دانشکده آموزش های الکترونیکی دانشگاه شیراز

مالتی مدیا در شبکه- بخش دوم

سیستم های چند رسانه ای

وحید اعتمادی

تابستان 94

Outline:

- We will review in this lecture:
  - Content distribution network(CDN)
  - Voice over IP
  - Network-level support for multimedia applications

CDN: introduction

- Today, many Internet video companies are distributing on-demand multi-Mbps streams to millions of users on a daily basis
- The most straightforward approach to providing streaming video service is to build a **single massive data center**

3 Major problems with single data center:

- the client is far from the data center

- فاصله زیاد بین مشتری و سرویس دهنده

- a popular video will likely be sent many times over the same communication links

- نیاز به ارسال یک ویدئو برای چندین بار از یک کانال مشخص

- a single data center represents a single point of failure

- وجود پتانسیل بالقوه برای خطای SPOF

CDN

- meet the challenge of distributing massive amounts of video data
- almost all major video-streaming companies make use of **Content Distribution Networks (CDNs)**
- provide a *system level* approach for delivering multimedia content

CDN

- A CDN manages servers in multiple geographically distributed locations, stores copies of the videos (and other types of Web content, including documents, images, and audio) in its servers, and attempts to direct each user request to a CDN location that will provide the best user experience

Type of CDN:

- **private CDN**: owned by the content provider itself
  - Ex. : Google

- **third-party CDN** that distributes content on behalf of multiple content providers
  - Ex. :Akamai

CDN Operation

- a browser in a user's host is instructed to retrieve a specific video (identified by a URL)
- CDNs take advantage of DNS to intercept and redirect requests
- CDN served as:
  1. determine a suitable CDN server cluster for that client at that time
  2. redirect the client's request to a server in that cluster

CDN Operation-page 604

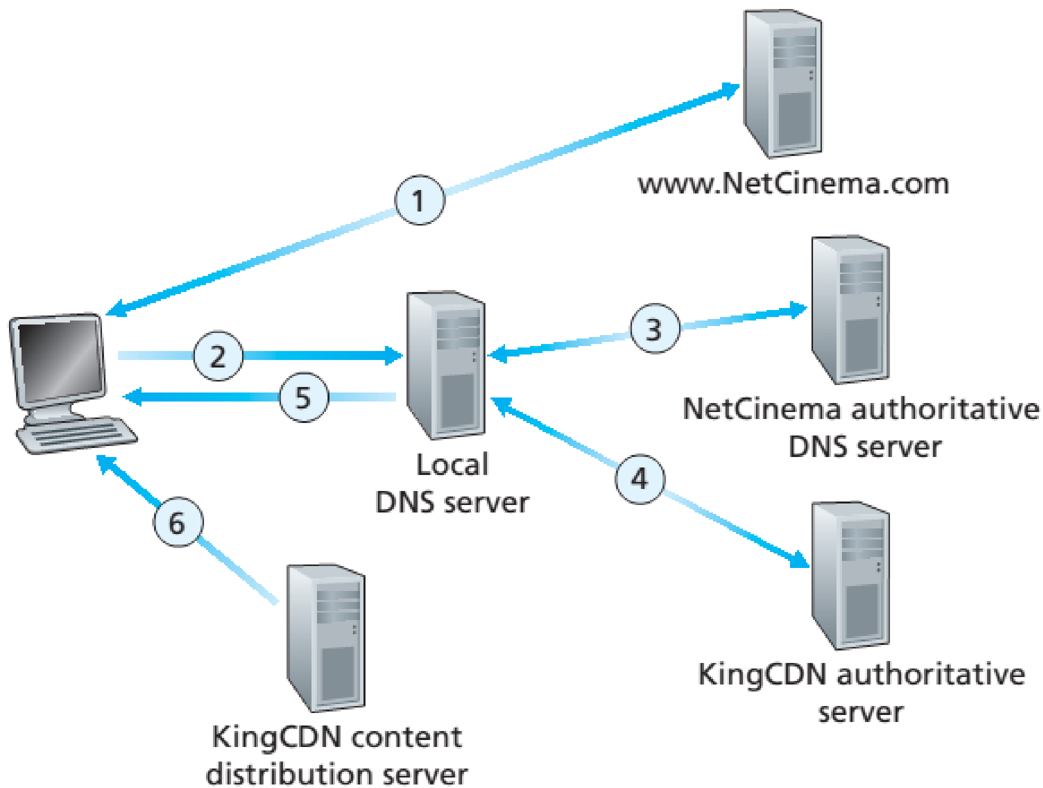

Figure 7.4 ♦ DNS redirects a user's request to a CDN server

CDN: A case studies

- Netflix, Kankan
- It will be Presented in the next lecture(lecture10)....

Voice-over-IP:

- **Internet telephony**
- principles and protocols underlying VoIP
- The Internet’s network-layer protocol, IP, provides best-effort service
- Sensitivity to packet delay, jitter, and loss
- **ways** in which the performance of VoIP over a best-effort network can be enhanced!!
  – focus will be on application-layer techniques

Voice-over-IP:

- The limitations of best-effort IP service in the context of a specific VoIP example(page 612)
  - The sender generates bytes at a rate of 8,000 bytes
  - every 20 msecs the sender gathers these bytes into a chunk
  - the number of bytes in a chunk is (8,000 bytes/sec) = 160 bytes
- the receiver must take more care in determining:
  - when to play back a chunk
  - what to do with a missing chunk

Packet Loss:

- Loss can occur in router buffers for IP datagrams(packet encapsulate as UDP packet)
- Loss could be eliminated by sending the packets over TCP (which provides for reliable data transfer) rather than over UDP
  – What’s problem for this approach?

Packet Loss:

- Is losing packets necessarily disastrous?
  - packet loss rates between 1 and 20 percent can be tolerated
  - Forward error correction (FEC) can help conceal packet loss
  - Lost original data can be recovered from the redundant information
  - Hint: best-effort service has its limitations

End-to-End Delay

- **End-to-end delay** is the accumulation of transmission, processing, and queuing delays in routers; propagation delays in links; and end-system processing delays
- End-to-end delay in real-time conversational applications
  - Smaller than 150 msecs are not perceived by a human listener
  - between 150 and 400 msecs can be acceptable but are not ideal
  - exceeding 400 msecs can seriously hinder the interactivity in voice conversations

End-to-End Delay

- packets that are delayed by more than the threshold are effectively lost

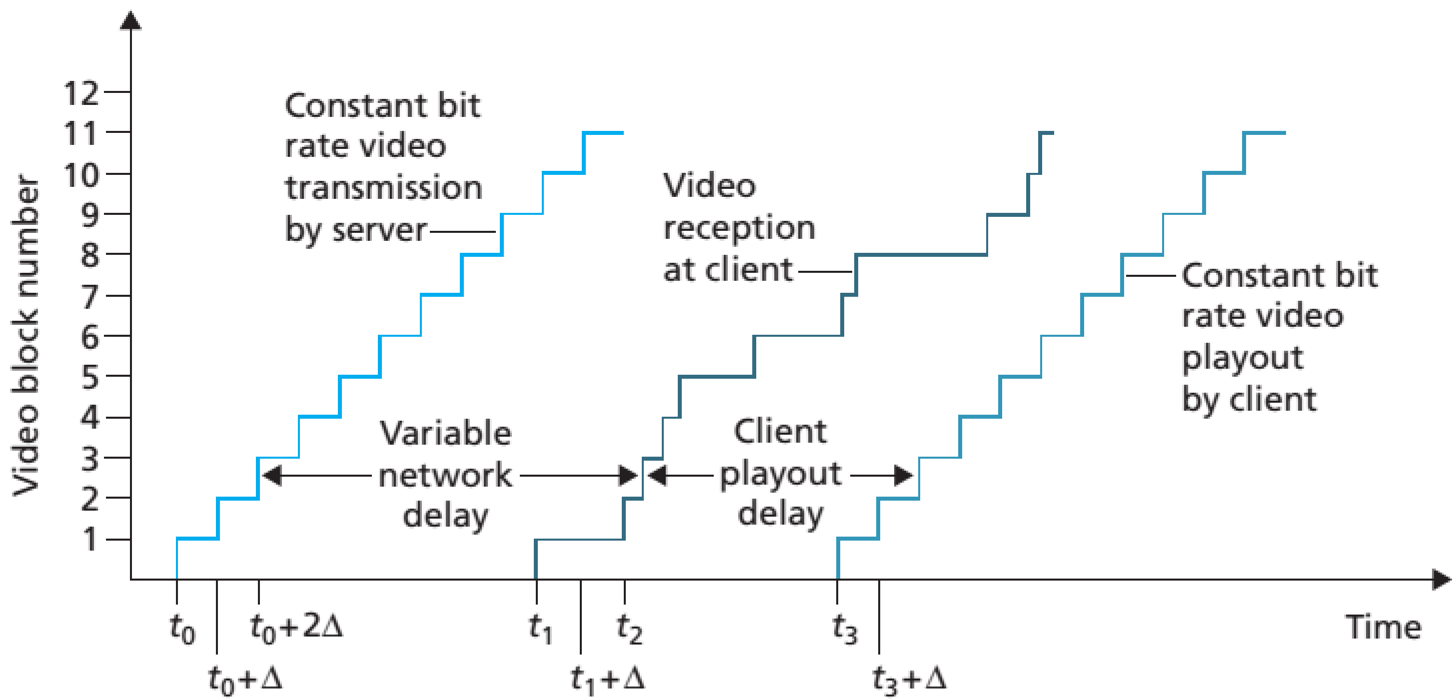

Figure 7.1 ♦ Client playout delay in video streaming

Packet Jitter

- The time from when a packet is generated at the source until it is received at the receiver can **fluctuate from packet to packet**
  - Example:
    - The sender sends the second packet 20 msecs after sending the first packet. But at the receiver, the spacing between these packets can become greater than 20 msecs.

Packet Jitter

- This situation is analogous to driving cars on roads
- If the receiver plays out chunks as soon as they arrive , then the resulting audio quality can easily become unintelligible at the receiver
- Removing jitter by:
  - **sequence numbers**
  - **Timestamps**
  - **playout delay**

Recovering from Packet Loss

- **Forward Error Correction (FEC):**
  - In this manner if any one packet of the group of n + 1 packets is lost, the receiver can fully reconstruct the lost packet
  - Other FEC mechanism is to send a lower-resolution audio stream as the redundant information
- **Interleaving:** the sender resequences units of audio data before transmission, so that originally adjacent units are separated by a certain distance in the transmitted stream

Forward Error Correction

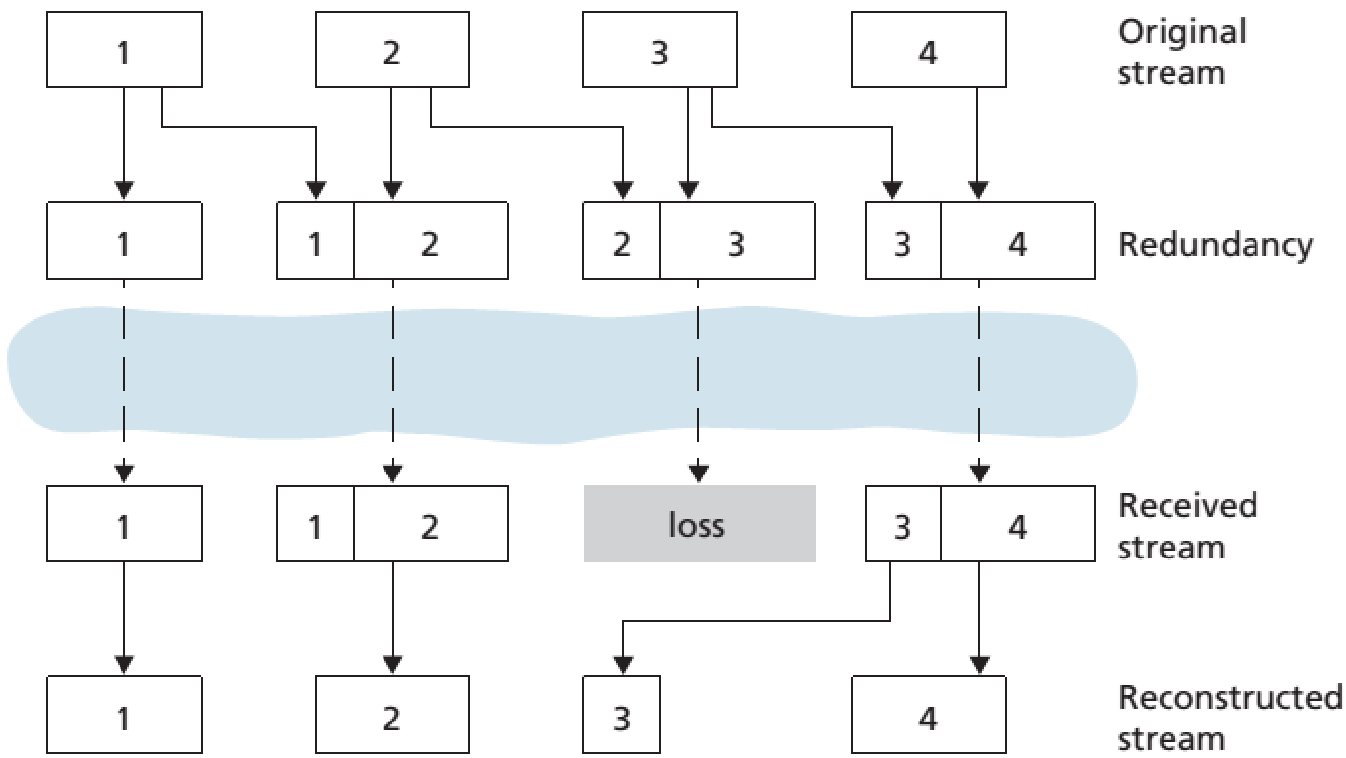

Figure 7.8 ♦ Piggybacking lower-quality redundant information

Recovering from Packet Loss: Interleaving

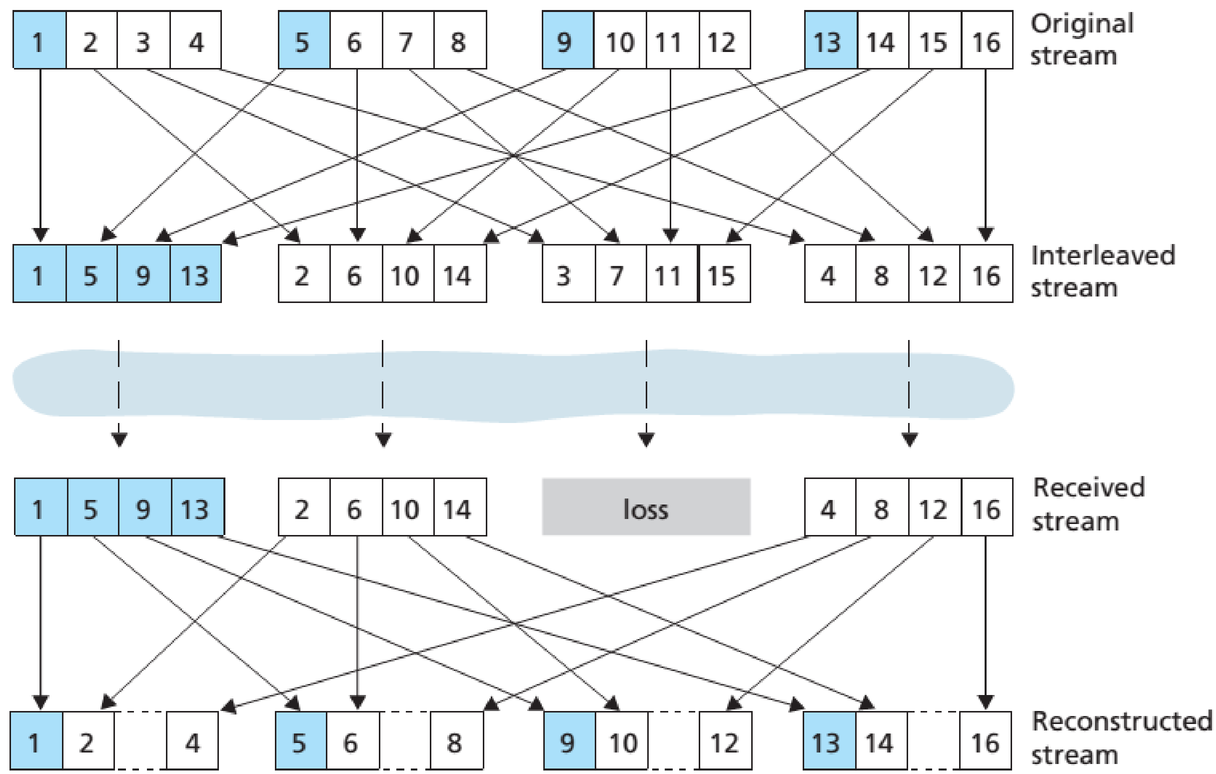

Figure 7.9 ♦ Sending interleaved audio

3 network-level support for multimedia applications

- ***Making the best of best-effort service***: The application-level mechanisms and infrastructure can be successfully used in a well-dimensioned network
- ***Differentiated service:*** providing different classes of service, rather than a single one-size-fits-all best-effort service
  – Type-of-Service field in the IP4v packet header

Differentiated service features:

- one type of traffic might be given strict priority over another class of traffic(in one time) when both types of traffic are queued at a router
  - Ex. : Packets belonging to a real-time conversational application might be given priority over other packets due to their stringent delay constraints

3 network-level support for multimedia applications

- ***Per-connection Quality-of-Service (QoS) Guarantees****:* each instance of an application explicitly reserves end-to-end bandwidth and thus has a guaranteed end-to-end performance
  - **hard guarantee(certainty)** vs. **soft guarantee(high probability)**

Approach	Granularity	Guarantee	Mechanisms	Complexity	Deployment to date
Making the best of best-effort service.	all traffic treated equally	none, or soft	application-layer support, CDNs, overlays, network-level resource provisioning	minimal	everywhere
Differentiated service	different classes of traffic treated differently	none, or soft	packet marking, policing, scheduling	medium	some
Per-connection Quality-of-Service (QoS) Guarantees	each source-destination flows treated differently	soft or hard, once flow is admitted	packet marking, policing, scheduling; call admission and signaling	light	little

Table 7.4 ♦ Three network-level approaches to supporting multimedia applications