Administrative

MP3 deadline Saturday April 28, 5pm

- Demonstrations of MP3, April 30, 5-7pm

  - Groups should sign up as follows:
    - 5-6pm – 3rd floor 3401 SC for groups that work with laptops and need wireless connectivity for MP3
    - 6-7pm - 216 SC basement for groups who work with PCs and need wired connectivity for MP3

  - Top four groups will be decided Monday, April 30 at 7pm (via email, also posted on the newsgroup/classwebsite) - these groups will compete in front of the judges on Tuesday, May 1
Administrative

**Competition of final four groups** on Tuesday 5-7pm in 3401 SC/ 216 SC

- ByteMobile Inc. company – judging competition (and TA/Instructor)

- The top four groups should prepare 3-4 power-point slides to present
  - Intro Slide – name of your system and your names (1 slide)
  - Surveillance System Design – overall architecture (1 slide)
  - Features of Your System - interface (1 slide)
  - Features of Your System – other features (1 slide)
Administrative

- **Homework 2** is posted
  - Deadline May 2, Wednesday midnight 11:59pm

- **Peer Evaluations** – due **Friday, May 4, midnight**
  - Peer Evaluation Form and Explanation - available on the class website
  - Submit your Peer Evaluation to klara@illinois.edu
  - Note: if you do not submit your peer evaluations, you get 0 for self-evaluation and 100% for your group mates.

- **¼ Unit projects** – due **Friday, May 4** midnight (if you need more time, arrange deadline with instructor)
Final Exam

- May 11, 1:30-4:30pm in two rooms
  - ROOMS 1105 SC and 1109 SC
  - Students with last names A-J in room 1105 SC
  - Students with last names K-Z in room 1109 SC
  - More information on Wednesday about final exam format/review session
Summary: Video Streaming Approaches

- IP Multicast
- Content Distribution Networks
  - Expensive
  - Akamai, Limelight, etc
- Application Layer Multicast
  - Alternative to IP Multicast
- Peer-to-Peer Based
  - Scalable
  - No setup cost
  - Viable
Outline

- Voice over IP via Telecom IP Networks
- Peer-to-Peer Internet Voice Distribution
Voice over IP (VoIP)

- **VoIP** – transport of voice over IP-based networks
- **Complexity ranges from**
  - Hobbyists using Internet to get free phone calls on peer-to-peer basis to
  - Full scale PSTN (Public-Switched Telephone Network) replacement networks
- **VoIP must address**
  - Types of end user terminals - IP phones, PC clients
  - **Quality of Service** – ensure agreed quality
  - **Security risks** must be clearly identified
  - Last mile bandwidth – which affects codec, packetization period and where to use compression to best meet service goals
  - **Signaling protocol** must support service set required
Next Generation VoIP Network (MSF – Multi-service Switching Forum Example)

CS 414 - Spring 2011
MSF VoIP

- Access Services Signaling protocol and network service signaling protocol: SIP
  - Use RTP packets for telephony events
  - Transport DTMF (Dual-tone multi-frequency signaling) tones out of band using the signaling protocol such as SIP

- Quality of Service (Delay, Jitter, Packet loss)
  - Use RSVP, DiffServ, MPLS, even ATM
  - RTP is used for media traffic
Skype

Source: An Analysis of the Skype Peer-to-peer Internet Telephony Protocol, S. Baset, H. Schulzrinne, 2004
Rapid Identification of Skype Traffic Flows, P. Branch et al., NOSSDAV 2008
Skype Overview

- **Peer-to-peer** (P2P) overlay network for Voice-over-IP (VoIP) and other application
- Developed by Niklas Zennstrom and Janus Friis (founders of KaZaA, file-sharing company)
- Users see Skype as an Instant Messaging (IM) software
- Free on-net VoIP service and fee-based off-net SkypeOut service (allows calling to PSTN and mobile phones)
- Runs on Windows, Linux, Pocket PC, ...
Skype Network

- **Super Nodes**: Any node with a public IP address having sufficient CPU, memory and network bandwidth is candidate to become a super node.

- **Ordinary Host**: this host needs to connect to super node and must register itself with the Skype login server.
Components of Skype

- **Ports**
  - Skype client (SC) opens TCP and UDP listening port configured in its connection dialog box

- **Host Cache (HC)**
  - List of super node IP address and port pairs that SC builds and refreshes regularly
  - SC stores HC in the Windows registry

- **Codecs**
  - Wideband coded allowing frequencies between 50Hz-8KHz (one of the codecs is implemented by Global IP Sound)
Skype Ports on which Skype listens for incoming connections

This is the port that Skype uses to listen for incoming communications from other Skype users (assuming your firewall does not block them). In addition, Skype will send out outgoing UDP packets from this port. Note that this setting does not affect outgoing communications -- i.e. your ability to connect to other Skype users and the Skype network.
Skype Host Cache List
Components of Skype

- **Buddy List**
  - Skype stores buddy information in Windows registry
  - Buddy list is digitally signed and encrypted, local to machine and not on a central server

- **Encryption**
  - Skype uses 256-bit AES encryption
  - Skype uses 1536 to 2048bit RSA to negotiate symmetric AES keys

- **NAT and Firewall**
  - SC uses variations of the STUN and TURN protocols to determine type of NAT and firewall
Skype Architecture
STUN and TURN

- **STUN (Simple Traversal of UDP through NAT)**
  - Client-server protocol
- **TURN (Traversal Using Relay NAT)**
  - Increase latency and packet loss
Techniques used in Skype

- Firewall and NAT traversal
- Global decentralized user directory
- Intelligent routing
- Security
- Super-simple UI
Login

- During login process SC:
  - Authenticates its user name and password with login server
  - Advertises its presence to other peers and its buddies
  - Determines type of NAT and firewall it is behind
  - Discovers online Skype nodes with public IP addresses

- Login server is the only central component in Skype network
Skype Login Algorithm

Start

Send UDP packet(s) to HC IP address and port

Response within 5 seconds

Yes

No

TCP connection attempt with HC IP address and port

Connected

Yes

No

TCP connection attempt with HC IP address and port 80 (HTTP port)

Connected

Yes

No

TCP connection attempt with HC IP address and port 443 (HTTPS port)

Connected

Yes

No

Connection Attempts == 5

Yes

Failure

No

Wait for 6 seconds

Success
Skype Login Process

- After installation and first time startup, HC was observed empty

- Bootstrap super nodes:
  - After login for the first time after installation, HC was initialized with seven (IP, port) pairs

- Bootstrap (IP, port) information either
  - Hard coded in SC
  - Encrypted and not directly visible in Skype Windows registry, or
  - One-time process to contact bootstrap node
Skype Login Process

- First time Login Process
  - SC sends UDP packets to some bootstrap SNs
  - SC establishes TCP connection with bootstrap SNs that respond
  - SC perhaps acquires address of login server from SNs
  - SC establishes TCP connection with login server, exchanges authentication information

- Subsequent Login Process
  - Similar to first-time login process
  - SC uses login algorithm to determine at least one available peer and establishes TCP connection
  - HC was periodically updated with new peers’ (IP,port)
Skype Login Process

Comparison of three network setups
- Exp A: both Skype users with public IP address
- Exp B: one Skype user behind port-restricted NAT
- Exp C: both Skype users behind port-restricted NAT and UDP-restricted firewall

Message flows for first time login process
- Exp A and Exp B are roughly the same;
- Exp C only exchange info over TCP

<table>
<thead>
<tr>
<th></th>
<th>Total Data Exchanged</th>
<th>Login Process Time</th>
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</thead>
<tbody>
<tr>
<td>Exp A</td>
<td>About 9 KB</td>
<td>3~7 seconds</td>
</tr>
<tr>
<td>Exp B</td>
<td>About 10 KB</td>
<td>3~7 seconds</td>
</tr>
<tr>
<td>Exp C</td>
<td>About 8.5 KB</td>
<td>About 34 seconds</td>
</tr>
</tbody>
</table>
User Search

- Skype uses **Global Index** technology to search for a user.
- Skype claims that search is distributed and is guaranteed to find a user if it exists and has logged in during last 72 hours.
- Search results are observed to be cached at intermediate nodes.
Call Establishment and Teardown

- Call signaling is always carried over TCP
- For user not present in buddy list, call placement is equal to user search plus call signaling
- If caller is behind port-restricted NAT and callee is on public IP, signaling and media flow through an online Skype node which forwards signaling to callee over TCP and routes media over UDP
- If both users are behind port-restricted NAT and UDP-restricted firewall, both caller and callee SCs exchange signaling over TCP with another online Skype node, which also forwards media between caller and callee over TCP
Media Transfer and Codec

- Bandwidth usage
  - 3-16 Kbytes/s

- Skype allows peers to hold a call.
  - To ensure UDP binding, SC sends three UDP packets per second to the call peer on average

- No silence suppression is supported in Skype

- min. and max. audible frequencies Skype codecs allow to pass through are 50 Hz and 8000 Hz.

- Uplink and downlink bandwidth of 2KB/s each is necessary for reasonable call quality
Conferencing

- Node A acts as **mixer**, mixing its own packets with those of node B and sending to C and vice versa.
- For three party conference, Skype does **not do full mesh conferencing**.
- Most **powerful machine will be elected as conference host and mixer**.
- Two-way call: 36kb/s
- Three-way call: 54kb/s
Impact of Skype

- Impact on fixed-line operator
  - Skype will introduce SkypIN

- Impact on mobile phone operator
  - Skype will be embedded in Wi-Fi/mobile phone
  - WLAN is now limited by
    - Batter life
Impact of Skype

Skype has shown, at least has suggested, the following

- **Signaling**, the most unique property of traditional phone systems, can now be accomplished effortlessly with self-organizing P2P networks

- **P2P overlay networks can scale** up to handle large-scale connection-oriented real-time services such as voice
Conclusion

- Statistics from the paper 2004 -
- More than 2 million on-line subscribers per day
- More than 2.7 billion minutes served (minutes of free Skype-to-Skype callees)
- More than 38 million of software download
- More than 7 million of registered subscribers
- More than 1 million concurrently on-line subscribers