

Laboratory Block – Voice over Internet Protocol (VoIP)

Alexander A. Kist
Version 2.3

Friday 31st July, 2015

Voice over IP uses the Internet Protocol Stack to provide telephone services. This includes the transport of voice data packets but also call signalling used for call setup and termination. VoIP has been used by telcos for many years. Increasingly, it is also penetrating the consumer market. Through applications such as Skype and others, VoIP has received growing attention. Skype is a VoIP application; however it uses proprietary protocols and does not interoperate with the IETF VoIP framework.

This exercise will demonstrate a simple VoIP setup in a local network. If you are in the position to purchase or upgrade a company Private Automatic Branch Exchange (PABX), it is worthwhile to look into VoIP. A similar setup can also be used for home phones, providing a second line and/or allow national calls at local call rates.

Aims The aims of this exercise are:

1. To familiarise yourself with VoIP technology.
2. To install and configure the *Software PABX Asterisk* using the FreePBX web interface.
3. Configure and test, VoIP telephones as well as softphones (software telephones).
4. Investigate various configuration options.
5. Observe the VoIP network traffic.

Hardware The following equipment will be used for this lab block. Purchase prices are per device as of 2008. Similar equipment is available from other vendors as well.

IP Phone	Linksys	SPA901	\$66
IP Video Phone	Grandstream	GXV 3000	\$350
VoIP Adapter	Linksys	SPA3102	\$65

Besides this VoIP equipment, personal computers and a switch are also required.

Software The software that is used in this experiment is freely available on the Internet.

The activity uses the Asterisk PBX (Private Branch Exchange) (<http://www.asterisk.org/>) with the FreePBX GUI.

Assessment and Documentation

- Complete the pre-lab activities and the pre-lab quiz on StudyDesk.
- Maintain a lab book for this activity. Please see for this post for details: <http://alexanderkist.com/2015/07/30/keeping-a-lab-book/> On completion, upload your lab book on StudyDesk.
- At the end of the lab, each group needs to present a brief overview of what has been achieved. What did you learn during completing this activity? How is this relevant in broader context of your discipline?

Pre-Lab and Background Research

- How does the traditional telephone system work?
- What are key components and key functions?
- What is a PBX and what is a telephone exchange?
- What is signalling?
- What signalling protocols are used in conventional telephone systems?
- How does VoIP differ?
- What are the key components of a VoIP system?

1 Laboratory Setup and Network Configuration

Since VoIP relies on IP networking, it is important that the computer network is configured correctly.

Each desktop computer in this lab has two network interface cards. Both cards are connected to a configurable switch. The switch determines which computer interfaces are connected to which physical network. Port 1 of all switches are reserved for the uplink connection to the Internet (Switch 0). The switches are configurable via a web interface. The IP addresses of the switches are on the 172.16.0.0/24 subnet with addresses between 172.16.0.201 and 172.16.0.205.

The local gateway provides a DHCP service for the 172.16.0.0/24 subnet. The dynamic address pool ranges from 172.16.0.50 to 172.16.0.199. If the computer is connected to the gateway, it can obtain an IP address automatically and is able to access the Internet. HTTP proxies should be disabled in this case.

For this activity all computers and telephones should use the laboratory network 172.16.0.0/24. Therefore, all switch ports have to be located on the same subnetwork.

Activity – Switch Configuration

1. On a local system, enable DHCP. You should get an IP address on the 172.16.0.0/24 subnet.
2. Open a web browser and connect to your switch (172.16.0.20?)
3. Both user name and password are *supervisor*.
4. **Do not change IP address or user credentials.**
5. To configure the local networks, use the “Port VLAN” option.
6. Assign all ports to the first VLAN.
7. Disable the second PC interfaces at the switch.
8. All PCs should be able to obtain an IP address automatically and be able to access the Internet.

2 PBX Installation

To simplify the installation, this activity uses the AsteriskNOW Software PBX distribution. It is an ISO image that installs Linux, Asterisk and the FreePBX GUI in a single install. AsteriskNow will be installed in a virtual environment under VirtualBox.

Background Research

- What does virtualisation software do?
- What network configurations are commonly used for virtual interfaces?
- What configuration is appropriate for this activity?

Activity – Virtual Machine Configuration

1. Verify that VirtualBox is installed. If it is missing, download and install VirtualBox with the default options.
2. Download the *AsteriskNOW* DVD image from the file server.
3. Open *Oracle VM VirtualBox*.
4. Create a new virtual machine. A wizard will assist you. Click on *New*.
5. Name the virtual machine *VoIP*, select *Linux* and *Red Hat 64bit*.
6. Use 512MB of memory.
7. Create a New Virtual Disk of 8 GB and use the dynamic option.
8. Complete the Wizard.
9. In the *Details* tab, click on *Storage*. Under *IDE Controller* click on the CD symbol. Under *Attributes* select the *AsteriskNOW* image. You might have to add the image to the list.
10. In the *Details* tab, click on *Network*. Select *Bridged Adapter* under *Attached to* and chose the active network interface.

Hints

- Click on the VirtualBox window to activate the keyboard and mouse for the guest operating system; press the right CTRL key to get back to the host operating system.

Background Research

- What is Asterisk?
- What is AsteriskNow?
- On what distribution is it based?

Activity – AsteriskNOW installation

1. Find the *AsteriskNow Quick Start Guide* and use it!
2. Click on the green arrow to power up the virtual machine.
3. At the digium screen, type 1 and press enter to commence the installation.
4. ...
5. Enter the root password as *usqz220*
6. Start the installation. Once it is running, you can assist your peers with their installations.
7. ... 5 to 10 minutes later... Reboot!
8. Skip the configuration tools.
9. Note the server IP address.

3 Asterisk/FreePBX Configuration

Background Research

- What is a dial plan?
- How does the Australian telephone numbering plan work?

Activity – Add Extensions

1. Install Google Chrome on the host computer.
2. Use Chrome to open the FreePBX server web interface.
3. Click on *FreePBX Administration*.
4. Both username and password are *admin*
5. Click on *Applications* to add a new extension.
6. Select *Generic SIP Device* and click *Submit*.
7. Define a *User Extension*, a *Display Name*, and a *secret – ele3915*.
8. Submit the changes.
9. Define and submit a second extension.
10. *Submit* stores changes. To apply the changes to the server, click the red *Apply Config* button.

Activity – Asterisk CLI

- Explore the Asterisk *Command Line Interface* (CLI)! How is it accessed?
- Find appropriate instructions!
- How do you show which SIP phones are currently registered?
- Call *60 on a registered phone and watch the interface! Which tasks are executed?
- Call another extension! Which tasks are evolved in the call handling?

Background Research and Activities – Signalling

- What signalling protocol is used by the setup/phones in the lab?
- What are the key network elements of this protocol?
- Briefly explain how this protocol works!
- What does the registration process do?
- Why is it necessary?
- Which messages are involved in the registration process?
- Which messages are involved in the call setup process?

Activity – Traffic Analysis

- Setup a configuration that allows you to monitor the message exchange between the PBX and a phone!
- Capture the traffic for a phone registration and a complete call!
- What connections are involved? Which protocols and ports are used?
- Draw call flow diagrams for both the registration and call setup message exchange!
- Compare your diagrams to RFC3665. Which examples do they match?
- Playback the recorded telephone call!
- What is the average bitrate?

4 IP Phone Configuration (SPA901)

Consult the phone manual if necessary.

4.1 Setup using the Keypad

- Connect the IP phone to your local switch. Also, connect the power supply.
- Pick up the handset and dial * * * * to get to the configuration menu.
- Configure an appropriate IP address for your subnet by using the options given below:

Option	check	set	Parameters
DHCP	100#	101#	0 disabled, 1 enabled
IP address	110#	111#	
Netmask	120#	121#	
Gateway	130#	131#	
Primary DNS	160#	161#	
Reset (factory default)	73738 #		

- Now your telephone should have a valid IP address. Try to ping the phone from another PC on your subnet.

4.2 Setup using the Web Interface

Access the telephone configuration by browsing the IP address that has been set above.

- Click on *Admin Login*
- Go to the *Ext 1* tab
- Under the *Proxy and Registration* section, set *Proxy:* to your Asterisk server's IP address.
- In the *Subscriber Information* section enter the *Password* as *ele3915* and *User ID:* to the extension.

Note, these entries have to match the configuration for this extension that you configured earlier.

- Click *SubmitAllChanges*.
- The light on the phone should turn off.
- Dial *60 to test your setup.
- Configure a second phone and see if you can make a call.

5 Configuring Softphones

5.1 Windows

5.1.1 Express Talk

- Download the SJphone from the file server.
- Install the phone on a MS Windows desktop on your subnet.
- Configure the SIP server settings and disable STUN.
- Change the SIP proxy to the IP address of your PABX (asterisk) server - OK.
- Use the credentials, configured above, e.g. Account *extention* and Password *ele3915*.

6 VoIP Gateway Configuration

The SPA3102 is a gateway that connects standard, analog telephones or fax machines to IP based networks (FXS port). It has also a connector that allows the link to the legacy telephone system (FXO port). In this exercise, we use the telephone interface only.

- Configure the analog telephone port as a local extension.
- Consult the product documentation on how to achieve this.
- Integrate the extension in your dialplan.

7 Bridging Servers

Asterisk, has the ability to connect several servers together.

- Configure your extensions according to your group number, e.g. Group 1, uses extensions beginning with 1; group two, extensions begin with 2 etc. Use three digits for local extensions, e.g. 103, 403, 201.
- Connect the local Asterisk servers to each other. The aim is to be able to call all phones in the room from all phones.

References

- [1] C. R. M Spencer, M Allison. (2008) The asterisk handbook. [Online]. Available: www.digium.com/handbook-draft.pdf
- [2] (2008) voip-info.org - asterisk. [Online]. Available: <http://www.voip-info.org/wiki/index.php?page=Asterisk>
- [3] A. Powell. (2008) Getting started with asterisk. [Online]. Available: <http://www.automated.it/guidetoasterisk.htm>
- [4] P. Mahler, *Asterisk and IP Telephony*. Signate, LLC, 2004.
- [5] S. Wintermeyer. (2008) Practical asterisk 1.4. [Online]. Available: <http://www.the-asterisk-book.com/unstable/>

Voice over Internet Protocol (VoIP), also called IP telephony, is a method and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. The terms Internet telephony, broadband telephony, and broadband phone service specifically refer to the provisioning of communications services (voice, fax, SMS, voice-messaging) over the public Internet, rather than via the public switched telephone network (PSTN), also known as Voice over Internet Protocol (VoIP) is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms commonly associated with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, IP communications, and broadband phone service. Two kinds of competitors are competing in this space: one set is focused on VoIP for medium to large enterprises, while another is targeting the small-to-medium business (SMB) market.[6]. VoIP allows both voice and data communications to be run over a single network, which can significantly reduce infrastructure costs.[7]. VoIP basically works on top of Internet Protocol (IP). Like every data communicated over IP, a big chunk of VoIP related data is divided into smaller chunks called packets. Those packets are sent to the other end individually. Packets will be properly addressed so that the network can navigate it to the correct destination.