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Practical Exercises

Communication Systems (Rechnernetze II)

Topic 21: Asterisk

Exercise 1:
Asterisk is a popular open source implementation of software based private branch exchange(PBX). It acts as a central switch to many extensions using various interface such as SIP, Traditional landline and ISDN. It also includes many features available in proprietary PBX systems: voice mail, conference calling, interactive voice response (phone menus), and automatic call distribution.

Asterisk could be a very complex software if combined with different interfaces. However, due to limited amount of time in our exercise session, we shall only briefly demonstrate the basic function such as voice communication between two parties as well as voice mail.

We shall first set up our AsteriskNow server:

1. Ask the CD-ROM/USB Stick from the tutors.
2. Copy the VMWare image of AsteriskNow to the Desktop or somewhere you have root access. Due to performance consideration, please do not launch it from the Flash Drive/CD-ROM.
3. Load the AsteriskNow VMWare image.
4. Look into Devices tab and change the mode of "Network Adapter" to "Bridged".
5. Press the launch button and wait until the login prompt appears. Press "Quit" if a configuration screen is shown
6. Enter root/12345678 as login.
7. Our predefined IP address is 10.0.0.100/255.255.255.0, change your computer's IP address to the same subnet such as 10.0.0.100/255.255.255.0
8. Open using your browser in the host computer and connect to http://10.0.0.100/admin/ with freepbx/fpbx as login. Your should see a web-based interface namely "FreePBX", a popular GUI extension of Asterisk.
9. Your Asterisk server should be running now.

Exercise 2:
Creating extension and establish your first call

1. Most of the configuration files are automatically generated by FreePBX, editing the sip.conf is deprecated.

2. In the web-based interface, try to add two new extension, 1001 and 1002 respectively. You will need to fill at least the “User Extension” field and “secret” field. In this case “secret” means login password for SIP client.

3. Go back to your host computer and launch “Ekiga Softphone”. If not available, please download it using package manger.

4. The Ekiga will prompt you to register to their SIP server and their PC-to-Phone plan. Ignore them, go to the main windows and open Edit/Account.

5. Click “Add”. Type in your name in “Account Name” field. “10.0.0.100” in registrar field. 1001 in User field and their corresponding password in Password field. Click OK.

6. Check the checkbox to activate your account you’ve just added.

7. If everything goes well, you should notice that a “connected” message will be shown in the state bar.

8. Try to dial a random number and you should hear a voice prompt that tells you your number is not valid.

9. Find another computer and change the IP address of this computer into same subnet. Repeat the previous steps but change the extension number to 1002.

10. When both computer were connected, try to establish a call from 1001 to 1002. Pick up the phone and go back to the VMWare screen. You should see a lot of debug message there. You could also try to make a conversation if you have a microphone.

Exercise 3:
Enabling the Voice Mailbox.

1. Try to enable the Voice Mailbox features when you are creating new extension. See what happened if the person you are trying to reach does not answer the phone.