The Application of Telephone Mobility Gateway Based on VPN Tunnel

Lin Jia-Ching and Sun Pei-Chen

Abstract—**From the earlier analog phone to smart phone for each person now, from analog to digital voice transmission, the fast-rising VoIP has reached an unprecedented peak in nowadays of flourishing smart phone. However, the analog phone technology has been mature, stable and of high penetration rate. This paper introduces the technology of integrating IP PBX and analog phone via VoIP gateway based on SIP protocol, so as to realize Telephone Mobility. It could convert the analog phone to be a part of extension either at home or company or anywhere. Through implementing the system, it's found the Dial Plan can make analog phone communicate with the extensions within the original PBX, but also be converted into an extension to call other extensions in the company through VoIP. When making PSTN Calls on outside, the call will go through the office external line via VoIP to avoid additional telephone bills from hotel, it is unnecessary to need to access the software IP phone in your PC. Such application will provide a solution to speed up the integration of VoIP and analog phone, it able to get security guaranteed when voice call through VPN Tunnel on Internet.**

Index Terms—**VoIP gateway, IP PBX, SIP, telephone, mobility, dial plan.**

I. Introduction

This chapter will briefly introduce the research background and purpose, literature review, system implementation and findings of this paper.

A. Research Background

With the rapid growth of broadband network in Taiwan in recent years, the penetration rate of FTTB/H has even ranked fifth over the world. In 2012, Chunghwa Telecom released symmetrical or asymmetrical network bandwidth 100MB. For the VoIP, it is the primary beneficiary under high-speed network infrastructure; however, its fast development also results in more threats to traditional PBX system. The real-time voice transmission via Internet greatly lowers the communication costs. More and more domestic enterprises apply open-source IP PBX for company communication, which are even comparable with that of business IP PBX, such as Cisco or Panasonic. In spite of this, the analog phone still occupies major market share. Some users install the VoIP applications like CSIPSimple and 3CX application on smart phone, but it requires complicated settings in prior to offer a good performance, which often takes a lot time. Besides, the users have to call with analog phone in case of improper settings.

Manuscript received June 13, 2013; revised August 6, 2013.

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In the open-source field, Asterisk is a powerful IP PBX, which supports the mainstream VoIP communication protocols, such as H.323, SIP, MGCP, etc, and can be also easily integrated with heterogeneous PBXs, such as Cisco's Call Manager or PSTN. Moreover, it supports the basic call function, such as transferring, pickup or meeting, as well as some advanced function, such as voice messages or auto operation service. This paper applies Asterisk PBX software together with Cisco Linksys SPA3000 to implement telephone mobility. It could convert the common analog phone into VoIP via SPA3000, which can communicate with other extensions under Asterisk and also registers back to the company via network and then make calls from the Analog/Voice E1 of the company when calling external line. Even when making phone calls on a business trip, you could connect your analog phone with FXS port, and set up dial plan design on SPA3000, making FXO port communicate with the extensions under the original switch. In this way, it keeps the communication channel with the original PBX; we also use VPN Tunnel to encrypt the voice call through the Internet, it will increase the safety.

B. Research Purpose

As stated in the above chapter, the prevalence rate of broadband network grows rapidly. As revealed in the statistics by Chunghwa Telecom in January, 2012, the broadband network users in Taiwan have reached 5,500,000. However, due to the capability of 3G network, it must wait for 4G LTE network service to dial VoIP phone on handy devices. Then the acceptable voice quality could be provided on the smart phone, which will take a long time. However, we could lower the communication costs by using the high prevalence rate of broadband network to build VoIP. In addition to digital voice service, it could even provide video & audio communication service. Apart from the original data transmission and network service, VoIP is also a killer-level value-added application based on the high-speed broadband network. All these services are unavailable in Public Switch Telephone Network (PSTN). With the integrated application through transparent VoIP gateway, you could convert the analog phone to VoIP even in the environment of traditional PBX system, rendering the analog phone two numbers: one is the original phone number, and the other is the internal extension number of the company. By doing so, the original phone number won't be restricted by installed gateway, also it is no need to use VoIP installed on your computer. Simply by following four installation steps, the analog phone will be converted into convenient VoIP, which meets the using habits of analog phone and may be easily acceptable to common users, on the other hand, it would has wiretap or embezzle concern once voice call deliver through the Internet, for this reason, it would increase the security when the Internet voice call encrypt by VPN Tunnel, so you don't worry about important information leaking on the Internet

II. Literature Review

This paper takes Asterisk as the company's VoIP switch, Draytek Vigor 2110 and 2110N as encryption channel of VPN Tunnel, and Linksys SPA3000s as Analog Telephone Adapter (ATA), to convert analog phone into VoIP. Through SIP protocol, it becomes a part of the company's extensions by registering Asterisk, but it could still communicate with the extensions under the analog phone switch through FXO port, without any impact on the structure of the original PBX system.

A. SIP Protocol [1]

SIP was initially developed in Columbia University in 1998. In March, 1999, IETF's Multipart Multimedia Session Control (MMUSIC) Team formulated the official standards, RFC2543. In September, 1999, IETF set up new team to formulate SIP V2.0 and later in July, 2000, RFC2543bis draft was released. In 2001, RFC3261 was officially released, which marks the SIP foundation was confirmed. Later on, some revision editions of RFC were released to enrich the content in some fields, such as security and identification. SIP is an information communication protocol formulated by Internet Engineer Task Force (IETF) for media meeting. Based on ASCII texts, SIP is a control protocol used to create, maintain and end the communication application layer between two points and multiple points. Just like other VoIP protocols, SIP is designed to define the signal sampling and communication management in the data network. Signal sampling converts the communication contents into digitalized information for transmission to the entire network; while the communication management controls all statuses from the very beginning to the end.

B. SIP Components [2]

SIP is a Peer-to-Peer communication protocol with distributed architecture. It names the addresses by URL and transmits message in the text format. By taking advantage of Internet model, SIP could build VoIP network and applications. If dividing from the perspective of hardware architecture, the devices in SIP network could be divided into two categories: Client and Server.

C. SIP Client [3]

SIP Phone – SIP phone could play the role of either UAC or UAS. For example, software phone (PC installed with software of telephone communication) or SIP-protocol IP phone, could send a SIP request, or respond to the calling party.

D. Gateways

Gateway executes the communication control, and provides various network service items, which mainly convert the SIP socket point and other terminal statuses. The feature includes converting data transmission format and the difference of communication programs. In addition, the gateway is also responsible for decoding between different voice and image coders, establishing and ending the communication calls between different communication media, such as LAN and packet switched network.

E. SIP Server [3]

Proxy Server plays a role of agent. When receiving the SIP request from Client, the proxy server will transfer the SIP request to the destination based on the incoming side. Generally, after receiving SIP information, the proxy server will transfer the information to the next SIP server in the network immediately, and execute the functions of user authorization, authentication, network access control, network routing judgment and security control.

F. IP PBX [4]

Currently many VoIP switches publicize the source codes on Internet. For example, 3CS PBX is developed on Windows platform, and Asterisk is the first PBX system implemented based on open source. The development platform of the system was Linux initially, which can operate on various platforms, including NetBSD, OpenBSD, FreeBSD, Mac OS X and Solaris. Also some people have transplanted the system onto Microsoft Windows platform, namely, AsteriskWin32. Asterisk contains many features available on the expensive business switching system, such as voice mails, multi-party IP conferences, interactive voice response (IVR), phone menus, phone customer center and other mechanisms. In this paper, Asterisk is an important role to control the calling rules. Below is the Calling Route Flow Dividing Diagram, as shown in Fig. 1:

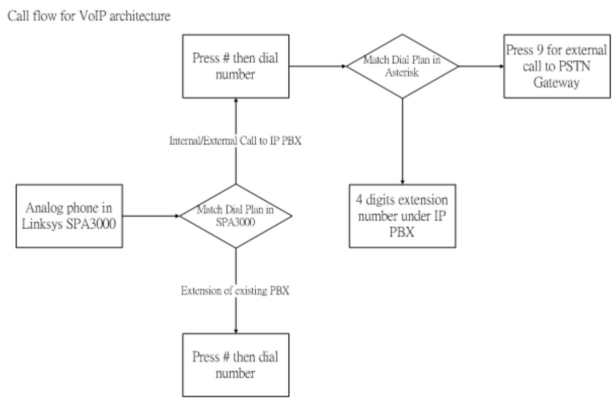


Fig. 1. Calling route diagram.

G. Voice Gateway

Linksys SPA3000 (FXO * 1 & FXS * 1) mainly provides a line incorporation function to allow the VoIP and telecom lines to share the telephones, and also the function of transferring in/out. The analog phone could be transferred on VoIP and it doesn't matter analog phone located anywhere around the world. It supports auto call routing(PSTN TO VOIP, VOIP TO PSTN) and dialing rules, which is regarded as independent hardware VoIP with 24-hour standby and no PC started. You can make cheaper VoIP calls from analog phone or extension under the company operator, with the standard SIP protocol supported. Below is the Comparison Chart of some Analog Telephone Adapter(ATA) Products, as shown in Table I:

TABLE I: COMPARISON CHART OF ATA PRODUCTS

Product Name / Manufacturer Feature Comparison	SPA3000 / Linksys	ATA171P / Welltech	TB110/ Soundwin
SIP	Yes	Yes	Yes
FXS	Yes	Yes	Yes
FXO	Yes	Yes	Yes
Built-in Dial Plan	Yes	No	Yes

H. Sipura Technology [5]

Cisco acquired Sipura Technology by USD68,000,000 on April 26, 2005, and categorized the related products under Linksys Department. Sipura which manufactured the network communication products at mid and low prices for the common users, was the leader of VoIP technology for the consumers before acquired by Cisco, and also the key technology supplier for Linksys' VoIP network devices.

I. IP Phone

IP phone could be the physical phone supporting SIP or VoIP software installed on PC that supports SIP, such as X-Lite - http://www.counterpath.com/x-lite.html, or applications installed on the smart phone, such as CSIPSimple, 3CX phone, etc.

J. Router

Router is network hardware, which is mainly used for Internet connection, NAT network redirecting service, firewall and VPN virtual private network. The system implementation adopts Draytek [6] IP share. Below is the detailed specification, as shown in Table II:

TABLE II: SPECIFICATION OF VIGOR 2110

Hardware interface	4 x 10/100M LAN Switch, RJ-45
	1 x 10/100M WAN Port, RJ-45
LAN	Port-based VLAN
	4-port 10/100 M Ethernet Switch
WAN Protocol	DHCP Client
	Static IP
	PPPoE
VPN	Up to 2 VPN Tunnels
	Protocol : PPP, IPSec, L2TP, L2TP over IPSec
	Encryption : AES, MPPE and DES / 3DES
	Authentication : MD5, SHA-1
	IKE Authentication : Pre-shared Key and Digital Signature (X.509)
	LAN-to-LAN, Teleworker-to-LAN
	DHCP over IPSec
	Dead Peer Detection (DPD)
	VPN Pass-through

III. SYSTEM IMPLEMENTATION

The implemented system architecture of this paper is shown in Fig. 2:

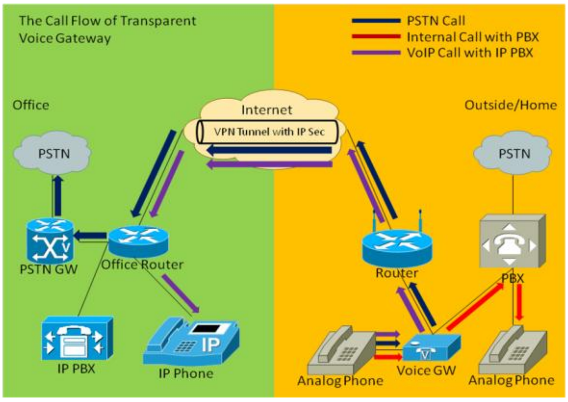
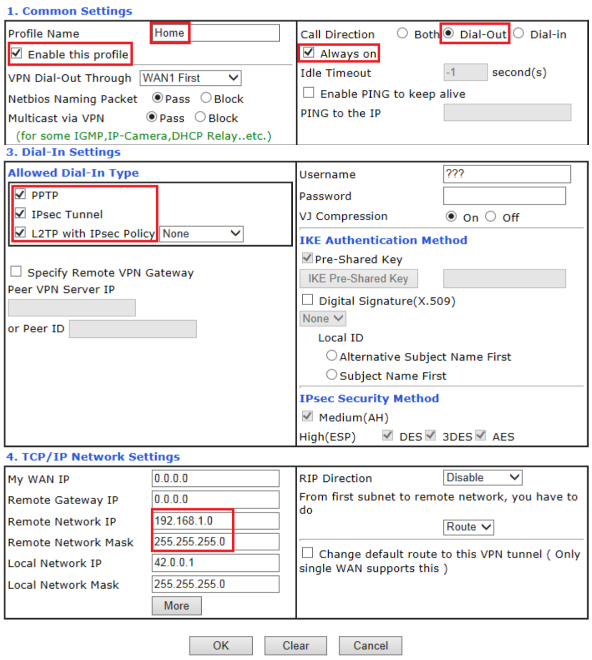
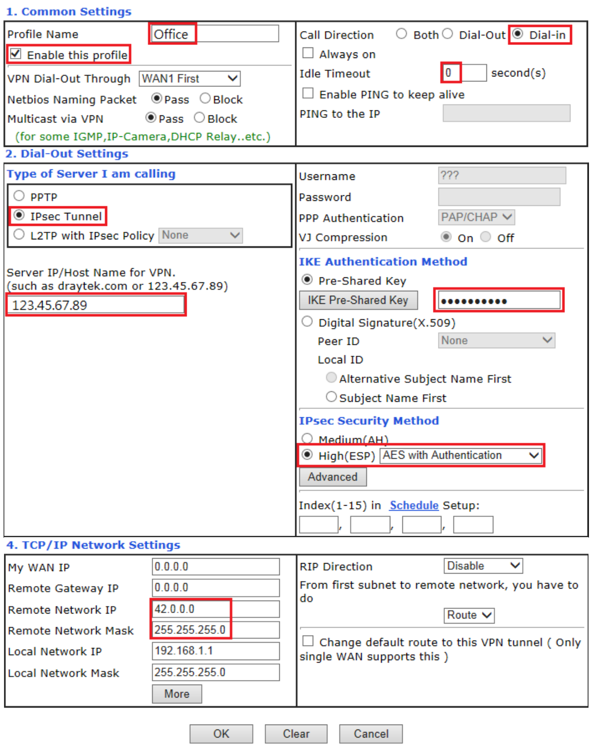


Fig. 2. Implemented system architecture.



(a). Mobility VPN gateway.



(b). Office VPN gateway.

Fig. 3. Vigor router VPN tunnel setup.

Here is system implementation environment as below:

CPU	Memory	Platform	OS type	PBX version
Intel i7 3.5G	32G	VMware Server 2.0	CentOS ver. 5.3	Asterisk V10

Vigor Router VPN Tunnel Setup is shown in Fig. 3.

The VPN Tunnel Connection Status is shown as below Fig. 4:

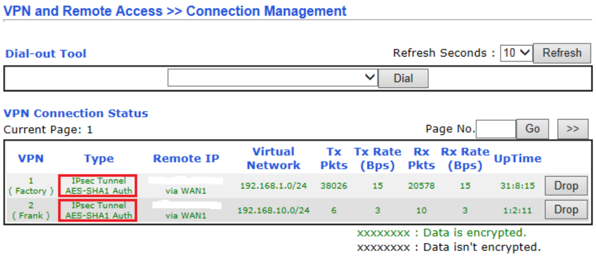


Fig. 4. The VPN tunnel connection status.

Dial plan for Linksys SPA3000 [7]:

( )	Pack all calling rules
|	Separate different numbers
x	It could be any number
x.	Any number appearing repeatedly
<25-7>	Any number among 2, 5, 6 or 7
<03:613>	Substitute 03 with 613
,	Calling tone
!	Calling prohibited
S0	Calling now
<:@gw0>	Calling through gateway 0

The FXO dial plan setup is shown in Fig. 5 below:

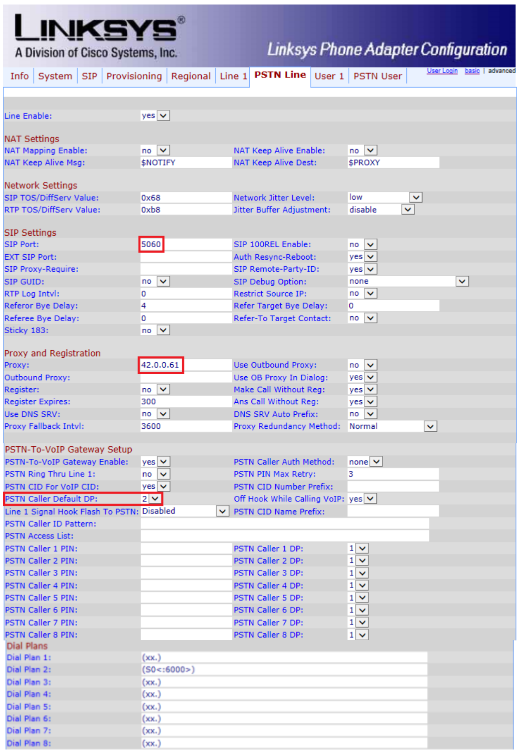


Fig. 5. FXO dial plan setup.

The FXS dial plan setup is shown in Fig. 6 below:

Fig. 6. FXS dial plan setup.

Analog phone calling X-Lite VoIP is shown as Fig. 7 below:

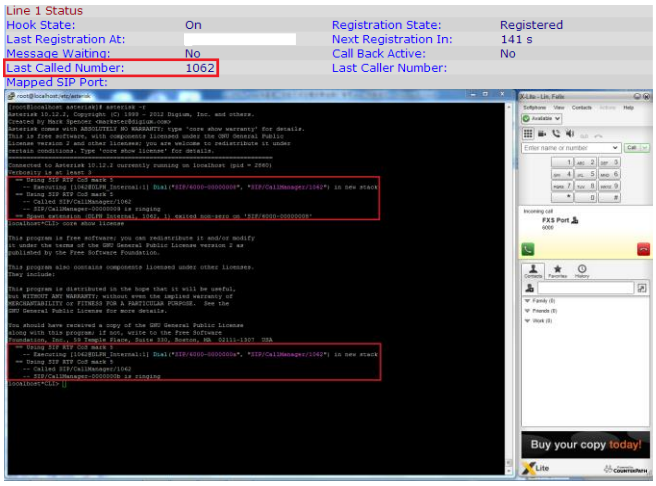


Fig. 7. Analog phone calling X-Lite VoIP.

Benefits Analysis

Cisco Extension Mobility [8] feature is the IP phone in the same cluster group, user use account name and password login to any IP phone as your extension number, through predefined Extension Mobility feature, the functional architecture diagram is show on below Fig. 8:

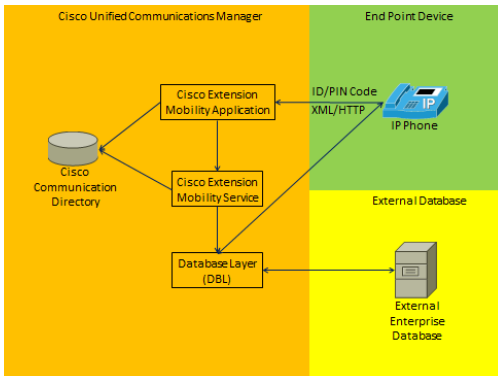


Fig. 8. The architecture of Extension mobility.

The mobile VoIP gateway design is similar with the Cisco Extension Mobility but it more flexible than the Extension Mobility, Cisco Extension Mobility either SCCP or SIP applies only to Cisco IP phone, through the pre-configured mobile VoIP gateway that allows any traditional analog phone become into an extension of the company, and is compatible with any SIP protocol support of IP PBX, such as the Asterisk we mentioned in this article, 3CX SIP Server, Microsoft Lync Server, and even Cisco CallManager Server is compatible with SIP protocol as well, here is the comparison sheet about telephone mobility gateway and the Extension Mobility are shown below in Table III:

TABLE III: COMPARISON SHEET OF TELEPHONE MOBILITY AND EXTENSION MOBILITY

IP PBX	Cisco CallManager	Asterisk Server	3CX SIP Server	Microsoft Lync 2010
Type				
Extension Mobility / Cisco IP phone	Yes / SCCP or SIP	Not compatible	Not compatible	Not compatible
Telephone Mobility / Linksys SPA3000	Yes	Yes	Yes	Yes

IV. CONCLUSION

In this paper, we propose using SIP protocol running under VPN Tunnel. Then the VoIP gateway is registered back to Asterisk via Internet to realize telephone mobility. Asterisk manages the calling rules of the extensions in the company, and the calling rules setup on Linksys SPA3000 allows the analog phone of FXS port to make phone calls directly through FXO Port and communicate with the extensions under the original switch. In this way, no one will need to use computer at any place. Simply by following some installation steps, the analog phone could be converted into company extension through gateway, which takes advantage of convenience and low cost of VoIP, and plays no impacts on the communication between the analog phone and the original PBX. SIP is a standard communication protocol on Internet, which is widely applied in the business networks now.

For other more valuable value-added applications such as voice conference and voice mail, or the solution combining server specification with VMware, with this application, they could use IP PBX to perform server redundancy and support larger VoIP network through cloud computation.

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