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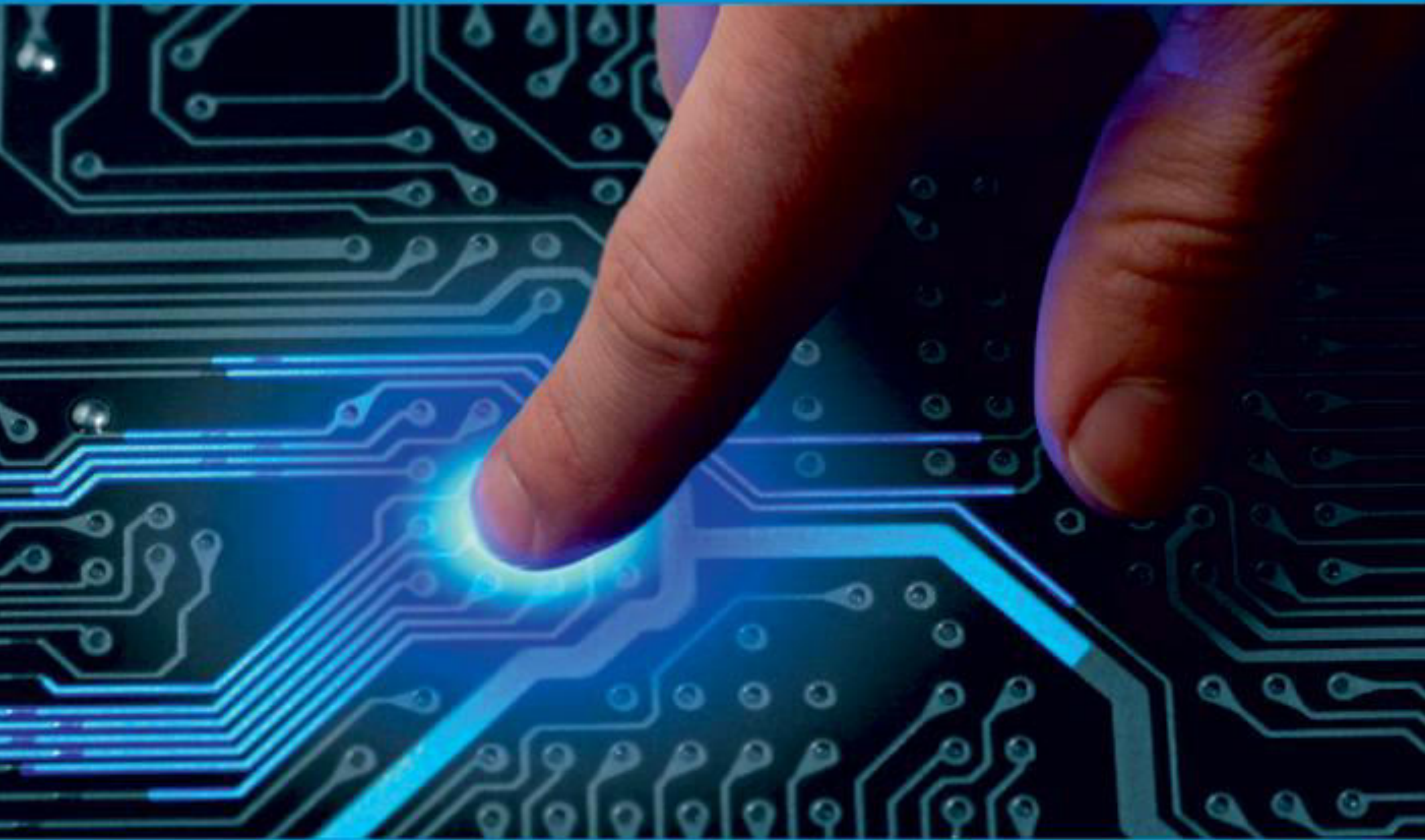

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Deploying and Configuring a VoIP Solution with Asterisk: A Comprehensive Guide

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ABSTRACT: This paper provides a comprehensive guide to the installation and configuration of a VoIP (Voice over IP) solution using the Asterisk server. Asterisk, an open-source telephony software, is introduced along with its core features and capabilities, such as handling multiple communication protocols (SIP, H.323, and IAX) to create a flexible telecommunication infrastructure.

The guide details the setup of a virtualized environment using Oracle VM VirtualBox, followed by the step-by-step process of installing Asterisk on a Linux-based operating system. It outlines the necessary dependencies, compilation, and installation of Asterisk, along with the configuration of key components like user accounts (SIP clients), extensions, voicemail, and conference rooms.

In addition, the document covers the configuration of a softphone (Linphone) as a SIP client, providing a practical example of managing SIP accounts and initiating voice calls through Asterisk. The guide also touches on setting up a voicemail system with email notifications using Postfix as the SMTP server.

Finally, this work serves as a foundational resource for anyone looking to implement a basic VoIP system using open-source tools.

KEYWORDS: Asterisk, VoIP, SIP, Virtualization, Softphone, Voicemail, Postfix, SMTP, VoIP Security, Dialplan.

I. INTRODUCTION

Asterisk is a project that started in 1999 by engineer Mark Spencer. It is open-source software released under the GPL (GNU General Public License) that transforms a computer or server into a private branch exchange (PBX). The reliability, performance, and interoperability of Asterisk have made it the most widely used open-source IP PBX globally. Asterisk provides all the functionalities of a telephone system for a business through various protocols such as H.323, SIP, and IAX.

In this document, we will show the steps to follow for installing and configuring our Asterisk server on the Linux operating system, as well as the installation and configuration of the communication tool Softphone.

II. DEPLOYING A VIRTUAL VOIP INFRASTRUCTURE

We will prepare our working environment using virtualization technology, which allows us to create multiple simulated environments or dedicated resources from a single physical system.

We need to install software called a hypervisor on the physical machine, which allows us to partition the host machine into several virtual machines.

In our case, we will install the open-source virtualization software published by Oracle, called Oracle VM VirtualBox, on the physical machine.

```
# apt-get install virtualbox
```

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Once VirtualBox is installed, we need to create three virtual machines:

- A machine called "Desktop," on which we will install Ubuntu 14 as the operating system and use it as a SIP client.
- A machine called "Hack," on which we will install Kali Linux, the GNU/Linux distribution based on Debian, as the operating system and use it to carry out attacks against the VoIP server.
- A third machine called "SIP Client," on which we will install Ubuntu 14 as the operating system and use it as a SIP client.

The next step is to create a virtual local network in VirtualBox by selecting the File menu and then "Settings." We will then go to the "Network" tab and manage our private networks. We will select our network card "vboxnet0" and then click on the wrench icon to modify its settings.

This is the IP address that we will have on our network card on the host machine.
Next, we can also modify the "DHCP Server" tab, which allows us to define the IP addresses that the virtual machines will have. For the virtual machines to communicate with the host and each other, the host and the IPs distributed by DHCP must be on the same IP range.

The final step is to change the "Promiscuous Mode" on the network card of each virtual machine to "Allow All" so that the guest operating system receives all the traffic observed on the network.

If all the previous steps are followed, the created virtual machines will be able to communicate with each other and with the host machine, on which we need to install the Asterisk server.

III. INSTALLATION OF ASTERISK

Before installing Asterisk, we need to update our Linux distribution and install the necessary dependencies for compiling Asterisk.

III.1 Updating the Linux Distribution

Running the apt-get update command regularly allows us to keep the list of available packages in the APT repositories up to date, as listed in the configuration file /etc/apt/sources.list.

The apt-get upgrade command updates all installed packages on the system to the latest available versions.
apt-get update &&apt-get upgrade

III.2 Installing Dependencies

To ensure the successful installation of the Asterisk server, the necessary prerequisites are presented in the following table, along with the package name and the command for installation:

Package Name	Installation Command	Description
build-essential	apt-get install build-essential	This package contains a list of packages necessary for building Debian packages.
libxml2-dev	apt-get install libxml2-dev	This package is required for development using the XML library.
libncurses5-dev	apt-get install libncurses5-dev	Development libraries for ncurses5.
linux-headers	apt-get install linux-headers	Linux kernel packages needed for the Asterisk installation.
libsqlite3-dev	apt-get install libsqlite3-dev	SQLite is a C library that allows linked programs to access SQL databases.
libssl-dev	apt-get install libssl-dev	This package is needed for the SSL project implementation of cryptographic protocols (SSL and TLS) to communicate securely on the internet.
uuid-dev	apt-get install uuid-	This package provides the development environment for the UUID library.

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	dev	
libjasson-dev	apt-get install libjasson-dev	C language library needed for encoding and decoding JSON data.

III.3 Downloading the Source Code

First, we will create a folder where we will place the Asterisk source files, as follows:
mkdir /usr/src/asterisk
cd /usr/src/asterisk
Next, we will retrieve Asterisk version 14.7.6 from the official website using the following command:
wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-14.7.6.tar.gz

III.4 Extracting the Packages

The downloaded packages are compressed archives that contain the source code. We will extract and then compile them using the following command:
tar -xzvf asterisk-14.7.6.tar.gz
cd asterisk-14.7.6

III.5 Compilation and Installation

To compile and install Asterisk, we will execute the following commands:
cd /asterisk-14.7.6
make clean
./configure
make
make install
make samples
make config

Once Asterisk is installed, we can start the server and connect to the Command Line Interface (CLI) with the following command:
asterisk –vvvc

Fig 1. CLI Console

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IV. CONFIGURATION OF THE ASTERISK SERVER

IV.1 Asterisk Configuration Files

Asterisk files are distributed across several directories to follow the conventional organization of Linux systems [2]. The directory containing the binary executables of the Asterisk server and its main components is located at /usr/bin/. The main directories are:[3]

- **/etc/asterisk/**: Contains all configuration files.
- **/usr/sbin/**: Contains the binary file of the main program.
- **/usr/lib/asterisk/**: Contains binary files that Asterisk uses to function.
- **/usr/lib/asterisk/modules/**: Contains modules for applications, codecs, and drivers.
- **/var/lib/asterisk/sounds/**: Contains audio files used by Asterisk, such as voicemail prompts.
- **/var/run/asterisk.pid**: File containing the current Asterisk process ID.
- **/var/spool/asterisk/outgoing/**: Contains outgoing calls from Asterisk.

The main configuration files are:

- **sip.conf**: Defines SIP protocol users and their options.
- **voicemail.conf**: Configures users' voicemail settings.
- **extension.conf**: Defines the DialPlan, which routes calls through the server.
- **meetme.conf**: Configures conference rooms.
- **rtp.conf**: Configures UDP ports for the RTP (Real-Time Protocol).
- **asterisk.conf**: Defines certain variables for Asterisk usage.
- **iax.conf**: Configures clients using the IAX (Inter-Asterisk Exchange) signaling protocol for VoIP conversations.

Now that our Asterisk platform is correctly installed, we move on to configuring client accounts and extensions.

a. Configuring Client Accounts

Our work is based on the SIP signaling protocol. The sip.conf file is used to define all SIP users.[4] It is divided into several sections, each beginning with a label in brackets. The [general] section defines parameters used by all users (default port number, call forwarding, call hold, etc.), and the following sections define account parameters (username, password, caller ID, etc.)[6].

Here is the configuration for the four clients we created in the file:

```
  GNU nano 2.2.6          Fichier : /etc/asterisk/sip.conf                Modifié

[general]
hasvoicemail = yes
hassip = yes
callwaiting = yes
threewaycalling = yes
callwaitingcallerid = yes
transfer = yes
canpark = yes
cancallforward = yes
callreturn = yes
callgroup = 1
pickupgroup = 1
nat = no
tcpenable=yes
tcpbindaddr=0.0.0.0
port=5060
```

Fig 2. General Configuration in sip.conf

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```
GNU nano 2.2.6              Fichier : /etc/asterisk/sip.conf                    Modifié

[ahmed]
type=friend                     ;Indique le type de compte, et les restrictions associées
host=dynamic                    ;Spécifie une adresse IP fixe ou dynamique fornit*
                                par un serveur DHCP à partir de laquelle l'utilisateur peut accéder à son compte
dtmfmode=rfc2833                ;Le type de tonalité DTMF appliqué
allow=ulaw                      ;Liste les codecs autorisés par l'utilisateur de ce compte
;disallow=all                   ;Interdit les codecs qui sont mentionnés à sa suite
username=Ahmed                  ;Identifiant de l'utilisateur
secret=secret                   ;Mot de passe associé au compte
callerid="LAOUITI Ahmed" <6001> ;Nom de l'utilisateur suivi de son extension téléphonique (son numéro d'appel)
contex=work                     ;Spécifie le type de routage à appliquer pour l'utilisateur
language=fr                     ;Spécifie la langue utilisée pour les fichiers audio
mailbox=601@default             ;Indique la boite vocale associée au compte
transport=tcp,udp               ;Le protocole de transport utilisé par l'equipement
```

Fig 3. Configuration of Client 1

```
GNU nano 2.2.6              Fichier : /etc/asterisk/sip.conf                    Modifié

[moez]
type=friend                     ;Indique le type de compte, et les restrictions associées
host=dynamic                    ;Spécifie une adresse IP fixe ou dynamique fornit
                                par un serveur DHCP à partir de laquelle l'utilisateur peut accéder à son compte
dtmfmode=rfc2833                ;Le type de tonalité DTMF appliqué
allow=ulaw                      ;Liste les codecs autorisés par l'utilisateur de ce compte
;disallow=all                   ;Interdit les codecs qui sont mentionnés à sa suite
username=Moez                   ;Identifiant de l'utilisateur
secret=secret                   ;Mot de passe associé au compte
callerid="NABLI Moez" <6002>    ;Nom de l'utilisateur suivi de son extension téléphonique (son numéro d'appel)
context=work                    ;Spécifie le type de routage à appliquer pour l'utilisateur
language=fr                     ;Spécifie la langue utilisée pour les fichiers audio
mailbox=602@default             ;Indique la boite vocale associée au compte
transport=tcp,udp
```

Fig 4. Configuration of Client 2

```
GNU nano 2.2.6              Fichier : /etc/asterisk/sip.conf                    Modifié

[salon]
type=friend                     ;Indique le type de compte, et les restrictions associées
host=dynamic                    ;Spécifie une adresse IP fixe ou dynamique fornit
                                par un serveur DHCP à partir de laquelle l'utilisateur peut accéder à son comp$
dtmfmode=rfc2833                 ;Le type de tonalité DTMF appliqué
allow=ulaw                       ;Liste les codecs autorisés par l'utilisateur de ce compte
;disallow=all                    ;Interdit les codecs qui sont mentionnés à sa suite
username=Salon                   ;Identifiant de l'utilisateur
secret=salon_password            ;Mot de passe associé au compte
callerid="Salon Commercial" <6003> ;Nom de l'utilisateur suivi de son extension téléphonique (son numéro d'appel)
context=maison                   ;Spécifie le type de routage à appliquer pour l'utilisateur
language=fr                      ;Spécifie la langue utilisée pour les fichiers audio
mailbox=603@default              ;Indique la boite vocale associée au compte
transport=tcp,udp
```

Fig 5. Configuration of Client 3

```
GNU nano 2.2.6              Fichier : /etc/asterisk/sip.conf                    Modifié

[bureau]
type=friend                     ;Indique le type de compte, et les restrictions associées
host=dynamic                    ;Spécifie une adresse IP dynamique fornit par un serveur DHCP
                                ou fixe à partir de laquelle l'utilisateur peut accéder à son compte
dtmfmode=rfc2833                 ;Le type de tonalité DTMF appliqué
allow=ulaw                       ;Liste les codecs autorisés par l'utilisateur de ce compte
;disallow=all                    ;Interdit les codecs qui sont mentionnés à sa suite
username=Bureau                  ;Identifiant de l'utilisateur
secret=bureau_password           ;Mot de passe associé au compte
callerid="Bureau Reception" <6004> ;Nom de l'utilisateur suivi de son extension téléphonique (son numéro d'appel)
context=maison                   ;Spécifie le type de routage à appliquer pour l'utilisateur
language=fr                      ;Spécifie la langue utilisée pour les fichiers audio
mailbox=604@default              ;Indique la boite vocale associée au compte
transport=tcp,udp
```

Fig 6. Configuration of Client 4

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b. Configuring Extensions (DialPlan)
Once the user accounts and terminals are defined, they need to be assigned phone numbers to be reachable. To configure an extension, use the following command:[5]
nano /etc/asterisk/extensions.conf

Assigning a Phone Number to a User
Multiple options can be used for a single number based on priorities or execution order. For example, if the caller dials 6003, they are connected to the terminal with that number. The Asterisk server answers, announces the date and time, waits for 3 seconds, and then terminal 6003 rings for 20 seconds. If there is no response, the caller is redirected to the voicemail associated with the number, where they can leave a message or hang up.

```
GNU nano 2.2.6          Fichier : /etc/asterisk/extensions.conf          Modifié

[maison]
exten => 6003,1,Answer
exten => 6003,2,SayUnixTime(,CET,AdbY \'digits/at\' kM)
exten => 6003,3,Wait(3)
exten => 6003,4,Goto(maison,6003,5)
exten => 6003,5,Dial(SIP/salon,20)
exten => 6003,6,VoiceMail(603@default)
```

Fig 7. Extension Configuration

Managing Voicemail
To manage voicemail, assign a dedicated phone number for voicemail access. In the following example, dialing extension 03 directly redirects the caller to voicemail. The VoiceMailMain() function handles the voicemail service, asking for the user's password and allowing them to listen to and manage their audio messages.

Inviting Participants to a Conference
To invite participants to join a conference room, redirect them using the DialPlan and the MeetMe application. To route a call to a conference, use the following in the extensions.conf file:
Meetme(100)
In our case, we need to create conference rooms and configure them in the meetme.conf file.

c. Configuring Voicemail
The voicemail service involves assigning a voicemail box number to a user. The audio messaging service is configured via three files. In our case, we modify the configuration using the following command:
nano /etc/asterisk/voicemail.conf
In the voicemail.conf file, we can configure email notifications for each new message received in the voicemail. Here are the variables used in the subject and body of the emails:

- VM_NAME: Username
- VM_DUR: Duration of the message
- VM_MSGNUM: Message number
- VM_MAILBOX: User's mailbox number
- VM_CIDNUM: Caller’s number
- VM_CIDNAME: Caller’s name
- VM_DATE: Date of the message
- \n: Newline
- \t: Tab

To allow our Asterisk server to send emails to users, we need to install an SMTP server. We will install Postfix using the following command:
apt-get install postfix
Next, we configure the SMTP relay by specifying the SMTP server address. We run the following command:

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nano /etc/postfix/main.cf
In this case, we will use Gmail's server to send emails. To do this, we add the following lines to the /etc/postfix/main.cf file:
relayhost = [smtp.gmail.com]:587

Now, we need to create the gmail_passwd file to store the authentication data.Then, we assign restrictive permissions to this file to limit unauthorized access using the following commands:
chmod 600 /etc/postfix/gmail_passwd

The final step is to restart the Postfix service to apply the changes:
systemctl restart postfix

IV.2 Starting Asterisk in Server Mode

Asterisk can be launched in server mode in two ways: either automatically at system startup or manually during the current session. For manual startup, use the following command:
asterisk -vvvvvc
The vvvv option increases the verbosity level of informative messages about the server's operations, and c indicates that the Asterisk control console is enabled. The prompt usually appears as *CLI> (for Command Line Interface).
Here is a list of commonly used commands:

Command	Description
show codecs	Displays codec numbers.
show conferences	Displays conference status.
show dialplan	Shows the different contexts and extensions.
show voicemail users	Displays voicemail users.
core restart now	Restarts Asterisk immediately.
core restart gracefully	Restarts Asterisk once no new calls are received.
core stop	Stops Asterisk immediately.
core stop gracefully	Stops Asterisk when no new calls are received.
core show channels	Shows active calls.
sip show peers	Displays all connected users.
sip show registry	Displays registered SIP clients.
sip reload	Reloads the sip.conf file.
core show version	Displays the installed Asterisk version.

V. INSTALLATION AND CONFIGURATION OF LINPHONE

Linphone is a free and open-source software available on multiple platforms that uses communication protocols like SIP.

V.1 Installation of Linphone

The installation is straightforward. Simply run the following command to download and install the Linphone package from the official Ubuntu repository:
sudo apt-get install linphone

V.2 Configuration of Linphone

To configure the Linphone client for the user "6003," follow these steps:
1. Access the Options menu.
2. Navigate to the Preferences submenu.
3. The Settings window will appear. Select Manage my SIP accounts and then Add.
4. The Configure a SIP account window will be displayed.

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Next, fill in the fields as shown:

- Display name
- SIP Identity
- SIP Proxy Address

The values entered should match those configured in the sip.conf file on the Asterisk server.

Once the configuration is complete, the softphone will display a window prompting you to enter the password associated with the account, as shown below:

Finally, the connection to the account is successful, and you can make your first call. Additional SIP accounts can be configured in the same way.

VI. CONCLUSION

This paper allowed us to understand how VoIP works, along with the basic configuration of a Voice over IP (VoIP) server, which in our case is Asterisk. We also explored the benefits of using free and open-source solutions.

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