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Establishment of VoIP Communication Service with Design and Implementation of SIP Client in Python using Asterisk Server

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Abstract: Voice over Internet Protocol (VoIP) is a way to carry out a telephone conversation over a data network. VoIP products promise converged telecommunications and data services that are cheaper, more versatile and provide good voice quality as compared to traditional offerings. Although VoIP is widely used, VoIP on mobile devices is still in its infancy. This paper intends to describe the voice calling on mobile phones and PCs'. The clients can register on the asterisk server through mobile phones or PCs'. The devices should be connected to the server via wireless or wired LAN. The registered clients can make free voice calls to communicate among themselves.
Keywords: RTC, SIP, PYTHON, GPRS, Asterisk, PBX, WiFi, SOFT PHONE.

I. INTRODUCTION

Voice over Internet Protocol (VoIP) has been established as an alternative to traditional telephony networks. VoIP allows telephone conversations to take placeover a data packet-switched networks like the Internet.

Asterisk is a complete phone system in software. It replaces large and expensive phone systems powering thousands of extensions and help users to save money on long international call usage. After implementing the software, it becomes extremely versatile, easy to customize and easy to extend.

Asterisk is an open source framework used for building multi-protocol real-time communications applications. It is stable, reliable, free and is used worldwide in production on thousands of systems. It can turn an ordinary computer into a communications server and offers many call and sms features. It can be used to build many communications applications like VoIP gateways, IP PBX systems, call distributers, conference bridge, voicemail, unified messaging and many more. It is used in small and large businesses, call centers, carriers and government agencies, worldwide. It can easily become the basis for a complete business phone system, or used to enhance an existing system, or can be used as a bridge between systems.

VoIP products promise converged telecommunications and data services that are cheaper, more versatile and provide improved voice quality as compared to traditional offerings.

II. BACKGROUND AND RELATED WORK

Until now the registered clients can make free voice calls by using SIP and IAX Protocols. The clients registered on the same server can communicate through SIP Soft phones, which can easily be installed on desktop as well as mobile phones. The VoIP signaling function can be performed using protocols such as SIP. Data transport can be performed by The Real-time Transport Protocol(RTP). This protocol is used to deliver voice data during conversation.

A. Various Packages Used

asterisk-13.5.9, asterisk-addons-1.6.2.3, dahdi-linux-complete-2.6.1, libpri-1.4.13, asterisk-sounds-1.2.1. Ekiga, Zoiper, X-lite are some softphones for linux and windows operating system. Siproid, IMSdroid are some SIP/VoIP clients for SIP supported mobile phone.

B. Scenarios

Fig. 1 and Fig. 2 shows communication between clients through asterisk, when asterisk is installed on same and different servers.

Fig. 1. Call between laptop, desktop and mobile phone on Wifi/ LAN through asterisk server.

Fig. 2. Calls between laptops and mobiles located in two different cities through asterisk servers.

C. *Session Initiation Protocol (SIP)*

The Session Initiation Protocol(SIP) is an application-layer control signaling protocol. SIP is used to create, modify, and terminate multimedia sessions or conferences such as Internet Telephone calls. The SIP message format is similar to the Hyper Text Transfer Protocol(HTTP) message format. Two main components in SIP are user agent(UA) and servers:

1) *UAs:* are regarded as a client that can send the request and response together. This includes a user agent client and a user agent server.
2) *Servers:* are used to receive requests from clients for servicing and sending responses back to the clients. The servers are typed as redirect server, proxy server and registrar.

D. *Real-time Transport Protocol (RTP)*

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Once signaling functions are implemented, voice data needs to be transmitted between clients. Real-time Transport Protocol(RTP) is viewed as the most powerful protocol to deliver multimedia packets in a session. RTP is defined by Internet Engineering Task Force(1ETF). It consists of two parts:

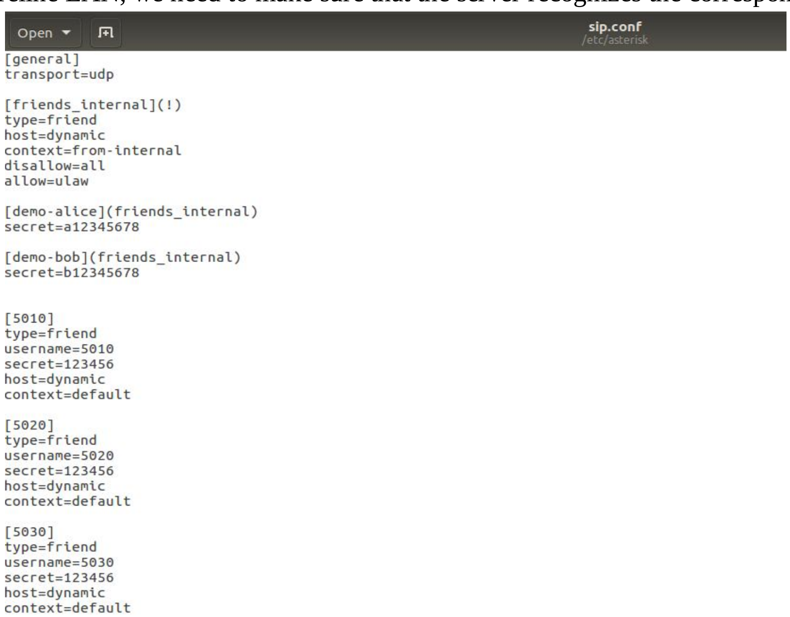
1) *RTP:* is used to cany voice data
2) *RTCP:* is used to monitor the quality of services and information about the participants who are in a sessions.

III. REGISTERING USERS WITH ASTERISK SERVER

To register on the asterisk server client has to make an account on the softphone specifying a valid username, password and server domain name. The softphone will initiate a request to the server, the server will check its entry in the sip.conf file. If there is a valid entry, the server will acknowledge the request for successful registration, and then the user can place calls using asterisk server. For clients registered on different servers to communicate, there should be trunking between the servers via IAX.

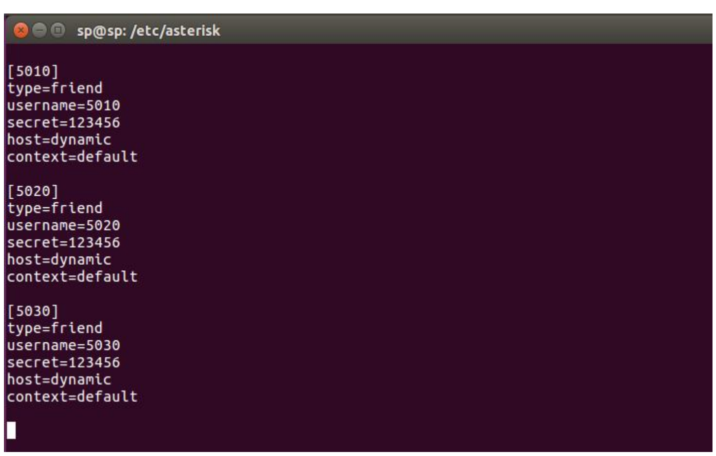
A. *Configuring the Files*

We configure the SIP, EXTENSIONS and IAX at the location /etc/asterisk. Each SIP client needs to be added to sip.conf file before client can register. We can set up the extension for each client in extensions. conf. Asterisk server is running over the internet or connected to a wireless or wireline LAN, we need to make sure that the server recognizes the corresponding IP address [4].


Open
sip.conf
/etc/asterisk

```
[general]
transport=udp

[friends_internal](!)
type=friend
host=dynamic
context=from-internal
disallow=all
allow=ulaw

[demo-alice](friends_internal)
secret=a12345678

[demo-bob](friends_internal)
secret=b12345678


[5010]
type=friend
username=5010
secret=123456
host=dynamic
context=default

[5020]
type=friend
username=5020
secret=123456
host=dynamic
context=default

[5030]
type=friend
username=5030
secret=123456
host=dynamic
context=default
```

Fig. 3.Clients registering on asterisk server


sp@sp: /etc/asterisk

```
[5010]
type=friend
username=5010
secret=123456
host=dynamic
context=default

[5020]
type=friend
username=5020
secret=123456
host=dynamic
context=default

[5030]
type=friend
username=5030
secret=123456
host=dynamic
context=default
```

Fig. 4.Sip.conf

The Fig.4., shows the sip registration done in the sip.conf so as to register user agents in the asterisk server, thus communication using voice calls can be established between them. Only these can regularly make voice calls from one client to another. Asterisk is a powerful server which can connect many calls at an instant and is thus, used for industrial applications.

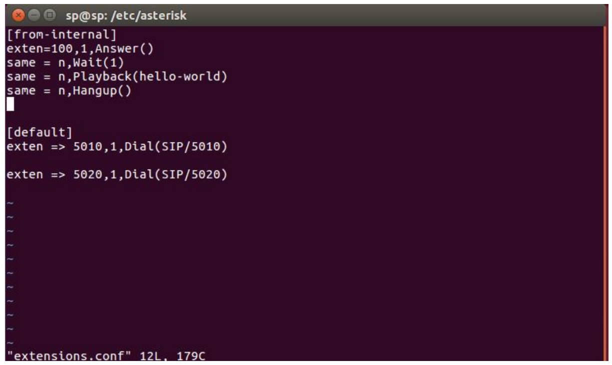


Fig. 5.Extensions.conf

B. Design and Implementation

We setup our asterisk server running on a laptop using Ubuntu 16.04 LTS. The version of asterisk is13.5.9. The sip phones used by us are XLite, Zoiper, etc.

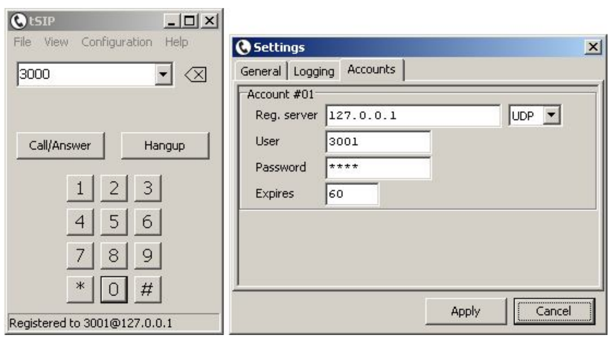


Fig. 6.SIP soft phone display and authentification of the user agent.

1) *Installation of Packages:* To configure our asterisk server some additional packages are also required to be installed, which can be done as follows:

cd /usr/sc/asterisk-11.2.1

./configure && make menu select

Select the add on chan mobile in the list, save and exit. Stop asterisk before compiling and installing, then again restart.

service asterisk stop

make && make install

service asterisk restart

We have to make a modification in the chan mobile source before it can recognize the phone: We have to make a modification in the chan mobile source file at/usr/src/asterisk 13.5.9/addons/chan_mobile.c, change addr.rc_channel=(uint8_t)1with addr.rc_ channel=(uint8 t) 0, so that our mobile device is shown connected to our server. Again build asterisk and reinstall it:

make && make install

IV. USER AGENT

User Agents are regarded as a client that can send the request and response together. This includes a user agent client and a user agent server. The user is the Client which is connected to the asterisk server to another user agent.the Clients used here is a softphone which is build in a Python environment. Also to be kept in mind is the fact that the soft phone built should be suitable to work in a linux operating system. There's afew obvious reasons for choosing Python for VoIP:

Python is easy to work with, and to debug. For implementing a network protocol from scratch, Python is way more superior compared to other languages. It's cross platform, so it is easy to use the source codes in other operating systems. There's a variety of

UI toolkits available from Python. Python would be fast enough to handle VoIP. Presence of Third Party Modules. Extensive Support Libraries. Open Source and Community Development. Learning Ease and Support Available. User-friendly Data Structures. Productivity and Speed.

The object oriented nature, the compact coding style and very small source code size of the implementation makes Python a very good choice for implementing application protocols such as SIP. There is some overhead because it is an interpreted language; however the overhead is comparable to that of Java run-time. The interpreter is now usually available for embedded platforms as well, making it more portable than other languages such as Action Script or C++.

The biggest advantage of an implementation in Python is that the code size is drastically smaller than other languages. I have implemented the basic SIP stack in less than 2000 lines of Python source code. Compare this with the Java implementation of SIP which has more than 1000 files. The lower size means that not only the development time is smaller, but also the testing, code review and maintenance cost is much lower.

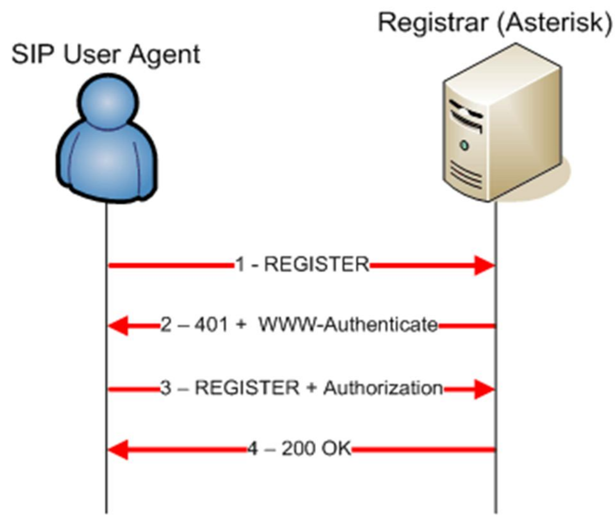


Fig. 7. SIP authentication in Asterisk server

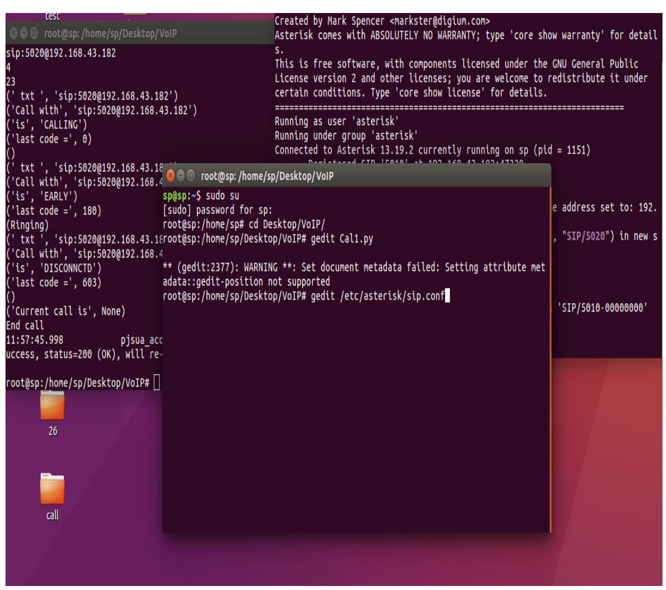


Fig. 8 .The User agents communicating with each other and the asterisk server running in the background.

V. CONCLUSION

The design and implementation of our model enables the SIP clients to communicate with each other through asterisk server using free voice calls. The client softphone is designed using python because to implement a network protocol using Python is way more superior compared to other languages. The model can be successfully implemented in remote areas where communication is sparce, Mining Industries, research organizations, etc. As SIP phones are installed on the desktops, employees can communicate with each other for free, thereby reducing the cost of the overall communication network.

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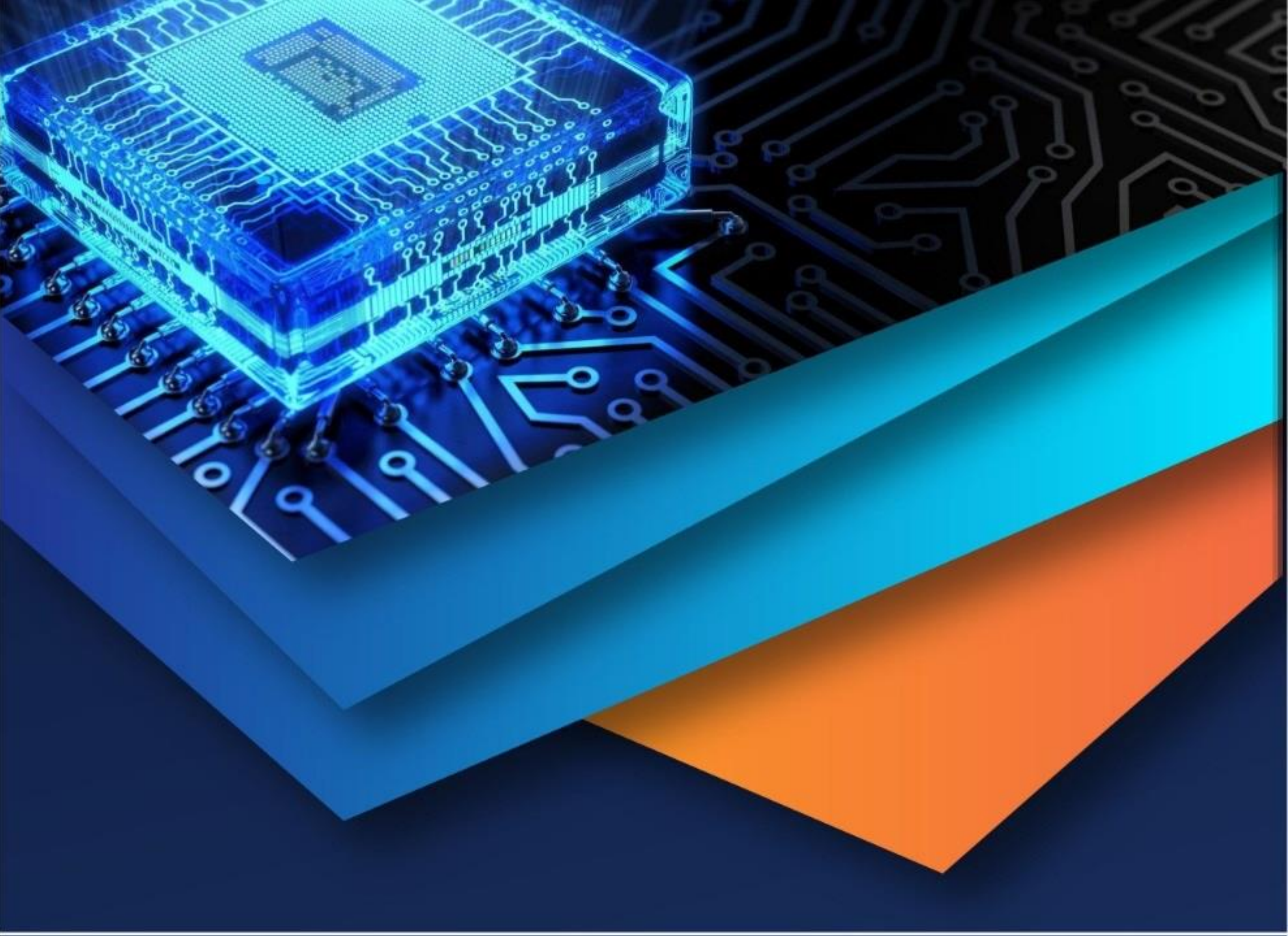

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