Learning SIP with Asterisk

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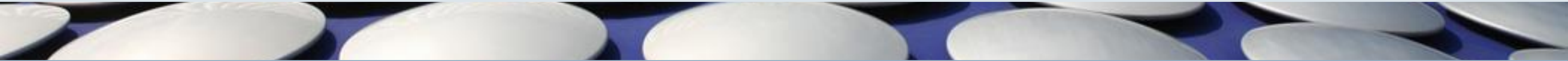

Install Mint 17.1 ‘Rebecca’

- **Step 0: Why Mint 17.1 ‘Rebecca’ Cinnamon?**
  - Asterisk requires a Linux platform on which to run on. New to Linux? Try Mint v. 17.1 ‘Rebecca’ – Cinnamon Desktop edition
  - Cinnamon is a beautiful & user friendly GUI environment
    - Available @ http://www.linuxmint.com/download.php
      - The hardest decision you’ll have to make is do you need the 32-bit download, or 64-bit download. This depends on your PC architecture, so I can’t tell you. Generally, newer machines are all 64-bit. If you’re unsure– experiment! The worse thing that can happen is it doesn’t install
  - Mint 17.1 is new as of 01/2015 and has Long Term Support (LTS) & it is an Ubuntu fork (so you can use the same CLI commands a you would in an Ubuntu or Debian environment)
- **Step 1: Install Mint 17.1 ‘Rebecca’ Cinnamon (2 options)**
  - **Option 1**
    - Download “Desktop – CD”, then burn to CD or DVD (*.iso)
    - Install Mint 17 LTS onto an old machine by booting from the CD or DVD (you’ll likely loose data on that drive unless you understand partitioning or dual booting
  - **Option 2**
    - Install to a virtual machine (such as VMWare Player)
    - Available @ http://www.vmware.com (you’ll need to make an account, then download VMWare player for home use… it’s free)
    - VMWare player is awesome you can have a lot of fun playing with, and learning about VMs
  - During install process, accept all defaults until you’re a Linux pro!
  - All you’ll need to supply is a host name, user name, and password to associate with that account
- **Step 2: Updating Mint 17.1 ‘Rebecca’ Cinnamon**
  - Open a command-line terminal (CTRL + ALT + T) then type:
    - `sudo apt-get update && time sudo apt-get dist-upgrade`
    - Default “Y” to all questions
- **Step 3: Installing Asterisk**
  - To install Asterisk, type:
    - `sudo apt-get install asterisk`
    - Default “Y” to any questions

Author’s Note: This is all you have to do in order to build an Asterisk test environment on a Linux server, so don’t be afraid to try new things. Worst case scenario is you have to reinstall Mint. The Linux and Asterisk worlds are massive, but very well documented by the open source community.

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Connecting to the Asterisk CLI

- **Step 4: Getting started with Asterisk**
  - Asterisk typically begins on Mint startup, but let's check to find out, in a command terminal type:
    - `ps aux | grep asterisk`
      - You'll always see one result (PID of your search!), but do you see a PID describing Asterisk?
  - Connect to the Asterisk Command Line Interface (CLI)
    - Step A: Start Asterisk (only if Asterisk isn't running)
      - `sudo asterisk`
    - Step B: Asterisk is running, connect to the Asterisk CLI in verbose mode
      - `sudo asterisk -rvvvvv`
  - After you are connected to the Asterisk CLI, leave this terminal open…
  - Use this screen to monitor Asterisk's behavior

- Common Asterisk CLI commands:
  - `core stop now`
    - stops Asterisk immediately
  - `core restart now`
    - restarts Asterisk immediately
  - `exit`
    - exits the Asterisk CLI
  - `reload`
    - reload all *.conf files – enter this command anytime you edit and save a *.conf file
  - `help`
    - list all Asterisk commands

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Configuration of Asterisk

- **Step 5: Configuring Asterisk (*.conf files)**
  - Asterisk is configured via a series of *.conf files found in /etc/asterisk/
  - A minimum of two *.conf files need to be configured (sip.conf and extensions.conf)
  - Editing /etc/asterisk/sip.conf (SIP settings file)
    - Open a new Linux CLI (CTRL + ALT + T) & type:
      - `gksudo gedit /etc/asterisk/sip.conf`
    - Asterisk installs with a working example, replace it with the version found on your instructor's website (copy and paste):
      - http://rzfeeser.com/2012/10/29/a-basic-asterisk-v1-8-sip-conf-configuration/
    - Review this new configuration, then save and exit
  - Editing /etc/asterisk/extensions.conf (dialplan settings file)
    - Open a new Linux CLI (CTRL + ALT + T)
      - `gksudo gedit /etc/asterisk/extensions.conf`
    - Asterisk installs with a working example, replace it with the version found on your instructor's website (copy and paste):
      - http://rzfeeser.com/2013/06/19/a-basic-asterisk-v1-8-extensions-conf-configuration/
    - Review this new configuration, then save and exit

- **Step 6: Restart Asterisk**
  - Connect to the Asterisk CLI (see step 4), then type:
    - `reload`
    - This command will make Asterisk reload all of its *.conf files

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Registering SIP User Agents to Asterisk

- **Step 7: Preparing to register a SIP UA to Asterisk**
  - If you examine sip.conf you'll find that Asterisk is expecting registrations from 3 extensions, they are:
    - Extension 1
      - UserID: 401
      - Password: secretpass401
    - Extension 2
      - UserID: 402
      - Password: secretpass402
    - Extens 3
      - UserID: 403
      - Password: secretpass403
  - Asterisk is currently configured to listen for SIP traffic on the IP address of Mint 17.1, we'll need that
    - Open a new Linux CLI (CTRL + ALT + T) & type:
      - `ifconfig`

- **Step 8 : Download a SIP Phone to a smart device**
  - **FREE** SIP Softphones for Smartdevices
    - Media5 Fone (iTunes & Google Play)
    - cSIPsimple (Google Play)
    - LinPhone (iTunes & Google Play)
    - Zoiper (iTunes & Google Play)
  - You can configure any of these clients to register to Asterisk, *but only when you are on your home wi-fi*
  - Basic UA config rundown (using the info collected in step 7)
    - Enter the correct UserID
    - Enter the correct Password
    - Set to 'domain' or 'server' or 'proxy' to the IP address of Mint

Author's Note: Try out many of the SIP UAs available, but be advised that many of the 'free' UAs found in the Google & Apple store will only allow you to register to a hosted SIP registrar / proxy they host (which costs money).

Install Wireshark

- **Step 9: Install Wireshark**
  - You won't learn anything unless you can see the SIP messaging
    - Open a new Linux CLI (CTRL + ALT + T) & type:
      - `sudo apt-get install wireshark`
  - After it is installed, you can launch Wireshark from the Mint GUI or type:
    - `sudo wireshark`

- Making a capture
  - Capture on the 'any' interface
  - Filter on `sip` (lower case)
    - Don't forget to press the "Apply" button

Beyond this document…

- **Step 10: Asterisk is stable and running**
  - It now might be time to get a SIP trunk provider. Find one you like, but before you buy from them, call/email and ask if they support Asterisk.
    - They should provide you with the copy/paste config for sip.conf and extensions.conf
    - SIP Trunking basic security notes
      - Do NOT 'link' a credit card to your account… ever! If your system is compromised, you may get stuck with a large credit card bill
      - Turn off international dialing with your SIP trunk provider until you are certain your system is secure
      - Watch the Asterisk log file for malicious SIP behaviors
        - `/var/logs/asterisk/message`

- Check around the internet, try to find a deal on some nice SIP desk phones
  - Grandstream – Cheap robust SIP UA
  - Linksys IP 941 – Reliable SIP UA
  - SNOM – Expensive and nice, often found in offices or call centers
- **Step 11: Other fun things to try…**
  - SIP lab with Raspberry-Pi (alternative)
    - Requires a working Rpi, SD card with the RPi Asterisk project installed
      - http://www.raspberry-asterisk.org/
    - Instructions to install a RPi project to SD card (follow NOOBs guide)
      - http://elinux.org/RPi_Easy_SD_Card_Setup

Final thoughts

- **Solutions for Traversing the NAT (ingress)**
  - At some point you might want to be able to register from outside of your home network
  - Easy solutions – both rely on DDNS & port-forwarding
    - Solution 1: DD-WRT enabled router with DDNS
      - Do your own research, but a great router to start with flashing is the Asus N-16 Router
        - The DD-WRT Wiki (study this before flashing) http://www.dd-wrt.com/wiki/index.php/Main_Page
        - Read the install instructions associated with the DD-WRT installs; possible to brick your router if you're not careful
    - Solution 2: Buy a new 802.11 AC routers with DDNS as a supported feature
    - Both of these solutions offer you a free FQDN that will resolve to the IP of your router's public interface. From your router you can then port forward to your Asterisk server inside your NAT
  - Alterative solution
    - VPN into your own network (don't port forward). Configuring a VPN server beyond the scope of this document, but VPN is a free service offered by the DD-WRT project

- **Further learning**
  - Asterisk has the benefit of a large community – Explore it! (with Google)
  - Your first goal should be understanding and rewriting the sip.conf and extensions.conf configuration files
  - Check out RZFeeser's YouTube account!
    - SIP Training 486 vs. 603 - https://www.youtube.com/watch?v=ZfwhHoGZ1a0
  - Check out Alta3 Research's YouTube account!
    - Asterisk Architecture (part 1) - https://www.youtube.com/watch?v=bgAX3HS-YE4

Good Luck!

Stuck? I mean, really, really, *REALLY* stuck? Don't get discouraged, just shoot an email to **z@rzfeeser.com**

- Extensions REGISTER to Asterisk
- Asterisk REGISTER to SIP trunk provider