US102 IP Phone

User Manual

Escene Communication Technology Co.Ltd

Tables of Contents

1. Getting Started

About

US102 is a popular type IP Phone in Sayhi phones series, with modern design, functional, practical and voice clarity characteristics. It accomplished the powerful telephony features by cooperating with the communications platform, such as call transfer, hotline, third-party conferences, voice mail, interruption-free, etc.

Feature Highlights:

- HD Voice: HD Codec
- Support unified maintenance and auto upgrade
- Enterprise Phone Book
- Support Headset interface
- Support PoE and AC power adapter
- Support HTTP/TFTP/FTP Auto-provision/TR069 for upgrade software

Technical Features

Item	Technical Features
Screen	Grayscale LCD with background light
	128*64 characters
Language	English, Chinese
Line	2
Function Keys	5 Navigation keys (Arrow button, OK button) Volume button(multiplex up and down keys) Hands-free Mute Headset Message Menu Hold Redial Conference Transfer
VoIP Protocol	SIP 2.0

Network Protocol	HTTP, BOOTP, FTP, TFTP, IEEE 802.1Q, *IEEE 802.1X
Codec	G.723、G.729 A 、G.711 A/U G.722
QoS	TOS, Jiffer Buffer, VAD, CNG, G.168 (32ms)
Network	2*RJ45 10/100M Ethernet interfaces(LAN/PC) IP Assignment: Static IP or DHCP VPN(L2TP) ,VLAN/QoS DNS Clients (Primary and Secondary)
Speech	Handset, Headset or Hand-free Mode, Call center headset and 3.5mm headset supported 9-levels volume adjustment
Call Processing	Call Waiting, Call Queuing Call Forward, Call Transfer, Call Holding, Call Pickup, Callback Redial,Auto-answer Phone directory speed dial, call record direct dial 3-way conference DnD Voice mail, Voice Prompt, Voice Message
Application	Enterprise phone directory XML Phonebook Private phone directory
Security	Password Login Web Signaling encryption Media encryption
Management	Upgrade: HTTP/TFTP/PnP auto-provision Configurations: Phone/Web/auto-provision Debug: Telnet/Phone/Web
Power Supply	Power adapter:AC100~240V input and DC 5V/1A output PoE(IEEE 802.af)
Specification	Storage Temperature: 0℃ ~ 60℃ Operating Humidity: 10%~90% Size: 335mm*219mm*68mm Net weight: 1.07kg

2. Connecting Your Phone

Your system administrator will likely connect your new SayHi US102IP Phone to the corporate IP telephony network. If that is not the case, refer to the graphic and table below to connect your phone.

1) Open the box of US102 IP Phone，carefully check the packing list as follow:

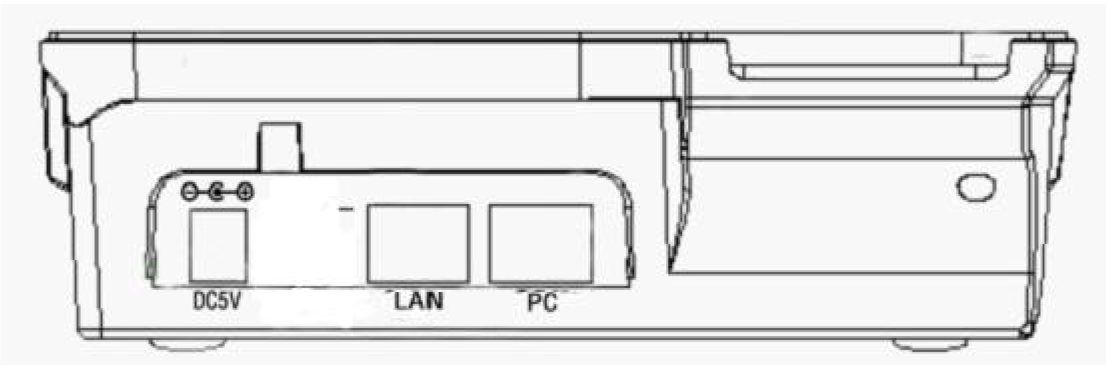
Item	Counts
IP Phone	1
Handset	1
Handset Cord	1
Power adapter	1 (Phone with PoE without Power adapter)
RJ45 cable	1
RJ11 cable	1
CD	1
Quick Installation	1
Quick User Guide	1
Product certification	1

2) As shown in figure 2.1，please plug Handset Cord into RJ11 interfaces (IP Phone and Handset)，RJ45 cable into the LAN interface；IP Phone will automatically start if IP Phone with POE function.

3) The phone must work together with power adapter without POE support.

4) If you want connect your computer into LAN at the same time, please connect your computer to PC interface of the phone with a RJ45 cable.

Figure2.1 Interfaces of SayHi US102



3. Phone overview

Understanding Buttons and Hardware

You can identify buttons and hardware on your SayHi US102 from figure 3.1.

Figure 3.1 SayHi US102

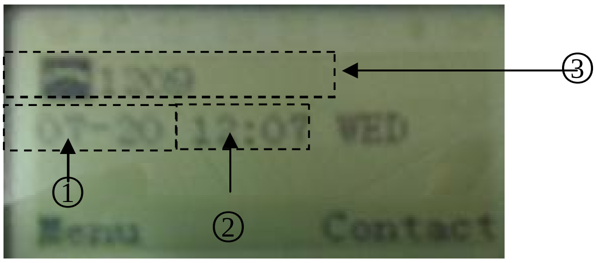
	Item	Description
1	LCD Screem	128*64 characters, grayscale LCD with background light

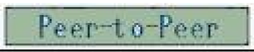
2		Menu button: which buttom make you enter the menu setting interface
3		Line button: US102 have two account ,one account have a corresponding line button .If the call coming or the line is used, the light will become red.
4		Received button: you can search the phone number which you have receive by press this button
5		Vol+ button: you can adjust the volume
6		Missed button: you can search the phone number which you have missed by press this button.
7		Dialed button: you can search the phone number whhich you hace dailed by press this button.
8		OK button: To confirm the action.
9	0-9，*，#	Basic Call Handling: press “#” send out a call by default.
10		Speaker button: Toggles the speakerphone on or off.
11		There button： Conference button: Connect calling / called party Transfer button： Transfer redirects a connected call. Redial button: To dial the last number.
12		light : It will flash if a call come in It will become red if you want to dail a phone number.
13		Blf button： You can set four type on blf . there are speed dial, Asterisk BLF,Speed Dial Prefix, DTMF.

Understanding Phone Screen Features

This is what your main phone screen might look like:

Figure 3.3 SayHi US102 Phone LCD



	Screen displays	Functions
1	Date	Show current date（You can set with different sources，the more *7. Web Setting*)
2	Time	Show current time（You can set with different sources，the more *7. Web Setting*)
3	Line status	Show the phone line status: 1): Disconnect into network. 2): Only Peer-to-Peer call. 3): Network connected normal, but the line is not successfully registered. 4): Network is OK and the line is available. 5): Line is turned on DND.

4. Basic Call Handling

You can perform basic call-handling tasks using a range of features and services. Feature availability can vary; see your system administrator for more information.

Note: The bold type of the following text in table signifies the phone's button.

Placing a Call

Here are some easy ways to place a call on SayHi US102 IP Phone:

<table>
<tr><th>If you want to...</th><th colspan="2">Then…</th></tr>
<tr><td>Place a call using the handset</td><td>Pick up the handset</td><td rowspan="3">--1) You can hear dial tone;
--2) Enter a number；
--3) Press # button (default)，
-or wait 5s (default), then it send the number automatically.</td></tr>
<tr><td>Place a call using a speakerphone</td><td>Press Speaker button</td></tr>
<tr><td>Place a call using a headset</td><td>Put on your headset, active Headset button so that the status light is Red, and then do as using speakerphone</td></tr>
<tr><td>Redial</td><td colspan="2">-- press Navigation button-Right (in Standby interface) > “Dialed ”, select a number, and press Select .</td></tr>
<tr><td>Dial from a call log</td><td colspan="2">--1) Press MENU or OK button > “Calls ”, you can select “Missed calls”, “Received calls” and “Dialed numbers”,
- or press Navigation button (in Standby interface) > select “Missed ” (down), “Received ” (left) and “numbers” (right))；
--2) Then press Select .</td></tr>
</table>

Tips

• You can dial on-hook, without a dial tone (pre-dial). To pre-dial, enter a number, and then go off-hook by lifting the handset or pressing **Headset** or **Speaker** button.

Answering a Call

You can answer a call by simply lifting the handset, or you can use other options if they are available on SayHi US102.

<table>
<tr><th>If you want to...</th><th colspan="2">Then...</th></tr>
<tr><td>Answer with a handset</td><td rowspan="3">--1) Your phone ring;
--2) Light strip is Red and flashing;</td><td>--Pick up the handset</td></tr>
<tr><td>Answer with the speakerphone (Non-headset mode)</td><td>--Press Speaker button</td></tr>
<tr><td>Answer with the a headset</td><td>--Put on headset, press Headset button , and then do as using speakerphone</td></tr>
<tr><td>Auto-answer</td><td colspan="2">--1) Press MENU or OK button > “Functions ” > “Auto answer”;
--2) Select “Enable”;
--3) Your phone answers incoming calls automatically after a few rings.</td></tr>
</table>

Tips

• Your system administrator configures Auto-answer to use either the speakerphone or a headset. You might use Auto-answer if you receive a high volume of incoming calls.

Ending a Call

To end a call, hang up. Here are some more details.

If you want to...	Then...
Hang up while using the Handset	-- Return the handset to its cradle
Hang up while using the speakerphone	-- Press **Speaker** button
Hang up while using the	--Press **Handset** button, （Do not keep the headset mode)

Headset	

Using Hold and Resume (Switch Calling Line)

You can hold and resume calls.

If you want to...	Then...
Put a call on hold	Press **HOLD** button
Resume a call	Press **HOLD** button

Tips

• Engaging the Hold feature typically generates music or a beeping tone.

Transfer Calls

Transfer redirects a connected call. The target is the number to which you want to transfer the call.

If you want to...	Then...
Talk to the transfer recipient before transferring a call (consult transfer)	--1) Press **TRANSFER** button; --2) Enter number; --3) press “**#**”（default) , -or wait five seconds(default)then transfer the call

Do Not Disturb

You can use the Do Not Disturb(DND) feature to block incoming calls on your phone with a busy tone（Can also be set to their voice mail or other extension numbers, etc.).

If you want to...	Then...
Enable DND on a line	--1) Press **MENU** or **OK** button > “Functions ” > “DND” > (select line) “Enable”

	--2) All enabled line on the phone would changes to  status.
Disable DND	Press **MENU** or **OK** button > “Functions” > “DND” >(select line) “Disable”

3-way Conference

You can establish a three-party conference, during the conversation three phone parties can communicate with each other.

If you want to...	**Then...**
Invite the transfer recipient into a conference in a transferring	--1) When the transfer recipient answer the call, press **CONFERCENCE** button on your phone; --2) Then the held one, transfer recipient and you will be into a conference.
Invite the third party into a conference in a active call	--1) Press **CONFERENCE** button in an active call; --2) Enter the third party number; --3) After connected the third party, press **CONFERENCE** button again

5. Advanced Call Handling

Using the Phone Book

You can store a large number of contacts in your phone's directory. You can add, edit, delete, dial, or search for a contact in this directory. However, it only can configure the phone book on web page in SayHi US102. For details, you can refer to *7.Web Settings*.

However, you can dial from Phone Book on the phone after setting phone book on web page.

Call from phone book	--1) Press **MENU** button > "Contact", --2) Select "Personal phone book">"View All", -or select a contact button beside the menu button.

Using Call Logs

Your phone maintains records of your missed, placed, and received calls.

If you want to…	Then…
View your call logs	--1) Press **MENU** button > "Calls > "Missed Calls", "Received Calls", or "Dialed numbers" --2) Use the navigation keys to view the call record information.
Dial from a call log	Please refer to the previous part *4.Basic call handing – Placing a call.*

Tips

• Each call log store up to 20 entries on SayHi US102 IP phone.

6. Keypad Instruction

SayHi series IP phones are can be configured in two ways. The first you can use the phone keypad where you can settings for you IP phones, the other you can log in to User Options web pages where you can settings for you IP phones.

Use phone keypad to setting. Press **MENU** or **OK** button to the main menu, Use the navigation keys to select menu, press **OK** button to confirm menu selections, press back button or cancel button to delete input information.

SIP Account Settings

SayHi US102 series IP phone make calls based on sip accounts, sayHi US102 series IP phones can support 2 independent SIP account, Each account can be configured to different SIP server.

If you want to…	**Then...**
Create an SIP account	--1) Select “Settings” > “Advanced settings”; --2) Enter the password required（The default is empty）； --3) Select “SIP” > “Account sip”; --4) Select one of the account you want to setting, you can configure the following parameters **-Enable account***: Select Enable **-Account Mode**: the type of account **-Display Name**: The name displayed on the screen **-User Name***: the account matched with the SIP server.（extension number) , **-Authen usr**: the Authenticated users matched with the SIP server.（The default With the same account) **-user pwd***: the user password matched with the SIP server **-Description**: description of this account, **-SIP1***: the primary SIP server, By default all calls through the server, **-SIP2**: the secondary SIP , When the primary server is

	unavailable ,use the SIP server **-Refresh time**: Registration refresh interval, the minimum value is 20 **The default value is 3600.** **-Con type: which protocol the phone used to send the voip packets** **-Amount of used lines: Maximum line are allowed to used** --5) Set up the above parameters, select "Submit changes" to saves settings, Complete the account creation.
Disable sip account	--1) Select "Settings" > "Advanced setting"; --2) Enter the password required（The default is empty）； --3) Select "SIP" > "Account sip"; --4) Select "Enable account" > "Disable"; --5) Select "Submit changes" to saves settings

Network Setting

If you want to …	**Then…**
network setting	--1) Select "Settings" > "Advanced settings"; --2) Enter the password required（The default is empty）； --3) Select "Network", you can configure the following parameters: **-Type**: static IP or DHCP - **DNS1**: enter IP address of the primary DNS server - **DNS2**: enter IP address of the secondary DNS server **-Web port**: the default Web port is 80,if you change it(for example change it to 88),you must use IP and Web port to login the web page (for example http://192.168.0.200:88).It will take effect on next reboot. **-Telnet port**: the default Telnet port is 23, if you change it (for example change it to 2003), you must use IP and Telnet port to login the manage page (for example telnet 192.168.0.200:2003).It will take effect on next reboot.

Customizing Rings and Volume

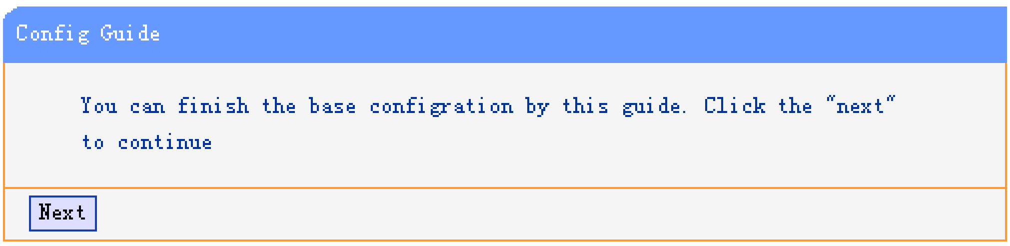
If you want to…	**Then…**
Change the ring tone	--1) Select "Settings" > "Phone settings" > "Ring type"; --2) Press navigation to Select ring tone
Adjust the volume level	--1) Select "Settings" > "Phone settings" > "Volume settings" --2) You can adjust the volume level of following types **-Ring volume**: Phone call ring volume, **-Handset volume**: Handle output volume, **-Handset mic volume**: Handle input volume, **-Speaker volume**: Hands-free speaker output volume, **-Speaker mic volume**: Hands-free input volume, **-Headset volume**: Headphone output volume, **-Headset mic volume**: Headset microphone input volume

7. Web Settings

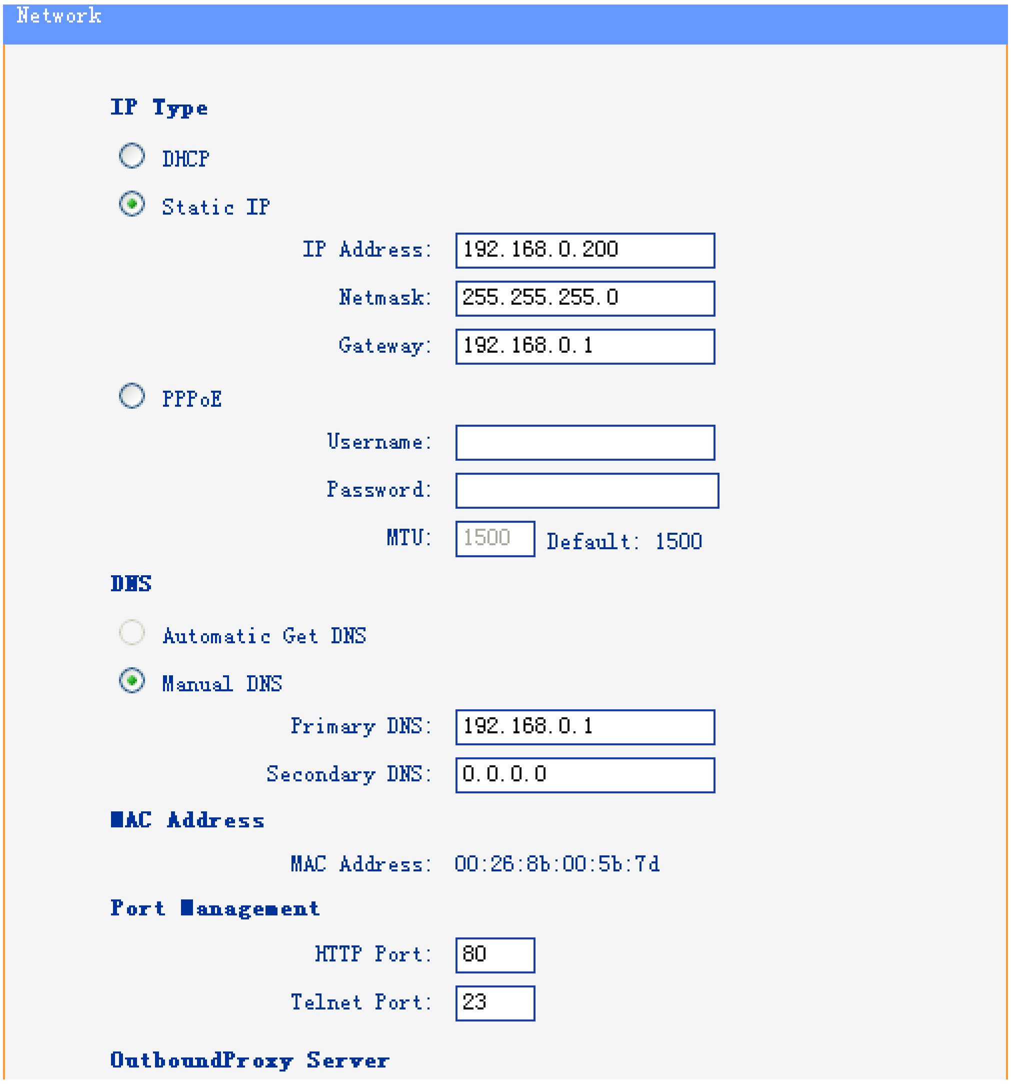
We can configure IP Phone more handy through web setting. Press OK button on the keypad of the phone to enter the status page and find out the IP address of IP phone. Enter it (for example http://192.168.0.200) into the address bar of web browser. The default login name and password are both "root".

Config Guide

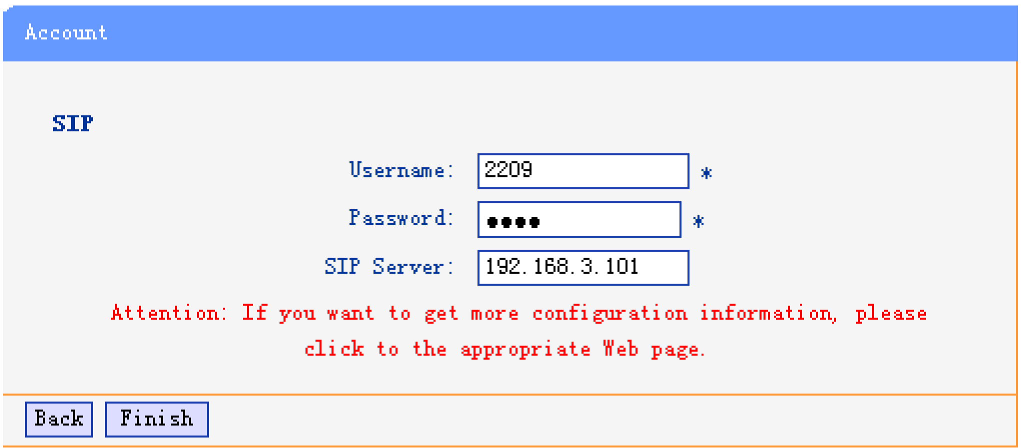
You can finish the base configration step-by-step by this guide.



When press 'next',you can configure the Network parameters for the phone,



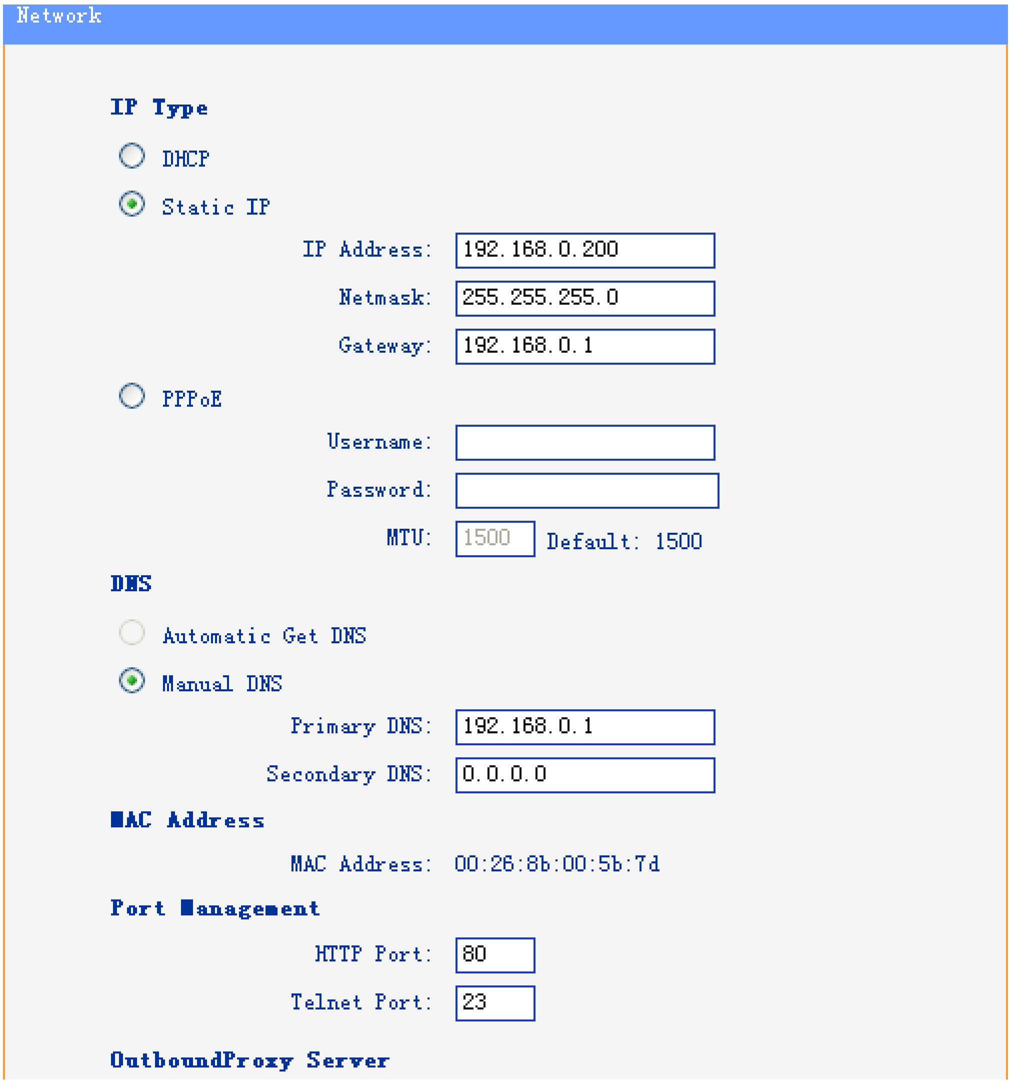
After config the network parameter,press next,then you can config sip account for the phone.



Press Finish,the base configuration of the phone is complete,now you can use the phone to call with sip.

Network

You can config the network parameters for the phone on the web page.



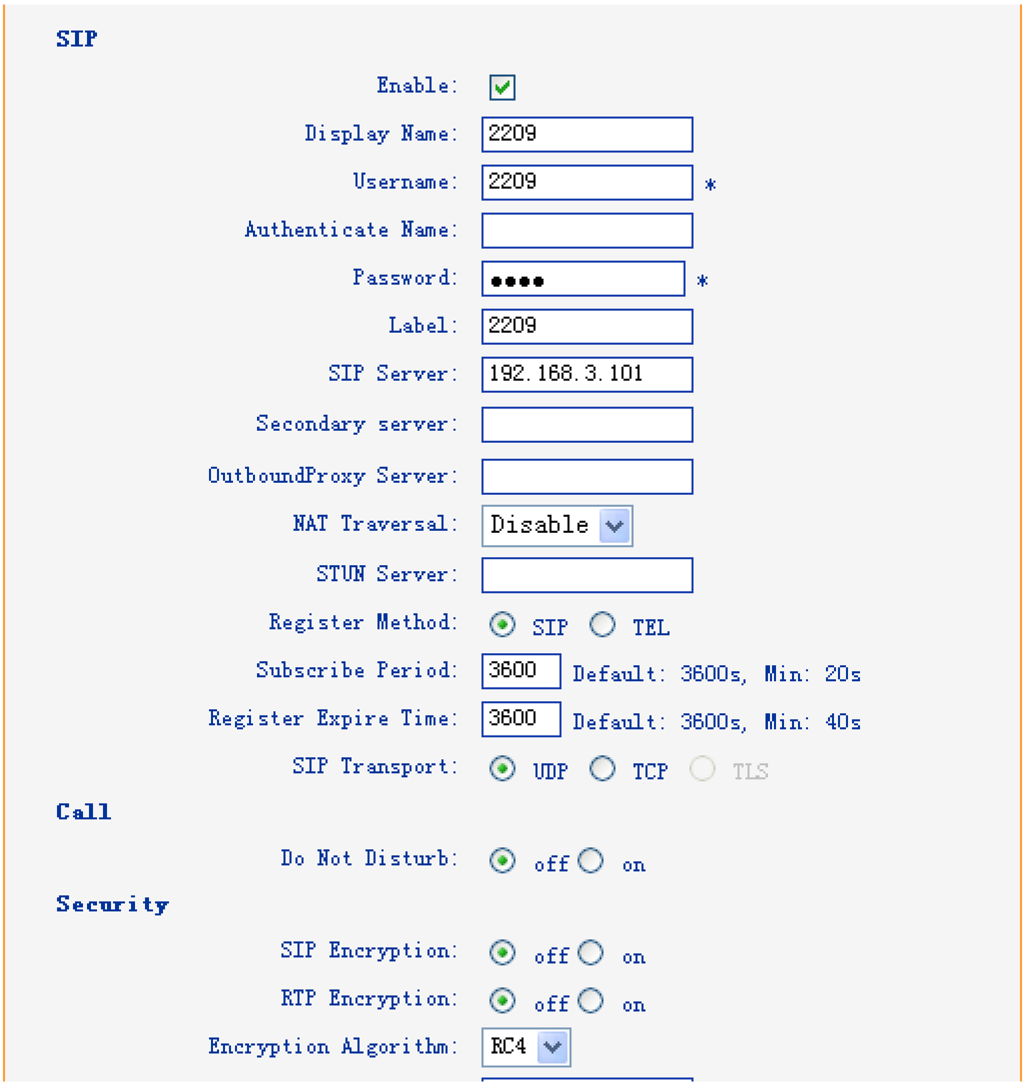
Choose network,you will find the following parameters:

Field	Description
DHCP	Config the phone get ip info from DHCP server
IP Address	Config the ip manual for phone
Netmask	Config the netmask manual for phone
Gateway	Config the gateway manual for phone
Username (pppoe)	The pppoe username
Password (pppoe)	The pppoe password
MTU (pppoe)	The mtu for pppoe,default is 1500
Primary DNS	The primary DNS server
Secondary	The secondary DNS server
MAC Address	Display the MAC of the phone

HTTP Port	The default web port is 80,if you change it(for example change it to88), You must use IP and Web port to login the web page(for example http://192.168.0.200:88). It will take effect on next reboot.
Telnet Port	the default Telnet port is 23,if you change it(for example change it to 2003),you must use IP and Telnet port to login the manage page (for example telnet 192.168.0.200:2003).It will take effect on next reboot.

SIP Account

The phone attempts to register to the SIP server using the account/registrar data provided by the automatic or manual initialization.



Choose one Account, you will find the following parameters:

Field	Description
Enable	You can choose on/off to enable/disable the line.

Account Mode	You can choose VOIP
Display Name	It is showed as Caller ID when making a phone call
Username	It is a username provide by SIP Server
Authenticate Name	It is authenticated ID for authentication
Password	It is a password provide by SIP Server
SIP Server	Server for registration, provided by administrator
Register Expire Time	IP phone automatically registered every time
Amount Of Line Account Used	The line key of account used,default is 1

Programmable Keys

Choose Programmable Keys,you will find the following parameters:

Field	Description
Speed Dial(Mode)	Use specific Key as Speed Dial key
Asterisk BLF(Mode)	Use specific Key as BLF key
Speed Dial Prefix(Mode)	Use specific Key as Speed Dial Prefix key
DTMF	Use specific Key as DTMF key

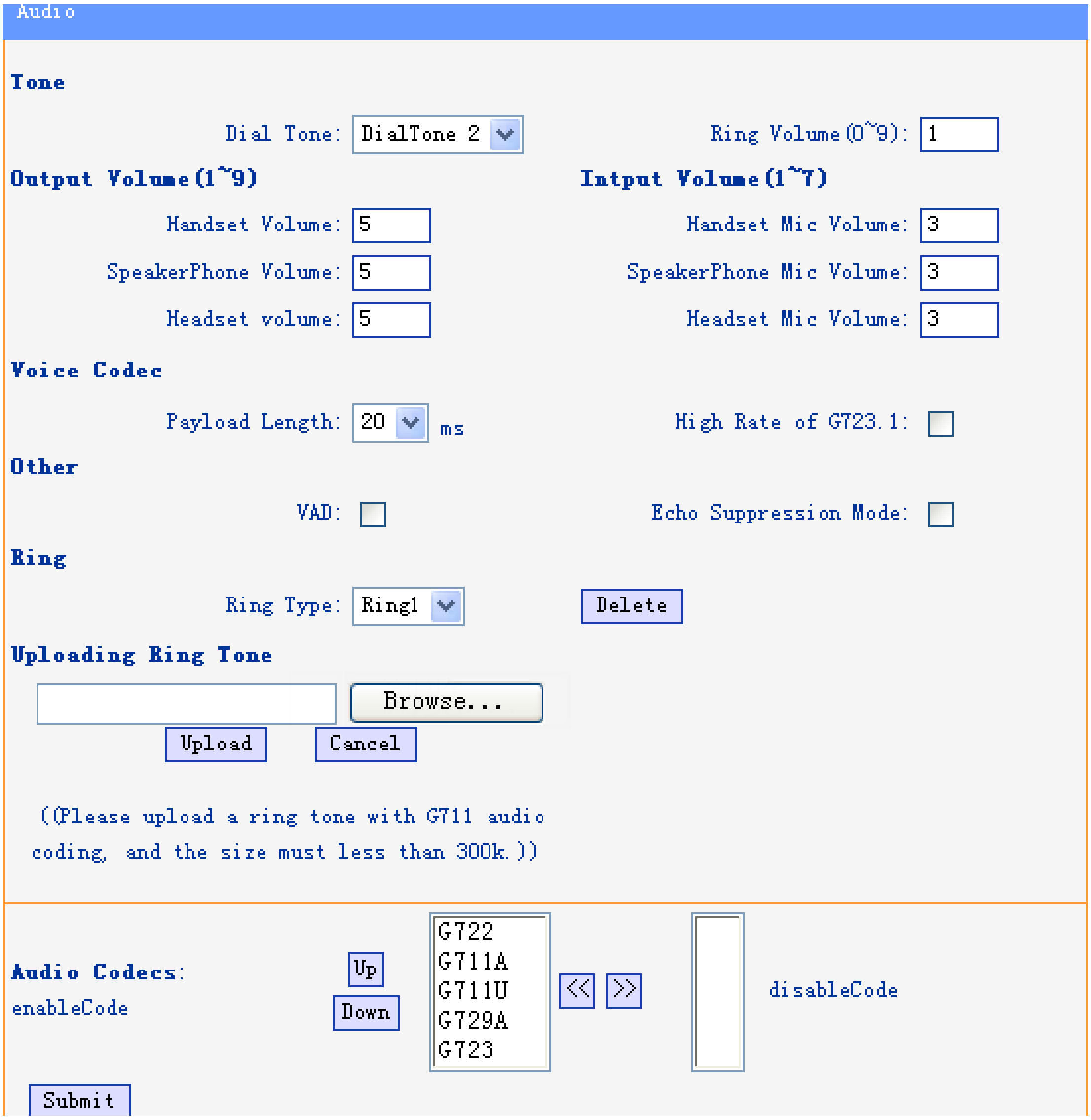
Audio

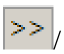
The IP phone supports the following voice codecs: G.722, G.711A, G.711U, G.723, and G.729A.

You can enable/disable the desired codecs via Web interface. Please contact your system administrator for more details about the codecs.

To enable/disable the codecs:

1) Choose Audio-> Audio Codecs



2) Use the navigation keys to highlight the desired one in the Enabled/Disable Codecs list, and press the 

/ 

to move to the other list.

3) Choose Submit to save the change.

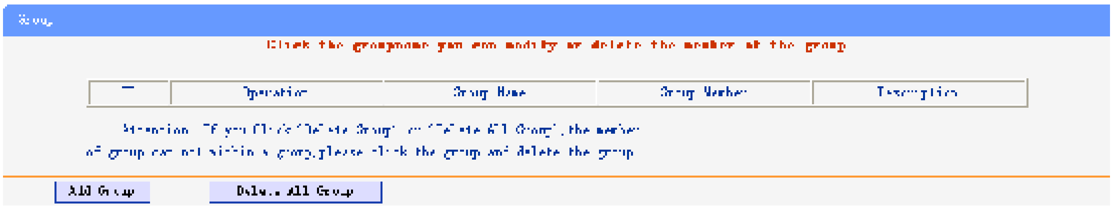
Of course, you can control the voice bulk in this choose.

PhoneBook

Group

You can add, edit and delete group in a phone book on web page of US102.

1) Click “PhoneBook” > “Group”,



If you want to add a Group, you just ought to click ‘Add Group’ .

You can edit an existed Group by click .

You can delete an existed Group by click , if you want to delete all Groups, you just ought to click ’Delete All Group’.

2) When you add a group or edit an existed group, you can set several parameters as follow:

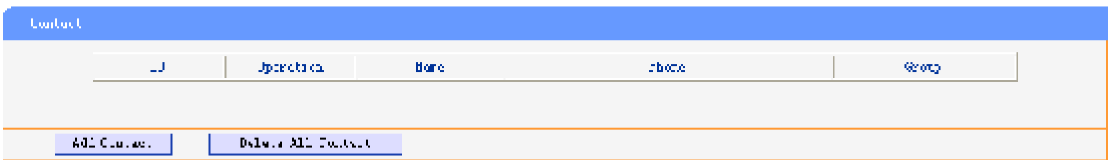
Group	
ID	Serial number of a group
Description	Description of a group
Group Name	Name of a group

Contact

You can add, edit and delete contact in a phone book on web page of US102.

The phonebook can storage 300 contact entry.

1) Click “PhoneBook” > “Contact”,



If you want to add a Group, you just ought to click ‘Add Contact’.

You can edit an existed Contact by click .

You can delete an existed Contact by click , if you want to delete all Contacts, you just ought to click 'Delete All Contact'.

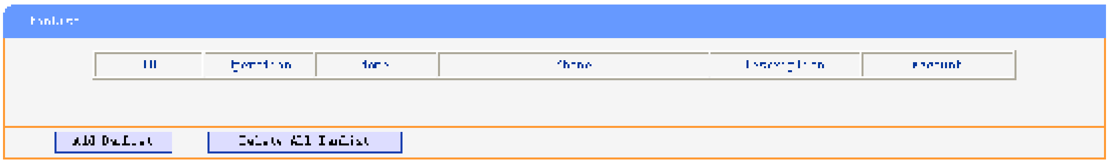
2) When you add a Contact or edit an existed Contact, you can set several parameters as follow:

Contact	
Serial Number	Serial number of a contact
First Name	The First Name of a contact
Last Name	The Last Name of a contact
Mobile Number	The Number1 phone number of a contact
Office Number	The Number2 phone number of a contact
OtherNumber	The Number3 phone number of a contact
Group	You can assign a contact to a specific group. If there isn't any group set on the phone, the group is None by default.
Account	Select a SIP account relating this contact, that is you can dial to the contact from this SIP account.

BanList

You can add, edit and delete banlist in a phone book on web page of US102..

1) Click "PhoneBook" > "BanList",



If you want to add a BanList, you just ought to click 'Add BanList'.

You can edit an existed BanList by click .

You can delete an existed BanList by click , if you want to delete all BanLists, you just ought to click 'Delete All BanList'.

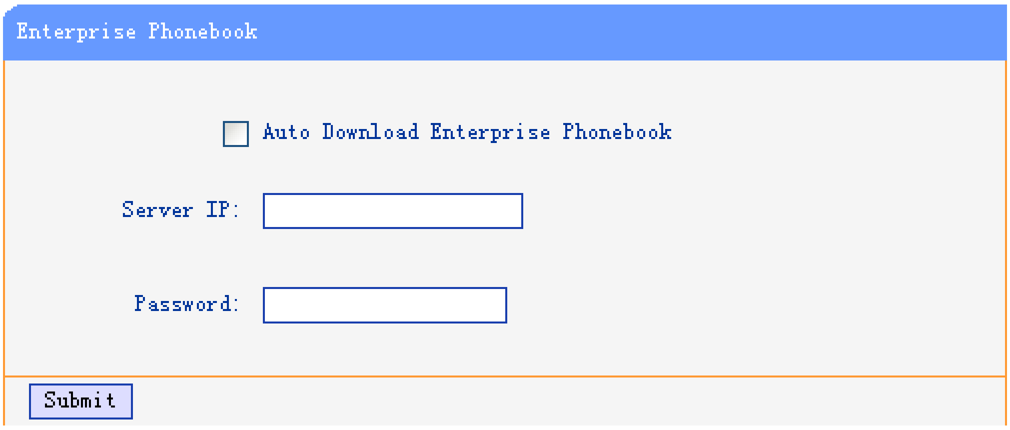
2) When you add a BanList or edit an existed BanList, you can set several parameters as follow:

BanList	
Serial Number	Serial number of a BanList
Description	Description of a BanList
First Name	The First Name of a ban contact
Last Name	The Last Name of a ban contact
Mobile Number	The number1 phone number of a ban contact
Home Number	The number2 phone number of a ban contact
Office Number	The number3 phone number of a ban contact
Account	Select a SIP account relating this ban contact, that is the ban contact can't dial to this SIP account.

Enterprise Phonebook

You can download Enterprise Phonebook from this web interface. But you should do second develop on the sip server to enable this function completely.

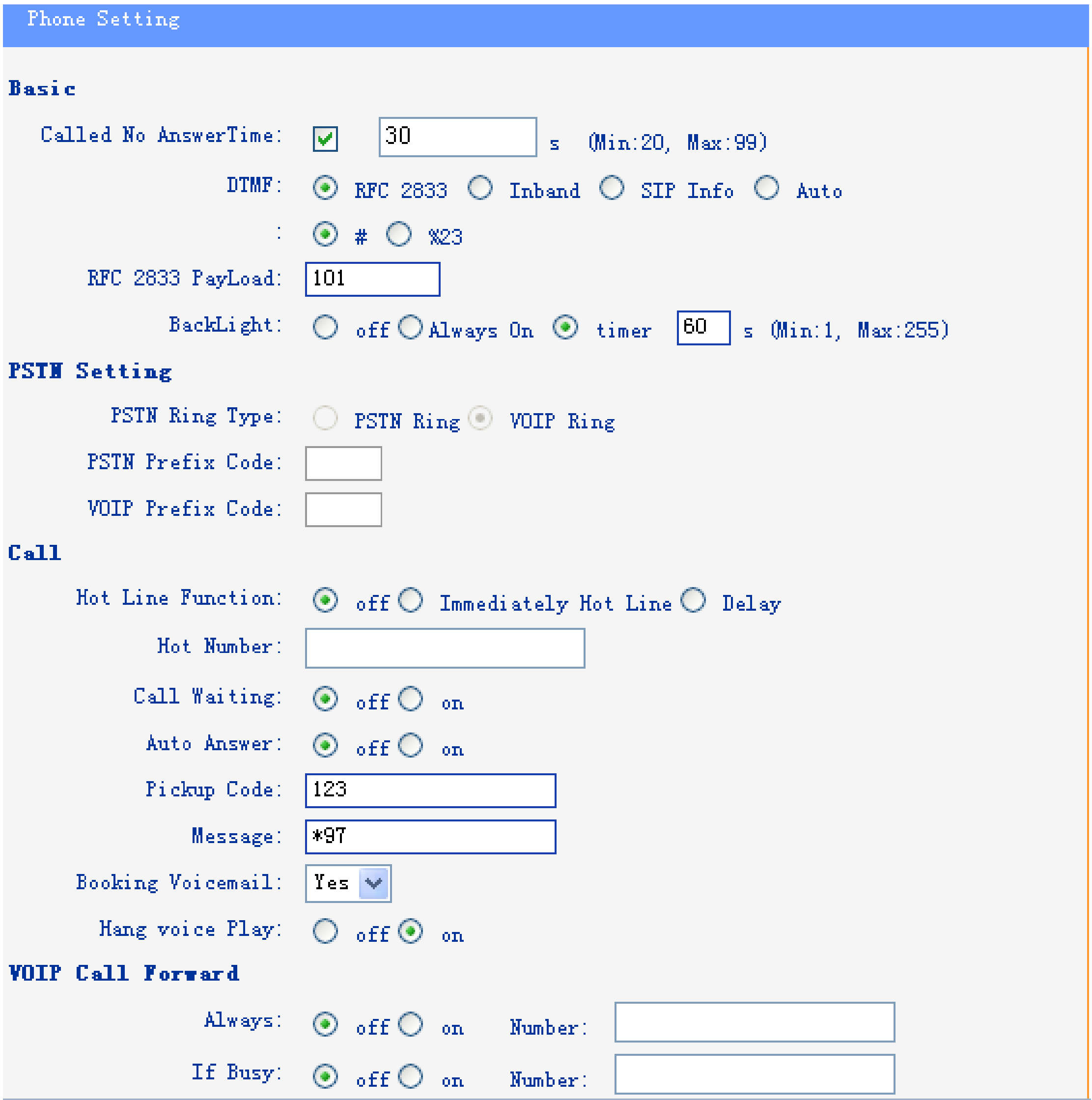
If the sip server no add some function to hold this option ,this option can be userd.



Advance

Phone Setting

You can use phone setting to set the time,qos,port Mirroring for the phone.

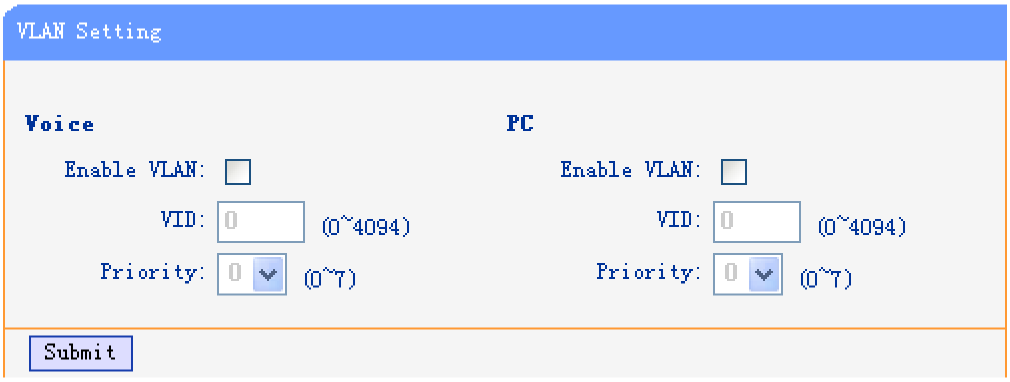


When used Phone Setting option,you can set several parameters as follow:

Phone Setting	
DTMF	The DTMF transmitted mode,include RFC 2833,Inband,SIP Info
BackLight	The backlight of the phone LCD
Set Time Mode	The mode of set time for phone,include SNTP/SIP Server/PSTN/Manual
Daylight Saving Time	Enable/disable the DST for the phone
Time Format	You can use 24 hour time format or 12 hour time format
Time Zone-GMT	You can select different time zone for the phone
Manual Setting	This used to manual set time for the phone
QoS	The qos priority,support diff-serv and precedence
Network Packet Mirroring	When select on,then you can capture the phone's packet use notebook which connect to pc port of the phone

VLAN Setting

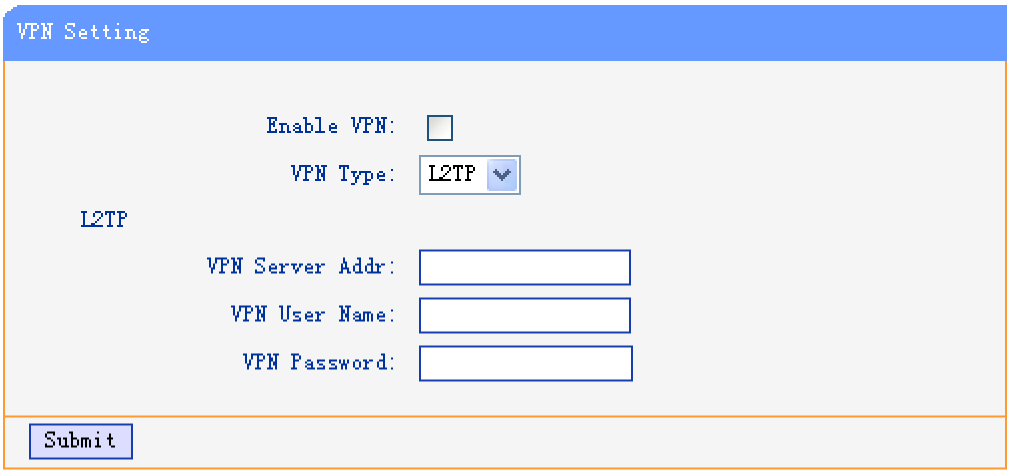
You can add the phone and PC to different VLAN used VLAN Setting option.



When used VLAN Setting option,you can set several parameters as follow:

VLAN Setting	
Enable VLAN	You can enable/disable vlan for phone and pc
VID	The vlan you want the phone or pc to join

VPN Setting



IF you need to serup a VPN Setting, you shoule fill below options.

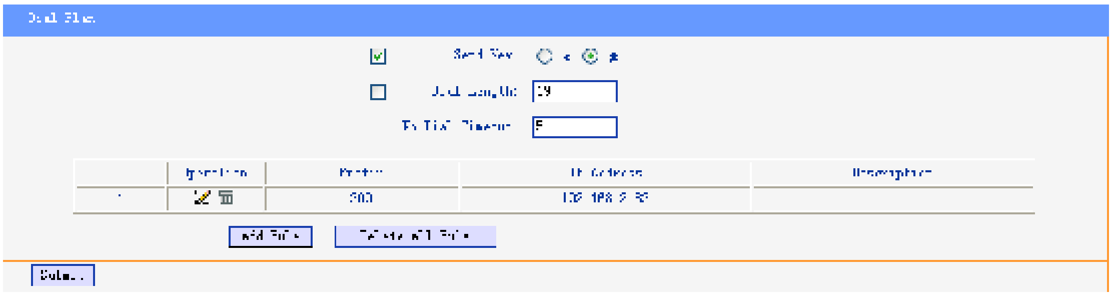
When used VPN Setting option,you can set several parameters as follow:

VLAN Setting	
Enable VPN	You can enable/disable VPN for phone and pc

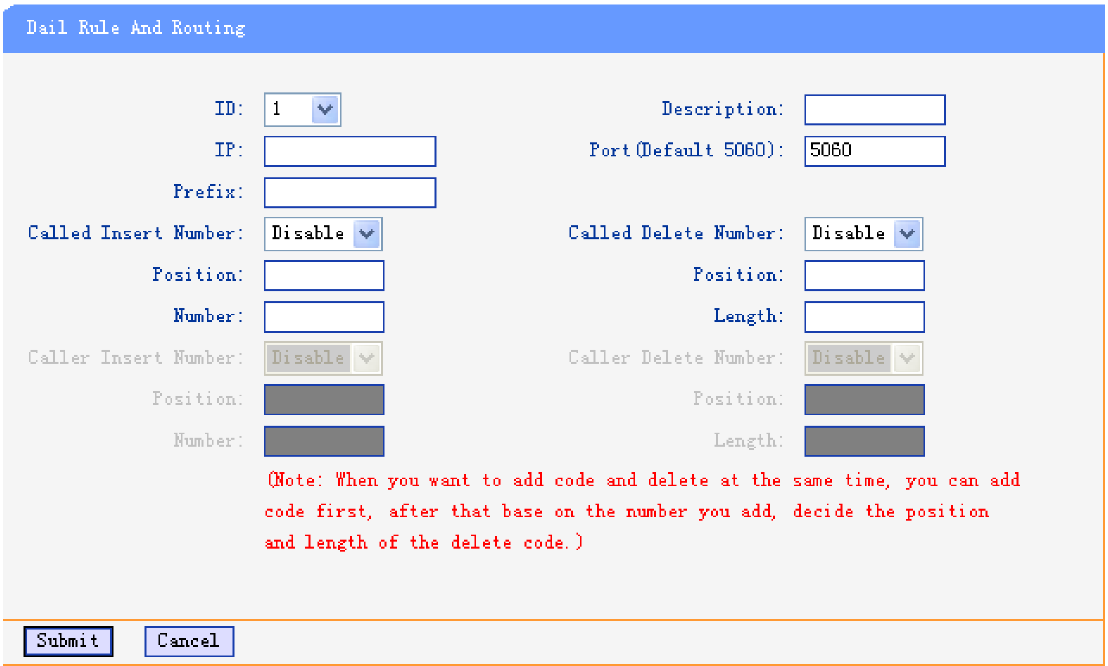
VPN Type:	There is one choose you can choice.
VPN Server Addr	VPN server'ip
VPN User Name	VPN iser's name
VPN User Name	A password be userd foe authentication

Dial Plan setting

If you want to setup a dial plan, you can click "Dial Plan".



Click "add rule" to entry this interface.



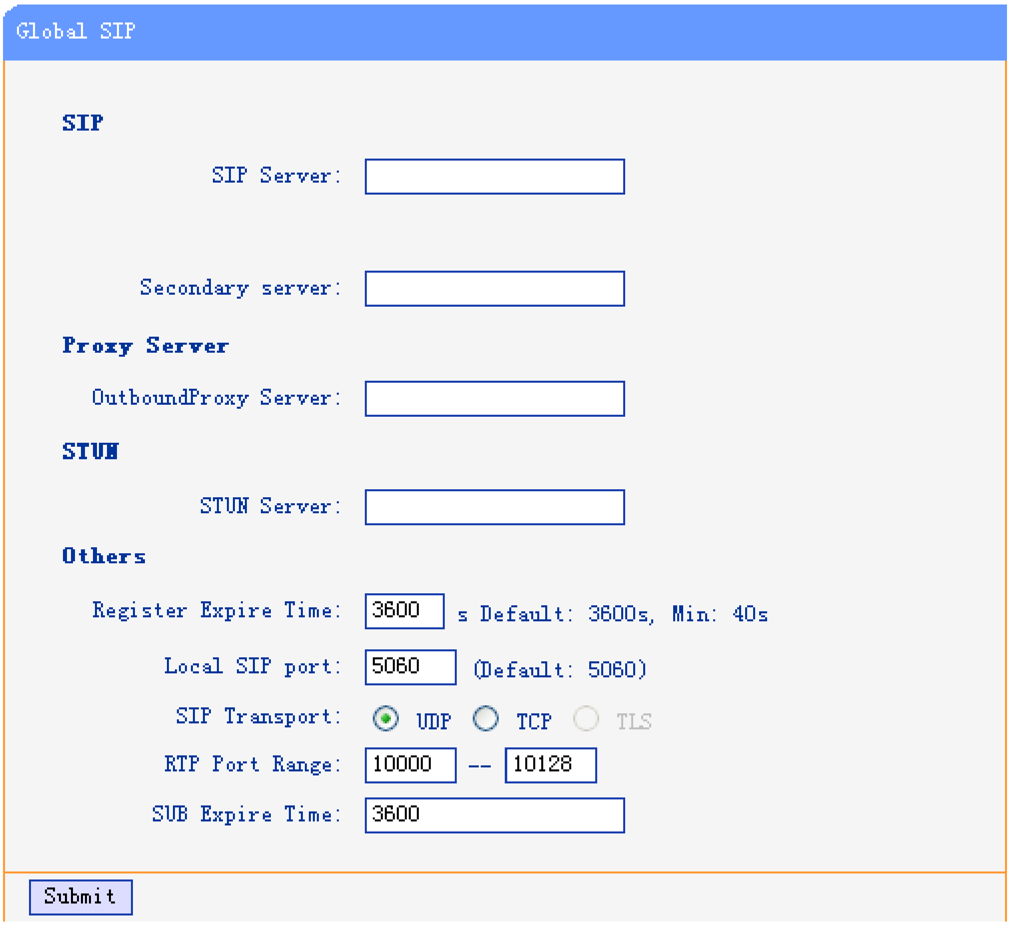
Dial Plan Setting	
ID	Dial Plan ID

IP	The ip of a phone which you want to call
prefix	The number which you need to press actually if you want to call the phone
Called Insert Number	There have two option, Enable or Disable.
Position	Which position you want insert the number
Number	Waht number you want to insert
Called Delete Number	There have two option, Enable or Disable.

Tips
(Note: When you want to add code and delete at the same time, you can add code first, after that base on the number you add, decide the position and length of the delete code.)

Global SIP

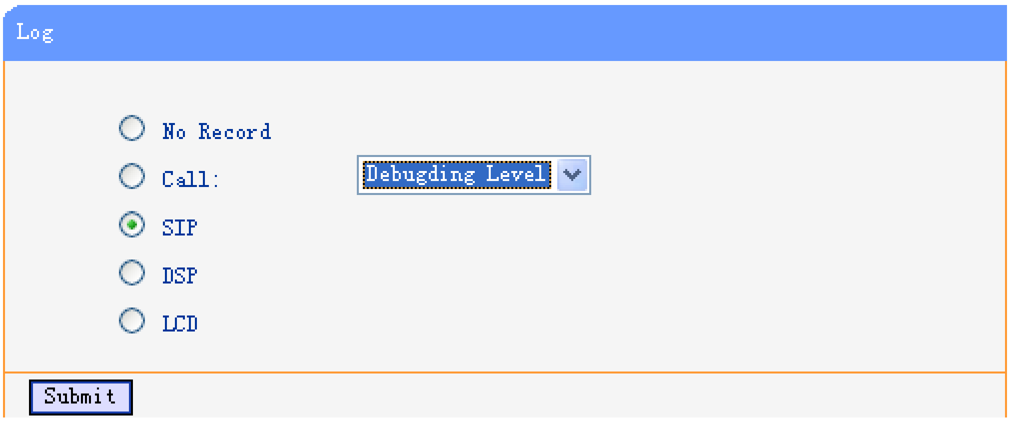
You also can setup the SIP server on Global SIP.



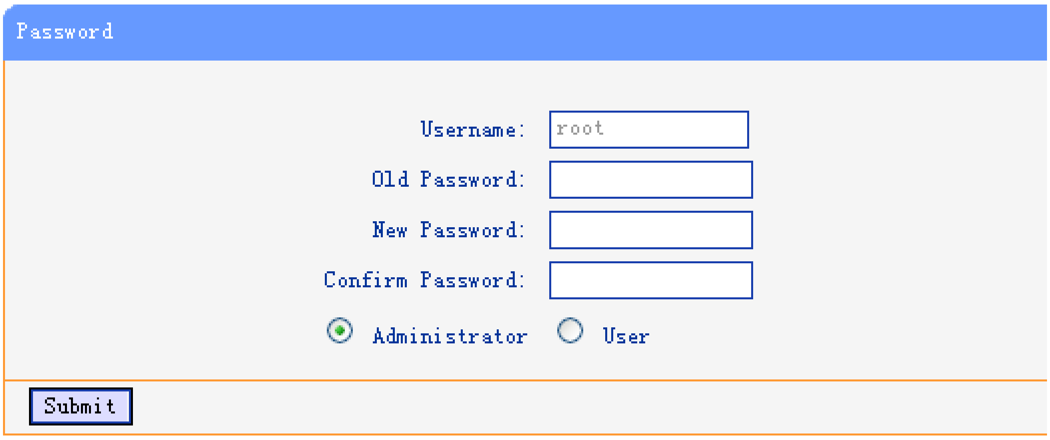
Phone Maintenance

Log

If you need to catch a debuging Level log, you need setup on this interface.



You can change the password used to login phone GUI in Password option.

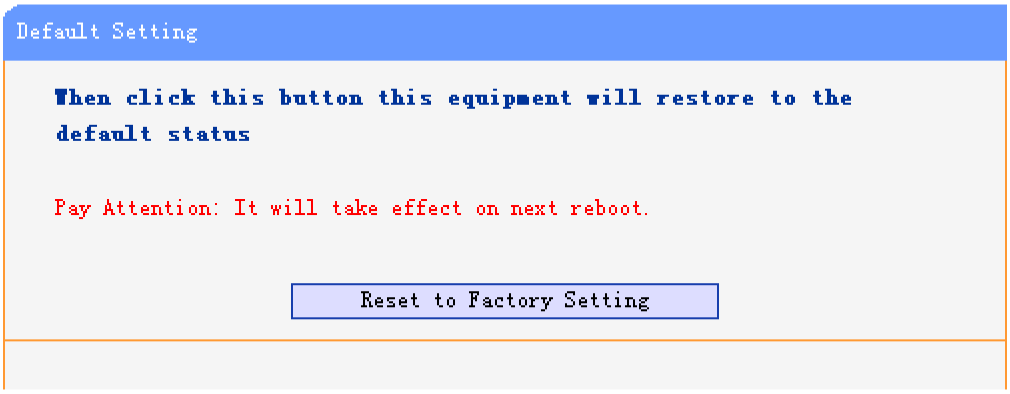


In Password option, you can set several parameters as follow:

Password	
Username	The login username of the web page
Old Password	The old password used to login of the web page
New Password	The new password used to login of the web page
Confirm Password	The new password used to login of the web page
Administrator	Login phone web page used administrator privileged
User	Login phone web page used general user privileged

Default Setting

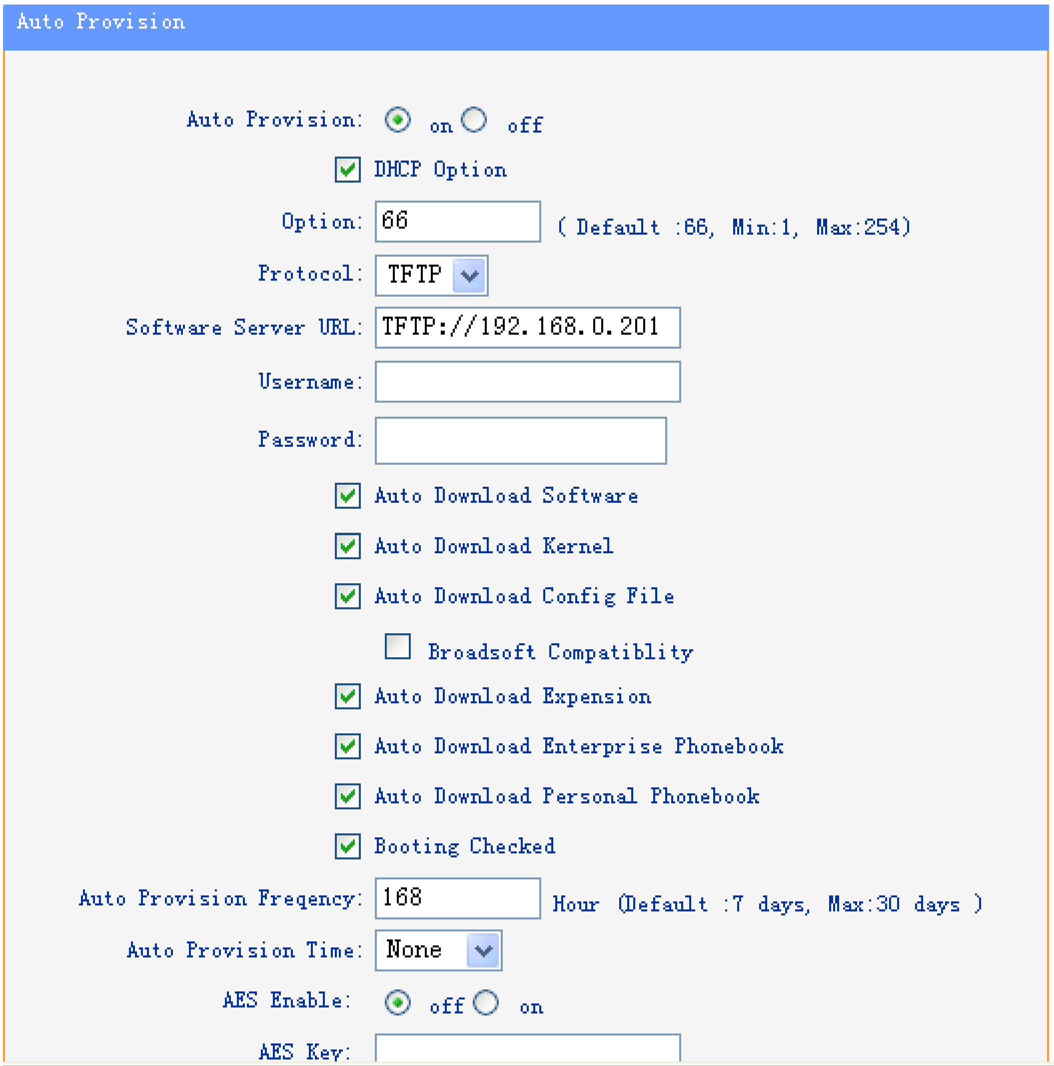
You can load the phone to the factory default setting in default setting option.



Press the 'Reset to Factory Setting' option,the phone will load to factory default setting on next reboot.

Auto Provision

When you open the auto provision function, the phone will auto provision if the phone detect a higher sofrwre or kernel which are put on the software server. The detail information about auto provision you can see the appendix.



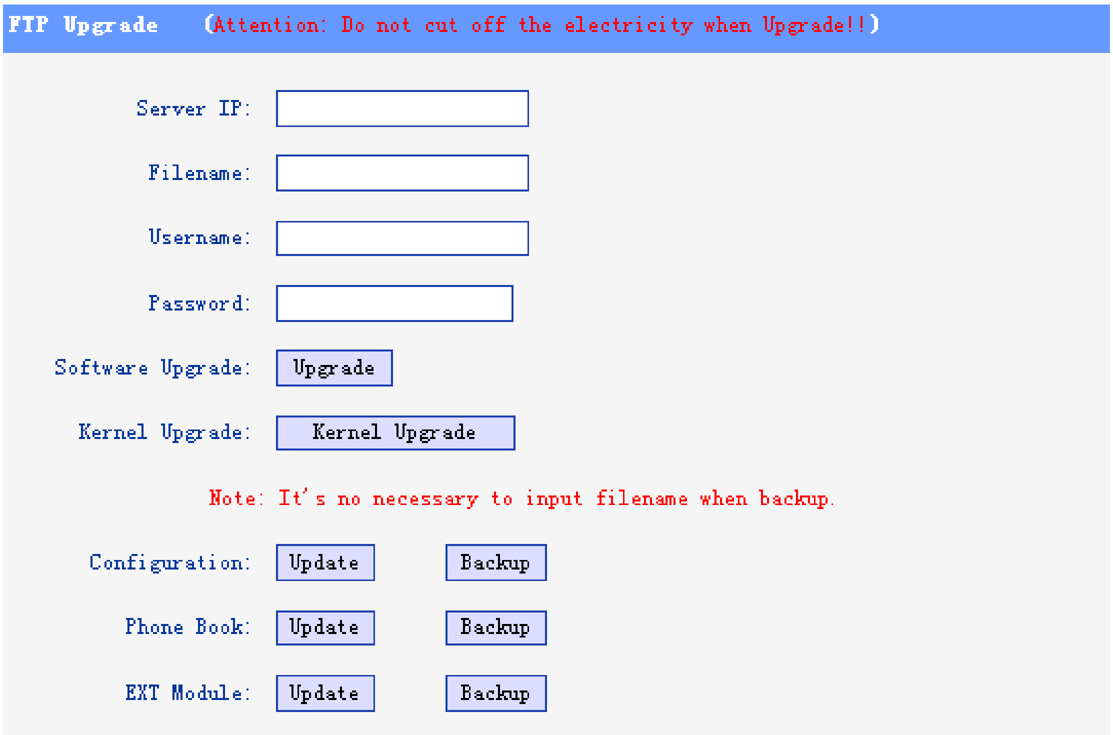
When use auto provision, you can set several parameters as follow:

Auto Provision	
Auto Provision	You can enable/disable auto provision by select on/off
Protocol	The protocol use for auto provision,it include tftp/http/ftp
Software Server URL	The server address of the auto provision
Username	The username provide by provision server
Password	The password provide by provision server
Auto Download Software	This used to auto download software from server
Auto Download Kernel	This used to auto download kernel from server
Auto Download Config File	This used to auto download config file from server
Broadsoft Compatiblity	This used to compatible the broadsoft format's config file
Auto Download	This used to auto download expansion's config from server

Expension	
Auto Download Enterprise Phonebook	This used to auto download enterprise phone from server
Auto Download Personal Phonebook	This used to auto download personal phonebook from server
Booting Checked	This used to checked the auto provision when phone booting
Auto Provision Freqency	This used to set the time interval for auto provision
Auto Provision Time	This used to the specific time for auto provision
AES Enable	You can enable/disable AES encrypt for auto provision
AES Key	The key of the AES
Auto Provision Now	This used to do auto provision immediately

FTP Upgrade

You can upgrade the software,kernel and configure file for the phone use ftp.



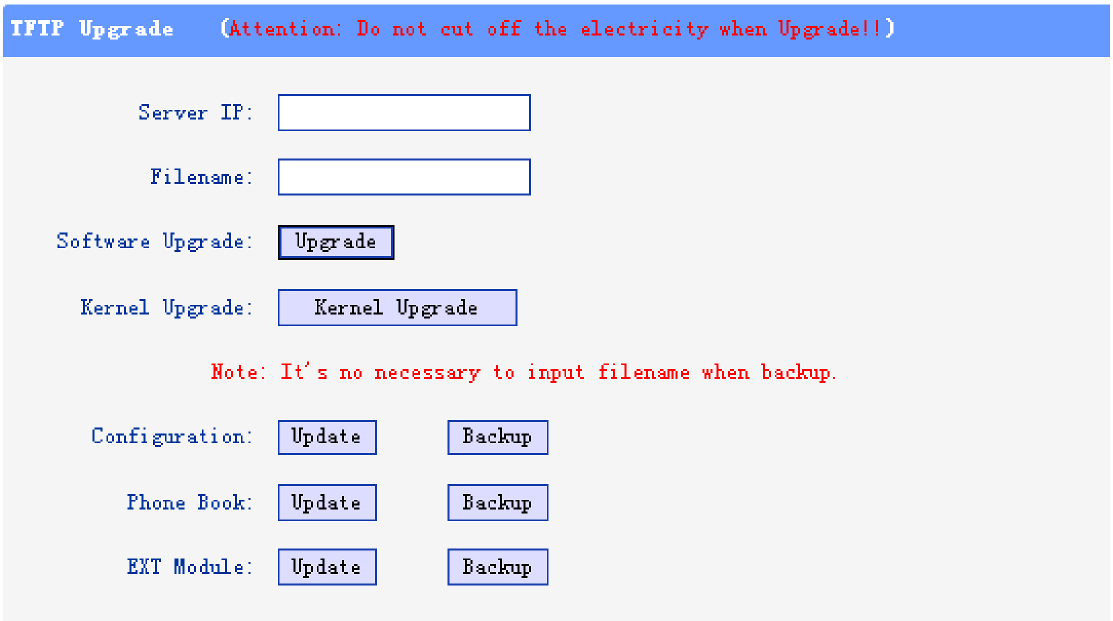
When use ftp upgrade, you can set several parameters as follow:

FTP Upgrade	
Server IP	The ip address of the ftp server
Filename	The name of the file want to download from ftp server
Username	The username provide by ftp server

Password	The password provide by ftp server
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone
Configuration	You can used update/backup to update/backup the configure file of the phone
Phone Book	You can used update/backup to update/backup the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

TFTP Upgrade

You can upgrade the software,kernel and configure file for the phone use tftp.



When use tftp upgrade, you can set several parameters as follow:

TFTP Upgrade	
Server IP	The ip address of the tftp server
Filename	The name of the file want to download from ftp server
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone
Configuration	You can used update/backup to update/backup the configure file of the phone
Phone Book	You can used update/backup to update/backup the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

HTTP Upgrade

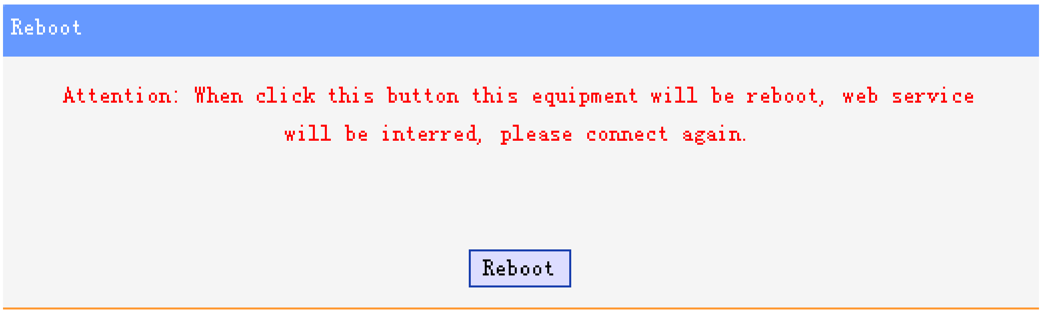
You can upgrade the software,kernel and configure file for the phone use http.

When use http upgrade, you can set several parameters as follow:

HTTP Upgrade	
Select a File	Browse the software/kernel/config file you want to upgrade from http
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone
Configuration	You can used upload/download to upload/download the configure file of the phone
Phone Book	You can used upload/download to upload/download the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

Reboot

You can use reboot option to reboot the phone.



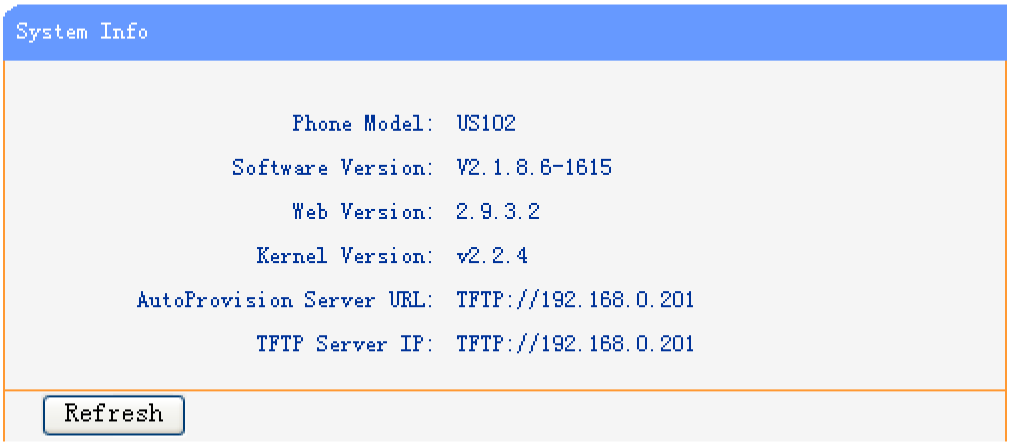
When you press ‘Reboot’,the phone will reboot.

Phone Status

You can see the currently status of the phone when use Phone Status option.

System Info

You can see the system information when used System Info option.



Attention:
On this interface ,you can see the software and kernel which we used for test and this user_manual is written base on this software and kernel.
This software version is V2.1.8.6-1615
This kernel version is v2.2.4

About

You can see the phone model when used About option.


Group
Contact
BanList
Enterprise Phoneboo
Advanced
Phone Setting
VLAN Setting
VPN Setting
Dial Plan
Global SIP
Phone Maintenance
Log
Password
Default Setting
Auto Provision
TR069
FTP Upgrade
TFTP Upgrade
HTTP Upgrade
Reboot
Phone Status
System Info
About
Please Select Language
English
IPPhone WEB Software
US102
Software Version: V2.1.8.6-1615
System Upgrade: 2011-06-29
Web Version: 2.9.3.2
Web Upgrade: 2010.07.12

Appendix：

Auto Provision

Pre-configuration on TFTP/HTTP/HTTPS/FTP Server

When the software or kernel auto-provision is enabled and want to run, IP Phone will check the software and kernel version at first, so we need make some pre-configuration on the provisioning server.

Auto Provision for Software:

1. Create a notepad file named"**F000X00.cfg**"(the "X" is decided by the model of the IP phone you are using, for example, if the model is ES620, the file name is "F000600.cfg");

   **Named rule of the file:*

   F00600.cfg: for ES620, ES610 and DS622;
   F00400.cfg: for ES410 and DS412;
   F00300.cfg: for ES310 and DS312;
   F00200.cfg: for ES210 and DS212.
2. Open the notepad file "F000X00.cfg" and write the new software name in it, for example,

   S_ES6xx_version2.0.4.6: for ES620, ES610 and DS622;
   S_ES410_version2.0.4.6: for ES410 and DS412;
   S_ES310_version2.0.4.6: for ES310 and DS312;
   S_ES210_version2.0.4.6: for ES210 and DS212

   Write down the new version you want to upgrade and save it on your provisioning server.

   **Please note that if the version is not older than (and same as) the one on your phone, auto-provision of your software would be not available.*
3. After it, upload the new software to the TFTP/HTTP/HTTPS/FTP provisioning server and complete the pre-configuration steps.

Auto Provision for Kernel:

1. Create a notepad file named"**K000X00.cfg**"(the "X" is decided by the model of the IP phone you are using, for example, if the model is ES620, the file name is "K000600.cfg");

   **Named rule of the file:*

   K00600.cfg: for ES620, ES610 and DS622;
   K00400.cfg: for ES410 and DS412;
   K00300.cfg: for ES310 and DS312;
   K00200.cfg: for ES210 and DS212.
2. Open the notepad file "K000X00.cfg" and write the new kernel name in it, for example,

   K_uImage_600.bin_version2.1.6: for ES620, ES610 and DS622;
   K_uImage_400.bin_version2.1.6: for ES410 and DS412;

K_uImage_300.bin_version2.1.6: for ES310 and DS312;
K_uImage_200.bin_version2.1.6: for ES210 and DS212

Write the new version you want to upgrade and save it on your provisioning server.

**Please note that if the version is not older than (and same as) the one on your phone, auto-provision of your kernel would be not available.*

3. After it, upload the new kernel to the TFTP/HTTP/HTTPS/FTP provisioning server and complete the pre-configuration steps.

Configuration files on TFTP/HTTP/HTTPS/FTP Server

- **Name of configuration file:**

The configuration file on the provisioning server is named as the MAC address of IP phone itself. Escene's IP phones support two different configuration files for auto-provision:

1. Normal Configuration file:

Normal Configuration file is the configuration file of your Escene IP phone. You can download it from your phone (You can see the following chapter to see how to download a configuration file from Escene IP hone) and modify by yourself. If the IP phone's MAC address is 00:11:22:33:44:55, the normal configuration file of it should be ***001122334455.xml***.

2. Broadsoft Configuration files:

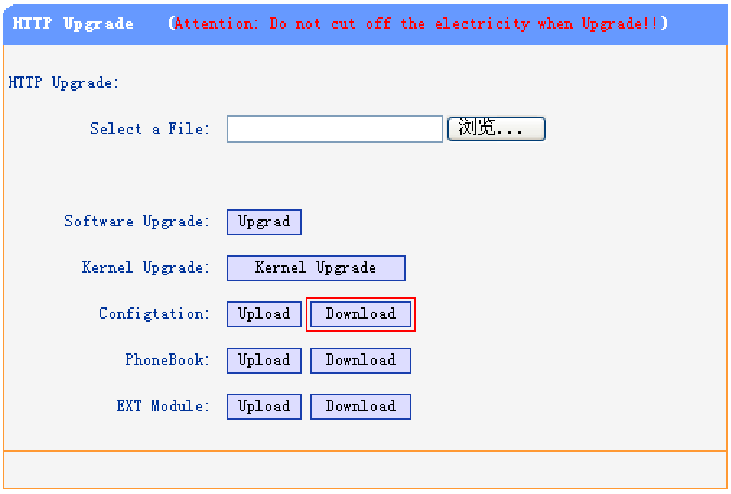
Broadsoft Configuration files support the format of Broadsoft IP-PBX. However, you can use them for provisioning. There are two files should be set on your provisioning server, they are also named by the MAC address of your phone

1) ***001122334455.cfg***: a configuration file for system settings, for example, network, audio and so on.
2) ***001122334455.txt***: a configuration file for SIP accounts.

- **Download a configuration file from your phone:**

You can download a configuration file from your phone by HTTP as follow:

1. Open the web page of your IP phone, click "Phone Maintenance">"HTTP Upgrade";
2. Then click "Download" of Configuration:



3. If you want to use this file to auto-provision, you just need to modify it by yourself and rename it to the MAC address of your IP Phone with .xml suffix.

Extern.xml file on TFTP/HTTP/HTTPS/FTP Server

The Extern.xml includes the settings of programmable buttons on the phone and all Expansion Modules. All the phones can download the settings from a same file and they will have the same settings (for example, Speed-dial, BLF and so on).

You can't rename the file on the provisioning server. The file name is fixed to* ***Account1_Extern.xml**.(Account1 is the first account you register)*

Phonebook on TFTP/HTTP/HTTPS/FTP Server

Escene IP phone supports Enterprise Phonebook and Personal Phonebook.

- **Enterprise Phonebook:**

Enterprise Phonebook is used for all staffs in your office. All phones will download a common phonebook for all staffs. The file's name must be ***Enterprise_Phonebook.xml*** on your provisioning server and you can not rename it.

- **Personal Phonebook:**

Personal Phonebook is individual for each IP phone. The file on your provisioning server is named by the first account of your IP phone. If the IP phone's first account is 1287, the Personal Phonebook of this phone is ***1287_Phonebook.xml***.

Automatic Provisioning using DHCP Option 66

The following steps will descript auto-provision by TFTP. You also can use HTTP and FTP for auto-provision with our phones.

DHCP Server: (Microsoft Windows 2003 server)

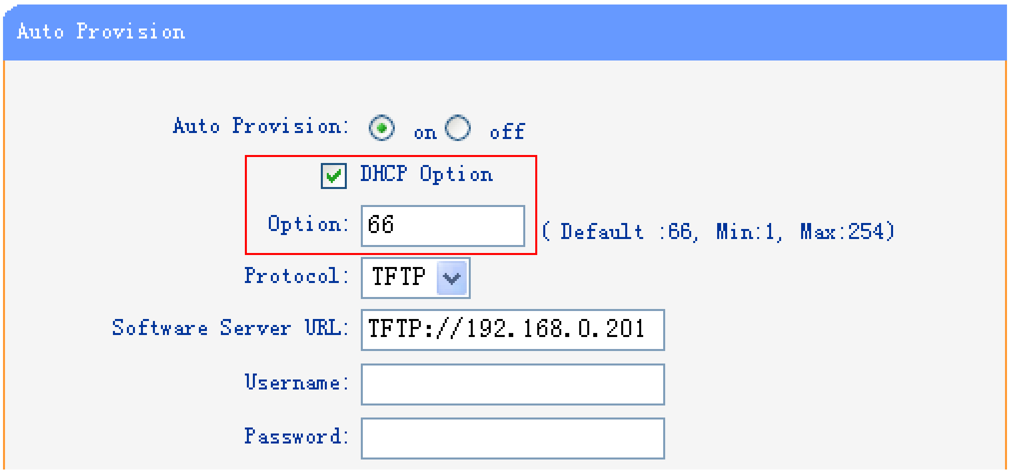
1. Start up the "DHCP Management Console";
2. Expand the DHCP scope which will contain the phones
3. Right-click on the "Scope Options" node
4. Select "Configure Options"
5. In the "General" tab, scroll down the list of options and identify the option labeled "066 Boot Server Host Name"
6. Enable the "066 Boot Server Host Name" and enter the string value according to the examples discussed previously

string value: 192.168.0.201(TFTP Server IP Address);

7. Click the "OK" button

IP Phone:

1. Input the IP Phone's IP Address in browser;
2. Enter user and password with "root" then open the web page;
3. Click "Phone Maintenance" and select "Auto Provision";
4. Select like as follows:

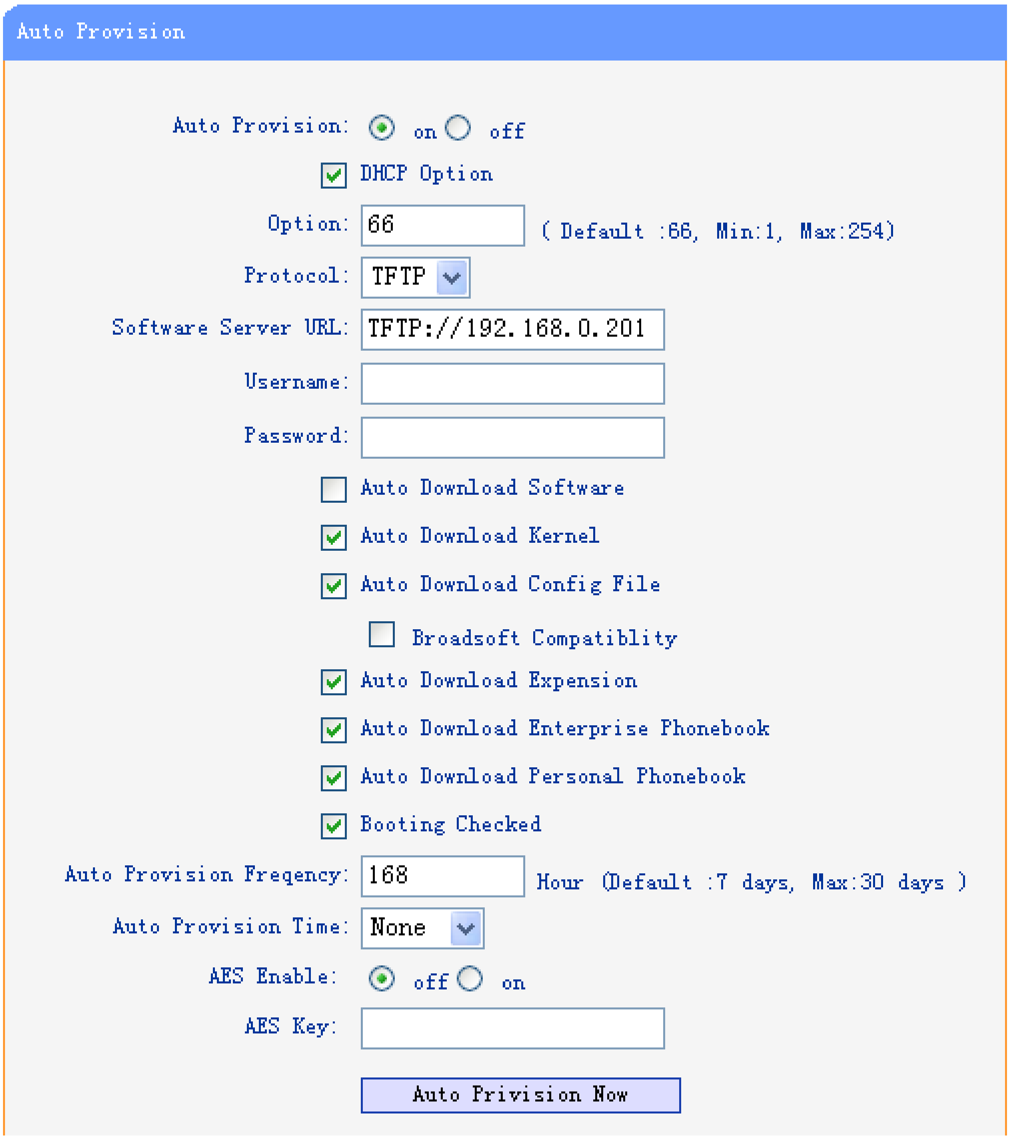


5. Click "Submit" to save it.

Auto-Provision via fixable TFTP/HTTP/HTTPS/FTP Server

IP Phone:

1. Input the IP Phone's IP Address in browser;
2. Enter user and password with "root" then open the web page;
3. Click "Phone Maintenance" and select "Auto Provision";
4. select like as follows:



It supports three protocols in Auto-Provision:TFTP,HTTP and FTP.
The format with provisioning server URL is:

TFTP:

TFTP://192.168.0.201(192.168.0.201 is the default Server IP address)

HTTP:

HTTP://192.168.0.201

HTTPS:

HTTPS://192.168.0.201

FTP:

FTP://192.168.0.201

Username: the user to login FTP/HTTP/HTTPS server
Password: the password of the user using to login FTP/HTTP/HTTPS server
**Username and password are available in FTP/HTTP/HTTPS only (unavailable in TFTP).*

Auto Download Software:

Download software from server and upgrade it automatically.

Auto Download Kernel:

Download kernel from server and upgrade it automatically.

Auto Download Config File:

Download configuration file from server and update it automatically.

BroadsoftCompatibility:

If you select this function, you need to put two configuration files (with Broadsoft format) on the provisioning server. Otherwise, you can download the configuration file from your phone via HTTP (regarding the steps, you can refer to *"Download a configuration file from your phone"* in this document.), modify it and upload it to the server for auto-provision.

Auto Download Expansion:

Download configuration file of the Programmable buttons on your phone or Expansion Modules automatically.

Auto Download Enterprise Phonebook:

Download Enterprise Phonebook from server and update it automatically.

Auto Download Personal Phonebook:

Download Personal Phonebook from server and update it automatically.

Booting Checked:

Check all items you had selected and upgrade/update them when the phone boot

Auto Provision Frequency:

The auto provision Frequency which you want.

Auto Provision Time:

The time you want to execute auto-provision.

Examples of Auto Provision Frequency and Time

1) When you set the **Auto Provision Frequency** and disable**Auto Provision Time** (set to None), the Auto Provision function will work after the **AutoProvision Frequency**;
2) When you set both **Auto Provision Frequency** and **Auto Provision Time**, for example:
   You set the **AutoProvision Frequency** to 24 hours, and the **Auto Provision Time** to 2:00 at 8:00 today (1, Jan), it will pass 24 hours at first and work at the nearest 2:00, it means that the Auto Provision function will work at 2:00 on the day after tomorrow (3, Jan).
   Therefore, if you want this function work at 23:00 tonight and it is 8:00 now, you need to set the **Auto Provision Frequency** to 0 hours and the **Auto Provision Time** to 23:00.

AES Encryption:

AES encryption is used for all the setting files of your phone (include configuration file, Expansion file, Enterprise/Personal Phonebook etc. You just need to enable the AES Encryption function and input the AES Key matching the one on your server on.

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