

How to set up a VoIP lab

(on a shoestring)

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Introduction

Have you been wondering about what sort of security vulnerabilities apply to the VoIP network that's coming up in your next assignment but have no equipment to test on yet?

Truth is that most of the times there is no need for a lot of expensive hardware to setup a basic lab for testing VoIP security. Of course some of the more interesting security threats apply to expensive proprietary hardware, but most of the times you can get away with making use of virtual machines and freely available software. In this tutorial I will be giving *step-by-step* instructions on how to setup a VoIP lab that can be used for testing security tools such as VOIPPACK.

Basic Requirements

- VMware Server (may also use other VM options)
- AsteriskNOW 1.0.2.1 <http://www.asterisknow.org/downloads>
- X-Lite <http://www.counterpath.net/X-Lite-Download.html>

After following this tutorial you will be able to:

- Install and setup a PBX on a VM
- Manager user accounts on the PBX
- Route calls through a 3rd party VoIP provider
- Make calls to landline numbers through the IP PBX

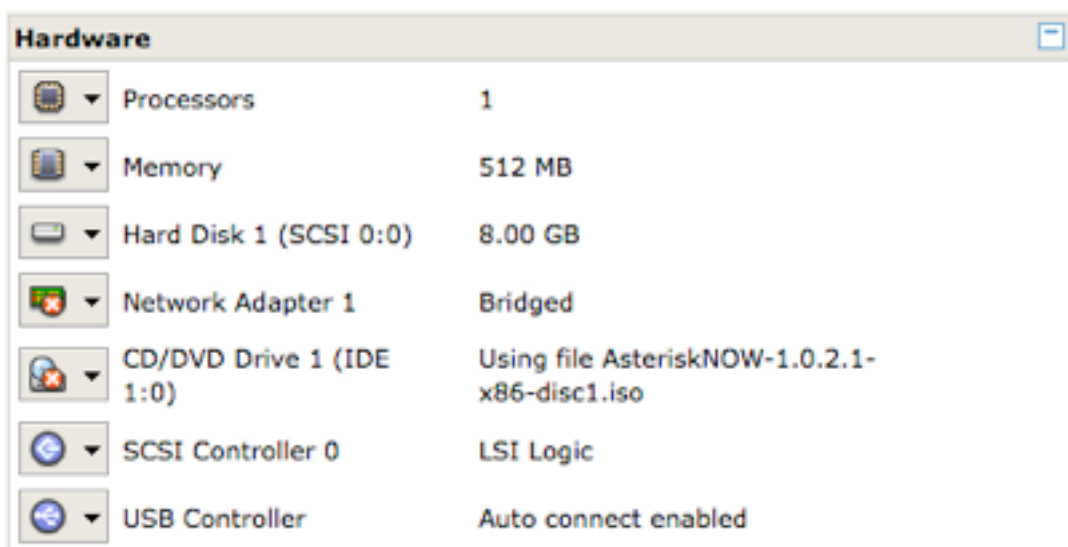
Installing AsteriskNOW on a VM

Preparation

We made use of the following default settings during creation of a new Virtual Machine in VMware server:

- Operating System: Linux, Version: Other 2.6x Linux (32-bit)
- Memory: 512 MB
- Virtual Hard Disk: created a new one of 8gb
- Network Adapter 1: Bridged
- CD/DVD Drive: Use an ISO image and selected the ISO image file downloaded from the AsteriskNOW download page

That's it, we're ready to start the installation. Here's a screenshot of how the virtual machine configuration looks like after following the above instructions:



Installation

The installation of AsteriskNOW is quite straightforward and even includes a GUI as you can see in the screenshot.



Steps during installation:

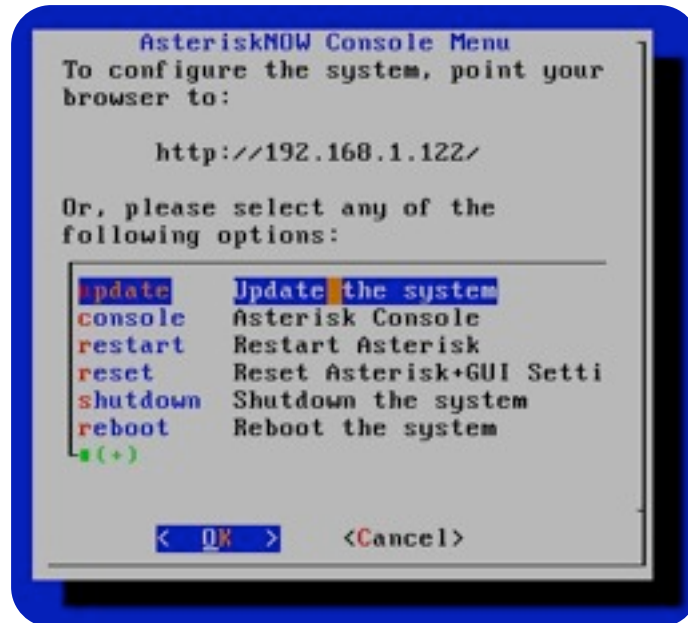
1. Installation Type: Express Installation
2. Automatic Partitioning: Remove all Linux partitions on this system
3. Network configuration: DHCP (default)
4. Administrator Password: remember this password

It will take a few minutes to install, so grab a coffee or your favorite drink. Once installed you will be asked to reboot.



Initial setup

After reboot the screen will show the IP address of the new AsteriskNOW box.



Browse to this address (192.168.1.122 in our case) by making use of your web browser and login using the user credentials that you set before. The username is “admin” and password is what you chose during installation. Once logged in, click on the “Users” button in the left side bar so that we create a new user extension. We set the following options and left the rest as set by default:

- Name: Joe Doe
- Password: 6000

Then click on the “Save” button at the bottom of the page and “Activate Changes” at the top.

Extension: 6000
Name: New User
Password:
VM Password: 1234
E-mail:
Caller ID:
Analog Phone:
Dial Plan:
Phone Serial:

Extension Options:

<input checked="" type="checkbox"/> Voicemail	<input type="checkbox"/> In Directory
<input checked="" type="checkbox"/> SIP	<input checked="" type="checkbox"/> IM
<input type="checkbox"/> CTI	<input checked="" type="checkbox"/> Is Agent
<input checked="" type="checkbox"/> Call Waiting	<input checked="" type="checkbox"/> 3-Way Calling
<input type="checkbox"/> Can Reinvite	<input type="checkbox"/> NAT

DTMFMode: rfc2833
Insecure:

Setting up your IP Phone

It is time to try out our new extension. Install X-Lite if you did not do so yet, and configure it by right clicking on the application (X-Lite) and selecting “SIP Account Settings”. Then click on the “add” button and set the following options in the “Account” tab:

- Display Name: Joe Doe (does not really matter)
- User name: 6000
- Password: 6000
- Authorization user name: 6000
- Domain: 192.168.1.122 (the IP address of the PBX)



X-lite should show up as “Ready, your name is 6000”. Make sure that the extension works by calling 7000 which by default is the “Voice Menu” system. You should hear an automated voice asking you to dial an extension.

Further configuration

PBX configuration

Accounts

Set up a few more accounts:

- Without a password

- With a pattern password such as 1212
- With a dictionary word

Setting up toll free calling

1. Select the user that you created before and select a dialplan “DialPlan1”. Remember to save and activate the changes

Extension: 6000
 Name: Joe Doe
 Password:
 VM Password: 1234
 E-mail:
 Caller ID:
 Analog Phone:
 Dial Plan: DialPlan1
 Phone Serial:
 Extension Options: _____

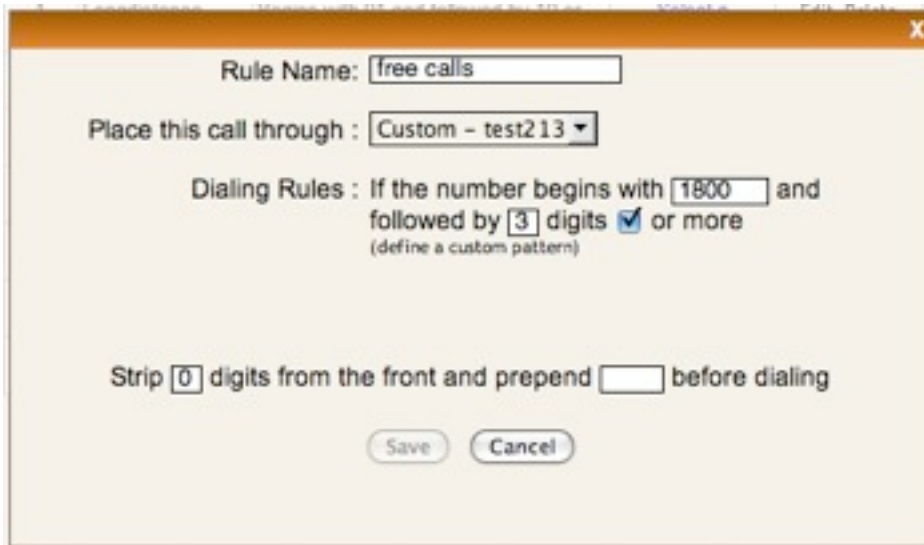
2. From the AsteriskNOW configuration click on the “Service Providers” and then “Add Service Provider”.
3. Select “Custom VoIP” and the fill in the following fields:
 1. Comment: anything
 2. Protocol: SIP
 3. Host: sip.tollfreegateway.com
 4. Username: anything
5. Activate changes

Edit Service Provider

Provider Type:
 Analog VoIP Custom VoIP

Comment: test213
 Protocol: SIP
 Register:
 Host: sip.tollfreegateway.com
 Username: user
 Password:
 Save Cancel

4. Finally create a call rule by clicking on the call rules button on the left bar and “add a calling rule” button. Use the following details:
 1. Rule name: free calls
 2. Place this call through: choose the provider that we just added
 3. If the number begins with: 1800
 4. Followed by 3 digits
 5. Select the “or more” option



5. Activate the changes
6. From X-Lite call the listing directory on 18005551212 - you should hear an automated voice.



Additional options to consider

You may choose to install more PBX servers for your lab. Here's a short list of available (some might be free others trial) and easy to setup PBX servers and softphones.

Other Asterisk-based PBX distributions

- Trixbox (<http://www.trixbox.org/>)
- PBX in a Flash (<http://pbxinaflash.net/>)
- Elastix (<http://www.elastix.org/>)

Other PBX servers available for download

- 3CX (<http://www.3cx.com>)
- pbxnsip
- Brekeke (<http://www.brekeke.com/>)
- CommuniGate

Other Software IP Phones

- SJPhone
- Zoiper (supports IAX2)
- Ekiga
- Gizmo5 (Skype competitor that makes use of SIP)

Conclusion

Hope that you enjoyed this tutorial and that at least it motivated you enough to consider setting up a VoIP lab. If you have any feedback or additional content please email me sandro@enablesecurity.com.

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