

# Advanced Computer Networks

VoIP

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# First things first

- Course project
  - ↳ Website and progress log by February 27
  - ↳ In-class update (5mins/project) in the same week
  - ↳ Feedback from the class, instructor and TA
  - ↳ Continue the discussion on crosscourse
- Reading summary 4
  - ↳ Pick a paper and have my approval by March 1
  - ↳ Related to your course project is highly encouraged
  - ↳ Summary&video present to the class by March 20

# Course project website

- Count for 5% in your final grade
- Updated throughout the project, by you
  - list URL on crosscourse->course projects
    - please populate with your project proposal
      - what's the problem and why is it important?
      - what have been done on it and why they are not enough? (including your previous and other ongoing projects)
      - what's your approach and expected deliverables?
      - a roughly biweekly schedule toward the end of March
      - **progress/milestone: keep updated at least biweekly**
      - *they are useful materials for your course project report*

# 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

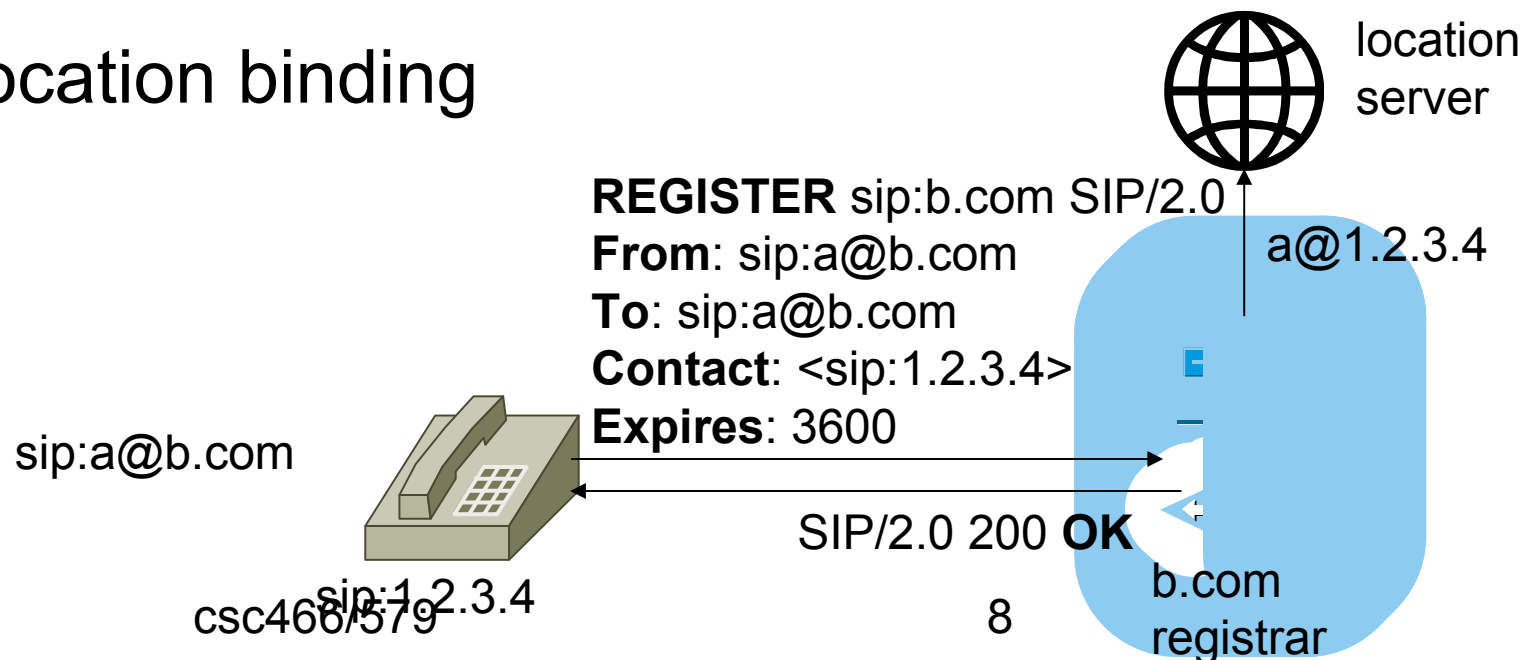
- Packet loss (or equivalently excessive delay)
  - ↳ application impacts
    - voice clipping and skipping, decoding dependence, etc
  - ↳ application strategies
    - loss concealment: add background noise, repeat the last packet, interpolate with the next packet, etc
    - effective up to around 20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

# 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

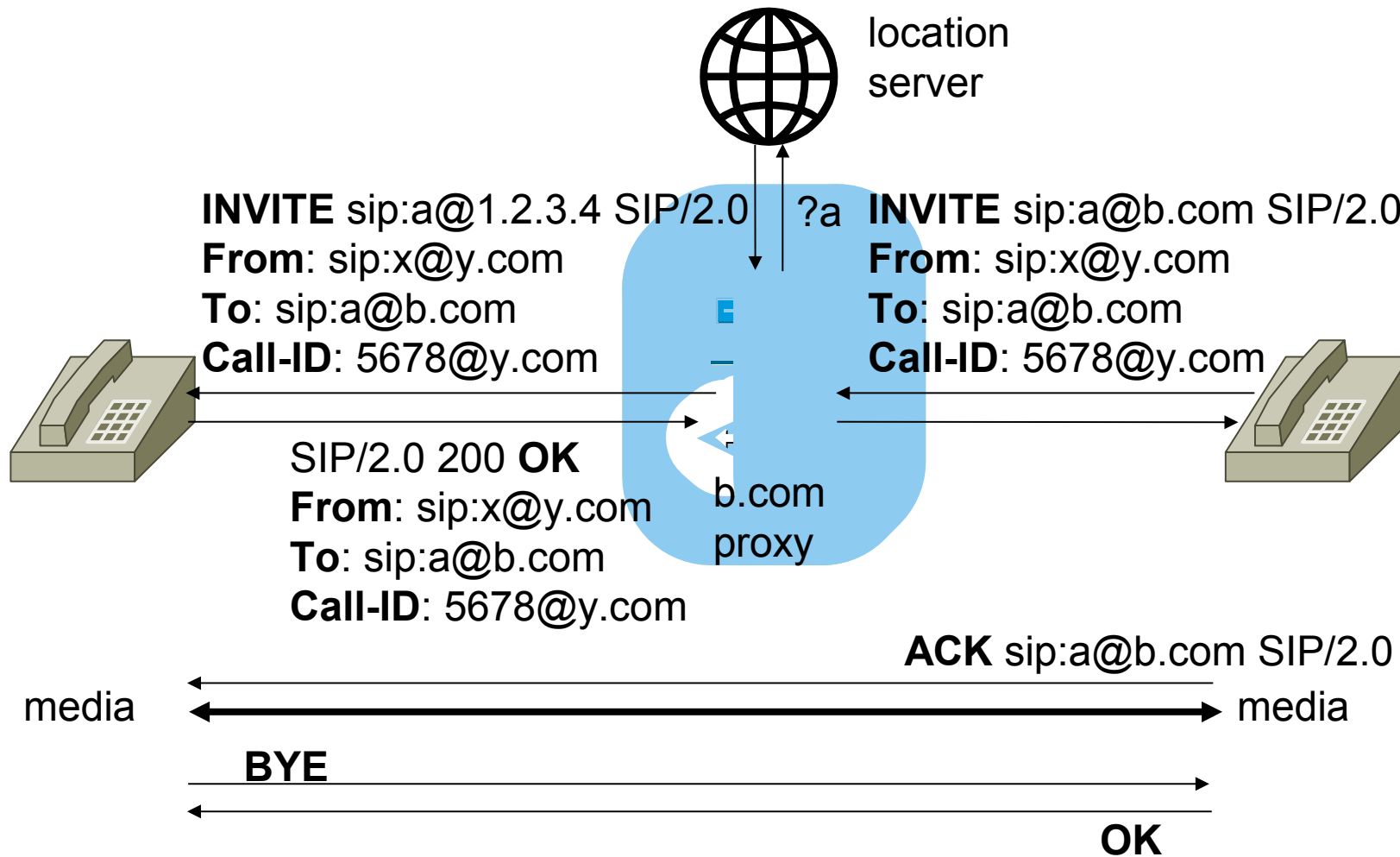
# SIP operations

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

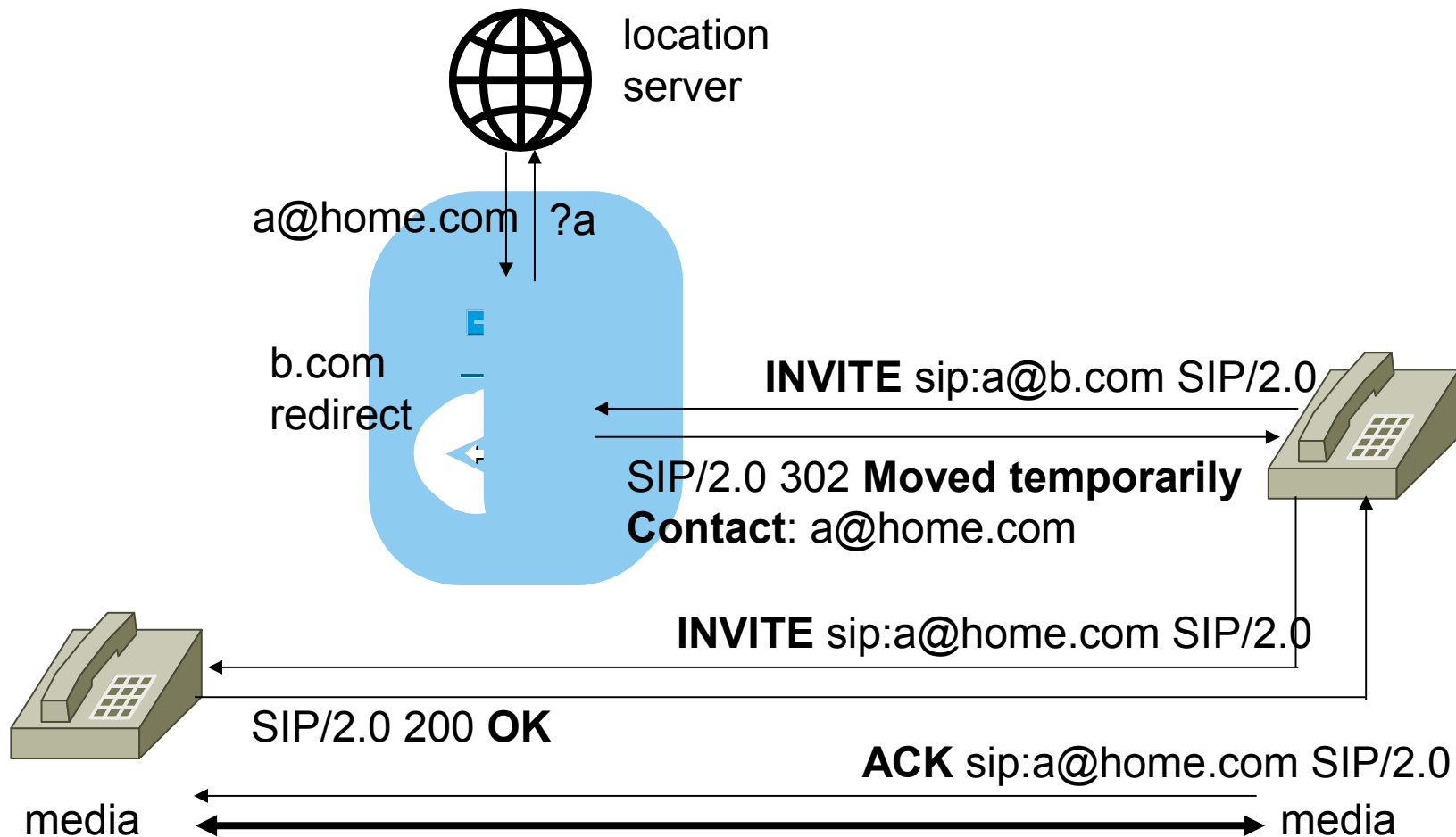




# SIP proxy



# SIP redirect



# SDP

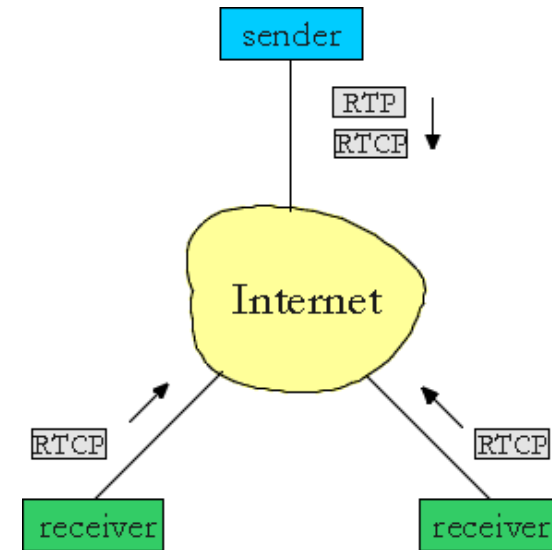
- Session description protocol
  - used in SIP to describe sessions
  - include media type, network/transport parameters
  - e.g., media: media, port, protocol, format\_list
    - m=audio 2000/2 RTP/AVP 0 98
  - format attributes
    - a=rtpmap:0 PCMU/8000
  - connection: net\_type, add\_type, address/TTL/#
    - 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

# 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

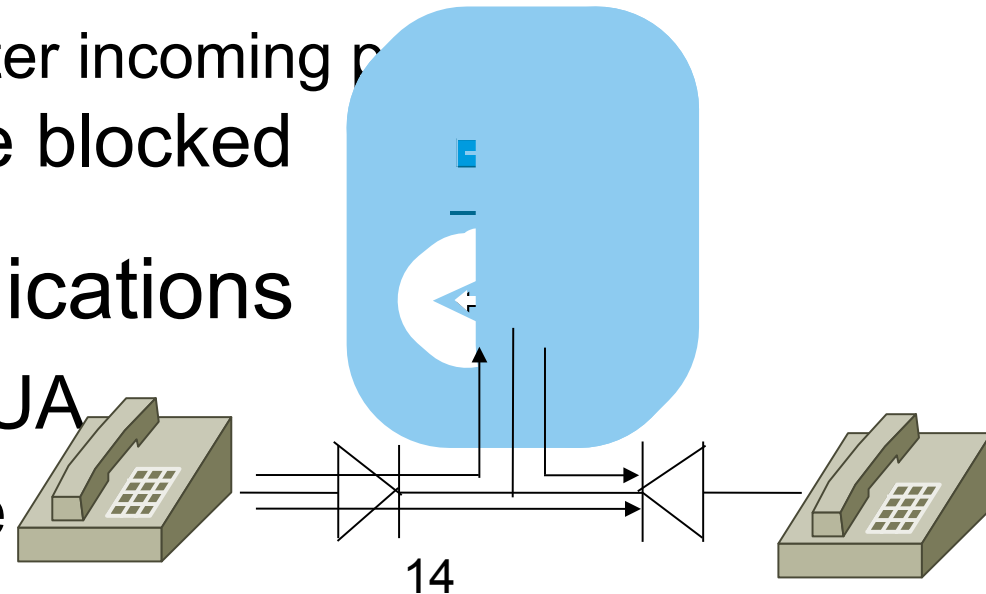


# NAT

- 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 p
  - incoming connections are blocked
- Problems with VoIP applications
  - how SIP server reaches UA
  - how caller reaches callee

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Q: problems when behind your home router?

# NAT traversal

- NAT characterization

- ↳ full cone

- ↳ restricted cone

- ↳ port-restricted cone

- ↳ symmetric cone

- NAT traversal

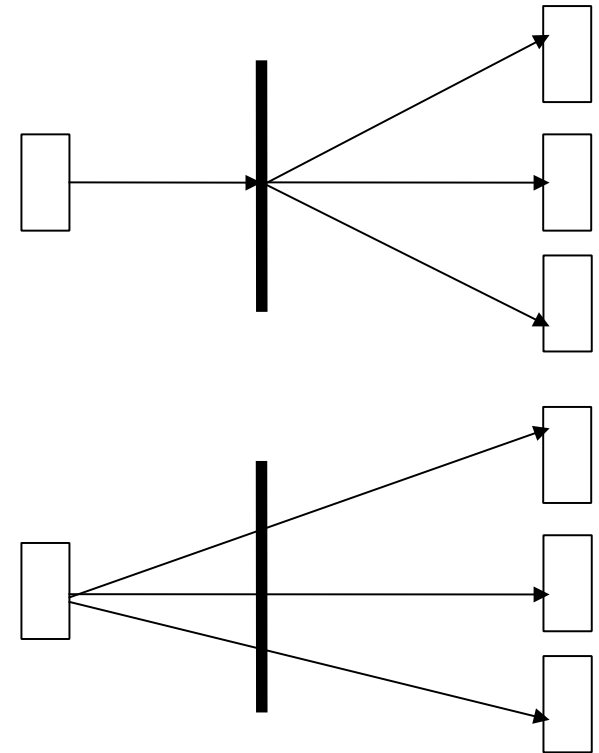
- ↳ static configuration

- ↳ UPnP (universal plug and play)

- ↳ application-layer gateway

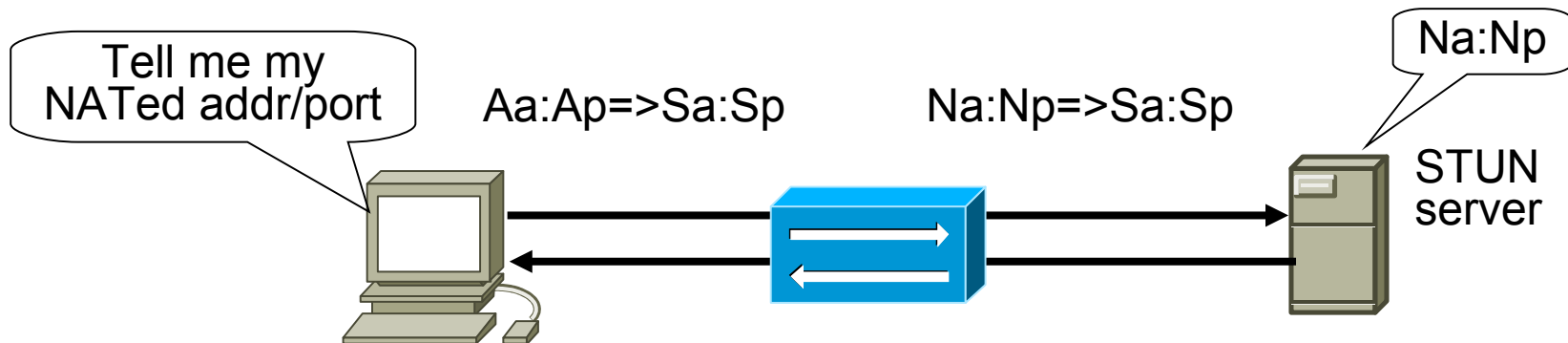
- ↳ STUN, TURN (relay)

- ↳ ICE (interactive connection establishment)

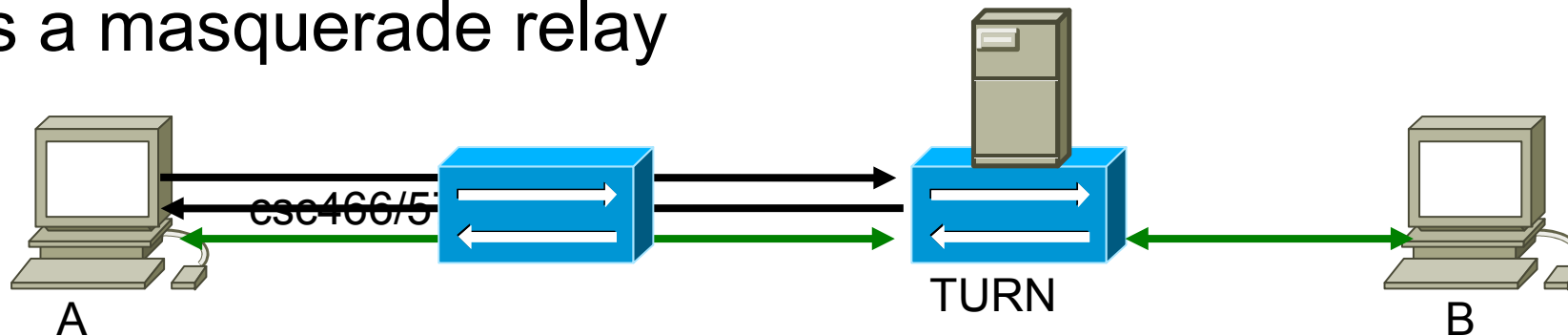


# STUN and TURN

- 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

# A case study

- 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 don't know much about Skype!

- ↳ Skype acquired by Microsoft in 2011

<http://www.cs.columbia.edu/~salman/skype/>