CS529
Multimedia Networking

Introduction

Objectives

- Brief introduction to:
  - Digital Audio
  - Digital Video
  - Perceptual Quality
  - Network Issues
- Get you ready for research papers!
- Introduction to:
  - Silence detection (for Project 1)

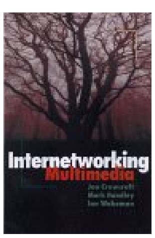
Groupwork

- Let's get started!
- Consider audio or video on a computer
  - Examples you have seen, or
  - Systems you have built
- What are two conditions that degrade quality?
  - Describing appearance is ok
  - Giving technical name is ok

Introduction Outline

- Foundation
  - ***Internetworking Multimedia (Ch 4)***
  - Perceptual Coding: How MP3 Compression Works (Sellars)
  - Graphics and Video (Linux MM, Ch 4)
  - Multimedia Networking (Kurose, Ch 7)
- Audio Voice Detection (Rabiner)
- Video Compression

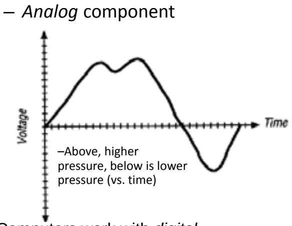
(These Slides)



[CHW99] J. Crowcroft, M. Handley, and I. Wakeman. Internetworking Multimedia, Chapter 4, Morgan Kaufmann Publishers, 1991, ISBN 1-55860-584-3.

Digital Audio

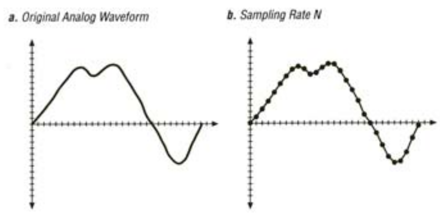
- Sound produced by variations in air pressure
  - Can take any continuous value
  - *Analog* component



- Computers work with *digital*
  - Must convert analog to digital
  - Use *sampling* to get discrete values

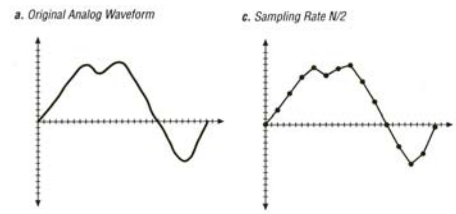
Digital Sampling

- *Sample rate* determines number of discrete values



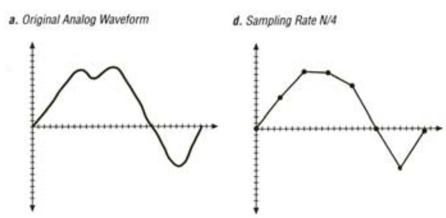
Digital Sampling

- Half sample rate



Digital Sampling

- Quarter sample rate



(How often to sample to reproduce curve?)

Sample Rate

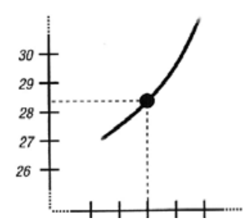
- *Shannon's Theorem*: to accurately reproduce signal, must sample at twice highest frequency
- Why not always use high sampling rate?

Sample Rate

- *Shannon's Theorem*: to accurately reproduce signal, must sample at twice highest frequency
- Why not always use high sampling rate?
  - Requires more storage
  - Complexity and cost of analog to digital hardware
  - Human's can't always perceive
    - Dog whistle
  - Typically want an "*adequate*" sampling rate
    - "Adequate" depends upon use of reconstructed signal

Sample Size

- Samples have discrete values



- How many possible values?
  - *Sample Size*
  - Say, 256 values from 8 bits

Sample Size

- *Quantization error* from rounding
  - Ex: 28.3 rounded to 28
- Why not always have large sample size?

Sample Size

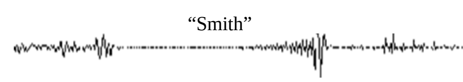
- *Quantization error* from rounding
  - Ex: 28.3 rounded to 28
- Why not always have large sample size?
  - Storage increases per sample
  - Analog to digital hardware becomes more expensive

Groupwork

- Think of as many uses of computer audio as you can
- Which require a high sample rate and large sample size? Which do not? Why?

Audio

- Encode/decode devices are called *codecs*
  - Compression is the complicated part
- For voice compression, can take advantage of speech:



- Many similarities between adjacent samples
  - Send differences (ADPCM)
- Use understanding of speech
  - Can ‘predict’ (CELP)

Audio by People

- Sound by breathing air past vocal cords
  - Use mouth and tongue to shape vocal tract
- Speech made up of phonemes
  - Smallest unit of distinguishable sound
  - Language specific
- Majority of speech sound from 60-8000 Hz
  - Music up to 20,000 Hz
- Hearing sensitive to about 20,000 Hz
  - Stereo important, especially at high frequency
  - Lose frequency sensitivity with age

Typical Encoding of Voice

- Today, telephones carry digitized voice
- 8000 samples per second
  - Adequate for most voice communication
- 8-bit sample size
- For 10 seconds of speech:
  - 10 sec x 8000 samp/sec x 8 bits/samp = 640,000 bits or 80 Kbytes
  - Fit 2 years of raw sound on typical hard disk
- Ok for voice (but Skype better), but what about music?

Typical Encoding of Audio

- Can only represent 4 KHz frequencies (why?)
- Human ear can perceive 10-20 KHz
  - Full range used in music
- CD quality audio:
  - sample rate of 44,100 samples/sec
  - sample size of 16-bits
  - 60 min x 60 secs/min x 44100 samp/sec x 2 bytes/samp x 2 channels (stereo)

    = 635,040,000, about 600 Mbytes (typical CD)
- Can use *compression* to reduce
  - mp3 ("as it sounds)", RealAudio
  - 10x compression rate, same audible quality

Sound File Formats

- Raw data has samples (interleaved w/stereo)
- Need way to 'parse' raw audio file
- Typically a header
  - Sample rate
  - Sample size
  - Number of channels
  - Coding format
  - …
- Examples:
  - .au for Sun µ-law, .wav for IBM/Microsoft
  - .mp3 for MPEG-layer 3

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MP3 – Introduction (1 of 2)

- "MP3" abbreviation of "MPEG 1 audio layer 3"
- "MPEG" abbrev of "Moving Picture Experts Group"
  - 1990, Video at about 1.5 Mbits/sec (1x CD-ROM)
  - Audio at about 64-192 kbits/channel
- Committee of the International Standards Organization (ISO) and International Electrotechnical Commission (IEC)
  - (Whew! That's a lot of acronyms (TALOA))
- MP3 differs in that it does not try to accurately reproduce PCM (waveform)
- Instead, uses theory of "perceptual coding"
  - PCM attempts to capture a waveform "as it is"
  - MP3 attempts to capture it "as it sounds"

MP3 – Introduction (2 of 2)

- Ears and brains imperfect and biased measuring devices, interpret external phenomena
  - Ex: doubling amplitude does not always mean double perceived loudness. Factors (frequency content, presence of any background noise…) also affect
- Set of judgments as to what is/not meaningful
  - *Psychoacoustic model*
- Relies upon "redundancy" and "irrelevancy"
  - Ex: frequencies beyond 22 KHz redundant (some audiophiles think it *does* matter, gives "color"!)
  - Irrelevancy, discarding part of signal because will not be noticed, was/is new

MP3 - Masking

- Listener prioritizes sounds ahead of others according to context (hearing is adaptive)
  - Ex: a sudden hand-clap in a quiet room seems loud. Same hand-clap after a gunshot, less loud (*time domain*)
  - Ex: guitar may dominate until cymbal, when guitar briefly drowned (*frequency domain*)
- Above examples of *time-domain* and *frequency-domain* masking, respectively
- Two sounds occur (near) simultaneously, one may be partially masked by the other
  - Depending relative volumes and frequency content
- MP3 doesn't just toss masked sound (would sound odd) but uses fewer bits for masked sounds

MP3 – Sub-Bands (1 of 2)

- MP3 not method of digital recording
  - Instead, removes irrelevant data from existing recording
- Encoding typically 16-bit sample size at 32, 44.1 and 48 kHz sample rate
- First, short sections of waveform stream filtered
  - How, not specified by standard
  - Typically *Fast Fourier Transformation* or *Discrete Cosine Transformation*
    - Method of reformatting signal data into spectral sub-bands of differing importance

MP3 – Sub-Bands (2 of 2)

- Divide into 32 "sub-bands" that represent different parts of frequency spectrum
- Why frequency sub-bands? So MP3 can prioritize bits for each
  - Ex:
    - Low-frequency bass drum, a high-frequency ride cymbal, and a vocal in-between, all at once
    - If bass drum irrelevant, use fewer bits and more for cymbal or vocals

MP3 – Frames

- Sub-band sections are grouped into "frames"
- Determine where masking in frequency and time domains will occur
  - Which frames can safely be allowed to distort
- Calculate *mask-to-noise* ratio for each frame
  - Use in the final stage of the process: bit allocation

MP3 – Bit Allocation

- Decides how many bits to use for each frame
  - More bits where little masking (low ratio)
  - Fewer bits where more masking (high ratio)
- Total number of bits depends upon desired bit rate
  - Chosen before encoding by user
- For quality, a high priority (music) 128 kbps common
  - Note, CD is about 1400 kbps, so *10x* less

MP3 – Playout and Beyond

- Save frames (header data for each frame). Can then play with MP3 decoder.
- MP3 decoder performs reverse, but simpler since bit-allocation decisions are given
  - MP3 decoders cheap, fast (ipod!)
- What does the future hold?
  - Lossy compression not needed since bits irrelevant (storage + net)?
  - Lossy compression so good that all irrelevant bits are banished?

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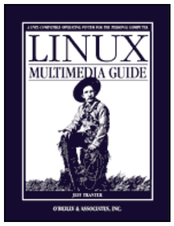


[Tr96] J. Tranter. Linux Multimedia Guide, Chapter 4, O'Reilly & Associates, 1996, ISBN: 1565922190

Graphics and Video

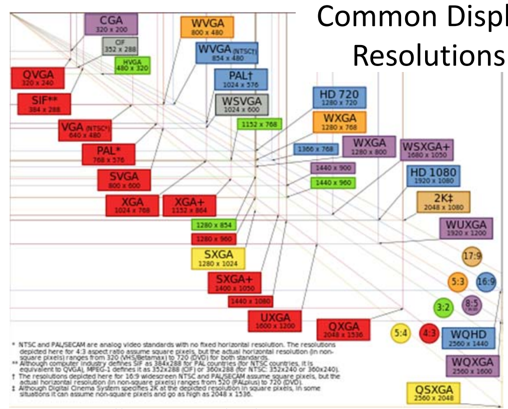
"A Picture is Worth a Thousand Words"

- People are visual by nature
- Many concepts hard to explain or draw
- Pictures to the rescue!
- Sequences of pictures can depict motion
  - Video!

Video Images

- Traditional television is 646x486 (NTSC)
- HDTV is 1920x1080 (1080p), 1280x720 (720p), 852x480 (480p)
- Often Internet video smaller
  - 352x288 (H.261), 176x144 (QCIF)
- Monitors higher resolution than traditional TV (see next slide)
- Computer video sometimes called "postage stamp"
  - If make full screen, then pixelated (jumbo pixels)

Common Display Resolutions



http://en.wikipedia.org/wiki/Display_resolution

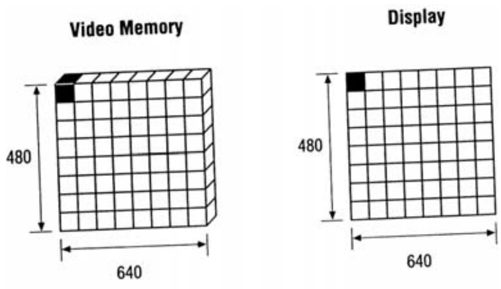
Video Image Components

- Luminance (Y) and Chrominance: Hue (U) and Intensity (V) - *YUV*
  - Human eye less sensitive to color than luminance, so those sampled with less resolution (e.g. 4 bits for Y, 2 for U, 2 for V – 4:2:2)
- YUV has backward compatibility with BW televisions (only had Luminance)
  - Monitors are typically Red Green Blue (RGB)
  - (Why are primary colors Red Yellow Blue?)

Graphics Basics

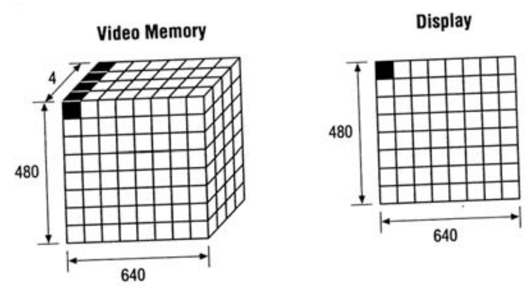
- Display images with graphics hardware
- Computer graphics (pictures) made up of pixels
  - Each pixel corresponds to region of memory
  - Called *video memory* or *frame buffer*
- Write to video memory
  - Traditional CRT monitor displays with raster cannon
  - LCD monitors align crystals with electrodes

Monochrome Display



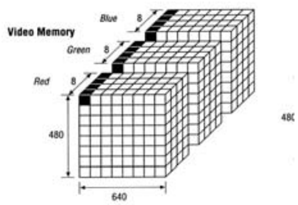
- Pixels are on (black) or off (white)
  - *Dithering* can make area appear gray

Grayscale Display



- *Bit-planes:* 4 bits per pixel, 2^4 = 16 gray levels
- Typically, 8 enough levels for perception (256 human max), but medical uses (e.g. x-ray) use 10- or 12-bit since sensors may detect. TIFF, PNG use 16-bit greyscale.

Color Displays



- Humans can perceive far more different colors than grayscales
  - Cones (color) and Rods (gray) in eyes
- All colors seen as combo of **red**, **green** and **blue** (additive)
- Visual maximum needed
  - 24 bits/pixel, 2^{24} ~ 16 million colors (true color)
- Requires 3 bytes per pixel

Sequences of Images – Video (Guidelines)

- Series of frames with changes appear as motion
- Units are frames per second (*fps* or *f/s*)
  - 24-30 fps: full-motion video
  - 15 fps: full-motion video approximation
  - 7 fps: choppy
  - 3 fps: very choppy
  - Less than 3 fps: slide show

Video Sizes

- *Raw* video bitrate:

  color depth * vertical rez * horizontal rez * frame rate

e.g. 1080p: 10-bit (4:4:2) @ 1920 x 1080 @ 29.97fps

= ~120 MB per/sec or ~430 GB per/hr

RESOLUTION	FORMAT	UNCOMPRESSED
720x480	480i 29.97fps	124.29 Mbps
1,280x720	720p30	663 Mbps
1,280x720	720p60	1.66 Gbps
1,920x1,080	1080i60	1.49 Gbps
1,920x1,080	1080p24	1.19 Gbps

Uncompressed video is big!

Video Compression

RESOLUTION	FORMAT	UNCOMPRESSED	COMPRESSED
720x480	480i 29.97fps	124.29 Mbps	2.49 Mbps
1,280x720	720p30	663 Mbps	13.26 Mbps
1,280x720	720p60	1.66 Gbps	33.2 Mbps
1,920x1,080	1080i60	1.49 Gbps	29.8 Mbps
1,920x1,080	1080p24	1.19 Gbps	25 Mbps

- Image compression: about 25 to 1
- Video compression: about 100 to 1
- Options: Lossless or Lossy
  - (Q: *why not always lossless?)*
- Intracoded or Intercoded
  - Take advantage of dependencies between frames → Motion (more later)

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[KR12] J. Kurose and K. Ross. *Computer Networking: A Top-Down Approach*, 6th edition, Pearson, ISBN-10: 0132856204, 2012.

Section Outline

- Overview: multimedia on Internet
- Audio
  - Example: Skype
- Video
  - Example: Netflix
- Protocols
  - RTP, SIP
- Network support for multimedia

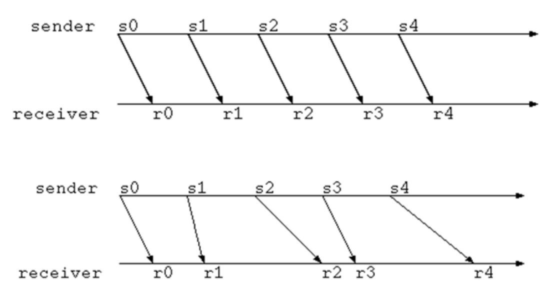
Internet Traffic

- Internet has many text-based applications
  - Email, File transfer, Web browsing
- Very sensitive to loss
  - Example: lose one byte in your `blah.exe` program and it crashes!
- Not very sensitive to delay
  - 10's of seconds ok for Web page download
  - Minutes ok for file transfer
  - Hours ok for email to delivery
- Multimedia traffic emerging (especially as fraction of bandwidth!)
  - Video already dominant on some links

Multimedia on the Internet

- Multimedia not as sensitive to *loss*
  - Words from speech lost still ok
  - Frames of video missing still ok
- Multimedia can be very sensitive to *delay*
  - Interactive session needs one-way delays less than ½ second!
- New phenomenon is effects of variation in delay, called *delay jitter* or just *jitter*!
  - Variation in bandwidth can also be important

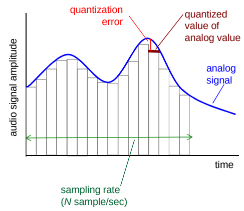
Jitter-Free



Jitter

Multimedia: Audio

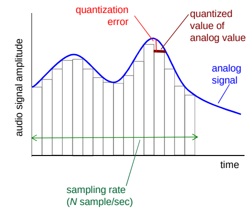
- Analog audio signal sampled at constant rate
  - phone: 8000 samples/sec
  - CD music: 44,100 samples/sec
- Each sample quantized (rounded)
  - e.g., 2^8=256 possible quantized values
  - each quantized value represented by bits, e.g., 8 bits for 256 values



Multimedia: Audio

- Example: 8000 samples/sec, 256 quantized values: 64,000 bps
- Receiver converts bits back to analog signal:
  - some quality reduction

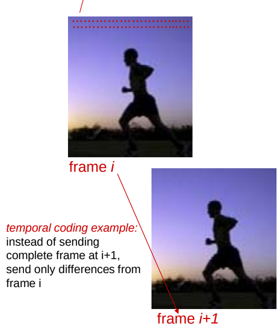
Example rates
- CD: 1.411 Mbps
- MP3: 96, 128, 160 Kbps
- Internet telephony: 5.3 Kbps and up



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)

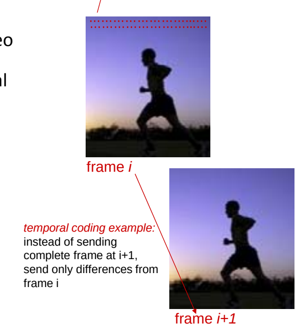
spatial coding example: instead of sending *N* values of same color (all purple), send only two values: color value (*purple*) and *number of repeated values* (N)



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 Mb/s
  - MPEG2 (DVD) 3-6 Mb/s
  - MPEG4 (often used in Internet, < 1 Mb/s)

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)



Some Types of Multimedia Activities over the Internet

- *Streaming, stored* audio, video
- *Conversational* voice (& video)
- *Streaming live* audio, video

Streaming Stored Media

- *Streaming, stored* audio, video
  - Pre-recorded
  - *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)
- 1-way communication, unicast
- Interactivity, includes pause, ff, rewind…
- Examples: pre-recorded songs, video-on-demand
  - e.g. *YouTube, Netflix, Hulu*
- Delays of 1 to 10 seconds or so tolerable
- Need reliable estimate of bandwidth
- Not very sensitive to jitter

Conversational Voice/Video

- *Conversational* voice/video
  - interactive nature of human-to-human conversation limits delay tolerance
- "Captured" from live camera, microphone
- 2-way (or more) communication
- e.g., *Skype, Facetime*
- Very sensitive to delay
  - < 150 ms one-way delay good
  - < 400 ms ok
  - > 400 ms bad
- Sensitive to jitter

Streaming Live Media

- *Streaming live* audio, video
  - *streaming:* can begin playout before downloading entire file
  - Not pre-recorded, so cannot send faster than rendered
- "Captured" from live camera, microphone
- May be 1-way communication, unicast but may be more
  - More potential for "flash crowd"
- Interactivity, includes pause, ff, rewind…
- Delays of 1 to 10 seconds or so tolerable
- Need reliable estimate of bandwidth
- Not very sensitive to jitter
- Basically, like stored but:
  - May be harder to optimize/scale (less time)
  - May be 2+ recipients (flash crowd)

Hurdles for Multimedia on the Internet

- IP is best-effort
  - No delivery guarantees
  - No bitrate guarantees
  - No timing guarantees
- So … how do we do it?
  - Not as well as we would like
  - This class is largely about techniques to make it better!

Groupwork: TCP or UDP?

- Above IP we have UDP and TCP as the de-facto transport protocols. Which to use?

Streaming, stored audio, video?

Conversational voice (& video)?

Streaming live audio, video?

TCP or UDP?

- TCP
  - \+ In order, reliable (no need to control loss)
  - \- Congestion control (hard to pick encoding level right)
- UDP
  - \- Unreliable (need to control loss)
  - \+ Bandwidth control (easier to control sending rate)

An Example: VoIP

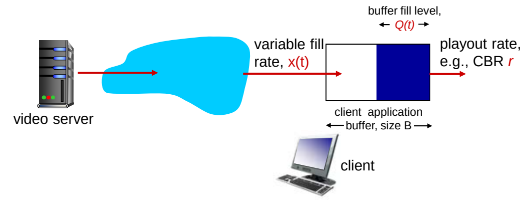
(Mini Outline)

- Specification
- Removing Jitter
- Recovering from Loss

VoIP: Specification

- 8000 bytes per second, send every 20 msec (*why every 20 msec*?)
  20 msec * 8000/sec = 160 bytes per packet
- Header per packet
  – Sequence number, time-stamp, playout delay
- End-to-end delay requirement of 150 – 400 ms
  – (So, why might TCP cause problems?)
- UDP
  – Can be delayed different amounts (need to remove jitter)
  – Can be lost (need to recover from loss)

Client-side Buffering, Playout



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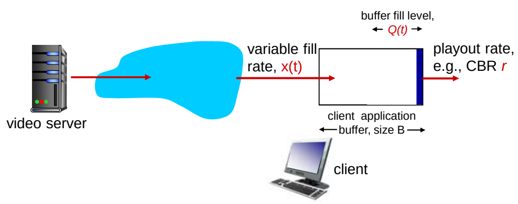
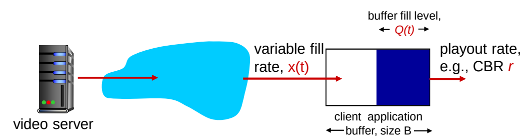


1. don't play immediately - initial fill of buffer t_0
2. playout begins at t_p,
3. buffer fill level varies over time as fill rate x(t) varies and playout rate r is constant

Client-side Buffering, Playout



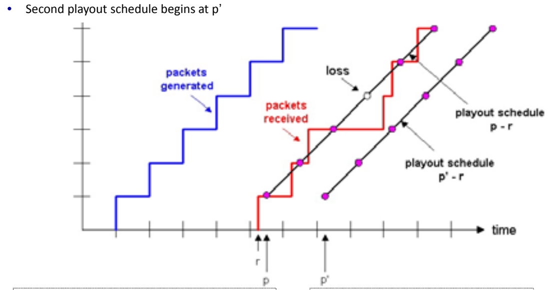
playout buffering: average fill rate ($\bar{x}$), playout rate (r):

- $\bar{x}$ < 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)
  - *tradeoff:* buffer starvation less likely with larger delay, but longer wait until user begins watching

VoIP: Playout Delay

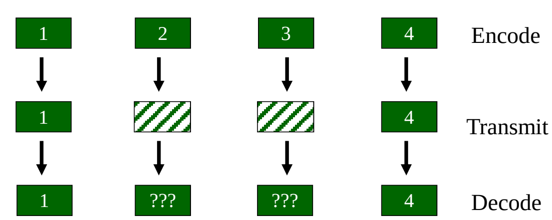
- Sender generates packets every 20 msec (during talk spurt)
- First packet received at time r
- First playout schedule begins at p
- Second playout schedule begins at p'

Two policies, ***wait p*** or ***wait p'***
- *p* has less delay, but one missed
- *p'* has no missed, but higher delay



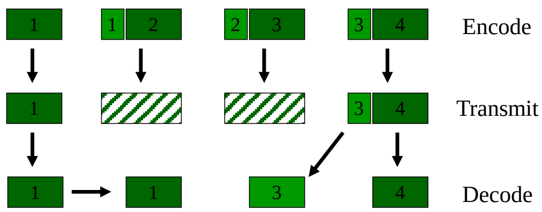
Playout delay can be *fixed* or *adaptive* → If adaptive, adapt each talkspurt

VoIP: Loss



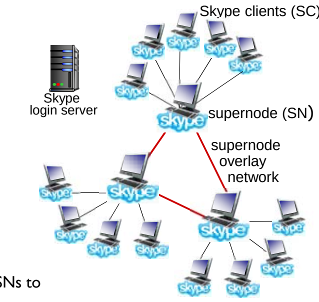
Q: What to do about missing packets?

VoIP: Recovering from Loss



Voice-over-IP: Skype

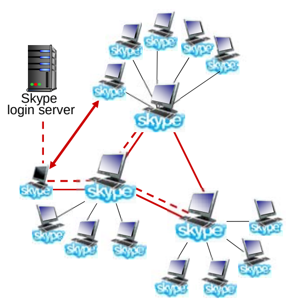
- Proprietary application-layer protocol (inferred via reverse engineering)
  - encrypted msgs
- P2P components:
  - clients (SC): Skype peers connect directly to each other for VoIP call
  - super nodes (SN): Skype peers with special functions
  - overlay network: among SNs to locate SCs
  - login server



P2P Voice-over-IP: Skype

Skype client operation:

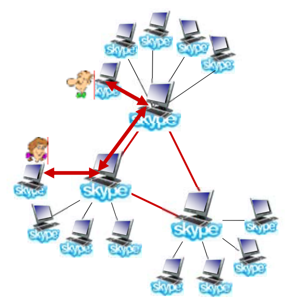
1. joins Skype network by contacting SN (IP address cached) using TCP
2. logs-in (usename, password) to centralized Skype login server
3. obtains IP address for callee from SN, SN overlay
   - or client buddy list
4. initiate call directly to callee



Q: when might this *not* work?

Skype: Peers as Relays

- *Problem:* both Alice, Bob behind “NATs”
  - NAT prevents outside peer from initiating connection to insider peer
  - inside peer *can* initiate connection to outside
- Relay solution: Alice, Bob maintain open connection to their SNs
  - Alice signals her SN to connect to Bob
  - Alice’s SN connects to Bob’s SN
  - Bob’s SN connects to Bob over open connection Bob initially initiated to his SN



Projects

- Project 1:
  - Read and Playback from audio device
  - Detect Speech and Silence
  - Evaluate (1a)
- Project 2:
  - Build a VoIP application
  - Evaluate (2b)
- Project 3:
  - Pick your own (video conf, thin game, repair …)

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  - RTP, SIP
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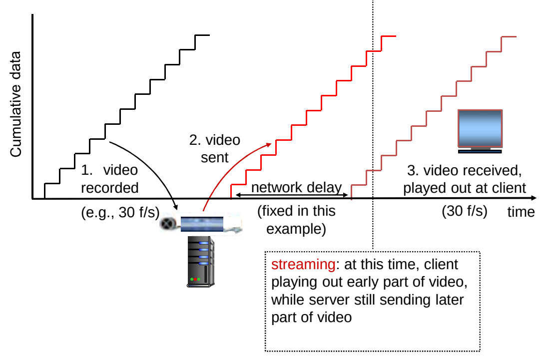
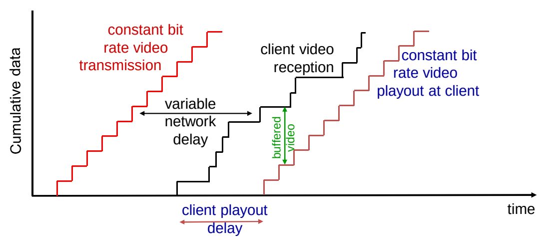
Streaming Stored Video



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

Streaming Stored Video: Revisted



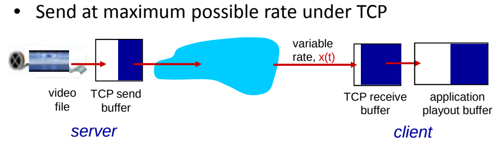
- *client-side buffering:* compensates for delay jitter and bitrate jitter

Streaming Multimedia: UDP

- Server sends at rate appropriate for client
  - Often: *send rate = encoding rate = constant rate*
  - Transmission rate can be oblivious to congestion levels!
- Short playout delay (2-5 seconds) to remove bandwidth (and delay) jitter
- Error recovery: application-level, time permitting
- RTP [RFC 2326]: multimedia payload types (later)
- UDP often *not* allowed through firewalls

Streaming Multimedia: HTTP

- Basis for many: *Apple*, *Microsoft Silverlight*, *Adobe*, *Netfilx*
- 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 to smooth out TCP delivery rate
- HTTP/TCP passes more easily through firewalls

Streaming Multimedia: DASH

- *DASH:* Dynamic, Adaptive Streaming over HTTP
  - Now a standard, basis for Netflix streaming
- 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
  - consulting 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

- *"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*

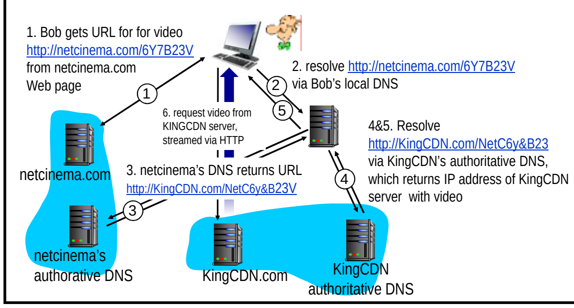
Content Distribution Networks

- *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 *(content distribution network, or 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 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



CDN Cluster Selection Strategy

- *challenge:* how does CDN DNS select "good" CDN node to stream to client
  - pick CDN node geographically closest to client
  - pick CDN node with shortest delay (or min # hops) to client (CDN nodes periodically ping access ISPs, reporting results to CDN DNS)
  - IP anycast – same addresses routed to one of many locations (routers pick, often shortest hop)

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

Case Study: Netflix

HTTP Adaptive Streaming in practice

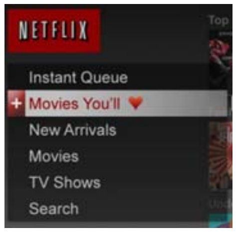
Mark Watson

(with thanks to the Netflix adaptive streaming team!)

ACM MMSys 2011 – 22-24 February 2011, San Jose, CA

Netflix Overview

- 20+ million subscribers in 2011 (15% of US households)
- 20% downstream US traffic at peak hours
- Bitrates up to 4.8 Mb/s
- Known for "recommendations"
- Many Netflix-ready devices (next slide)



Netflix Partner Products

Netflix Network Approach

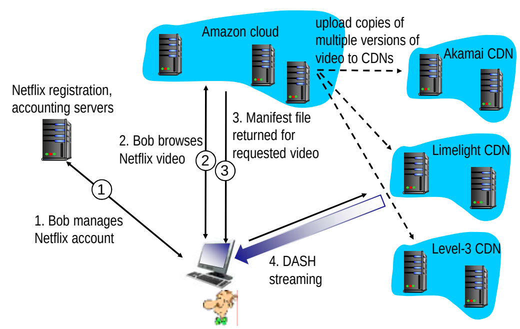
Client-centric

- Client has best view of network conditions
- No session state in network
  - Better scalability
- But, must rely upon client for operational metrics
  - Only client knows what happened, really

CDN

- Own little infrastructure, use 3rd parties
  - Own registration, payment servers
- Amazon cloud services:
  - Cloud hosts Netflix web pages for user browsing
  - Netflix uploads studio master to Amazon cloud
  - create multiple version of movie (different encodings) in cloud
  - upload versions from cloud to CDNs
- *Three* 3rd party CDNs host/stream Netflix content: *Akamai*, *Limelight*, *Level-3*

Netflix – Initiate Request

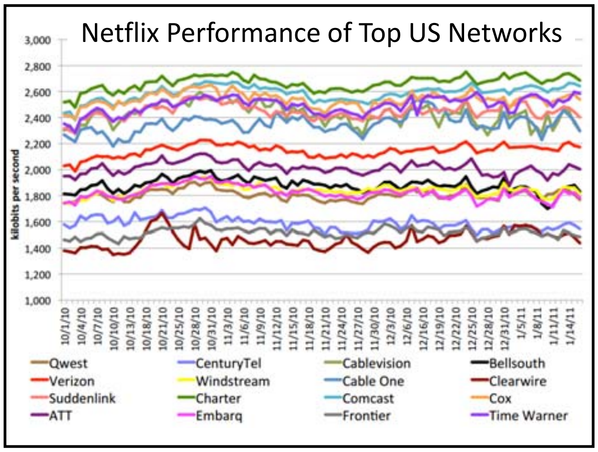


Netflix Importance of Client Metrics

- Metrics are essential
  - Detecting and debugging failures
  - Managing performance
  - Experimentation (new interfaces, features)
- Absence of server-side metrics places onus on client
- What is needed?
  - Reports of what user did (or didn't) see
    - Which part of which stream when
  - Reports of what happened in network
    - Requests sent, responses received, timing, throughput

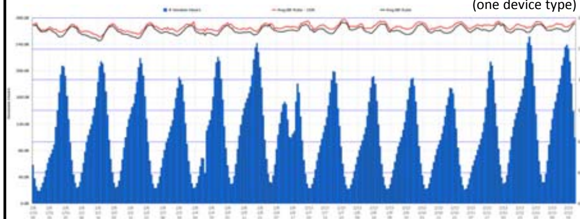
Netflix Quality

- Reliable transport (HTTP is over TCP)
- Quality characterized by
  - Video quality (how it looks)
    - At startup, average and variability (different layers)
  - Startup delay
    - Time form use action to first frame displayed
  - Rebuffer rate
    - Rebuffers per viewing hour, duration of rebuffer pauses



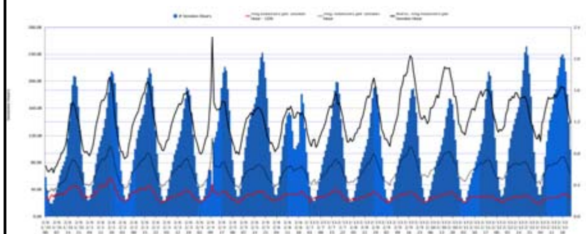
Netflix Streaming Bitrates

(one device type)



- Cyclic session hours (Q: *why*?)
- Average bitrate stays relatively flat (but not totally)

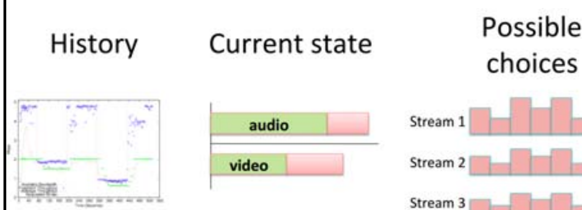
Netflix Rebuffer Rates



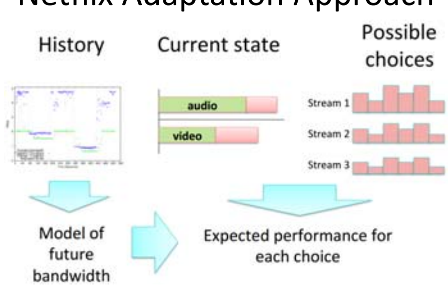
- Rebuffers at peaks for sessions (usually)
- Worst is about 1-2 per hour
- CDN performance is better

Netflix Adaptation Problem

- At client, pick sequence and timing of requests in order to:
  - minimize probability of rebuffering
  - maximize visual quality



Netflix Adaptation Approach



- Example:
  - Model future bandwidth: constant? avg over last 10s?
  - Analyze choices: construct "plan" for each choice, know visual quality, estimate rebuffering

NetFlix Future Work Needs

- Good models of future bandwidth (based on history)
  - Short term history
  - Long term history (across multiple sessions)
- Tractable representations of future choices
  - Including scalability, multiple streams
- Quality goals with "right" mix of visual quality and performance (rebuffering)
- Convolution of future bandwidth models with possible plans
  - Efficiently, maximizing quality goals

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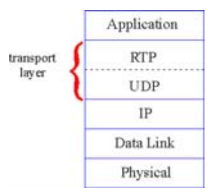
Real-Time Protocol (RTP) [RFC 3550]

- RTP specifies packet structure for packets carrying audio, video data
- RTP packet provides
  - payload type identification
  - packet sequence number
  - time stamp
- RTP runs in end systems, not routers
- RTP packets encapsulated in UDP segments
- Interoperability potential
  - e.g. if two VoIP applications run RTP, they *may* be able to work together

RTP Runs on Top of UDP

- RTP libraries provide transport-layer interface that extends UDP:
  - Port numbers, IP addresses
  - Payload type identification
  - Packet sequence numbers
  - Time stamps



RTP Example

Example: sending 64 kb/s PCM-encoded voice over RTP

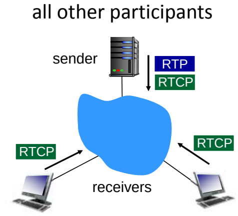
- application collects encoded data in chunks, e.g., every 20 msec = 160 bytes in chunk
- audio chunk + RTP header form RTP packet → encapsulated in UDP segment
- RTP header indicates type of audio encoding in each packet
  - sender can change encoding during conference
- RTP header also contains sequence numbers, timestamps

RTP and Quality of Service (QoS)

- RTP does *not* provide any mechanism to ensure timely data delivery or other QoS guarantees
- RTP encapsulation only seen at end systems (*not* by intermediate routers)
  - routers provide best-effort service, making no special effort to ensure that RTP packets arrive at destination in timely matter

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

SIP: Session Initiation Protocol [RFC 3261]

Long-term vision:

- All telephone calls, video conference calls take place over Internet
- People identified by names or e-mail addresses, rather than by phone numbers
- Can reach callee *(if callee so desires)*, no matter where callee roams, no matter what IP device callee is currently using

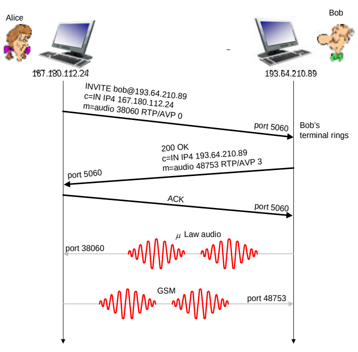
- SIP comes from IETF: borrows much of its concepts from HTTP
  - SIP has "Web flavor"
  - Alternative approaches (e.g. H.323) have "telephony flavor"
- SIP uses KISS principle: Keep It Simple Stupid

SIP Services

- SIP provides mechanisms for call setup:
  - for caller to let callee know s/he 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

Example: Setting Up Call to Known IP Address



- Alice's SIP invite message indicates her port number, IP address, encoding she prefers to receive (PCM µlaw)
- Bob's `200 OK` message indicates his port number, IP address, preferred encoding (GSM)
- SIP messages can be sent over TCP or UDP; here sent over RTP/UDP
- Default SIP port is 5060

Setting Up a Call (more)

- Codec negotiation:
  - suppose Bob doesn't have PCM µlaw encoder
  - Bob will instead reply with `606 Not Acceptable` reply, listing his encoders. Alice can then send new INVITE message, advertising different encoder

- Rejecting call
  - Bob can reject with replies "busy," "gone," "payment required," "forbidden"
- Media can be sent over RTP or some other protocol

SIP Name Translation, User location

- Caller wants to call callee, but only has callee's name or e-mail address.
- Need to get IP address of callee's current host:
  - user moves around
  - DHCP protocol
  - user has different IP devices (PC, smartphone, car device)

- Result can be based on:
  - time of day (work, home)
  - caller (don't want boss to call you at home)
  - status of callee (calls sent to voicemail when callee is already talking to someone)

SIP Registrar

- One function of SIP server: registrar
- When Bob starts SIP client, client sends SIP REGISTER message to Bob's registrar server

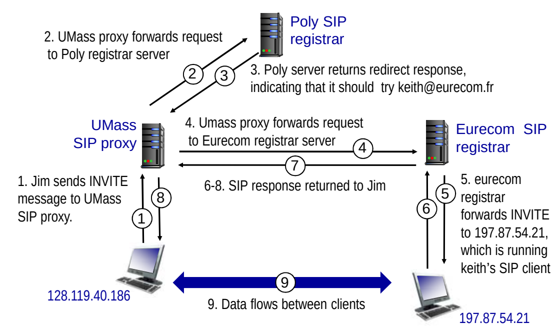
register message:

```
REGISTER sip:domain.com SIP/2.0
Via: SIP/2.0/UDP 193.64.210.89
From: sip:bob@domain.com
To: sip:bob@other-domain.com
Expires: 3600
```

SIP Proxy

- Another function of SIP server: *proxy*
- Alice sends invite message to her proxy server
  - contains address `sip:bob@domain.com`
  - proxy responsible for routing SIP messages to callee, possibly through multiple proxies
- Bob sends response back through same set of SIP proxies
- Proxy returns Bob's SIP response message to Alice
  - contains Bob's IP address
- SIP proxy analogous to local DNS server plus TCP setup

SIP Example: jim@umass.edu Calls keith@poly.edu



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Network Support for Multimedia

Approach	Granularity	Guarantee	Mechanisms	Complex	Deployed?
Making best of best effort service	All traffic treated equally	None or soft	No network support (all at application)	low	everywhere
Differentiated service	Traffic "class"	None of soft	Packet market, scheduling, policing.	med	some
Per-connection QoS	Per-connection flow	Soft or hard after flow admitted	Packet market, scheduling, policing, call admission	high	little to none

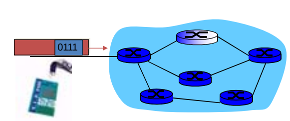
- Most of Internet is "best effort" and is focus of this class
- But there is some differentiated services
- And issues are useful for all

Capacity Planning in Best Effort Networks

- *Approach:* deploy enough link capacity so that congestion doesn't occur, multimedia traffic flows without delay or loss
  - low complexity of network mechanisms (use current "best effort" network)
  - high bandwidth costs
- Challenges:
  - *capacity planning:* how much bandwidth is "enough?"
  - *estimating network traffic demand:* needed to determine how much bandwidth is "enough" (for that much traffic)

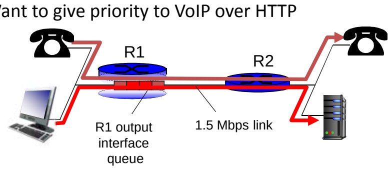
Providing Multiple Classes of Service

- Thus far: making the best of best effort service
  - "one-size fits all" service model
- Alternative: multiple classes of service
  - partition traffic into classes
  - network treats different classes of traffic differently (analogy: VIP service versus regular service)
- Granularity: differential service among multiple classes, not among individual connections



Scenario: Mixed HTTP and VoIP

- Example: 1 Mbps VoIP & HTTP share 1.5 Mb/s link.
  - HTTP bursts can congest router, cause VoIP loss
  - Want to give priority to VoIP over HTTP

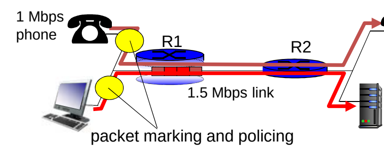


Principle 1

Packet marking needed for router to distinguish between different classes; and new router policy to treat packets accordingly

Principles for QOS Guarantees

- What if applications misbehave (VoIP sends higher than declared rate)?
  - policing: force source adherence to bandwidth allocations
- *Marking*, *policing* at network edge

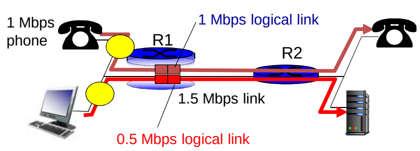


Principle 2

Provide protection (isolation) for one class from others

Principles for QOS Guarantees

- Allocating *fixed* (non-sharable) bandwidth to flow? *Inefficient* use of bandwidth if flows doesn't use its allocation



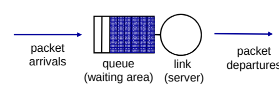
Principle 3

While providing isolation, it is desirable to use resources as efficiently as possible

Scheduling and Policing Mechanisms

- *Scheduling:* choose next packet to send on link
- *FIFO (first in first out) scheduling:* send in order of arrival to queue
  - real-world example?
  - *discard policy:* if packet arrives to full queue: who to discard?
    - *tail drop:* drop arriving packet
    - *priority:* drop/remove on priority basis
    - *random:* drop/remove randomly

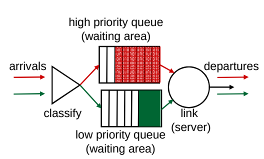
Q: other policies?

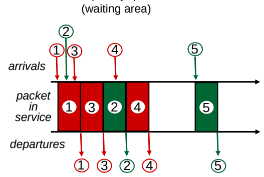


Scheduling Policies: Priority

Priority scheduling: send highest priority queued packet

- Multiple *classes*, with different priorities
  - class may depend on marking or other header info, e.g. IP source/dest, port numbers, etc.
  - real world example?

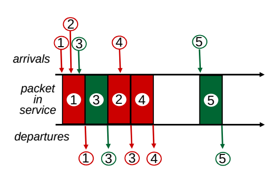




Scheduling Policies: Still More

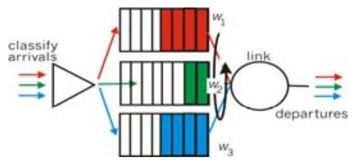
Round Robin (RR) scheduling:

- multiple classes
- cyclically scan class queues, sending one complete packet from each class (if available)
- real world example?



Scheduling Policies: Still More

Weighted Fair Queuing (WFQ):
- Generalized Round Robin
- Each class gets weighted amount of service in each cycle
- real-world example?



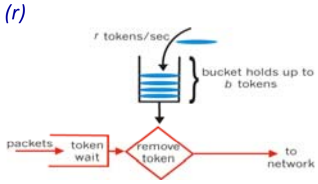
Policing Mechanisms

Goal: limit traffic to not exceed declared parameters

Three commonly-used criteria:
- *(long term) average rate:* how many packets can be sent per unit time (in long run)
  - crucial question: what is the interval length: 100 packets per sec or 6000 packets per min have same average!
- *peak rate:* e.g., 600 pkts per min (ppm) avg.; 1500 ppm peak rate
- *(max) burst size:* max number of pkts sent consecutively (with no intervening idle)

Policing Mechanisms: Implementation

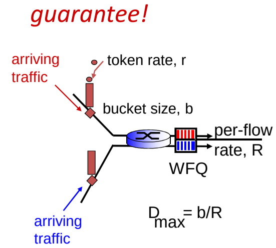
token bucket: limit input to specified *burst size (b)* and *average rate (r)*

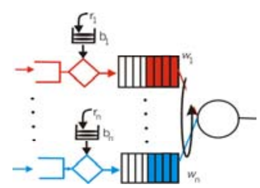


- Bucket can hold *b* tokens
- Tokens generated at rate *r token/sec* unless bucket full
- *Over interval of length t: number of packets admitted less than or equal to (r t + b)*

Policing and QoS Guarantees

- Token bucket, WFQ combine to provide guaranteed upper bound on delay, i.e., *QoS guarantee!*





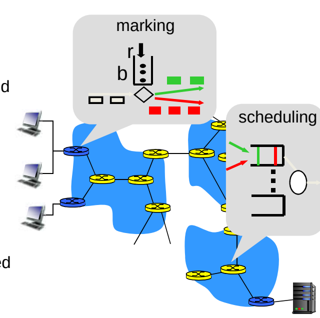
Differentiated Services (DiffServ)

- Want "qualitative" service classes
  - "behaves like a wire"
  - Relative service distinction: **Platinum**, **Gold**, **Silver**
- *Scalability:* simple functions in network core, relatively complex functions at edge routers (or hosts)
  - signaling, maintaining per-flow router state difficult with large number of flows
- Don't define service classes, provide functional components to build service classes

DiffServ Architecture

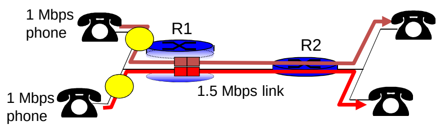
Edge router:
- per-flow traffic management
- marks packets as in-profile and out-profile

Core router:
- per class traffic management
- buffering and scheduling based on marking at edge
- preference given to in-profile packets over out-of-profile packets



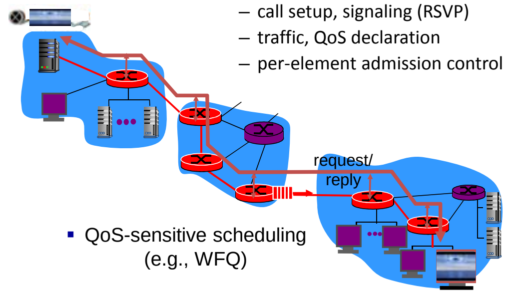
Per-connection QoS Guarantees

- *basic fact:* cannot support traffic demands beyond link capacity



Principle 4
call admission: flow declares its needs, network may block call (e.g., busy signal) if it cannot meet needs

QoS Guarantee Scenario



- *Resource reservation*
  - call setup, signaling (RSVP)
  - traffic, QoS declaration
  - per-element admission control
- QoS-sensitive scheduling (e.g., WFQ)

Introduction Outline

- Foundation (done)
  - Internetworking Multimedia (Ch 4) (done)
  - Perceptual Coding: MP3 Compression (done)
  - Graphics and Video (Linux MM, Ch 4) (done)
  - Multimedia Networking (Kurose, Ch 7) (done)
- Audio Voice Detection (Rabiner) (done)
- Video Compression (next)
  - (Next slide deck)