

The Administrator Guide Of SIP Phone



Specification Note

In this manual, the “★” mark indicates that the specification or feature described there may be not available yet and is reserved for the future.

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Chapter 1

1. The Introduction For Administration

The SIP phone provides a variety of functions for user and can support a wide range of application from simple to complicated environment. Therefore, it needs an Administrator to plan, install and set the whole environment of VoIP communications. Especially for a powerful architecture or requirements, the knowledge of modern networking and Internet technologies are essential for the IP phone application.

In principle, the Administration is required to do the following things:

- (1) To understand the architecture, resources, and devices of whole environment which will be involved with the VoIP communications.
- (2) To plan the VoIP network and prepare all necessary materials what will need for or happen to the introduction of the IP phone.
- (3) To build a common setting file for most users.
- (4) To configure each phone and install then into the network.
- (5) And to solve the problems what users encounter during operation.

For (3), (4) and (5) items above, the Administrator could utilize the following items in the Table 1.1 to work out.

Table 1.1 The Major Support Items For Administration

	Media	Items for usage	The Description and What to do with
1	Inside the CD-ROM	Firmware image file	To be uploaded to the IP phone. It can be put into the TTP download server.
2	PC's browser	Web Page of IP Phone	To configure the IP phone with a browser running in a PC.
3	PC's DOS Mode	Telnet Command	Use Telnet to communicate, control, monitor and do setting with the IP phone. You may collect trace log with Telnet.
4	The Phone	Menus Mode	For setting major configurations of the phone

1.1 Associating VoIP Phone System With Traditional PBX

At present, there are a lot of companies still using traditional PBX/KTS as the voice communication central system of office. And the Administrator is designated to merge the new VoIP phones with the existing PBX/KTS. For a large group of users starting with IP phones, in general consideration, a VoIP Control system will be introduced. The new trend for the VoIP Control system is SIP protocol based.

The SIP Server provides call registration, connect/disconnect, control and extension services, as well as manages all VoIP call activities, like a traditional PBX does. Normally,

the SIP Server must have a fixed real IP address anyway. But the IP phones don't need fixed real IP addresses. The office internal DHCP server could automatically assign virtual IP addresses to the IP phones.

With the association of VoIP system and traditional PBX, you can place different kinds of VoIP calls. In Figure 1.1, there are 3 types of call thru the connection of SIP Server. Path (A) shows a VoIP call from IP phone to another IP phone by extension number. Path (B) shows a VoIP call from IP phone to the attached analog phone of VoIP Gateway by Gateway's extension number. And Path (C) illustrates a VoIP phone calls to an extension analog phone of PBX system.

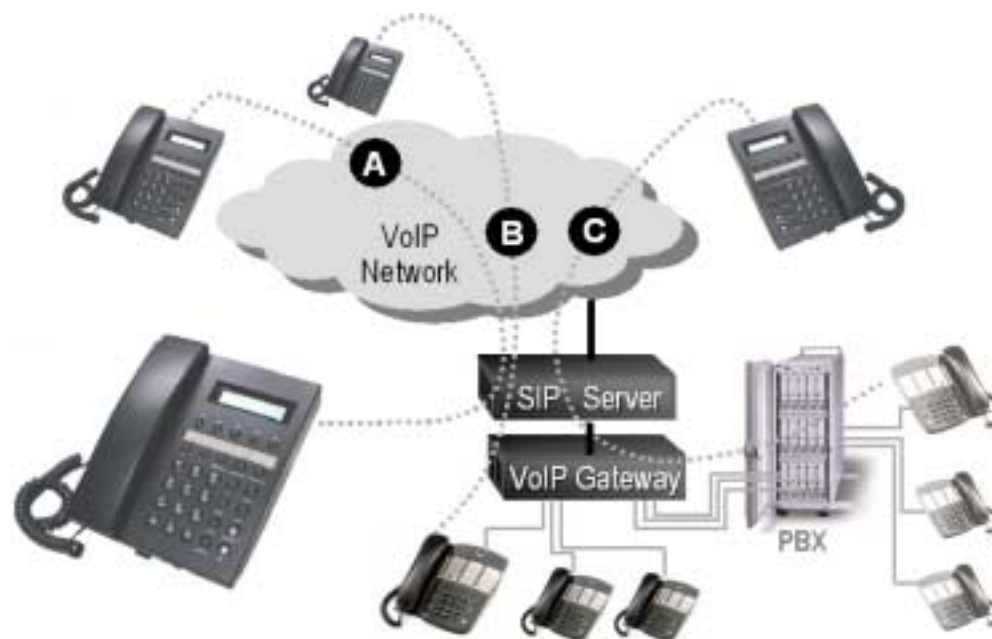


Figure 1.1. The Association of VoIP System With PBX system

1.2 New Application with IP-Enabled PBX System

The communication system architecture of modern enterprise is illustrated in Figure 1.2. IP-enabled PBX system is equipped with VoIP Gateway modules and associated with a SIP Server. As usual, it supports internal Extension lines to traditional analog phones and external Trunks to the PSTN public telephone network. At the same time, it may be installed with a Voice Mail System (VMS) for message recording service. Here, the SIP Server will act as a call agent to control the routes of VoIP calls. Therefore, all users can take the advantages of IP-enabled PBX features as well as VoIP's benefits.

With QoS (Quality of Service) voice priority support on the LAN connection, you also can connect your notebook or PC to the phone. It will keep the voice quality of VoIP calls as usual.

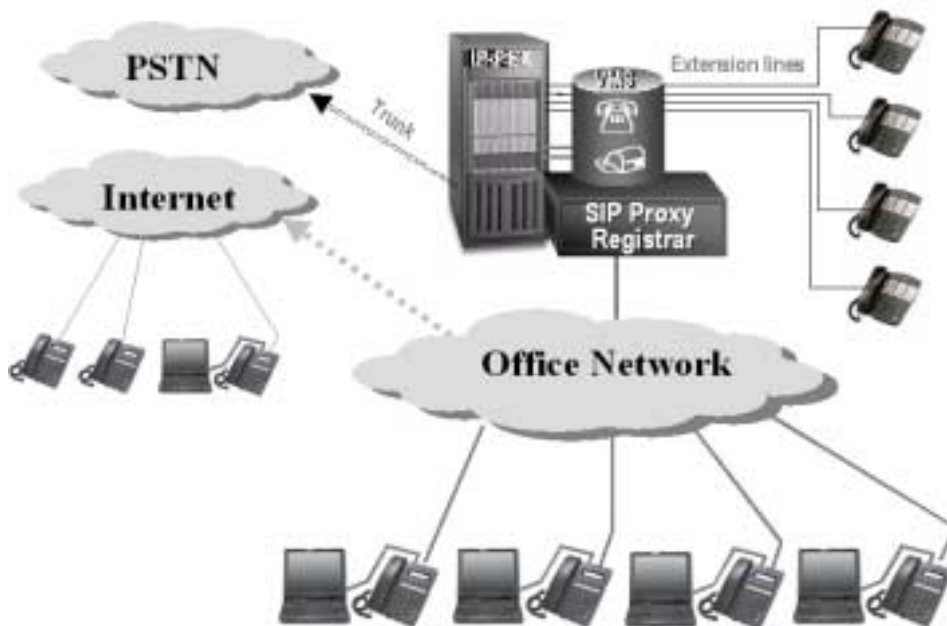


Figure 1.2: The VoIP Network Architecture of Enterprise Office System

1.3 Advanced VoIP Solution in Enterprise Office

For the sake of better management, some enterprises would like to utilize Intranet /Internet Servers in order to improve administration capabilities. And now, this becomes a trend in modern VoIP communication environment.

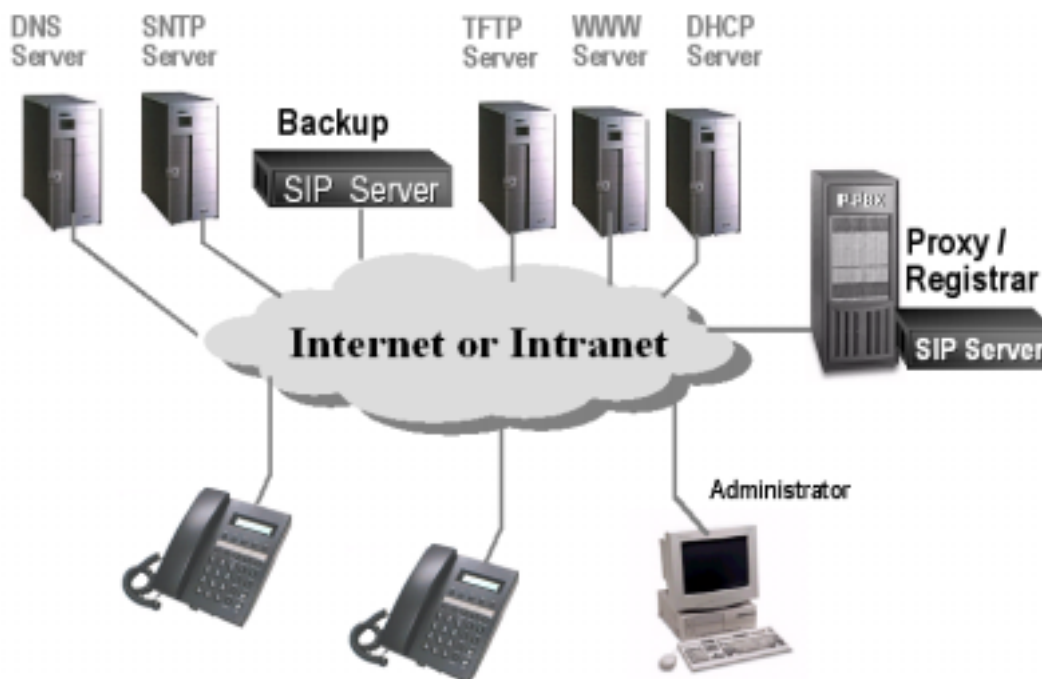


Figure 1.3. The Advanced System Architecture For VoIP Application

There are some general servers, such as DHCP server, WWW (Web) server, FTP/TFTP server, DNS server, SNTP server and SIP Server usually used to support the VoIP communication

service. Different server could offer different function. However, they are all NOT essential equipments but useful for administrator to do management job better. Regarding the function and the protocol of each server, please refer to following table.

PC or Server	To	IP Phone as	Description of Function
Http Client (Browser S/W Running on Administrator PC.)	▶▶ ▶▶ ◀◀	Web Server	(1) Configure the phone through web interface. (2) Download the current profile and save to PC. (3) Upload a profile to the phone. (4) Issue a firmware upgrade through web page.
SIP Server	◀◀	Endpoint	Phone will register to the SIP server. All phones register to the same server can call each other by number. [Remarks]: It is recommended for enterprise or office user.
DHCP Server	▶▶	DHCP client	IP Phone can lease an IP address from DHCP server and configure the network settings according to the DHCP server's response.
DNS Server	◀◀	DNS client	All SIP related address could be a domain name. The IP Phone will do a query to DNS server for the real IP address of this domain name.
SNTP Server	◀◀	SNTP client	The IP Phone can automatically synchronize its system time with the NTP server.
FTP Server	▶▶	FTP client	For downloading firmware and automatic configuration.
Telnet client (Telnet S/W Running on Administrator PC.)	▶▶	Telnet Server	PC to use Telnet command to manage IP phone [Remarks]: It is an optional but very common for Administrator. Telnet can get more data and report from the IP phone. Besides, Troubleshooting engineer likes to use it for issuing debugging commands.

Chapter 2

2. The Definition of Parts

2.1 Definition of Keys

MENUS



[MENUS]

Press this key to enter Menu mode for configuring the IP Phone. Press this key again to Save and/or Exit from the Menu mode.



Soft keys

There are 3 soft keys with vertical lines connecting to the LCD screen. With these soft keys, user may select the desired corresponding item from which shows on the screen. For next group of selectable items, user may press the corresponding key of “Next”.



[τ] [v]

These 2 keys are located inside the Soft key array. Usually, these two keys is used in menu mode as the navigation keys to browse previous / next available options. Sometimes, it will be used as increase/decrease volume in menu mode. During editing

string or IP address, the  will be used as backspace.

Outside Menu mode, they can be used to adjust the volume of speakerphone or handset, when the handset has been picking up or using speakerphone.



[↵] (Same as ENTER)

This key is also located in soft key array. Press this key to Enter edit mode or confirm the setting in current page.

CANCEL



[CANCEL]

Press this key to Cancel current setting(s) and go back to the upper level menu without change. During dialing phase, this key is also used as a backspace.

MUTE



[MUTE]

Press this key to mute the microphone.

XFR/CNF

**[XFR / CNF]**

Press this key to transfer the current call to another one. Currently this button is for transfer function only.

HOLD

**[HOLD]**

Press this key to hold current call and pick up another line.

SPKR

**[SPKR]**

Press this key to pick up the phone in hand-free mode. If the key is already in hand-free mode, pressing this key will hang up the phone.

REDIAL

**[REDIAL]**

Press this key to invoke the last number dialed and make call immediately.

MSG

**[MSG](* Reserved)**

Press this key to retrieve the voice mailbox.

**Programmable Feature buttons**





There are 5 buttons located above the numeric keypad and without any printing above. Starting from the most left one, these buttons are referred to Key P1, Key P2, ... to Key P5 respectively. The detail description of available functions will be shown in section 2.3.

2.2 The Definition of LED Status

On the panel of phone, there are two LED indicators. One is associated with **SPKR** key (referred as LED1) and the other is marked “**MSG / LINE**”(referred as LED2). The first one LED is mainly for the status display of Line # 1. And the second LED is for the status display of Line # 2 and Message Waiting. The following table describes the detailed definition of LED status.

Table 2.1. The Definitions Of LED Status

	RED	Green
--	-----	-------

	 Red Steady	 Red Flashing	 Green Steady	 Green Flashing
MSG / LINE (LED2)	<ul style="list-style-type: none"> ◆ Downloading ◆ Booting 	<ul style="list-style-type: none"> ◆ Line 2 has a holding call or incoming call. (When engage line 1) ◆ Message Waiting (When IDLE) 	<ul style="list-style-type: none"> ◆ Line 2 is engaging 	N/A
SPKR (LED1)	<ul style="list-style-type: none"> ◆ Downloading ◆ Booting 	<ul style="list-style-type: none"> ◆ Line 1 has a holding call or incoming call. (When engage line 2) 	<ul style="list-style-type: none"> ◆ Line 1 is engaging 	<ul style="list-style-type: none"> ◆ Ringing ◆ DND, Call Forward

[Remark] Line #1 status is always shown on the “SPKR” LED while the Line #2 status is always displayed on the “MSG / LINE” LED. And when a line is held, the LED is turned to flashing. When the line is resumed again, the LED will light steadily again.

2.3 The Definition of Programmable Features

The programmable features can be enabled and mapped to the programmable feature buttons on the phone panel. The Administrator (or user) may configure these features via Menus, Web-management, Telnet, or Administrator tool. The following list shows basic programmable features for selection.

(1) Speed Dial:

This function aim to provide a shortcut for users to access most frequently used numbers or important numbers. Up to 10 sets of numbers mapping to the number keys (0,1...9). User may invoke predefined numbers by a speed dial key plus a number key. For example, the programmable button P1 is programmed to perform Speed Dial function and the predefined number for keypad 1 is defined as “100”. User can click P1 then click keypad 1 to invoke the speed dial number “100”. User will see a number “100” show on the screen. The following operation is exactly the same as he just dialed the three digits (“100”) one by one.

(2) Call Record:

This function provides a way to review the call history stored in memory. There are 3 different types of call history the outgoing calls, the incoming calls and missed calls. The call history records the latest 10 sets of numbers for each type. User may review the record, maintain the record (delete single number or delete entire record) and dial the number in the record.

(3) Phone Book:

When pressing this programmable button, the phone book is prompted to the Menus display. A maximum of 100 sets of phone number can be stored in the phone book. User may view, modify and delete the phone number (or IP address) and its associated Name.

(4) Call Forward:

With this feature, the IP phone will execute the extension service of “Call Forward”. The call-forward function will be invoked based on 3 different conditions. First, unconditional forward, user may enable this and all incoming call will forward to another number defined by the user immediately. Second, forward on busy, as the name explained, the incoming call will be forward to another number defined by the user as long as all lines on this phone is occupied. Third, forward on no answer, user may enable this function and all incoming call will forward to another number defined by the user when the incoming call was not answered for a certain period of time (Also defined by the user). With this call-forward feature, user can implement a so-called “follow me” on this phone without the SIP server support.

(5) Conference:

By selecting this function this phone can bridge two existing calls and do a 3-Way conference through this button.

(6) Lock IP-Phone:

Upon pressing this key, the phone will jump to lock state. User will need to provide his password to unlock the phone before he can access the service. This function aim to provide a way to protect the phone form unauthorized user.

(7) Use DND:

By selecting this function user can turn on/off Do Not Disturb (DND) function by a single click on this button.

(8) HOT Speeddial:

By selecting this function user can immediately place a call to predefined number by a single click on this button. The predefined numbers are the first 5 entries of the speed dial numbers. (Speed dial 0 to Speed dial 4) The predefined hot-speed-dial number for the programmable button 1 is the speed dial 0 and the predefined hot-speed-dial number for the programmable button 5 is the speed dial 4.


Chapter 3

3. The Administration Setting in Menus

The LCD Screen provides 2 lines information with a total of 32 characters for the operation of Menus. Administrator may use it to configure the IP phone. For detailed descriptions, please refer to User's Manual of IP phone. In this Guide, only the portion related with Administration will be introduced. You may press [MENUS] key on the phone to enter Menus mode for Administrative configuration. Basically, the menu operation is exactly the same as user mode menu. Except, the [Admin] menu for administrator has more advance settings. Here we focus on the advanced settings for administrator, please refer to the user's manual for regular settings.

[MENUS OPERATION]

	Menu Selection: TCPIP Admin Next	TCPIP: TCPIP network settings. Admin: The advance setting goes here. [CANCEL]: to go back to the upper page.
--	--	---

To enter Administration page, please select [**Admin**] section by pressing  in this menu. You have to provide password to enter this section. For the administrator, please enter the administrator's password. The Admin section will unveil much more advance settings than using user's password.

[Admin]:

A password is required for entering the Administration mode.

Index	LCD Screen	Description of Action
0	Admin Password: ■ Error, Password: ■	Administrator authentication: The factory default password is [1234] If password is matched, the following page will be entered. If Password is wrong, this screen will show up and prompt you to enter password again. [CANCEL]: to go back to the upper page
1 2	Admin Setting: Phone Acct Next	1'st Administration menu. [τ]: Phone: Configure the phone number of this phone. [←]: Acct: Configure the account of this phone. [v]: Next: Next Administration menu. [CANCEL]: to go back to the upper page.

3 4	Admin Setting: NAT SIP Next	2'nd Administration menu: [τ]: NAT: Configure the NAT traversal mechanism. [↵]: SIP: Configure the SIP settings of this phone. [v]: Next: Next Administration menu. [CANCEL]: to go back to the upper page.
5	Admin Setting: Program Next	3'rd Administration menu: [τ]: Program: Configure the function of 5 programmable buttons. [↵]: No action. [v]: Next: Next Administration menu. [CANCEL]: to go back to the upper page.
6	Admin Setting: Download Next	4'th Administration menu: [τ]: Download: Download firmware. [↵]: No action. [v]: Next: Next Administration menu. [CANCEL]: to go back to the upper page.
7 8	Admin Setting: Reset Boot Home	5'th Administration menu: [τ]: Reset: Reset the setting to factory default. [↵]: Reboot the phone. [v]: Go to 1'st Administration menu. [CANCEL]: to go back to the upper page.

Phone number setting

1	Edit Phone No: 100■	Edit the phone number of this phone. [τ]: Backspace. [↵]: Confirm the current editing result. [v]: No action. [CANCEL]: to go back to the upper page.
---	------------------------	---

Account setting

2	Account Setting:	Edit the phone number of this phone.
2.1	Admin User	[τ]: Backspace.
2.2		[↵]: Confirm the current editing result. [v]: No action. [CANCEL]: to go back to the upper page.
2.1	Admin Setting:	Edit the administrator user name and password of this phone.
2.1.1	Name Password	[τ]: Name: Administrator user name.
2.1.2		[↵]: Password: Administrator password. [v]: No action. [CANCEL]: to go back to the upper page.
2.1.1	Name Setting: admin■	Edit the administrator user name [τ]: Backspace. [↵]: Confirm and save current editing result.

- 2.1.2 Change Password:
New: ■

[v]: No action.
[CANCEL]: to go back to the upper page.
Change administrator password:
[τ]: Backspace.
[↵]: Confirm and save current editing result.
[v]: No action.
- 2.1.2.1 Confirm Again:
New: ■

[CANCEL]: to go back to the upper page.
Confirm the new password:
[τ]: Backspace.
[↵]: Confirm current editing result. If the confirmed password and the new password are the same, the 2.1.2.3 shows up. If they are not consistent, the next screen shows up.
- 2.1.2.2 Password Error:
Retry Cancel

[v]: No action.
[CANCEL]: to go back to the upper page.
The two input are not consistent:
[τ]: Retry: Go to 2.1.2.
[↵]: No action.
[v]: Cancel: Go to 2.1.
- 2.1.2.3 Password
Change OK Back

[CANCEL]: to go back to the upper page.
Password changed OK.
[τ]: No action.
[↵]: No action.
[v]: Back: Go to 2.1.
- 2.2 User Setting:
2.2.1 Name Password
2.2.2

[CANCEL]: to go back to the upper page.
Edit the administrator user name and password of this phone.
[τ]: Name: Administrator user name.
[↵]: Password: Administrator password.
[v]: No action.
- 2.2.1 Name Setting:
user ■

[CANCEL]: to go back to the upper page.
Edit the user username
[τ]: Backspace.
[↵]: Confirm and save current editing result.
[v]: No action.
- 2.2.2 Change Password:
New: ■

[CANCEL]: to go back to the upper page.
Change user password:
[τ]: Backspace.
[↵]: Confirm and save current editing result.
[v]: No action.
- 2.2.2.1 Confirm Again:
New: ■

[CANCEL]: to go back to the upper page.
Confirm the new password:
[τ]: Backspace.
[↵]: Confirm current editing result. If the confirmed password and the new password are the same, the 2.2.2.3 shows

		up. If they are not consistent, the next screen shows up.
		[v]: No action.
		[CANCEL]: to go back to the upper page.
2.2.2.2	Password Error: Retry Cancel	The two input are not consistent:
		[τ]: Retry: Go to 2.2.2.
		[↵]: No action.
		[v]: Cancel: Go to 2.2.
		[CANCEL]: to go back to the upper page.
2.2.2.3	Password Change OK Back	Password changed OK.
		[τ]: No action.
		[↵]: No action.
		[v]: Back: Go to E.1.2.1.
		[CANCEL]: to go back to the upper page.
NAT setting		
3	TCP/IP NAT:	1'st NAT setting page.
3.1	NATTyp STUN Next	[τ]: NATTyp: Select the NAT traversal mechanism.
3.2		[↵]: Configure STUN related settings.
		[v]: Next NAT setting page.
		[CANCEL]: to go back to the upper page.
3.1	NAT Type: < None >	Select the NAT traversal mechanism:
		[τ]: Previous option.
		[↵]: Confirm and save current selection.
		[v]: Next option.
		The available options are:
		◆ None
		◆ STUN
		◆ SIP PING
		◆ Port Mapping
		◆ UDP Heartbeat
		[CANCEL]: to go back to the upper page.
3.2	Edit STUN Server:	STUN setting page.
3.2.1	IP IP Port Back	[τ]: IP: STUN server address.
3.2.2		[↵]: Port: STUN server port.
		[v]: Back: Go back to 3.1.
		[CANCEL]: to go back to the upper page.
3.2.1	Edit STUN Server 0.0.0.0█	Configure the STUN server.
		[τ]: Backspace.
		[↵]: Confirm and save current editing result.
		[v]: No action.
		[CANCEL]: to go back to the upper page.
3.2.2	STUN Server Port 3478█	Configure the STUN server port.
		[τ]: Backspace.
		[↵]: Confirm and save current editing result.
		[v]: No action.

TCP/I		[CANCEL]: to go back to the upper page.
P		2'nd NAT setting page.
NAT:		[τ]: SPing: SIP PING related settings.
SPing		[↵]: PMap: Port mapping related settings.
PMap		[v]: Next NAT setting page.
Next		[CANCEL]: to go back to the upper page.
3.3	SIPPing Time(ms) 10000█	Configure the SIP PING interval. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action.
3.4	Edit Port Map	[CANCEL]: to go back to the upper page.
3.4.1	Esrv Signal Next	1'st Port mapping setting page.
3.4.2		[τ]: Esrv: External router address. [↵]: Signal: External SIP signaling port. [v]: Next Port mapping setting page.
3.4.1	Edit Extern Srv 61.222.54.166█	[CANCEL]: to go back to the upper page. Configure the external router address. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action.
3.4.2	Extrn signalport 36000█	[CANCEL]: to go back to the upper page. Configure the external signaling port. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action.
3.4.3	Edit Port Map	[CANCEL]: to go back to the upper page.
3.4.4	Rtp1 Rtp2 Back	2'nd Port mapping setting page. [τ]: Rtp1: External RTP port for channel 1. [↵]: Rtp2: External RTP port for channel 2. [v]: Back: Back to first Port mapping setting page.
3.4.3	Extern RTP Port1 36010█	[CANCEL]: to go back to the upper page. Configure the external RTP port for channel 1. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action.
3.4.4	Extern RTP Port2 36020█	[CANCEL]: to go back to the upper page. Configure the external RTP port for channel 2. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action. [CANCEL]: to go back to the upper page.

3.5	TCP/IP NAT: OpHeader	Back	3'rd NAT settings page [τ]: OpHeader: Optional header settings page. [↵]: No action. [v]: Back: Back to 1'st NAT settings page. [CANCEL]: to go back to the upper page.
3.5 3.5.1 3.5.2	NAT Option Head OpHead1 OpHead2		3'rd NAT settings page [τ]: OpHeader: Optional header settings page. [↵]: No action. [v]: Back: Back to 1'st NAT settings page. [CANCEL]: to go back to the upper page.
3.5.1	Optional Header1 ■		Configure first optional header. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action. [CANCEL]: to go back to the upper page.
3.5.2	Optional Header2 ■		Configure second optional header. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action. [CANCEL]: to go back to the upper page.
SIP setting			
4	SIP Setting:		1'st SIP setting page.
4.1	Srv Authen Next		[τ]: Srv: Configure SIP related servers.
4.2			[↵]: Authen: Configure the authentication id and password. [v]: Next SIP setting page. [CANCEL]: to go back to the upper page.
4.1	SIP Server Setup:		1'st SIP server setting page.
4.1.1	Proxy OBSrv Next		[τ]: Proxy: Configure the proxy (domain).
4.1.2			[↵]: OBSrv: Configure the outbound proxy. [v]: Next SIP server setting page. [CANCEL]: to go back to the upper page.
4.1.1	Edit Proxy Addr: 192.168.2.17■		Configure proxy. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action. [CANCEL]: to go back to the upper page.
4.1.2	Outbound proxy : 192.168.2.18■		Configure outbound proxy. [τ]: Backspace. [↵]: Confirm and save current editing result. [v]: No action. [CANCEL]: to go back to the upper page.
4.1.3	SIP Server Setup:		2'nd SIP server setting page.
4.1.4	Reg. OutReg Port		[τ]: Reg.: Configure the REGISTRAR.
4.1.5			[↵]: OutReg: Configure the outbound REGISTRAR.

- 4.1.3 Registrar Server:
192.168.2.17█

[v]: Port: Configure the server port.
[CANCEL]: to go back to the upper page.
Configure REGISTRAR.
[τ]: Backspace.
[↵]: Confirm and save current editing result.
[v]: No action.
- 4.1.4 Outbound Registr
192.168.2.18█

[CANCEL]: to go back to the upper page.
Configure outbound REGISTRAR.
[τ]: Backspace.
[↵]: Confirm and save current editing result.
[v]: No action.
- 4.1.5 SIP Server Port
4.1.5.1 Local Srv Back
4.1.5.2

[CANCEL]: to go back to the upper page.
Configure SIP related port setting.
[τ]: Local: Local port.
[↵]: Srv: Server Port.
[v]: Back: Back to first SIP server setting page.
- 4.1.5.1 SIP Local Port
5060█

[CANCEL]: to go back to the upper page.
Configure SIP local port (client port).
[τ]: Backspace.
[↵]: Confirm and save current editing result.
[v]: No action.
- 4.1.5.2 Outbound Port
5060█

[CANCEL]: to go back to the upper page.
Configure SIP server port.
[τ]: Backspace.
[↵]: Confirm and save current editing result.
[v]: No action.
- 4.3 SIP Setting:
4.4 DTMF CODEC Next

[CANCEL]: to go back to the upper page.
2'nd SIP setting page.
[τ]: DTMF: Configure the DTMF option.
[↵]: CODEC: Configure the CODEC.
[v]: Next SIP setting page.
- 4.3 DTMF Type:
< RFC2833 >

[CANCEL]: to go back to the upper page.
DTMF configuration:
[τ]: Previous option.
[↵]: Confirm and save current selection.
[v]: Next option.
The options are:
◆ NONE
◆ Tone In Band
◆ RFC2833
◆ RFC2833+Tone
◆ SIP INFO
[CANCEL]: to go back to the upper page.

- 4.4 Codec Setting:
 <G.711u-law>
- Preferred Codec settings:
You have to select a specific codec to set the priority of them one by one.
[τ]: Previous codec.
[↵]: Go on to modify the priority of this codec.
[v]: Next codec.
The codecs are:
◆ G.711u-law
◆ G.711a-law
◆ G.723.1
◆ G.729
- 4.4.1 G711u-law Pri:
 < First >
- [CANCEL]: to go back to the upper page.
Change the codec priority. (In this case we change the priority of G.711u-law).
[τ]: Previous option.
[↵]: Confirm and save current selection.
[v]: Next option.
Currently we have 4 different codecs: So the valid options are:
◆ NONE (Don't use)
◆ First
◆ Second
◆ Third
◆ Forth
- Once, you change the priority of one codec, the other codecs will change their priority accordingly.
[CANCEL]: to go back to the upper page.
- 4.5 SIP Setting:
4.6 Ptime CalID Next
- 3'rd SIP setting page.
[τ]: Ptime: Configure the packet time.
[↵]: CalID: Configure the caller-ID.
[v]: Next SIP setting page.
- 4.5 Ptime Setting:
 < 20ms >
- [CANCEL]: to go back to the upper page.
Packet time setting
[τ]: Previous option.
[↵]: Confirm and save current selection.
[v]: Next option.
The valid options are:
◆ 10ms
◆ 20ms
◆ 30ms
◆ 40ms
◆ 50ms
◆ 60ms
[CANCEL]: to go back to the upper page.

<p>4.6 Caller ID Setting</p> <p>4.6.1 Enable Name Back</p> <p>4.6.2</p>	<p>Caller ID related setting</p> <p>[τ]: Enable/Disable sending caller ID.</p> <p>[↵]: Configure the DisplayName.</p> <p>[v]: Back to 3'rd SIP setting page.</p> <p>CANCEL]: to go back to the upper page.</p>
<p>4.6.2 Caller ID Name:</p> <p>█</p>	<p>Configure the DisplayName.</p> <p>[τ]: Backspace.</p> <p>[↵]: Confirm and save current editing result.</p> <p>[v]: No action.</p> <p>CANCEL]: to go back to the upper page.</p>
<p>4.7 SIP Setting:</p> <p>4.8 UrlName Back</p>	<p>4'th SIP setting page.</p> <p>[τ]: UrlName: Configure username in SIP URL.</p> <p>[↵]: No action.</p> <p>[v]: Back to first SIP setting page.</p> <p>CANCEL]: to go back to the upper page.</p>
<p>4.7 UrlName Setting:</p> <p>█</p>	<p>Configure the username used in the SIP URL.</p> <p>[τ]: Backspace.</p> <p>[↵]: Confirm and save current editing result.</p> <p>[v]: No action.</p> <p>CANCEL]: to go back to the upper page.</p>
<p>Configure programmable button features</p>	
<p>5 Program Key:[P1]</p> <p>< Speed Dial ></p>	<p>Browsing the current setting of programmable buttons. The upper right corner is the index of programmable button (P1~P5).</p> <p>[τ]: Pervious programmable button.</p> <p>[↵]: Configure the selected programmable button.</p> <p>[v]: Next programmable button.</p> <p>CANCEL]: to go back to the upper page.</p>
<p>5.1 Edit Program:[P1]</p> <p>< Speed Dial ></p>	<p>Edit selected programmable button</p> <p>[τ]: Pervious function.</p> <p>[↵]: Confirm to set the selected function to this programmable button.</p> <p>[v]: Next function.</p> <p>The functions are:</p> <ul style="list-style-type: none"> ◆ Speed Dial ◆ Call Record ◆ Phone Book ◆ Call Forward ◆ Conference ◆ Lock Phone ◆ Used DND ◆ Hot Speed Dial <p>CANCEL]: to go back to the upper page.</p>

Download Settings

6	Download Setting	Firmware and profile download setting:
6.1	FTP Image Profil	[τ]: Configuration FTP related settings.
6.2		[↵]: Download firmware image.
6.3		[v]: Configure the profile file name.
		CANCEL]: to go back to the upper page.
6.1	FTP Server:	FTP related setting:
6.1.1	IP User Pswd	[τ]: Configure FTP server address.
6.1.2		[↵]: Configure the username for FTP account.
6.1.3		[v]: Configure the password for FTP account.
		CANCEL]: to go back to the upper page.
6.1.1	FTP Server IP:	Configure the FTP server address
	192.168.5.20█	[τ]: Backspace.
		[↵]: Confirm and save current editing result.
		[v]: No action.
		CANCEL]: to go back to the upper page.
6.1.2	FTP User:	Configure the FTP username
	root█	[τ]: Backspace.
		[↵]: Confirm and save current editing result.
		[v]: No action.
		CANCEL]: to go back to the upper page.
6.1.3	FTP Password:	Configure the FTP password
	█	[τ]: Backspace.
		[↵]: Confirm and save current editing result.
		[v]: No action.
		CANCEL]: to go back to the upper page.
6.2	Download Image	Download image:
	Yes No	[τ]: Yes: Start downloading the firmware.
		[↵]: No: Back to previous page.
		[v]: No action.
		CANCEL]: to go back to the upper page.
6.3	Download Image	Download image:
	Yes No	[τ]: Yes: Start downloading the profile.
		[↵]: No: Back to previous page.
		[v]: No action.
		CANCEL]: to go back to the upper page.
Reset to factory default setting		
7	Reset Default:	Reset to factory default:
	Yes No	[τ]: Yes: Reset to factory default.
		[↵]: No: Back to previous page.
		[v]: No action.
		CANCEL]: to go back to the upper page.

Boot

7 Warn Boot :
Yes No

Reboot the phone

[τ]: Yes: Reboot the phone.

[↵]: No: Back to previous page.

[v]: No action.

CANCEL]: to go back to the upper page.

Chapter 4

4. How to Configure IP Phone Through Web Page

Accessing the phone through web browser, just simply enter the “http://192.168.1.10” in the location field of the browser. (If you are not sure about the IP address, you can examine the current IP address through Info menu.)

4.1 Login:

The following dialog box will pop up and prompt you to provide the user name and password in order to prevent unauthorized user access the phone. Please enter the administrator’s username and password.



Link to 192.168.1.10

SIP Phone

Username: (U)

Password: (P)

To remember your password (R)

Confirm Cancel

4.2 Default Page:

The following is the default page you will see when you login the phone's web page.

SIP Phone IP Phone

Phone Personal PhoneBook TCP/IP SIP System Reboot

Version: 1.0.08

Phone Settings			
Handset Mic	7	Handset Speaker	8
Speaker Mic	7	Speakerphone	9
Ring Tone Volume	13	Side Tone Volume	2
Enable Echo Canceller	<input checked="" type="checkbox"/>	Enable VAD+CNG	<input type="checkbox"/>
Tone Type	User Defined	Ring Type	User Defined
DTMF Relay	RFC2833+Tone	Enable DND	<input type="checkbox"/>
Enable Auto Answer	<input type="checkbox"/>	Enable Phone Lock	<input checked="" type="checkbox"/>
Programmable Button Settings			
Programmable Button 1	Lock IP-Phone		
Programmable Button 2	Speed Dial		
Programmable Button 3	Speed Dial		
Programmable Button 4	Speed Dial		
Programmable Button 5	Conference		

4.3 Administrator's Menu:

The upper area of the web page is the menu of all configuration categories. You can click on the menu items to select different group of settings.

SIP Phone IP Phone

Phone Personal PhoneBook TCP/IP SIP System Reboot

Version: 1.0.08

When mouse pointer move over the menu item, the menu item will highlighted and the cursor will change to a hand indicating this item can be clicked.

SIP Phone IP Phone

Phone Personal PhoneBook TCP/IP SIP System Reboot

Version: 1.0.08

When you click on the menu item, the web page will change to different setting page.

Basically, there are only two pages that are different to what user can see through web. The first one is system page. The system page for administrator is much more complicated than user's one. All major maintenance functions are available only to the administrator. The second different page is the SIP page. Only administrator can access this page. Other regular pages have been described in detail in the user's manual. Please refer to the user's manual for other pages. In this manual we focus on the administrator specific function that can be performed through web.

4.4 Administrator's system page

Account Settings			
Administrator Name	<input type="text" value="admin"/>	Administrator Password	<input type="password" value="****"/>
User Name	<input type="text" value="user"/>	User Password	<input type="password" value="****"/>
NAT Settings			
NAT Type	<input type="text" value="None"/>		
STUN Items			
STUN Server IP	<input type="text" value="0.0.0.0"/>	STUN Server Port	<input type="text" value="3478"/>
SIP Ping			
SIP PING Interval Time(ms)	<input type="text" value="6000"/>		
Optional SIP Header			
Optional Header 1	<input type="text"/>	Optional Header 2	<input type="text"/>
Port Mapping			
Extern Router IP	<input type="text"/>	Extern Signal Port	<input type="text" value="5060"/>
Extern RTP Port 1	<input type="text" value="8006"/>	Extern RTP Port 2	<input type="text" value="8008"/>
System Optional Items			
Enable Multi-Line	<input checked="" type="checkbox"/>	Enable Conference	<input checked="" type="checkbox"/>
Enable MM	<input type="checkbox"/>	Voice Mail Server	<input type="text"/>
Enable QoS	<input type="checkbox"/>	VLAN ID	<input type="text" value="0x000"/>
VLAN Priority	<input type="text" value="0"/>	TOS/Differentiated Services	<input type="text" value="0x00"/>
Provisioning Settings			
Enable Auto Provisioning	<input type="checkbox"/>		
FTP Server IP Address	<input type="text" value="192.168.5.20"/>	FTP Directory	<input type="text"/>
FTP User Name	<input type="text" value="root"/>	FTP Password	<input type="password" value="****"/>
FTP Image Name	<input type="text" value="IpPhone.bin"/>	FTP Profile Name	<input type="text" value="0003c93043e8.txt"/>

Select Profile to Upload:

Here is the description of each field.

Field Name	Function
Administrator Name	The administrator username.
Administrator Password	The administrator password.
User Name	The user username.
User Password	The user password.
NAT Type	NAT traversal mechanism selection.
STUN Server IP	STUN server address.
STUN Server Port	STUN server port.

SIP PING Interval Time(ms)	SIP PING frequency.
Optional Header 1	Optional SIP header in message.
Optional Header 2	Optional SIP header in message.
Extern Router IP	The external router address for port mapping
Extern Signal Port	The external SIP signaling port for port mapping.
Extern RTP Port 1	The external RTP port for port mapping.
Extern RTP Port 2	The external RTP port for port mapping.
Enable Multi-Line	Enable the phone to use 2-lines. (default enabled)
Enable Conference	Enable the phone bridged 3-way conference. (default enabled)
Enable MWI	Subscribe to voice mail server.
Voice Mail Server	Voice mail server address.
Enable QOS	Enable VLAN.
VLAN ID	VLAN ID.
VLAN Priority	VLAN Priority selection.
TOS/Differentiated Services	TOS/Differentiated Services ID.
Enable Auto Provisioning	Enable Auto Provisioning.
FTP server IP Address	FTP server address.
FTP Directory	FTP directory where the firmware and profile located.
FTP User Name	Username of the account on the FTP server.
FTP Password	Password of the account on the FTP server.
FTP Image Name	Firmware file name.
FTP Profile Name	Profile file name.

Buttons:

Save Settings	Save changes in this page to the phone.
FTP Image Update	Click on this button to issue a firmware download.
FTP Profile Update	Click on this button to issue a profile download.
Download Settings	Save the current settings on the phone to local file in PC.
Logout	Logout and close the browser window.

4.4.1 Profile upload field:

Select Profile to Upload:

Buttons:

Save Settings	Save changes in this page to the phone.
Browse	Click this button will invoke the explorer to locate the profile to be uploaded.
Upload Profile	Upload the selected profile to phone. The phone will modify the settings according to the profile.

4.4.2 Feature Key field

Feature key
<input type="text"/> <input type="button" value="Send Key"/>

Administrator can input specific keys to turn on some optional features.

4.5 SIP page:

SIP Settings			
Phone Number	<input type="text" value="5022"/>		
SIP Proxy Server	<input type="text" value="192.168.2.17"/>	Outbound Proxy Server	<input type="text"/>
Server Port	<input type="text" value="5060"/>	Local Port	<input type="text" value="5060"/>
Authorized Account Name	<input type="text" value="5022"/>	Authorized Password	<input type="password" value="*****"/>
Registrar Server	<input type="text"/>	Registrar Outbound Server	<input type="text"/>
Registrar Expire Time (sec)	<input type="text" value="60"/>		
Enable Caller ID	<input checked="" type="checkbox"/>	Display Name	<input type="text"/>
User Name For URL	<input type="text"/>		
Media Settings			
RTP Port 1	<input type="text" value="8006"/>	RTP Port 2	<input type="text" value="8008"/>
Codec G.711u-law	<input type="text" value="None"/>	Codec G.711a-law	<input type="text" value="First"/>
Codec G.723.1	<input type="text" value="None"/>	G.723.1 Bit Rate	<input type="text" value="5.3kb/s"/>
Codec G.729	<input type="text" value="None"/>	Media Packet Time	<input type="text" value="20(ms)"/>
<input type="button" value="Save settings"/> <input type="button" value="Cancel"/> <input type="button" value="Logout"/>			

Here is the description of each field.

Field Name	Function
Phone Number	Phone number assigned to the phone.
SIP Proxy Server	SIP proxy server address.
Outbound Proxy Server	Outbound proxy server address.
Authorized Account Name	Authentication username.
Authorized Password	Authentication password.
Registrar Server	Registrar server address.
Registrar Outbound Server	Registrar outbound server address.
Registrar Expire Time	Expiration time for REGISTER request.
Enabled Caller ID	Enable sending Display Name.
Display Name	The Display Name field in SIP message.
User Name For URL	The User field for the SIP URI.
RTP Port 1	RTP Port for channel 1.
RTP Port 2	RTP Port for channel 2.
Codec G.711u-law	Set the preference of G.711u-law.

Codec G.711a-law	Set the preference of G.711a-law.
Codec G.723.1	Set the preference of G.723.1
G.723.1 Bit Rate	Set the bit rate for G.723.1
G.729	Set the preference of G.729
Media Packet Time	Packet Time.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.
Logout	Logout and close the browser window.

Chapter 5

5. Provisioning

5.1 Upgrade Firmware Through FTP

This section describes how to download the image file from FTP server to upgrade the firmware in IP-Phone. The following figure (Figure 5.1) shows the basic framework for firmware upgrade through FTP.

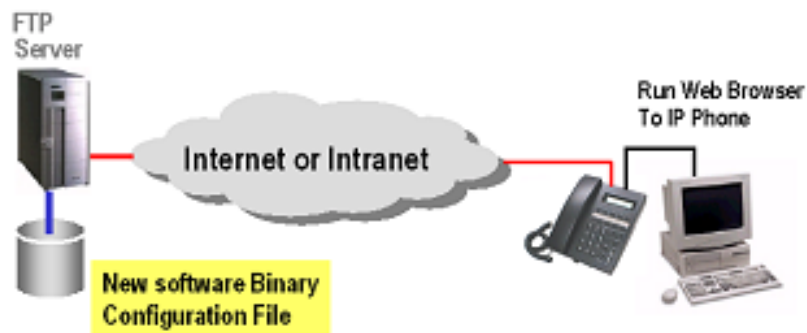


Figure 5.1. The Basic Download Architecture

Prepare the FTP server:

1. Make sure the FTP server has been properly installed in the server.
2. Create an account for this IP-Phone. (You may use pre-created account as long as it still valid.)
3. Put the firmware file in the directory where the specific account has authorization to access.

Configure the phone to be ready for FTP access:

1. Configure the correct FTP server address.
2. Configure the correct account username.
3. Configure the correct account password.
4. Configure the file name to be downloaded.
5. Configure the directory where the file located on the server. If the file is putted in the root directory of this account, clear this field if there is anything.

You can issue a firmware upgrade via WEB or Menus. For using Menus to issue a firmware upgrade please refer to the chapter 3 “Download Settings Menu” (Menu 6). For using web to issue a firmware upgrade please refer to the chapter 4 “Administrator’s system page”. No matter you issue the upgrade through web or menus, the underlying operation is exactly the same, the following is a description of what you will see on the LCD screen in the whole procedure.

At first you will see this screen and the two LEDs will all turn to red steady.

```
UPGRADE PROGRESS  
ERASING FLASH...
```

Later there should be a progress percentage in the second line.

```
UPGRADE PROGRESS  
28%
```

Then, the phone gets the complete firmware and reboot automatically

```
UNREGISTERING...
```

```
Booting...
```

Note: The phone will take a little longer time to reboot this time, because, IP-Phone will do some extra checking to make sure the integrity of the newly downloaded firmware.

Error:

If you see the following screen that means the phone fail to contact the FTP server.

```
CANT CONNECT SRV
```

There are several possibilities:

1. FTP server is not working. (You can PING the server from your computer. You can even login to the server from your computer to verify this.)
2. The account information wrong. (Wrong username or wrong password)
3. Wrong FTP server address.
4. The network configuration of the phone is wrong. (You can issue a PING from IP-Phone to verify this.)

If you see the following screen that means the phone contact to the FTP server, but fail to download file form server.

```
DOWNLOAD ERROR
```

Usually, this is caused by the following reasons:

1. The file is not located in where you expect it to be. You can use your FTP client login to the FTP server (using the same account) to verify this.

2. The file is in position, but the folder setting on the phone is wrong.
3. The file is in position, the folder is correct, but the file name is wrong.
4. FTP server died for any reason. (Very unusual case.)

5.2 Update Profile Through FTP

This section describes how to download the profile of the phone from FTP server to update some specific settings on the IP-Phone. Actually, the procedure is almost the same as upgrade firmware. The only difference is the file to be downloaded from FTP server. Just put the well-formatted settings file on the specific directory on the FTP server and issue a profile download via web or menus. The phone will refresh the settings according to the profile.

The format of the profile will be described in detail in the **Auto Provisioning** chapter.

5.3 Download Current Settings Of IP-Phone

This section describes how to download the current settings form the IP-Phone. This function is only available on the web interface. Please login to the web page (Using administrator account) and select system page. Press the following button in system page:

DownLoad Settings

You will be prompted to provide the file name and location on your PC. After you confirm the dialog box the whole settings of the phone will be saved in your PC. It is like a snapshot of the current settings of the phone. Later, if you want to trace back to the settings of this moment, you can simply upload this file to the phone, and the phone will be configured as if nothing ever changed. Usually, you can use this as a backup of current settings, just incase you lost some important SIP account or settings. You can always use this file to recover the settings.

5.4 Upload Profile Through Web

This section describes how to upload a profile through web interface. Please login to the web page (Using administrator account) and select system page. Look for the following field in the system page. It is right below the form and the buttons.

Select Profile to Upload:

Press the button to locate where you put the profile. After that, press the button

to upload the file through web.

Chapter 6

6.NAT Traversal

6.1 Mechanisms

The NAT traversal has become a very important function for VOIP industry. Since more and more network devices (especially client devices) are behind a NAT, the network devices has to make itself visible outside the NAT. Several mechanisms have been proposed. Basically, all try to keep some specific port mapping unchanged in the NAT mapping table so that outer world can traverse the NAT to communicate with the devices behind the NAT. In the following sections we will discuss those NAT traversal mechanism implemented in our phone.

6.2 STUN (Simple Traversal of UDP through NAT)

The basic framework is to setup a STUN server in the public Internet. STUN client using a so called BINDING request to query whether this client is behind a NAT and if it is behind a NAT, find out the behavior of this NAT. Also, using binding request, STUN client can find out what is the address and port look like form the point of view of STUN server (i.e. From the outer world). To enable STUN, you have to do the following:

1. You can use web or menu to set the NAT type to STUN.
2. Set the STUN server address.
3. Set the STUN server port. (Default setting is 3478).
4. Save the setting and reboot the phone.

6.3 SIP PING

This mechanism is only for keeping the SIP signaling port mapping on the NAT. The SIP server has to support a special optional method “PING”. Whenever the SIP server receives this message it reply a 200 OK. Basically, if the “PING” request is frequent enough, the SIP signaling port will keep mapping to the same port. Usually, the SIP server has to support media route (RTP relay) to relay the RTP. To enable SIP PING, you have to do the following:

1. You can use web or menu to set the NAT type to SIP PING.
2. Set the PING interval.
3. Save setting and reboot the phone.

6.4 Port Mapping

This might be the most efficient and guarantee to be working way. Most of the NAT has a so-called virtual server setting. You can set several ports on the NAT that will be mapped to some specific device behind the NAT. Some NAT can even let you set the transport protocol

(TCP/UDP). Whenever there is a packet receives from one of these ports, the NAT will transfer this packet to the mapped devices. Actually, this is force the NAT to keep the port mapping directly. To enable Port Mapping you have to configure both the NAT and phone settings.

1. Set the virtual server on the NAT to **reserve 3 UDP ports for each phone**.

For example, your phone has following configuration:

Address: 192.168.1.10

SIP signaling port: 5060

RTP port for channel1: 8006

RTP port for channel2: 8008

You wish to use the following ports on the NAT:

SIP signaling port: 36000

RTP port for channel1: 36010

RTP port for channel2: 36012

Then you have to set the port mapping on the NAT to map 36000 to 192.168.1.10:5060, map 36010 to 192.168.1.10:8006 and map 36012 to 192.168.1.10:8008.

2. Select the NAT type via web or menu as Port Mapping.
3. Set the Extern Router IP to the public address of NAT.
4. Set the Extern Signal Port to 36000.
5. Set the Extern RTP Port 1 to 36010.
6. Set the Extern RTP Port 2 to 36012.
7. Save the setting and reboot.

6.5 UDP Heartbeat

The UDP Heartbeat is very similar to SIP PING. Except, UDP Heartbeat sending different message to SIP server. This mechanism is keeping the SIP signaling port mapping only. Usually the SIP server will support RTP relay to relay the media. The client will send packet periodically through the SIP signaling port to keep the port mapping. No further settings are required to enable this feature except to select the NAT type.

1. Select the NAT Type via web or menu as Port Mapping.
2. Save the setting and reboot.

Chapter 7

7. SIP

7.1 Phone Number

Usually, the phone number will be the user part of the SIP URI. Usually, the SIP phone will be assigned a number so that others can make call to this phone easily.

7.2 SIP server configuration

There are 4 server fields to be configured, SIP Proxy Server, Outbound Proxy Server Registrar Server and Registrar Outbound Server. There are several possibilities will require different settings combination on these 4 server.

1. Most simple case, you just configure the SIP Proxy Server and leave the other 3 empty. If there is no outbound proxy required and no domain required and the proxy does registration. This is the most common case.
2. If the outbound proxy is required, you have to set the SIP Proxy Server and the Outbound Proxy Server accordingly. Keep the Registrar and Registrar Outbound Server empty.
3. If the SIP server requires using domain in your SIP URI, you have to set the domain to the SIP Proxy Server and set the SIP server address to the Outbound Proxy Server.
4. If the SIP server require to register to different server other than the proxy and outbound proxy, then set the SIP server and outbound server as required and set the Registrar server also.
5. If the SIP server require to register to different server other than the proxy and outbound proxy and it also requires using domain in your SIP URI when doing registration, than you have to set the SIP server and outbound server as required, set the domain for registration to the Registrar Server and set the registrar to the Registrar Outbound Server.
6. The server might use different service port. You have to change the Server Port setting accordingly.
7. Some SIP server will require that your local port have to be larger than some specific number for any reason. You have to change the local port accordingly. Be careful, when you change the local port, you have to reboot the phone to make it work.

7.3 Registration

The settings for Registration have been described in the previous section in case 4 and 5. There are several cases that will affect the registration behavior.

1. If the Registrar Server is empty, the phone will use SIP Proxy Server as the host field in

SIP URI for registration.

2. If the Outbound Proxy Server is not empty, then the registration message will be sent to the Outbound Proxy Server.
3. If the Registrar Server is not empty, the host field in SIP URI uses the Registrar Server setting.
4. If the Registrar Outbound Server is not empty, the registration message will be sent to the Registrar Outbound Server.

7.4 Authentication

Both registration and call service will require authentication. You have to set the username and password via web or menu.

7.5 Caller ID

You can decide to send or hide the caller ID when making a call to others. In SIP we use the DisplayName field in the SIP signaling as caller ID. If the caller ID is enabled, what you set in the DisplayName field will be sent to the callee as the caller ID. IF the DisplayName is empty, the phone will send the phone number instead. If caller ID is disabled, the DisplayName field in the SIP message will be empty.

7.6 RTP Port

You can change the RTP port if it is required for any reason. There is one restriction, that is the RTP port has to be even number and the port is for both send and receive.

7.7 Audio CODEC

1. Audio Codec Preference list: There are 4 different audio codec supported in this phone. (G.723.1 and G.729 are optional and might not be available in your phone.) You can arrange them by assign priority for each of them. When you modify the priority of a single audio codec, the others will rearrange accordingly.
2. Packet Time: You can select the preferred packet time according to the network traffic. Sometimes, in a heavy network a larger packet (Larger Packet Time) will get better voice quality because the packet has better chance to be received. But, larger packet time also introduces larger delay time. So, it's a trade off.
3. For G.723.1 Audio codec, there is another settings for choosing high bit rate or low bit rate.

7.8 User Name For URL

Sometimes, the SIP server requires using a non-numeric username for your SIP URI. Since we use phone number as default username, it can't be a non-numeric string. So, you have to set this field, to force the phone using this setting instead of the phone number in your SIP URI.

7.9 Optional SIP Header

If there is any additionally SIP headers that are not exist in the regular SIP message of this phone and is a must for some SIP server, you can use these two fields to force the phone using these headers in all SIP messages. The format for optional header field is:

headername: header value

Chapter 8

8. Auto Provisioning

8.1 Auto Provisioning Mechanism

There are two major reasons to create auto-provision framework. First, this framework aim to provide a way to make the phone automatically get ready without tedious SIP setting. Second, this framework aim to provide a way to let the phone automatically sync with the latest settings and the latest firmware. Only FTP server and IP-Phone are involved in the framework. Also, the administrators who maintain the database on the FTP server are the key person of this framework.

Basically, our Auto Provisioning System (APS) follows a so-called “plugndial” mechanism. This mechanism requires the special folder structure, which reflects the MAC address of the devices. Thus, all devices have its unique folder. When the phone boot up, it will automatically download and examine a special formatted configuration file and update the phone’s setting according to this file. The operation of this phone is slightly different to the original plugndial mechanism. This phone will download the checksum file first. Usually, the checksum file is prepared to verify the downloaded file is valid. In our APS mechanism, we use this checksum file to verify whether there is any change in the configuration file. Our phone also calculates the checksum according to the current settings on the phone. If the two checksums are exactly the same, that means nothing changed since last synchronization, download the whole profile is not necessary in this case. The following is a description of how to prepare the FTP server to perform this function.

1. Install and make sure the FTP server is working well.
2. For each device, create an account for it. (You may use one account for all devices or group of devices)
3. For each device, create its own unique folder according to its MAC address. For example, a device has MAC address 0003C9005050 will require to create a folder [/0003/c900].
4. There should be at least 2 files in this folder, and if necessary, a firmware file also.
 - A. 0003c9005050.txt: This is the configuration file. This is the core of whole framework.
 - B. 0003c9005050.md5: This is the MD5 encoded checksum of the configuration file.
 - C. Ipphone.bin: This is the firmware file. (Not necessary to be here)

The format of the configuration file will be described later. The MD5 checksum widely used in the Internet I will describe how to generate this file by a free tool later.

(Note: The filename before the extension name is exactly the same as MAC address in Hex format and in lower case.)

After preparing the server, you can turn on the “Enable Auto Provisioning” flag of the phone through web. The following is a description of how the phone performs the auto-provisioning function.

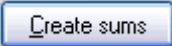
1. Download the **0003c9005050.md5** (/0003/c900/0003c9005050.md5). Calculate the checksum and compare with the downloaded file to see whether they are the same. If they are the same, go to step 2. If they are not the same, go to step 3.
2. If the result is the same, the phone does nothing.
3. If they are not the same that means there should be something changed in the configuration file. Download the **0003c9005050.txt** (/000c/c900/0003c9005050.txt)
4. Parse the configuration file and configure the phone is any settings changed.

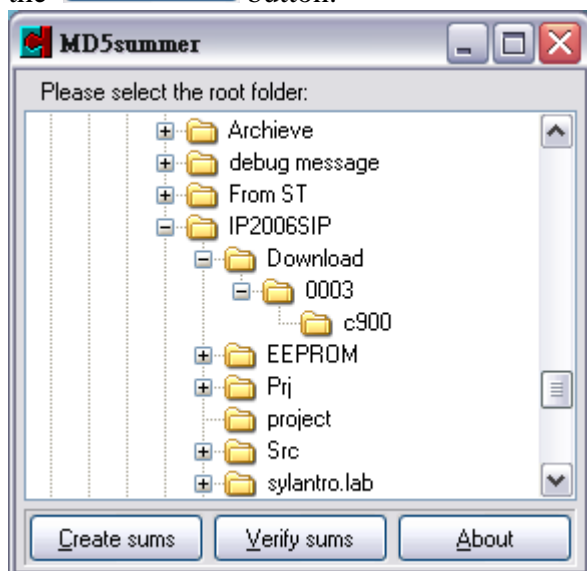
5. Examine the firmware version in the configuration file compare with the current running version. If the firmware version didn't change but some configuration has been changed, the phone will reboot automatically. If the firmware version changed, the phone will download the firmware according to the settings in the configuration file. Then reboot the phone. No matter the firmware or the phone settings changed, the phone will reboot automatically after updating.

The phone will automatically perform this procedure when booting and every 24 hours.

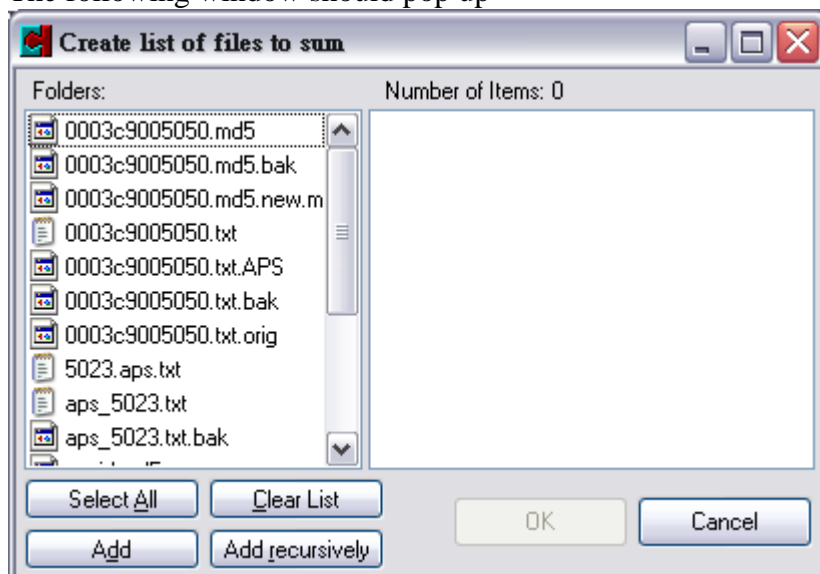
Checksum file:


Here's a description of how to use a free tool (md5summer.exe) to generate the .md5 file in your FTP server:

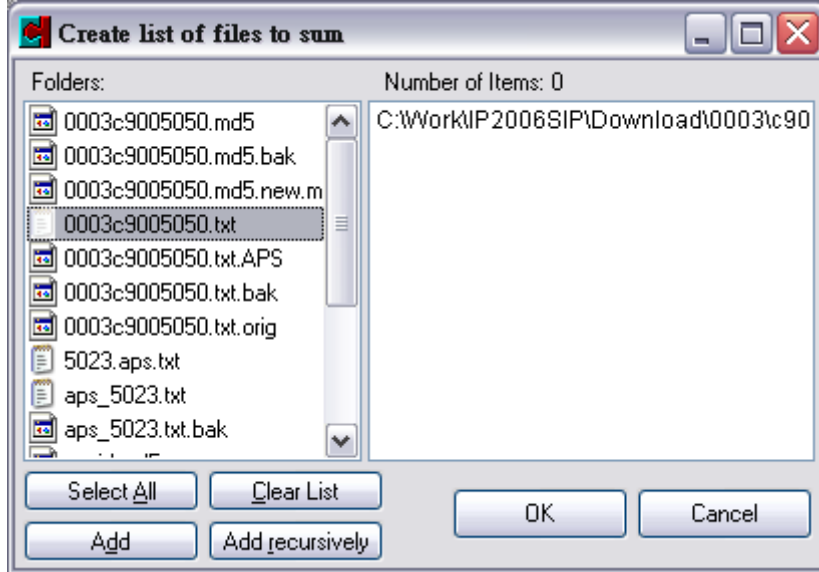
1. Get an md5 tool. You can get a free tool md5summer.exe from the following link:
<http://www.md5summer.org/download.html>
2. Prepare the configuration file and put it in the specific location in the server. The format of the configuration file is described in the next section.
3. Launch the md5summer.exe. The following window will pop up and prompt you to select the folder where the file you are going to calculate is located. Click on the folder and press the  button.




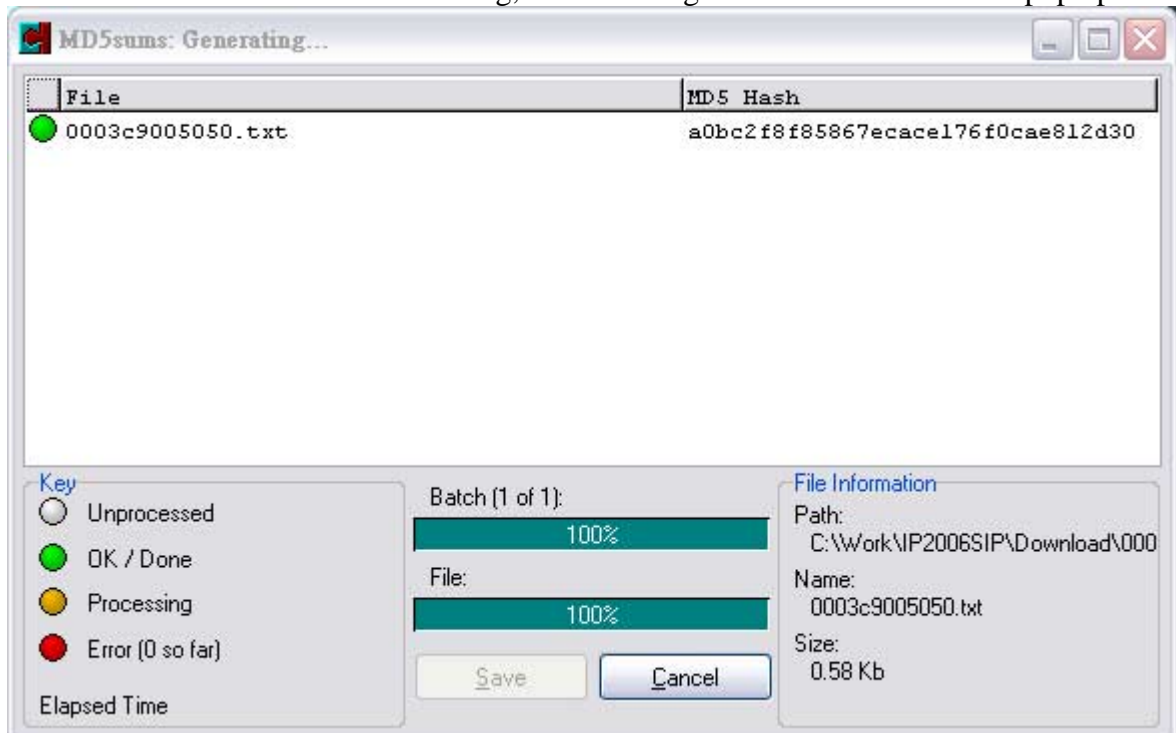
4. The following window should pop up



5. Click on the list in the left to select the file and click  to add this file to list in the right. Or you can just double click on the list in the left to add it.



6. Press  to start calculating, the following calculation window will pop up.



7. It won't take long a save to dialog box will pop up and prompt you to specify the location and file name to save the calculation result. By default, the file name will be the same as the source filename, in this case "0003c9005050" and by default the extension is "md5". I suggest just using the default. Now you have the 0003c9005050.md5 in the specific folder.

The resulting md5 file should have the content look like the following:

```
a0bc2f8f85867ecace176f0cae812d30 *0003c9005050.txt
```

8.2 The Format of Configuration File

The format of the configuration will look like this:

```
[COMMON]
dhcp_enable=0
GetFtpConfigFile=0
GetFtpCodeUpgrade=0
SaveInFlashAfterSetup=0
RebootAfterSetup=0
ftp_server=192.168.5.20
ftp_user=root
ftp_password=1234
ftp_folder=null
config_filename=0003c9000050.txt
firmware_filename=IpPhone.bin
BoardName=Ip2006SIP
fix_ip=192.168.5.21
subnet_mask=255.255.255.0
default_gateway=192.168.5.1
outboundproxy=null
NbPhonesInUse=1
NbCallsPerChannel=2
echo_cancel_enable=1
silence_suppression=0
codec_order=0,1
g723_bitrate=1
g726_bitrate=5
dsp_code_type=1
packet_time=1
min_jitter_time=0
ui_language=0
additional_codec=0
vlan_id=0x000
vlan_priority=0
vlan_enable=0
dial_tone=1,425,425,-8,8000,0,0,0,0,0,0
busy_tone=1,425,425,-3,6000,6000,0,0,0,0,0
ringback_tone=1,425,425,-3,3200,1600,3200,16000,0,0,0,0
ring_pattern=1,645,20,-10,8000,16000,8000,16000,0,0,0,0
lcd_contrast=20
telnet_password=df 8c bc 5b 9e 0e f7 91
max_jitter_time=60
authentic_id=5021
authentic_password=password
tx_gain_handset=8
tx_gain_handfree=8
rx_gain_handset=9
rx_gain_handfree=13
rx_gain_ring=14
side_tone_gain=3
send_callerid=1
```

```
sip_displayname=null
sip_server=192.168.2.17
primary_dns=168.95.1.1
send_dtmf=3
sip_registrar=null
ntp_server=140.112.2.189
time_zone=40
daylight_saving=0
oem_model=SIP Phone
pppoe_ac_name=null
pppoe_service_name=null
ppp_id=null
ppp_password=null
pppoe_reconnect=0
phone_number=5021
ChannelName=null
rtp_port=8006,8008
sip_client_port=5060
clir_enable=1
always_forwardto=null
handfree_mode=1
mute_mode=1
ipqos=0x00
tone_type=9
ring_type=9
do_not_disturb=0
dial_timeout=20000
wait_dial_timeout=20000
busy_timeout=40000
ringback_timeout=60000
ringing_timeout=60000
autoanswer_time=1000
callsent_timeout=3000
calling_timeout=20000
answer_timeout=5000
register_expiration=60
alphabetic_timeout=2500
dtmf_payload_type=97
multi_line=1
subscribe_voicemail=0
redial_directly=0
conference_enable=1
fwd_always=0
fwd_always_uri=null
fwd_onbusy=0
fwd_onbusy_uri=null
fwd_noanswer=0
fwd_noanswer_uri=null
pppoe_enable=0
outbound_registrar=null
```

sip_server_port=5060
admin_id=admin
admin_password=1234
user_id=user
user_password=1111
auto_answer=0
program_button_1=0
program_button_2=0
program_button_3=0
program_button_4=0
program_button_5=0
phone_lock=0
speeddial_0=null
speeddial_1=null
speeddial_2=null
speeddial_3=null
speeddial_4=null
speeddial_5=null
speeddial_6=null
speeddial_7=null
speeddial_8=null
speeddial_9=null
secondary_dns=63.250.206.138
sip_userid=null
nat_type=0
stun_server=0.0.0.0
stun_server_port=3478
custom_sipheader_1=null
custom_sipheader_2=null
sip_ping_interval=6000
oem_company=Company Name
oem_firmware_version=null
oem_config_version=Cfg:1.0.04
agc_enable=1
http_service_port=80
service_number=Service TEL Number
external_router_ip=null
external_sip_port=5060
external_rtp_port1=8006
external_rtp_port2=8008
frame_per_packet=1
prefix_number=null
CallerBlocking_0=null
CallerBlocking_1=null
CallerBlocking_2=null
CallerBlocking_3=null
CallerBlocking_4=null
CallerBlocking_5=null
CallerBlocking_6=null
CallerBlocking_7=null

CallerBlocking_8=null
 CallerBlocking_9=null
 voicemail_server=null
 auto_provisioning=0

Here's a detail description of each field and valid values:

Field Name	Description	Valid Value
[COMMON]	Reserved, do not change.	
dhcp_enable	Enable / Disable DHCP	1: enable 0: disable
GetFtpConfigFile	Reserved, do not change.	
GetFtpCodeUpgrade	Reserved, do not change.	
SaveInFlashAfterSetup	Reserved, do not change.	
RebootAfterSetup=0	Reserved, do not change.	
ftp_server	FTP server address.	IP address.
ftp_user	FTP account user name.	Valid FTP user name.
ftp_password	FTP account password	Valid FTP user password.
ftp_folder	Folder where firmware located. Note: In APS system, the configuration file folder is fixed can't be changed. This folder setting only indicates where the firmware located. Also, this folder will be used to indicate the configuration file location when doing a manual profile download.	Valid FTP folder
config_filename	Configuration file name. Note: In APS system, the configuration file name is fixed can't be changed. However, you still can issue a profile download through web by specifying the file name here.	Valid file name.
firmware_filename	Firmware file name.	Valid file name.
BoardName	Reserved	
fix_ip	The static IP address.	IP address
subnet_mask	The static Subnet Mask.	Valid subnet mask
default_gateway	The static Router (Default Gateway)	IP address
outboundproxy	The SIP outbound proxy	Domain name
NbPhonesInUse	Reserved, do not change.	
NbCallsPerChannel	Reserved, do not change.	
echo_cancel_enable	Enable/Disable echo cancellation	1: enable 0: disable
silence_suppression	Enable/Disable silence suppression	1: enable 0: disable
codec_order	Preferred codec order. Currently, you can specify up to 4 in the list. However, the G.723 and G.729 are optional. You have to apply the license to use these codec and then enable it via feature keys.	0:G.711-ulaw 1:G.711-alaw 2:G.723 3:G.729
g723_bitrate	Bit rate used by G.723.1	0:5.3KBps 1:6.3KBps
g726_bitrate	Reserved, do not change.	
dsp_code_type	Reserved, do not change.	

packet_time	Packet time (currently, only 0,1,2 is valid.)	0:10 1:20 2:30 3:40 4:50 5:60 6:70 7:80
min_jitter_time	Minimum jitter time	0~255
ui_language	Reserved, do not change.	
additional_codec	Reserved, do not change.	
vlan_id	VLAN ID	3 digits hex value
vlan_priority	VLAN Priority	0-7. 0 is the highest.
vlan_enable	Enable / Disable VLAN	1: enable 0: disable
dial_tone	The dial tone configuration. The format of this pattern is Mix,F1,F2,F3,dbm,t1on,t1off,t2on,t2off,t3on,t3off,t4on,t4off. Mix: Mixing way, 0:F1XF2, 1:F1+F2: 3:F1/F2 F1: Frequency 1 F2: Frequency 2 dbm: loudness factor. (-30~+3) t1on: On duration 1, 1 second = 8000. t2off: Off duration 1, 1 second= 8000. The other on,off period is using the same format as t1on and t1off.	
busy_tone	Same format as dial_tone	
ringback_tone	Same format as dial_tone	
ring_pattern	Same format as dial_tone	
lcd_contrast	Reserved, do not change.	
telnet_password	Telnet password in hash format.	
max_jitter_time	Max jitter time.	
authentic_id	Authentication user ID	
authentic_password	Authentication user password	
tx_gain_handset	Handset transfer gain	1~12
tx_gain_handfree	Handfree transfer gain	1~12
rx_gain_handset	Handset receive gain	1~12
rx_gain_handfree	Handfree receive gain	1~14
rx_gain_ring	Ring gain	1~14
side_tone_gain	Side tone gain	1~12
send_callerid	Send caller ID or not	1:Send 0:Hide
sip_displayname	Display name	32 characters string
sip_server	SIP Server	64 characters string
primary_dns	Primary DNS	IP address
send_dtmf	DTMF type	0: None 1: In-band 2: RFC-2833 3: RFC-2833 & inband 4. SIP INFO
sip_registrar	SIP REGISTRAR	64 characters string
ntp_server	NTP server	IP address
time_zone	Time zone	0~51
daylight_saving=0	Enable / Disable daylight saving time	1: enable 0: disable

oem_model	Model name	20 characters string
pppoe_ac_name	Reserved, do not change.	
pppoe_service_name	Reserved, do not change.	
ppp_id	PPPoE user ID	64 characters string
ppp_password	PPPoE user password	64 characters string
pppoe_reconnect	Reconnect if disconnect by ISP	1: enable 0: disable
phone_number	Phone number assign to this phone.	20 characters string
ChannelName	Reserved, do not change.	
rtp_port	RTP Port for channel 0 and channel 1.	Two valid port number separated by a ','
sip_client_port	SIP client port	Valid port number
clir_enable	Reserved, do not change.	
always_forwardto	Reserved, do not change.	
handfree_mode=1	Reserved, do not change.	
mute_mode=1	Reserved, do not change.	
ipqos=0x00	QOS id.	0~255
tone_type	Build in tone type.	0~9
ring_type	Build in ring type.	0~9
do_not_disturb	Enable/Disable do not disturb	1: enable 0: disable
dial_timeout	Dialing timeout	16bit word in ms
wait_dial_timeout	Dial tone timeout	16bit word in ms
busy_timeout	Busy timeout	16bit word in ms
ringback_timeout	Ringback wait remote answer timeout	16bit word in ms
ringing_timeout	Ringling timeout	16bit word in ms
autoanswer_time	Auto Answer timeout	16bit word in ms
callsent_timeout	Call Sent timeout	16bit word in ms
calling_timeout	Calling timeout	16bit word in ms
answer_timeout	Answer timeout	16bit word in ms
register_expiration=60	Registration expiration time	16bit word in seconds
alphanumeric_timeout	Alphanumeric input timeout	16bit word in ms
dtmf_payload_type	DTMF payload type for RFC2833	96,97
multi_line	Multi-Line support. (Use 2-lines)	1: enable 0: disable
subscribe_voicemail	Subscribe the Message Summary (Standard MWI implementation)	1: enable 0: disable
redial_directly	Redial immediately dial out.	1: enable 0: disable
conference_enable	Enable conference function.	1: enable 0: disable
fwd_always	Enable forward unconditional	1: enable 0: disable
fwd_always_uri	Unconditional forward number	64 characters string
fwd_onbusy	Enable forward on busy	1: enable 0: disable
fwd_onbusy_uri	Forward on busy number	64 characters string
fwd_noanswer	Enable forward on no answer	1: enable 0: disable
fwd_noanswer_uri	Forward on no answer number	64 characters string
pppoe_enable	Use PPPoE	1: enable 0: disable
outbound_registrar	Outbound server for REGISTRAR. Similar to the outbound proxy for SIP server.	64 characters string
sip_server_port	SIP Server port	Valid port number

admin_id	Administrator user name	20 characters string
admin_password	Administrator password	20 characters string
user_id	User level user name	20 characters string
user_password	User level password	20 characters string
auto_answer	Enable / Disable Auto Answer	1: enable 0: disable
program_button_1	Function of programmable button 1	0:Speed dial 1:Call Record 2:Phone book 3:Call Forward 4:Conference 5:Lock IP-Phone 6:Use DND 7:Hot Speed dial
program_button_2	The same as programmable button 1	The same as above
program_button_3	The same as programmable button 1	The same as above
program_button_4	The same as programmable button 1	The same as above
program_button_5	The same as programmable button 1	The same as above
phone_lock	Lock the phone.	1: enable 0: disable
speeddial_0	Speed dial map to keypad 0	64 characters string
speeddial_1	Speed dial map to keypad 1	64 characters string
speeddial_2	Speed dial map to keypad 2	64 characters string
speeddial_3	Speed dial map to keypad 3	64 characters string
speeddial_4	Speed dial map to keypad 4	64 characters string
speeddial_5	Speed dial map to keypad 5	64 characters string
speeddial_6	Speed dial map to keypad 6	64 characters string
speeddial_7	Speed dial map to keypad 7	64 characters string
speeddial_8	Speed dial map to keypad 8	64 characters string
speeddial_9	Speed dial map to keypad 9	64 characters string
secondary_dns	Secondary DNS server	IP address
sip_userid	SIP user ID use in contact and from header.	64 characters string
nat_type	NAT traversal mechanism	0: None 1: STUN 2: SIP PING 3: Port Mapping 4: UDP Heartbeat
stun_server	STUN server address	IP address
stun_server_port	STUN server port	Valid port number
custom_sipheader_1	Customer specific SIP header 1	48 characters string
custom_sipheader_2	Customer specific SIP header 2	48 characters string
sip_ping_interval	The interval for sending SIP PING message.	16 bit word
oem_company	Company name show on the phone.	48 characters string
oem_firmware_version	Firmware version shows on the phone.	32 characters string.
oem_config_version	Profile version shows on the phone.	32 characters string.
agc_enable	Reserved, do not change.	
http_service_port	Service port for HTTP (WEB) configuration.	Valid port number
service_number	Service telephone number shows on the phone.	20 characters string
external_router_ip	External router IP address for port mapping.	IP address
external_sip_port	External port for SIP signaling for port mapping	Valid port number

external_rtp_port1	External port for RTP channel 1 for port mapping.	Valid port number
external_rtp_port2	External port for RTP channel 2 for port mapping.	Valid port number
frame_per_packet	Reserved, do not change.	
prefix_number=null	Reserved, do not change.	
CallerBlocking_0	Prevent the call from this ID.	64 characters string
CallerBlocking_1	Prevent the call from this ID.	64 characters string
CallerBlocking_2	Prevent the call from this ID.	64 characters string
CallerBlocking_3	Prevent the call from this ID.	64 characters string
CallerBlocking_4	Prevent the call from this ID.	64 characters string
CallerBlocking_5	Prevent the call from this ID.	64 characters string
CallerBlocking_6	Prevent the call from this ID.	64 characters string
CallerBlocking_7	Prevent the call from this ID.	64 characters string
CallerBlocking_8	Prevent the call from this ID.	64 characters string
CallerBlocking_9	Prevent the call from this ID.	64 characters string
voicemail_server	Voice mail server.	64 characters string
auto_provisioning	Enable / Disable auto-provisioning.	1: enable 0: disable

The End of Document =====

Part No. : 671-000594 (AG) D-Code : R2-30714
