Advanced Network Technologies

Multimedia 1/2

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Multimedia

- Multimedia
- Streaming stored video
- Voice-over-IP
- RTP/SIP

Multimedia

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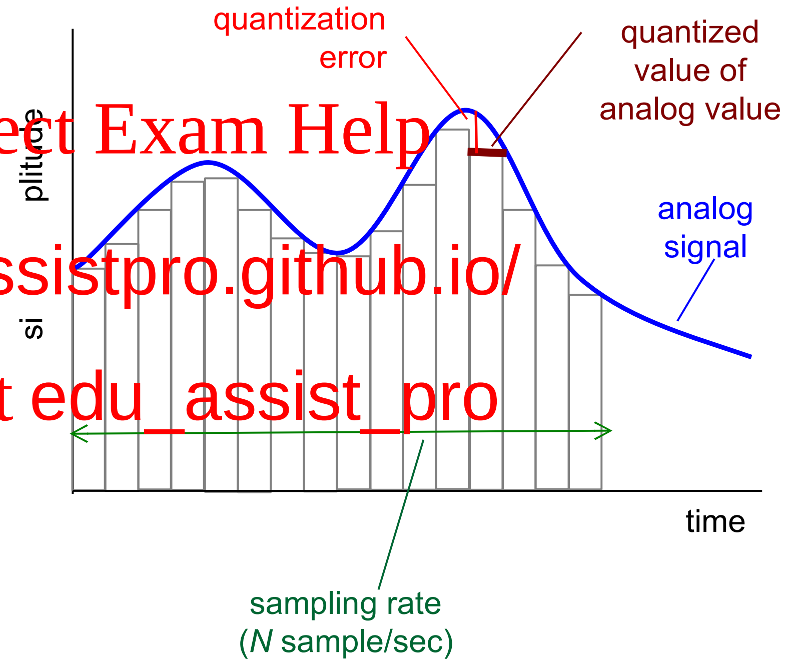
Multimedia networking: 3 application types

- *streaming, stored* audio, video
  - *streaming:* can begin playout before downloading entire file
  - *stored (at server):* can transmit faster than audio/video will be rendered (implies storing/buffering at client)
  - e.g., YouTube, N
- *conversational* voice/video ove
  - interactive nature of human-to-hu ation limits delay tolerance
  - e.g., Skype
- *streaming live* audio, video
  - e.g., live sporting event

Multimedia audio

- analog audio signal sampled at constant rate
  - telephone: 8,000 samples/sec
  - CD music: 44,100 samples/sec
- each sample quantiz rounded
  - e.g., 2^8=256 possible quantized values
  - each quantized value represented by bits, e.g., 8 bits for 256 values



Rate=44100 samples/sec * 8bit/sample = 352800 bps

Video

- Video: sequence of images displayed at constant rate
  - e.g. 24 images/sec
- Each image: array of pixels: Resolution: e.g. 480*640

  - each pixel: 3 colors
  - Red, Green, Blue (R
  - Each color has 2^8=256 possible quantize)

  - Data rate: 8*3*480*640*24 = 177 Mbps.

- 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)

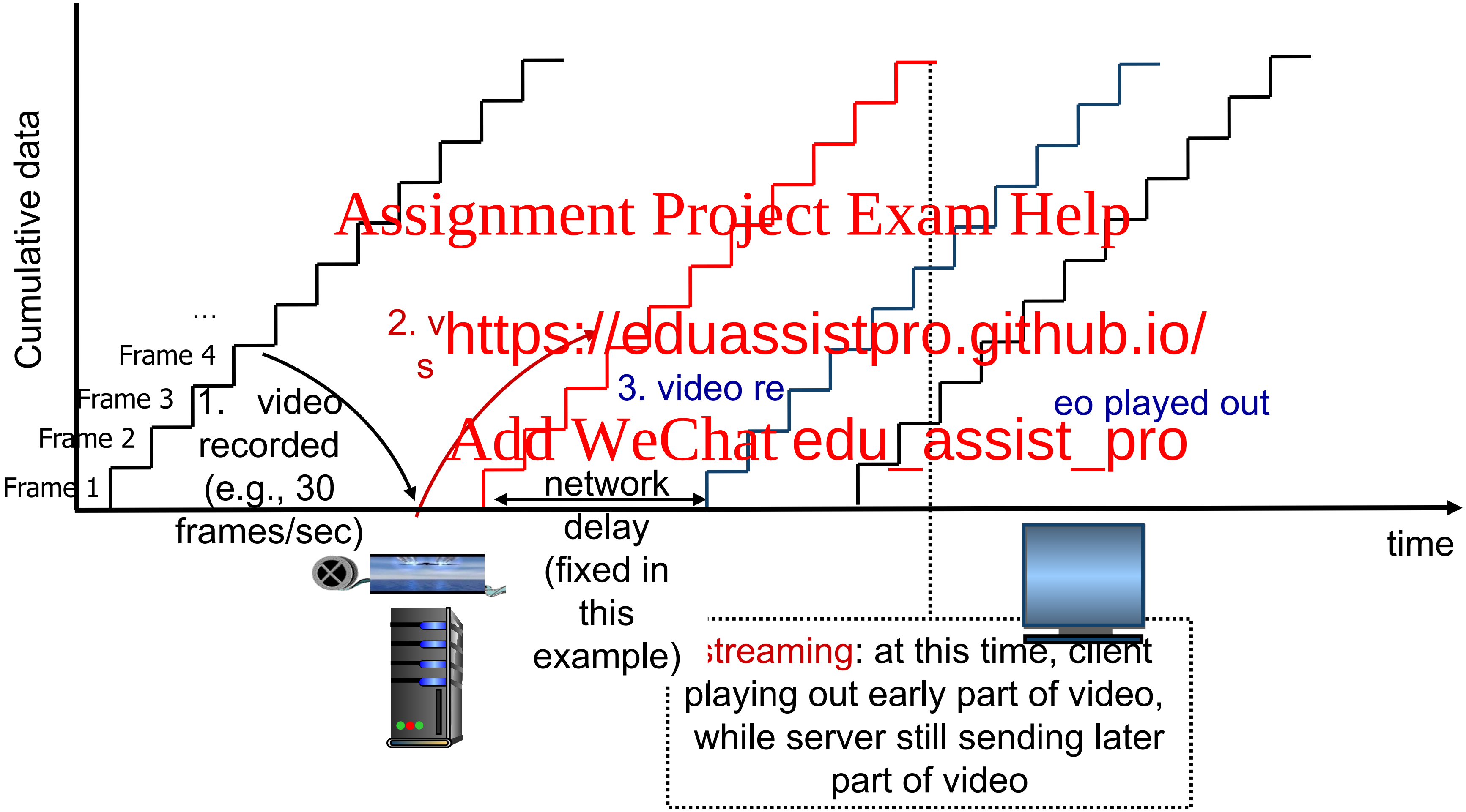
- examples:
- MPEG 1 (CD-ROM) 1.5 Mbps
- MPEG2 (DVD) 3-6 Mbps
- MPEG4 (often used in Internet, < 1 Mbps)
  - MPEG: Moving Picture Experts Group

temporal coding example: instead of sending complete frame at i+1, send only differences from frame i

frame *i+1*

Streaming Stored Video

Streaming stored video



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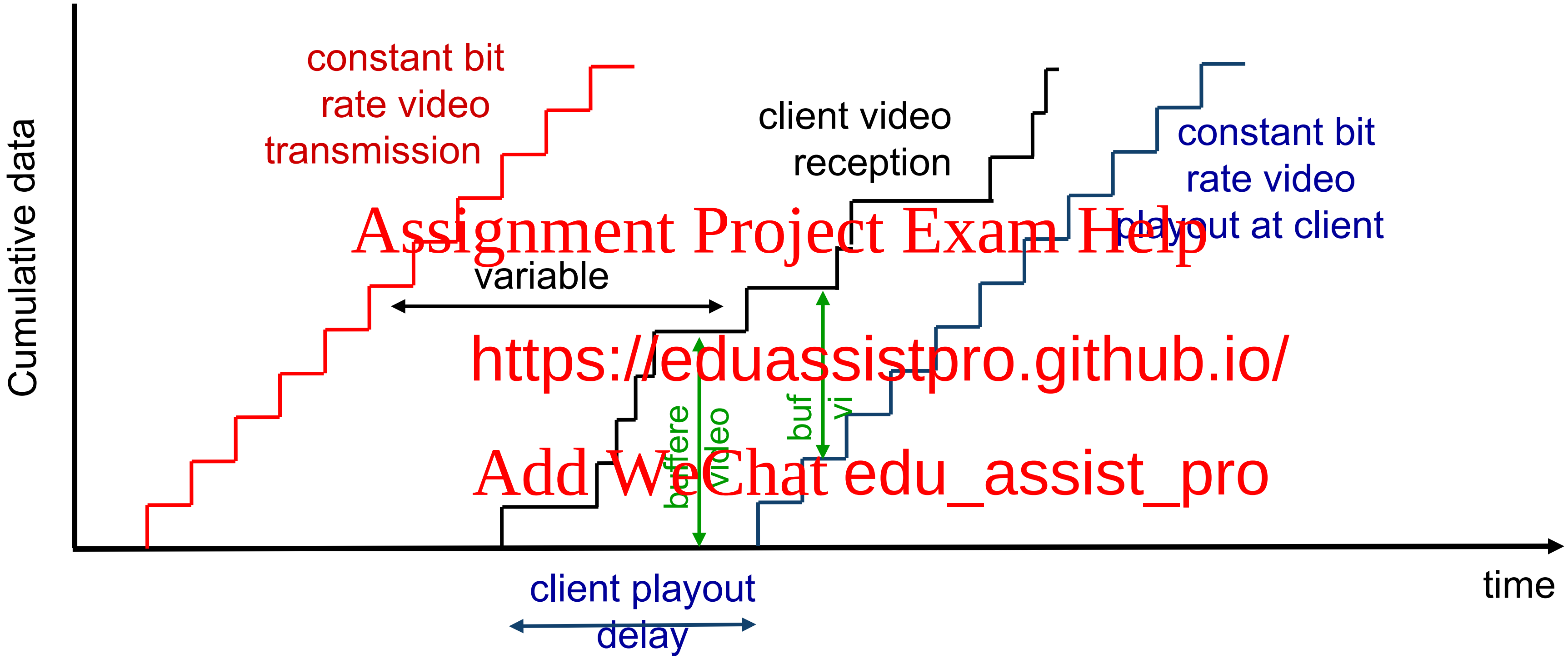
Streaming stored video: challenges

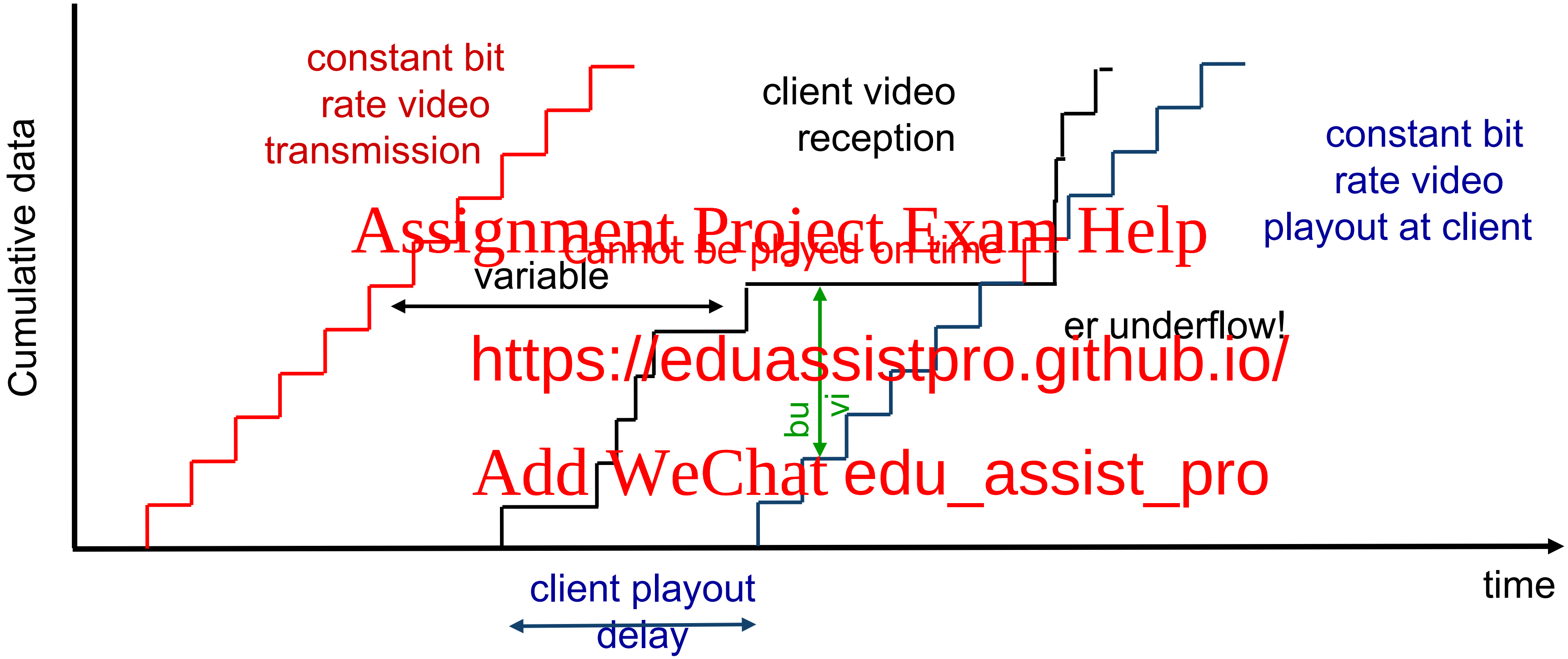
- continuous playout constraint: once client playout begins, playback must match original timing
  - … but network delays are variable (jitter), so will need client-side buffeuirements
- other challeng
  - client interactivity: pause, fasjump through video
  - video packets may be lost, retransmitted

Streaming stored video: revisited

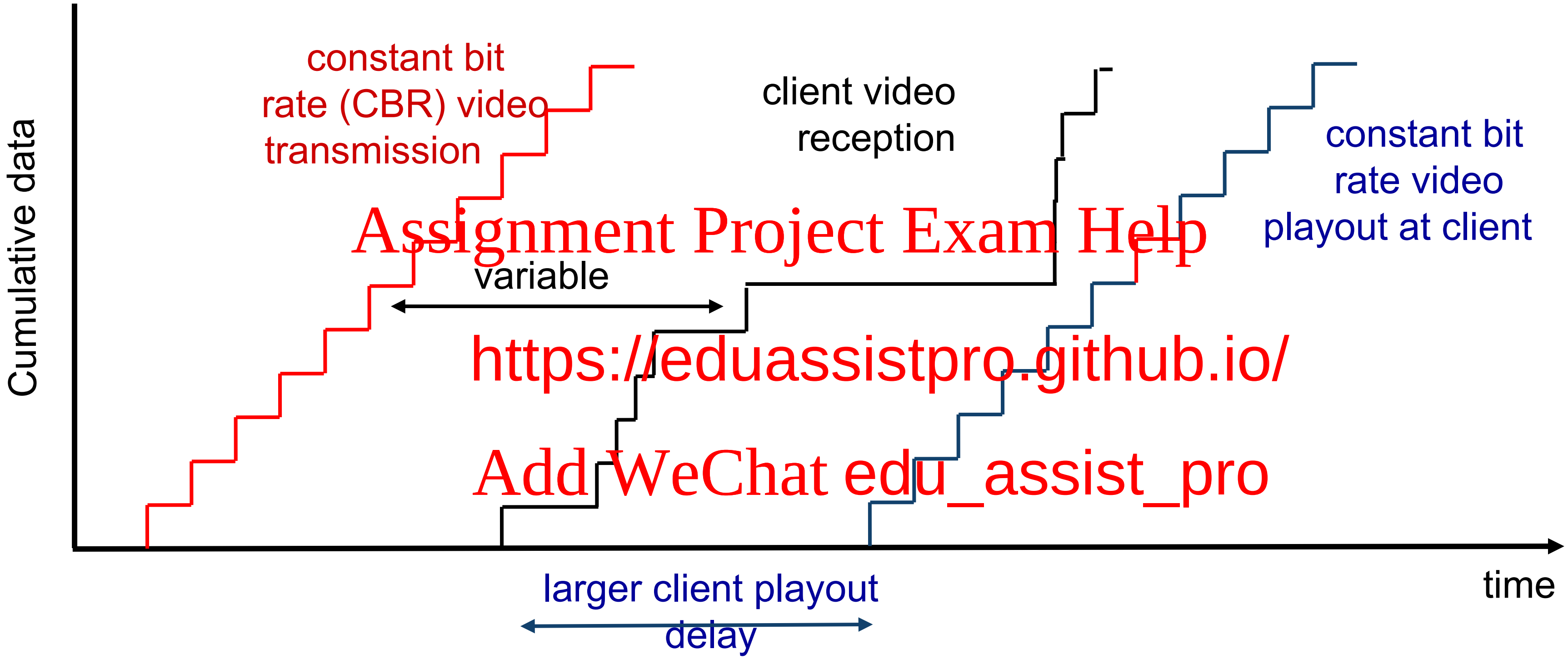


› *client-side buffering and playout delay:* compensate for network-added delay, delay jitter

Streaming stored video: revisited

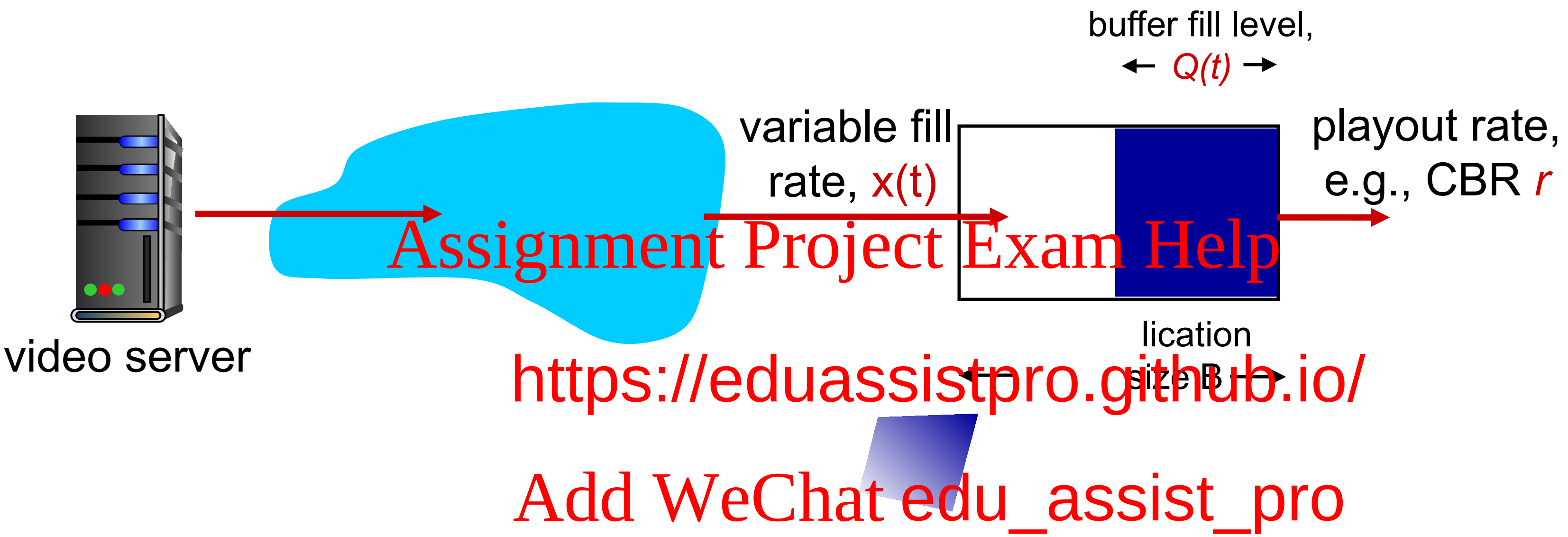


Streaming stored video: revisited

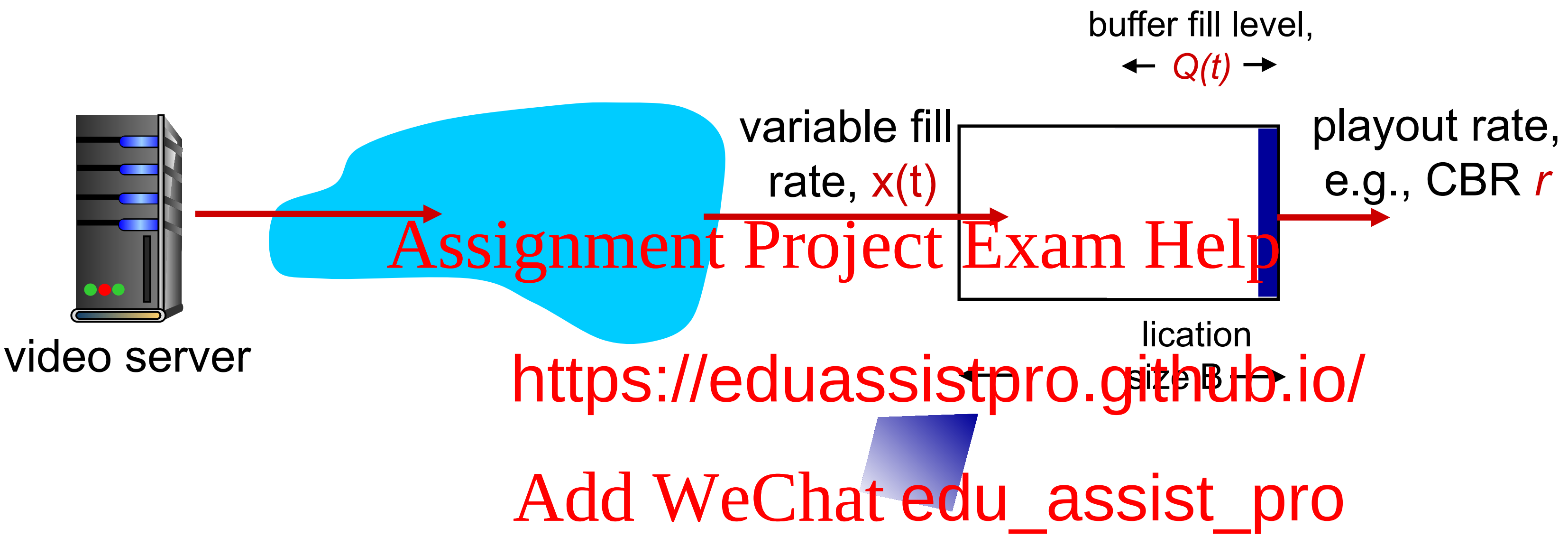


- *Increase playout delay: fewer buffer underflows*
- *initial playout delay tradeoff*

Client-side buffering, playout

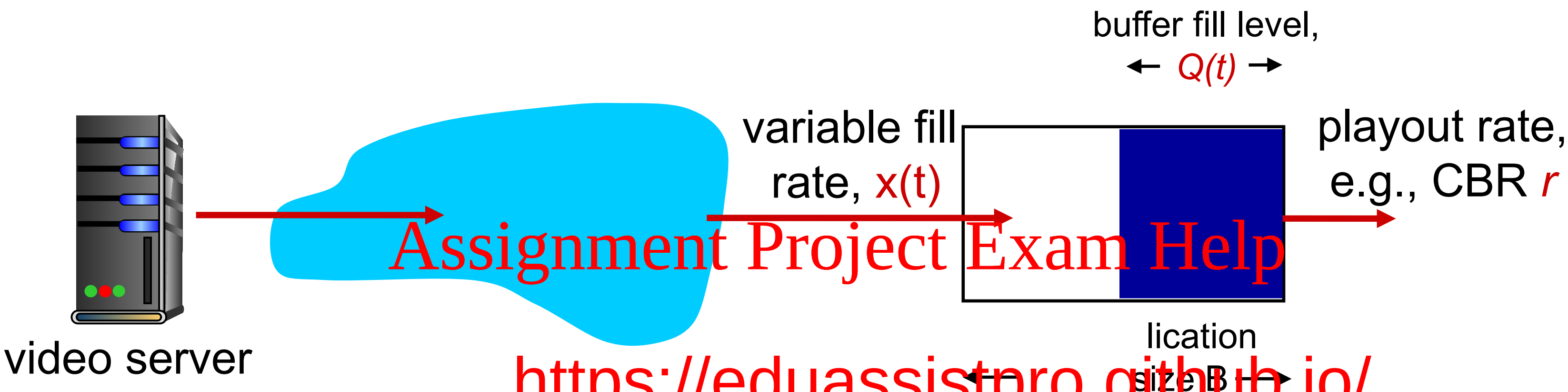


Client-side buffering, playout



1. Initial fill of buffer until playout begins at t_p
2. playout begins at t_p,
3. buffer fill level Q(t) varies over time as fill rate x(t) varies and playout rate r is constant
4. $Q(t+1)=Q(t)+x(t)$, $t \le t_p$; $Q(t+1)=\max[Q(t)+x(t)-r, 0]$, $t > t_p$
5. $Q(t)+x(t)-r<0$: buffer underflow

Client-side buffering, playout



playout buffering: average fill rate E(x ... r

- E(x) < r: buffer eventually empties (causing freezing of video playout until buffer fills again)
- E(x) ≥ r: buffer will not empty, provided initial playout delay is large enough to absorb variability in x(t)
  - *initial playout delay tradeoff:* buffer starvation less likely with larger delay, but larger delay until user begins watching

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 ork jitter
- error recovery: application-level, tim
- RTP [RFC 2326]: multimedia payload types
- UDP may *not* go through firewalls

Streaming multimedia: HTTP

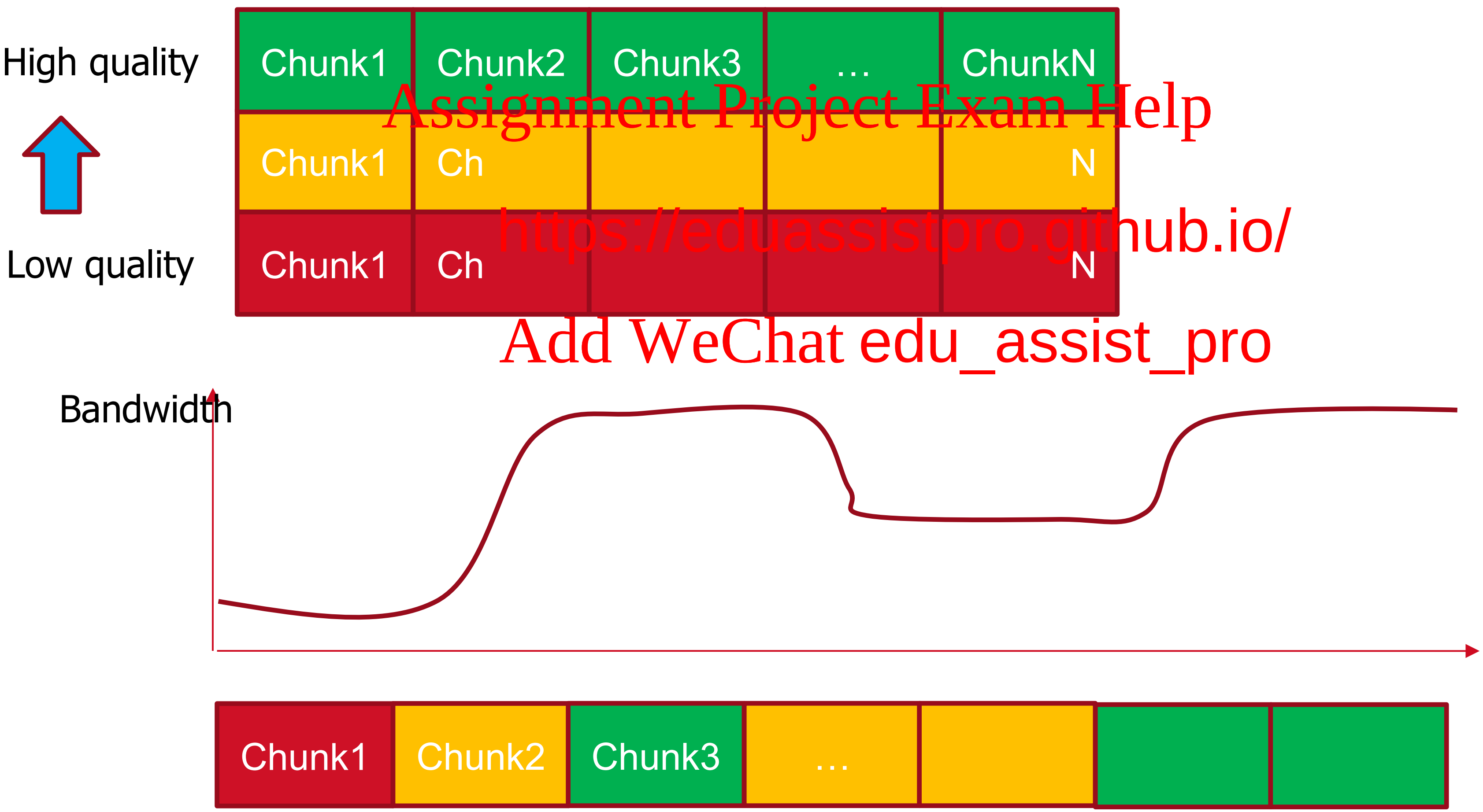
- multimedia file retrieved via HTTP GET
- send at maximum possible rate under TCP

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

Streaming multimedia: DASH

- *DASH: D*ynamic, *A*daptive *S*treaming over *H*TTP
- *server:*
  - divides video file into multiple chunks
  - each chunk store
  - *manifest file:* pro ks
- *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 current available bandwidth)

Streaming multimedia: DASH



Streaming multimedia: DASH

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

Content distribution network

- *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 failu
  - point of network co
  - long path to distant clients
  - multiple copies of video sent over outgoing link

….quite simply: this solution *doesn't scale*

Content distribution network

- *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 dist

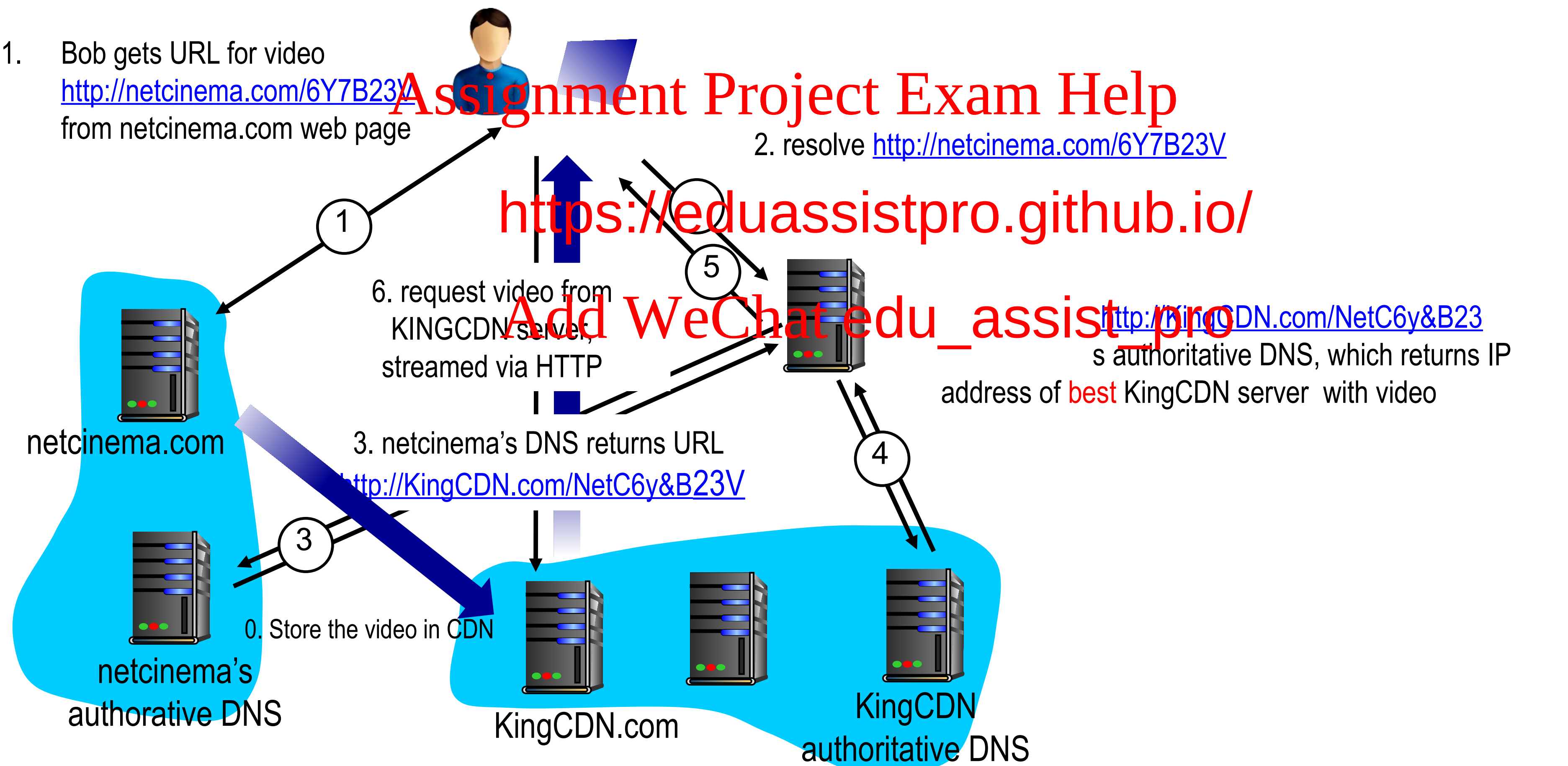
Akamai

Limelight NETWORKS

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 ac CDN DNS)

› *alternative:* let *client* decide - give cli everal CDN servers

- client pings servers, picks “best”
- Netflix approach

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Case study: Netflix

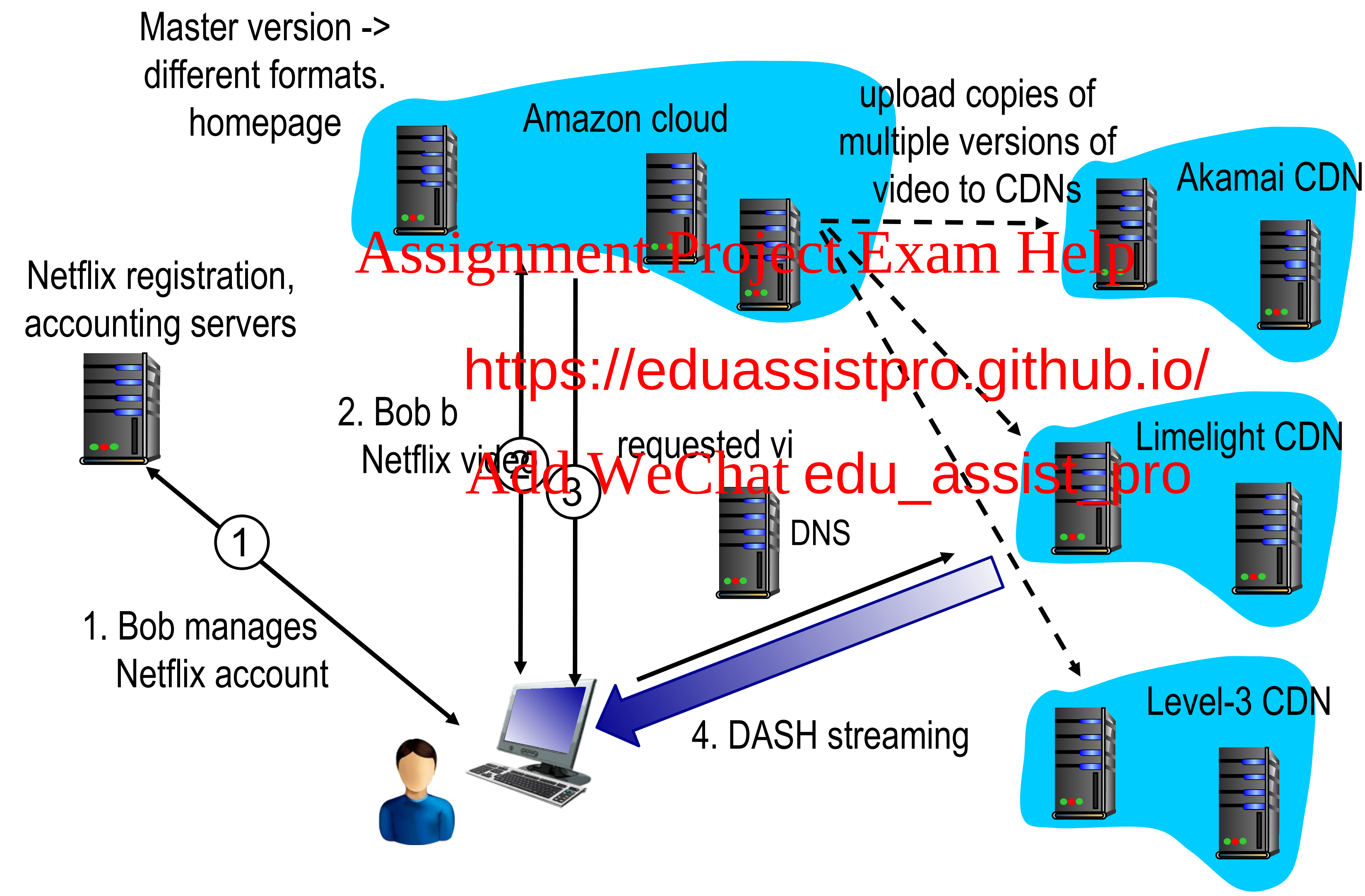
- 30% downstream US traffic in 2011

- Owns very little infrastructure, uses 3rd party services:

  - own registration, payment servers
  - Amazon (3rd part
    - Create multiple versions of movie (di gs) in Amazon cloud

    - Upload versions from cloud to CDNs
    - Cloud hosts Netflix web pages for user browsing
  - *three* 3rd party CDNs host/stream Netflix content: Akamai, Limelight, Level-3

Case study: Netflix



Voice over IP

Voice-over-IP (VoIP)

› *VoIP end-end-delay requirement*: needed to maintain "conversational" aspect

- higher delays noticeable, impair interactivity
- < 150 msec: good
- > 400 msec: bad
- includes application

› *session initialization:* how does calle address, port number, encoding algorithms?

› *value-added services:* call forwarding, screening, recording

› *emergency services:* 911/000

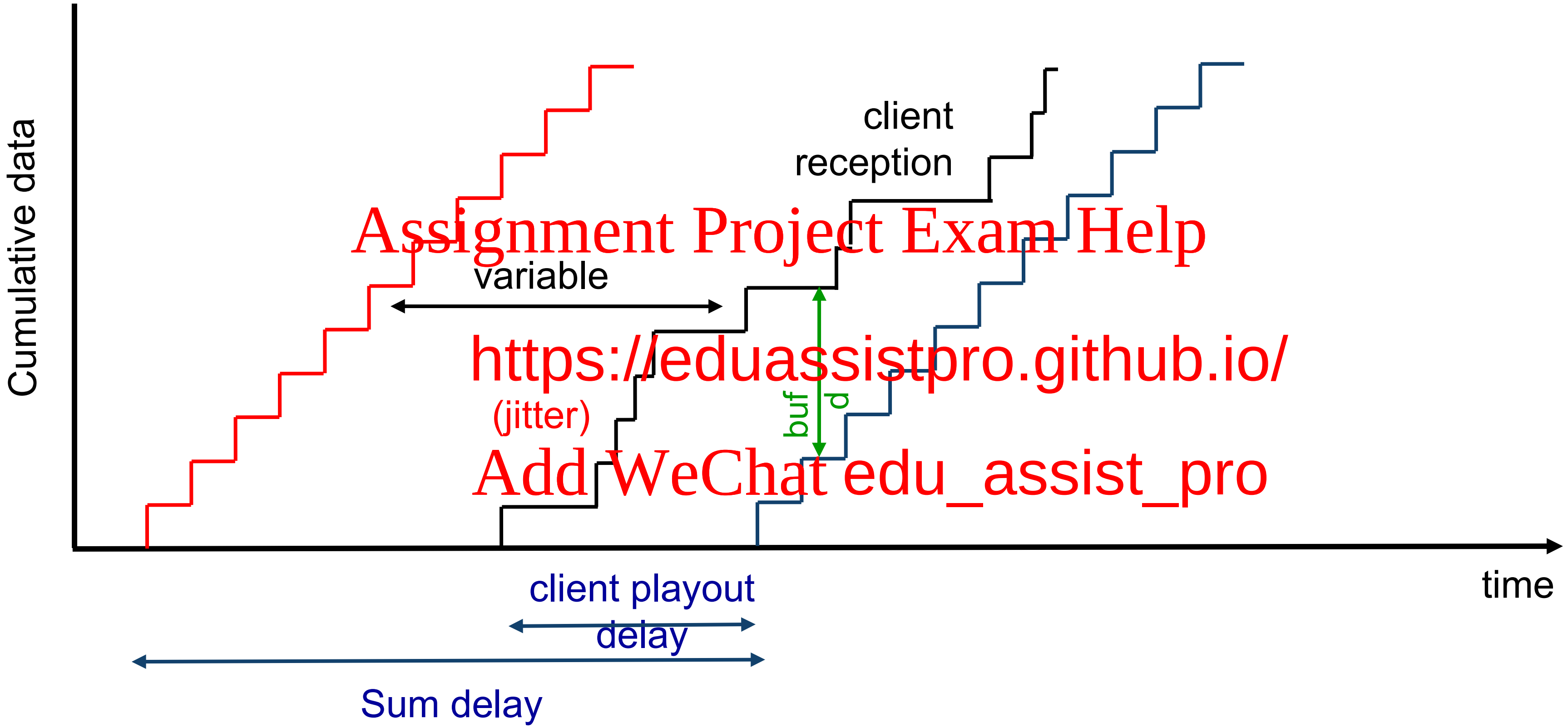
VoIP characteristics

- speaker's audio: alternating talk spurts, silent periods.
  - 64 kbps during talk spurt
  - chucks generated only during talk spurts
  - 20 msec: chucks at 8 Kbytes/sec: 160 bytes of data
- application-layer h
- chunk+header encapsulated into U   gment
- application sends segment into soc   sec during talkspurt

VoIP: packet loss, 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, transmission, proporgation.
  - typical maximum t
- *loss tolerance:* de loss concealment, packet loss rates between 1% and olerated

Delay jitter

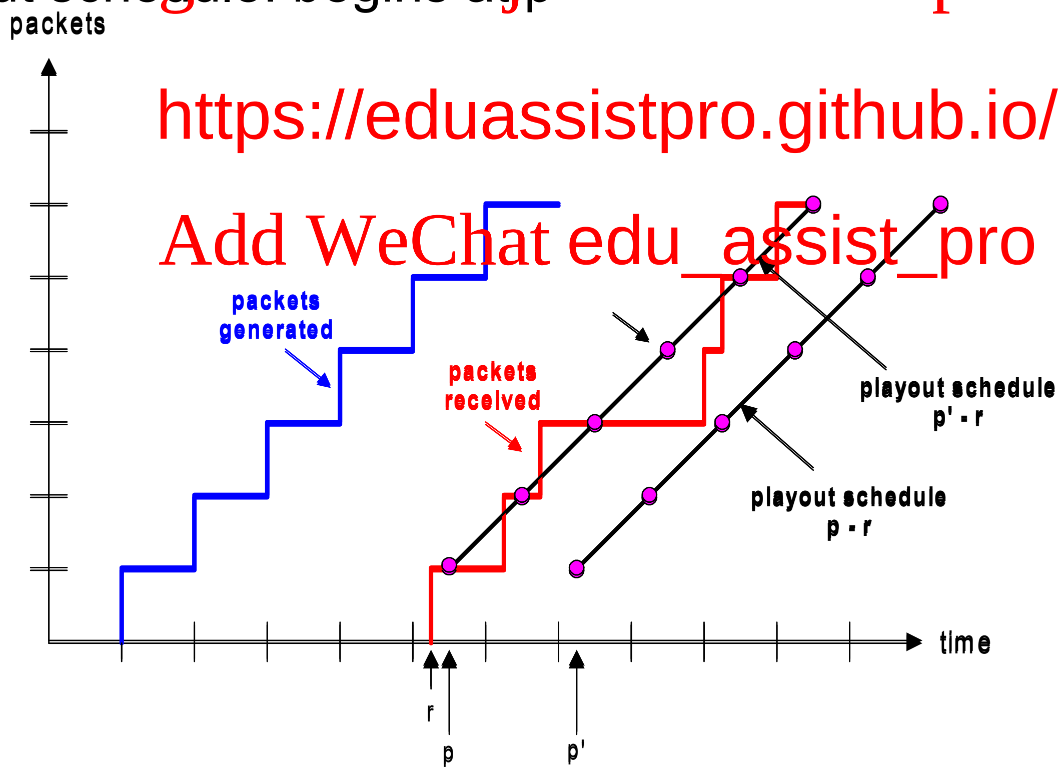


VoIP: fixed playout delay

- receiver attempts to playout each chunk exactly *q* msecs after chunk was generated.
  - chunk has time stamp *t:* play out chunk at *t+q*
  - chunk arrives after *t+q*: data arrives too late for playout: data
- tradeoff in choosing *q*:
  - *large q:* less packet loss
  - *small q:* better interactive experience

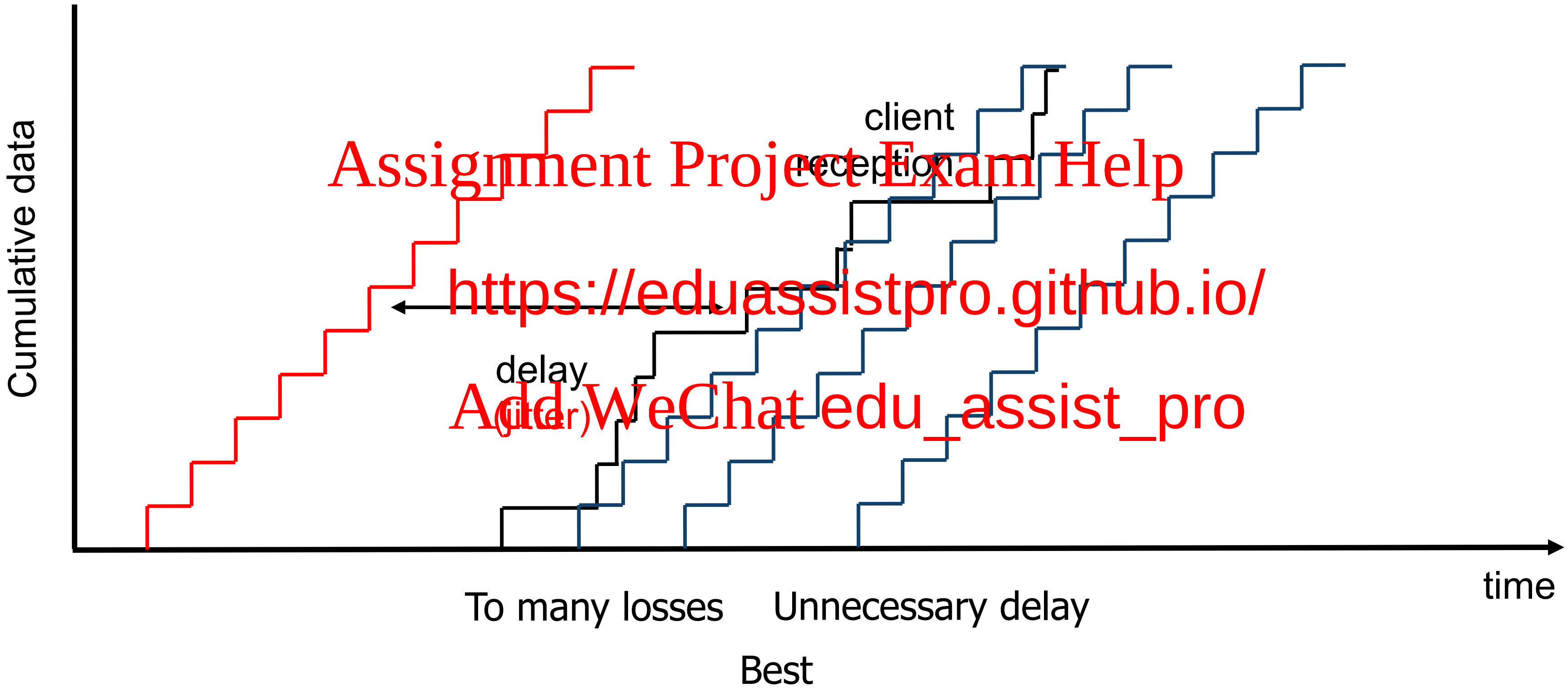
VoIP: fixed playout delay

- 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'



Adaptive playout delay

- *goal:* low playout delay, low late loss rate



Adaptive playout delay

- *goal:* low playout delay, low late loss rate
- *approach:* adaptive playout delay adjustment:
  - estimate network delay, adjust playout delay at beginning of each talk spurt
  - silent periods compressed and elongated
- adaptively estimate onentially weighted moving average, re

$$d_i = (1-\alpha)d_{i-1} + \alpha\,\underbrace{(r_i - t_i)}_{\text{measured delay of ith packet}}$$

delay estimate after ith packet (d_i)

small constant, e.g. 0.01 (α)

time received (r_i) *-* *time sent (timestamp)* (t_i)

measured delay of ith packet

Adaptive playout delay (cont'd)

- also useful to estimate average deviation of delay, v_i :

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

- estimates d_i, v_i calculated for every received packet, but used only at start of talk spurt

- for first packet in talk spurt, playout

$$playout\text{-}time_i = t_i + d_i + Kv_i$$

Delay jitter

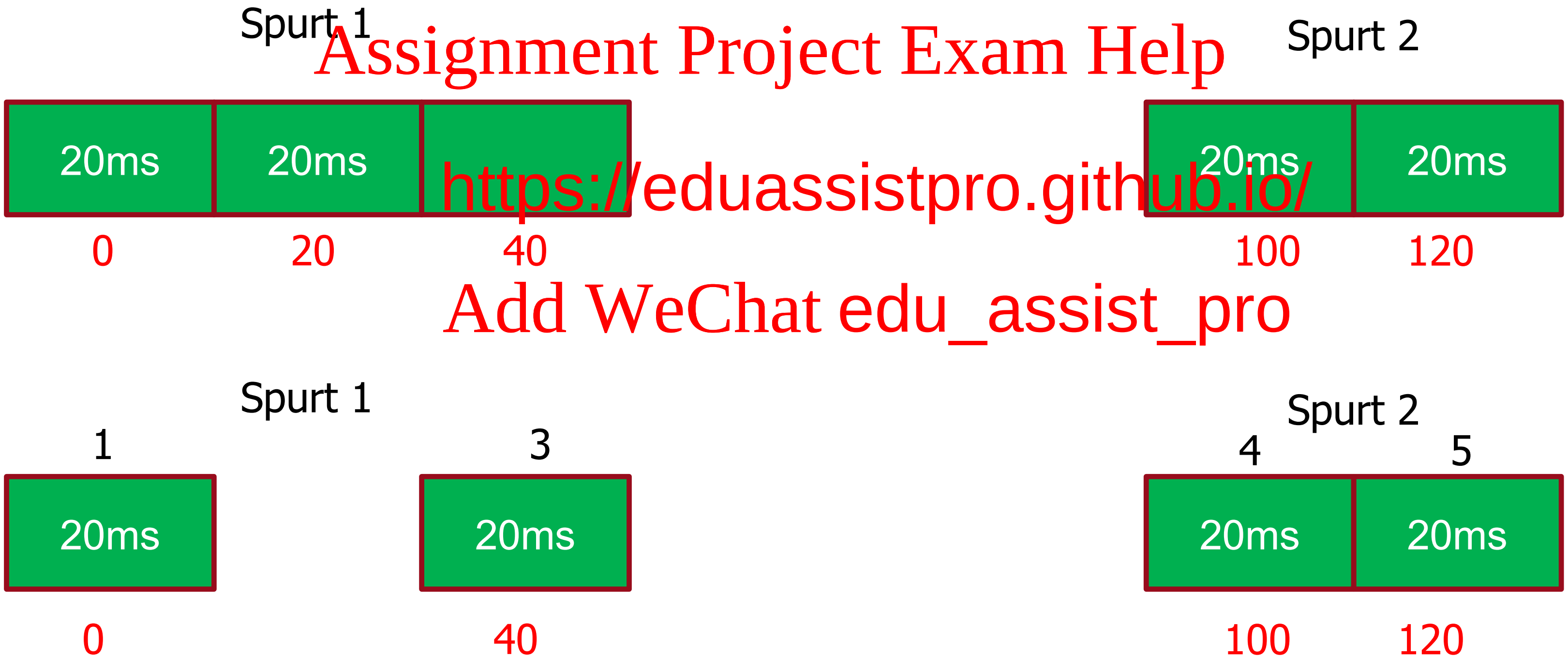
Adaptive playout delay (cont'd)

Q: How does receiver determine whether packet is first in a talkspurt?

- if no loss, receiver looks at successive timestamps
  - difference of succe rt begins.

- with loss possible, receiver must loo stamps and sequence numbers
  - difference of successive stamps > 20 msec *and* sequence numbers without gaps ⇒ talk spurt begins.

Adaptive playout delay (cont'd)



VoIP: recovery from packet loss

Challenge: recover from packet loss given small tolerable delay between original transmission and playout

- each ACK/NAK takes ~ one RTT
- alternative: *Forward Error Correction (FEC)*
  - send enough bits to allow recovery without retransmission

simple FEC

- for every group of *n* chunks, create redundant chunk by exclusive OR-ing *n* original chunks
- send *n+1* chunks, increasing bandwidth by factor *1/n*
- can reconstruct original *n* chunks if at most one lost chunk from *n+1* chunks
- Send x_1, x_2, x_3, … x_n, and $y=x_1$ xor x_2 xor x_3,…, xor x_n,   1 0 1 0
- If x_3 is lost, can re-compute x_3 from x_1, x_2, x_4, … x_n, and y

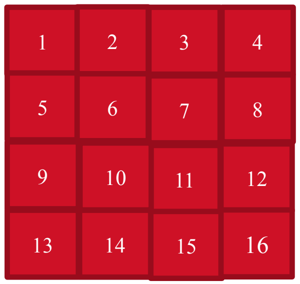
1 0 ? 0   1 XOR 0 XOR x_3 =0 x_3 =1

VoIP: recovery from packet loss (cont'd)

another FEC scheme:

- "piggyback lower quality stream"
- send lower resolution audio stream as redundant inform
- e.g., nominal stream at 64 kbps and redundant stream at 13 kbps

VoiP: recovery from packet loss (cont'd)



interleaving to conceal loss:

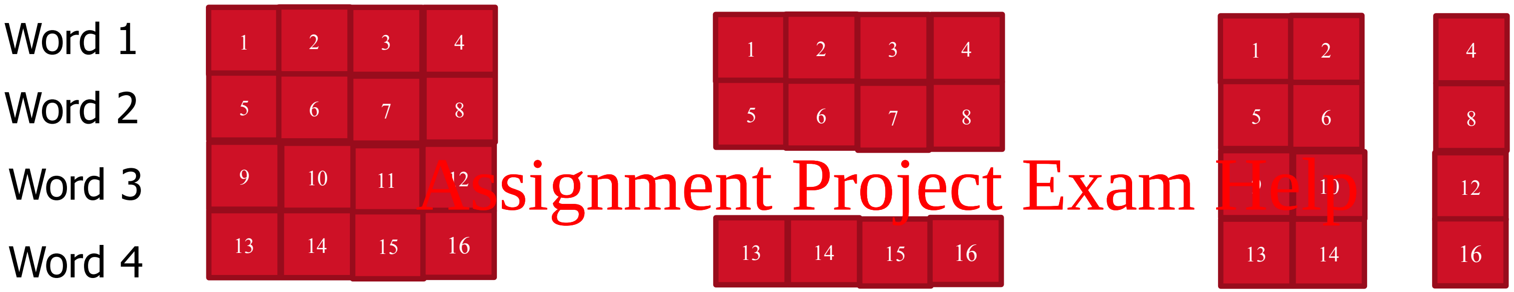
› audio chunks divided into smaller units, e.g. four 5 msec units per 20 msec audio chunk

› packet contains small units from different chunks

› if packet lost, still have *most* of every original chunk

› no redundancy overhead, but worse delay performance

VoiP: recovery from packet loss (cont'd)



e.g., word mis llable missing, acceptable

Subjective feeling is improved

Real-time Conversational

Real-Time Protocol (RTP)

- RTP specifies packet structure for packets carrying audio, video data
- RFC 3550
- RTP packet provides
  - payload type identification
  - packet sequence numbering
  - time stamping

- RTP runs in end systems
- RTP packets … lated in UDP … s
- I… bility: 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 (already existing)
- payload type identification
- packet s
- time-stamping

RTP example

example: sending 64 kbps PCM μ-law encoded voice over RTP

PCM: Pulse-code modulation: a method used to digitally represent sampled analog signals

μ-law: Special quati

Sample rate 8000sa

Quantization 8bit/sample

application collects encoded data in chunks, e.g., every 20 msec = 160 bytes in a chunk

- audio chunk + RTP header form RTP packet, which is encapsulated in UDP segment

- RTP header indicates type of audio/video encoding in each packet

r can change
g during
nce

- RTP header also contains sequence numbers, timestamps

RTP and QoS

- RTP does *not* provide any mechanism to ensure timely data delivery or other QoS guarantees

- RTP encapsulation *ot* by intermediate routers)

  - routers provide best-effort service, ecial effort to ensure that RTP packets arrive at d timely manner

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RTP header

payload type	sequence number type	time stamp	Synchronization Source ID (SSRC)	Miscellaneous fields

- payload type (7 bits): indicates type of encoding currently being used. If sender changes encoding during call, sender informs receiver via payload type field
  - w, 64 kbps
    13 kbps
  - Payload typ kbps
  - Payload typ JPEG
    - Payload type 31: H.261
  - Payload type 33: MPEG2 video
- sequence # (16 bits): increment by one for each RTP packet sent
  - detect packet loss, restore packet sequence

RTP header

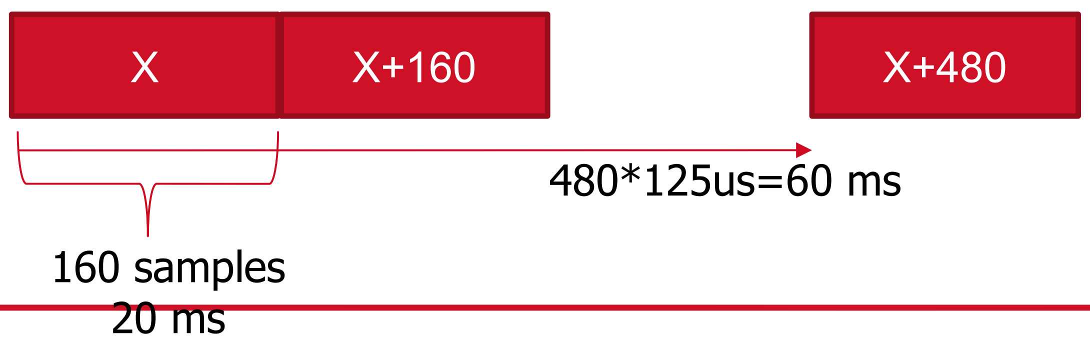
payload type	*sequence number type*	*time stamp*	*Synchronization Source ID (SSRC)*	*Miscellaneous fields*

› *timestamp field (32 bits long):* sampling instant of first byte in this RTP data packet

- for audio, timestam ach sampling period (e.g., each 125 use

- if application generates chunks of 160 les (20ms), 20ms/125us=160

- timestamp increases by 160 for each RTP packet when source is active. Timestamp clock continues to increase at constant rate when source is inactive.



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RTP header

payload type	*sequence number type*	*time stamp*	*Synchronization Source ID (SSRC)*	*Miscellaneous fields*

- sequence # + *timestamp: **knows new spurts***
- *SSRC field (32 bits long):* identifies source of RTP stream. Each stream in t SSRC

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

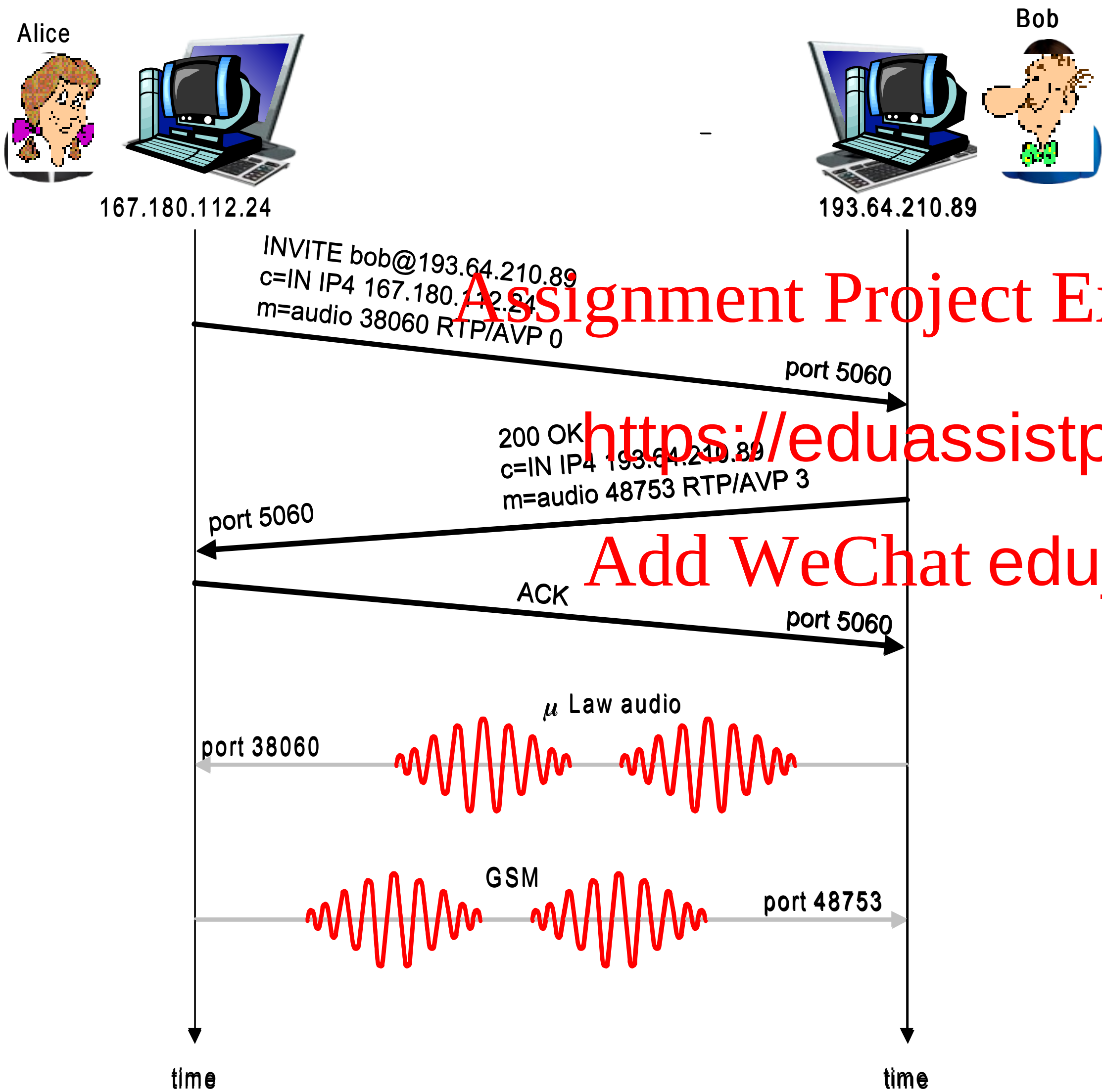
› can reach callee *(if* r where callee roams, no matter w ty using

- SIP provides mechanisms for call setup:
  - for caller to le
    know she wa
    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

gement:

  - dia streams
  - 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
- Actually, Bob and Alice talks simultaneoulsy
- SIP is out-of-band

Setting up a call (cont'd)

- codec negotiation:
  - suppose Bob doesn't have PCM μlaw encoder
  - Bob will instea
    606 Not Accep
    listing his encoders. Alice can then send new INVITE message, advertising different encoder

- rejecting a call
  - Bob can reject with replies "busy," "gone," "payment required," en"

    n be sent
    or some
    other protocol

Name translation, user location

- caller wants to call callee, but only has callee's name or e-mail address.
- need to get IP
  callee's curren
  - user moves around
  - DHCP protocol (dynamically assign IP address)
  - 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
    t home)
  - callee (calls sent
    il when callee is
    king 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@domain.com
Expires: 3600
```

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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 respons SIP proxies
- proxy returns Bob's SIP response m e
  - contains Bob's IP address
- SIP proxy analogous to local DNS server

SIP example: alice@umass.edu calls bob@poly.edu