

VoIP System based on Asterisk for Enterprise Network

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Abstract— This paper describes VoIP system for the enterprise network based on Asterisk. First the main ideas and the features of the VoIP system are described that we have developed by using Asterisk in the Intranet environment. Then the new scheme to realize high security by using Open VPN is described when developing the large scale enterprise network.

Keywords— VoIP, Enterprise network, Asterisk, IP-PBX, Open VPN

I. INTRODUCTION

This paper describes VoIP system for the enterprise network (e.g. company, university) based on Asterisk (<http://www.asterisk.org>). Asterisk is a kind of open source software to implement IP-PBX system and supports various necessary protocols to realize the VoIP system such as SIP, H.323, MGCP, SCCP. First the main ideas and the features of the VoIP system are described that we have developed by using Asterisk in the Intranet environment. Then the new scheme to realize high security by using Open VPN is described when developing the large scale enterprise network.

II. BASIC IDEA

The following are the main requirements to develop the VoIP system for the enterprise network.

A. Scalability

In the environment of the enterprise network, it is not easy to anticipate the traffic because there are lots of uncontrollable factors. So developing various scale systems based on the same architecture is necessary to meet the unpredictable change of traffic.

B. Cost

It is obviously desirable to develop the system at reasonable cost because generally the budget is rather limited.

C. High security

Also obviously high security is indispensable.

Considering the above requirements, the following are our basic ideas.

A. Developing VoIP system by using Asterisk

Obviously considering development cost it is desirable to use the open software. So we have selected three open software as candidates, that is, OpenSIPS, FreeSwitch and Asterisk. As the SIP server's viewpoint, OpenSIPS and FreeSwitch are superior to Asterisk in terms of functions, but Asterisk support various protocols (e.g. H.323, MGCP, SCCP) other than SIP and also has lots of additional PBX services (e.g. Voice Conference, Automatic Call distributor). So we have decided to use Asterisk to development VoIP system for the enterprise network.

B. Realizing high security by using Open VPN

When we develop the large scale enterprise network by connecting multiple Asterisk servers located in different sites based on Asterisk proprietary protocol (i.e. IAX2), some method is necessary to realize the high security because the voice data among sites is not encrypted. For this purpose we have introduced a new scheme to establish VPN by using Open VPN.

III. OVERVIEW OF ASTERISK

Asterisk is a kind of open source software executed on Linux to implement IP-PBX system and support various VoIP protocols such as SIP, H.323, MGCP, SCCP. It can be connected with IP network and also can be connected with the existent telephone networks via analog/digital interfaces. Figure1 shows the architecture of Asterisk. Channel portion in Figure1 consist of various logical communication interface modules and Application portion consist of the additional PBX service modules. In the following the main modules of the channel

A. Channel Modules

1) DAHDI (Digium/Asterisk Hardware Device Interface):

To connect with the ordinary existent telephone terminal it is necessary to insert the telephony card (e.g. telephone card of Digium or of Voicetronix) as the physical interface and then the DAHDI interface module will be used. In case of connecting with existing POTS (Plain Old Telephone Service), FXS (Foreign eXchange Subscriber) and FXO (Foreign eXchange Office) interfaces will be used. In case of connecting with ISDN terminal it is necessary to insert the

extension card as the physical interface.

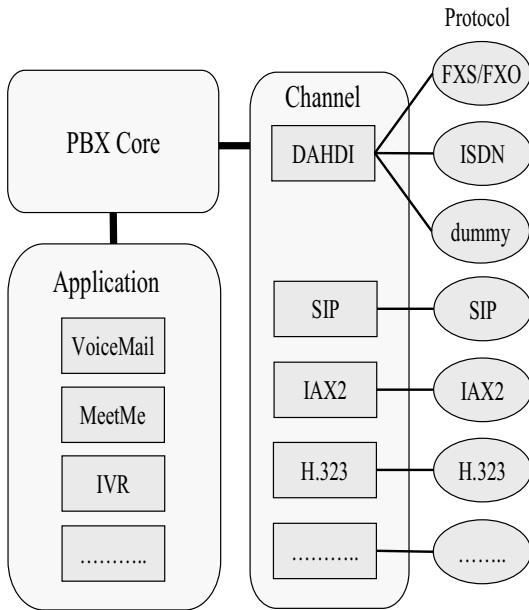


Figure 1. Architecture of Asterisk

2) SIP: This is the most basic signaling protocol to perform call processing in Asterisk and RTP/RTCP are used in order to transmit user data (e.g. voice data).

3) IAX2 (Inter-Asterisk eXchange2): IAX2 is Asterisk proprietary protocol to connect with multiple Asterisk servers located in the different sites. The same port (i.e. 4569 as the default port) is used to transmit the call control signal and voice data.

B. Application Modules

1) Voice Conference: The voice conference service in Asterisk is called as “MeetMe”. User can join the conference by inputting the designated number as the service number.

2) IVR (Interactive Voice Response): The automatic voice response can be performed by integrating voice response data file and dial number plan

3) AGI (Asterisk Gateway Interface): AGI is an API to connect the outside program with Asterisk in order to include some additional functions. Various programming language (e.g. C, Java, Perl, PHP, Bone Shell) are supported.

4) SLA (Shared Line Appearances): Multiple telephone terminals can share a subscriber line.

IV. VOIP SYSTEM BASED ON ASTERISK

A. VoIP System based on Asterisk

Figure 2. shows the VoIP system that we have developed by using an Asterisk in the Intranet environment (i.e. enterprise network).

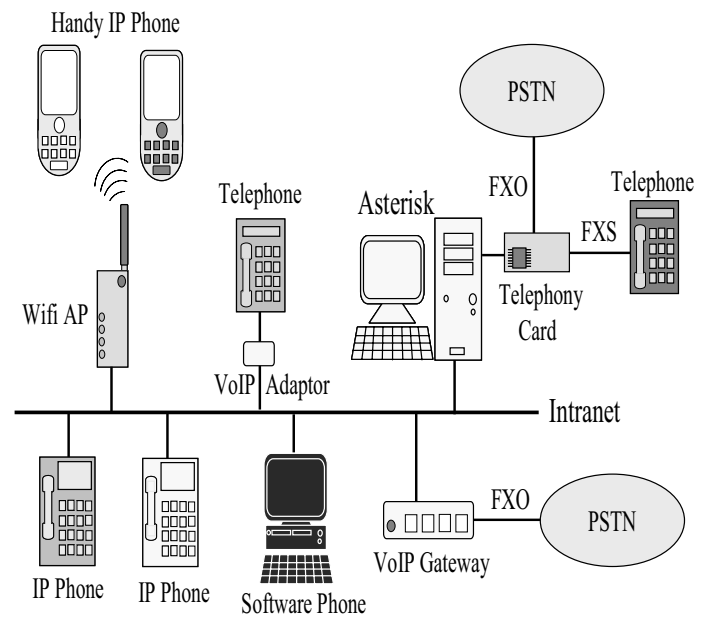


Figure 2 VoIP system developed in Intranet

In the Figure 2 all telephone terminals are connected to one Asterisk server, but it is possible to use multiple Asterisk servers depending on the scale of the Intranet (i.e. the number of terminals). As the IP phones, we have accommodated Grandstream BT101 and Snom 105 (Figure 3), and also Grandstream HT286 (Figure 4) has been used as VoIP adaptor. Grandstream HT488 and Sipura SPA1000 have been used as VoIP gateways to connect with PSTN.



Figure 3 Snom 105 as IP Phone



Figure 4 Grandstream HT286 as VoIP Adaptor

B. Multiple Location Connection by IAX2

As described previously IAX2 is the Asterisk proprietary protocol to connect with multiple Asterisk servers (see Figure 5). So it is possible to connect with multiple Asterisk servers located in different Intranets easily.

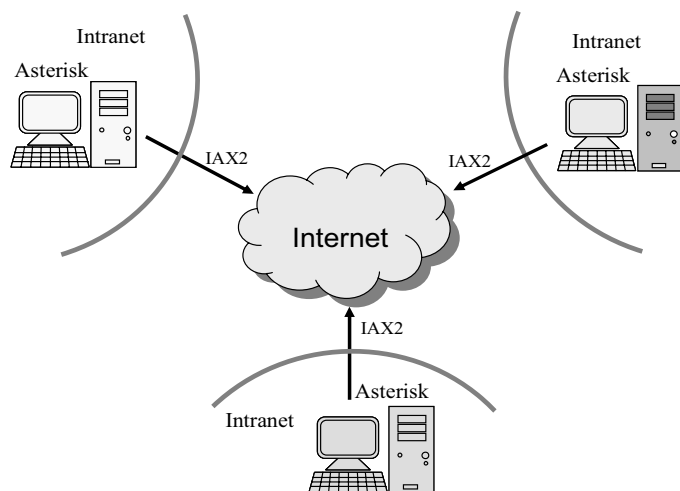


Figure 5. Connecting with multiple Asterisks

V. DEVELOPMENT PROCESS OF VOIP SYSTEM

A. Basic development process

1) DAHDI compile and install : First of all Asterisk should be installed, but before that it is necessary to complete the DAHDI compile and install.

2) Asterisk compile and install : Next Asterisk compile and install has been performed.

3) Define Dial Plan : Dial Plan is the core portion of the call processing in Asterisk. Dial Plan is defined in `/etc/asterisk/extensions.conf`. Extensions.conf consist of general section, globals section and context blocks as follows;

- General section: General parameters to cover the whole Dial Plan are defined in this section.

- Globals section: Variables used in the content blocks are defined in this section.

- Context blocks: Multiple dial plan are defined in the context blocks independently. So Asterisk can realize flexible dial plan by selecting appropriate block based on the conditions. The format of each line in the context block is as follows;

exten => Extension, Priority, Application

Extension in the right side is generally telephone number and Priority is the order of processing. Application is the processing to be performed to the Extension.

B. Application development process

1) MeetMe (Voice conference): MeetMe, voice conference service, can be easily realized in Asterisk. In order to register the service, first, registration data is defined in the `[room]` section of `/etc/asterisk/meetme.conf`.

```
[default]
.....
exten => 9000,1,Goto(incoming,s,1)
.....

[incoming]
exten => s,1,Answer()
exten => s,n,Wait(1)
exten => s,n(again),Background(vm-enter-num-to-call)
exten => s,n,WaitExten(10)
exten => s,n,Playback(vm-goodbye)
exten => s,n,Hangup()
;
exten => i,1,Playback(invalid)
exten => i,n,Goto(s,again)
;
exten => 1,1,Dial(SIP/1000)
exten => 2,1,Dial(SIP/1001)
exten => 3,1,Dial(DAHDI/1)
```

Figure 6. Example of IVR definition

2) IVR (Interactive Voice Response): In order to realize the automatic voice response service, detailed Dial Plan should be defined in `/etc/asterisk/extensions.conf` as shown in Figure 6. Answer in the `[incoming]` context shows that Asterisk will perform automatic response processing. The function of

Background shows that the voice file of vm-enter-num-to-call will be played and that the control signal from the terminal can be processed even during the voice response. Playback is also a kind of function to play the voice file, but the user's signal cannot be processed during the voice response. WaitExtern is a function that suspend the signal processing for the defined time.

VI. OPEN VPN

In order to realize high security to connect multiple Asterisks located in different Intranets, we have implemented VPN capability.

Figure 7 shows the procedure to establish VPN between two Asterisk servers by using OpenVPN (<http://openvpn.net/>) based on the regular SIP sequence. To realize this procedure we have developed a program (i.e. sip_app) to have SIP client function with the function to invoke the external application. It is developed by using oSIP2 (<http://www.gnu.org/software/osip>) and eXosip2 (<http://www.antisip.com/as/en/products.php>) libraries in GNU, and has the SIP client function, SDP control function and the function to invoke the external process as child process. In the Asterisk server1, OpenVPN is registered as the external process and sip_app send the REGISTER message to SIP server (1). In the Asterisk server2, sip_app send the REGISTER message to SIP server (1) and send INVITE message to the Asterisk server1(2, 3). Asterisk server1 invoke the OpenVPN as the server mode (4) and reply 200 OK after inserting the necessary connection information into "a" record in SDP (5,6). Asterisk server2 invoke OpenVPN as the client mode after getting the necessary information from "a" record in SDP (7). OpenVPN in the Asterisk server2 communicate with OpenVPN in the Asterisk server1 and VPN between two servers has been established(8).

Table 1 shows the values of SDP at the process (6) in Figure 7. Record "m" shows media type (i.e. application /VPN) and the kind of protocol (i.e. OpenVPN). Record "a" is used by sip_app to control external process invoke. IP4 in Table1 is the IP address of the Asterisk server1 and PORT is

the port to receive OpenVPN connection of Asterisk server1. VPN_LOCAL_ADDR is the IP address of Asterisk server1 and VPN_REMOTE_ADDR is the IP address of Asterisk server2.

VII. CONCLUSION

This paper describe VoIP system for the enterprise network (e.g. company, university) that we have developed based on Asterisk which is a kind of open source software to implement IP-PBX system. Through the development and evaluation, we have confirmed that VoIP system based on Asterisk is very powerful as a whole and most PBX functions to be required for the enterprise network can be realized.

Compared with the general SIP server, it can be said that Asterisk is more focused on providing basic functions. But Asterisk can connect with SIP server easily, so it is possible to implement the necessary additional functions by just connecting with other outside SIP servers. Also Asterisk can connect with the existent PSTN by using FXO telephony card, so it is possible to be used as the VoIP gateway. When developing the large scale enterprise network by connecting multiple Asterisk servers located in different sites based on IAX2, to realize high security is the issue because the voice data is not encrypted. To solve this issue, we have proposed the method to establish VPN by using Open VPN and have also described the development process in detail.

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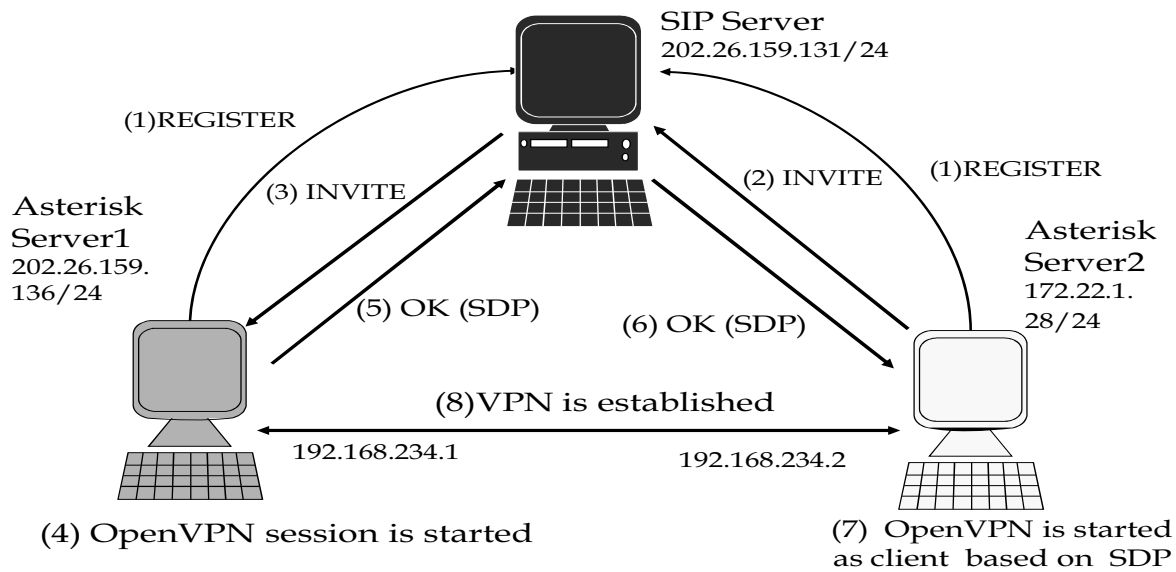


Figure 7. VPN establishing procedure

TABLE 1. RECORD VALUE OF SDP

Record Type	Value
v	0
o	2500 1169538046 1169538046 IN IP4 202.26.159.131
s	-
t	0 0
m	application/VPN 7084 OpenVPN 0
c	IN IP4 202.26.159.131
a	IP4:202.26.159.136
a	PORT:8000
a	VPN_LOCAL_ADDR:192.168.234.1
a	VPN_REMOTE_ADDR:192.168.234.