COMS3200 – Week 10 Multimedia Applications and Protocols

School of Information Technology and Electrical Engineering
The University of Queensland

Context …

- So far
  - Physical -> Transport Layer
  - Last time: Routing
- Coming up
  - Quality of Service
  - Network Security

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Application layer

- Covered in COMP2303
  - WWW (HTTP protocol)
  - Email (message format, SMTP, POP3/IMAP)
  - Domain Name System (DNS) and Name Servers
- Application layer in COMS3200
  - Routing protocols
  - Multimedia applications

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Outline

- Multimedia applications
  - Types of applications
  - Quality of Service (QoS) parameters
  - QoS requirements for multimedia applications
- Digital audio
- Streaming audio
- Internet radio
- Voice over IP
- Video on demand
  - Video servers
  - Distribution networks

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Learning objectives

After this week, you should be able to

- describe multimedia applications and their QoS requirements
- describe application layer protocols supporting multimedia

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Multimedia

- Multimedia – combination of two or more continuous media
  - E.g. audio and video
- Streaming audio, Internet telephony (VoIP), Internet radio are also often called "multimedia"
- Real-time multimedia have stringent Quality of Service (QoS) requirements

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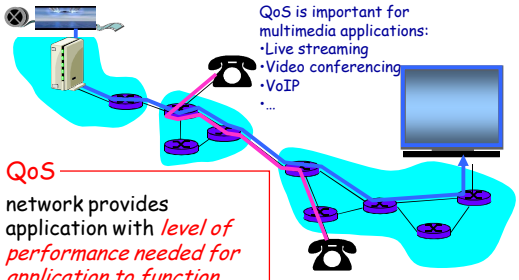
Quality of Service: What is it?



QoS
network provides application with *level of performance needed for application to function.*

QoS Parameters?

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QoS Parameters

- Data rate (bandwidth)
- Delay (latency)
- Jitter (Delay variation)
- Reliability
  - Bit error rate
  - Packet error rate

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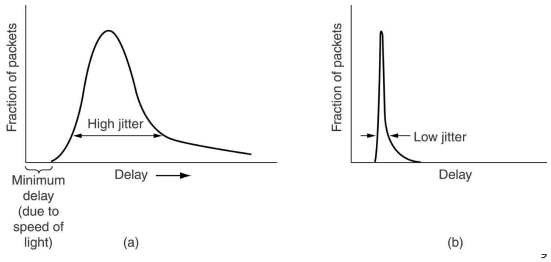
QoS Parameters: Jitter

Jitter: Variation (standard deviation) in packet delay



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QoS Requirements of Multimedia Applications

Classes of MM applications:
1) Streaming stored audio and video
2) Streaming live audio and video
3) Real-time interactive audio (VoIP) and video

Fundamental characteristics:
- Typically **delay sensitive**
  - end-to-end delay
  - delay jitter
- But **loss tolerant**: infrequent losses cause minor glitches
- Antithesis of data, which are loss intolerant but delay tolerant.

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Digital Audio (1)

- Audio – acoustic (pressure) wave
  - Generates electrical signal in microphones (analog signal)
  - Audio signals can be converted to digital form
    - Recall Analog Digital Converter from Lecture on Physical Layer (amplitude sampling)
    - Conversion to Digital introduces some error (quantization noise)

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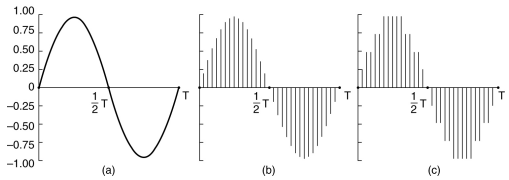
Digital Audio (2)

(a) A sine wave
(b) Sampling the sine wave
(c) Quantizing the samples to 4 bits



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Digital Audio (3)

- Examples:
  - Telephone
    - 8000 samples/sec, frequencies above 4kHz lost
    - 7 or 8-bit samples
    - requires bandwidth of 64 kbps
  - Audio compact discs
    - 44,100 samples/sec, frequencies up to 22,050 KHz
    - 16-bit samples
    - Requires bandwidth of 1.411 Mbps (if not compressed)
- Most popular compression – MP3 (MPEG audio layer 3)

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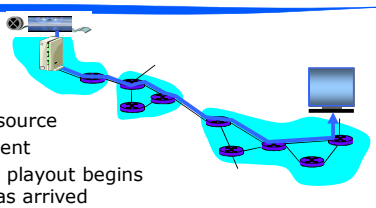
Streaming Stored Multimedia (audio, video)



Streaming:
- media stored at source
- transmitted to client
- streaming: client playout begins *before* all data has arrived

- timing constraint for still-to-be transmitted data: in time for playout

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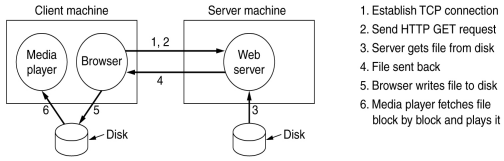
Streaming stored media

- Listening to sound or requesting video on demand over the Internet
- Simple implementation:



1. Establish TCP connection
2. Send HTTP GET request
3. Server gets file from disk
4. File sent back
5. Browser writes file to disk
6. Media player fetches file block by block and plays it

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Streaming stored media

- Due to long downloading time media players use different approach
  - The file is downloaded by a media player (HTTP not used)
  - Real Time Streaming Protocol is used to control the media server
  - The media player starts playing before all data has arrived

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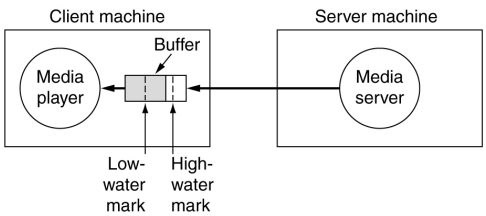
Streaming stored media

The media player buffers input from the media server and plays from the buffer rather than directly from the network



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Streaming stored media

- RTSP is an *out-of-band* protocol
  - RTSP packets sent to a different port than data

Command	Server action
DESCRIBE	List media parameters
SETUP	Establish a logical channel between the player and the server
PLAY	Start sending data to the client
RECORD	Start accepting data from the client
PAUSE	Temporarily stop sending data
TEARDOWN	Release the logical channel

RTSP commands from the player to the server

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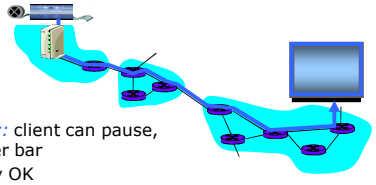
Streaming stored media: Interactivity



- *VCR-like functionality:* client can pause, rewind, FF, push slider bar
  - 10 sec initial delay OK
  - 1-2 sec until command effect OK
  - RTSP often used (Real Time Streaming Protocol)

timing constraint for still-to-be transmitted data: in time for playout

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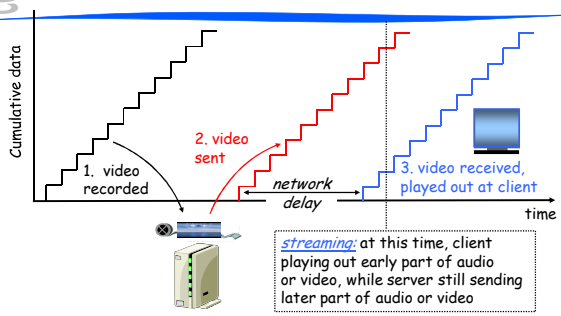
Streaming stored media: delayed playback



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Streaming stored media - summary

- Clients request compressed audio/video files that reside on servers
- Media player is responsible for
  - Decompression on the fly during playout
  - Jitter removal (buffering, delayed playback)
  - Error correction for lost packets
    - Transmission of redundant packets
    - Interpolation of missing data

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Streaming Live Multimedia

Examples:
- Internet radio talk show
- Live sporting event

Streaming
- playback buffer
- playback can lag tens of seconds after transmission
- still have timing constraint

Interactivity
- fast forward impossible
- rewind, pause possible!

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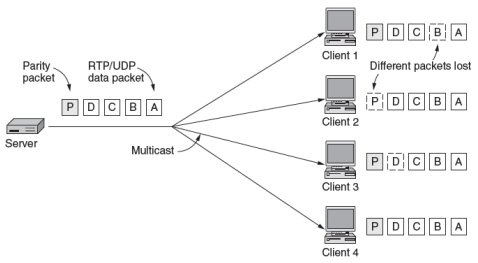
Streaming Live Media



Multicast streaming media with a parity packet

Internet radio

- Two approaches
  - Prerecorded, stored on disk
    - Streaming audio
  - Broadcast live over the Internet
    - Differences to streaming audio
      - Removing jitter is more difficult
      - Packets have to be delivered at the rate they are played back
      - Many simultaneous listeners not point to point as streaming audio
        - » should use multicast but is still mostly delivered over TCP

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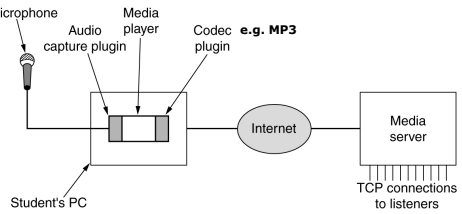
Internet Radio

A simple radio station



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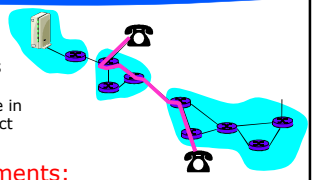
Interactive, Real-Time Multimedia

- applications: IP telephony (VoIP), video conference, distributed interactive worlds
  (an internet-based multi-user VR system where participants navigate in 3D space and see, meet and interact with other users and applications)



- end-end delay requirements:
  - audio: < 150 msec good, < 400 msec OK
    - includes application-level (packetisation) and network delays
    - higher delays noticeable, impair interactivity or make it impossible

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QoS for Real-time interactive applications

- PC-2-PC phone
  - instant messaging services are providing this
- PC-2-phone
  - Dialpad (part of Yahoo! Messenger with Voice)
  - Net2phone
- videoconference with Webcams

Going to now look at a PC-2-PC Internet phone example in detail

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Interactive Multimedia: Voice over IP (VoIP)

Internet Phone (VoIP) by way of an example

- speaker's audio: alternating talk spurts, silent periods
  - 64 kbps during talk spurt
- pkts generated only during talk spurts
  - 20 msec chunks at 8 Kbytes/sec: 160 bytes of data
- application-layer header added to each chunk
- Chunk+header encapsulated into UDP segment
- application sends UDP segment into socket every 20 msec during talkspurt

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VoIP Packet Loss and 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, losses concealed, packet loss rates up to 10% can be tolerated.

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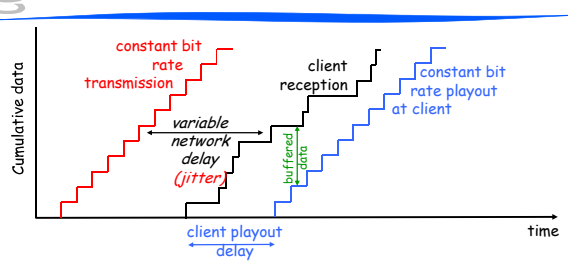
Delay Jitter



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

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VoIP: Fixed Playout Delay

- Fixed playout delay: q msecs
- Receiver attempts to playout each chunk exactly q msecs after chunk was generated (synchronised)
  - chunk has time stamp t: play out chunk at t+q .
  - chunk arrives after t+q: data arrives too late for playout, data "lost"
- Tradeoff for q:
  - large q: less packet loss
  - small q: better interactive experience
- → Adaptive playout delay

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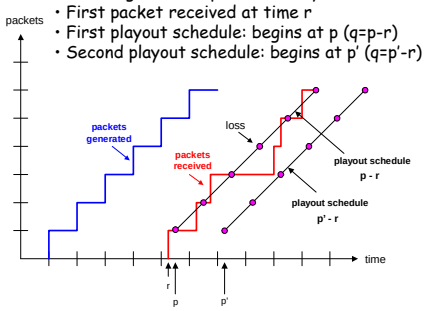
Fixed Playout Delay

- Sender generates packets every 20 msec during talk spurt.
- First packet received at time r
- First playout schedule: begins at p (q=p-r)
- Second playout schedule: begins at p' (q=p'-r)



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Adaptive Playout Delay

- Packet requires
  - Sequence number
  - Timestamp
- Estimation of network delay is calculated
  - Emphasis on recent delays
  - Similar to round-trip estimation in TCP
- Packet playout time depends on delay estimation

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VoIP Protocols

- Two approaches
  - Telecommunication community
    - H.323 ITU recommendation
      - References a number of protocols (speech coding, call setup, signaling, data transport, etc.)
  - Internet community
    - SIP - Session Initiation Protocol (IETF RFC)

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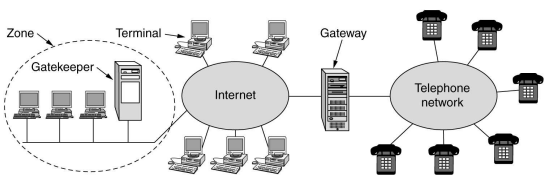
Voice over IP – H323

The H323 architectural model for Internet telephony – ITU model



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Voice over IP – H323

The H323 protocol stack

Speech	Control			
G.7xx	RTCP	H.225 (RAS)	Q.931 (Call signaling)	H.245 (Call control)
RTP				
UDP			TCP	
IP				
Data link protocol				
Physical layer protocol				

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Voice over IP – H323

- G.7xx – speech encoding and decoding
- RTP, RTCP – Real time transport protocol (will be described in lecture on QoS)
- H.245 - negotiation of encoding
- Q.931– call signaling
  - establishing and releasing connections
  - dial tones
  - making ringing sounds, etc
- H.225 – protocol for talking to gatekeeper (registration/admission/status)

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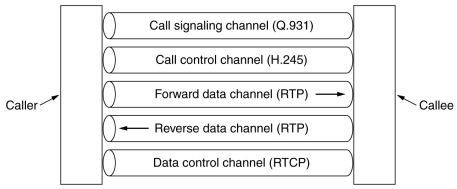
Voice over IP – H323

Logical channels between the caller and callee during a call



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Voice over IP - SIP

Another approach to Voice over IP - Session Initiation Protocol from IETF

- RFC 3261
- Lightweight protocol for
  - Two-party sessions
  - Multi-party sessions
  - Multicast sessions
- Telephone numbers are defined as URLs
- SIP is a text-based protocol similar to HTTP

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Voice over IP - SIP

The SIP methods defined in the core specification

Method	Description
INVITE	Request initiation of a session
ACK	Confirm that a session has been initiated
BYE	Request termination of a session
OPTIONS	Query a host about its capabilities
CANCEL	Cancel a pending request
REGISTER	Inform a redirection server about the user's current location

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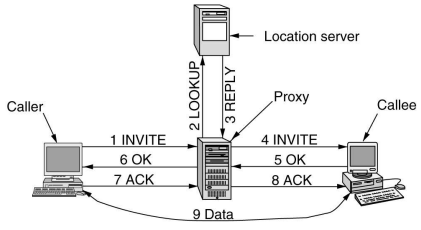
Voice over IP - SIP

A proxy and redirection servers are used in SIP



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Comparison of H.323 and SIP

Item	H.323	SIP
Designed by	ITU	IETF
Compatibility with PSTN	Yes	Largely
Compatibility with Internet	Yes, over time	Yes
Architecture	Monolithic	Modular
Completeness	Full protocol stack	SIP just handles setup
Parameter negotiation	Yes	Yes
Call signaling	Q.931 over TCP	SIP over TCP or UDP
Message format	Binary	ASCII
Media transport	RTP/RTCP	RTP/RTCP
Multiparty calls	Yes	Yes
Multimedia conferences	Yes	No
Addressing	URL or phone number	URL
Call termination	Explicit or TCP release	Explicit or timeout
Instant messaging	No	Yes
Encryption	Yes	Yes
Size of standards	1400 pages	250 pages
Implementation	Large and complex	Moderate, but issues
Status	Widespread, esp. video	Alternative, esp. voice

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Context-aware SIP (not examinable)

- Self-adapting SIP communications (developed by Indulska et al)
  - Context information and preference based
    - Call connection
      - Route call to best device based on current user context
    - Call redirection
      - Context change
        - » Device failure
        - » Location change
      - Transfer call based on new context

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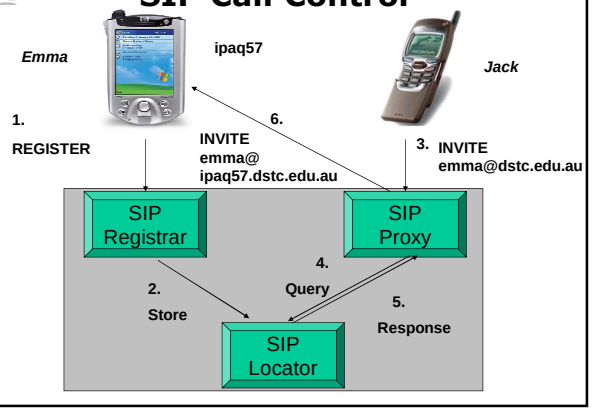
SIP Call Control



Context-aware SIP

- Make SIP locator responses context sensitive
  - Return contact addresses based on context information, user preferences, and locator system preferences

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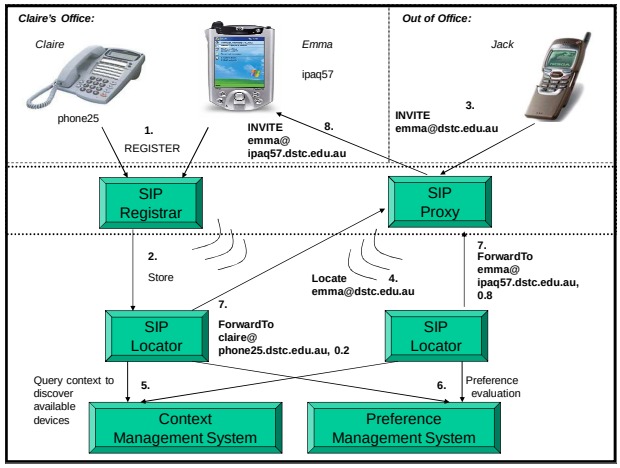
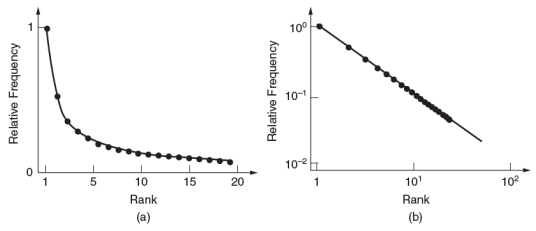


Content Delivery

- Content and internet traffic
- Server farms and web proxies
- Content delivery networks
- Peer-to-peer networks

Content and Internet Traffic



Zipf distribution (a) On a linear scale. (b) On a log-log scale.

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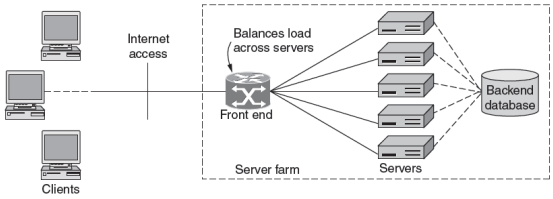
Server Farms and Web Proxies



A server farm

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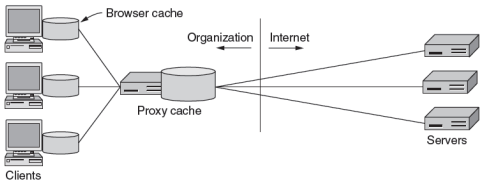
Server Farms and Web Proxies



A proxy cache between Web browsers and Web servers

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Content Delivery Networks

- Delivery from content providers to clients
- Content Delivery Networks (CDNs)
  - Content providers are their clients
  - Content provider pays CDN to deliver video to user

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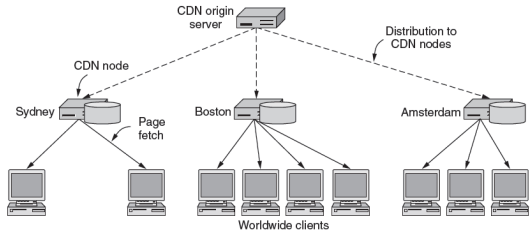
Content Delivery Networks



CDN distribution tree

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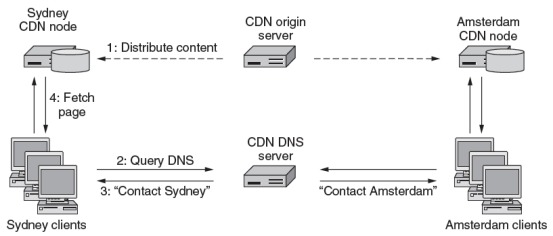
Content Delivery Networks



Directing clients to nearby CDN nodes using DNS

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Peer-to-Peer Networks

- There are many P2P protocols that allow to share content
- Currently the most popular is BitTorrent

Problems to be solved with BitTorrent sharing:

1. How is content replicated by peers to provide high-speed downloads
2. How do peers encourage each other to upload content to others
3. How does a peer find other peers

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Bit Torrent Terminology

- Torrent – a content descriptor, often from website
- Tracker – leads peers to content
- Chunks – content divided into small slices
- Swarm – set of peers currently working with desired content, as listed by tracker
- Seeders – members of swarm with complete content, needed to start download
- Free-riders or leechers don't upload, swarm members will "choke" their access to content

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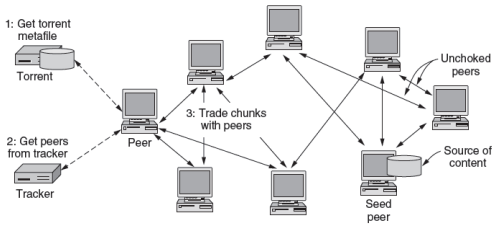
Peer-to-Peer Networks



BitTorrent

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Summary

- Multimedia delivery
  - Audio, video streaming
  - Interactive multimedia applications
- Stringent QoS requirements
  - Bandwidth
  - Delay
  - Jitter
- Application layer protocols supporting multimedia
- Lower level protocols will be covered in lecture on QoS

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Readings

- Tanenbaum 5th ed., sections
  - 7.4 – 7.4.1, 7.4.3, 7.4.4, 7.4.5,
  - 7.5 – 7.5.1, 7.5.2, 7.5.3, 7.5.4(BitTorent)

4th ed., 7.4 – Multimedia (without 7.4.2, 7.4.6 and 7.4.7)

- Next week – Quality of Service
  - Tanenbaum 5th ed: 5.4, 5.6.5, 6.4.3

(4th ed 5.3 (5.3.1, 5.3.2), 5.4, 6.4.3)

- NOTE: no lecture on Friday

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