Advanced Computer Networks

VoIP

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Feedback on reading & summaries

- Be aware of deadlines
  - the day before the presentation
- Read papers
  - not someone else's reading summaries
- Write summaries in your own words
  - the problem, main ideas
  - strengths, weaknesses, improvement
- Ask questions
  - papers cited by this one and citing this one
  - be active in class

http://www.cs.uvic.ca/~pan/485/csc485-rs.txt
http://www.cs.uvic.ca/~pan/485/reading.txt

Today's topics

- Network support for voice over IP (VoIP)
  - application
  - session
  - transport
  - network
  - and challenges
- A peer-to-peer implementation
  - Skype

VoIP

- Voice over IP
  - voice is still a major means of communication
  - trend: analog, digital, packetized
- Application requirements
  - reasonable bandwidth with a non-zero minimum
    - dependent on encoding schemes (10~100 Kbps)
  - tolerate some packet losses
    - normally less than 1%
  - sensitive to packet delay and jitter
    - one-way mouth-to-ear delay: less than 150 ms
    - average one-way delay jitter: less than 30 ms

Deal with network impairments

- Packet loss (or equivalently excessive delay)
  - application impacts
    - voice clipping and skipping, decoding dependence, etc
  - application strategies
    - loss concealment: add background noise, repeat the last packet, interpolate with the next packet, etc
    - effective up to around 20 ms (about one packet)
- End-to-end delay
  - encoding and decoding
  - transmission, propagation, processing, queuing
- Delay jitter
  - playback buffering: tradeoff

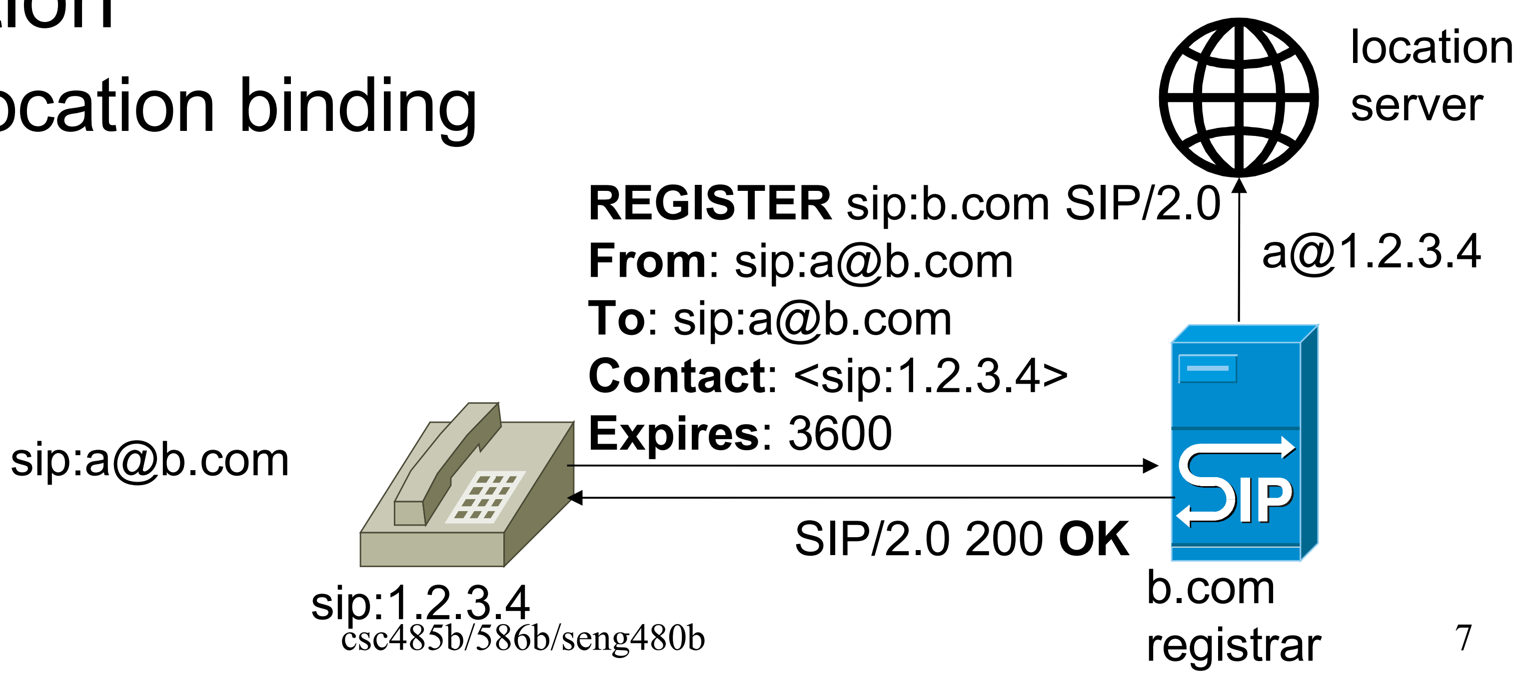
Q: small buffer or big buffer?

SIP

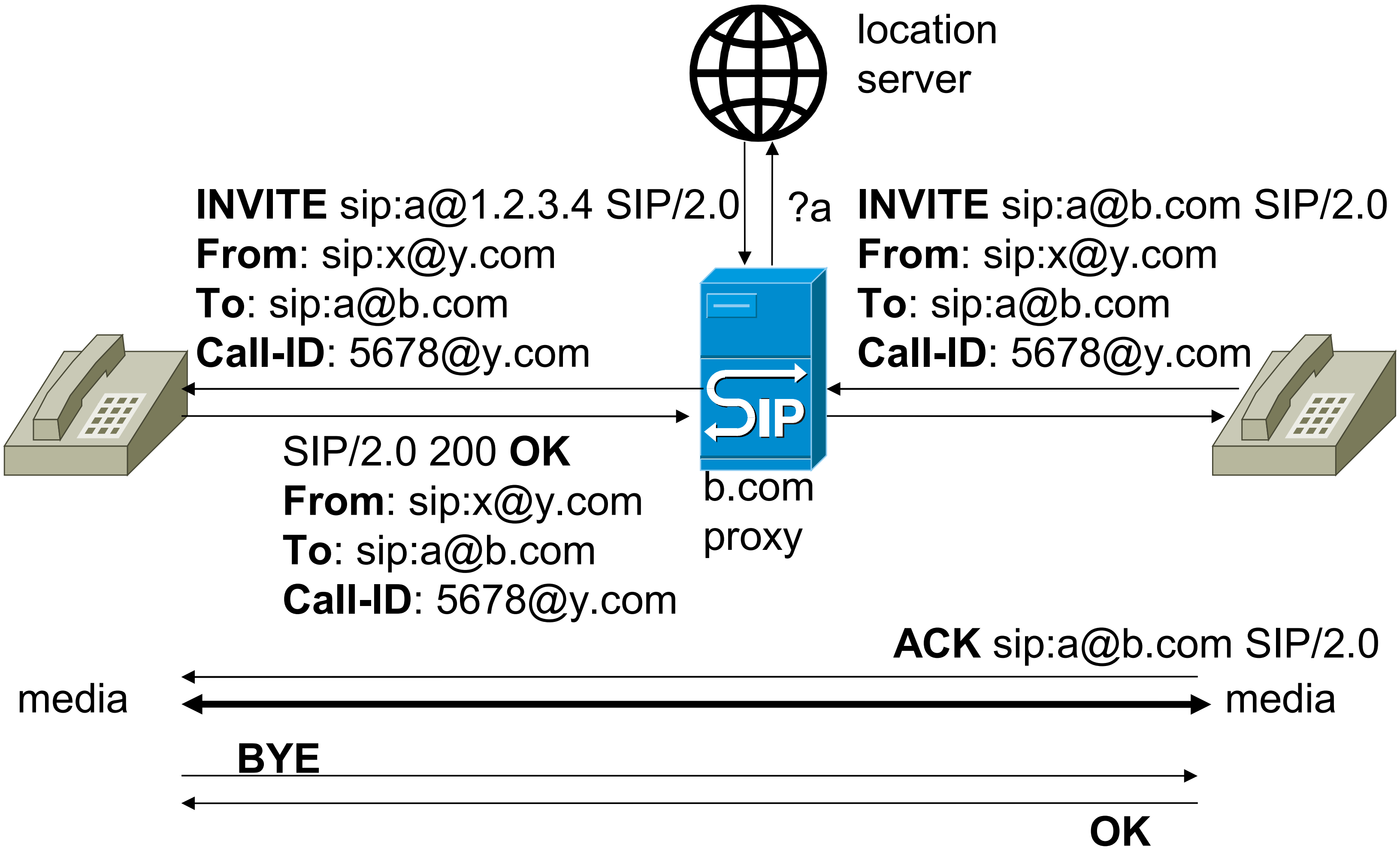
- Session initiation protocol
  - signaling: similar to SS7 in PSTN
  - SIP is not just limited to VoIP
- SIP functions
  - setup calls
  - make changes to ongoing calls
  - terminate calls
  - and more (e.g., presence)
- SIP does not offer
  - media transport, QoS support, server control, etc

SIP operations

- Design guidelines
  - client-server model, HTTP+MIME syntax
- SIP entities
  - UA, registration, proxy, redirect server
- Registration
  - name/location binding



SIP proxy


location
server
INVITE sip:a@1.2.3.4 SIP/2.0
From: sip:x@y.com
To: sip:a@b.com
Call-ID: 5678@y.com
?a
INVITE sip:a@b.com SIP/2.0
From: sip:x@y.com
To: sip:a@b.com
Call-ID: 5678@y.com
SIP
SIP/2.0 200 OK
From: sip:x@y.com
To: sip:a@b.com
Call-ID: 5678@y.com
b.com
proxy
ACK sip:a@b.com SIP/2.0
media
media
BYE
OK

SIP redirect

Explore further: http://www.cs.columbia.edu/~hgs/sip/

SDP

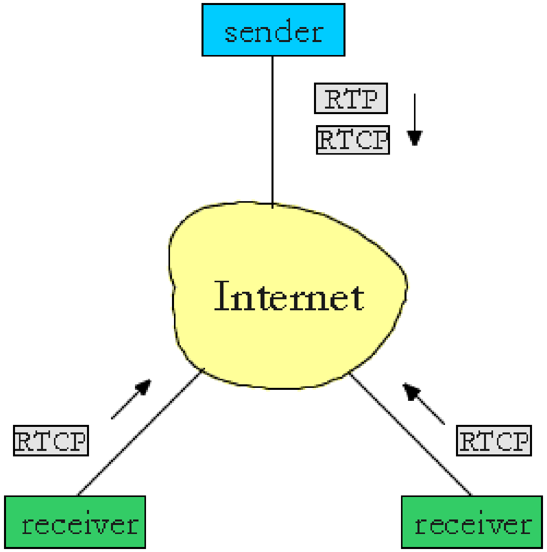
- Session description protocol
  - used in SIP to describe sessions
  - include media type, network/transport parameters
  - e.g., media: media, port, protocol, format_list
    - m=audio 2000/2 RTP/AVP 0 98
  - format attributes
    - a=rtpmap:0 PCMU/8000
  - connection: net_type, add_type, address/TTL/#
    - c=IN IP4 1.2.3.4/127/3
- Ref: http://www.ietf.org/rfc/rfc4566.txt

RTP/RTCP

- Real-time transport protocol
  - does NOT guarantee real-time itself
  - but does provide mechanisms to achieve so
- RTP profile
  - Marker: e.g., the start of a talk spurt
  - Payload Type: e.g., specific audio codec
    - PT=0: uPCM 64Kbps
    - PT=3: GSM 13Kbps
  - timestamp: e.g., sampling rate, 8KHz PCM
  - packet size: e.g., about 20ms samples in PCM
    - packets independent as much as possible: ALF
  - other issues: e.g., mixed audio channels

Explore further: http://www.cs.columbia.edu/~hgs/rtp/

RTCP

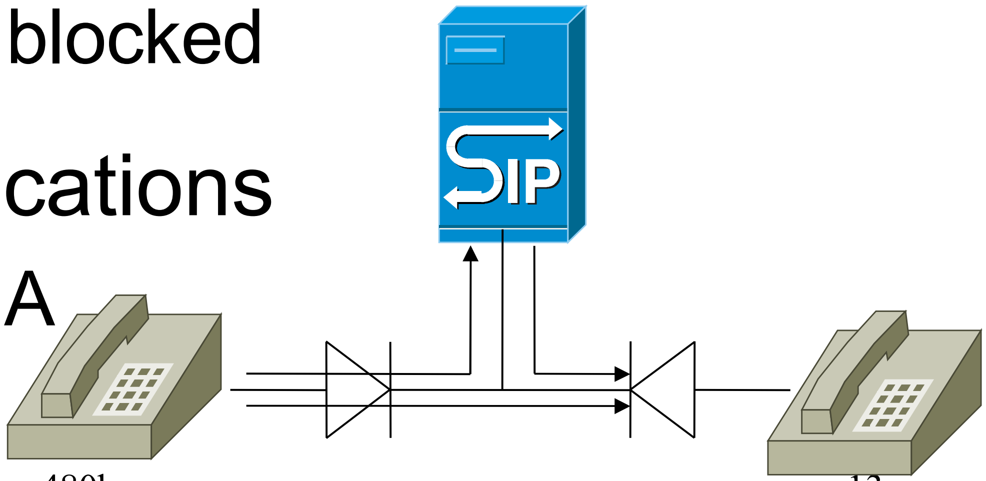


- Real-time control protocol
  - RTP's control companion
  - purpose: feedback control information
    - for flow/error/congestion/quality control
  - two consecutive UDP ports for RTP and RTCP
- Sender report
  - offer sending/reception statistics
  - NTP/RTP time stamp, byte/packet count, etc
- Receiver report
  - offer reception statistics
  - short/long-term loss ratio, time stamp, jitter, etc

Q: how does RTCP deal with scalability?

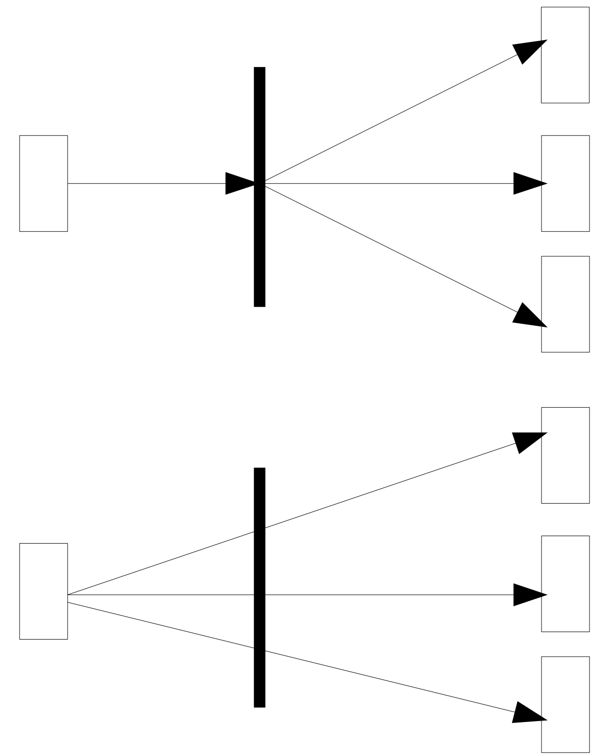
NAT

- Network address translation
  - was to deal with IPv4 address shortage
  - now pervasive in all networking scenarios
- "Directional" connectivity
  - outgoing connections are OK
    - mappings are created to filter incoming packets
  - incoming connections are blocked
- Problems with VoIP applications
  - how SIP server reaches UA
  - how caller reaches callee



Q: problems when behind your home router?

NAT traversal

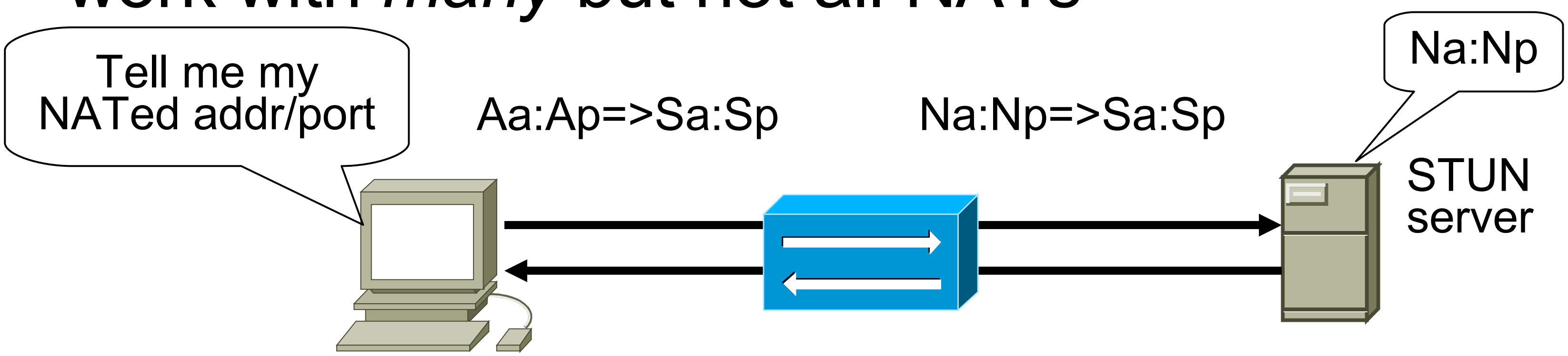


- NAT characterization
  - full cone
  - restricted cone
  - port-restricted cone
  - symmetric cone
- NAT traversal
  - static configuration
  - UPnP (universal plug and play)
  - application-layer gateway
  - STUN, TURN (relay)
  - ICE (interactive connection establishment)

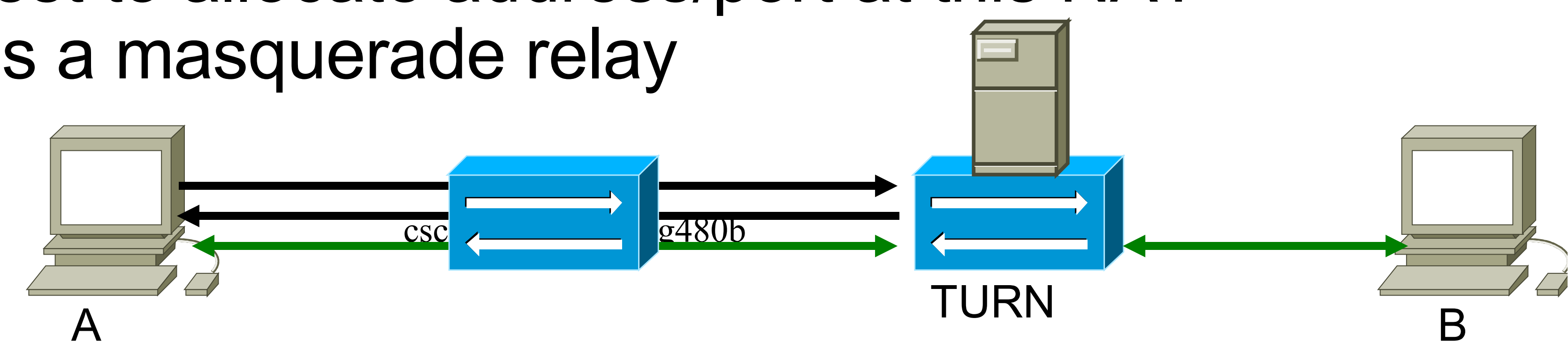
Explore further: http://www.cs.uvic.ca/~pan/seng490

STUN and TURN

- Simple traversal of UDP through NAT
  - probe and learn allocated address/port at NATs
  - work with *many* but not all NATs



- Traversal using relay NAT
  - request to allocate address/port at this NAT
  - act as a masquerade relay



Challenges

- Internet: best-effort service
  - no guarantee on minimal throughput
  - excessive packet loss, excessive delay, jitter, etc
  - better than best-effort services?
- Application: client-server model
  - scalability issues
  - peer-to-peer models?
- NAT and firewall
  - NAT traversal is not bullet-proof
- Security
  - “who else can hear you?”

IP telephony examples

- Vonage: proprietary VoIP infrastructure
  - good PSTN interworking
  - SIP compatible
    - phone adapter: SIP UA and more
- Skype: without specialized infrastructure
  - better NAT traversal capability
    - with the help of other users
    - voice encryption
  - proprietary protocols
  - an peer-to-peer implementation

Student presentation

- Chun-Hung Chiu: Skype
  - [BS06] Salman A. Baset and Henning Schulzrinne, "An Analysis of the Skype Peer-to-Peer Internet Telephony Protocol", IEEE Infocom 2006. [Skype]

This lecture

- VoIP and P2P
  - network support for VoIP applications
    - application, session, transport, network
  - challenges
    - from the viewpoint of applications and networks
  - Skype
    - a peer-to-peer implementation
- Explore further
  - Q and “Explore further” footnotes
  - we still know little about Skype!

Next lectures

- Congestion Control
  - June 11: [JK88] V. Jacobson and M. Karels, Congestion Avoidance and Control, In Proc. ACM SIGCOMM '88. [TCPCC]
  - June 13: [BOP94] L. Brakmo, S. O'Malley and L. Peterson TCP-Vegas: new techniques for congestion detection and avoidance, In Proc. of ACM SIGCOMM '94. [TCPVegas]
  - June 18: [PFTK98] Padhye, J., Firoiu, V., Towsley, D., and Kurose, J., "Modeling TCP Throughput: a Simple Model and its Empirical Validation". In Proceedings of ACM S IGCOMM 1998. [TCPmodel]
  - June 20: [KDR02] Dina Katabi, Mark Handley, and Chalrie Rohrs. Congestion Control for High Bandwidth-Delay Product Networks. In the proceedings on ACM Sigcomm 2002. [XCP]
- Bring up your course project web page at the google group