SIP-based VoIP Lab

Step 1: Connect Your PC to The Network

- Get your laptop connected to the campus network (both WLAN or wireline will work).
  - Run ipconfig to show your own IP address
  - Ping NCNU-SIP.ipv6.club.tw (163.22.20.155)
  - Ping NCNU-GW1.ipv6.club.tw
- We interconnect via D-Link DE-1824 Hub.

IP Addresses We Obtained

- 300 – (Cisco 7960)
- 301 – 10.10.19.173
- 302 - 10.10.19.117
- 303 - 10.10.19.64
- 304 - 10.10.19.162
- 305 - 10.10.19.148
- 306 - 10.10.19.142
- 307 – 10.10.19.158
- 308 – 10.10.19.3
- 309 –

- 331 – (WiFi phone)
- 332 – (SENAO)

Step 2: Install Ethereal under Windows

- Install WinPcap 3.1.
  - WinPcap is an architecture for packet capture and network analysis for the Win32 platforms.
  - It includes
    - a kernel-level packet filter,
    - a low-level dynamic link library (packet.dll), and
    - a high-level and system-independent library (wpcap.dll, based on libpcap version 0.6.2)
- Install Ethereal 0.10.13.
  - It will automatically install WinPcap.

Where to Get Ethereal

- Official site:
  - http://www.ethereal.com/
- Local mirror:
  - http://Download.ipv6.club.tw/

Step 3: Install a SIP UA

- Windows Messenger 4.7 (4.7.2009)
  - Windows XP SP2 will remove the SIP support in Messengers
- X-Lite
- NBEN UA
- Cisco 7960 IP phone
- SENAO WiFi phone

Step 4: REGISTER

- Configure the proxy server to be
  - NCNU-SIP.ipv6.club.tw (163.22.20.155)
- Start your Ethereal
  - Disable promiscuous mode
  - Capture Filter: udp port 5060
  - Display the captured packets in real-time
  - OK
- Start Windows Messenger
- Sign-in and observe the packets you capture

- Ex 1:
  - Everybody sign-in and sign-off to see the SIP messages.
  - Sign-in
    - Expires: 3600
  - Sign-off
    - Expires: 0
- Ex 2:
  - Two UAs sign in with the same username.
    - 2 Bindings
  - One UA signs off
    - 1 Binding
  - The other UA signs off
    - 0 Bindings

- Now everybody restores to your original username.
- Keep Ethereal running.

Test Your Microphone & Speaker

Call Scenarios

- Practice using this tool to capture SIP signaling in the following call flows
  - REGISTER – 200 OK
  - INVITE – 200 OK - ACK
  - BYE – 200 OK
  - Hold/Retrieve

INVITE

- Pairing students into groups
- Let the student with lower username be the caller, and the other student be the callee. Make the call.
- Exchange the role, and make the call again.

1. Call instructor's Cisco 7960
   - 300@iPBX.ipv6.club.tw
2. Call NCNU PBX
   - 4161 (COM), 4131 (CSIE), 4101(EE)

- Compare the signaling flow of the above two cases

Try to capture RTP packets

- Analyze – Decode As – RTP

X-Lite

- INVITE/200 OK/ACK
- BYE/200 OK
- Hold/Retrive
- RTP

SIP Forking

- Ex 1:
  - A calls B, C.
  - B answers.
  - C receives a CANCEL request.
- Ex 2:
  - B calls A, C
  - Nobody answers.
  - C hangs up.

Contacts (using Messenger)

- Presence
  - SUBSCRIBE/NOTIFY
  - RFC 3265
- Instant Message
  - MESSAGE
  - RFC 3428