

Bridge Implementation Between IP Network and GSM Network

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ABSTRACT: *The goal of this paper is to implement and establish an interconnection between Asterisk server and phone networks in general and mobile networks in particular. The arrival of new generations of phones that are equipped with Android system facilitates this project. More than the main goal is to remove the cost of internal calls using asterisk with the possibility to call to the external network : PSTN and GSM.*

Keywords: VoIP, Asterisk , SIP, Android, GSM

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1. Introduction

The development of the Internet has changed the way to use our phone. Indeed, the technology of conventional telephony PSTN and GSM is now about to be superseded by IP telephony.

Similar to traditional telephone, voice over IP (VoIP: Voice over Internet Protocol) is used to transmit voice, referring to IP (Internet Protocol). This allows us to make phone calls over existing data networks without any additional infrastructure or additional cost. All we need is three tools in one: A desktop computer, IP phone in hardware or software form. But a call from the local network to the outside telephone network requires a gateway. In paper [1] [2] we can see the devices used for interconnections between the computer network and the PSTN by a bridge (ATA), which is a kind of RJ45/RJ11 converter.

In our case we set up a device that is equipped with a android system provided the gateway phone.

In a first step we will look around our technology project, then we will finally presented our architecture and implementation of a bridge between the IP network and the GSM network.

2. State of Art

2.1 Presentation Of Asterisk

Asterisk [3] is software that is installed on a PC, acts as PABX (Private Automatic Branch eXchange). It is free software (open source), published under the GPL (General Public License) and created by Mark Spencer of Digium company. It runs on Linux, BSD (Berkeley Software Distribution) and Mac OS X.

2.2 Android

Android [4], is an open source operating system using the Linux kernel, for smartphones, tablet PCs, PDAs and mobile devices designed by Android. In our case we need Android to facilitate the interconnection between the server computer and the GSM network.

2.3 Connecting asterisk with the PSTN

Interfacing with the telephone operators and enterprise switches Traditional is one of the main strengths of Asterisk. Asterisk can interface with the PSTN on a link small capacity [3].

```
[general]
port = 5060
bindaddr = 123.45.67.3
realm = domainebis.fr
disallow = all
allow = alaw
allow = ulaw
allow = gsm
language = fr
allowguest = yes
Authentication sur le serveur Asterisk
principal:
[authentication]
auth = voicesserverbis:notresecret@domaine.fr
[voicesserver]
type = friend
host = 123.45.67.1
context = from-voicesserver
secret = notresecret
```

Figure 1. Sample of /etc/asterisk/sip.conf file

2.4 VoIP protocols

The main protocols used for establishing VoIP connections are:

- H.323
- SIP
- MGCP
- IAX (Asterisk)

2.5 SIP (Session Initiation Protocol)

SIP is a protocol which has been standardized by the IETF (Internet Engineering Task Force) in 1999. It comes from the world of the internet. It is responsible for authentication and location of participants. But it does not carry data.

SIP gradually replaces H.323 because it is more flexible and scalable. It is chosen to integrate telephones with Asterisk protocol. Today SIP has reached the level of maturity that allows to reproduce from one based architecture all the features of a sophisticated phone system business. The other base material of the architecture of the internal telephone network of the company is naturally Asterisk. A signaling protocol is a protocol layer 5 (session) of the OSI model. It manages an IP telephone call as follows:

1. Note the coordinates of the contact person (dialing)
2. indication corresponding to a call (the receiver's phone rings)
3. Acceptance of the call by the corresponding (for answer)
4. Information to third parties seeking to join the two parties of their availability (busy)
5. End of availability and communication lines (hook)

SIP is a protocol request / response as http [5]. A SIP address is in the form sip: 1000@192.168.1.2. 1000 is the SIP extension

number and 192.168.1.2 is the IP address of the SIP server.

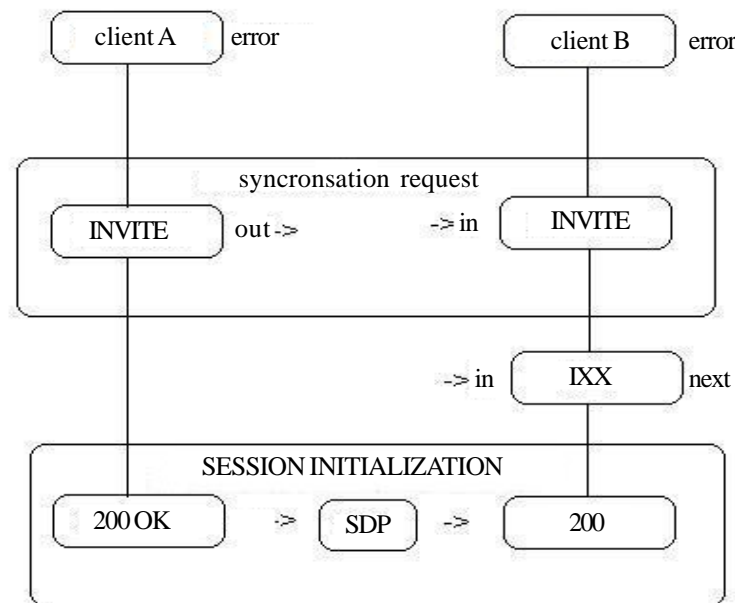


Figure 2. SIP operation

3. Implementation

3.1 The SIP client

X-Lite and Twinkle is free software used as the Asterisk server clients in the field of IP telephony, also called softphone based on the open standard SIP protocol. X-Lite is multiplatform software for Mac OS X, Windows and Linux, associated with a SIP account. Twinkle is only for Linux, they can enjoy all the traditional telephone services.



Figure 3. SIP account on X-lite

3.2 Interconnection-Softphone Softphone

3.2.1 Architecture of network

We have completed the interconnection of two different softphones, this option is a basic activity of Asterisk.

3.2.2 File sip.conf

The configuration section of / etc / asterisk / sip.conf treats terminal connected to the main Asterisk server. The sip.conf file will define all our users.

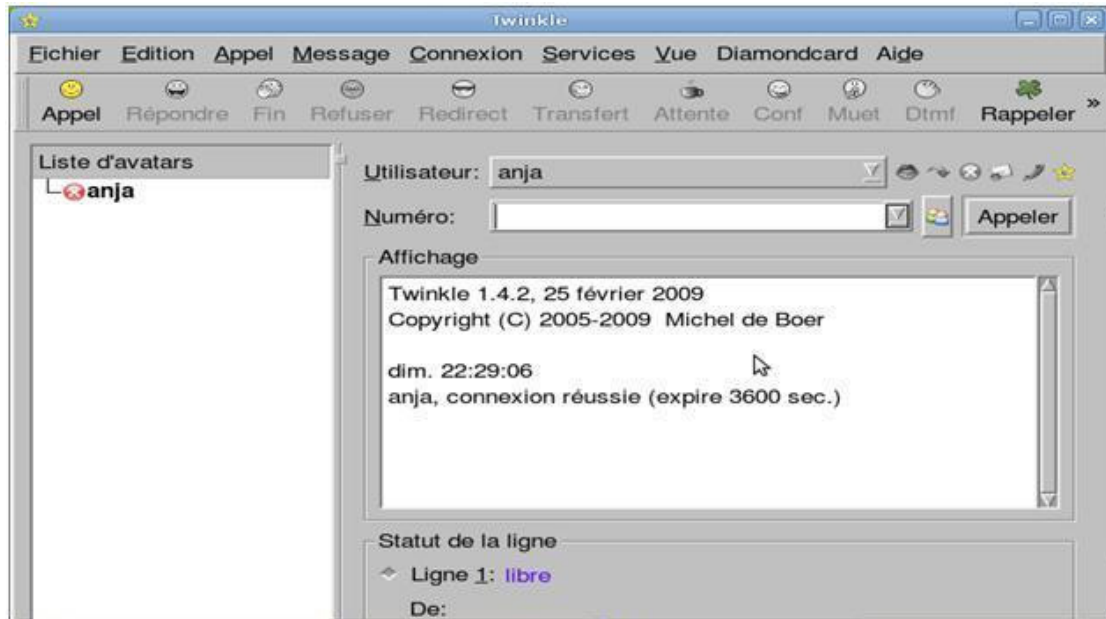
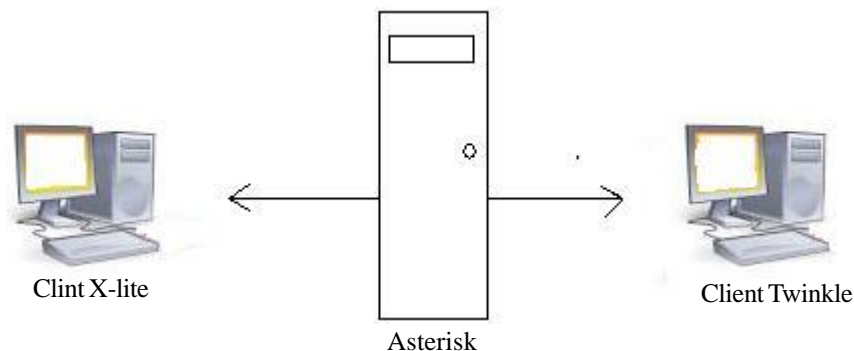


Figure 4. SIP account on twinkle



3.2.3 File extension.conf

Dial plan is the most important element of Asterisk. It consists of a sequence of commands that will be executed when a call is received by Asterisk. It is in the dial plan you can configure call routing.

Like any program worthy of the name, the dial plan requires a set of syntactic rules to the administrator. Historically, the configuration file is dial plan etc / asterisk / extensions.conf. Extensions.conf to define routing rules that is to say actions to do during a call to a phone number.

For interconnecting X-LITE – TWINKLE

3.3 Softphones – GSM interconnection

Our goal was to ensure that a SIP client can called a client via a GSM mobile phone connected via USB to the Asterisk server. The GSM phone therefore serve as the gateway for the SIP client and mobile phone number called by the SIP client. To set the gateway, it uses adb.sh is a bash script (Linux) that launches the program adb (Android Debug Bridge) for controlling the phone via USB to make the call specified by the number argument \$ {EXTEN}.

4. Result

When the external call, we see the exchanges of SIP Message, following is a Request between softphone whose number is 1000 and the GSM number 26133 ...

```
[general]
bindaddr = 0.0.0.0
port = 7000
language = fr
[1000]
username = 1000
type = friend
secret = 1000
host = dynamic
callerid = "Anja" <1000>
[1001]
username = 1001
type = friend
secret = 1001
host = dynamic
callerid = "Olona" <1001>
[1002]
username = 1002
type = friend
secret = 1002
host = dynamic
callerid = "Sankey" <1002>
```

Figure 6. Sip.conf file

```
[default]
exten => _1XXX,1,Dial(SIP/${EXTEN},10)
exten => _1XXX,n,VoiceMail (${EXTEN},@default,u)
exten => _1XXX,n,Hangup ()
exten => 777,1,VoiceMail ()
```

Figure 7. Extension.conf file

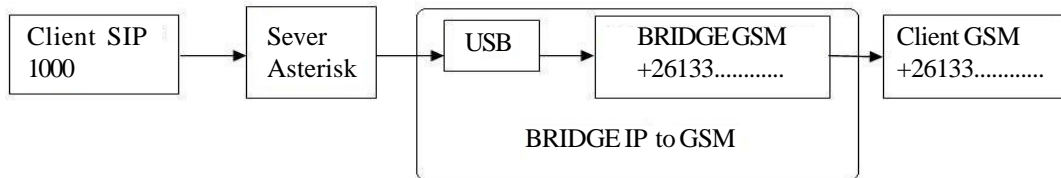


Figure 8. Operating diagram of Softphone-GSM interconnection

```
exten => _03XXXXXXXX, 1, Answer ()
exten =>
_03XXXXXXXX, n, System (/home/anja/adb/adb.sh
${EXTEN})
exten => _03XXXXXXXX, n, Wait (3)
exten => _03XXXXXXXX, n, Hangup()
```

Figure 9. Extension.conf for interconnection

```
# !/bin/bash
/home/anja/adb/adb shell service call phone 2 s16 $1
```

Figure 10. Adb script sample

When the external call, we see the exchanges of SIP Message, following is a Request between softphone whose number is 1000 and the GSM number 26133 ...

33.159426	192.168.1.1	192.168.1.2	SIP/SDP	Request: INVITE sip:03307357020192.168.1.2, with session description
43.160743	192.168.1.2	192.168.1.1	SIP	Status: 401 Unauthorized
53.162177	192.168.1.1	192.168.1.2	SIP	Request: ACK sip:03307357028192.168.1.2
63.270838	192.168.1.1	192.168.1.2	SIP/SDP	Request: INVITE sip:03307357020192.168.1.2, with session description
73.272260	192.168.1.2	192.168.1.1	SIP	Status: 100 Trying
83.273121	192.168.1.2	192.168.1.1	SIP/SDP	Status: 200 ok, with session description

Figure 11. Sip message request

4. Conclusion

We have found that with only Asterisk an android system is equipped and then we control the internal and external communication of our local phone, Asterisk has powerful features that should enable it to succeed in the future with VoIP, an emerging technology that tries several companies saw its benefits. Implementation point of view, the absence of additional cost of migration to the telephone network to the computer network leads us to do more research about voice over IP. The VoIP is currently the center of all research on telecommunications and IT. As an improvement to our case, the integration of a module for video calling on mobile phone is one of the next steps in our research that the VoIP already has this technology, so we will have spent not only vocals but also calls to the outside of videoconferencing networks.

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