CS 3251- Computer Networking: Multimedia Networking

Professor Patrick Traynor
12/3/13
Lecture 28

Announcements

- All assignment/project drop-boxes are now closed.
  - If you have late work, it's now *too* late.

- This is our last lecture with new material for the semester.
  - Interested in learning more? There's so much more to be done!
  - Next time, we'll review for the final.
    - Details available then...

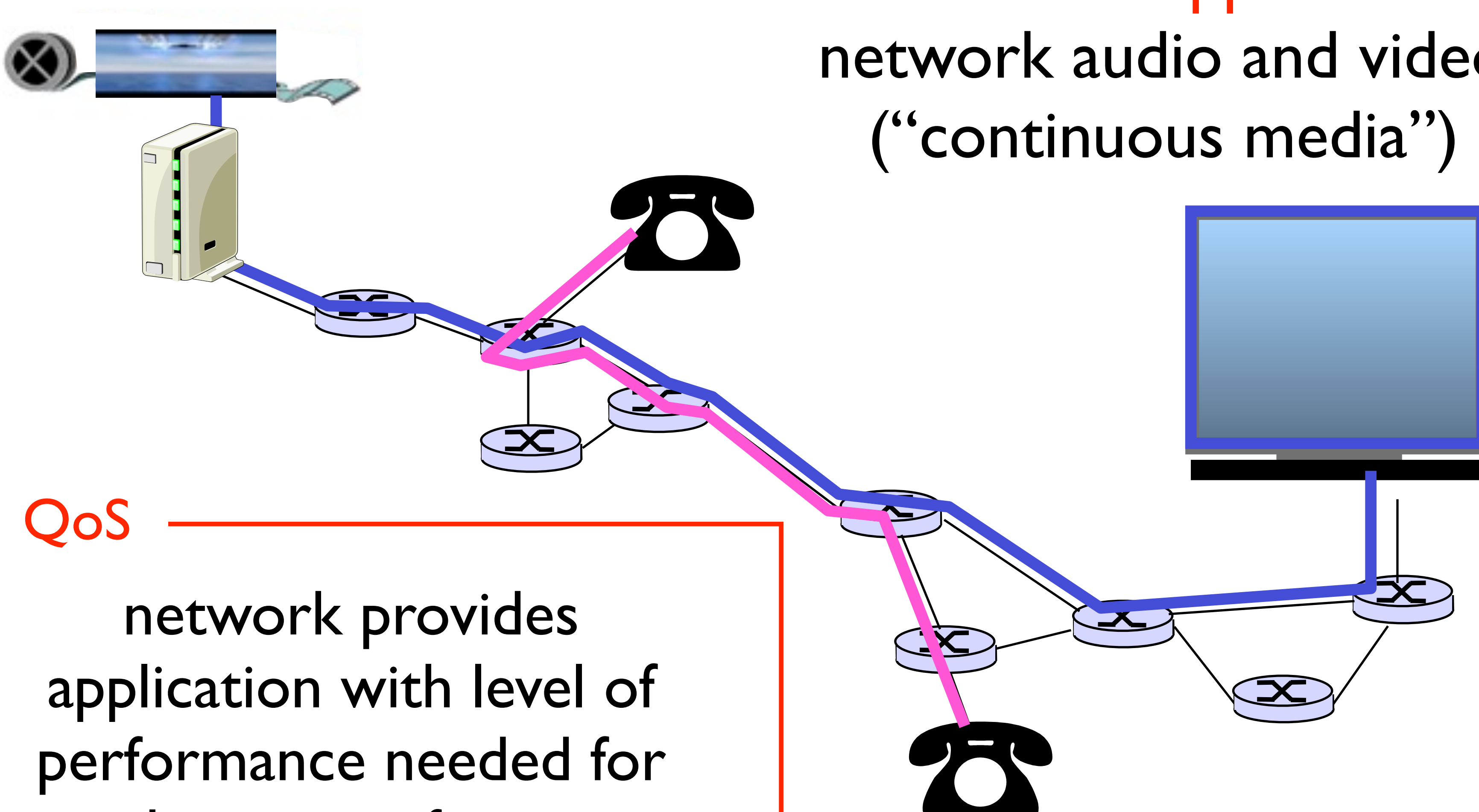
- Take the CIOS survey!

Last Time

- What is a VPN?
  - What technology/protocol suite is generally used to implement them?
- How much protection does WEP offer in practice?
  - What other options should you use?
- How does a firewall work?
- Why are attacks on BGP difficult to detect?

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.

Chapter 7: Goals

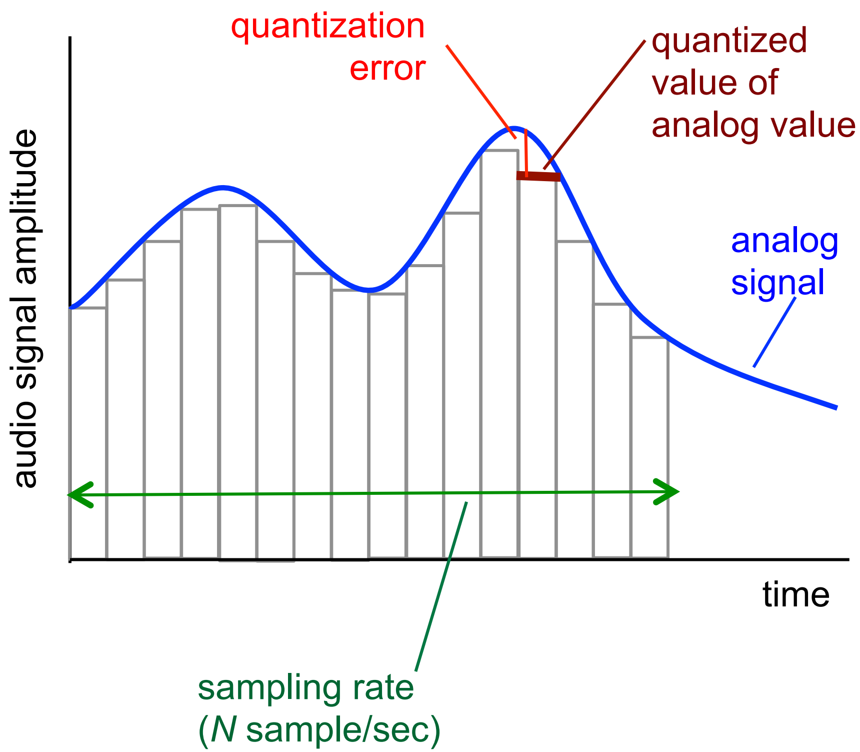
- Principles
  - classify multimedia applications
  - identify network services applications need
  - making the best of best effort service
- Protocols and Architectures
  - specific protocols for best-effort
  - mechanisms for providing QoS
  - architectures for QoS

Chapter 7 Outline

- 7.1 Multimedia Networking Applications
- 7.2 Streaming stored video
- 7.3 Voice-over-IP
- 7.4 Protocols for Real-Time Interactive Applications
  - RTP,RTCP,SIP
- 7.5 Network Support for Multimedia

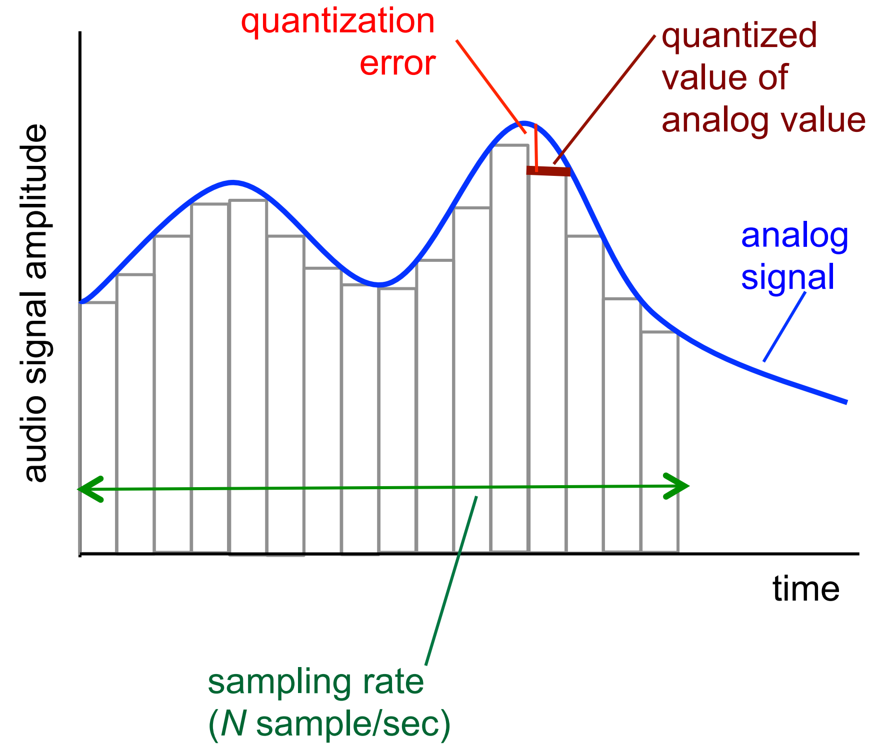
Multimedia: Audio

- analog audio 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, e.g., 8 bits for 256 values



Multimedia: Audio

- example: 8,000 samples/sec, 256 quantized values: 64,000 bps
- receiver converts bits back to analog signal:
  - some quality reduction
- <u>example rates</u>
  - CD: 1.411 Mbps
  - MP3: 96, 128, 160 kbps
  - Internet telephony: 5.3 kbps and up



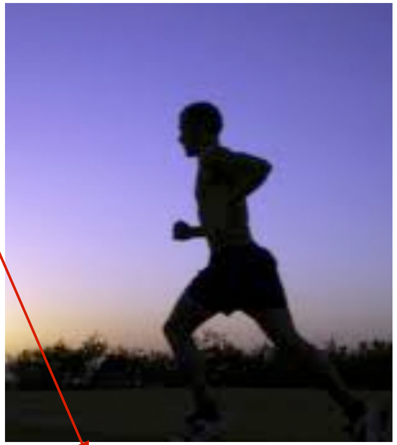
Multimedia: Video

- video: sequence of images displayed at constant rate
  - e.g. 24 images/sec
- digital image: array of pixels
  - each pixel represented by bits
- coding: use redundancy *within* and *between* images to decrease # bits used to encode image
  - spatial (within image)
  - temporal (from one image to next)

spatial coding example: instead of sending *N* values of same color (all purple), send only two values: color value (*purple*) and *number of repeated values* (N)

frame *i*

temporal coding example: instead of sending complete frame at i+1, send only differences from frame i



frame *i+1*

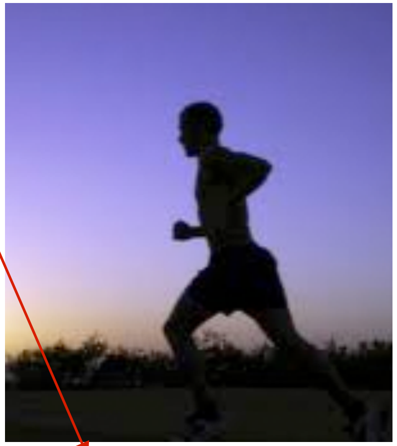
Multimedia: Video

- *CBR*: (constant bit rate): video encoding rate fixed
- *VBR*: (variable bit rate): video encoding rate changes as amount of spatial, temporal coding changes
- <u>examples</u>:
  - MPEG 1 (CD-ROM) 1.5 Mbps
  - MPEG2 (DVD) 3-6 Mbps
  - MPEG4 (often used in Internet, < 1 Mbps)

spatial coding example: instead of sending *N* values of same color (all purple), send only two values: color value (*purple*) and *number of repeated values* (N)

frame *i*

temporal coding example: instead of sending complete frame at i+1, send only differences from frame i

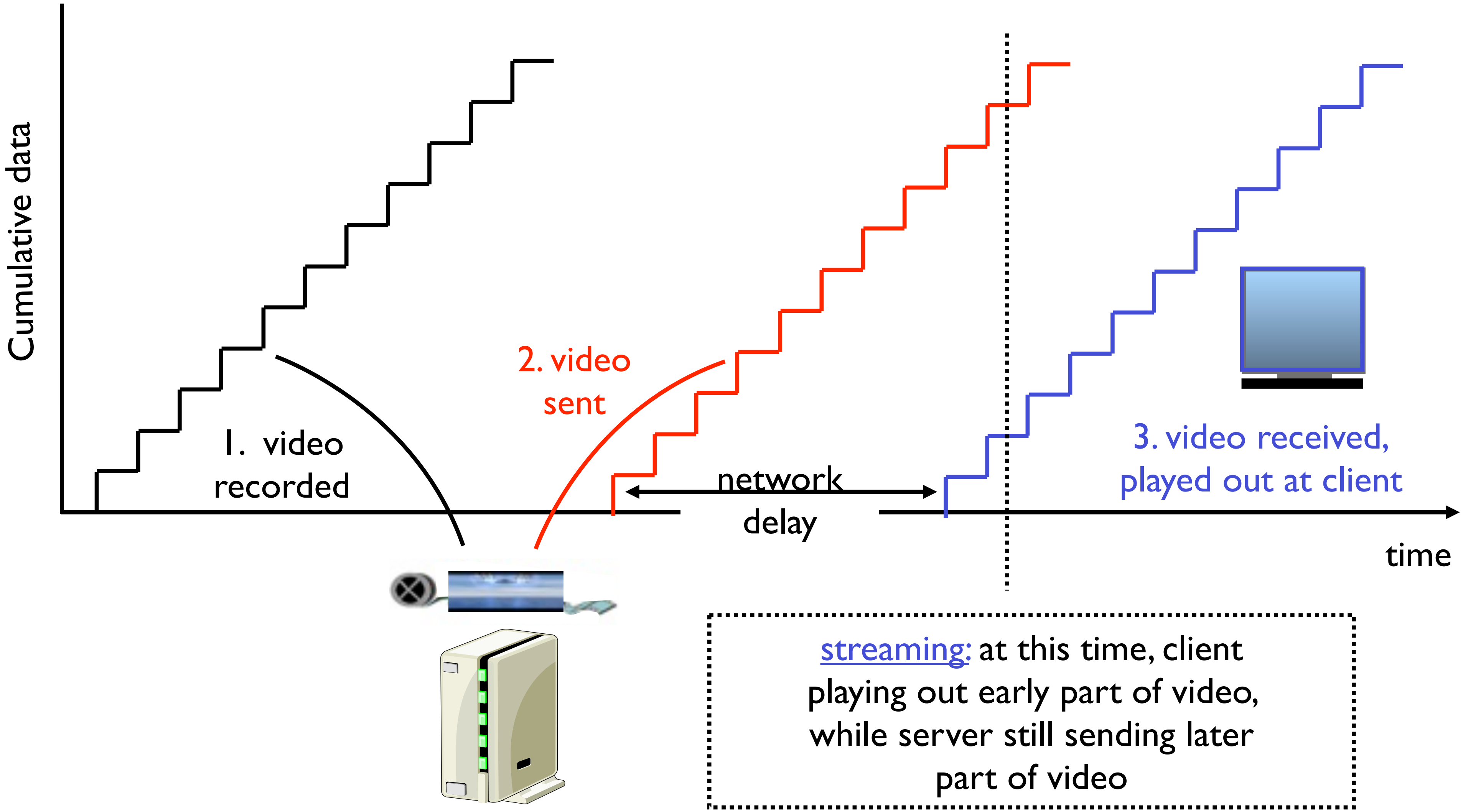


frame *i+1*

Multimedia Networking: Three Application Types

- streaming, stored audio, video
  - streaming: can begin playout before downloading entire file
  - stored (at server): can transmit faster than audio/video will be rendered (implies storing/buffering at client)
  - e.g., YouTube, Netflix, Hulu
- conversational voice/video over IP
  - interactive nature of human-to-human conversation limits delay tolerance
  - e.g., Skype
- streaming live audio, video
  - e.g., live sporting event (futbol)

Streaming Stored Multimedia: What is it?

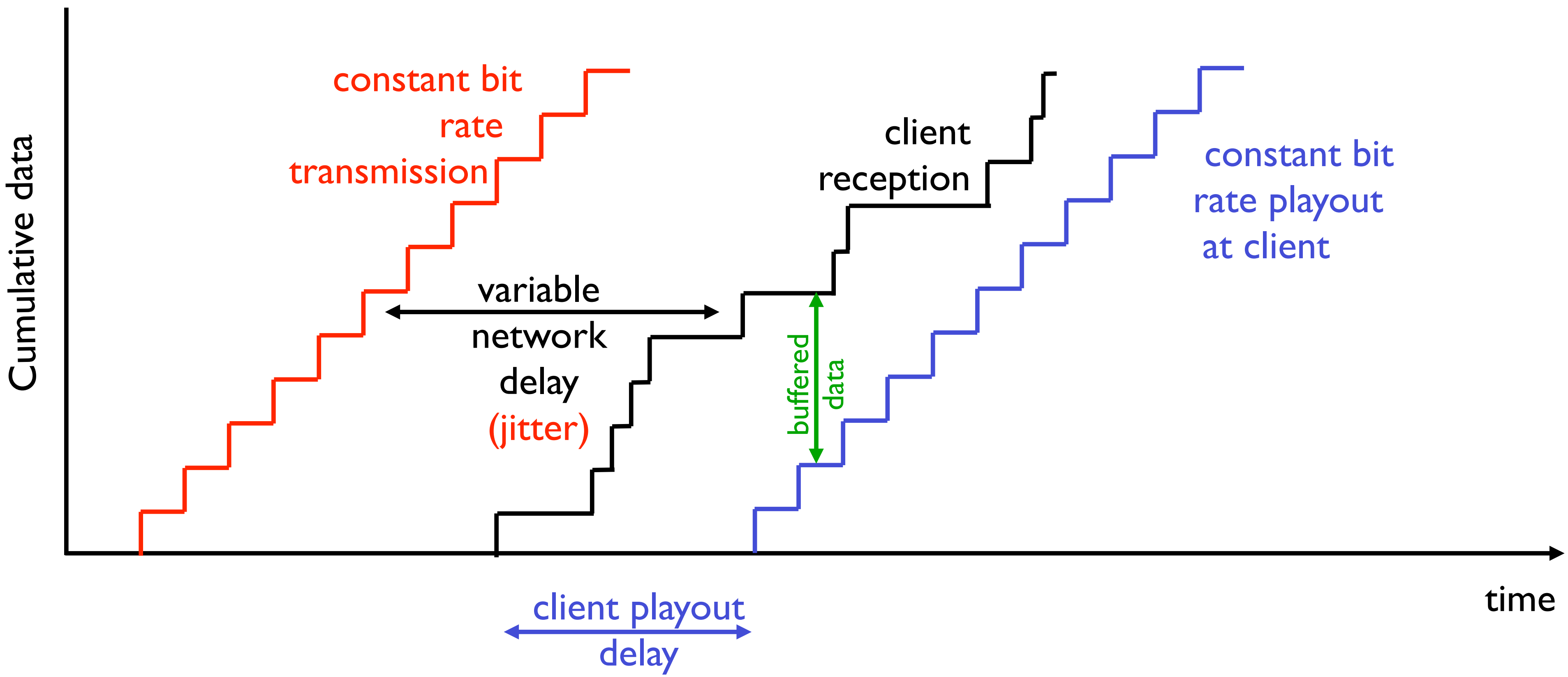


streaming: at this time, client playing out early part of video, while server still sending later part of video

Streaming stored video: challenges

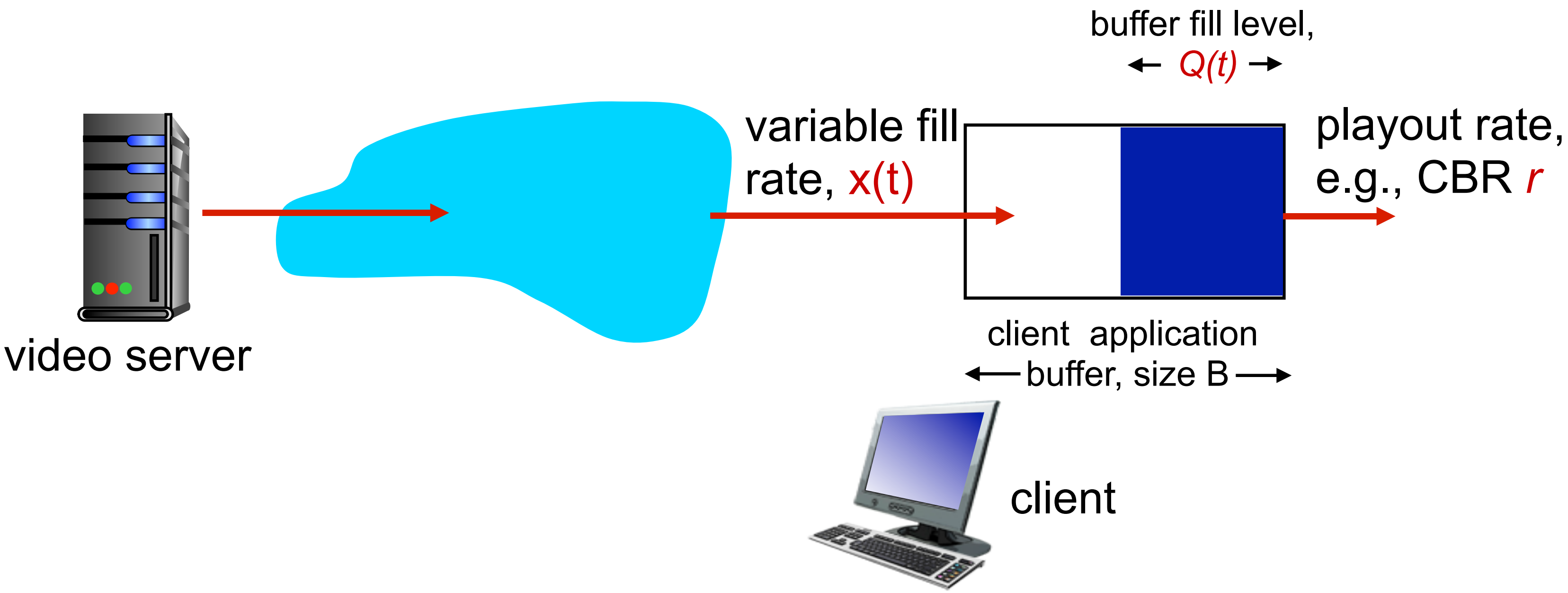
- continuous playout constraint: once client playout begins, playback must match original timing
  - … but network delays are variable (jitter), so will need client-side buffer to match playout requirements
- other challenges:
  - client interactivity: pause, fast-forward, rewind, jump through video
  - video packets may be lost, retransmitted

Delay Jitter

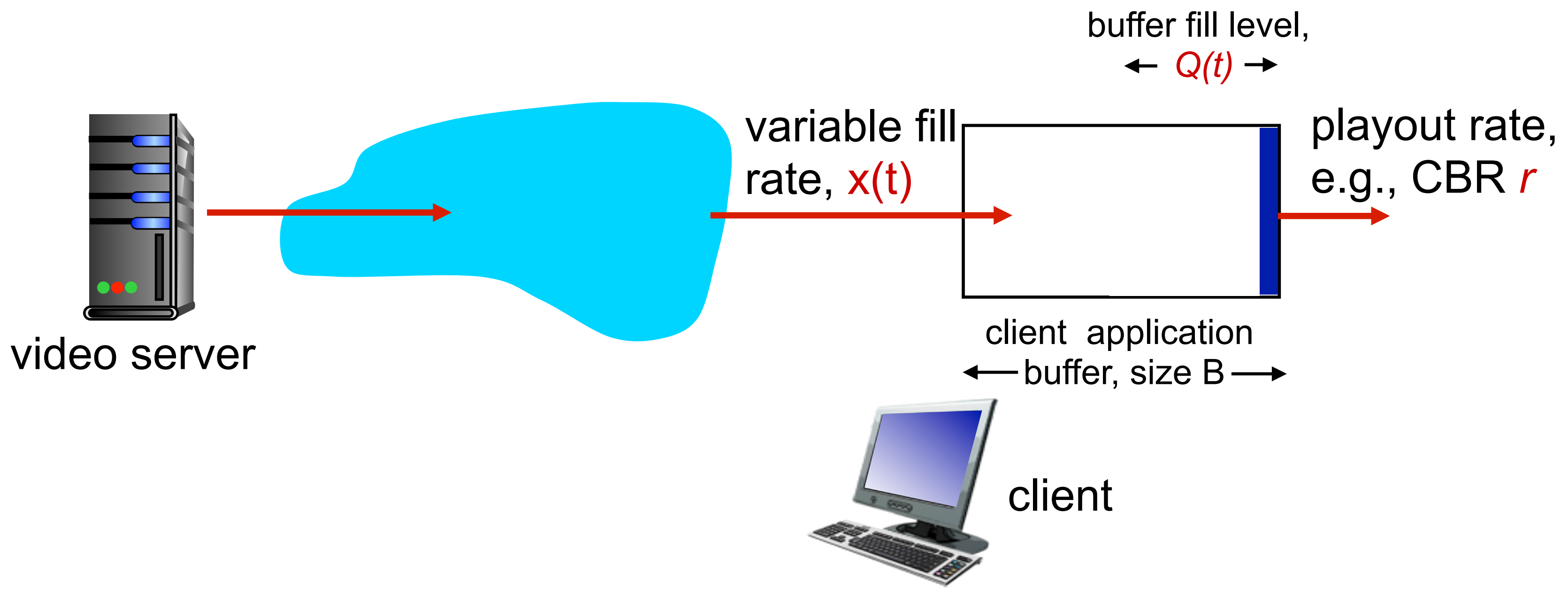


- Consider the end-to-end delays of two consecutive packets: difference can be more or less than 20 msec

Client-side buffering, playout

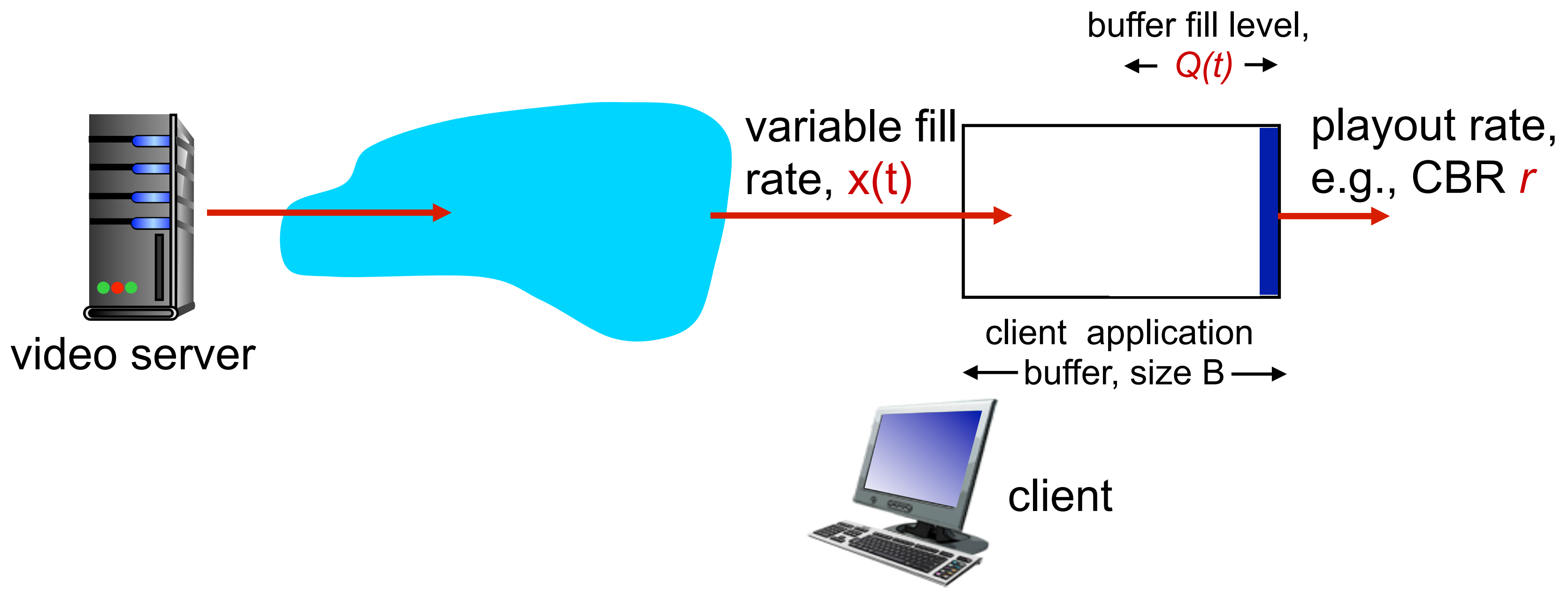


Client-side buffering, playout



1. Initial fill of buffer until playout begins at t_p
2. playout begins at t_p,
3. buffer fill level varies over time as fill rate $x(t)$ varies and playout rate r is constant

Client-side buffering, playout



- *playout buffering: average fill rate ($\bar{x}$), playout rate (r):*
  - $\bar{x} < r$: buffer eventually empties (causing freezing of video playout until buffer again fills)
  - $\bar{x} > r$: buffer will not empty, provided initial playout delay is large enough to absorb variability in x(t)
    - initial playout delay tradeoff: buffer starvation less likely with larger delay, but larger delay until user begins watching

Content distribution networks

- *challenge*: how to stream content (selected from millions of videos) to hundreds of thousands of simultaneous users?

- *option 1:* single, large "mega-server"
  - single point of failure
  - point of network congestion
  - long path to distant clients
  - multiple copies of video sent over outgoing link
- ….quite simply: this solution *doesn't scale*

Content distribution networks

- *challenge*: how to stream content (selected from millions of videos) to hundreds of thousands of simultaneous users?
- *option 2:* store/serve multiple copies of videos at multiple geographically distributed sites (CDN)
  - enter deep: push CDN servers deep into many access networks
    - close to users
    - used by Akamai, 1700 locations
  - bring home: smaller number (10's) of larger clusters in POPs near (but not within) access networks
    - used by Limelight

Case study: Netflix

- 30% downstream US traffic in 2011
- owns very little infrastructure, uses 3rd party services:
  - own registration, payment servers
  - Amazon (3rd party) cloud services:
    - Netflix uploads studio master to Amazon cloud
    - create multiple version of movie (different endodings) in cloud
    - upload versions from cloud to CDNs
    - Cloud hosts Netflix web pages for user browsing
  - three 3rd party CDNs host/stream Netflix content: Akamai, Limelight, Level-3

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Voice-over-IP (VoIP)

- *VoIP end-end-delay requirement*: needed to maintain "conversational" aspect
  - higher delays noticeable, impair interactivity
  - < 150 msec: good
  - > 400 msec bad
  - includes application-level (packetization,playout), network delays
- *session initialization:* how does callee advertise IP address, port number, encoding algorithms?
- *value-added services:* call forwarding, screening, recording
- *emergency services:* 911

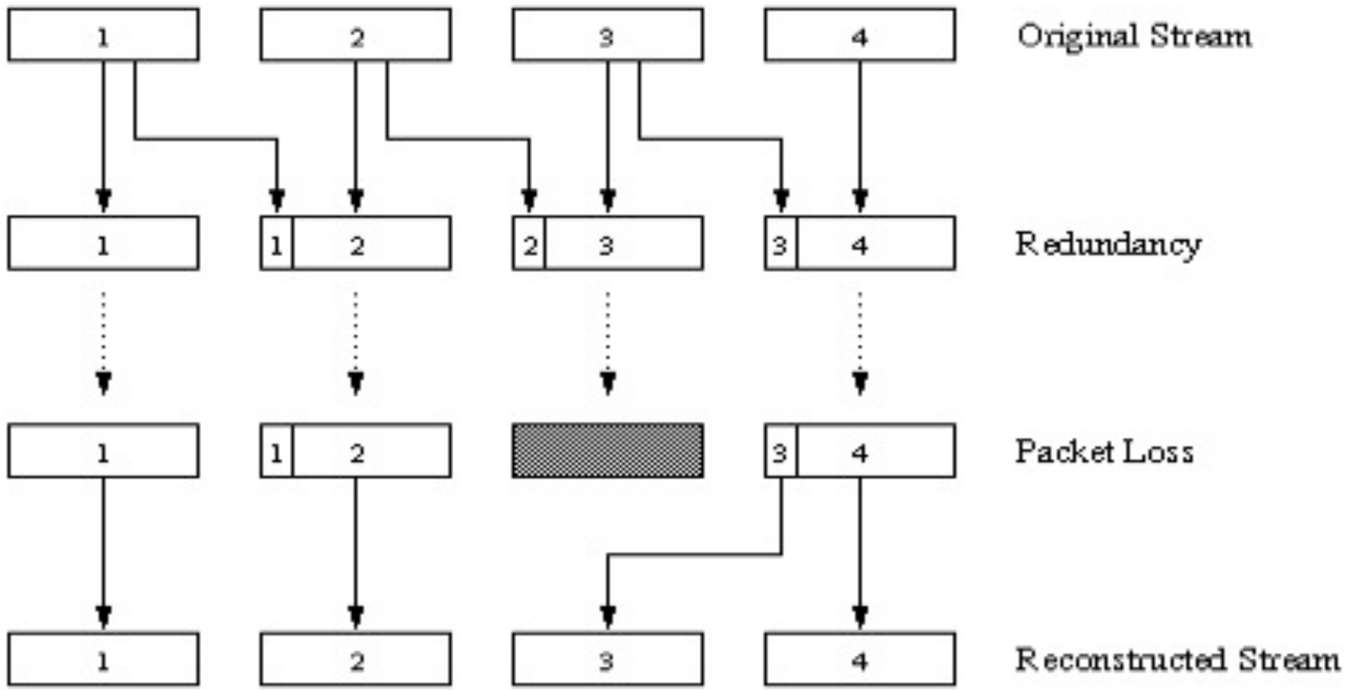
VoIP: packet loss, delay

- *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, loss concealment, packet loss rates between 1% and 10% can be tolerated

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 (II)

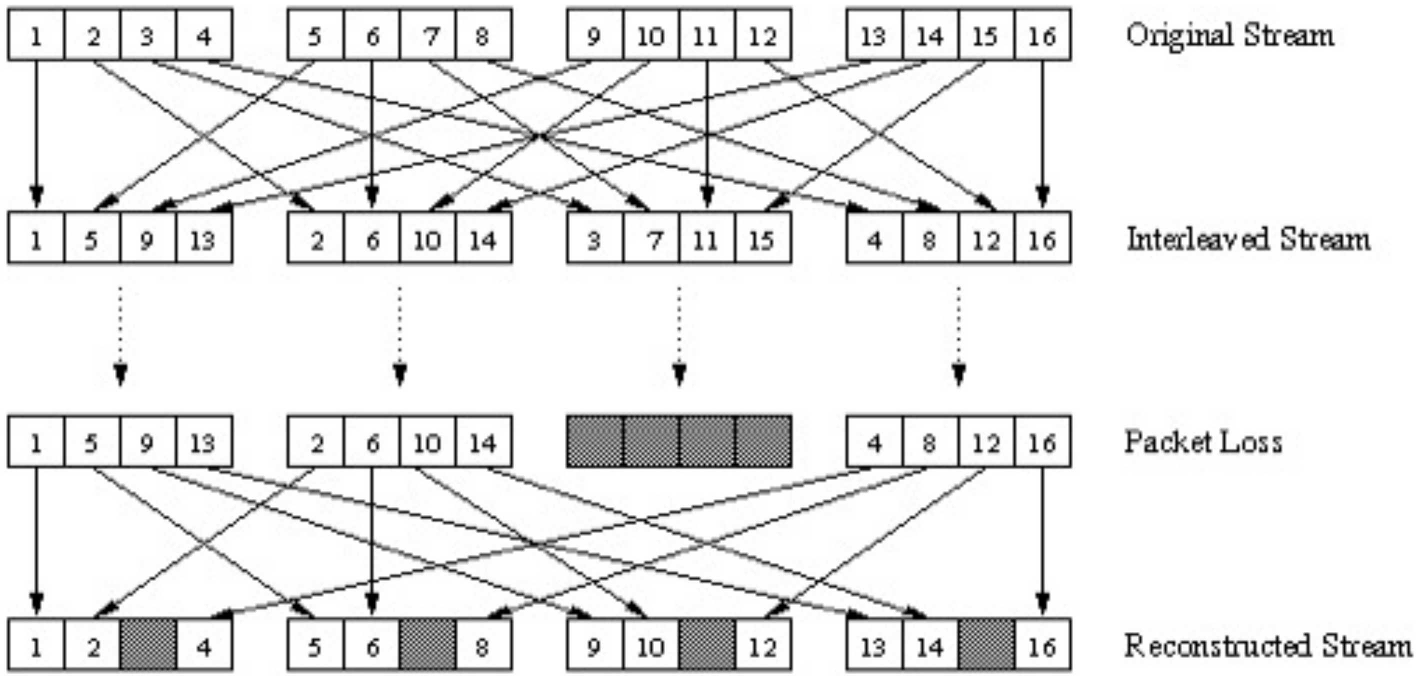


2nd FEC scheme

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

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

Recovery from Packet Loss (III)



Interleaving

- chunks are broken up into smaller units
- for example, 4 5 msec units per chunk
- Packet contains small units from different chunks

- if packet is lost, still have most of every chunk
- has no redundancy overhead
- but adds to playout delay

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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 VoIP applications run RTP, they may be able to work together

Real-Time Control Protocol (RTCP)

- works in conjunction with RTP
- each participant in RTP session periodically sends RTCP control packets to all other participants

- each RTCP packet contains sender and/or receiver reports
  - report statistics useful to application: # packets sent, # packets lost, interarrival jitter
- feedback used to control performance
  - sender may modify its transmissions based on feedback

Session Initiation Protocol (SIP)

- SIP provides mechanisms for call setup:
  - for caller to let callee know she wants to establish a call
  - so caller, callee can agree on media type, encoding
  - to end call
- determine current IP address of callee:
  - maps mnemonic identifier to current IP address
- call management:
  - add new media streams during call
  - change encoding during call
  - invite others
  - transfer, hold calls

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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?

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

Next Time

- That's all for new material...
  - Wow... that went quickly!
- Get ready for the final!
  - Next class (conveniently) is a review!