Network Support for Next-generation Applications

Data Communication: A Historical Perspective

- Traditionally voice, video, and textual data have been handled by different communication networks
  - Voice over telephone networks
  - Video over cable TV networks
  - Textual data over computer networks
- Reasons for the separation: fundamental differences in the characteristics of voice and video from textual data
  - Voice and video: real-time, requires bounded delay, tolerant to brief loss of information
  - Textual data: Tolerant to delay but cannot tolerate any transmission error or loss
- Two fundamental mechanisms: circuit switching and packet switching

Circuit Switching

- Characteristics:
  - Dedicated physical path from source to destination for the entire call duration
  - Fixed and dedicated bandwidth allocation
  - Fixed end-to-end delay
  - No data processing at intermediate nodes $\Rightarrow$ no overhead at intermediate nodes
  - Information delivery guaranteed to be sequenced
- Disadvantages:
  - Bursty data may yield severe underutilization of network resources
  - Call setup requires $\approx$ 1 sec $\Rightarrow$ short sessions are not cost effective
  - Unicast by nature

Packet Switching

- Characteristics:
  - Dynamic allocation of bandwidth
  - Store-and-forward switching
  - Extremely flexible: supports both unicast and multicast transmission
  - Suitable for bursty traffic, permits network resources to be multiplexed among several channels
- Disadvantages:
  - Variable end-to-end delay due to queuing at switches
  - Information delivery may not be sequenced or reliable

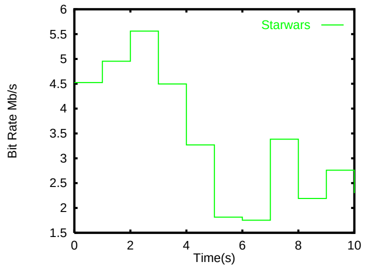
Integrated Networks

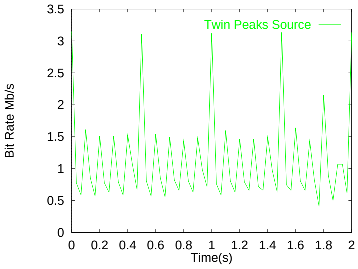
- Motivations for integration:
  - Availability of high speed networks at modest cost; relatively inexpensive hardware for digitization and compression of media streams
  - Convenience, flexibility, efficiency, and economy
- Levels of integration:
  - Integrated access
  - Integrated transmission
  - Integrated switching
    * Complete integration
    * Technically most challenging
    * Conventional media networks: circuit switched, data networks: packet switched, integrated networks: ??

Protocols for Video Transport

Network Protocols for Video

- Variable Bit Rate (VBR) compressed video
- Key characteristic: significant variation at multiple time-scale
  - Slow time-scale variation: scene complexity
  - Fast time-scale variation: interframe compression algorithms





- Overbooking to utilize resources efficiently

Designing Protocols for Video

- Overbooking $\Rightarrow$ network congestion $\Rightarrow$ packet loss
- Compressed video not resilient to packet loss
- Possible solutions:
  - Techniques to recover from packet losses
    - Retransmission-based schemes (ARQ)
    - Forward error correction (FEC) schemes
    - Multi-level or multi-resolution coding + prioritized transmission
  - Avoid packet losses
    - Traffic shaping at the source
    - Rate control using network feedback

ARQ and FEC Schemes

- Both ARQ and FEC are transport protocol functions
- ARQ:
  - Closed-loop technique based on retransmissions
  - Each retransmission adds at least one round-trip time latency $\Rightarrow$ not applicable for low latency applications
  - Overhead for maintaining large number of outstanding packets
- FEC:
  - Open-loop technique $\Rightarrow$ well suited for high bandwidth - delay product networks
  - Basic idea: Transmit redundant information along with the original data

Lossless Smoothing of Video

- **Objective**: smooth the rate fluctuations in a video sequence
- Simple approach:
  - Transmit all the frames in a group-of-pictures (or in repeated encoding pattern) at the same rate
  - Problem: increased delay
- Key problem: given a delay bound D, derive a transmission schedule

Lossless Smoothing Algorithm

- **Parameters**:

D	maximum delay for each frame
K	number of frames buffered prior to initiating transmission
τ	inter-frame period
S_i	size of ith frame
H	Lookahead interval
t_i	time when transmission of ith frame begins
d_i	time when the ith frame is completely transmitted
r_i	rate of transmission for ith frame

- **Constraints**:
  - Delay: $D \geq (K+1)\tau; \quad t_i = \max\{d_{i-1}, (i+K-1)\tau\}$
  $$d_i - (i-1)\tau \leq D$$
  - Continuous service: $t_i = d_{i-1}$

Determination of r_i

- Deadline constraint:

$$t_i + \frac{\sum_{m=0}^{h} S_{i+m}}{r_i} - (i+h-1)\tau \leq D$$

$$\Rightarrow r_i \geq \frac{\sum_{m=0}^{h} S_{i+m}}{D + (i+h-1)\tau - t_i}$$

- Continuous service:

$$d_{i+h} = t_i + \frac{\sum_{m=0}^{h} S_{i+m}}{r_i} \geq (i+h)\tau$$

$$r_i \leq \frac{\sum_{m=0}^{h} S_{i+m}}{(i+h)\tau - t_i}$$

- Goal: find the value of r_i that is between the bounds for as large a value of h as possible

Protocol for Stored Video Transport

Observations

- Stored video characteristics:
  - Delay insensitive
  - Bit rate requirement known a priori
- Protocols for stored video transport:
  - Smooth out fast time-scale and manage slow time-scale
  - Combine open-loop and feedback based control techniques

Smoothing of Stored Video

- Key difference: delay constraint is less stringent $\Rightarrow$ larger smoothing intervals are possible
- Simple approach:
  - Convert a VBR video stream into a CBR video stream by buffering
  - Disadvantages: large buffer space requirement and initiation latency
- **Solution**: piecewise CBR transmission
- Objective: derive a piecewise CBR schedule that minimizes the rate changes

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Reducing Rate Variability

- Parameters:

b	client buffer capacity
$D(t)$	Amount of data consumed by client by time t
$B(t)$	Maximum cumulative data that can be received by client by time t

- Deriving rate bounds:

$$c_{\max} = \min_{a+1 \le t \le b} \frac{B(t) - (D(a) + q)}{t - a}$$

$$c_{\min} = \max_{a+1 \le t \le b} \frac{D(t) - (D(a) + q)}{t - a}$$

- Rate selection criteria:
  - Select a rate that is valid as long as possible
  - Change the rate as early as possible

Congestion Control Protocol for Stored Video

- Exploit predictability
  - Sources: send bit rate requirement
  - Network: send feedback rate to avoid congestion
  - Sources: send at feedback rate
- Packet loss at source
- Infrequent and regular rate changes of source

Multimedia Servers: Review

- Periodic nature of video playback: service in *rounds*
- Round: access a fixed number of frames
- Data retrieved in one round transmitted in next
- Network transmission
  - Infrequent and regular rate changes: transmit at average rate of a round

Admission Control

- Histogram-based traffic characterization
  - Computed at time of storage
- Admission control based on loss probability
- Algorithm:
  - Order the sources in decreasing order of QoS
  - Ensure that

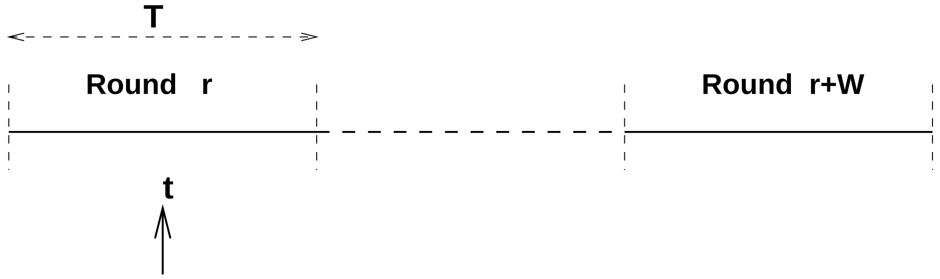
$$P\left(\sum_{i=1}^{i=j} \mathcal{X}_i > C\right) \leq q_j$$

  - Utilize Z-transform or CLT to evaluate the admission control

Overload Control

- Statistical admission control $\Rightarrow$ transient overloads
- Overload: aggregate rate requirement exceeds link capacity
- Key challenge: avoid overload
- Requirements:
  - Rate changes at most at round boundary
  - Rate available for a round should be known at least Δ_o prior to initiation
- Assumptions:
  - Round time of each source T
    - No phase relationship assumed
  - Switches periodically detect and control overload (period=T)

Overload Detection



- t: time at which switch detects overload
- Control overload for round $r + W$

$$W = \left\lceil \frac{\Delta_d + \Delta_c + \Delta_f + \Delta_o}{T} \right\rceil + 1$$

Δ_d : maximum delay of data packet
Δ_c : computation time of overload procedure
Δ_f : maximum delay of feedback packet

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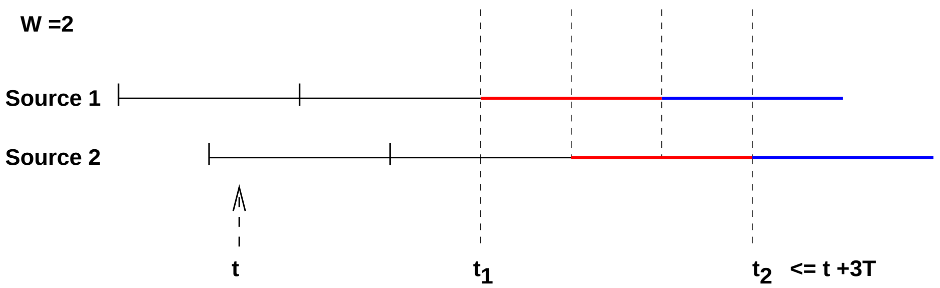
Overload Detection (Cont'd)



- No sub-interval intersects with more than one round of any source
- Rate of rounds $[r_i, r_i + W + 1]$ should be known

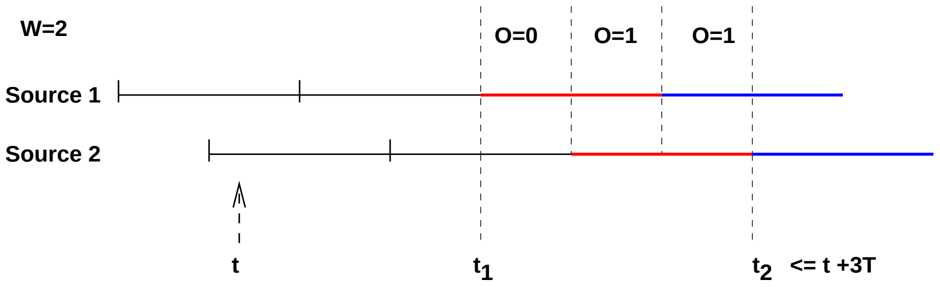
Rate Assignment

- Objective: determine rates such that
  - Overload avoided in each of the sub-intervals
  - QoS of the sources satisfied
- Loss affordability: rate that can be reduced without violating QoS

$$\mathcal{A}_i = \max \left\{ 0, q_i \sum_{j=1}^{j=r_i+W+1} \gamma_i^j - \sum_{j=1}^{j=r_i+W+1} (\gamma_i^j - \hat{\gamma}_i^j) \right\}$$

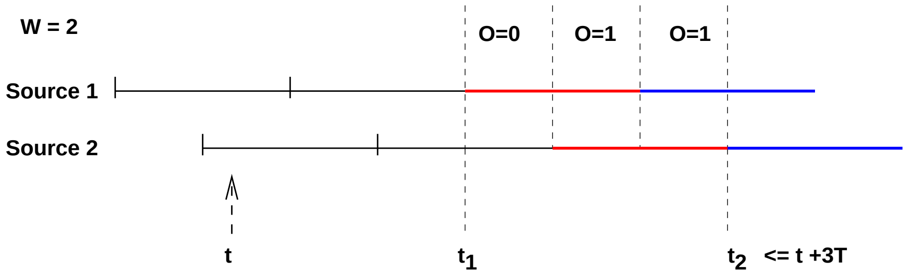
Rate Assignment (Continued)

- Naive approach
  - Consider each sub-interval in isolation
  - **Problem**: higher aggregate rate reduction than necessary



Rate Assignment Algorithm

- Rate assignment problem formulated as a linear program
- Objective: minimize the aggregate rate reduction
- Optimal $O(N)$ algorithm
  - N: number of sources
- Basic idea:
  - Process sub-intervals form left-to-right
  - Reduce rate of sources from bottom-to-top



Overload Control Protocol

- Source sends bit rate requirement of first $W + 1$ rounds in connection establishment message
- Once connection is established:
  - Identify first packet of round (e.g., RM cell)
  - In first packet of round n, send bit rate of round $n + W + 1$
- Each switch independently controls overload and sends feedback
- No feedback: source sends at requested rate
- Multiple feedback: source sends at minimum feedback rate

Salient Features

- Combines open-loop and feedback based control
- Heterogeneous QoS without scheduling support
- Packet losses at source that understands semantics
- Δ_o parameter allows disk bandwidth to be utilized as per network bandwidth
- Integrated with the architecture of a multimedia server