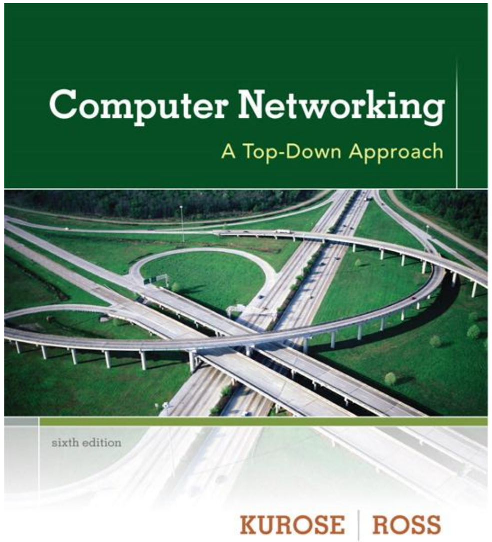
Chapter 7
Multimedia Networking



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Computer Networking: A Top Down Approach
6th edition
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Multimedia networking: outline

7.1 multimedia networking applications
7.2 streaming *stored* video
7.3 voice-over-IP
7.4 protocols for *real-time* conversational applications
7.5 network support for multimedia

A multimedia network application is a network application that uses audio or video.

Before diving into an in-depth discussion of Internet multimedia applications, it is useful to consider the intrinsic characteristics of the audio and video media themselves.

Properties of Video

Perhaps the most salient characteristic of video is its high bit rate.

Video distributed over the Internet typically ranges from 100 kbps for low-quality video conferencing to over 3 Mbps for streaming high-definition movies.

To get a sense of how video bandwidth demands compare with those of other Internet applications, let’s briefly consider three different users, each using a different Internet application.

User 1:

Browsing photos: Flipping through photos every 10 seconds, and that photos are on average 200 Kbytes in size.

User2:

Streaming music from the Internet to the system: Listening to many MP3 songs, one after the other, each encoded at a rate of 128 kbps.

User3:

Watching a video that has been encoded at 2 Mbps.

Finally, let's suppose that the session length for all three users is 4,000 seconds (approximately 67 minutes).

Table 7.1 compares the bit rates and the total bytes transferred for these 3 users.

	Bit rate	Bytes transferred in 67 min
Facebook Frank	160 kbps	80 Mbytes
Martha Music	128 kbps	64 Mbytes
Victor Video	2 Mbps	1 Gbyte

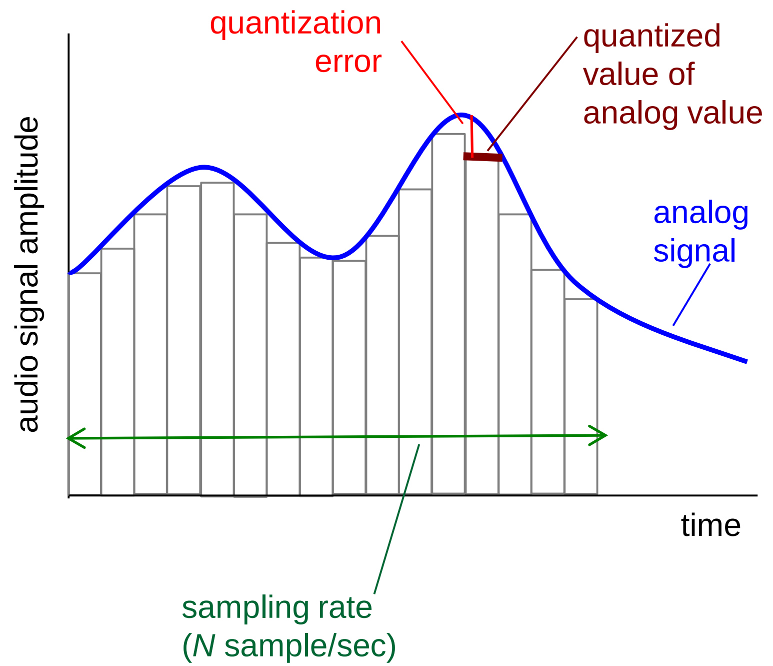
Table 7.1 ♦ Comparison of bit-rate requirements of three Internet applications

Therefore, when designing networked video applications, the first thing we must keep in mind is the high bit-rate requirements of video.

Given the popularity of video and its high bit rate, it was predicted that streaming live and stored video will be approximately 90 percent of global consumer Internet traffic by 2015.

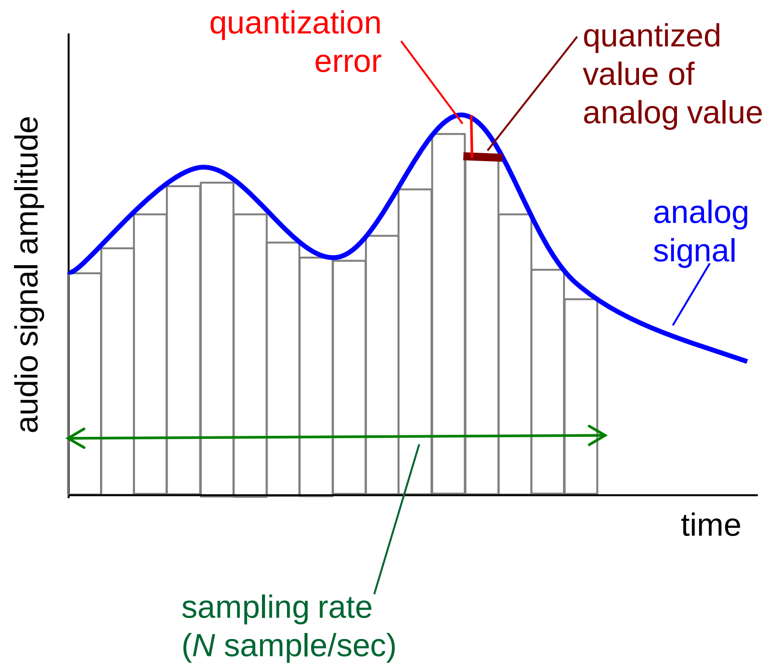
Multimedia: audio

- 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

- **Another important property of video is that it can be compressed.**
  - 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*

An important characteristic of video is that it can be compressed, thereby trading off video quality with bit rate.

A video is a sequence of images, typically being displayed at a constant rate, for example, at 24 or 30 images per second.

An uncompressed, digitally encoded image consists of an array of pixels, with each pixel encoded into a number of bits to represent luminance and color.

There are two types of redundancy in video, both of which can be exploited by video compression.

i. Spatial redundancy is the redundancy within a given image.
ii. Temporal redundancy reflects repetition from image to subsequent image.

Today's off-the-shelf compression algorithms can compress a video to essentially any bit rate desired.

Of course, the higher the bit rate, the better the image quality and the better the overall user viewing experience. Multiple versions of same video are available.

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
- *examples:*
  - 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: 3 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 (football)

Streaming stored Audio or Video

It has three key distinguishing features:

- *Streaming.*
- *Interactivity.*
- *Continuous playout.*

The most important performance measure for streaming video is average throughput.

In order to provide continuous playout, the network must provide an average throughput to the streaming application that is at least as large as the bit rate of the video itself.

By using buffering and prefetching, it is possible to provide continuous playout even when the throughput fluctuates, as long as the average throughput (averaged over 5–10 seconds) remains above the video rate.

For many streaming video applications, prerecorded video is stored on, and streamed from, a CDN or P2P network.

Conversational Voice- and Video-over-IP

Also called as **Internet telephony or Voice-over-IP (VoIP)**.

Conversational voice and video are widely used in the Internet today, with the Internet companies Skype, QQ, and Google Talk.

Application requirements: Timing considerations and tolerance of data loss.

The audio and video conversational applications are highly **delay-sensitive**.
On the other hand, conversational multimedia applications are **loss-tolerant**—occasional loss only causes occasional glitches in audio/video playback, and these losses can often be partially or fully concealed.

These delay-sensitive but loss-tolerant characteristics are clearly different from those of elastic data applications such as Web browsing, e-mail, social networks, and remote login.

Streaming Live Audio and Video

This class of applications is similar to traditional broadcast radio and television, except that transmission takes place over the Internet.

These applications allow a user to receive a live radio or television transmission—such as a live sporting event or an ongoing news event—transmitted from any corner of the world.

Can be accomplished through multicasting techniques or multiple unicasting techniques. May use CDN or P2P networks.

As with streaming stored multimedia, the network must provide each live multimedia flow with an average throughput that is larger than the video consumption rate. Techniques used are similar to streaming stored content.

Because the event is live, delay can also be an issue.

Delays of up to ten seconds or so from when the user chooses to view a live transmission to when playout begins can be tolerated.

Multimedia networking: outline

7.1 multimedia networking applications

7.2 streaming *stored* video

7.3 voice-over-IP

7.4 protocols for *real-time* conversational applications

7.5 network support for multimedia

Streaming stored video systems can be classified into three categories:

UDP streaming, HTTP streaming, and adaptive HTTP streaming.

A common characteristic of all three forms of video streaming is the extensive use of client-side application buffering to mitigate the effects of varying end-to-end delays and varying amounts of available bandwidth between server and client.

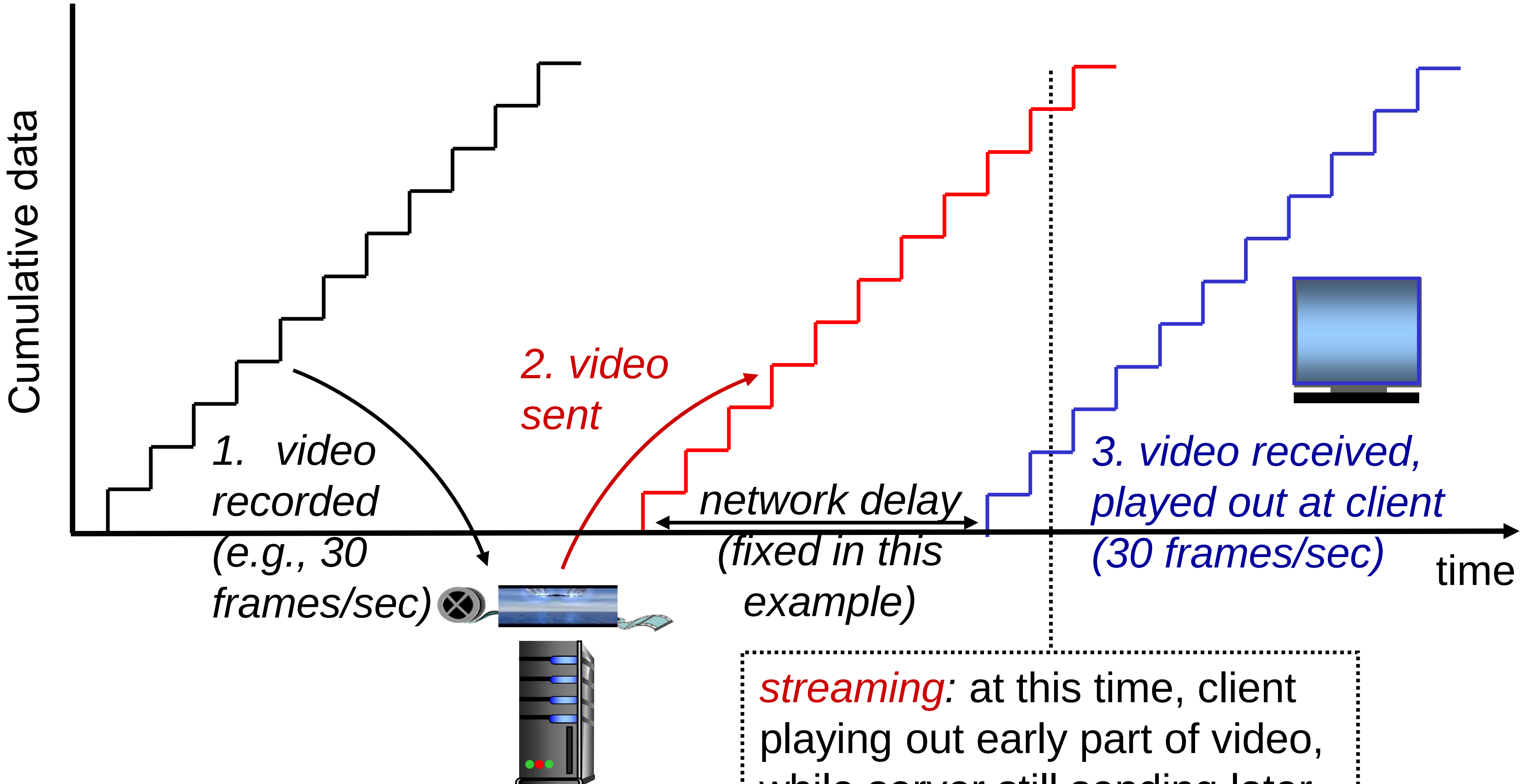
There are two important advantages provided by such client buffering.

First, client side buffering can absorb variations in server-to-client delay.

If a particular piece of video data is delayed, as long as it arrives before the reserve of received-but-not yet- played video is exhausted, this long delay will not be noticed.

Second, if the server-to-client bandwidth briefly drops below the video consumption rate, a user can continue to enjoy continuous playback, again as long as the client application buffer does not become completely drained.

Streaming stored video:



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

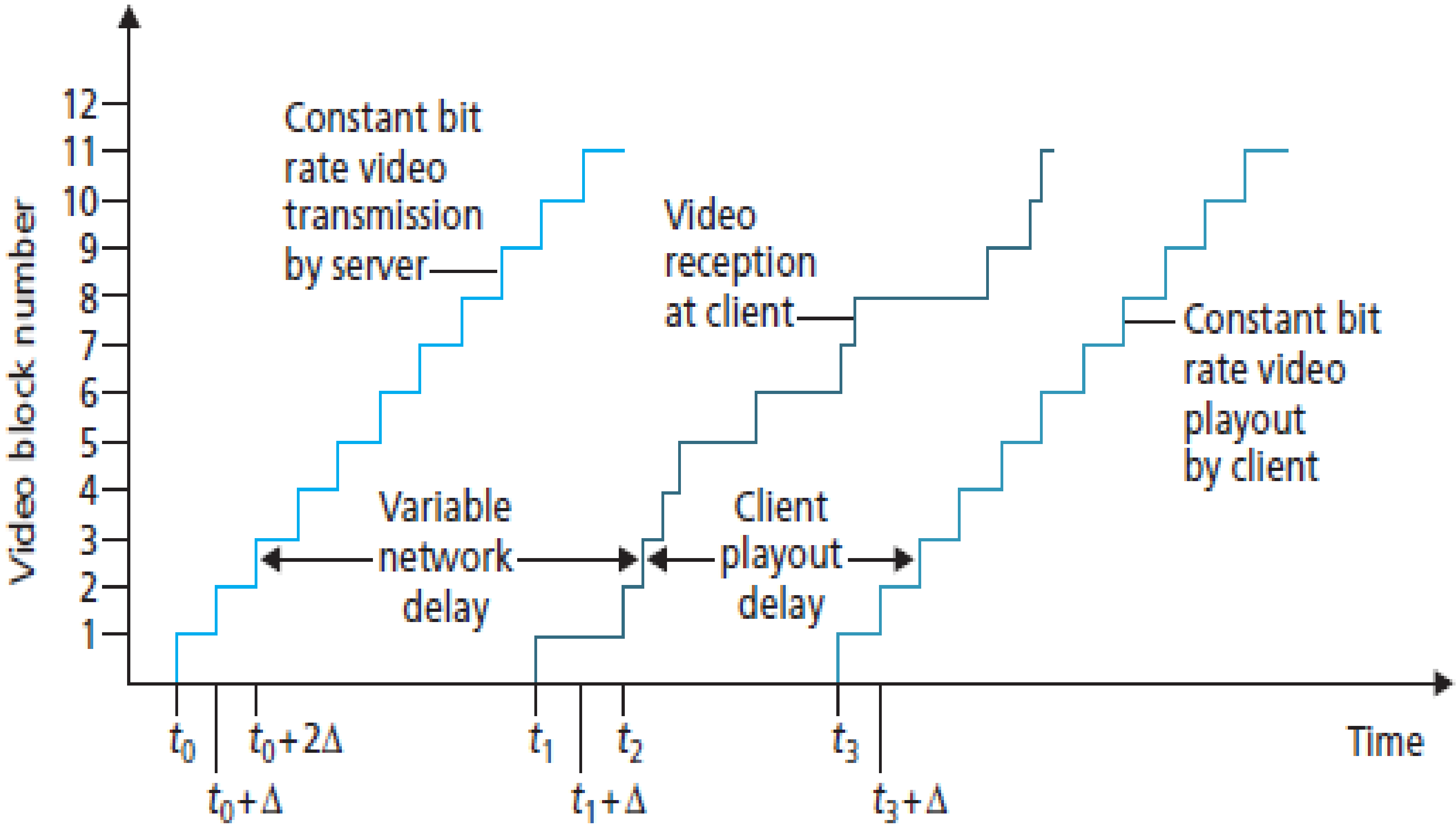
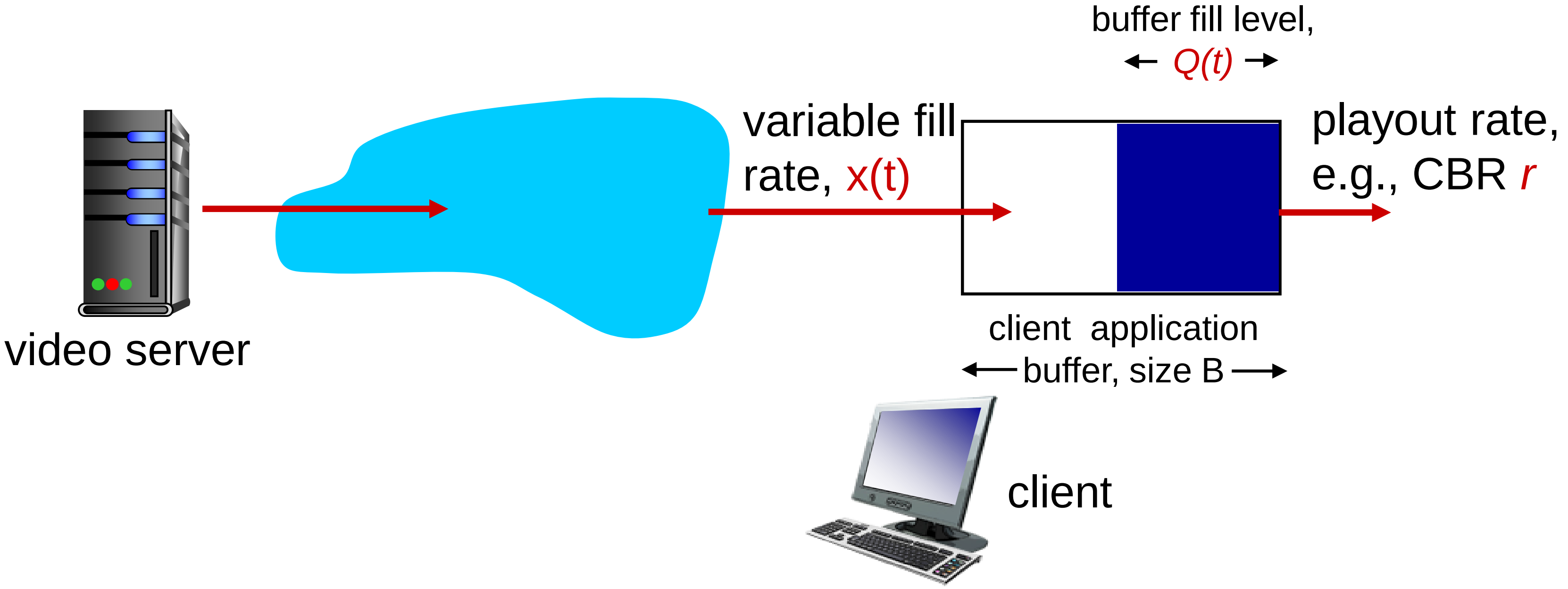


Figure 7.1 ♦ Client playout delay in video streaming

Client-side buffering, playout



Streaming multimedia: UDP

- The server transmits video at a rate that matches the client's video consumption rate by clocking out the video chunks over UDP at a steady rate.

- For example, if the video consumption rate is 2 Mbps and each UDP packet carries 8,000 bits of video, then the server would transmit one UDP packet every 4 msec.

- As UDP does not employ a congestion-control mechanism, the server can push packets into the network at the consumption rate of the video without the rate-control restrictions of TCP.

- UDP streaming typically uses a small client-side buffer, big enough to hold less than a second of video.

- Before passing the video chunks to UDP, the server will encapsulate the video chunks using the Real-Time Transport Protocol (RTP).

- Another distinguishing property of UDP streaming is that the client and server maintain a separate control connection over which the client sends commands regarding session state changes (such as pause, resume, reposition etc.,). This control connection is in many ways analogous to the FTP control connection.

- The Real-Time Streaming Protocol (RTSP) is a popular open protocol for such a control connection.

- Although UDP streaming has been employed in many open-source systems and proprietary products, it suffers from three significant drawbacks.

- First, due to the unpredictable and varying amount of available bandwidth between server and client, constant-rate UDP streaming can fail to provide continuous playout.

- The second drawback of UDP streaming is that it requires a media control server, such as an RTSP server, to process client-to-server interactivity requests and to track client state (e.g., the client’s playout point in the video, whether the video is being paused or played, and so on) for each ongoing client session. This increases the overall cost and complexity of deploying a large-scale video-on-demand system.

- The third drawback is that many firewalls are configured to block UDP traffic, preventing the users behind these firewalls from receiving UDP video.

HTTP Streaming

In HTTP streaming, the video is simply stored in an HTTP server as an ordinary file with a specific URL.

When a user wants to see the video, the client establishes a TCP connection with the server and issues an HTTP GET request for that URL.

The server then sends the video file, within an HTTP response message, as quickly as possible, that is, as quickly as TCP congestion control and flow control will allow.

On the client side, the bytes are collected in a client application buffer. Once the number of bytes in this buffer exceeds a predetermined threshold, the client application begins playback.

- In HTTP streaming the transmission rate may vary in a "saw-tooth" manner and the packets can also be significantly delayed.

Advantages:

- The use of HTTP over TCP allows the video to traverse through firewalls and NATs.

- There is no need for RTSP server, thereby reducing the cost.

- Most video streaming applications today—including YouTube and Netflix—use HTTP streaming (over TCP) as its underlying streaming protocol.

- Video prefetching on client side occurs naturally with TCP streaming, since TCP's congestion avoidance mechanism will attempt to use all of the available bandwidth between server and client.

- [Wang 2008] shows that when the average TCP throughput is roughly twice the media bit rate, streaming over TCP results in minimal starvation and low buffering delays.

Client Application Buffer and TCP Buffers

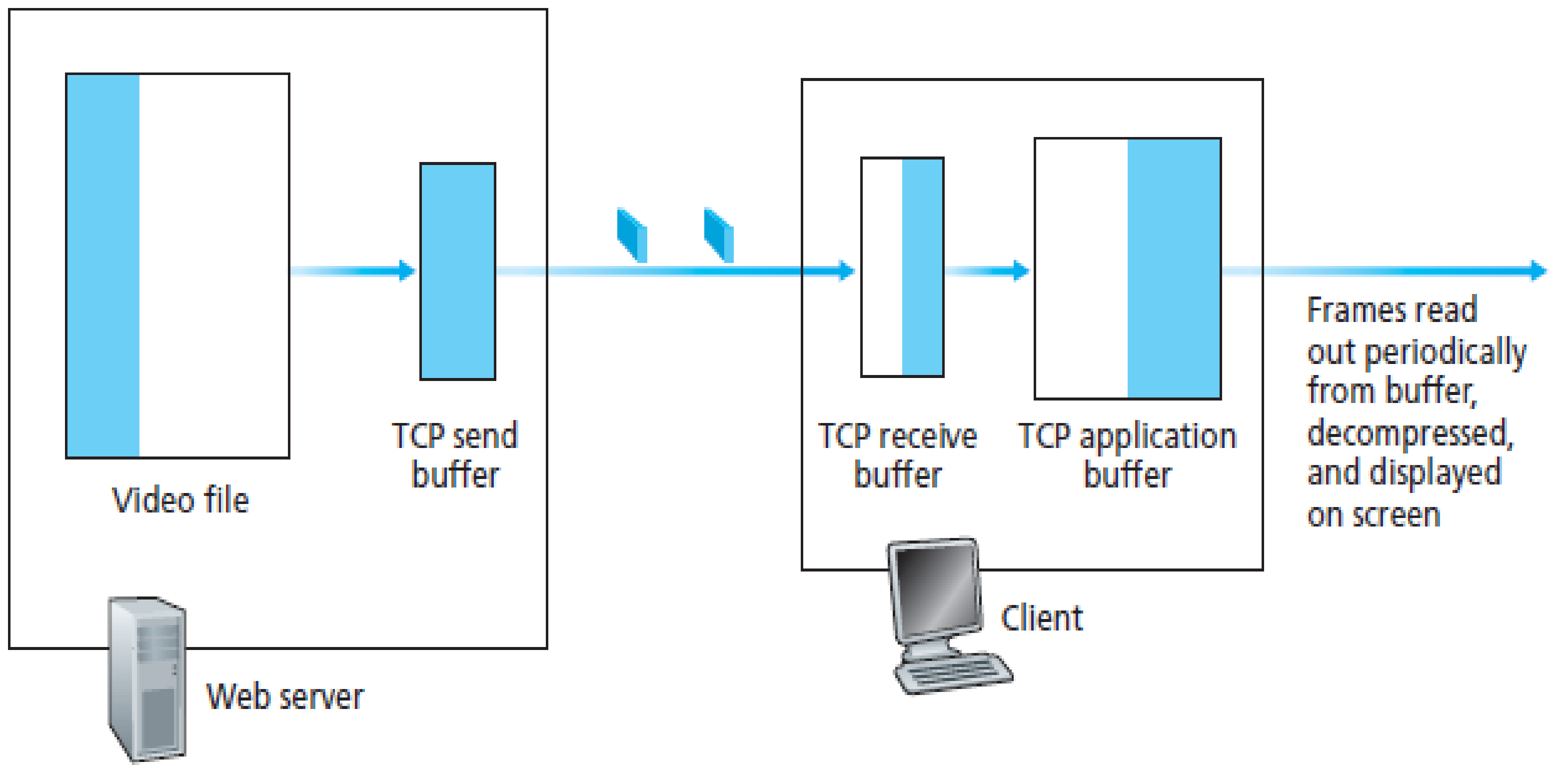


Figure 7.2 ♦ Streaming stored video over HTTP/TCP

Back pressure occurs if the user pauses the video, if the client application buffer becomes full.

Note that when the client application removes *f* bits, it creates room for *f* bits in the client application buffer, which in turn allows the server to send *f* additional bits.

Thus, the server send rate can be no higher than the video consumption rate at the client.

Therefore, *a full client application buffer indirectly imposes a limit on the rate at which the video can be sent from server to client when streaming over HTTP.*

Analysis of Video Streaming

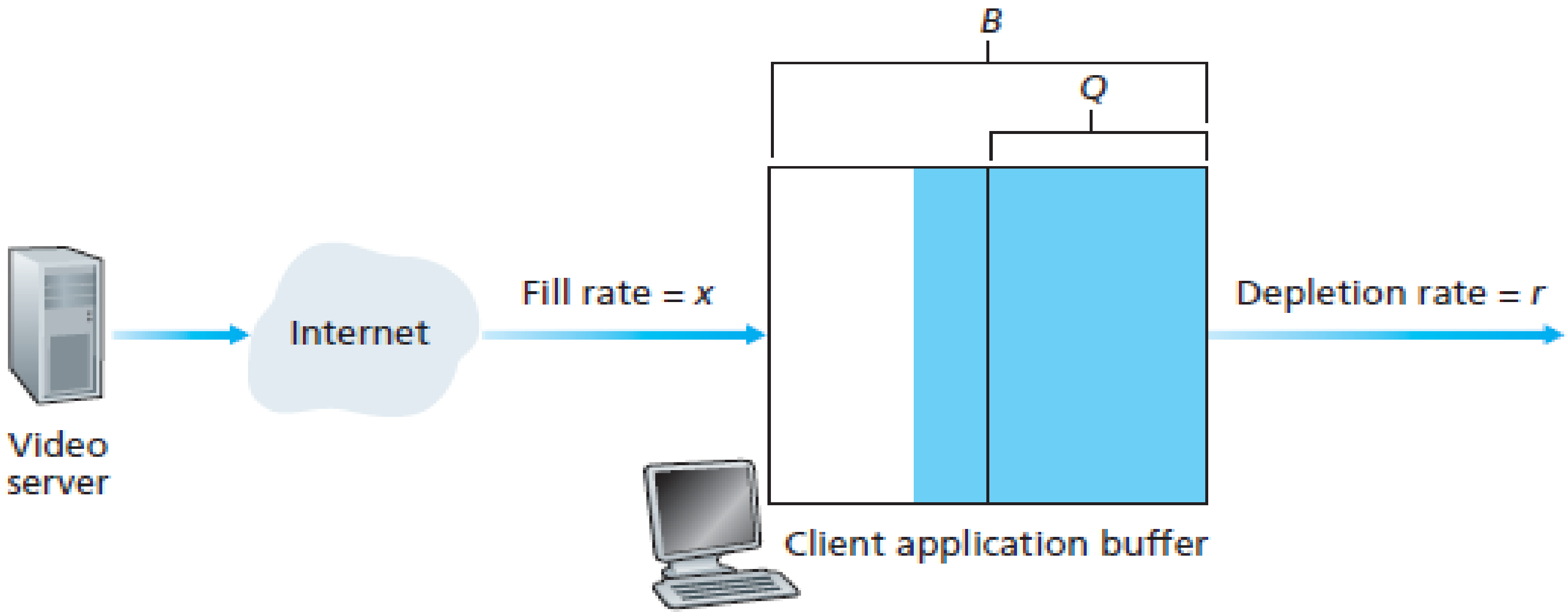


Figure 7.3 ♦ Analysis of client-side buffering for video streaming

The amount of time required to build up Q bits (the initial buffering delay) is $t_p = Q/x$ *where* t_p *is the playout time.*

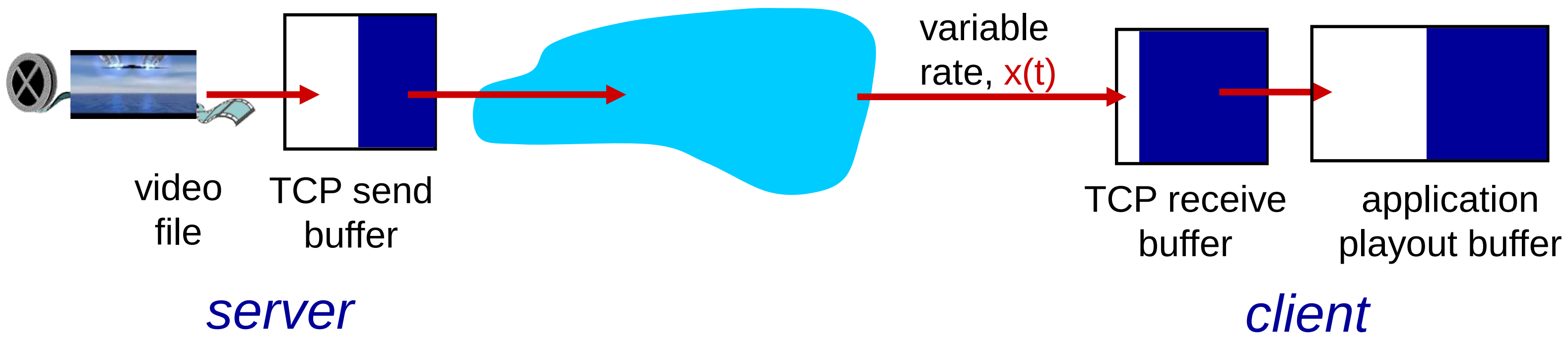
If $x < r$*, that is the available rate in the network is less than the video rate, playout will alternate between periods of continuous playout and periods of freezing.*

- When $x > r$. In this case, starting at time t_p , the buffer increases from Q to B at the rate $x-r$ since bits are being depleted at rate r but are arriving at rate x.

- ***Thus when the available rate in the network is more than the video consumption rate, after the initial buffering delay, the user will enjoy continuous playout until the video ends.***

Streaming multimedia: HTTP

- multimedia file retrieved via HTTP GET
- send at maximum possible rate under TCP



- fill rate fluctuates due to TCP congestion control, retransmissions (in-order delivery)
- larger playout delay: smooth TCP delivery rate
- HTTP/TCP passes more easily through firewalls

Early Termination and Repositioning the Video

HTTP byte-range header in the HTTP GET request message.

When the user repositions to a new position, the client sends a new HTTP request, indicating with the byte-range header from which byte in the file should the server send data.

When the server receives the new HTTP request, it can forget about any earlier request and instead send bytes beginning with the byte indicated in the byte range request.

Repositioning leads to waste of network bandwidth and server resources.

For example, suppose that the client buffer is full with B bits at some time t0 into the video, and at this time the user repositions to some instant $t > t0 + B/r$ into the video, and then watches the video to completion from that point on.

- In this case, all B bits in the buffer will be unwatched and there is significant wasted bandwidth in the Internet due to early termination, which can be quite costly, particularly for wireless links.

- For this reason, many streaming systems use only a moderate-size client application buffer, or will limit the amount of prefetched video using the byte-range header in HTTP Requests.

Adaptive Streaming and DASH

Although HTTP streaming, has been extensively deployed in practice , it has a major shortcoming:

All clients receive the same encoding of the video, despite the large variations in the amount of bandwidth available to a client, both across different clients and also over time for the same client.

This has led to the development of a new type of HTTP-based streaming, often referred to as Dynamic Adaptive Streaming over HTTP (DASH).

Streaming multimedia: DASH

- *DASH: D*ynamic, *A*daptive *S*treaming over *H*TTP
- *server:*
  - divides video file into multiple chunks
  - each chunk stored, encoded at different rates
  - *manifest file:* provides URLs for different chunks
- *client:*
  - periodically measures server-to-client bandwidth
  - consults manifest, requests one chunk at a time
    - chooses maximum coding rate sustainable given current bandwidth
    - can choose different coding rates at different points in time (depending on available bandwidth at time)

Streaming multimedia: DASH

- *DASH: Dynamic, Adaptive Streaming over HTTP*
- *"intelligence"* at client: client determines
  - *when* to request chunk (so that buffer starvation, or overflow does not occur)
  - *what encoding rate* to request (higher quality when more bandwidth available)
  - *where* to request chunk (can request from URL server that is "close" to client or has high available bandwidth)

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

CDN: "simple" content access scenario

Bob (client) requests video http://netcinema.com/6Y7B23V

- video stored in CDN at http://KingCDN.com/NetC6y&B23V

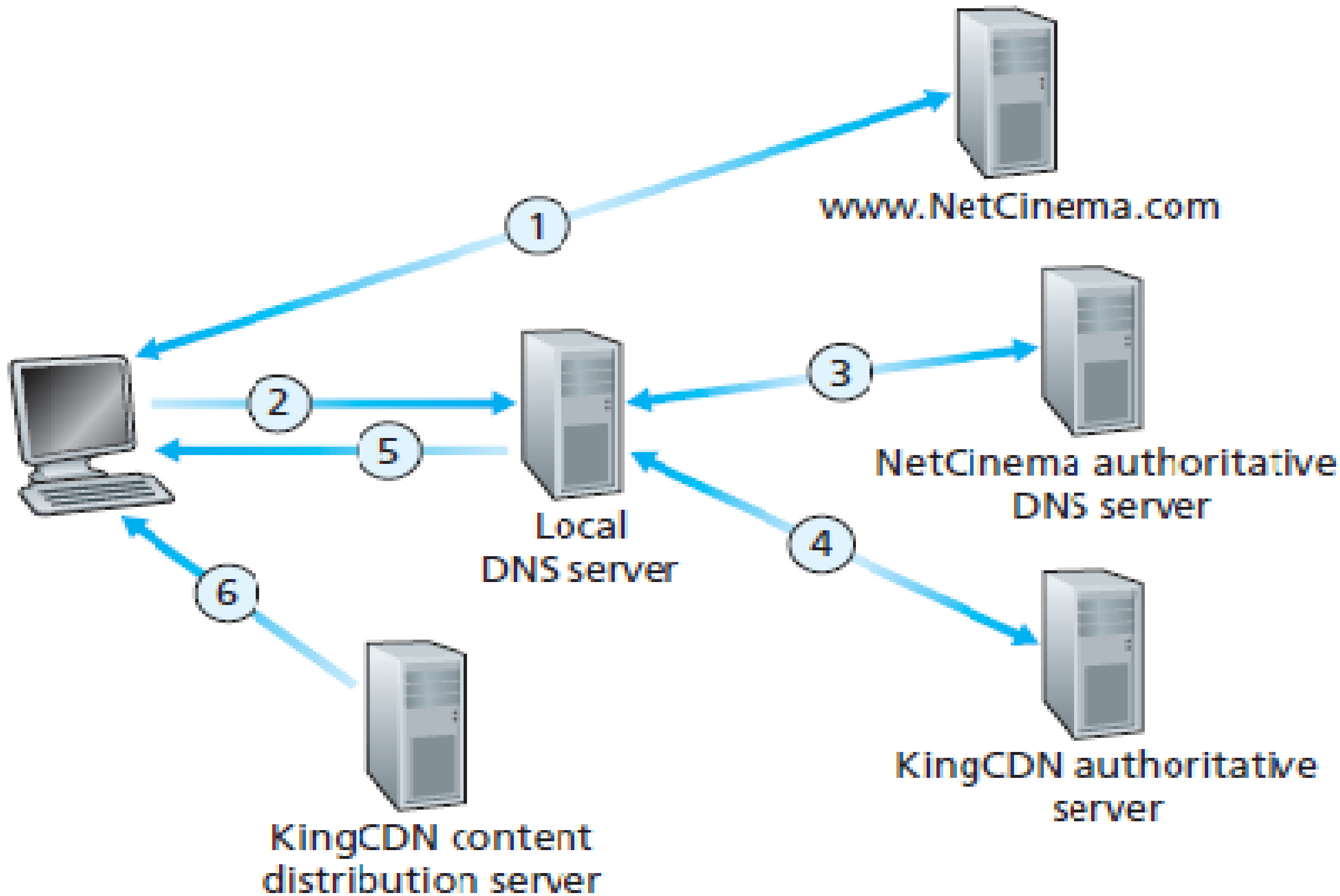


Figure 7.4 ♦ DNS redirects a user's request to a CDN server

Google's CDN infrastructure

It has three tiers of server clusters:

• Eight "mega data centers," with six located in the United States and two located in Europe, with each data center having on the order of 100,000 servers. These mega data centers are responsible for serving dynamic (and often personalized) content, including search results and gmail messages.

• About 30 "bring-home" clusters, with each cluster consisting on the order of 100–500 servers. The cluster locations are distributed around the world, with each location typically near multiple tier-1 ISP PoPs. These clusters are responsible for serving static content, including YouTube videos.

• Many hundreds of "enter-deep" clusters , with each cluster located within an access ISP. Here a cluster typically consists of tens of servers within a single rack. These enter-deep servers perform TCP splitting and serve static content, including the static portions of Web pages that embody search results.

All of these data centers and cluster locations are networked together with Google’s own private network, as part of one enormous AS (AS 15169).

When a user makes a search query, often the query is first sent over the local ISP to a nearby enter-deep cache, from where the static content is retrieved.

While providing the static content to the client, the nearby cache also forwards the query over Google’s private network to one of the mega data centers, from where the personalized search results are retrieved.

For a YouTube video, the video itself may come from one of the bring-home caches, whereas portions of the Web page surrounding the video may come from the nearby enter-deep cache, and the advertisements surrounding the video come from the data centers.

In summary, except for the local ISPs, the Google cloud services are largely provided by a network infrastructure that is independent of the public Internet.

CDN cluster selection strategy

- *challenge:* how does CDN DNS select "good" CDN node to stream to client

  - pick CDN node geographically closest to client

  Drawback: ignores the variation in delay and available bandwidth over time of Internet paths, always assigning the same cluster to a particular client.

  - pick CDN node with shortest delay (or min # hops) to client (CDN nodes periodically ping access ISPs, reporting results to CDN DNS-makes periodic real-time measurements of delay and loss performance between their clusters and clients.)

  Drawback: Many LDNSs are configured to not respond to such probes.

- Use the characteristics of recent and ongoing traffic between the clients and CDN servers. For instance, the delay between a client and a cluster can be estimated by examining the gap between server-to-client SYNACK and client-to-server ACK during the TCP three-way handshake.

- Another alternative for cluster-to-client path probing is to use DNS query traffic to measure the delay between clients and clusters. Specifically, during the DNS phase the client's LDNS can be occasionally directed to different DNS authoritative servers installed at the various cluster locations, yielding DNS traffic that can then be measured between the LDNS and these cluster locations.

- IP anycast

- *alternative:* let *client* decide - give client a list of several CDN servers client pings servers, picks "best"

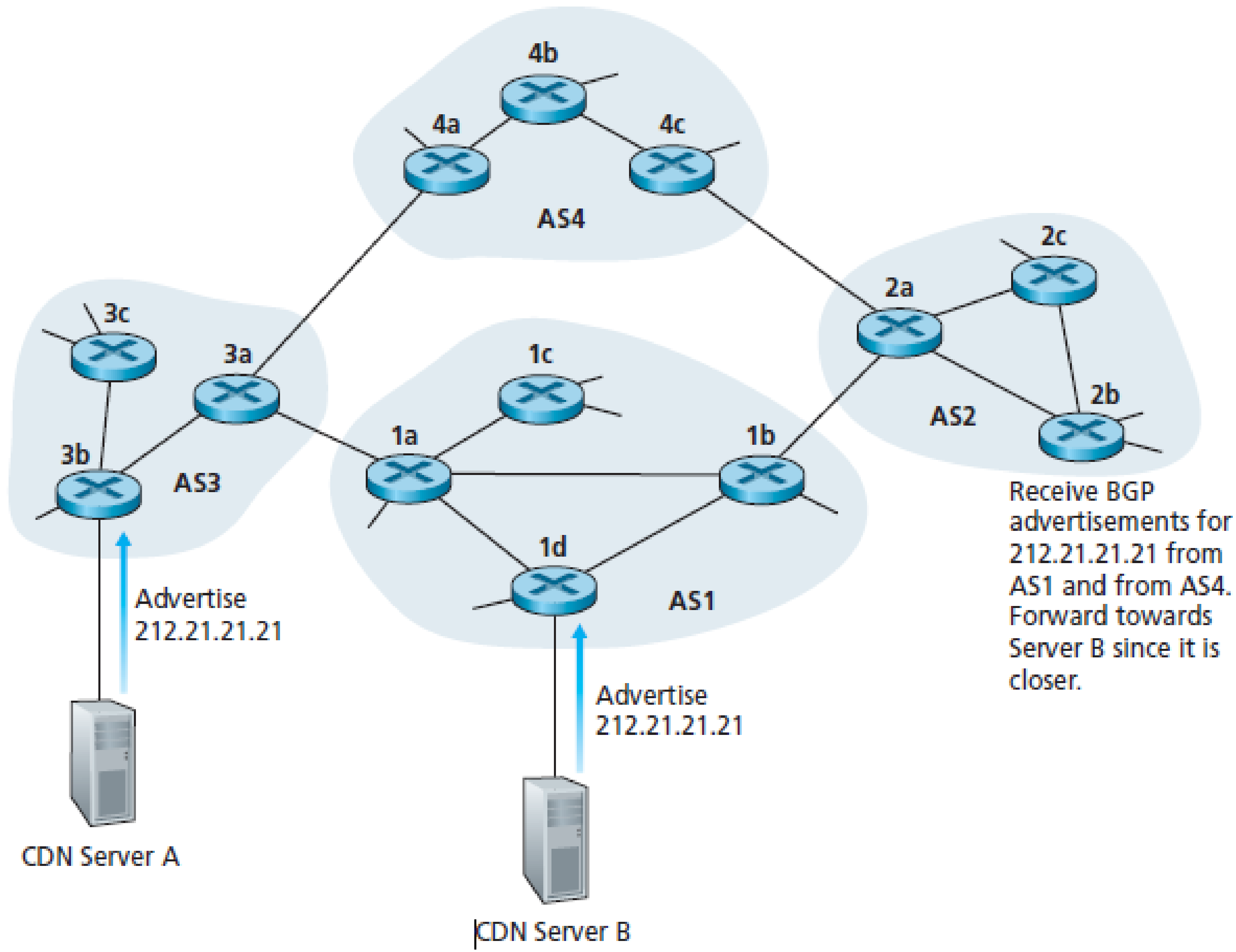
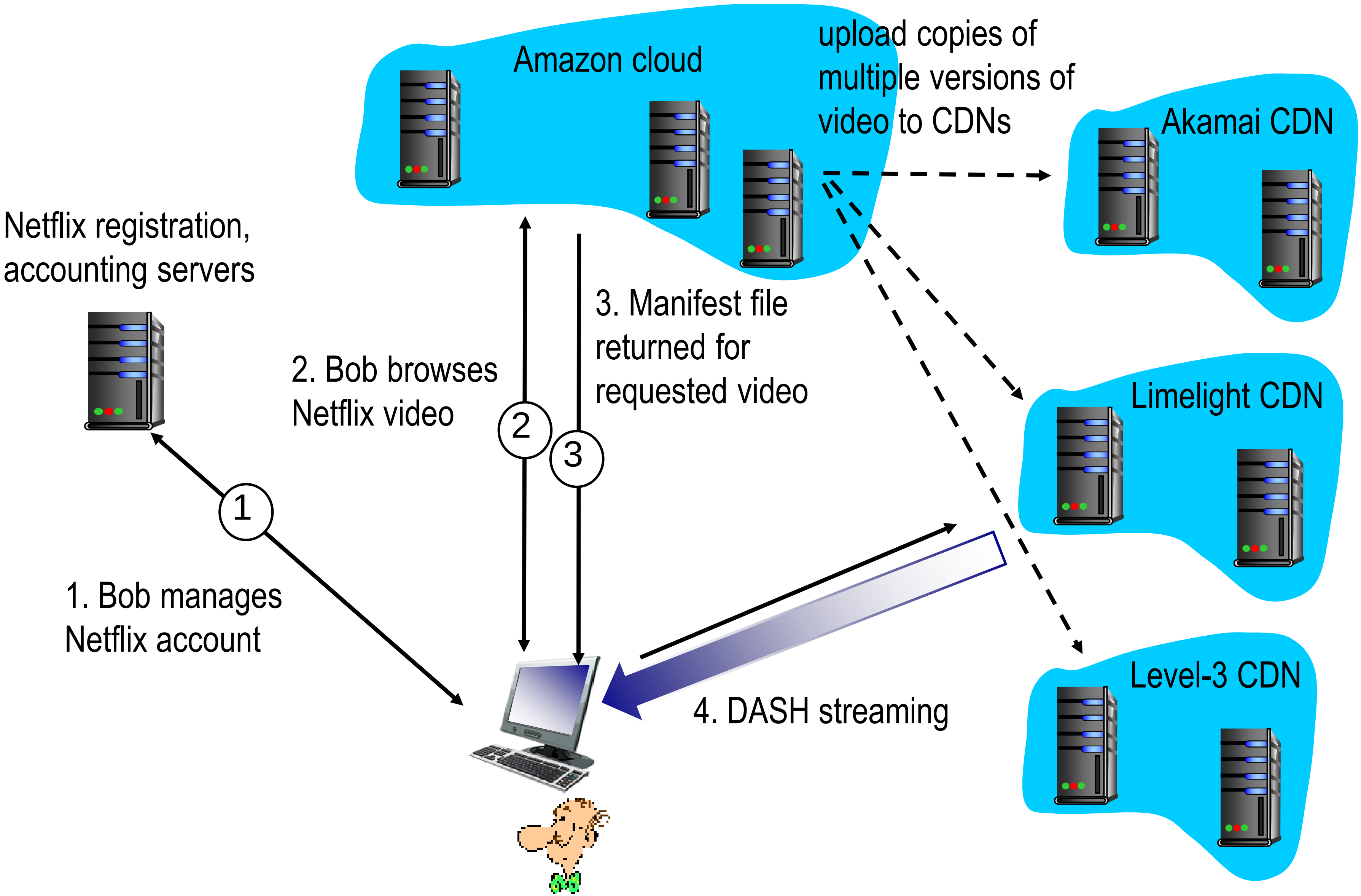


Figure 7.5 ♦ Using IP anycast to route clients to closest CDN cluster

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

Case study: Netflix



Some of the functions taking place in the Amazon cloud include:

- *Content ingestion.*

- *Content processing.*

- *Uploading versions to the CDNs.*

- **Limelight Networks** is a company that provides global (CDN) services that enable organizations to deliver their digital content (e.g. videos, operating system updates, online games, etc.) to any device, anywhere in the world.

- As of December 2014, the company's network has over 80 points-of-presence and 11 Terabits per second of egress capacity across the globe. The company is based in Tempe, Arizona**,** U.S.A.

- **Level3 communications** is a big company that has been providing a wide range of content delivery services for over 20 years. Level3 owns a Tier 1 backbone and has over 5.6 Tbps of peering capacity.

- The Level3 CDN services allow for token-based authentication, traffic prioritization, resource popularity tracking etc.

- **Akamai Technologies, Inc.** is a CDN and cloud service provider headquartered in Cambridge,USA. Akamai's CDN is one of the world's largest distributed computing platforms.

- The company operates a network of servers around the world and rents capacity on the servers to customers who want their websites to work faster by distributing content from locations close to the user.

- Some of its customers are apple, facebook, bing, google, ebay etc.

- The company was founded in 1998 by Daniel M Lewin (then a graduate student at MIT) and MIT applied maths professor Tom Leighton.

Case study: YouTube

- YouTube employs HTTP streaming.
- YouTube uses the HTTP byte range header in request to limit the flow of transmitted data after a target amount of video is prefetched.
- Supports Uploading.
- Google runs the entire YouTube service within its own vast infrastructure of data centers, private CDN, and private global network interconnecting its data centers and CDN clusters.

Case study: Kankan

- Kankan uses P2P delivery instead of client-server (via CDNs) delivery.

- This allows the service provider to significantly reduce its infrastructure and bandwidth costs.

- P2P video delivery approach is used with great success by several companies in China, including Kankan (owned and operated by Xunlei), PPTV (formerly PPLive), and PPs (formerly PPstream).

- Kankan, is currently the leading P2P-based video-on-demand provider in China, with over 20 million unique users viewing its videos every month.

- ✓ *The Kankan design employs a tracker and its own DHT for tracking content.*

- ✓ *Swarm sizes for the most popular content involve tens of thousands of peers, typically larger than the largest swarms in BitTorrent.*

- ✓ *The Kankan protocols—for communication between peer and tracker, between peer and DHT, and among peers—are all proprietary.*

- ✓ *Interestingly, for distributing video chunks among peers, Kankan uses UDP .*