Multimedia Networking

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Audio/Video recordings of this lecture are available on-line at:

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- Multimedia Networking Applications
- Skype
- Real-Time Transport Protocol (RTP)
- Session Initiation Protocol (SIP)

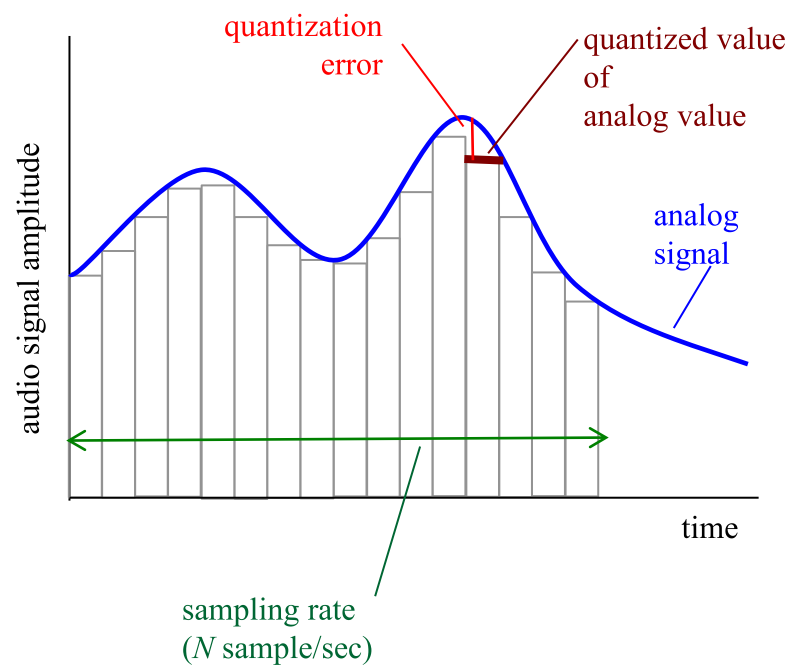
Note: This class lecture is based on Chapter 9 of the textbook (Kurose and Ross) and the figures provided by the authors.

Multimedia Applications

1. Audio Digitization
2. Playout Buffers
3. Streaming Using UDP
4. Streaming Using HTTP

Audio Digitization

- **Sampling**: Analog audio signal sampled at constant rate
  - Telephone: 8,000 samples/sec
  - CD music: 44,100 samples/sec
- **Quantization**: Each sample
  - 8 bits: 2^8=256 values
  - 16 bit: 2^{16} values
- 8 k samples/s each 8 bit ⇒ 64 kbps

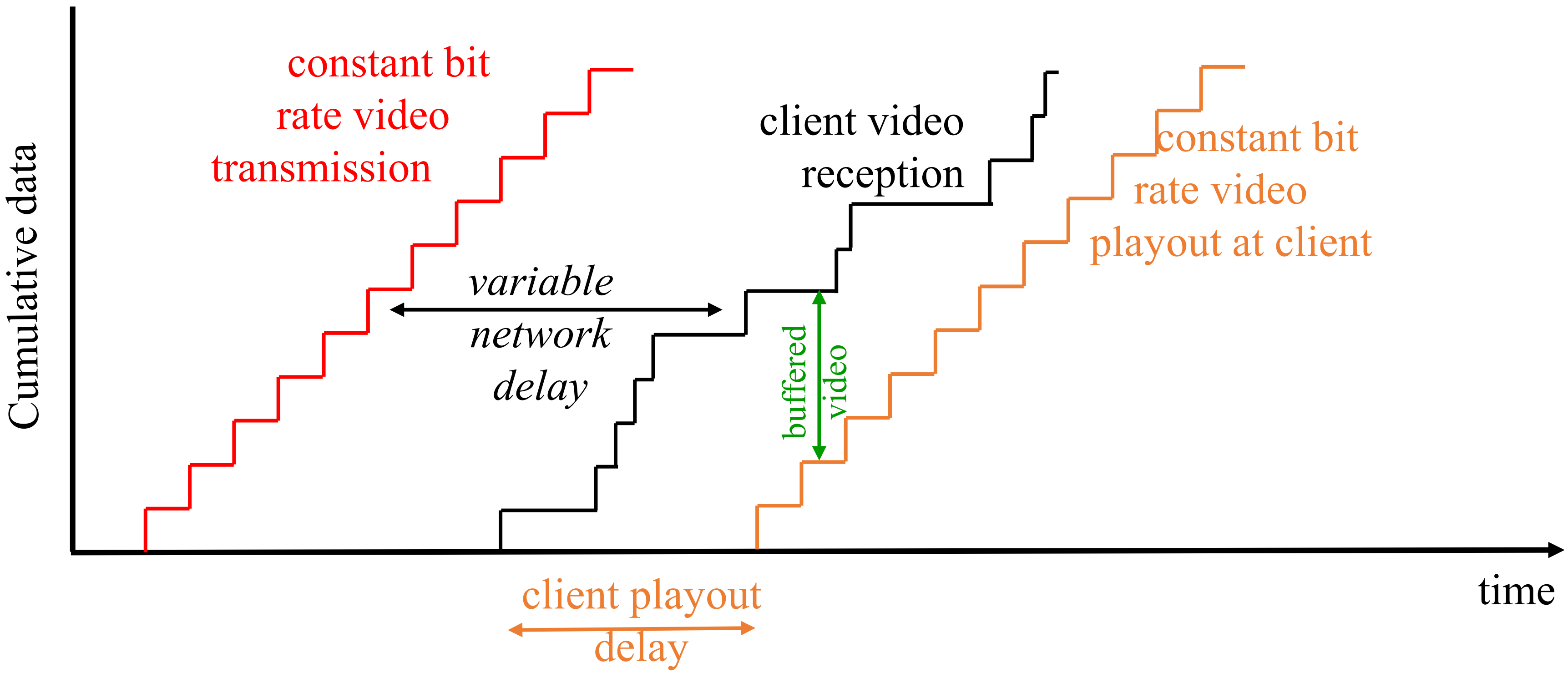


- **Compression**: Compress to 5-16 kbps, e.g., using differences
- Lower bits/sample, samples/sec, or higher compression ⇒ Lower Quality

Multimedia Networking Applications

- **Streaming Stored Multimedia**
  - Stored Media: Fast rewind, pause, fast forward
  - Streaming: simultaneous play out and download
  - Continuous play out: Delay jitter smoothed by playout buffer
- **Streaming Live Multimedia**: IPTV and Internet Radio
  - No fast-forward
  - High data rate to large number of users
    $\Rightarrow$ multicast or P2P,
  - Delay jitter controlled by caching,
- **Real-Time Interactive Multimedia**: Internet Telephone, Video Conferencing
  - Delay<400 ms.

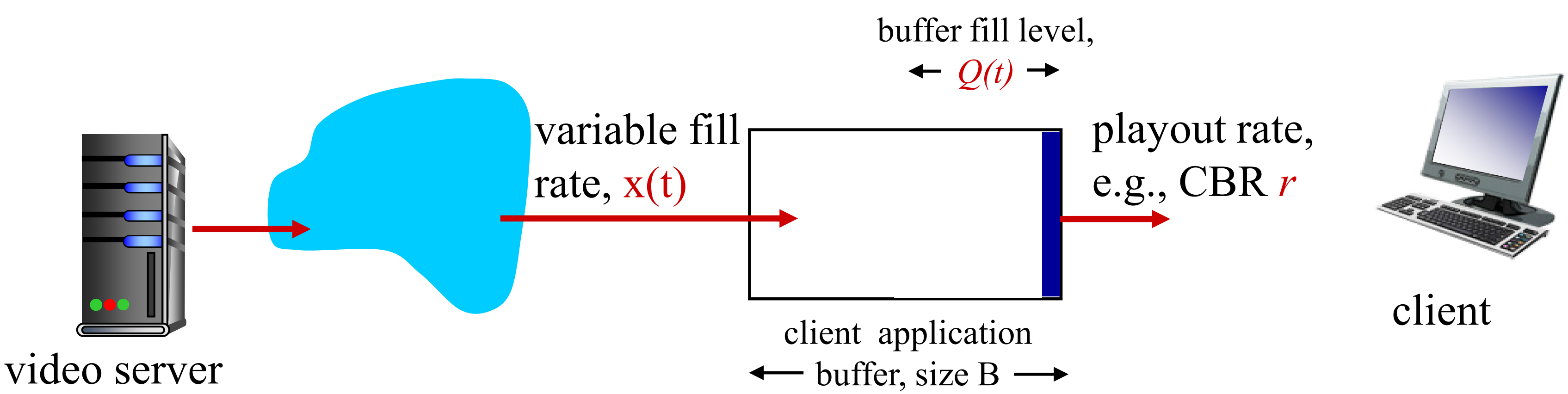
Playout Buffers



- Playout delay compensates for network delay, delay jitter
- Delay > Playout Delay ⇒ Packet late ⇒ Same as a lost packet

Client-side Buffering

1. Initial fill of buffer until playout begins at t_p
2. Fill rate $x(t)$ varies and playout rate r is constant
3. $x < r$: Buffer eventually empties causing freezing of video
4. $x > r$: buffer will not empty, Flow control to avoid overflow
5. *Tradeoff: Large initial playout delay* ⇒ *B*uffer starvation less likely but larger delay until user begins watching

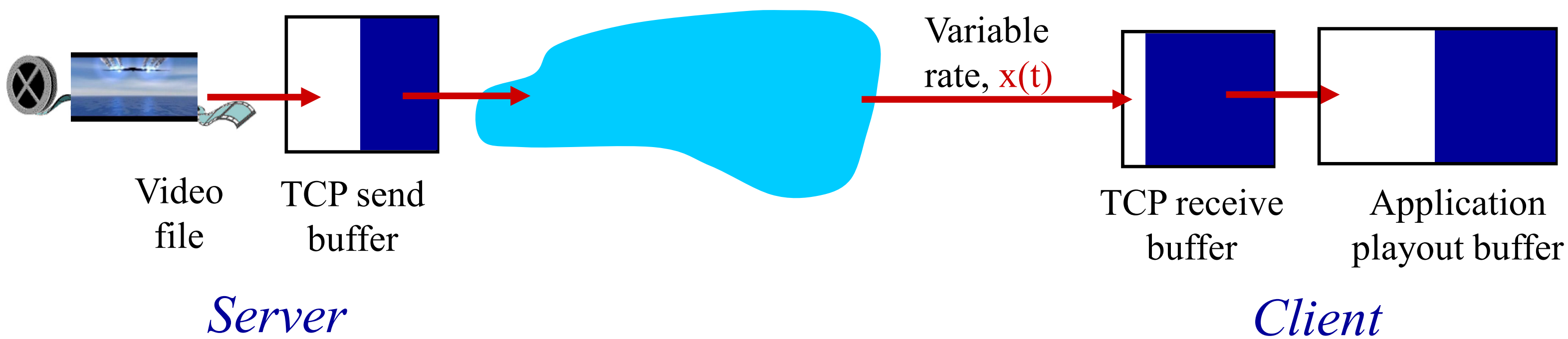


Streaming Using UDP

- Server sends at rate appropriate for client
  - Often: Send rate = Encoding rate = Constant
  - Transmission rate can be oblivious to congestion levels
- Short playout delay (2-5 seconds) to remove network jitter
- Application level error recovery
- UDP may *not* go through firewalls

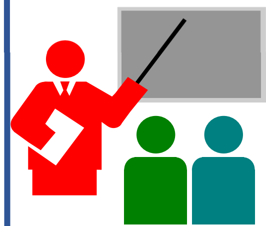
Streaming Using HTTP

- 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 to smooth TCP delivery rate
- HTTP/TCP passes more easily through firewalls

Review



Multimedia Applications

1. Audio is sampled, digitized, and compressed
2. Initial playout delay helps overcome the jitter in delay
3. UDP results in lower jitter but may not go through firewall
4. HTTP uses TCP and so the delay variation can be large

Ref: Section 9.1 and 9.2, Review Questions R1-R8, Problems P1-P5

Voice Over IP

1. VoIP Packet Losses
2. VoIP with Fixed Playout Delay
3. VoIP with Adaptive Playout Delay
4. Recovering From Packet Loss
5. Skype

Voice-over-IP (VoIP)

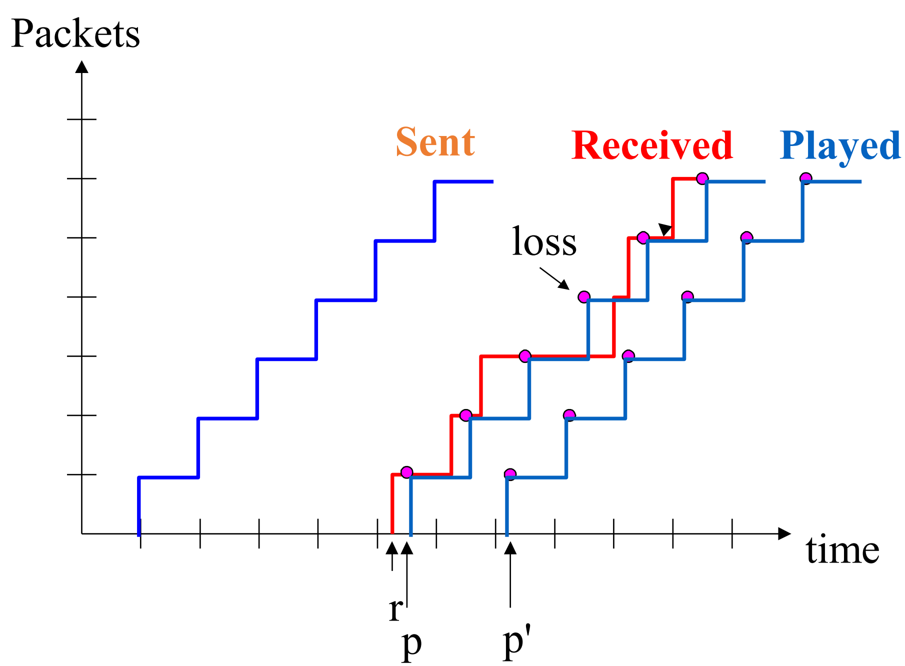
- **End-end-delay Requirement**: needed to maintain "conversational" aspect
  - Higher delays noticeable, impair interactivity
  - < 150 ms: good
  - > 400 ms: bad
  - Includes application-level (packetization, playout), network delays
- Alternating talk spurts, silent periods.
  - 64 kbps during talk spurt
  - Packets generated only during talk spurts
  - 20 ms chunks at 8 kbytes/sec: 160 bytes of data
- Application sends a segment every 20 ms during talk spurt

VoIP Packet Losses

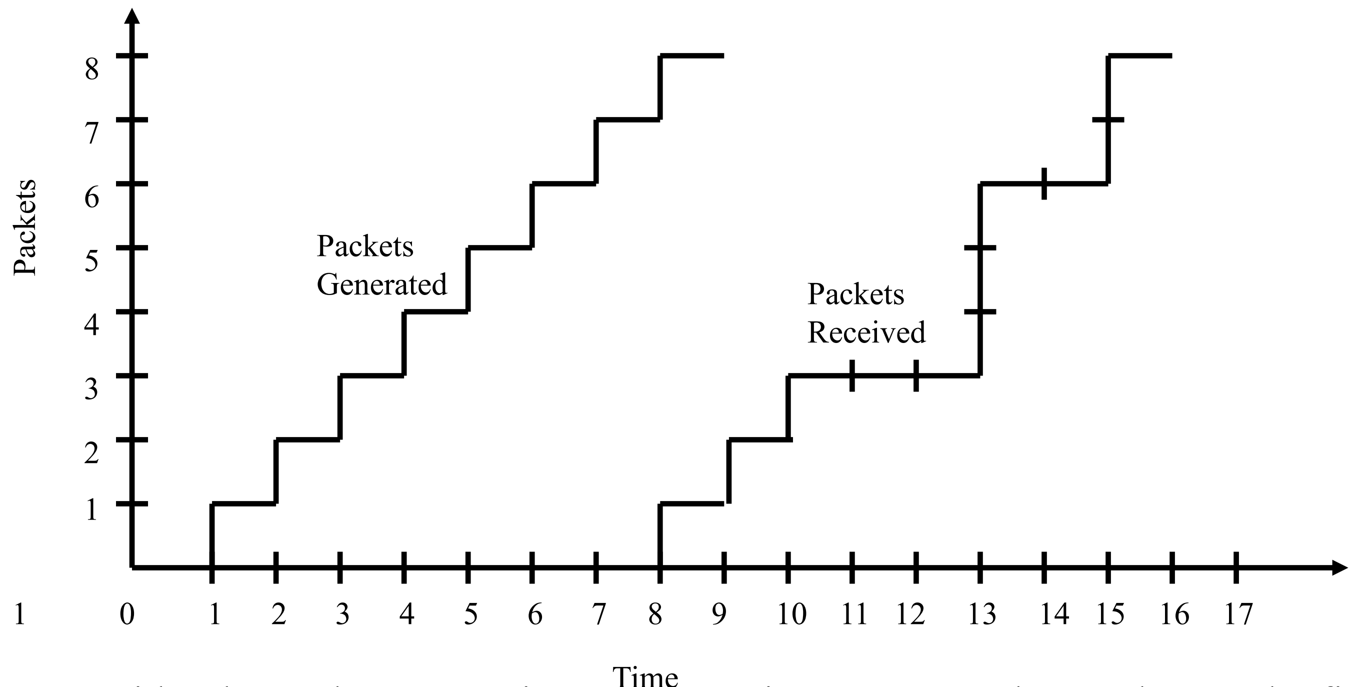
- **Network Loss**: IP datagram lost due to network congestion (router buffer overflow)
- **Delay Loss:** IP datagram arrives too late for playout
  - typical maximum tolerable delay: 400 ms
- **Loss Tolerance:** *P*acket loss rates up to 10% can be concealed

VoIP with Fixed Playout Delay

- Example: Packets sent every 20 ms during talk spurt.
- First packet received at time r
- If playout begins at p, 4th packet will arrive too late
- If playout begins at p', all packets can be played on time



Homework 9: Playout Delay



- [4 points] Consider the packet generation and reception sequence shown above. The first packet is generated at t=1 and is received at t=8.
- A. If Playout delay is zero and playout begins at t=8, which of the packets will not arrive in time? Show in a table.
- B. What is the minimum playout delay at the receiver that result in all of the first eight packets arriving in time for their playout?

Adaptive Playout Delay

- Estimate network delay, adjust playout delay at beginning of each talk spurt
- Silent periods compressed and elongated
- Chunks still played out every 20 ms during talk spurt
- Adaptively estimate packet delay: Similar to TCP RTT estimate

Adaptive Playout Delay

- t_i =Sending time
- r_i= Receiving time
- Measured delay sample = r_i-t_i
- d_i= Average network delay

$$d_i = (1-a)d_{i-1}+a(r_i-t_i)$$

- v_i= Variation of the delay

$$v_i = (1-a)v_{i-1}+a|r_i-t_i-d_i|$$

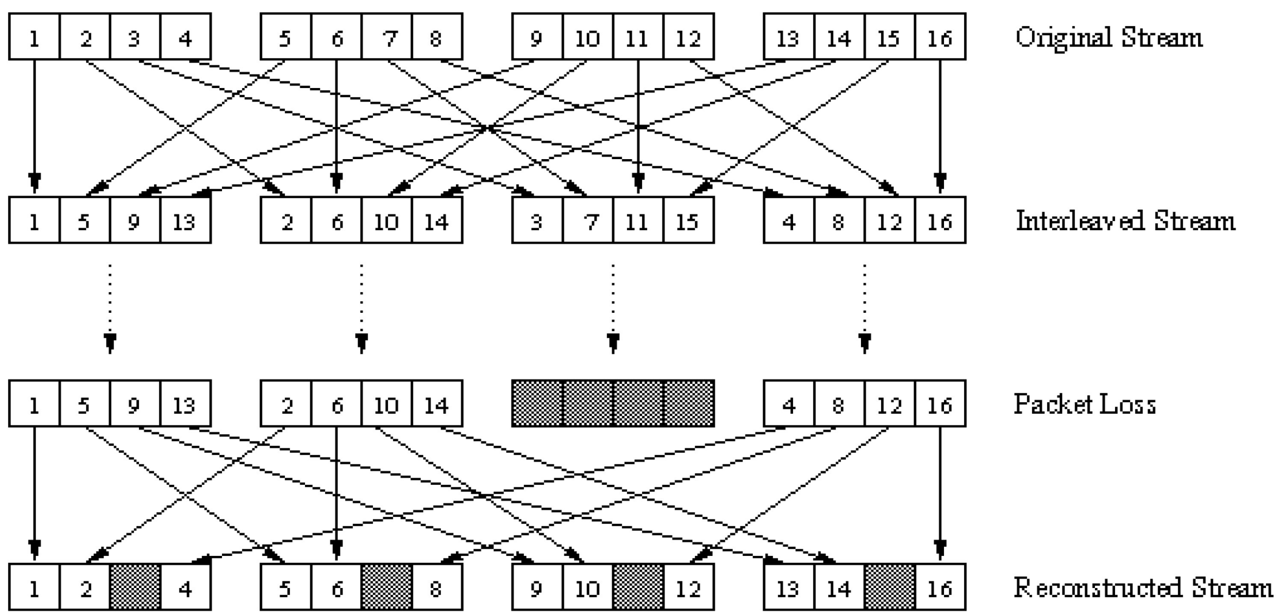
- p_i= Playout time

$$p_i = t_i + d_i + Kv_i$$

- Here K is a constant, say 4.
- Sequence numbers and timestamps used to determine talk spurts and silence

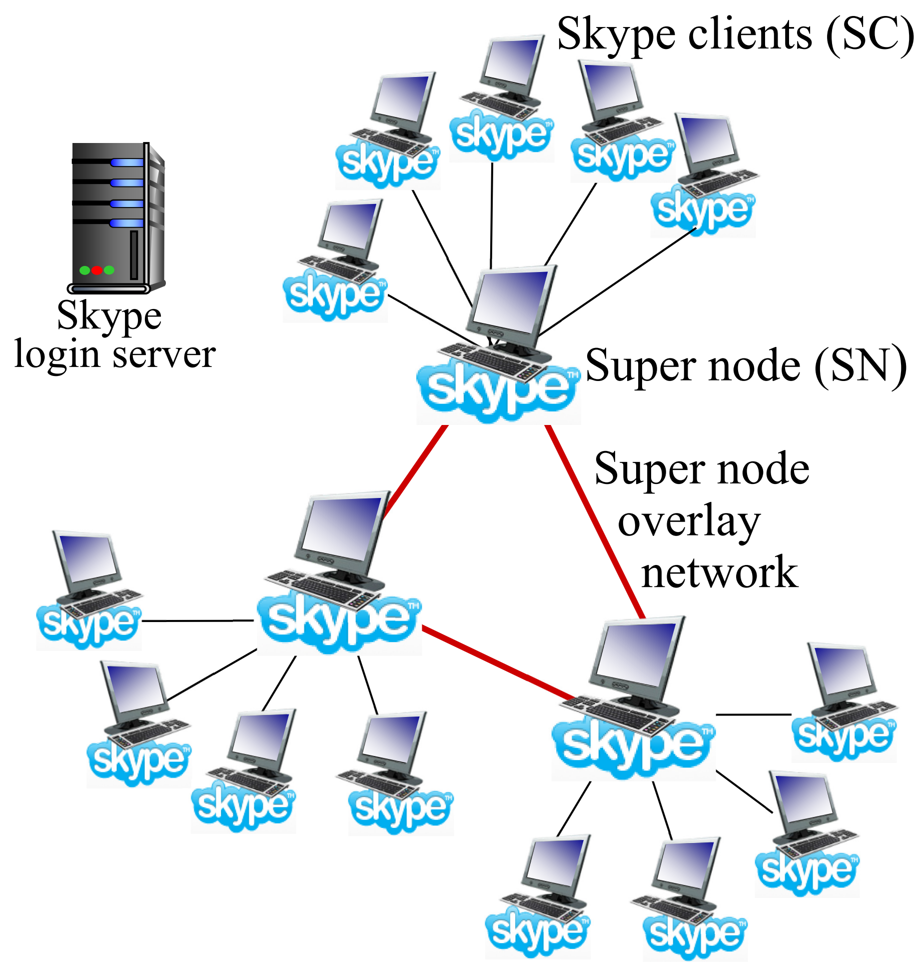
Recovering From Packet Loss

- **Forward Error Correction**
- Send n+1 packets in place of n packets
- Send a lower-resolution stream in addition
- Play out the old syllable
- Busty Loss ⇒ **Interleave** audio/video frames



Voice-over-IP: Skype

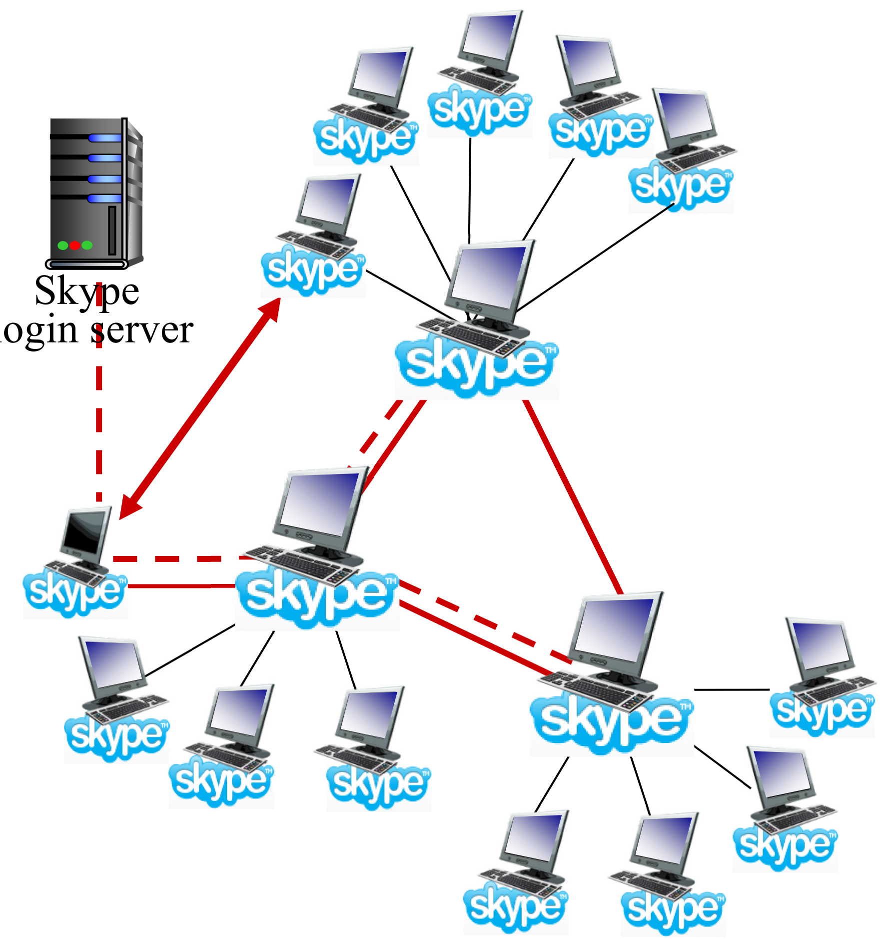
- **Proprietary** application-layer protocol (inferred via reverse engineering)
- **Encrypted** messages
- **P2P**: Media does not go through a central server
- **Clients**: Skype peers connect directly to each other for VoIP call
- **Super Nodes** (SN): Skype peers with special functions
- **Overlay Network**: Among SNs to locate clients
- **Login server**



P2P voice-over-IP: Skype

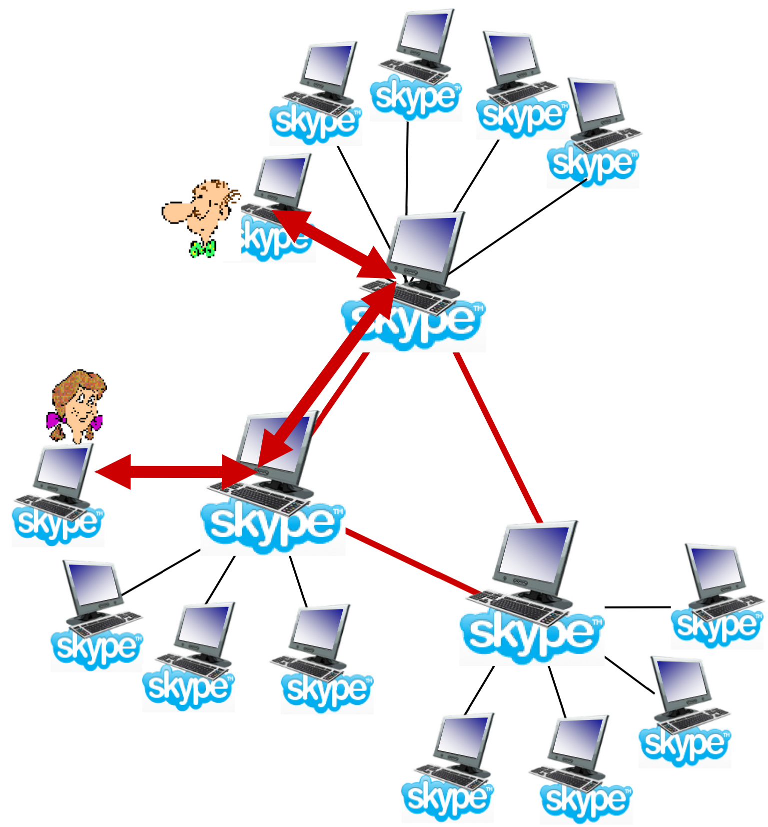
Skype client operation:

1. Joins Skype network by contacting SN (IP address cached) using TCP
2. Logs-in: Username, password to centralized Skype login server
3. Obtains IP address for callee from SN, SN overlay
4. Initiate call directly to callee via SN

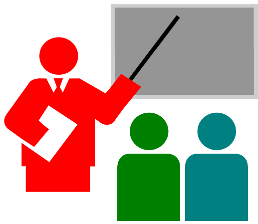


Skype: Super Nodes as Relays

- **Problem**: both Alice, Bob are behind "NATs"
    - NAT prevents outside peer from initiating connection to inside 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 initiated



Review



Voice over IP

1. Talk spurts are delayed to overcome jitter
2. Playout delay is estimated adaptively using mean and standard deviation
3. Forward error correction and interleaving is used to overcome losses and burst errors
4. Skype uses super nodes to help connect peers. A login server is used for authentication.
5. Skype nodes maintain an outgoing connection with the super nodes. These connections are used for incoming VoIP packets.

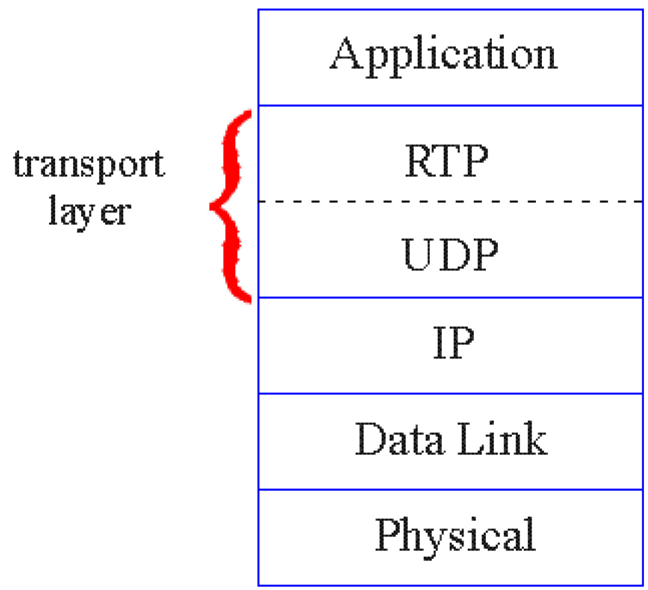
Ref: Section 9.3, Review Question R9-R11, Problems P6-P14

Protocols for Real-Time Applications

1. Real-Time Transport Protocol (RTP)
2. Session Initiation Protocol (SIP)
3. H.323 Protocols

Real-Time Transport Protocol (RTP)

- Common sublayer between applications and UDP
- Provides sequence numbers, timestamps, and other facilities
- Supports both unicast and multicast



RTP Packet Format

Marker	Payload Type	Sequence Number	Time-stamp	Synchronization Source Identifier	Miscellaneous Fields
1b	7b	16b	32b	32b	

- Marker indicates that the packet contains special data required by some applications
- SSRC = Synchronization Source = Stream #

Payload Type	Coding	Rate
0	PCM mu-law	64 kbps
3	GSM	13 kbps
7	LPC	2.4 kbps
26	Motion JPEG	
31	H.261	
33	MPEG2 video	

Session Initiation Protocol (SIP)

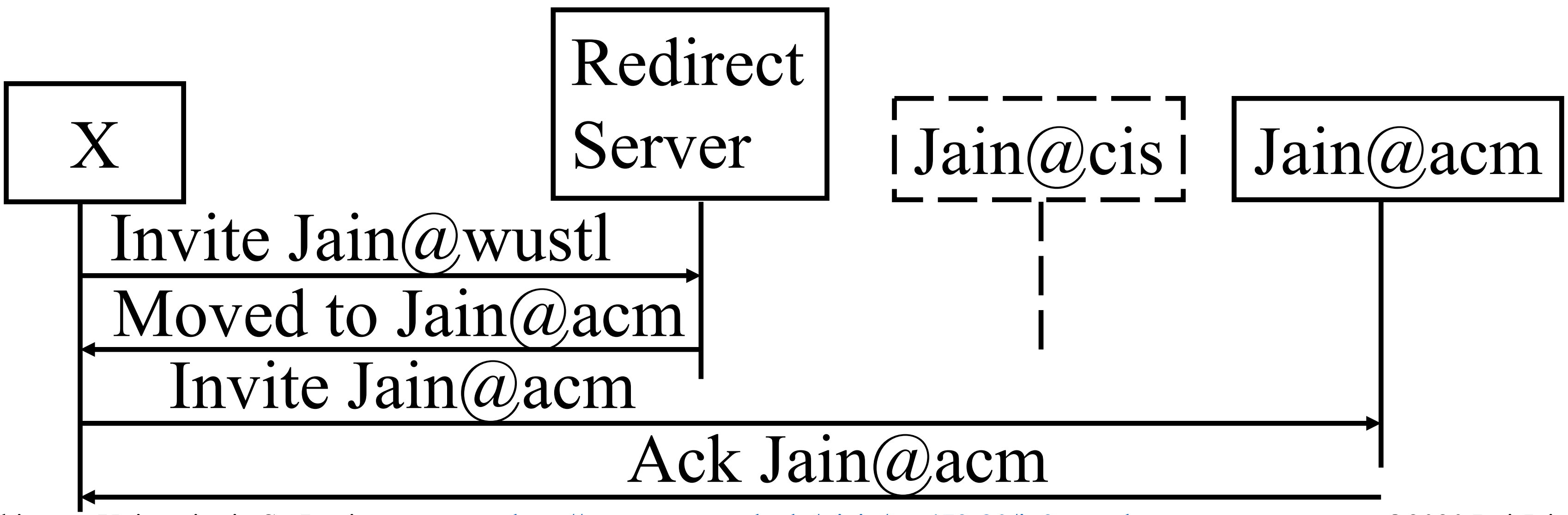
- Application level signaling protocol for voice and video conferencing over Internet
- Allows creating, modifying, terminating sessions with one or more participants
- Carries session descriptions (media types) for user capabilities negotiation
- Supports user location, call setup, call transfers
- Supports mobility by proxying and redirection

SIP (Cont)

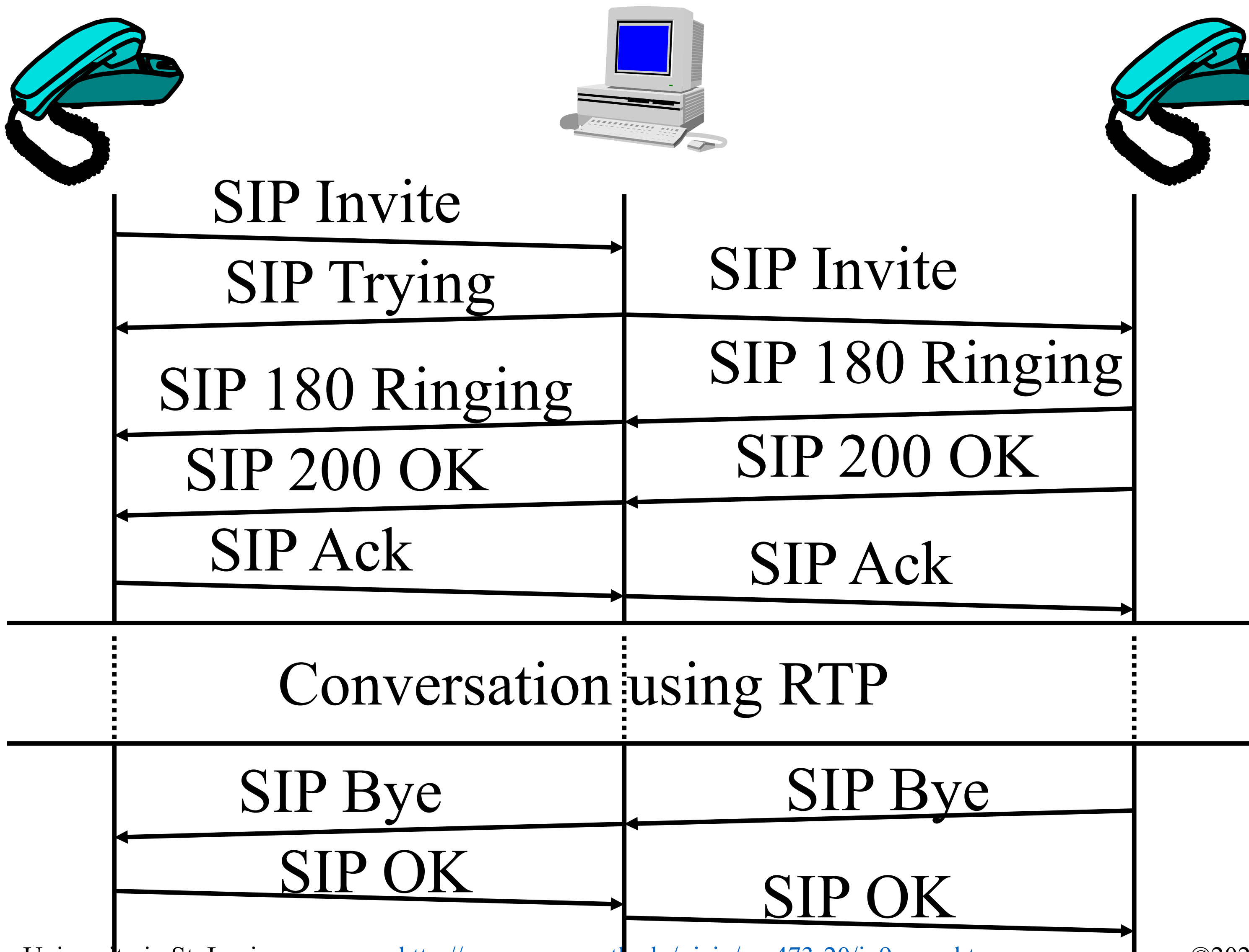
- SIP Uniform Resource Identifiers (URIs):
  Similar to email URLs
  sip:jain@wustl.edu
  sip:+1-614-292-3989:123@wustl.edu?subject=lecture
- SIP can use UDP or TCP
- SIP messages are sent to SIP servers:
  - Registrar: Clients register and tell their location to it
  - Location: Given name, returns possible addresses for a user. Like Directory service or DNS.
  - Redirect: Returns current address to requesters
  - Proxy: Intermediary. Acts like a server to internal client and like a client to external server

Locating using SIP

- Allows locating a callee at different locations
- Callee registers different locations with Registrar
- SIP Messages: Ack, Bye, Invite, Register, Redirection, ...



SIP Proxy

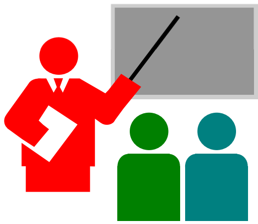

SIP Invite
SIP Trying
SIP Invite
SIP 180 Ringing
SIP 180 Ringing
SIP 200 OK
SIP 200 OK
SIP Ack
SIP Ack
Conversation using RTP
SIP Bye
SIP Bye
SIP OK
SIP OK

H.323 Protocols

- Multimedia over LANs, V1 (June 96), V2(Feb 98)
- Provides component descriptions, signaling procedures, call control, system control, audio/video codecs, data protocols

<table>
<tr><th>Video</th><th>Audio</th><th colspan="4">Control and Management</th><th>Data</th></tr>
<tr><td>H.261
H.263</td><td>G.711, G.722,
G.723.1, G.728,
G.729</td><td rowspan="2">RTCP</td><td>H.225.0
RAS</td><td>H.225.0
Signaling</td><td>H.245
Control</td><td>T.124</td></tr>
<tr><td colspan="2">RTP</td><td colspan="3">X.224 Class 0</td><td>T.125</td></tr>
<tr><td colspan="3">UDP</td><td colspan="3">TCP</td><td rowspan="2">T.123</td></tr>
<tr><td colspan="6">Network (IP)</td></tr>
<tr><td colspan="7">Datalink (IEEE 802.3)</td></tr>
</table>

Review



Protocols for Real-Time Applications

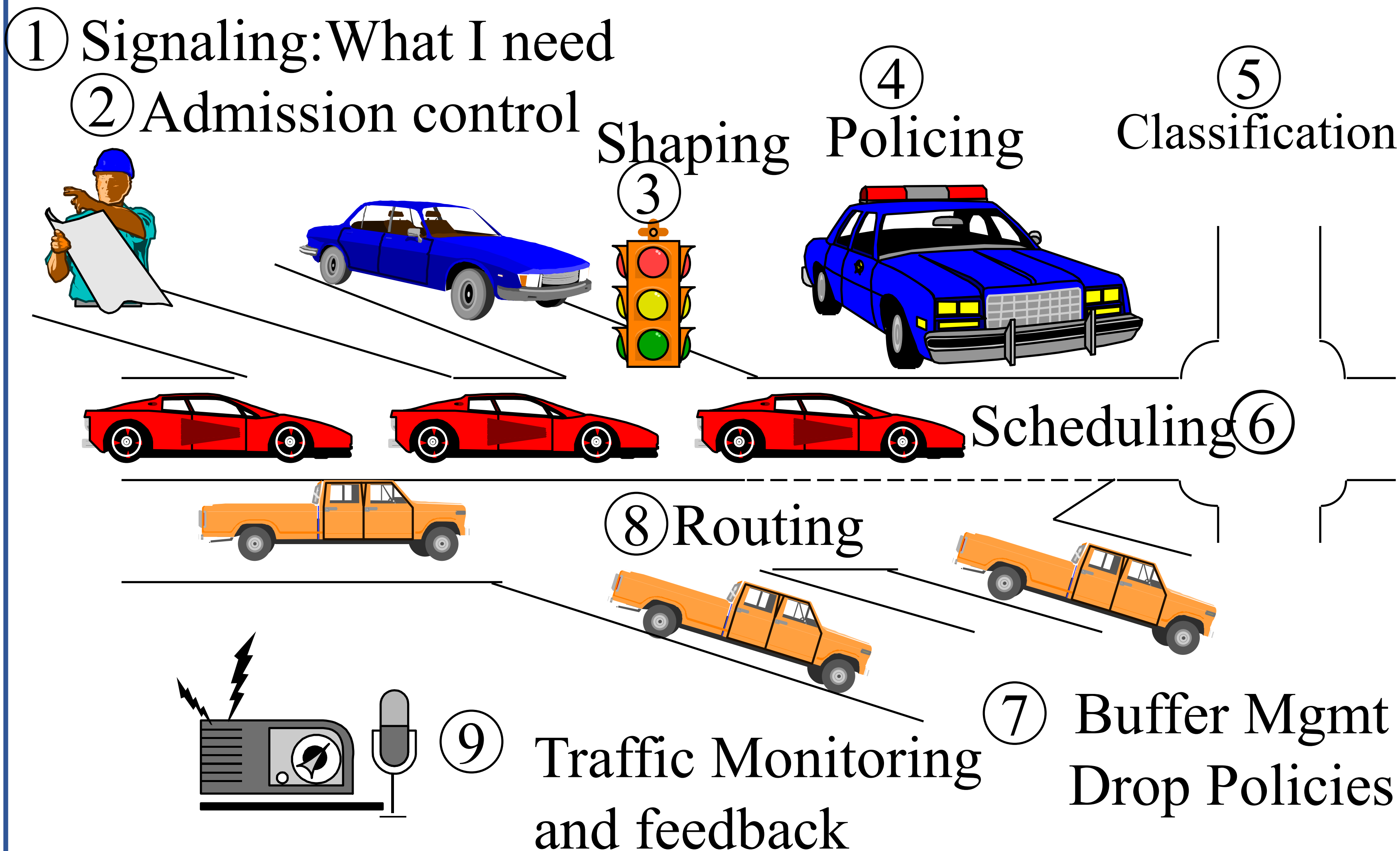
1. RTP is used to transmit multimedia over UDP
2. SIP is a signaling (control) protocol to establish multimedia connections
3. H.323 is a framework for a group of protocols used for multimedia

Ref: Section 9.4, Review questions R12-R13, Problems P15-P16

Networking Support for Multimedia

1. QoS Components
2. Traffic Shaping
3. Token Bucket Shaper
4. Traffic Policing
5. Differentiated Services

QoS Components

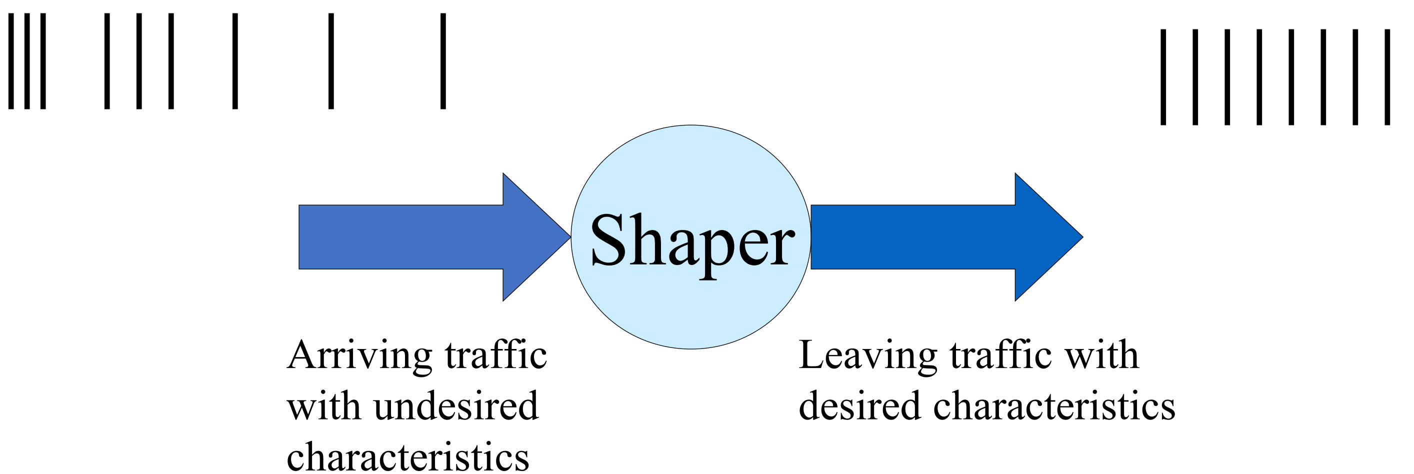


QoS Components (Cont)

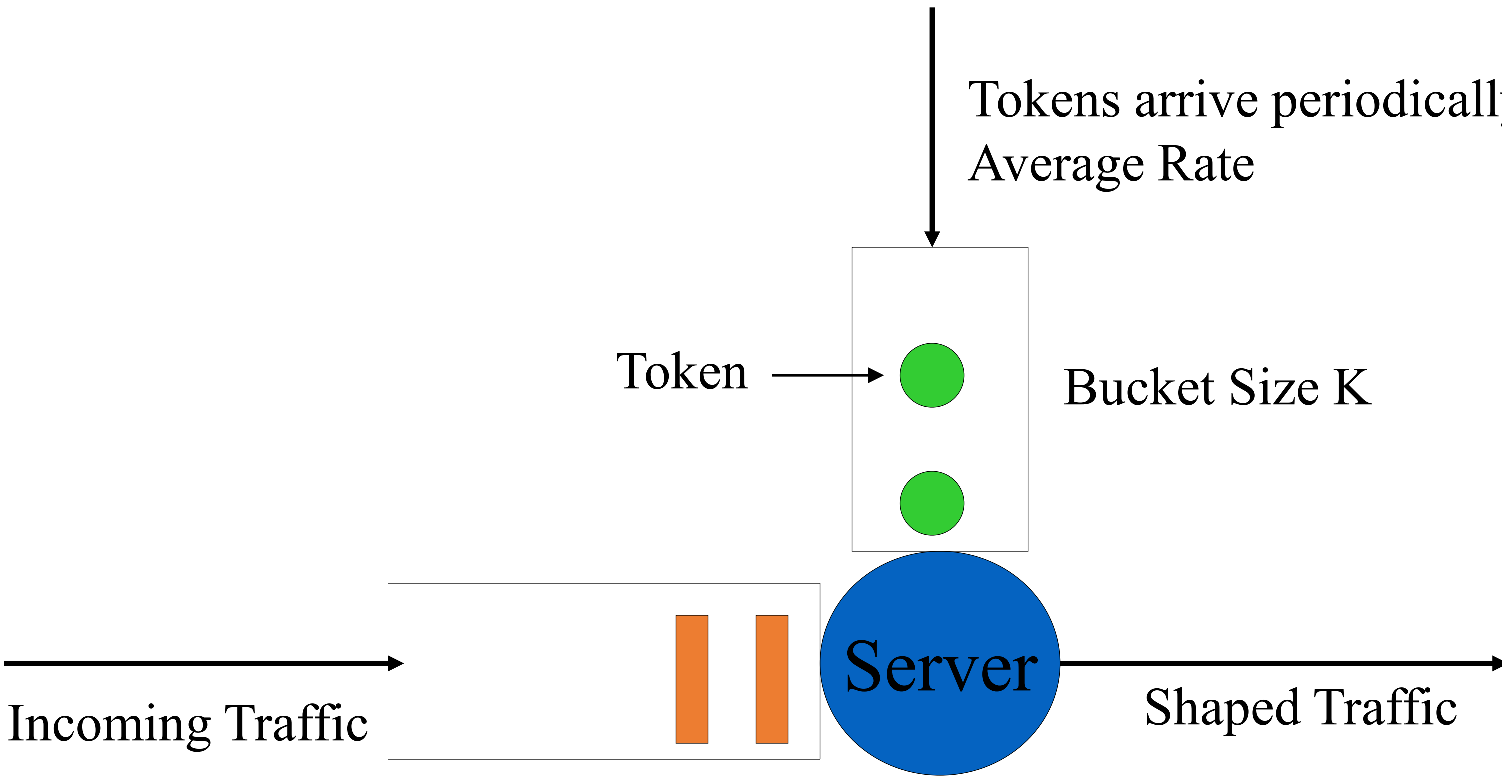
1. **Signaling**: Users need to tell/negotiate their QoS requirements with the network
2. **Admission Control**: Network can deny requests that it can not meet
3. **Shaping**: Traffic is smoothed out so that it is easier to handle
4. **Policing**: Ensuring that the users are sending at the rate they agreed to
5. **Marking/Classification**: Packets are classified based on the source, destination, TCP ports (application)
6. **Scheduling**: Different flows get appropriate treatment
7. **Drop Policies**: Low priority packets are dropped.
8. **Routing**: Packets are sent over paths that can meet the QoS
9. **Traffic Management**: Sources may be asked to reduce their rates to meet the loss rate and delay guarantees

Traffic Shaping

- Altering the traffic characteristics of a given flow is called traffic shaping
- The source must shape its traffic prior to sending it to network so it does not violate traffic contract

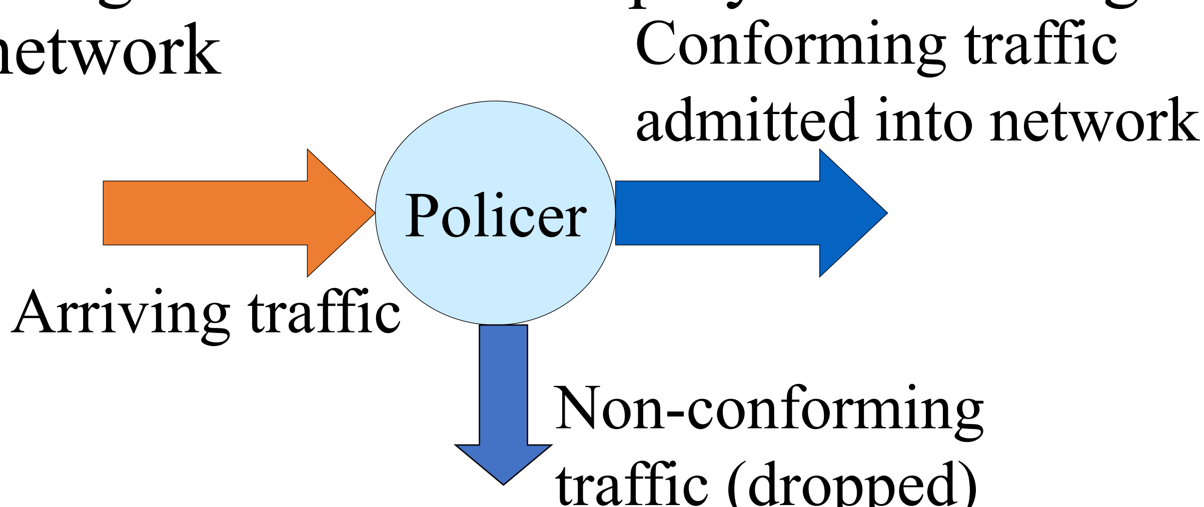


Token Bucket Shaper



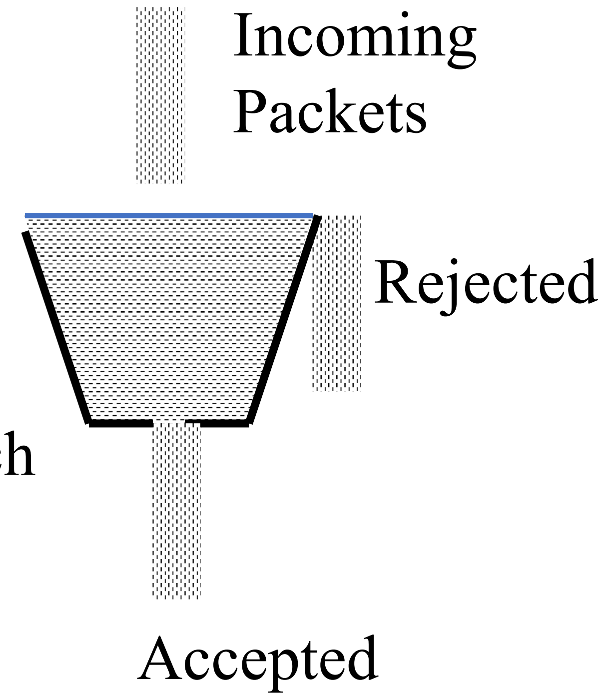
Traffic Policing

- Users violating the traffic contract can jeopardise the QoS of other connections
- The network must protect well behaving users against such traffic violations
- Policing functions are deployed at the edge (entry) of the network

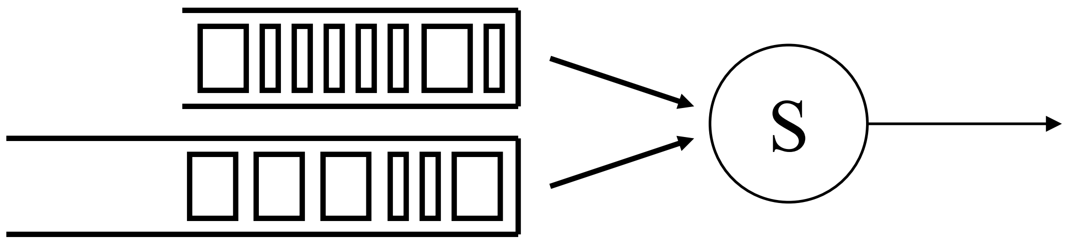


Peak Rate Policing with Leaky Bucket

- Enforces sustained rate and maximum burst size
- Requires only one counter
  - counter is decremented, to a minimum of zero, at the avg rate
  - counter is incremented by one, to a maximum of a limiting value, for each packet arrival
- An arriving packet is non-conforming if counter is at its limit

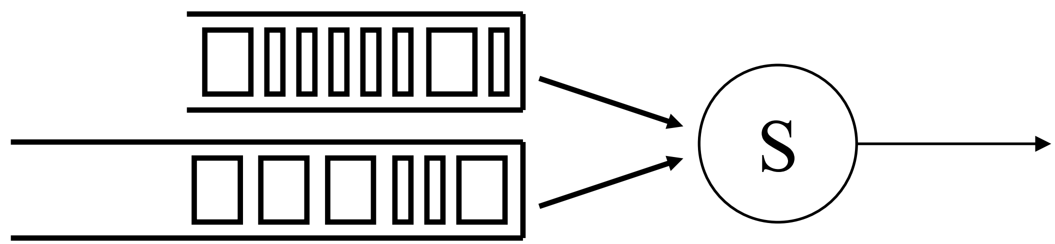


Fair Queueing



- Bit-level round robin but packet level scheduling
- Count the packet size and determine which packet would finish first. Serve that packet.
- Each flow gets the same number of bits/sec

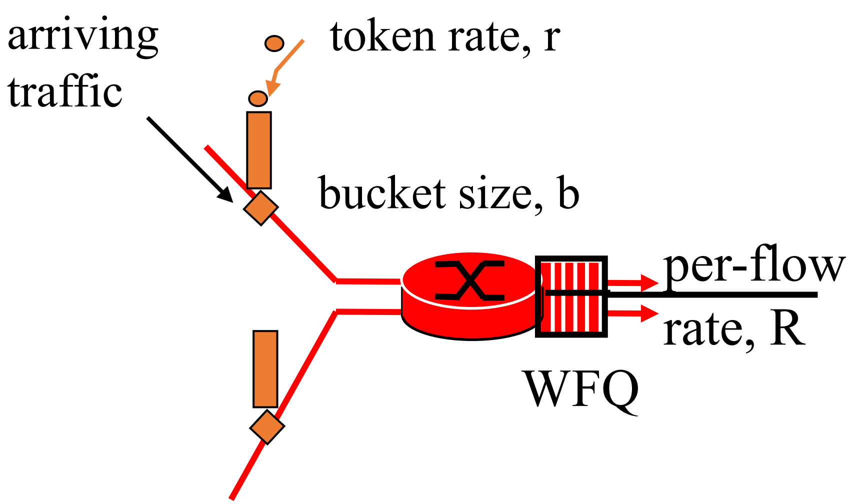
Weighted Fair Queueing (WFQ)



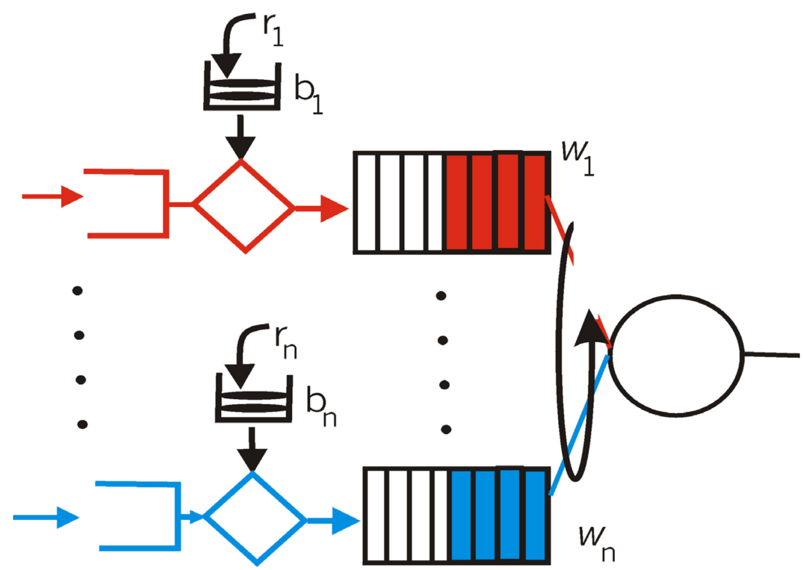
- Fair queueing with different weight for each queue
- Flow 1 gets x bit/sec
- Flow 2 gets y bit/sec
- Flow n gets z bit/sec
- Here, x, y, z are weights

Maximum Delay with WFQ and Policing

- Max Delay $d_{max} = b_i/(R\, w_i/\Sigma\, w_j)$
- Here,
- b_i=Burst size of ith flow
- R=Service Rate
- W_i=Weight of ith flow



$D_{max} = b/R$



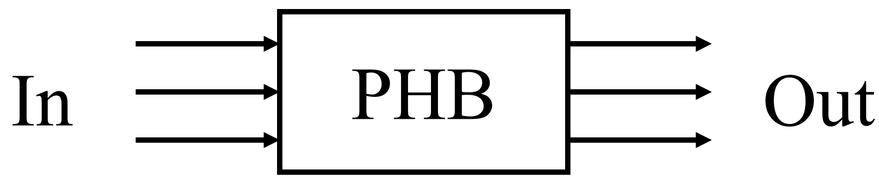
Differentiated Services

Ver	Hdr Len	Precedence	ToS	Unused	Tot Len
4b	4b	3b	4b	1b	16b

DSCP	ECN
6b	2b

- IPv4: 3-bit precedence + 4-bit ToS
- OSPF and integrated IS-IS can compute paths for each ToS
- Many vendors use IP precedence bits but the service varies
  ⇒ Need a standard ⇒ Differentiated Services
- Edge routers can mark the packets ⇒ Set ToS field
- Core routers use ToS field to provide "Per-Hop-Behavior"

Per-hop Behaviors

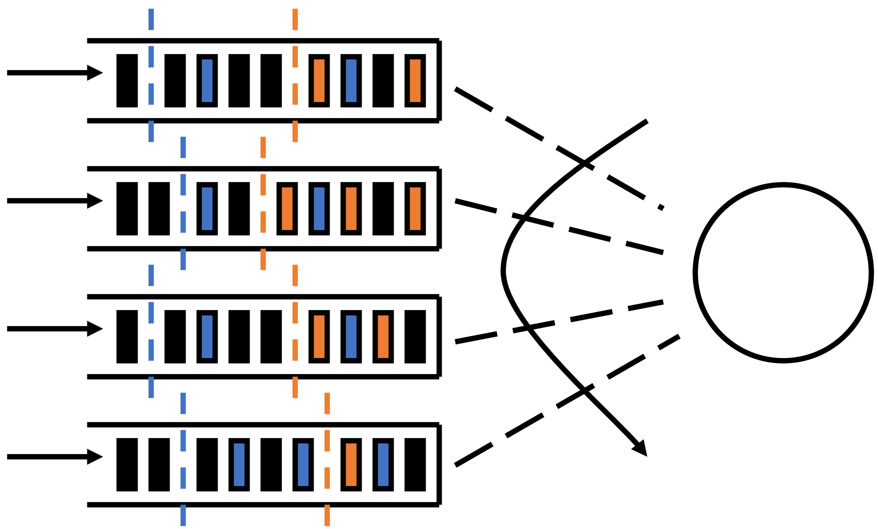


- Externally Observable Forwarding Behavior
- x% of link bandwidth
- Minimum x% and fair share of excess bandwidth
- Priority relative to other PHBs

Expedited Forwarding

- Also known as "Premium Service"
- Virtual leased line
- Guaranteed minimum service rate
- Policed: Arrival rate < Minimum Service Rate
- Not affected by other data PHBs
  $\Rightarrow$ Highest data priority (if priority queueing)
- Code point: 101 110

Assured Forwarding



- PHB Group
- Four Classes: No particular ordering
- Three drop preference per class

Assured Forwarding (Cont)

- DS nodes SHOULD implement all 4 classes and MUST accept all 3 drop preferences. Can implement 2 drop preferences.
- Similar to nrt-VBR/ABR/GFR
- Code Points:

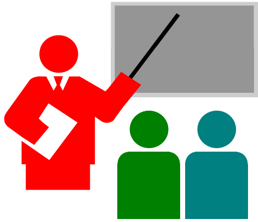
Drop Prec.	Class 1	Class 2	Class 3	Class 4
Low	010 000	011 000	100 000	101 000
Medium	010 010	011 010	100 010	101 010
High	010 100	011 100	100 100	101 100

- Avoids 11x000 (used for network control)

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

Review

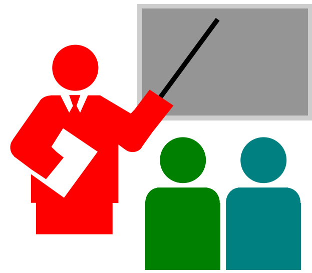


Network Support for Multimedia

1. QoS is obtained using several components including shaping, policing, differentiated services
2. Shaping is done by a token bucket
3. Policing is done using a leaky bucket
4. Differentiated services specifies per-hop behaviors
    1. Expedited Forwarding: min service rate
    2. Assured Forwarding: 4 classes, 3 drop precedence's

Ref: Section 9.5, Problems P17-P22

Summary



1. Multimedia applications require bounded delay, delay jitter, and minimum throughput
2. RTP allows sequencing and time stamping
3. SIP allows parameter negotiation and location
4. QoS requires shaping, policing, scheduling, etc.
5. Diffserv allows different packets to get different service

Ref: Entire Chapter 9, Review Exercises R1-R12, Problems P2-P4,P9, P11, P16, P19

Acronyms

ABR	Available Bit Rate
CBR	Constant Bit Rate
CD	Compact Disk
DNS	Domain Name System
DS	DiffServe
GFR	Guaranteed Frame Rate
HTTP	HyperText Transfer Protocol
IEEE	Institution of Electrical and Electronics Engineers
IP	Internet Protocol
IPTV	Internet Protocol Television
IPv4	Internet Protocol Version 4
IS	Integrated Services
LAN	Local Area Network
NAT	Network Address Translator
OSPF	Open Shortest Path First
PHB	Per-Hop Behavior

Acronyms (Cont)

QoS	Quality of Service
RAS	Registration, Admission, and Status
RTCP	Real-Time Transport Protocol Control Protocol
RTP	Real-Time Transport Protocol
RTSP	Real-Time Streaming Protocol
RTT	Round Trip Time
SC	Skype Clients
SIP	Session Initiation Protocol
SN	Super Node
SSRC	Synchronization Source
TCP	Transmission Control Protocol
ToS	Type of Service
UDP	User Datagram Protocol
URI	Uniform Resource Identifiers
URL	Uniform Resource Locator
VBR	Variable Bit Rate

Acronyms (Cont)

- VoIP Voice over IP

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