of storage costs, increase in high speed net access from homes, enhancements to caching and introduction of QoS in IP networks**

- **Audio/Video file is segmented and sent over either TCP or UDP.**
  - **public segmentation protocol:** Real-Time Protocol (RTP)

Streaming

- **User interactive control is provided**
  - **public protocol** Real Time Streaming Protocol (RTSP)
- Helper Application: **displays content, which is typically requested via a Web browser; e.g. RealPlayer; typical functions:**
  - **Decompression**
  - **Jitter removal**
  - **Error correction: use redundant packets to be used for reconstruction of original stream**
  - **GUI for user control**

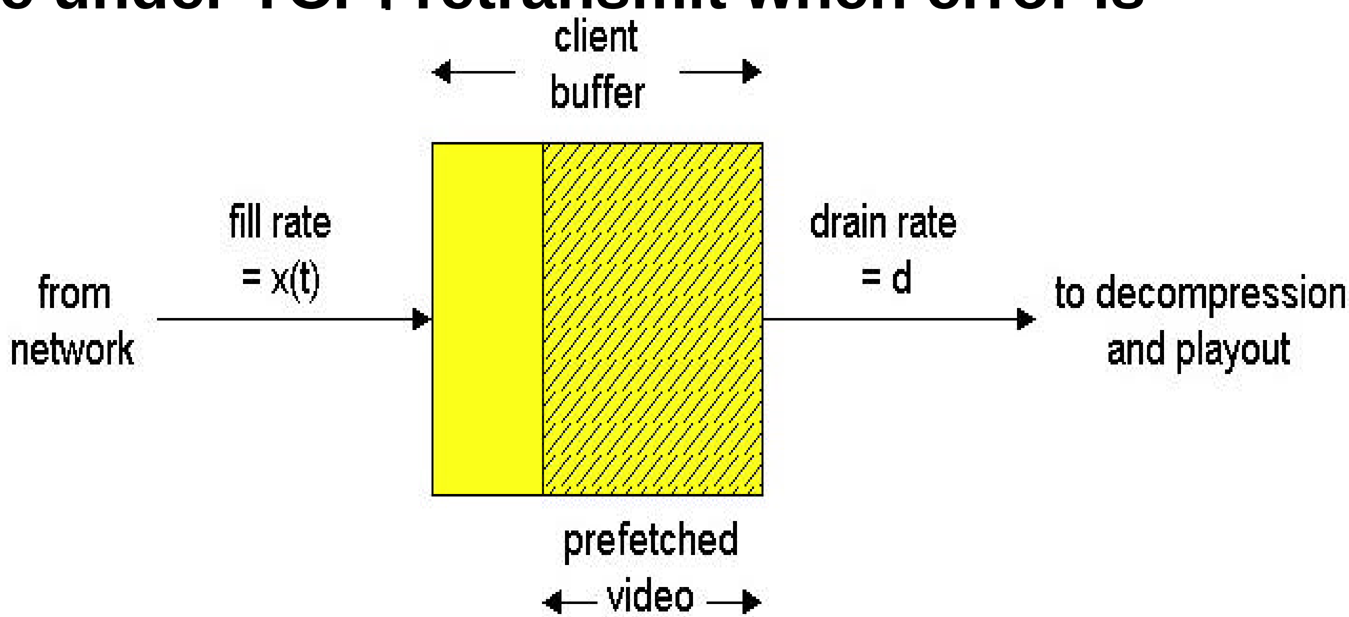
Using a Streaming Server



Options When Using a Streaming Server

- **Use UDP and Server sends at a rate (Compression and Transmission) appropriate for client; to reduce jitter, Player buffers initially for 2-5 seconds, then starts display**

- **Use TCP and sender sends at maximum possible rate under TCP; retransmit when error is**



Real Time Streaming Protocol (RTSP)

- **For user to control display: rewind, fast forward, pause, resume, etc…**
- **Out-of-band protocol (uses two connections, one for control messages (Port 554) and for media stream)**
- **RFC 2326 permits use of either TCP or UDP for the control messages connection, sometimes called the RTSP Channel**
- **As before, meta file is communicated to web browser which then launches the Player; Player sets up an RTSP connection for control messages in addition to the connection for the streaming media**

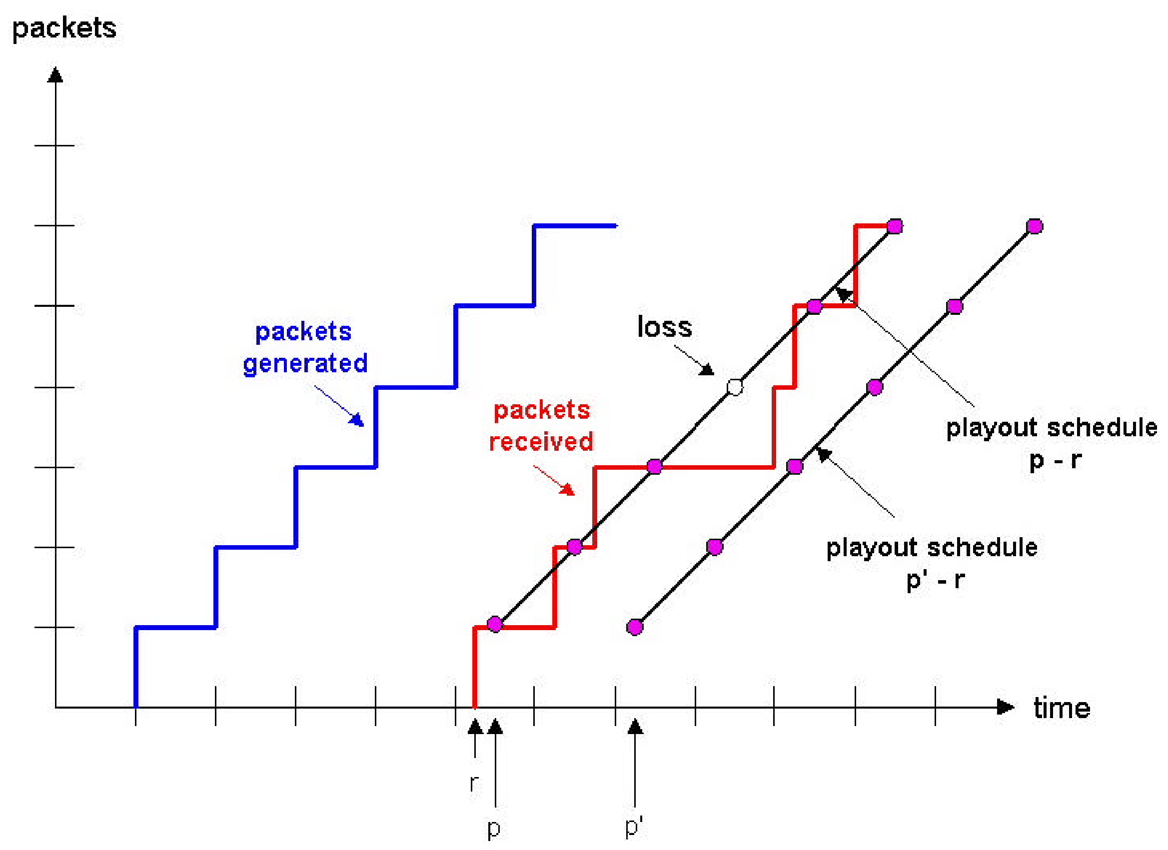
Real-Time (Phone) Over IP's Best-Effort

- **Internet phone applications generate packets during talk spurts**
- **Bit rate is 8 KBytes, and every 20 msec, the sender forms a packet of 160 Bytes + a header to be discussed below**
- **The coded voice information is encapsulated into a UDP packet and sent out; some packets may be lost; up to 20 % loss is tolerable; using TCP eliminates loss but at a considerable cost: variance in delay; FEC is sometimes used to fix errors and make up losses**

Real-Time (Phone) Over IP's Best-Effort

- **End-to-end delays above 400 msec cannot be tolerated; packets that are that delayed are ignored at the receiver**
- **Delay jitter is handled by using timestamps, sequence numbers, and delaying playout at receivers either a fixed or a variable amount**
- **With fixed playout delay, the delay should be as small as possible without missing too many packets; delay cannot exceed 400 msec**

Internet Phone with Fixed Playout Delay



Adaptive Playout Delay

- **Objective is to use a value for playout delay that tracks the network delay performance as it varies during a phone call**
- **The playout delay is computed for each talk spurt based on observed average delay and observed deviation from this average delay**
- **Estimated average delay and deviation of average delay are computed in a manner similar to estimates of RTT and deviation in TCP**
- **The beginning of a talk spurt is identified from examining the timestamps in successive and/or sequence numbers of chunks**

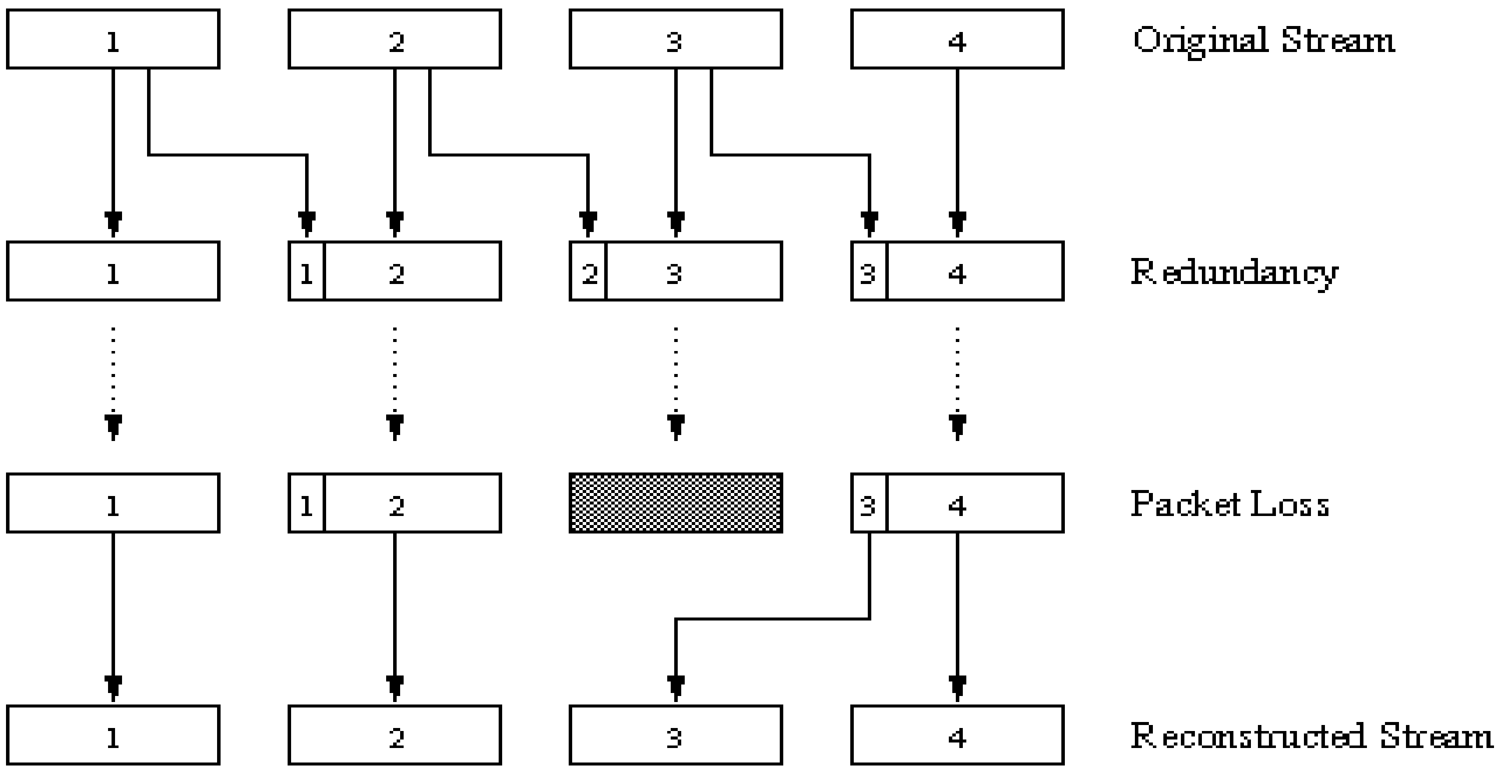
Recovery From Packet Loss

- **Loss is in a broader sense: packet never arrives or arrives later than its scheduled playout time**
- **Since retransmission is inappropriate for Real Time applications, FEC or Interleaving are used to reduce loss impact.**
- **FEC is Forward Error Correction**
- **Simplest FEC scheme adds a redundant chunk made up of exclusive OR of a group of n chunks; redundancy is 1/n; can reconstruct if at most one lost chunk; playout time schedule assumes a loss per group**

Recovery From Packet Loss

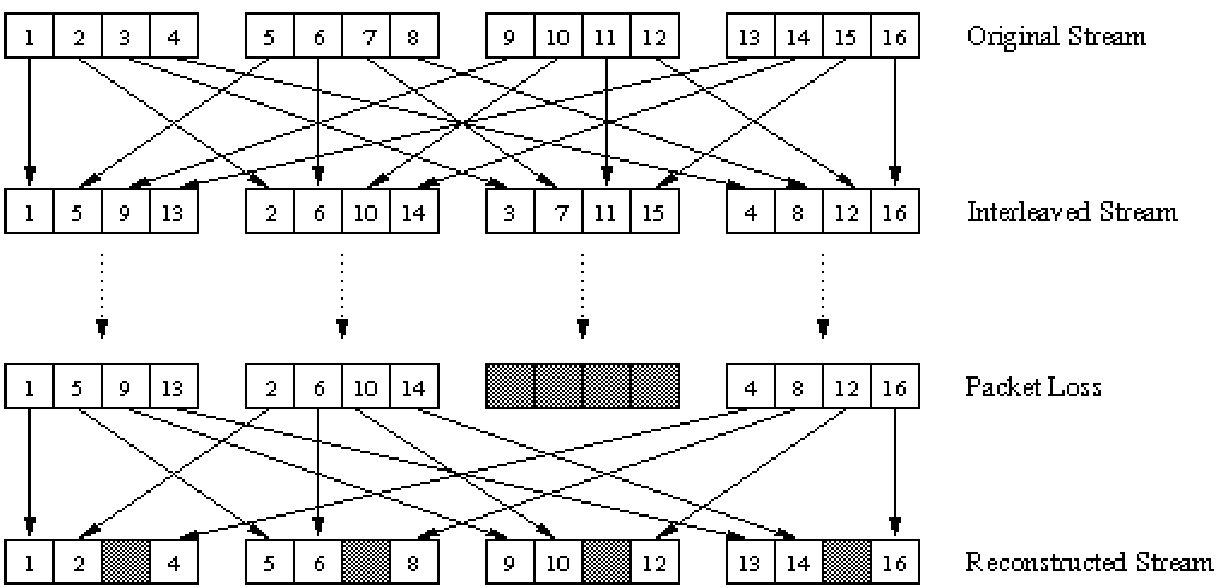
- **Mixed quality streams are used to include redundant duplicates of chunks; upon loss a lower quality redundant chunk is available.**
- **With one redundant chunk per chunk can recover from single losses**

Piggybacking Lower Quality Stream



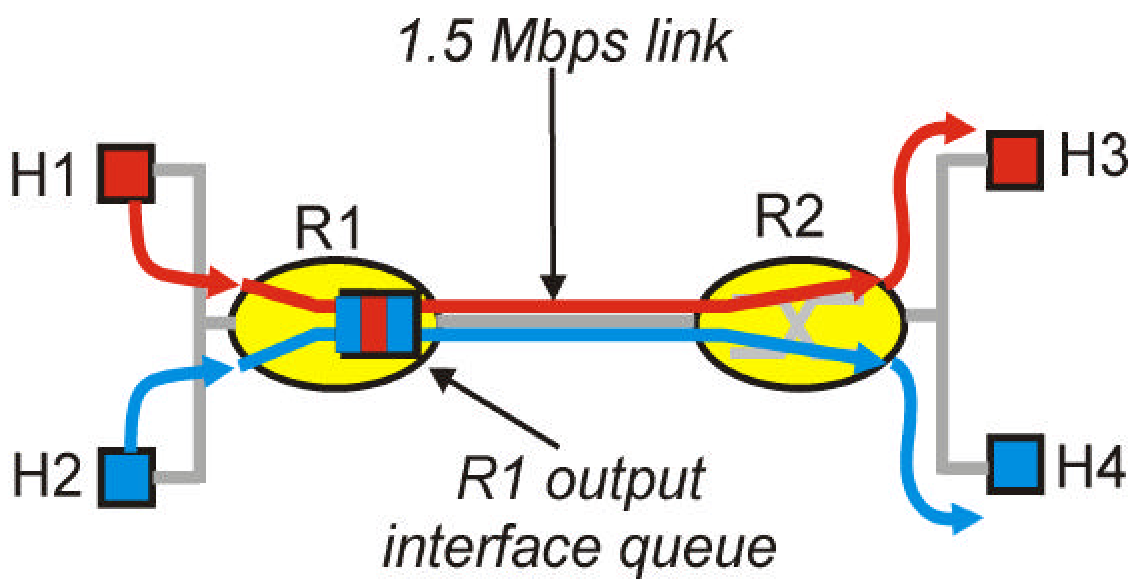
Interleaving

- **Has no redundancy, but can cause delay in playout beyond Real Time requirements**
- **Divide 20 msec of audio data into smaller units of 5 msec each and interleave**
- **Upon loss, have a set of partially filled chunks**



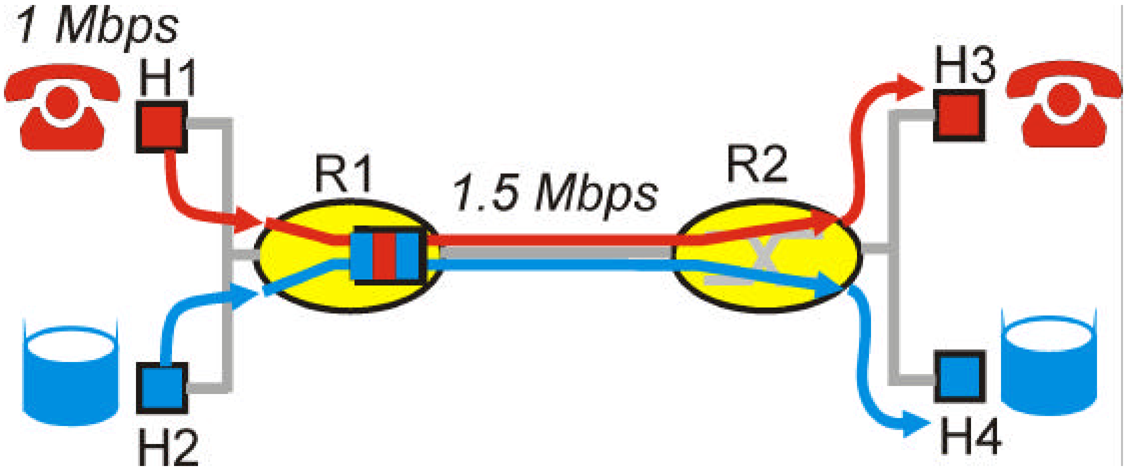
Improving QOS in IP Networks

- **IETF groups are working on proposals to provide better QOS control in IP networks, i.e., going beyond best effort to provide some assurance for QOS**
- **Work in Progress includes RSVP, Differentiated Services, and Integrated Services**
- **Simple model for sharing and congestion studies:**



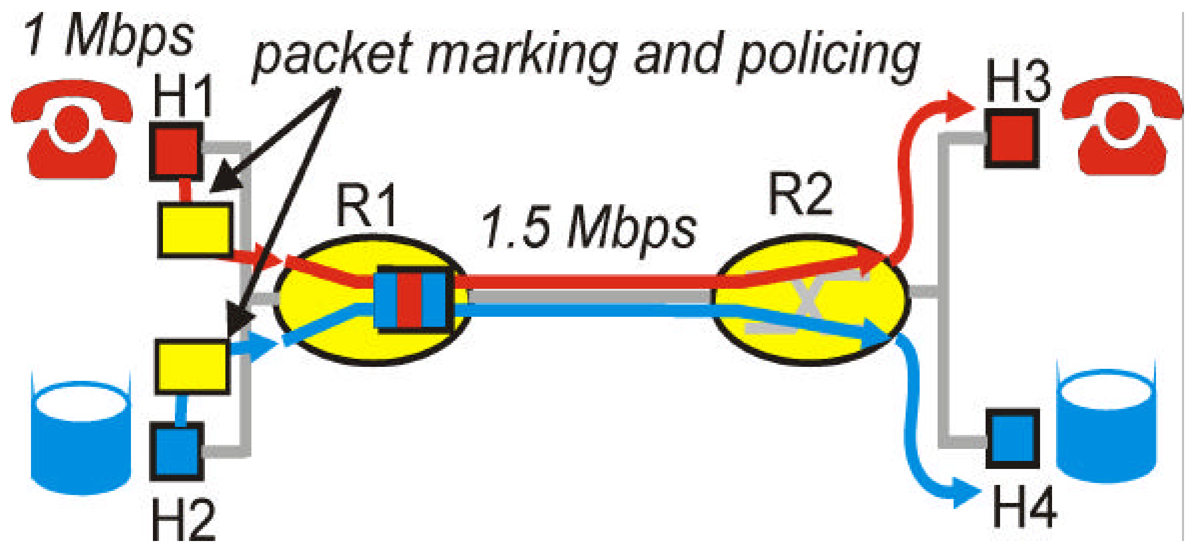
Principles for QOS Guarantees

- **Consider a phone application at 1Mbps and an FTP application sharing a 1.5 Mbps link.**
  - **bursts of FTP can congest the router and cause audio packets to be dropped.**
  - **want to give priority to audio over FTP**
- PRINCIPLE 1: Marking of packets is needed for router to distinguish between different classes; and new router policy to treat packets accordingly



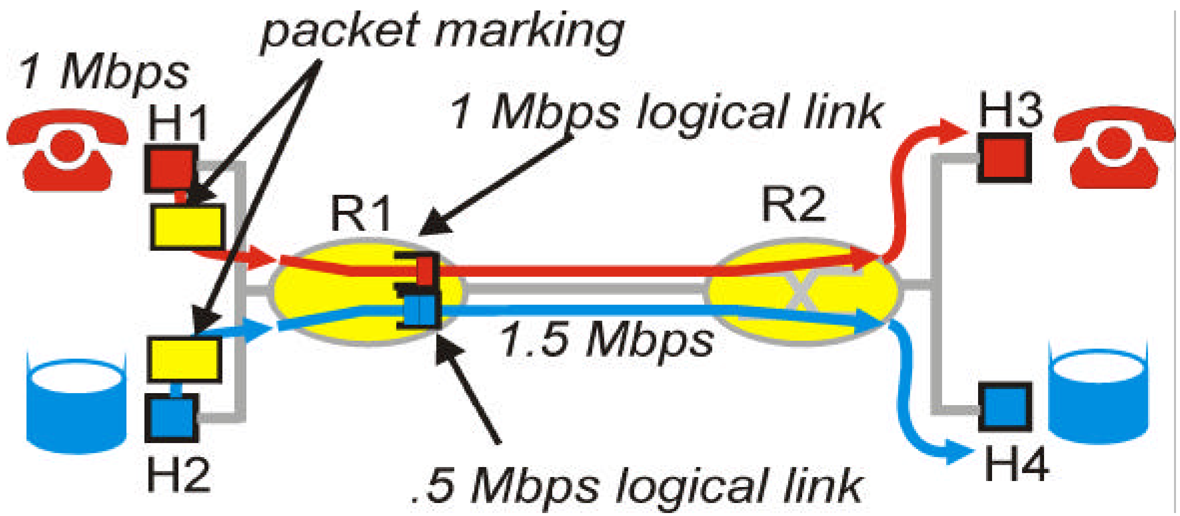
Principles for QOS Guarantees (more)

- **Applications misbehave (audio sends packets at a rate higher than 1Mbps assumed above);**
- PRINCIPLE 2: provide protection (isolation) for one class from other classes
- **Require Policing Mechanisms to ensure sources adhere to bandwidth requirements; Marking and Policing need to be done at the edges:**



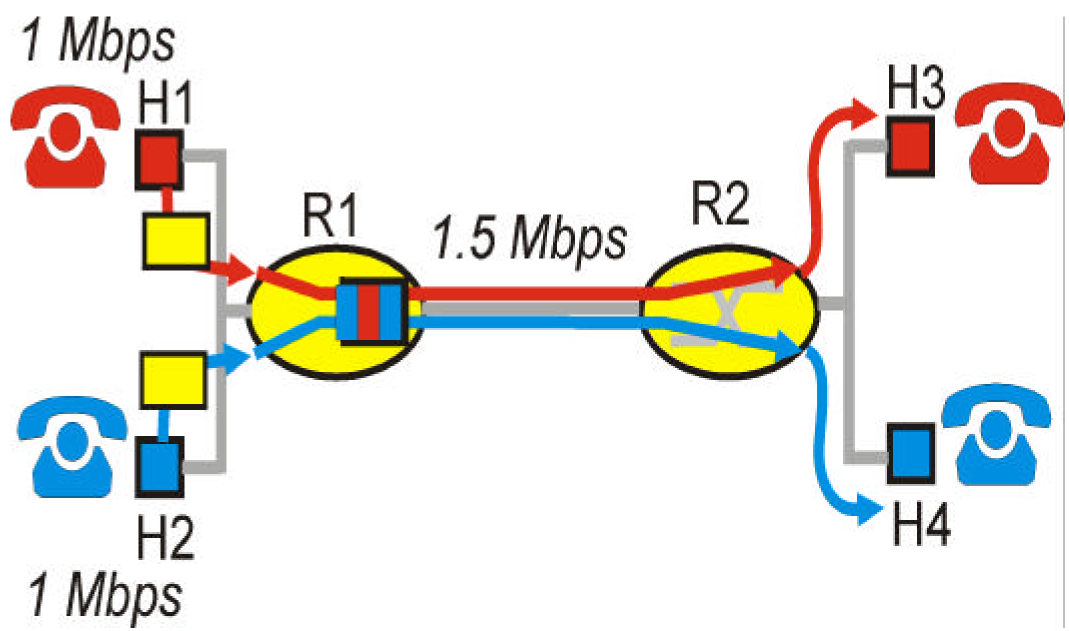
Principles for QOS Guarantees (more)

- **Alternative to Marking and Policing: allocate a set portion of bandwidth to each application flow; can lead to inefficient use of bandwidth if one of the flows does not use its allocation**
- PRINCIPLE 3: While providing isolation, it is desirable to use resources as efficiently as possible

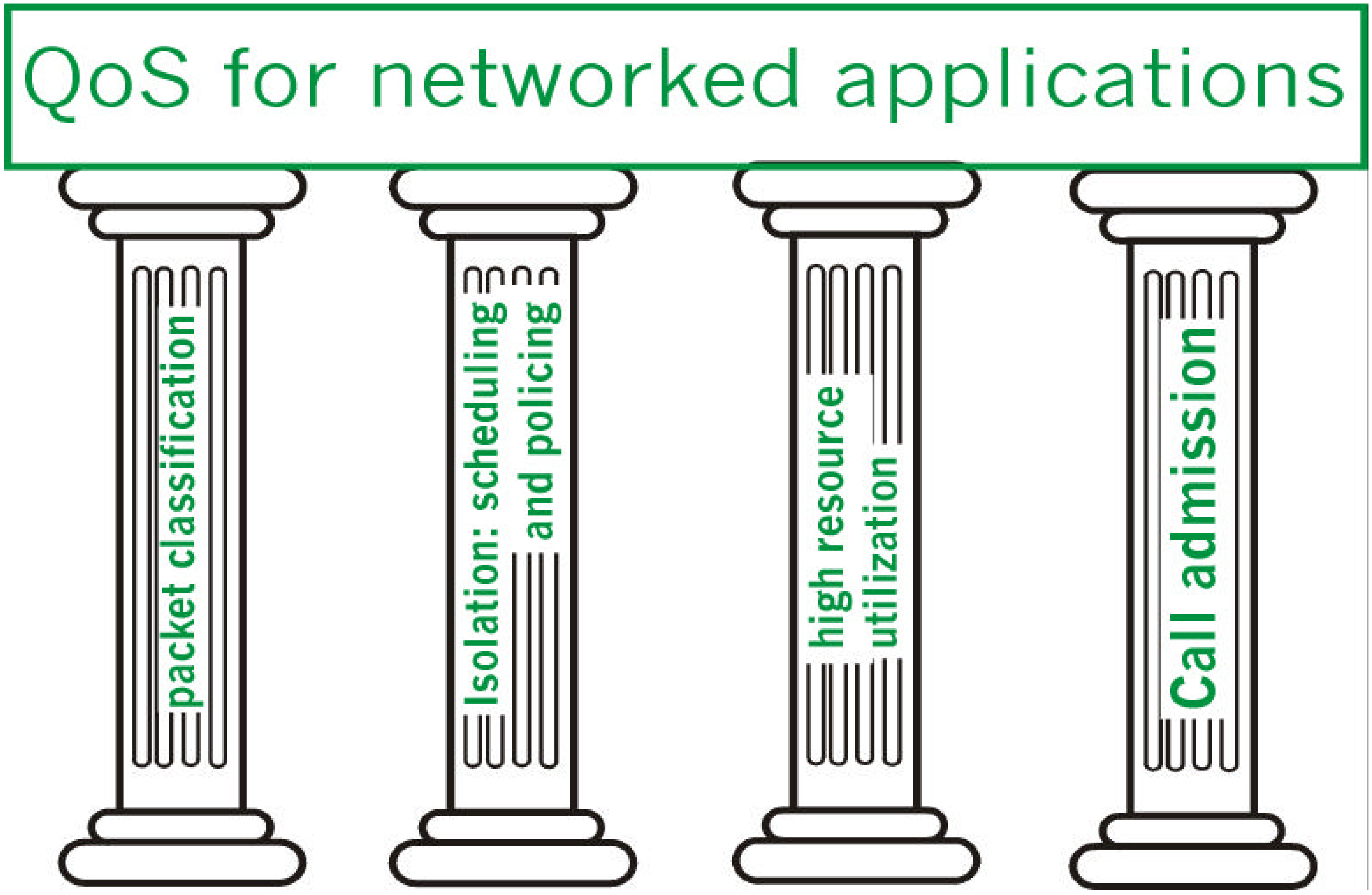


Principles for QOS Guarantees (more)

- **Cannot support traffic beyond link capacity**
- **PRINCIPLE 4: Need a Call Admission Process; application flow declares its needs, network may block call if it cannot satisfy the needs**

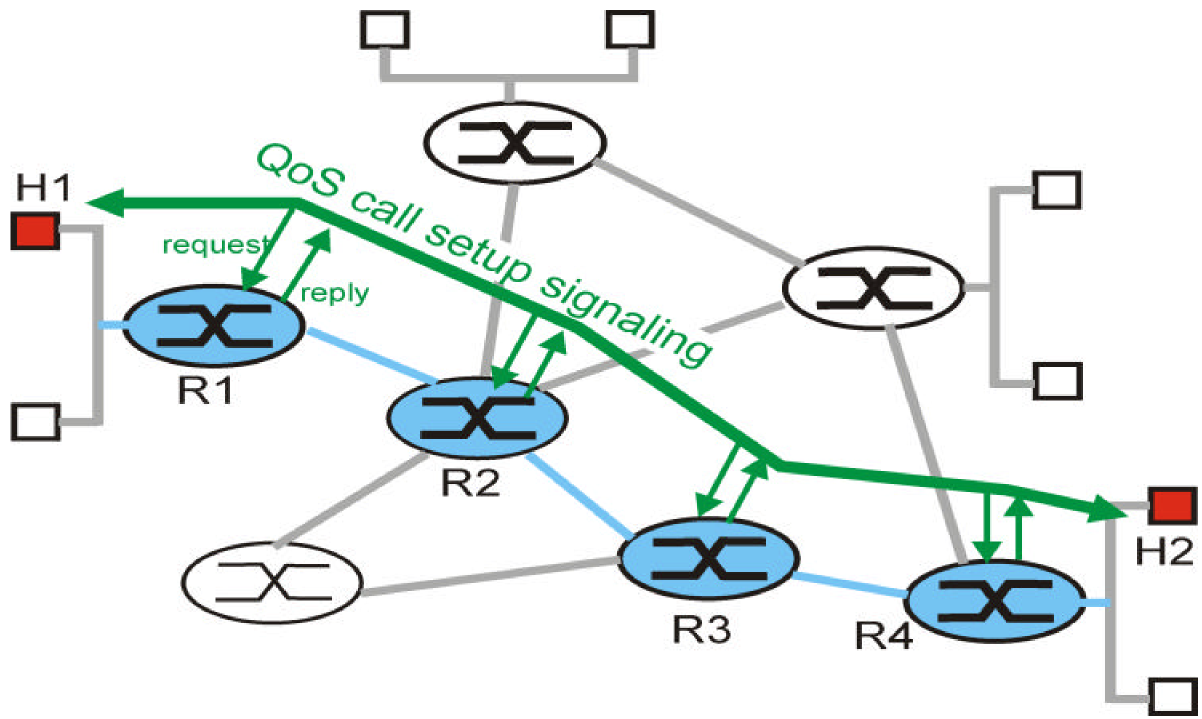


Summary


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

Integrated Services

- **An architecture for providing QOS guarantees in IP networks for individual application sessions**
- **relies on resource reservation, and routers need to maintain state info (Virtual Circuit??), maintaining records of allocated resources and responding to new Call setup requests on that basis**



Call Admission

- **Session must first declare its QOS requirement and characterize the traffic it will send through the network**
- R-spec**: defines the QOS being requested**
- T-spec**: defines the traffic characteristics**
- **A signaling protocol is needed to carry the R-spec and T-spec to the routers where reservation is required; RSVP is a leading candidate for such signaling protocol**

Differentiated Services

- **Intended to address the following difficulties with Intserv and RSVP;**
- Scalability**: maintaining states by routers in high speed networks is difficult due to the very large number of flows**
- Flexible Service Models**: Intserv has only two classes, want to provide more qualitative service classes; want to provide 'relative' service distinction (Platinum, Gold, Silver, Lead …)**
- Simpler signaling**: (than RSVP) applications and users may only want to specify a more qualitative notion of service**

Differentiated Services

- **Approach:**
  - **Only simple functions in the core, and relatively complex functions at edge routers (or hosts)**
  - **Do not define service classes, instead provides functional components with which service classes can be built**