VoIP
Lab 4 - SIP Servers, Skype, and H.323

Goals

- Examine the operation of SIP phones with a public SIP server.
- Examine the signaling in Skype calls.
- Analyze the signaling in H.323.

Some Notes on the Equipment

- You have several soft-phones devices to work with:
  - Windows with “X-Lite”.
  - Windows and Mac OS with “sjphone”.
  - Windows and Mac OS versions of Skype

Steps to Complete

1. In the Options menu of sjphone on Windows 7 and the Mac, locate the tab for profiles. Make the SIP profile active and test that you can make a call as in lab 2.
2. Change the active profile on both systems to H.323. Make a call while capturing packets (either system is fine); examine the signaling exchange. Quit sjphone on Windows.
3. Configure X-Lite, and two sjphone instances to register with the Ekiga SIP server. Make sure you have a packet capture running when you start the registration process to capture the SIP registration. (You can always shut down the phone application and re-launch it to see the registration again).
   - In all cases your SIP proxy aka host aka server is “ekiga.net”.
   - You extension or SIP id and your account username are the same. You will receive the password in the lab.
   - In sjphone, you need to create a new profile with type “Calls through SIP proxy”. One system will use id “its441.first”, the other one will use id “8411411”.
   - For X-Lite, use id “its441.second”.
   Confirm that all three clients are registered.
4. Go to www.ekiga.net in a browser, and select “Ekiga.net Free VoIP Service”. The page lists test numbers, namely an echo test and a “call me back” test. Use these to confirm that your phones are correctly configured. These calls currently do not work on Windows XP. Use Wireshark to capture the incoming test call. Traceroute to the IP address that originated the call.
5. Make a call between two of your phones using the Ekiga.net sip addresses, while capturing packets on both systems. Examine the signaling and see if you can match up out-bound messages on one system with in-bound messages on the other system (remember the SIP triangle).
6. On one sjphone, edit the profile you created for ekiga, and select the DTMF tab. Set the DTMF method to SIP INFO. Make a call from that machine while capturing on both systems. While you are in the call, dial a few digits using the numbers on the keyboard; you should hear DTMF on the other side. Examine the packets to locate the SIP packets that signaled the DTMF.
7. For each of the calls below, make sure you capture packets on all systems while you experiment with the calls.

(a) Make a call between two of your stations; while the call is active, place the call on hold, make a second call, and switch between the two calls.

(b) Make a call between two stations, then attempt to transfer the call to the third.

(c) Make a call between two systems, put the call on hold, place a call to the third system. Put the calls into conference mode.

(d) Look at the instructions on www.ekiga.net for their conference service. Create a conference with two phones.

8. Shut down all soft-phone applications. Bring up Skype on two of your systems while running packet captures.

• On one system, use Skype ID “its441aaaa”.

• On the other one, use Skype ID “its441bbbb”.

Examine the packet captures for indications of how the registration was done.

9. Place a Skype call between the Mac and Windows while capturing packets. Skype uses a proprietary peer-based protocol. Examine the packets and see how much sense you can make of how the call completed.

10. Place a video call between the two systems. Use wireshark to determine the data rate used by the combined audio and video traffic.
Lab Report Guidelines

Each report is to be written individually, although the data for the lab is collected during the lab with your partner/group. Reports should be typed/word processed and brought to class in printed form.

Lab writeups are due in class on the Monday following the lab. They don’t generally need to be more than a few (several) pages. Officially, they need to be “long enough to answer the questions”. Each lab writeup must have a header on the first page that includes:

- Your name
- The lab section (i.e. the day) that you attended
- Your lab partner’s names

In Your Report

1. Describe the H.323 signaling process you observed.
2. Explain the SIP registration sequence you observed.
3. Show the signaling that took place for the inbound test call from the ekiga.net system. Identify the DNS name of the system the placed the call (or one near it in the traceroute).
4. Explain the DNS queries that took place prior to one of your calls through ekiga.net.
5. Show the signaling for the setup of a call through ekiga.net; attempt to draw the SIP triangle.
6. Show how the DTMF tone was transmitted in your experiment with DTMF.
7. Show the SIP signaling and the RTP flows for
   (a) the two-line call,
   (b) the transferred call (or attempt),
   (c) the local conference call (or attempt),
   (d) the ekiga.net hosted conference call.
8. Show the Skype registration process, explain as much of it as you can.
9. Show the Skype call setup process, explain as much of it as you can.
10. Show the data rate of the video Skype call.