SayHi™

CC800v2 IP Phone
User Manual

Escene Communication Co.Ltd

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1.Getting Started

About

ESCENE CC800v2 is one of the SayHi series IP Phone in the Call Center. It has has the unique style, good utility, clear voice etc feature. Cooperating with communication platform to finish strong phone functions, such as: call transfer, hotline function (immediately/delay), a key cancellation and registration, a key automatic response, etc.

Feature Highlights:

- One key enable or disable accounts register function.
- One key enable or disable auto-answer function.
- One key change the ringing type.
- Multi-language, e.g. Chinese, English, Russian, French etc .
- Two SIP accounts and support three-way conference, SMS.
- 2xLAN, PoE, RJ9Headset.
- 5 programmable keys.
- USB port for external unit charging.
- XML/LDA, BLF/BLA
- Auto-provision, HTTP/TFTP/FTP, TR069 .
- Light of status.

Technical Features

Item	**CC800v2**
Screen	Grayscale LCD with background light
	128*64 pixel, 4 display, 2.3 inch
Language	Multi-Language (e.g.CN/EN/Spain/Portugal/Poland/Turkey/French/Italy etc.)
Line	2 ,Light status: Coming call & Hold(Red flashing);Talking (Red)
Function Keys	2 Line keys, Auto-ANS , Hold, RD & Mute, these five keys also support programmable function. Hands-free,Volume adjust, VOL,
VoIP Protocol	SIP 2.0

Network Protocol	HTTP, BOOTP, FTP, TFTP, IEEE 802.1Q, *IEEE 802.1X
Codec	PCMA,PCMU, G.722 ,G.729 A,G.723.1(5.3Kb/s, 6.4Kb/s),iLBC
QoS	TOS, Jiffer Buffer, VAD, CNG, G.168 (32ms)
Network	2×RJ45 10/100M Ethernet Interfaces (LAN/PC) IP Assignment: static IP, DHCP, PPPoE PC port support Bridge and Router DNS SRV,STUN, VPN(L2TP), VLAN/QoS STUN,DTMF(In-band/RFC2833/SIP INFO) DC 5V Power Port, USB Power Port RJ9 Call Center Headset Port, 3.5mm PC Headset Port
Voice	Hands-free model available by Full-duplex Separated 9 Level Volume Adjustment
Function APP	Quickly Register\Down Auto-Answer PC APP control calling Call Waiting, Call Queuing, Line Switching Call Forward, Call Transfer, Call Holding, Call Pickup, Callback One Key Dial, Redial Phone directory speed dial, Call record direct dial Mute
PBX	Call Transfer, Call Pick-Up, Network-Meeting, DND, Call Waiting, Call Hold. Call Barring, Call Back On Busy, Anonymous Call ,Intercom, Paging
Application	LDAP Enterprise phone directory, download with server, and it support 800 contacts Public phone directory XML Phonebook : Search /Input/ Out put Private phone directory: input/output 300 contacts, every contact can save 3 numbers and the size of number is 19 byte. Call History(600): every records is 200 with Miss Calls /Received Calls/Dialed Calls. Voice Message, Voice Mail Box, Light of Message. Ringing Update, Input, Del, *we also support to order the other APP.
Security	Login the website by password Login the LCD by password Signaling encryption(RC4) Media encryption(RC4) VPN, 802.1X, VLAN QoS(802.1pq), *LLDP TLS, MD5,AES, ROOT/USER Management

Management	Upgrade: HTTP/TFTP/FTP Auto-provision/TR069 Configurations: Phone/Http/Auto provision/TR069 Debug: Telnet/Phone/Web Keyboard Setting
Power Supply	Power adapter:AC100~240V input and DC 5V/1A output PoE(IEEE 802.af),USB
Specification	Storage Temperature: 0℃-60℃ Operating Humidity: 10%-90% Size 162x105x62MM
Certificate	CE、FCC、RoHS、Avaya、Broadsoft、Alcatel、Yeastar、Digium、Metaswitch etc.

Note: “*” Sign means function has not been published yet.

2.Connecting Your Phone

Your system administrator will likely connect your new SayHi CC800v2 IP 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 CC800v2 IP Phone; carefully check the packing list, Packing List as follows:

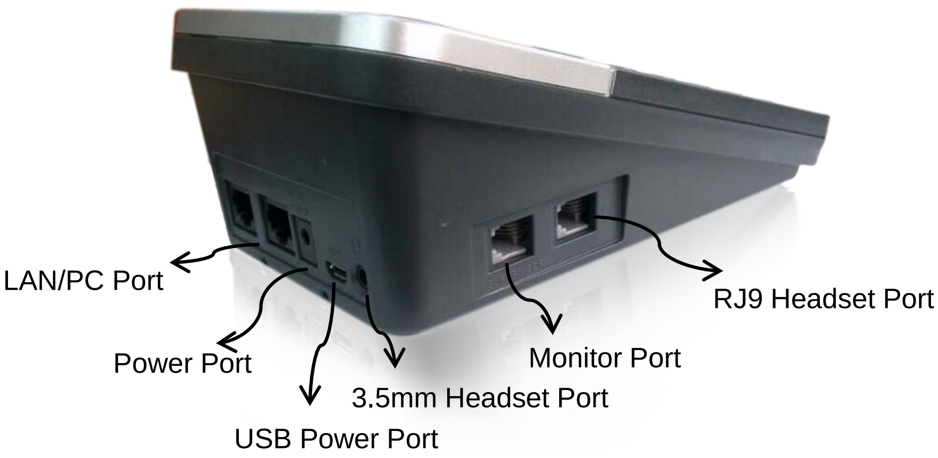
Item	Counts
IP Phone	1
Power adapter	1(Non Standard)
RJ45 cable	1
Quick Installation	1
Product certification	1

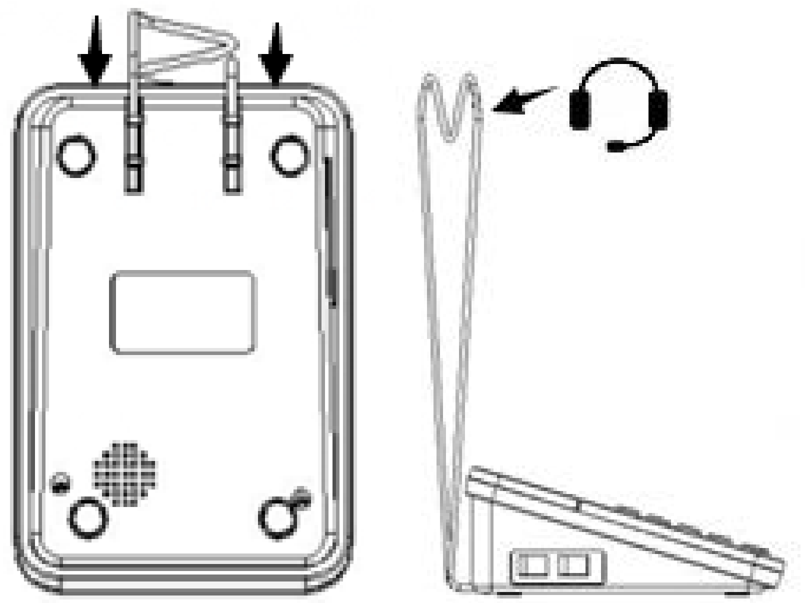
2) As shown in figure 2.1 and figure 2.2 interface；When the power up， IP Phone will automatically start if IP Phone with POE function. Connect your computer to PC interface of the phone with cable. RJ45 cable into the LAN interface

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

* More detailed description please refers to the *3.Phone overview-Understanding phone buttons and hardware*.

Figure 2.1 Interfaces of SayHi CC800v2





3.Phone overview

Understanding Buttons and Hardware

From figure 3.1 to figure 3.2, you can understand buttons and hardware about SayHi CC800v2

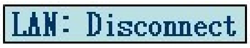
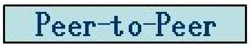
Figure 3.1 Buttons and Hardware of SayHi CC800v2

Num	Buttons & HD	Description
1	LCD	132*64 Pixel LCD.
2	Soft key	Operating function with what is the LCD show.
3	Line keys	Select the phone line（Call or Answer）； Different colors for different status: 1) Red, flashing: There is an incoming call. 2) Red, steady: Pick up and enter normal call.
4	Auto-ANS	Turn on or Shut down the auto answer service.
5	Hold	Hold button: Put a call on hold
6	RD/MUTE	RD: Redial a call. Mute button: Toggles the Mute feature on or off. Red means the feature is enabled.
7	Headset 	Headset button: Toggles the headset on or off. Red means the feature is enabled.
8		VOL±: Controls the volume and other settings VOL: Change the voice model with 
9	0-9, *, #	Basic Call Handling: press “#” send out a call(default) Navigation buttons : “Up”: -2 ; “Down”-8; “Left”-4; “Right”-6;

Understanding Phone Screen Features

This is what your main phone screen might look like:

Figure 3.3 LCD of SayHi CC800v2/CC800v2

Num	Screen	Functions
1	Time and Date	Show current time and date.
2	Auto-answer	Enabled Auto-answer, displays “AA” 
3	Missed calls	Show the number of missed calls. 
4	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.
5	Soft key labels	Each displays a softkey function (displayed on your phone screen), and the function is different when menu changes.

3.1Basic 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.

Network Setting

If you want to …	Then…

network setting	--1) Choose “Menu” > “System setting” > “Advanced setting”； --2) Enter the password required（The default is empty）； --3) Choose “Network”, you can configure the following parameters: -**Type**: static IP or DHCP -**IP**: enter IP address , Note: Do not duplicate the IP address with other devices on the network -**Mask**: enter appropriate subnet mask -**GW**: enter appropriate gateway - **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.

Placing a Call

Here are some easy ways to place a call on SayHi CC800v2 IP Phone:

If you want to...	**Then...**	
Place a call using the handset	Pick up the handset	--1) You can hear the dial tone； --2) The first line light is ； --3) Enter a number； --4) Press ‘**#’** button（default) , -or press **Send**; -or wait 5s (default), then it send the number automatically.

Redial	--Press **REDIAL** button to dial the last number - "Dialed number", select a number, and press RD.
Dial from a call log	--1) Press **MENU** or **OK** button **>** "Call history", you can select "Missed calls", "Received calls" and "Dialed numbers", - or press **Navigation button** (in Standby interface) > select "Missed calls" (**down**), "Received calls" (**left**) and "Dialed numbers" (**right**))； --2) Then press **Enter** button follow the tips and press **Dial**.
Place a call while Another call is active	--1) Press **Hold** button or Resum; --2) Select another account and enter a number; --3) Press '**#'** button (default) ; -or press **Send** to send the number.

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 CC800v2.

If you want to…	Then…	
Answer with a handset	--1) Your phone ring; --2) **Line** button of the ringing line is Red	--Pick up the handset or press the flashing **Line** button,
Switch from a connected Call to answer a ringing call	--1) Another **Line** button is Red and flashing, Light strip is Red and flashing; --2) Press the flashing **Line** button to answer (at this time, the original call will be hold.)	
Auto-answer	--1) Press **MENU** or **OK** button > "Function setting" > "Auto answer" or press AUTO ANS; --2) Select "Enable"; --3) Your phone answers incoming calls automatically after a few rings.	

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, -or press END
Hang up while using the Speakerphone	--Press **Speaker** button , -or press **Line** button for the appropriate line, -or press END
Hang up while using the Headset	--Press **Handset** button, （Do not keep the headset mode) , -or press END (keep the headset mode)
Hang up one call, but preserve another call on the other line	--Press END, -or refer to the above three methods

Using Hold and Resume (Switch Calling Line)

You can hold and resume calls. You can take a call in one line at any time, and the other lines would be hold. As a result of that, you can switch different calling line on our phone.

If you want to...	Then...
Put a call on hold	--Press **HOLD** button under the LCD, -or press **Hold**
Hold a line and switch to another line	Press another **Line** button for the appropriate line
Resume a call on current line	--Press **Line** button,
Release a call on different line	Select the line want to release hold, press the line, so recovery;

Tips

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

• A held call is indicated by the red line and flashing red line button.

Transferring 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 XFER soft key under the LCD; --2) Enter number; --3) press “**#**”（default) , -or press **Send** then transfer the call, -or wait five seconds(default)then transfer the call
Transferred to idle lines or other numbers without talking to the transfer recipient (Blind transfer)	--1) Press XFER soft key under the LCD;； --2) Press **Blind**; --3) Enter number; --4) Press “**#**”（default) -or press **Send**, then transfer the call; -or wait five seconds(default)then transfer the call
Blind transfer to the held line	--1) Press XFER soft key under the LCD;; --2) Press the **Line** button of held line

Using Mute

With Mute enabled, you can hear other parties on a call but they cannot hear you. You can use mute in conjunction with the handset, speakerphone, or a headset。

If you want to...	Then...
Toggle Mute on	Press **Mute** button, then the button is Red
Toggle Mute off	Press **Mute** button, then the button light off

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 single line	Press **MENU** or **OK** button > "Function setting" > "DND" > (select line) "Enable", e.g. 
Disable DND	--Line DND enabled, press twice **DND**, -or press **MENU** or **OK** button > "Function setting" > "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 CONF under the LCD; --2) Then the held one, transfer recipient and you will be into a conference, and the LCD will display conferenc 0:0:10 status.
Invite the third party into a conference in a active call	--1) Press **CONF** button under the LCD; --2) Enter the third party number； --3) press "**#**"（default) , -or press **Send** then transfer the call,

establish a conference with held line	--1) when one phone line is holding on and the other line is busy; --2) Press CONF Soft key --3) press the held line's button, the 3-way Conference will establish.

VOIP Call Forwarding

If you want to...	**Then...**
Unconditional transfer	--1) Press **MENU** or **OK** button > “Function setting” > “voip call forwarding”; --2)select “unconditional transfer”, select enable. --3)input number which you want to transfer, when have a call in ,it will unconditional transfer.
Busy transfer	--1) Press **MENU** or **OK** button > “Function setting” > “voip call forwarding”; --2)select “busy transfer”, select enable. --3) input number which you want to transfer, when have a call in conversation ,it will transfer.
No answer transfer	--1) Press **MENU** or **OK** button > “Function setting” > “voip call forwarding”; --2)select “no answer transfer”, select enable. --3) input number which you want to transfer, when have a call in but you don’t have time to answer ,it will transfer.

3.2 Advanced Call Handling

Speed Dialing

Speed dialing allows you to enter an index number, press a button, or select a phone screen item to place a call.

If you want to...	Then…
Set up Speed Dials on your phone	--1) Press **MENU** or **OK** button > “Function setting” > “Hot line”; --2) Press Enter and to select Enable --3) Number: Need to speed dial numbers --4) Press OK to submit --1) Press **MENU** or **OK** button > “Function setting” > “Delay line”; --2) Press Enter and to select Enable --3) Number: delay dial the number after 5 second --4) Press OK to submit

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.

If you want to…	Then…

Add Contacts	--1) Press **MENU** button > “Phone book”, --2) Press Opt. [Tips: opt. means modify.] --3) Select “Add contact”, press **ENTER** --4) Use the navigation keys to select content, press **opt.** button to modify: -Name: set the name of contact, -NO.1-5: you can set up 5 contacts’ numbers, -Group: the contacts be divided into different user’s groups --5) Press **Save** soft key to complete
Add group	--1) Press **MENU** button > “Phone book” --2) Press Opt. [Tips: opt. means modify.] --3) Select the “add group” then press **ENTER** button --4) Use the navigation keys to select content, press **opt.** button to modify -Group name: name of the group -Description: description of the group --5) Press **Save** soft key to complete
Modify group	--1) Press **MENU** button > “Phone book”, --2) Press Opt. [Tips: opt. means modify.] --3) Select the “Modify group” then press **OK** button --4) Select the group you want to modify, press the **OK** button save the change
Delete group	--1) Press **MENU** button > “Phone book”, --2) Press Opt. [Tips: opt. means modify.] --3) Select the “Delete group” --4) Select a group you want to delete, press **OK** button

View/Edit Contacts	--1) Press **MENU** button > "Phone book", --2) Select "View ALL", -or select a contact who are belong to different group; --3) Select the contact, press the **ENTER** button or Opt. (to edit the contact's information)
LDAP	--1) Press **MENU** button > "Phone book" --2) Select "LDAP", press the **ENTER** button. --3) Select"Search name->name", then input the name ,and press **OK** or **Del.** --4)Select "Search number->Number", then input the number ,and press **OK** or **Del.** **Pay attention: before you use LDAP function, you need to configure LDAP rule in the web configure page.**
Call from phone book	--1) Press **MENU** button > "Phone book", --2) Select "View ALL", -or select a contact who are belong to different group; --3) Select a contact, then press **Dial**, (If there are multiple numbers of one contact, press **Dial** to enter the interface of "call options", select the one you want to call and press **Dial**)
Modify the relative account of a contact	--1) Open your web browser, enter the "web" interface. (For details, you can refer to *7.Web Settings*.) --2) Open "Contact" > "Phone book", select the contact who are needed to be modified, click  --3) Select the account in the drop-down column of the account, click "**Submit**" to complete it.

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 > "Call history" > "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.*
Erase your call logs	--1) If you want to delete a call record, you have to select this record from the logs and press **DEL;** --2) If you want to delete an entire call record list, you have to select this record list from the logs and press **Clear**

Black List

You can add, edit or delete black list in a phone book.

If you want to…	**Then…**
View your phone book	--1) Press **MENU** button > "Phone book" > "Personal Phone Book", "View all", or "Groups member" --2) Use the navigation keys to view the members information.
Put into the Black List	Use the navigation keys to select "Put into the Black List", Press the soft key "**Opt.** or **5**" to submit.
Erase your Black List members	--1) If you want to delete a black list member. Press **MENU** button > "Phone book" > "Black List" --2) Pls select "Move to personal phone contacts" and press **ENTER**

Fuzzy search

Search by phone number to identify someone by their landline or cell phone number using a digital number to accurate results.

If you want to…	Then…
Open this function	--1) Press **MENU** button > “Function Setting” > “Fuzzy Search” --2) Press **ENTER** and make it **Enable**.

Time & Date

If you want to...	Then...
Time & Date	--1) Press **MENU** or **OK** button > “Function setting” > “time & date”, --2)you can select : SNTP: select “enable ”to set parameter: time 、server 、daylight SIP server: select “enable ” to set parameter: root can modify date . manual Settings: select “enable ”to set parameter: date and time

3.3 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** button to the main menu, Use the navigation keys to select menu, press **ENTER** button to confirm menu selections, press **BACK** button or **DEL** to delete input information.

Language

SayHi CC800v2 IP Phone supports Chinese English Russian French Polish Spanish Portuguese Turkish Italian Portuguese (Brazil). As the following sample is how to setting English.

If you want to…	**Then...**
To change the language via Phone interface	--1) Choose "Menu" > "Language"; --2) Scroll through the list of available languages. --3) Press **ENTER** button when the desired language is highlighted. The language appears on the graphic display will be changed to the one you chose.

SIP Account Settings

SayHi CC800v2 series IP phone make calls based on sip accounts, SayHi CC800v2 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) Choose “Menu” > “System setting” > “Advanced setting”； --2) Enter the password required（The default is empty) ； --3) Choose “SIP” > “Account sip”； --4) Choose one of the account you want to setting, you can configure the following parameters -**Enable account***: choose Enable -**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. --5) Set up the above parameters, Press **Save** soft key to submit, Complete the account creation； * **Note**: the parameters with the * mark must be set.
Disable sip account	--1) Choose “Menu” > “System setting” > “Advanced setting”； --2) Enter the password required（The default is empty) ； --3) Choose “SIP” > “Account sip”； --4) Choose “Enable account” > “Disable”； --5) Press **Save** soft key to submit.

Load default settings

If you want to...	Then...
Load default settings	--1) Choose “Menu” > “System settings” > “Advanced settings”； --2) Enter the password required（The default is empty)； --3) Choose “load default settings” and press ’**OK**’, then “**Reboot”** the phone.

Customizing Rings and Volume

If you want to…	Then…
Change the ring tone	--1) Choose “Menu” > “System setting” > “Phone setting” > “Ring type”； --2) Press navigation to choose ring tone, it will auto play the voice. --3) Press OK soft key to set the ring tone, Press **BACK** soft key to cancel
Adjust the volume level	--1) Choose “Menu” > “System setting” > “Phone setting” > “Volume setting”; --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

View status

If you want to see the phone status, Press **MENU** button > “view status” , you can see the detail information of the phone. Also you can press **INFO** button under the LCD, it can quickly into the summary [Software version\IP\Mask\MAC\Network type\Kernel version\Phone Mode]

If you want to	**Then......**
Network	You can see the network detail information of the phone
Lines	You can see the SIP account
software	It include phone Mode、software version、kernel version、Upgrade date、Running time
Expansion	Can check the expansion

Diagnose

If you want to check the phone hardware function,Press **MENU** button > “diagnose” ,or press **ENTER** button > “diagnose”, you can check the phone item as below.

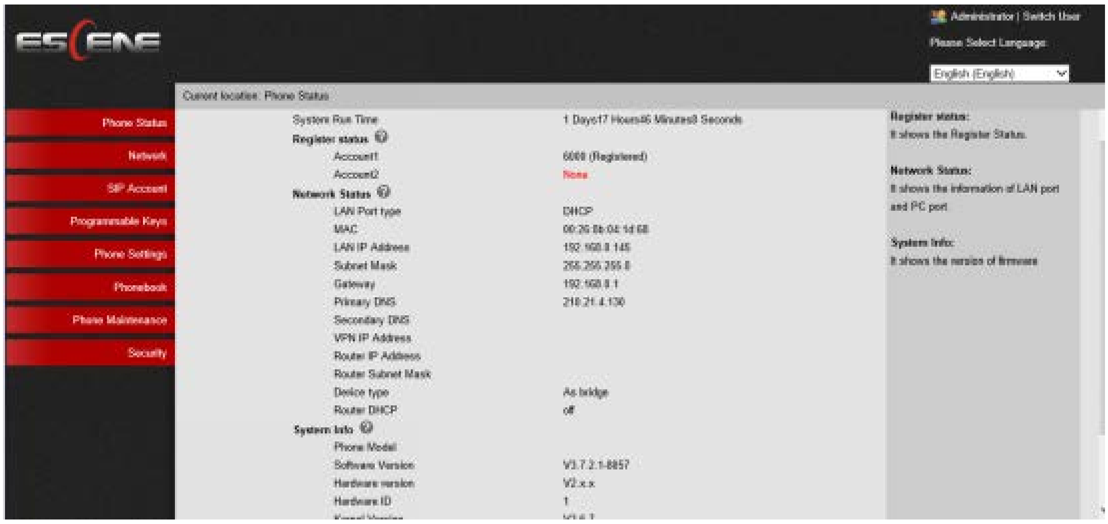
If you want to	**Then ....**
Keys	You can check the phone keys
LCD	Press’**ENTER**’to start,press’**BACK**’to exit
Lights	Press’**ENTER**’to start,press’**BACK**’to exit
Sound	Press’**OK**’to start , press’**BACK**’to exit

4.WEB User Interface

In addition to the phone user interface, you can also customize your phone via web user interface. In order to access the web user interface, you need to know the IP address of your new phone. To obtain the IP address, press the C key on the phone. Enter the IP address (e.g. HTTP://192.168.0.10 or 192.168.0.10) in the address bar of web browser on your PC. The default user name is root (case-sensitive) and the password is root (case-sensitive).

Main Interface-Phone Status

Here you can see as below information: System Run Time, Register Status, Network Status, System Information,

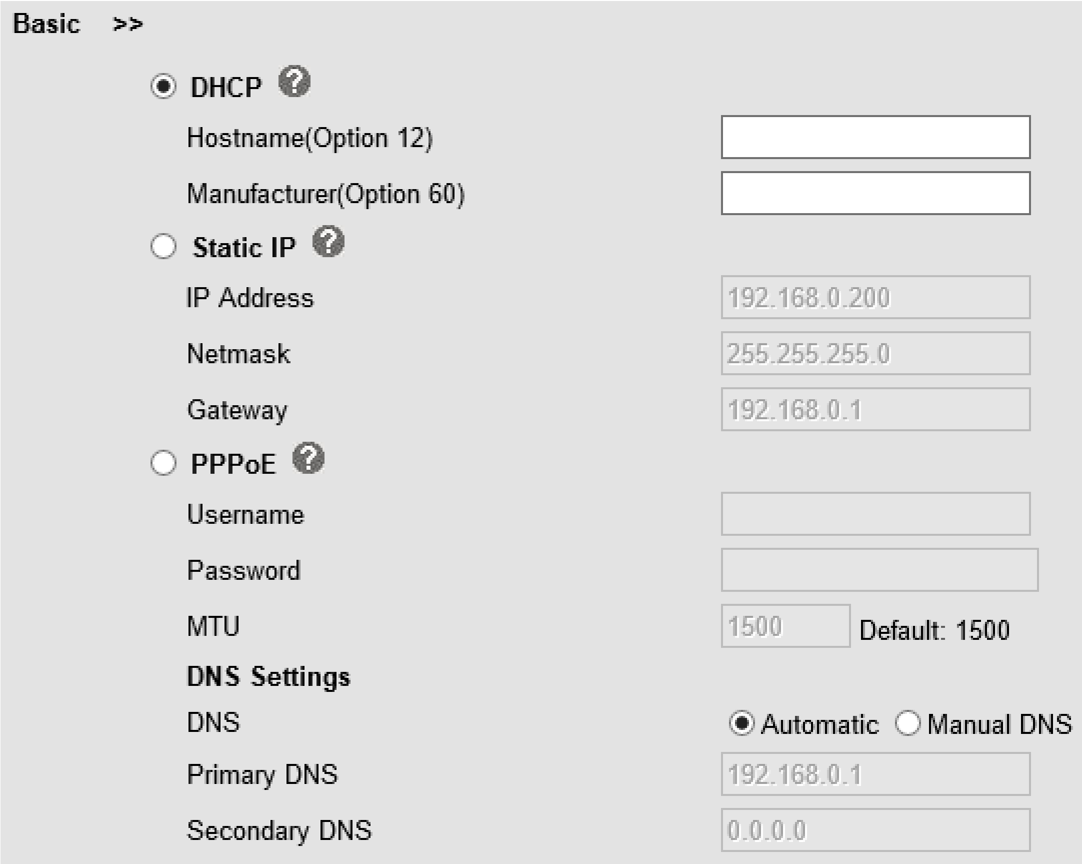


ITEM	DESCRIPTION
System Run Time	The phone system normal running time.
Register Status	The status with Account 1~2.
Network Status	The status with LAN, MAC, LAN IP, Net mask, Gateway, Primary DNS, Secondary DNS, VPN IP, PC IP, PC Net mask, Device Type, DHCP Server.
System Information	The status with Phone Model, Software Version, Hardware Version, Hardware ID, Kernel Version, Auto-Provision Server URL, TFTP Server IP.

4.1 Net Work

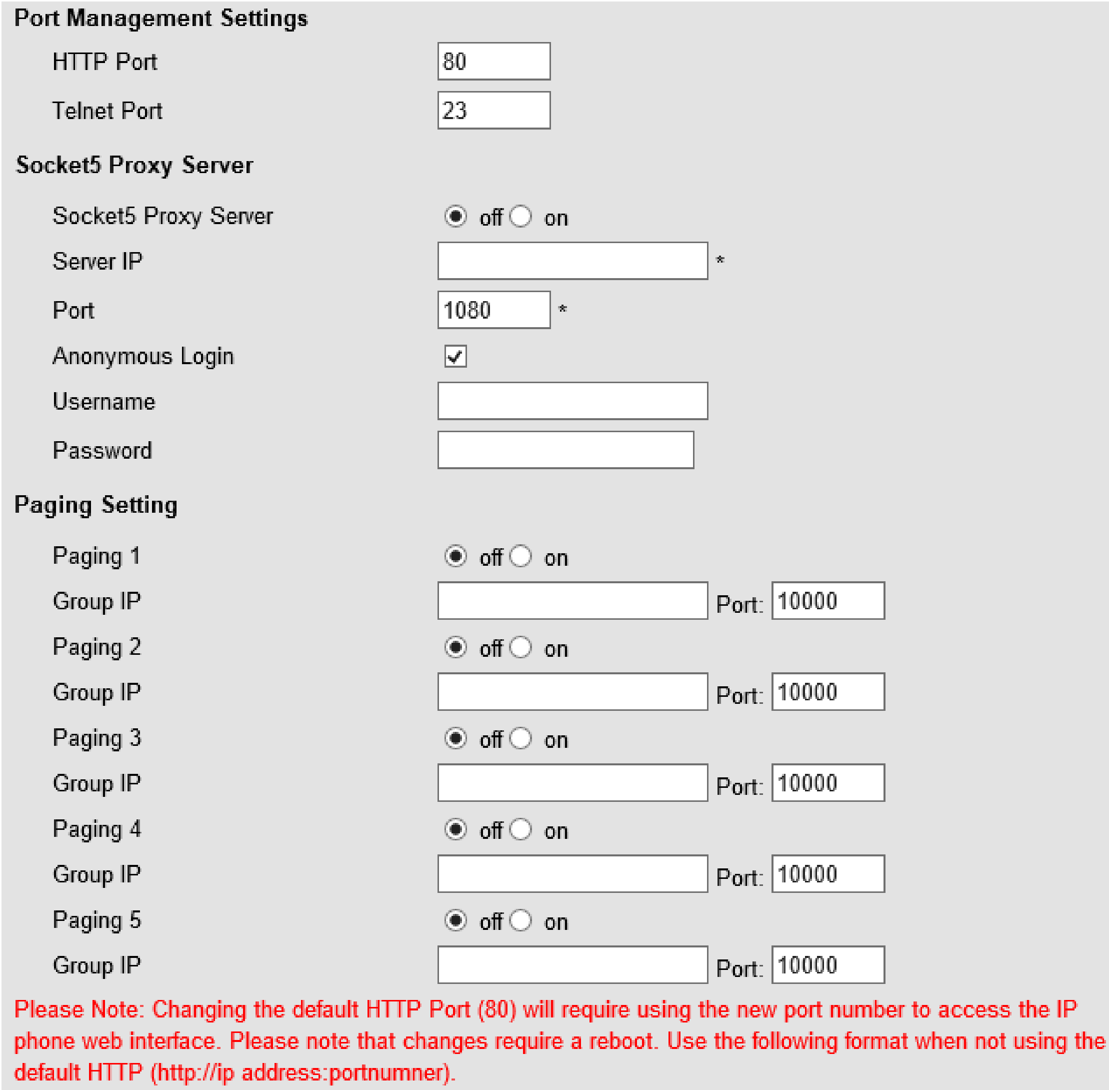
4.1.1 LAN Port

Basic



ITEM	DESCRIPTION
Network Connection Mode	Network Connection Mode has DHCP, Static IP, PPPoE.
DNS Settings	Select the DNS mode that you want.

Advanced



ITEM	DESCRIPTION
Port Management Settings	
HTTP Port	The default web port is 80,if you want to change it(for example change it to88), You must input 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 want to change it(for example change it to 2003). You must input IP and Telnet port to login the manage page (for example telnet 192.168.0.200:2003).It will take effect on next reboot.
Socket5 Proxy Server	
Socket5 Proxy Server	Enable/Disable Socket5 Proxy Server.
Server IP	Socket5 Proxy Server IP address.
Port	Socket5 Proxy Server port, default is 1080.

Anonymous Login	Enable/Disable Socket5 Proxy Server login username.
Paging Setting(NOTE: This feature priority is followed the serial number, In other words, "paging 1" is the highest priority)	
Paging1	Enable/Disable Paging feature.
Group IP and Port	Group IP and Port with Paging.

4.1.2 PC Port

Normally choose Bridge, if you choose Router ,you need to input router IP address ,net mask.

Bridge

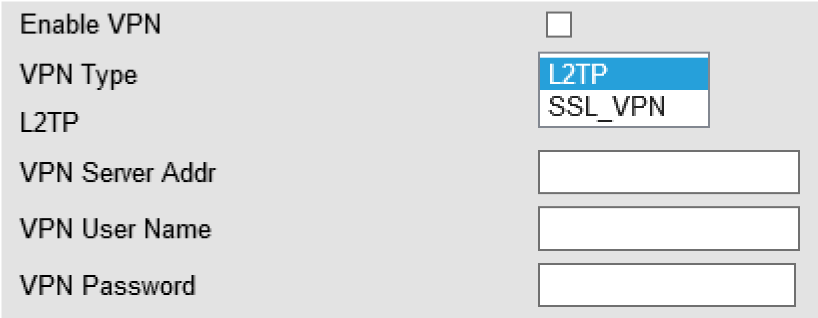
Normally, you should choose “bridge” feature, it means that pc port and LAN port will share the same network.

Router

Router feature is for the phone PC Port. You must input IP address (it’s equivalent to a gateway) and Net mask. If you want to use DHCP function, please turn it on, input start IP and end IP.

4.1.3 Advanced

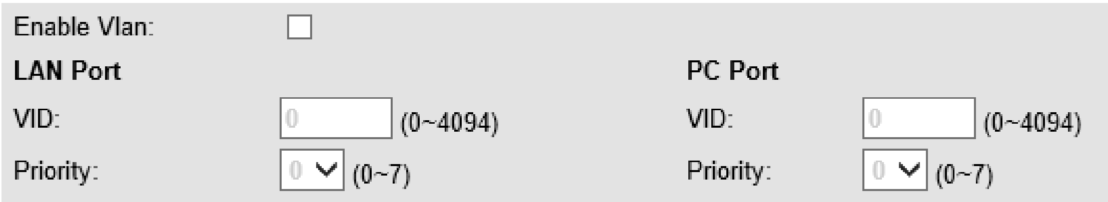
VPN Setting



When using 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:	Choose the appropriate type of VPN.
VPN Server Addr	VPN server's IP.
VPN User Name	VPN user's name
VPN Password	A password be used for authentication

VLAN Setting

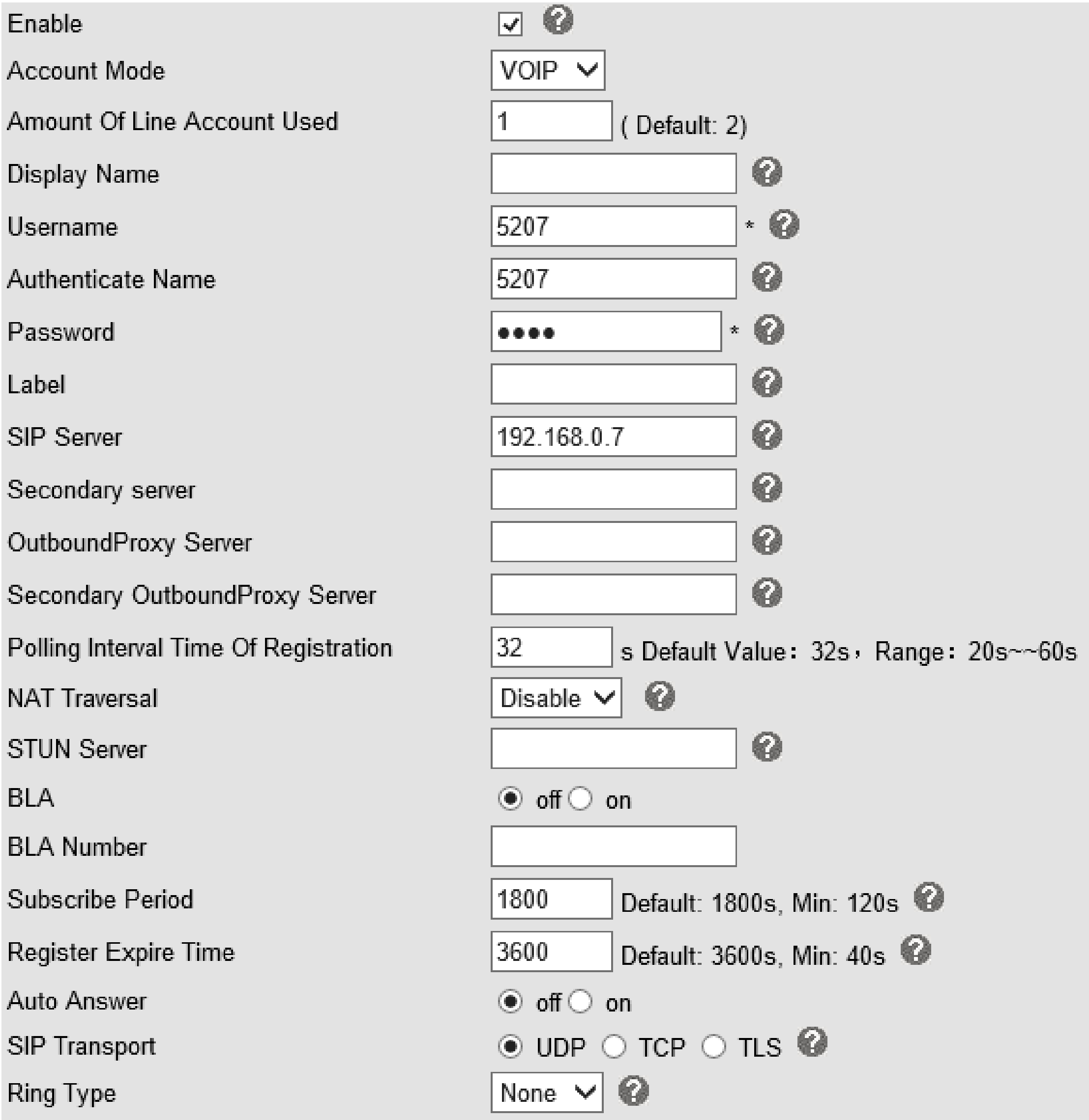


When using VLAN Setting option, you can set several parameters as follow:

VLAN Setting	
Enable VLAN	You can enable/disable vlan for phone and pc
VID [LAN/PC Port]	The vlan ID you want the phone or pc to join

5 SIP Account

5.1Basic

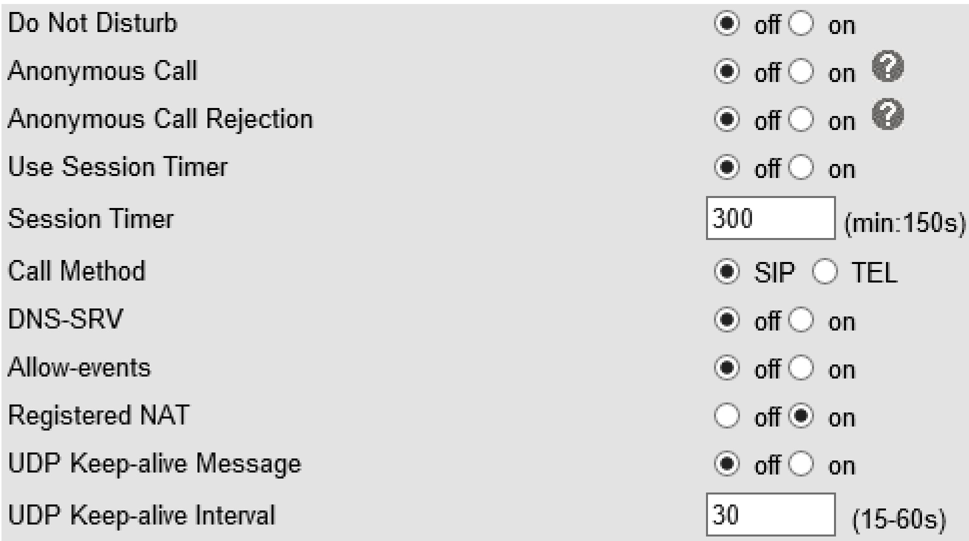


Choose one Account, you will find the following parameters:

ITEM	DECSRIPTIO
Enable	You can choose on/off to enable/disable the line.
Account Mode	You can choose VOIP/PSTN, but this model nonsupport PSTN, If you need, Pls contact us to buy another model that can supports PSTN.
Amount Of Line Account Used	The line key of account used, default is 2
Display Name	It is showed as Caller ID when making a phone call
Username	It is a username provided by SIP Server
Authenticate Name	It is authenticated ID for authentication
Password	It is a password provided by SIP Server
Label	Label with this account.
SIP Server	Server for registration, provided by administrator

Secondary server	When the main server can't work, it also can register in this secondary server.
Outbound Proxy	Put into the address with the outbound proxy server.
Secondary Outbound Proxy Server	When the main out bound server can't work, it also can use this secondary server.
Poling Interval Time Of Registration	Poling Interval Time Of Registration, default is 32 s.
NAT Traversal	Defines the STUN server will be active or not
STUN Server	Session traversal utilities for NAT.
BLA	Share with the line.
BLA Number	BLA Number
Subscribe Period	Subscribe expire time.
Register Expire Time	IP phone automatically registered every time
SIP Transport	There are UDP/TCP/TLS three options
Ring Type	Select this account ringing type.

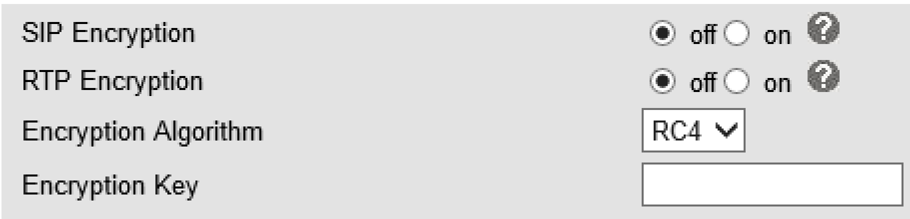
5.2 Call



ITEM	DECSRIPTIO
Call	
Do Not Disturb	Enable/Disable Do Not Disturb
Anonymous Call	Enable/Disable anonymous call.

Anonymous Call Rejection	Enable/Disable anonymous call rejection.
Use Session Timer	Enable/Disable refresh session function. The device will send an Invite packet to refresh the session during a call if it enable.
Session Timer	The refresh session time interval.
Call Method	This method include SIP and TEL.
DNS-SRV	Enable/Disable DNS-SRV.
Allow-events	Enable/Disable Allow-events.
Registered NAT	Enable/Disable Registered to NAT
UDP Keep-alive Message	The phone periodically sends a UDP packet to keep the port active and to avoid the server to shut down the port
UDP Keep-alive Interval	Default is 30 second.

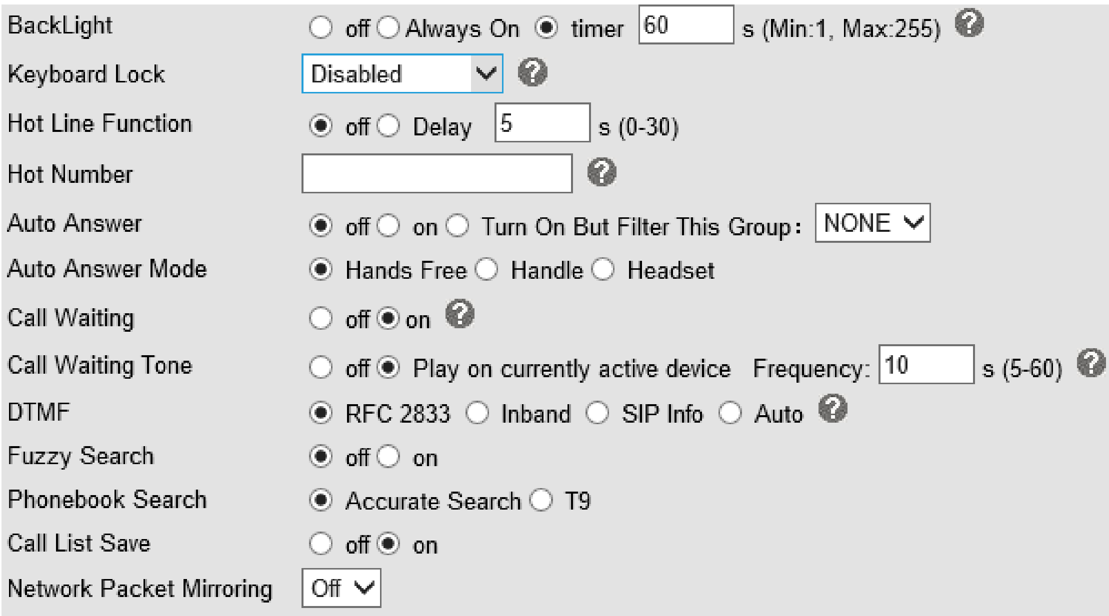
5.3 Security



ITEM	DECSRIPTIO
Security	
SIP Encryption	Enable/Disable SIP encryption.
RTP Encryption	Enable/Disable RTP encryption.
Encryption Algorithm	The encryption algorithm at this time we only have RC4.
Encryption Key	The key with encryption.

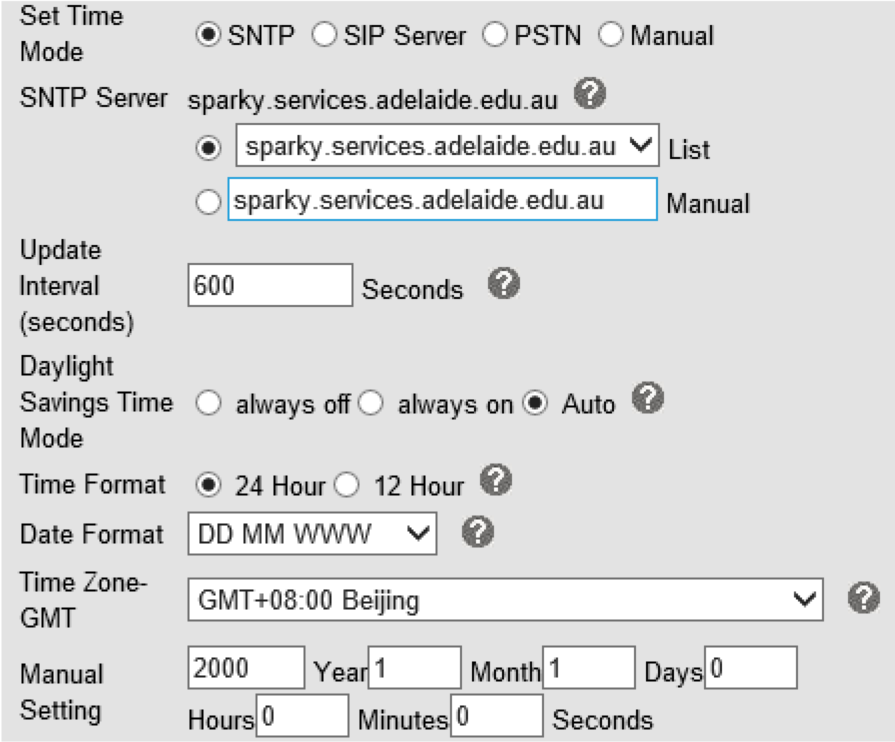
6 Phone Setting

6.1 Basic



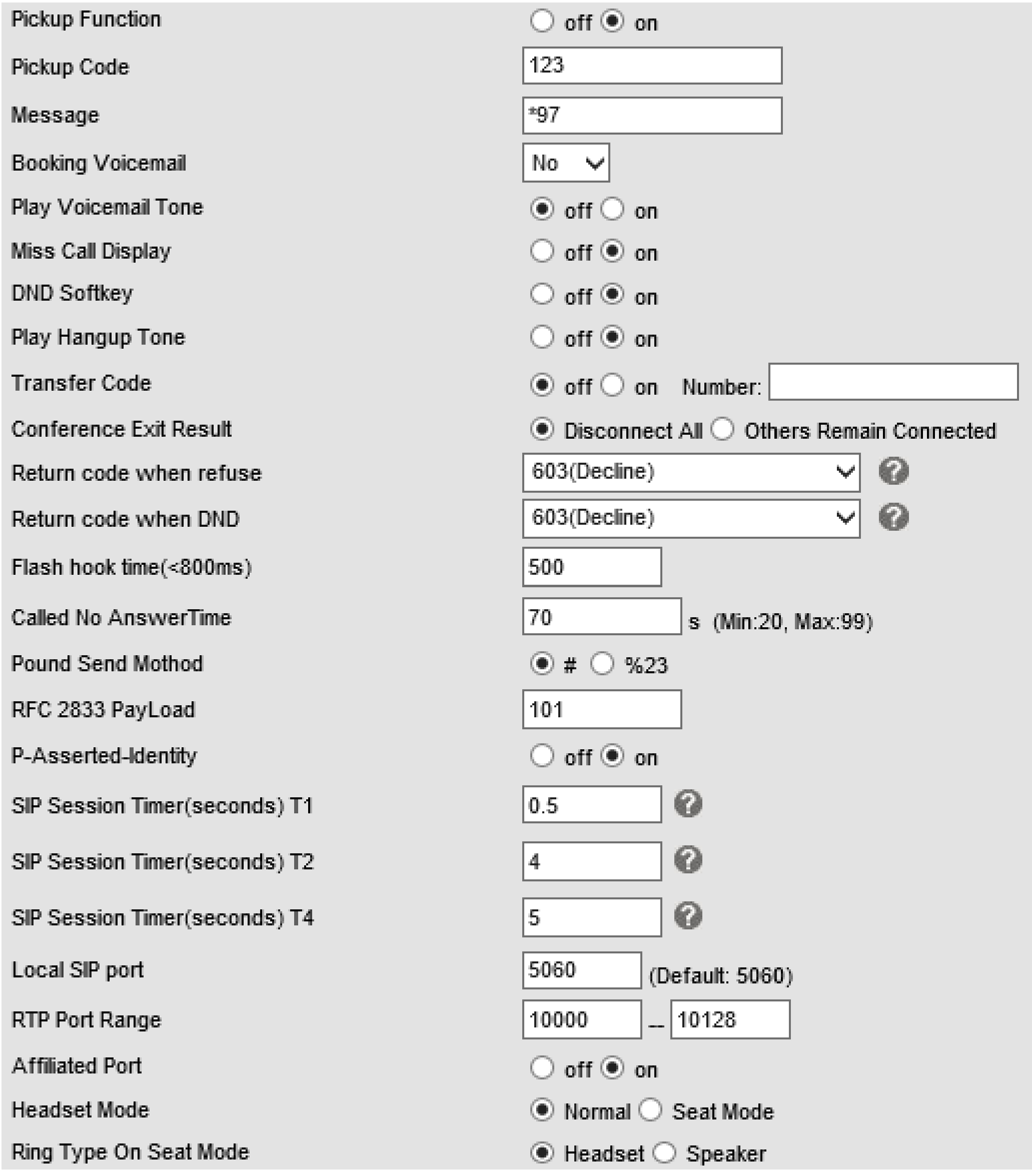
ITEM	DECSRIPTIO
Basic	
Back Light	The backlight of the phone LCD.
Keyboard Lock	Enable/Disable keyboard lock, you can lock: MENU Key, FUNCTION Key., ALL Keys, LOCK all keys but auto Answer.
Hot Line function	When you pick up the handset, it will dial out with the hot number.
Hot Number	Input the number what you want to.
Auto Answer	Auto-answer the coming call, it also can filter a contact group.
Auto Answer Mode	Auto-answer the coming call, it also can filter a device to answer.
Call Waiting	When there's coming a call or the phone is talking, the second call will be in the queue.
Call Waiting Tone	Select the frequency with the tone when call waiting.
DTMF	The DTMF transmitted mode, include RFC2833,Inband,SIP Info, Auto
Fuzzy Search	Fuzzy search someone with the phone book in the idle.
Phone Book Search	Enable/Disable the phone book search feature with accurate or T9 mode.
Call List Save	You can choose to save the call list into the phone or not.
Network Packet Mirroring	When select on, then you can capture the phone’s packet use notebook which connect to pc port of the phone

6.1.1 Time Settings



ITEM	DECSRIPTIO
Time Settings	
Set Time Mode	Include SNTP/SIP Server/PSTN/Manual
SNTP Server	You can select in the list or input owner server address.
Update Interval	The update interval with SNTP.
Day Light Saving Time	Enable/disable the DST for the phone
Time Format	You can use 24 hour time format or 12 hour time format
Date Format	You can choose the appropriate time format.
Time Zone-GMT	You can select different time zone for the phone
Manual Setting	Setting time manually.

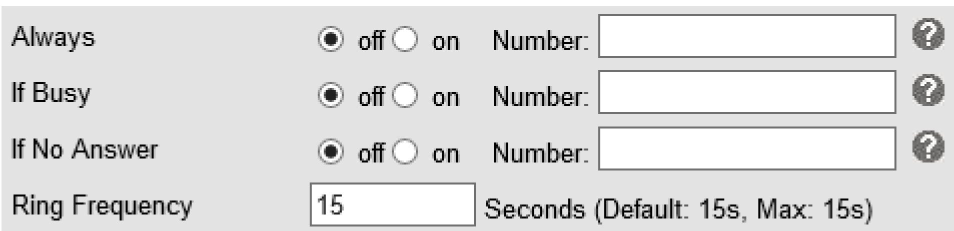
6.1.2 Call



ITEM	DECSRIPTI
Call	
Pickup Function	When you are not in the position, others can help you to answer.
Pickup Code	Fill in server's pickup code.
Message	The code with voice message.
Booking Voice Mail	Open this feature, the phone light(Message) will be bright when it get message.
Play Voice Mail Tone	Open this feature, it will be ringing when it get message.
Miss Call Display	Turn on or off the display with Miss call in the phone LCD.
DND Soft key	Enable/Disable the DND feature.
Play Hang-up Tone	The tone with hang up in busy.
Transfer Code	The code with transfer.

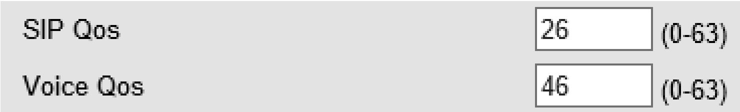
Conference Exit Result	Conference originator hang up the phone, hang up two ways of it.
Return Code When Refuse	Select the code feedback to the server when you reject the call.
Return Code When DND	Select the code feedback to the server when you open DND function.
Flash Hook Time(<800ms)	The time with the flash hook.
Called No Answer Time	When it has coming call and enable this feature, the caller will be request time out in the stipulated time.
Pound Send Method	When you to use the code, such as: #28#123 or %23123, you need to set this feature.
RFC 2833 Play Load	Default is 101, RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
P-Asserted-Identity	Enable/Disable the P-Asserted-Identity feature.
SIP Session Timer T1	The SIP Session Timer setting.
SIP Session Timer T2	The SIP Session Timer setting.
SIP Session Timer T4	The SIP Session Timer setting.
Local SIP Port	The port range setting with SIP, default is 5060.
RTP Port Range	The port range with RTP
Affiliated Port	Enable/Disable the affiliated port feature.
Headset Mode	Select headset mode with normal or seat.
Ring Type On Seat Mode	Select ring type mode with headset or speaker.

6.1.3 VoIP Call Forward



ITEM	DECSRIPTIO
Always	All ways transfer the call to others.
If Busy	If the phone was busy working, the call will be transfer to others.
If No Answer	If the phone was no answer, the call will be transfer to others.
Ring Frequency	The ring frequency with the VOIP Call Forward.

6.1.4 QoS

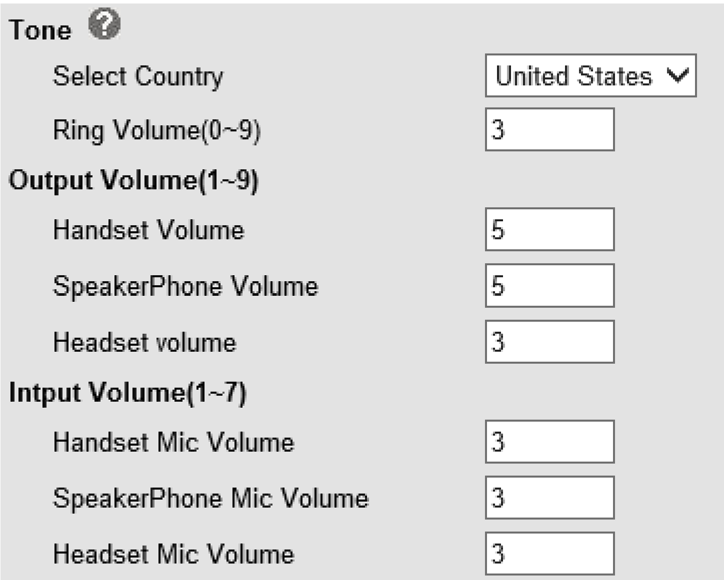


ITEM	DECSRIPTIO
SIP QoS	The range is 0~63,default is 26
Voice QoS	The range is 0~63,default is 46

6.2 Advanced

6.2 .0 Audio

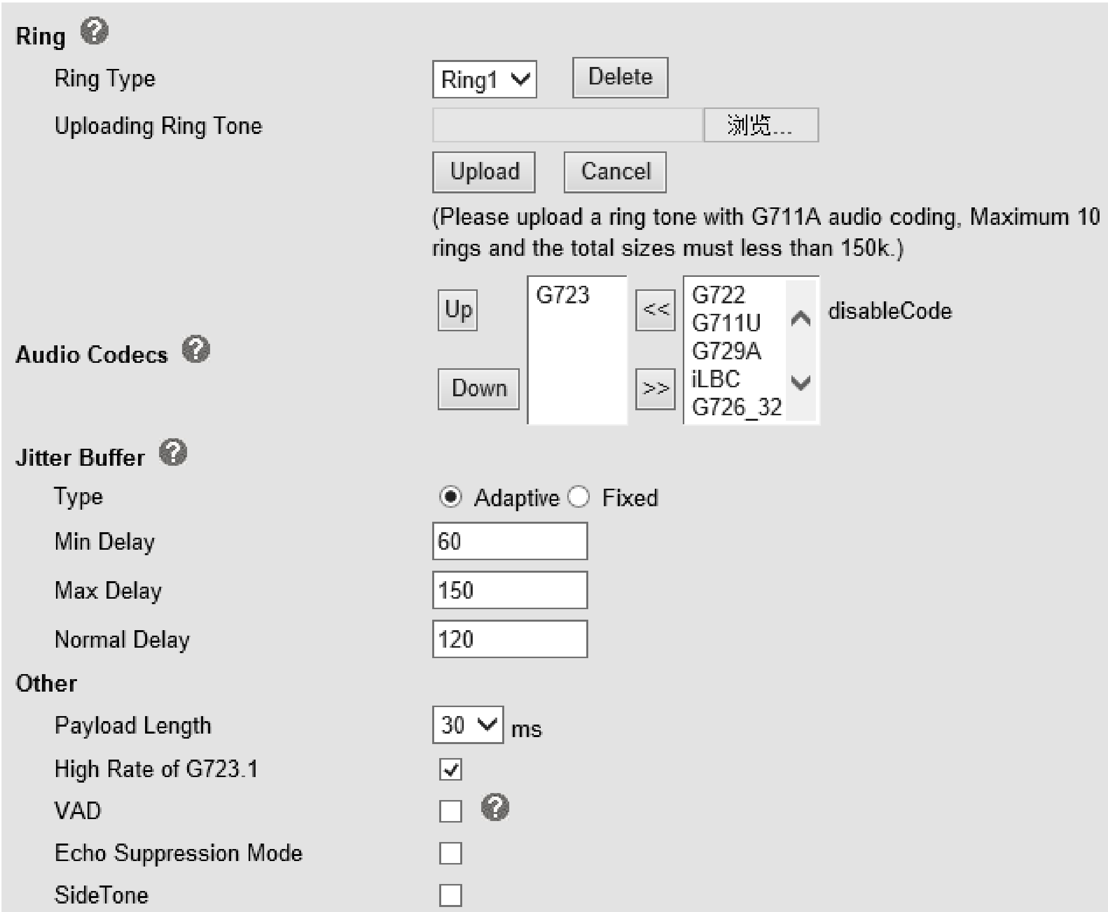
6.2.1 Basic



ITEM	DECSRIPTIO
Basic	
Select Country	Select the country dial tone. Default is United States.
Ring Volume	The ring volume default is Lv3, the range is 0~9.
Handset Volume	The handset volume default is Lv5, the range is 1~9.
Speaker Phone Volume	The speaker volume default is Lv5, the range is 1~9.

Headset Volume	The headset volume default is Lv3, the range is 1~9.
Handset MIC Volume	The handset MIC volume default is Lv3, the range is 1~7.
Speaker Phone MIC Volume	The speaker MIC volume default is Lv3, the range is 1~7
Headset MIC Volume	The headset MIC volume default is Lv3, the range is 1~7

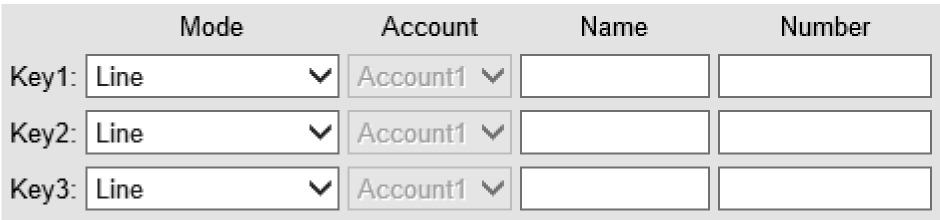
6.2.2 Advanced

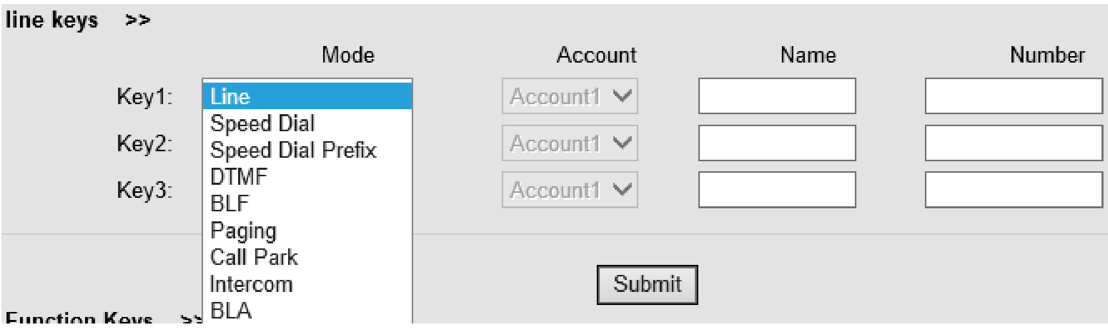


ITEM	DECSRIPTIO
Ring	
Ring Type	Select the ring type. Default is Ring 1.
Uploading Ring Tone	Please upload a ring tone with G711A audio coding, Maximum 10 rings and the total sizes must less than 150k.
Audio Codec	Use the navigation keys to highlight the desired one in the Enabled/Disable Codes list, and press the >>/ << to move to the other list.
Jitter Buffer	

Type	The type of Jitter Buffer is Adaptive or Fixed, default is adaptive.
Min Delay	The min delay range setting , default is 60.
Max Delay	The max delay range setting , default is 150.
Normal Delay	The normal delay range setting , default is 120.
Other	
Play Load Length	The play load length setting, default is 30ms.
High Rate Of G723.1	Enable/Disable High Rate of G723.1 feature.
VAD	Enable/Disable VAD feature.
Echo Suppression Mode	Enable/Disable Echo Suppression Mode feature.
Side Tone	Enable/Disable Side Tone feature.

6.3 Line Keys





ITEMS	DESCRIBES
Line	The default value.
Speed Dial	You can use this key feature to speed up dialing the numbers often used or hard to remember.
Speed Dial Prefix	You can use this key feature to speed up dial a call with a specified prefix number.
DTMF	You can use this key feature to send the specification of arbitrary key sequences via DTMF.
BLF	You can use the BLF feature to monitor a specific user for status changes on the phone.
Paging	You can use multicast paging to quickly and easily forward time sensitive announcements out to people within the multicast group.

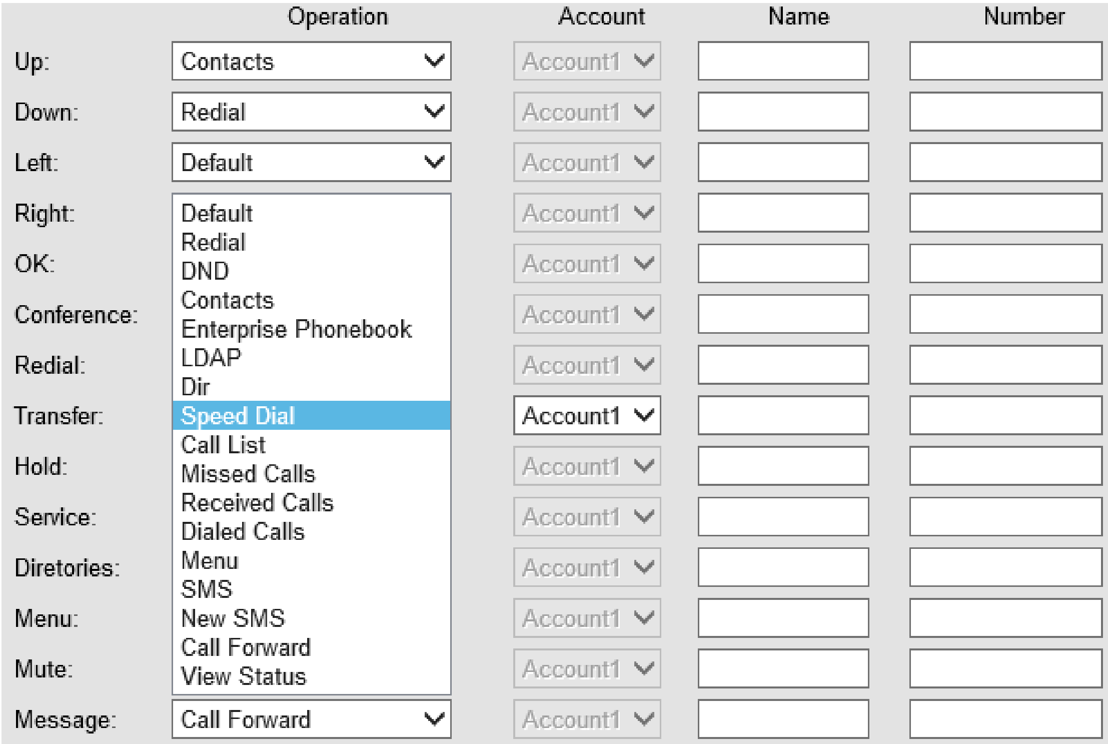
Call Park	You can use call park feature to place a call on hold, and then retrieve the call from another phone in the system (for example, a phone in another office or conference room).
Intercom	You can press the configured intercom key to automatically connect with a remote extension for outgoing intercom calls, and the remote extension will automatically answer the incoming intercom calls
BLA	This feature such as the BLF.

NOTE: ONLY WHEN YOU CHOOSE “SPEED DIAL”, THE RIGHT OF “NAME”,”NUMBER” WILL TAKE EFFECT.

6.4 Function Keys

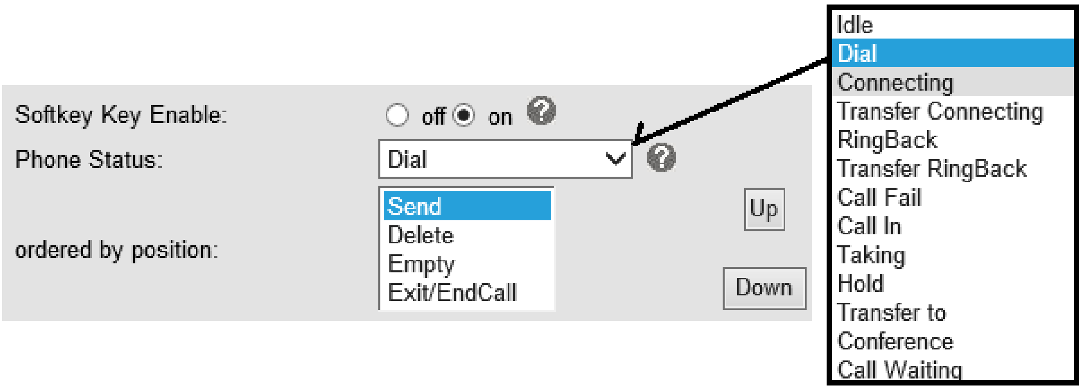
Function Keys: If you do not like the default setting with the function keys feature. You can change to whatever you like.

NOTE: IF THE PHONE WITHOUT THE KEY, YOU CAN IGNORE IT.



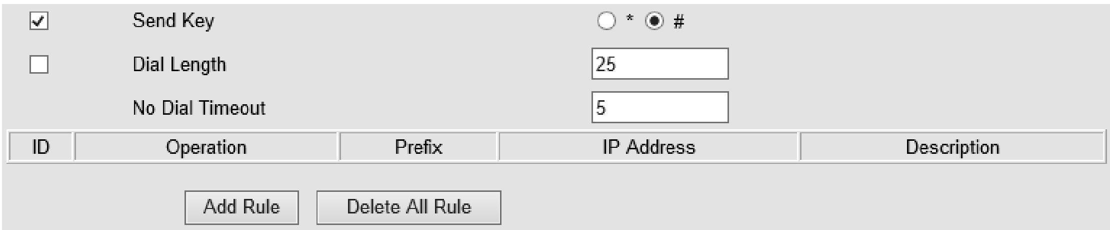
6.5 Soft Key

Soft Keys: Soft key is the key with below display in the LCD. You can change it for your mind to the other features in many all kinds of status. As below example, when you dialing with someone, the LCD display soft key is Send \Del \Empty\End, Empty means nothing in it.

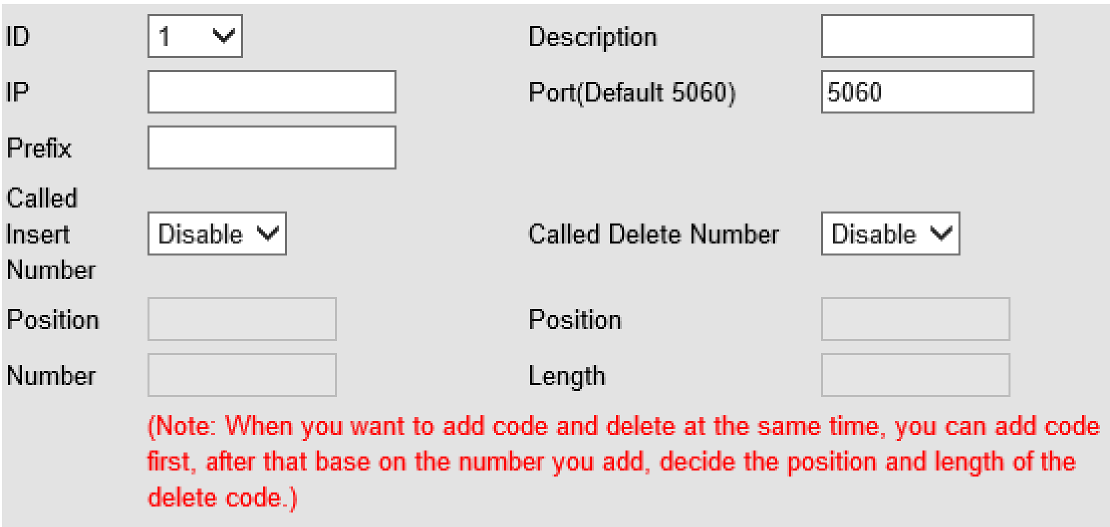


6.6 Dial Plan

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



ITEM	DECSRIPTIO
Send Key	Select the default send key mode you want to use.
Dial Length	Enable this feature will limit the dial length. Default is 25.
No Dial Timeout	Setting the range with no dial timeout, default is 5.
Dial Rule	Select the Add Rule button to add dial rule, pls see as below detail.



ITEM	DECSRIPTIO

ID	Dial Plan ID
IP	The ip of a phone which you want to call
Description	Description with this dial rule.
Port	Setting the Port with this dial rule, default is 5060.
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	Which number you want to insert
Called Delete Number	There have two option, Enable or Disable.

NOTES: If you want to know more detail about Dial Rule, pls find it in the official website to download the specific document. HTTP://www.escene.cn/en.

6.7 IP Strategy

You can use IP Strategy feature to make a list which can be set to only allow the incoming call on the list.

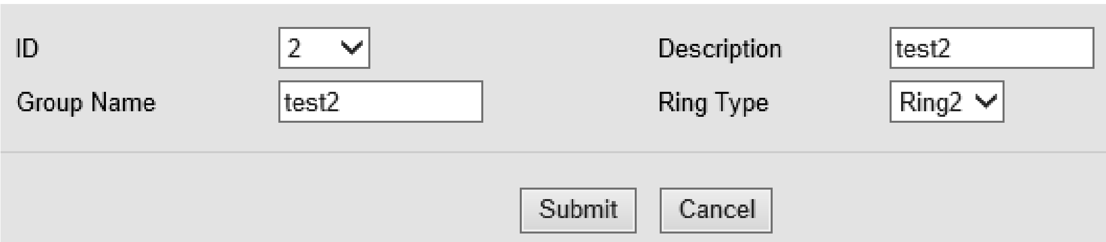
e.g. As following picture you can see it has 192.168.0.248 in the list. When you open this feature. It means you just allow come from this IP address meeting

7 Phone Book

The phone book including Group, Contact, LDAP and Ban list, please review the following for more details:

7.1 Group

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



Click the groupname you can modify or delete the member of the group

ID	Operation	Group Name	Group Member	Description	Ring Type
1		test	0	test	Ring1

Attention: If you Click 'Delete Group' or 'Delete All Group',the member of group can not within a group,please click the group and delete the group.

Add Group | Delete All 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'.

7.2 Contact

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

The phonebook can storage 300 contacts entry

Serial Number	1		
First Name	test	Last Name	test
Mobile Number	1111	Office Number	1111
OtherNumber	1111	Account	Account1
Group1	test	Group2	None

Submit | Cancel

Delete	ID	Operation	Name	Phone	Group
	1		test test	Number1:1111 Number2:1111 Number3:1111	test

Attention:If you want to download or upload the contact,please go to the "Phone Maintenance" page

Add Contact | Delete All Contact

If you want to add a Contact, 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'.

You can edit or move this contact to Ban List after you select .

You can download and save this contact to PC after you select .

7.3 LDAP

NOTES: If you want to know more detail about LDAP, pls find it in the office website to download the specific document. HTTP://www.escene.cn/en. As below figure is

an example.

e.g.

LDAP Name Filter:(sn=%s)

LDAP Number Filter:(telephoneNumber=%s)

Server Address:192.168.0.65

BASE:DC=ldap,DC=escene,DC=com

User Name：bb@ldap.escene.com

Pass Word：escene_2012

LDAP Name Attributes 1:sn

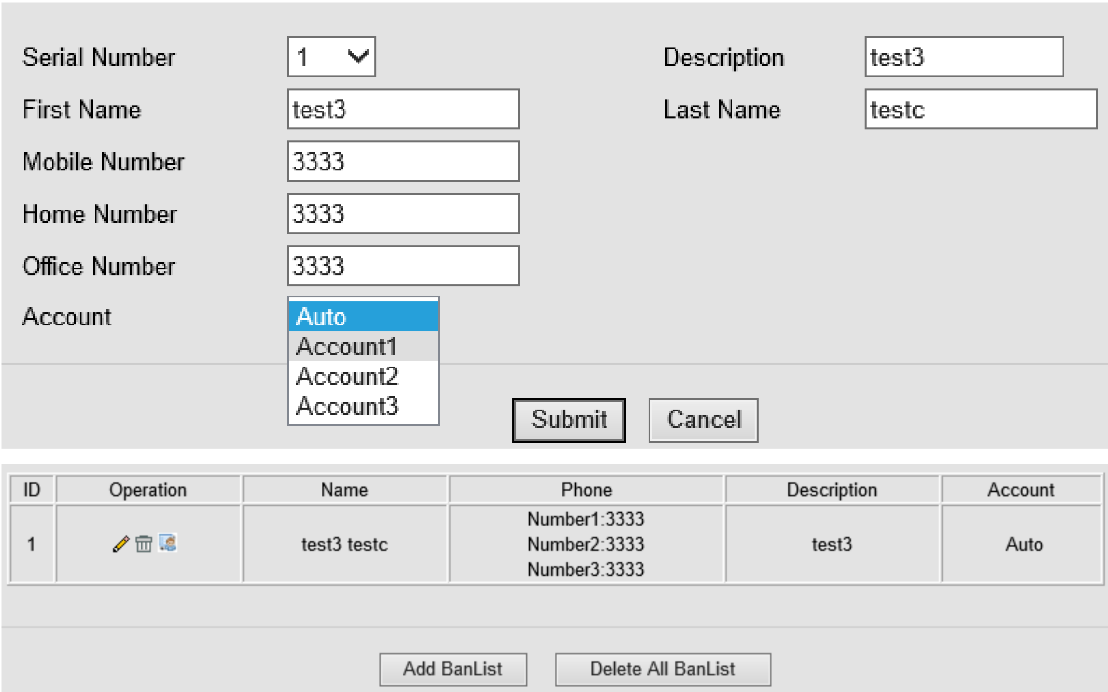
LDAP Name Attributes 2:cn

LDAP Number Attributes 1:telephoneNumber

LDAP	○ on ◉ off
LDAP Name Filter	(sn=%s)
LDAP Number Filter	(telephoneNumber=%:
Server Address	192.168.0.65
Cwmp Port	389
Base	DC=ldap,DC=escene,
Username	bb@ldap.escene.com
Password	escene_2012
Max. Hits(1~32000)	50
LDAP Name Attributes 1	sn
LDAP Name Attributes 2	cn
LDAP Name Attributes 3	
LDAP Number Attributes 1	telephoneNumber
LDAP Number Attributes 2	
LDAP Number Attributes 3	
Protocol	○ Version2 ◉ Version3
Search Delay(ms)(0~2000)	0
LDAP Lookup For Incoming Call	◉ on ○ off
LDAP Lookup For PreDial/Dial	◉ on ○ off

7.4 Ban List

You can add, edit and delete contact in a Ban List on this web page .



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

You can edit an existed Ban List by click .

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

You can edit or move this contact to Contact after you select .

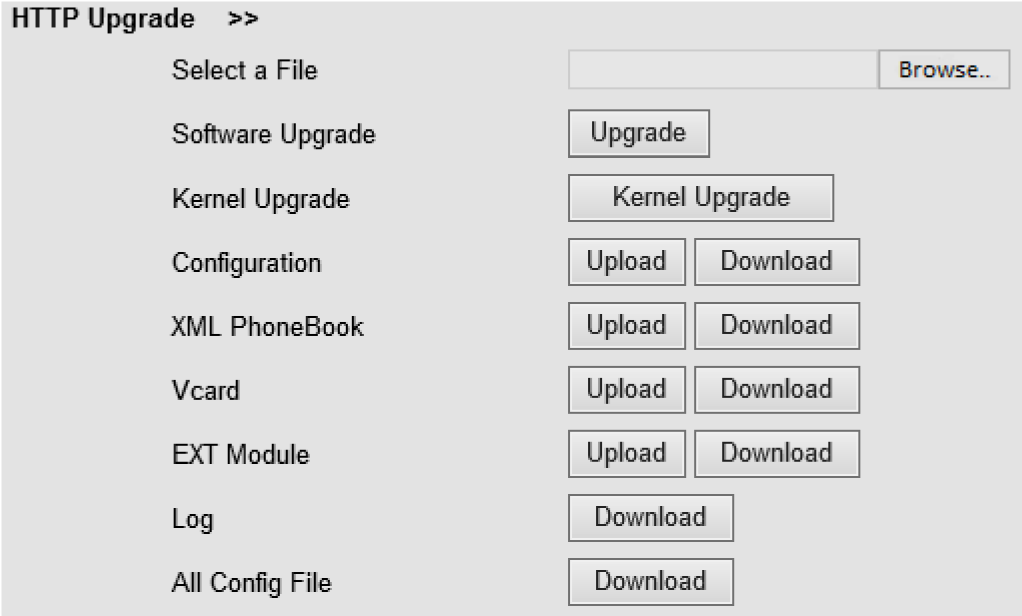
8 Phone Maintenance

8.1 Basic

NOTES: Don't cut off the electricity or network cable when doing upgrade in the below ways!

8.1.1 HTTP Upgrade

You can upgrade the software, kernel and configuration etc. files by HTTP.

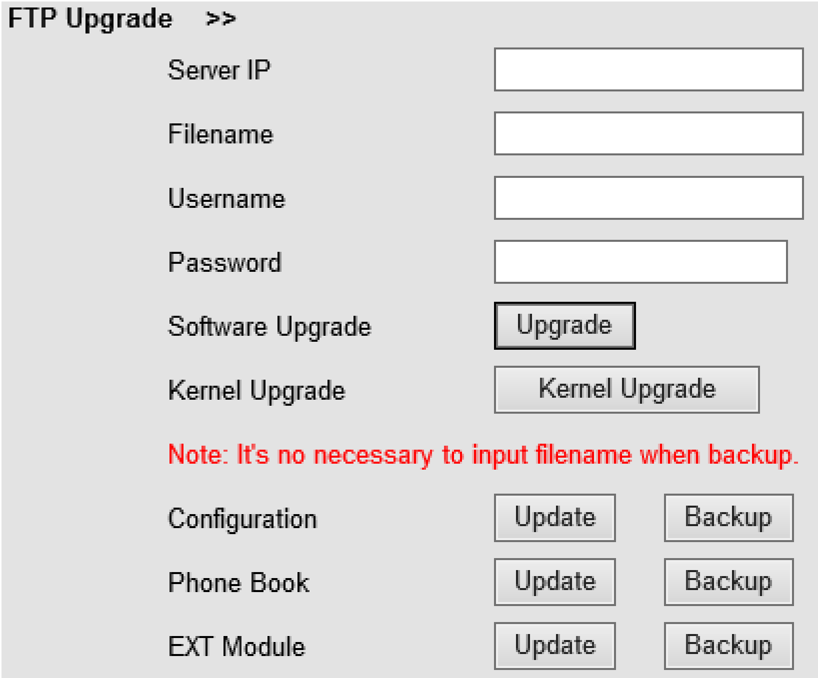


When using HTTP upgrade, you can set several parameters as follow:

HTTP Upgrade	
Select a File	Browse the software/kernel/configuration file which you need to upgrade from HTTP
Software Upgrade	Used for upgrading the software of the phone
Kernel Upgrade	Used for upgrading the kernel of the phone
Configuration	You can used upload/download to upload/download the configure file of the phone
XML Phone Book	Used for uploading/downloading the XML phonebook of the phone
Vcard	Downloading all contacts in the Vcard mode, but upload only support one by one.
EXT Module	Used for updating/backup the expansion of the phone **[NOTES：The mode doesn't support this feature]**
Log	Used for the administrator to find out or making sure the problem with this equipment.
All Config File	All Config File includes: Configuration, Extern, Log, XML Phone book, Enterprise Phone Book.

8.1.2 FTP Upgrade

You can upgrade the software, kernel and configure files by FTP.



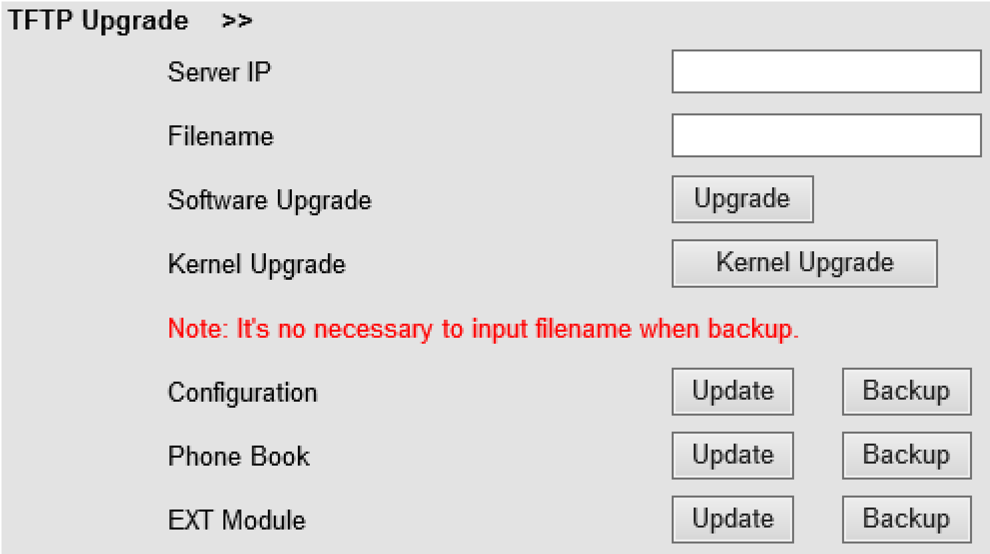
When using FTP upgrade, you can set several parameters as follow:

FTP Upgrade	
Server IP	The IP address of the FTP server
Filename	Downloading from FTP server
Username	Providing by FTP server
Password	Providing by FTP server
Software Upgrade	Used for upgrading the software of the phone
Kernel Upgrade	Used for upgrading the kernel of the phone
Configuration	Used for updating/backup to update/backup the configure file of the phone
Phone Book	Used for updating/backup to update/backup the phonebook of the phone
EXT Module	Used for updating/backup the expansion of the phone **[NOTES：The mode doesn't support this feature]**

NOTES: It's not necessary to input filename when doing backup Configuration, Phone Book, EXT Module.

8.1.3 TFTP Upgrade

You can upgrade the software, kernel and configure files by TFTP.



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

TFTP Upgrade	
Server IP	The IP address of the TFTP server
Filename	Downloading from FTP server
Software Upgrade	Used for upgrading the software of the phone
Kernel Upgrade	Used for upgrading the kernel of the phone
Configuration	Used for updating/backup the configure file of the phone
Phone Book	Used for updating/backup the phonebook of the phone
EXT Module	Used for updating/backup the expansion of the phone **[NOTES：The mode doesn't support this feature]**

NOTES: It's not necessary to input filename when doing backup Configuration, Phone Book, EXT Module.

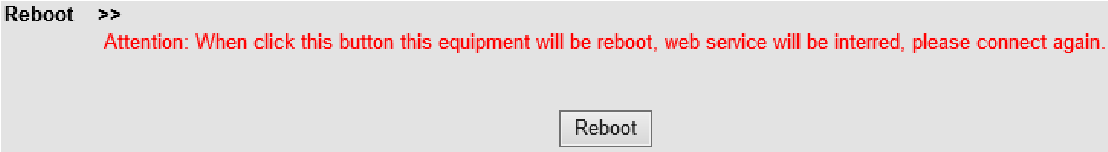
8.1.4 Default Setting

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.

8.1.5 Reboot

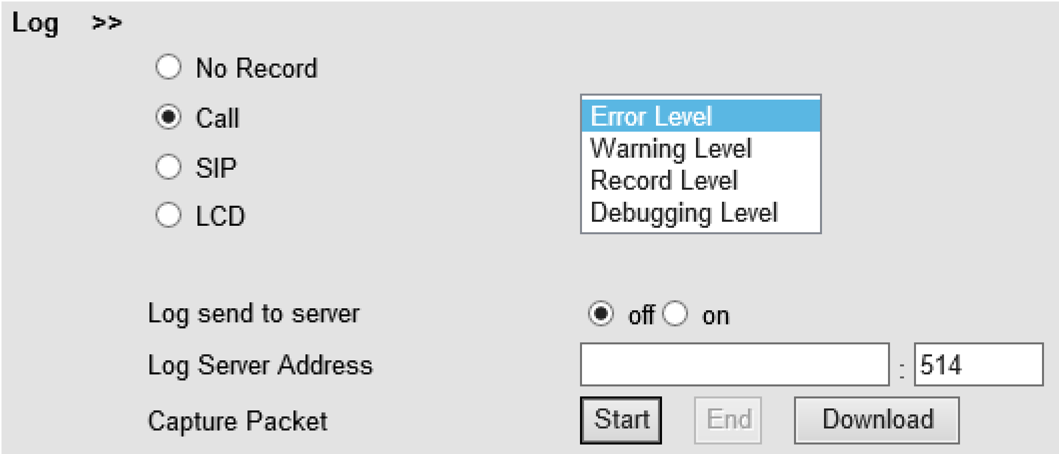
You can use reboot option to reboot the phone.



8.2 Advanced

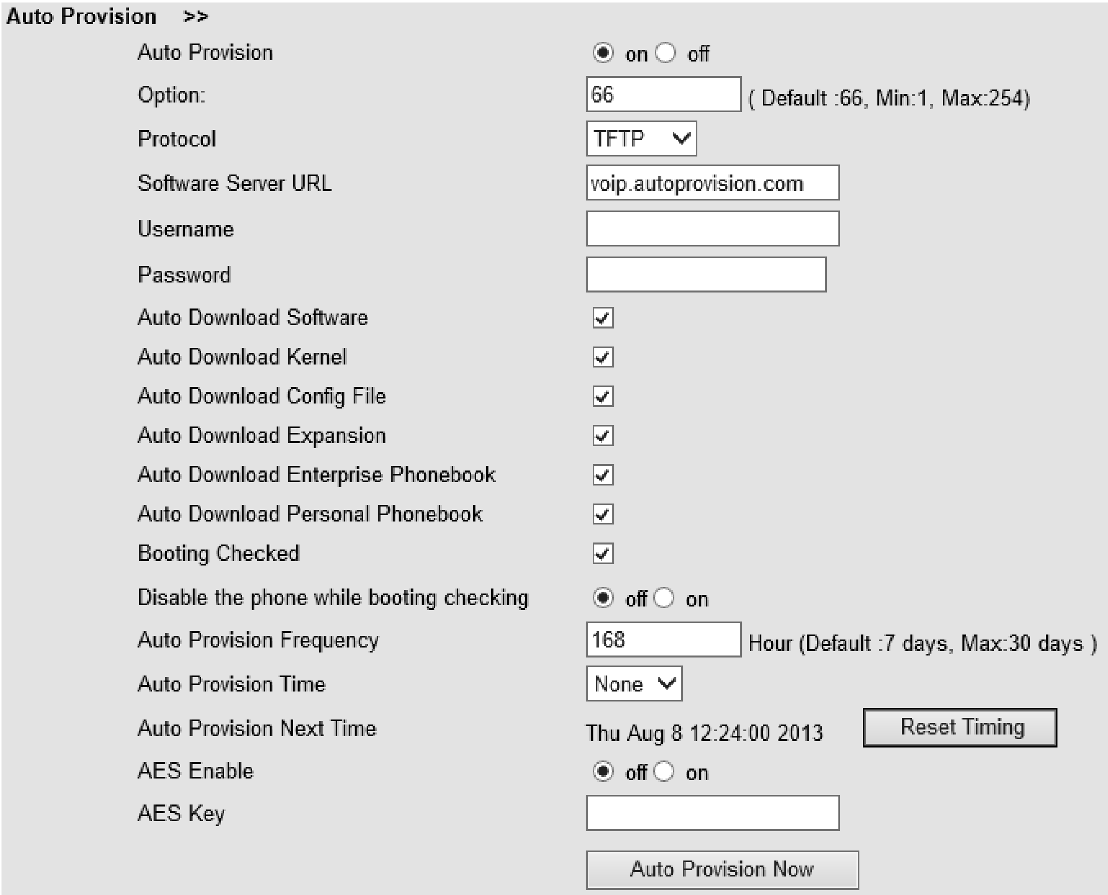
8.2.1 Log

This feature is use for the administrator to managing the equipment, like debugging, SIP etc,. If you need to catch a debugging Level, you need to setup on this interface.



8.2.2 Auto Provision

When you open this auto provision feature, the phone will do auto provision after it detect a different software or kernel (Higher or Lower) which are putted on the TFTP,HTTP,HTTPS,FTP, server. For the detailed information about auto provision, you can find it in the official website: HTTP://www.escene.cn/en



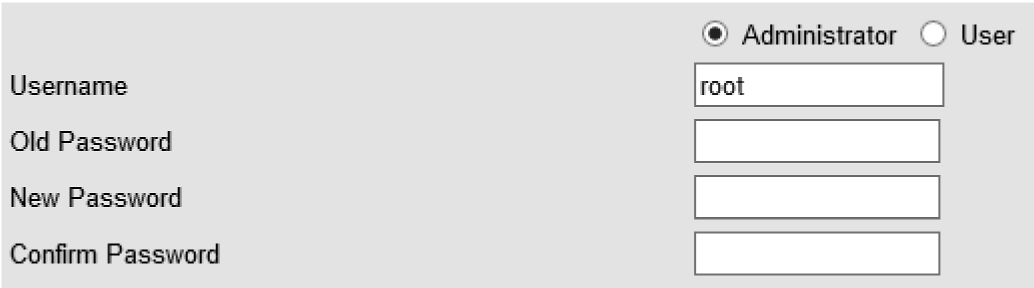
When using auto provision, you can set several parameters as follow:

Auto Provision	
Auto Provision	You can enable/disable auto provision by select on/off
Protocol	Used for auto provision, it includes TFTP/HTTP/FTP
Software Server URL	The server address of the auto provision
Username	Providing by provision server
Password	Providing by provision server
Auto Download Software	Used for auto download software from server
Auto Download Kernel	Used for auto download kernel from server
Auto Download Config File	Used for auto download config file from server
Auto Download Expansion	NOTES: The model doesn't support this feature.
Auto Download Enterprise Phonebook	Used for auto download Enterprise Phonebook from server
Auto Download Personal Phonebook	Used for auto download personal phonebook from server
Booting Checked	Used for checking the auto provision when phone booting
Disable the phone while booting checking	Enable/Disable the booting checking feature.
Auto Provision Frequency	Used for setting the time interval for auto provision
Auto Provision Time	Used for the specific time for auto provision
Auto Provision Next Time	Reset the Auto Provision Next Upgrading time.

AES Enable	You can enable/disable AES encrypt for auto provision
AES Key	The key of the AES
Auto Provision Now	Used for doing auto provision immediately

9 Password

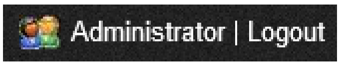
Here you can setting the administrator or user WEB password management. Select your type. If you login as an administrator, you can modify both the user's and admin's passwords.



10 WEB Other Settings or Information - Appendix

10.1 WEB User

In the upper right corner of the website page, you can select the user or logout.



10.2 Multi-Language

In the upper right corner of the website page, you can select the language in the below list.

10.3 Note Tips

In the right middle of the website page, there is a Note tips in every function page. Hope it can help you to know something about that.