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# CC800v2 IP Phone User Manual



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Escene Communication

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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	2 Line keys, Auto-ANS , Hold, RD & Mute, these five keys also support	
Keys	programmable function.	
	Hands-free,Volume adjust, VOL,	
VoIP	SIP 2.0	
Protocol		

Network	HTTP, BOOTP, FTP, TFTP, IEEE 802.1Q, *IEEE 802.1X	
Protocol		
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	Quickly Register\Down	
APP	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	Power adapter:AC100~240V input and DC 5V/1A output	
Supply	PoE(IEEE 802.af),USB	
Specification	Storage Temperature: 0°C-60°C	
	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 0	Headset button: Toggles the headset on or off.	
		Red means the feature is enabled.	
8	VOL- VOL+ VOL	VOL±: Controls the volume and other settings VOL: Change the voice model with	
	Û Û Î Î Î Î Î Î Î Î Î Î Î Î Î Î Î Î Î Î		
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) LAN: Disconnect into network.	
		2) Peer-to-Peer : Only Peer-to-Peer call.	
		3) Retwork 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**

lf you want	Then
to	

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:
	- <b>Type</b> : 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	Pick up the handset	1) You can hear the dial tone;
5		2) The first line light is e_;
the handset		3) Enter a number;
		4) Press ' <b>#'</b> button(default),
		-or press <b>Send</b> ;
		-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 <b>MENU</b> or <b>OK</b> button > "Call history", you can select	
	"Missed calls", "Received calls" and "Dialed numbers",	
	- or press Navigation button (in Standby interface) > select	
	"Missed calls" ( <b>down</b> ), "Received calls" ( <b>left</b> ) and "Dialed	
	numbers" ( <b>right</b> ) );	
	2) Then press <b>Enter</b> button follow the tips and press <b>Dial</b> .	
Place a call while	1) Press Hold button or Resum;	
Another call is active	2) Select another account and enter a number;	
	3) Press '#' button (default) ;	
	-or press <b>Send</b> 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	1) Your phone ring;	Pick up the handset or press the
handset	2) Line button of the ringing line is Red	flashing <b>— Line</b> button,
Switch from a	1) Another Line button is F	Red end flashing, Light strip is
connected Call	Red end flashing;	
to answer a	2) Press the flashing	ine button to answer (at this time, the
ringing	original call will be hold.)	
call		
Auto-answer	1) Press MENU or OK buttor	n > "Function setting" > "Auto answer"
	or press AUTO ANS;	
	2) Select "Enable";	
	3) Your phone answers inco	oming calls automatically after a few
	rings.	

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# Ending a Call

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

If you want to	Then
Hang up while using the	Return the handset to its cradle,
Handset	-or press END
Hang up while using the	Press <b>Speaker</b> button ,
Speakerphone	-or press Line button for the appropriate line,
	-or press END
Hang up while using the	Press Handset button, (Do not keep the headset mode),
Headset	-or press END (keep the headset mode)
Hang up one call, but	Press END,
preserve another call on	-or refer to the above three methods
the other line	

## 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	Press another Line button for the appropriate line
another line	
Resume a call on	Press Line button,
current line	
Release a call on	Select the line want to release hold, press the line, so
different line	recovery;

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#### 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	1) Press XFER soft key under the LCD;
recipient before	2) Enter number;
transferring a call	3) press " <b>#</b> " (default),
(consult transfer)	<ul> <li>-or press Send then transfer the call,</li> <li>-or wait five seconds(default)then transfer the call</li> </ul>
Transferred to idle	1) Press XFER soft key under the LCD;;
lines or other	2) Press Blind;
numbers without	3) Enter number;
talking to the	4) Press " <b>#</b> " (default)
	-or press <b>Send</b> , then transfer the call;
transfer recipient	-or wait five seconds(default)then transfer the call
(Blind transfer)	
Blind transfer to the	1) Press XFER soft key under the LCD;;
held line	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 <b>Mute</b> 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	Press MENU or OK button > "Function setting" > "DND" >
single line	(select line) "Enable", e.g.
Disable DND	Line DND enabled, press twice <b>DND</b> ,
	-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	1) When the transfer recipient answer the call, press CONF
recipient into a	under the LCD;
conference in a	2) Then the held one, transfer recipient and you will be into a
transferring	conference, and the LCD will display conferenc 0:0:10
	status.
Invite the third party	1) Press <b>CONF</b> button under the LCD;
into a conference in	2) Enter the third party number;
a active call	3) press " <b>#"</b> (default),
	-or press Send then transfer the call,

establish a	1) when one phone line is holding on and the other line is
conference with held	busy;
line	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.
	1) Press MENU or OK button > "Function setting" >
No answer transfer	"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	1) Press <b>MENU</b> or <b>OK</b> button > "Function setting" > "Hot line";
Dials on your	2) Press Enter and to select Enable
phone	3) Number: Need to speed dial numbers
	4) Press OK to submit
	<ul> <li>1) Press MENU or OK button &gt; "Function setting" &gt; "Delay line";</li> <li>2) Press Enter and to select Enable</li> <li>3) Number: delay dial the number after 5 second</li> <li>4) Press OK to submit</li> </ul>

### 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 <b>MENU</b> 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 <b>opt.</b> 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 <b>Save</b> soft key to complete
Add group	1) Press <b>MENU</b> 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 <b>MENU</b> button > "Phone book",
	2) Press Opt. [Tips: opt. means modify.]
	3) Select the "Modify group" then press <b>OK</b> button
	4) Select the group you want to modify, press the OK button
	save the change
Delete group	1) Press <b>MENU</b> button > "Phone book",
	2) Press Opt. [Tips: opt. means modify.]
	3) Select the "Delete group"
	4) Select a group you want to delete, press <b>OK</b> button

View/Edit Contacts	1) Press <b>MENU</b> 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 <b>MENU</b> 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 <b>OK</b> or <b>Del.</b>	
	Pay attention: before you use LDAP function, you need to	
	configure LDAP rule in the web configure page.	
Call from phone	1) Press <b>MENU</b> button > "Phone book",	
book	2) Select "View ALL",	
	-or select a contact who are belong to different group;	
	3) Select a contact, then press <b>Dial</b> ,	
	(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 <b>Dial</b> )	
Modify the relative	1) Open your web browser, enter the "web" interface. (For	
account of a	details, you can refer to 7. Web Settings.)	
contact	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,	

# **Using Call Logs**

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	1) If you want to delete a call record, you have to select this	
logs	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	

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

### **Black List**

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

If you want to	Then	
View your phone	1) Press MENU button > "Phone book" > "Personal Phone	
book	Book", "View all", or "Groups member"	
	2) Use the navigation keys to view the members information.	
Put into the Black	Use the navigation keys to select "Put into the Black List", Press	
List	the soft key " <b>Opt.</b> or <b>5</b> " to submit.	
Erase your Black	1) If you want to delete a black list member. Press MENU	
List members	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 <b>MENU</b> button > "Function Setting" > "Fuzzy Search"	
	2) Press ENTER and make it Enable.	

### Time & Date

If you want to	Then
Time & Date	1) Press <b>MENU</b> or <b>OK</b> 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	<ul> <li>1) Choose "Menu" &gt; "Language";</li> <li>2) Scroll through the list of available languages.</li> <li>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.</li> </ul>

### **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	1) Choose "Menu" > "System setting" > "Advanced setting";
account	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
	- <b>Refresh time</b> : 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;
	* <b>Note</b> : 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 <b>Save</b> 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 ' <b>OK</b> ', then " <b>Reboot"</b> the phone.

### **Customizing Rings and Volume**

If you want to	Then	
Change the ring	1) Choose "Menu" > "System setting" > "Phone setting" > "Ring	
tone	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	1) Choose "Menu" > "System setting" > "Phone setting" > "Volume	
volume level	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,

SENE			Please Soloci Language
			English (English)
Current lo	caties: Phone Status	ALC: ADDRESS AND ADDRESS ADDRESS	AUG 1993 (1997)
Phone Status	System Run Time Register status ©	1 Days17 Hours26 Minutes8 Seconds	Register status: # shows the Register Status.
Network	Account	6000 (Registered)	
	Account2	None	Network Status:
SIP Account	Network Status		It shows the information of LAN port
	LAN Port type	DHOP	and PC port
Programmable Keys	MAC	00.26 Bb-04 5d 68	and the second
Phone Settings	LAN IP Address	192 968.8 145	System Info:
1 more containing a	Subnet Mask	266.266.266.8	it shows the renzion of iterminent
Phonebook	Gateway	192.168.8.1	
	Primary DNS	210.21.4.130	
Phase Maintenance	Secondary DNS		
	VPN IP Address		
Security	Routel IP Address		
	Router Subnet Mask		
	Device type	As bridge	
	Router DHCP	off	
	System lato @		
	Phone Model		
	Software Version	V3.7.2.1-8857	
	Hardware version	V2.x.x	
	Hardware ID	1	
	Name of Manadam	Vie 1	

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

Basic	>>	
	• DHCP 🕜	
	Hostname(Option 12)	
	Manufacturer(Option 60)	
	O Static IP 🔞	
	IP Address	192.168.0.200
	Netmask	255.255.255.0
	Gateway	192.168.0.1
	🔿 РРРоЕ 🕜	
	Username	
	Password	
	MTU	1500 Default: 1500
	DNS Settings	
	DNS	Automatic O Manual DNS
	Primary DNS	192.168.0.1
	Secondary DNS	0.0.0.0

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

#### Advanced

Port Management Settings		
HTTP Port	80	
Telnet Port	23	
Socket5 Proxy Server		
Socket5 Proxy Server	● off ○ on	
Server IP		*
Port	1080 *	
Anonymous Login	✓	
Username		]
Password		
Paging Setting		
Paging 1	● off ○ on	
Group IP		Port: 10000
Paging 2	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on	
Group IP		Port: 10000
Paging 3	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on	
Group IP		Port: 10000
Paging 4	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on	
Group IP		Port: 10000
Paging 5	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on	
Group IP		Port: 10000

Please Note: Changing the default HTTP Port (80) will require using the new port number to access the IP phone web interface. Please note that changes require a reboot. Use the following format when not using the default HTTP (http://ip address:portnumner).

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.	

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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 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.

<ul> <li>Bridge Ø</li> <li>Router Ø</li> </ul>	
IP Address	*
Netmask	*
DHCP Server	$\odot$ off $\bigcirc$ on
Start IP	
End IP	

#### 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** 

Enable VPN	
VPN Type	L2TP
L2TP	SSL_VPN
VPN Server Addr	
VPN User Name	
VPN Password	

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**

Enable Vlan:			
LAN Port		PC Port	
VID:	0 (0~4094)	VID:	0 (0~4094)
Priority:	0 🗸 (0~7)	Priority:	0 🗸 (0~7)
	where any inclusion and any and		fallow

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

Enable	
Account Mode	VOIP V
Amount Of Line Account Used	1 (Default: 2)
Display Name	0
Username	5207 * 🕜
Authenticate Name	5207
Password	••••
Label	0
SIP Server	192.168.0.7
Secondary server	0
OutboundProxy Server	0
Secondary OutboundProxy Server	0
Polling Interval Time Of Registration	32 s Default Value: 32s, Range: 20s~~60s
NAT Traversal	Disable V
STUN Server	0
BLA	● off ○ on
BLA Number	
Subscribe Period	1800 Default: 1800s, Min: 120s 🕜
Register Expire Time	3600 Default: 3600s, Min: 40s 🕜
Auto Answer	● off ○ on
SIP Transport	$\odot$ UDP $\bigcirc$ TCP $\bigcirc$ TLS 🕜
Ring Type	None 🗸 🕜

#### 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, PIs 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

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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	When the main out bound server can't work, it also can use this
Proxy Server	secondary server.
Poling Interval Time	Poling Interval Time Of Registration, default is 32 s.
Of	
Registration	
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

Do Not Disturb	● off ○ on
Anonymous Call	● off ○ on 🙆
Anonymous Call Rejection	● off ○ on 🕜
Use Session Timer	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on
Session Timer	300 (min:150s)
Call Method	$\odot$ SIP $\bigcirc$ TEL
DNS-SRV	$\odot$ off $\bigcirc$ on
Allow-events	$\odot$ off $\bigcirc$ on
Registered NAT	$\bigcirc$ off $\odot$ on
UDP Keep-alive Message	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on
UDP Keep-alive Interval	30 (15-60s)

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

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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

SIP Encryption	● off ○ on 🕜
RTP Encryption	● off ○ on 🕜
Encryption Algorithm	RC4 🗸
Encryption Key	

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

BackLight	○ off ○ Always On ④ timer 60 s (Min:1, Max:255) 🚱
Keyboard Lock	Disabled V
Hot Line Function	● off ◯ Delay 5 s (0-30)
Hot Number	0
Auto Answer	$ullet$ off $\bigcirc$ on $\bigcirc$ Turn On But Filter This Group : NONE $\checkmark$
Auto Answer Mode	● Hands Free ○ Handle ○ Headset
Call Waiting	○ off ● on 🚱
Call Waiting Tone	○ off ● Play on currently active device Frequency: 10 s (5-60) 𝚱
DTMF	● RFC 2833 ○ Inband ○ SIP Info ○ Auto 🚱
Fuzzy Search	● off ○ on
Phonebook Search	Accurate Search      T9
Call List Save	○ off ● on
Network Packet Mirroring	Off V

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

Set Time Mode	● SNTP ○ SIP Server ○ PSTN ○ Manual
SNTP Server	sparky.services.adelaide.edu.au 🚱
	● sparky.services.adelaide.edu.au ∨ List
	Sparky.services.adelaide.edu.au Manual
Update	
Interval (seconds)	600 Seconds
Daylight Savings Time Mode	$\bigcirc$ always off $\bigcirc$ always on $ullet$ Auto 🚱
Time Format	● 24 Hour 〇 12 Hour 🕜
Date Format	DD MM WWW 🗸 🚱
Time Zone- GMT	GMT+08:00 Beijing V
Manual	2000 Year 1 Month 1 Days 0
Setting	Hours <sup>0</sup> Minutes <sup>0</sup> Seconds

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

Pickup Function	○ off ● on
Pickup Code	123
Message	*97
Booking Voicemail	No 🗸
Play Voicemail Tone	● off ○ on
Miss Call Display	○ off ● on
DND Softkey	○ off ● on
Play Hangup Tone	$\bigcirc$ off $\odot$ on
Transfer Code	● off ○ on Number:
Conference Exit Result	${\small \bullet}$ Disconnect All $\bigcirc$ Others Remain Connected
Return code when refuse	603(Decline) V
Return code when DND	603(Decline) V
Flash hook time(<800ms)	500
Called No AnswerTime	70 s (Min:20, Max:99)
	70 s (Min:20, Max:99) • # • %23
Called No AnswerTime	o (mm.20, max.00)
Called No AnswerTime Pound Send Mothod	<ul> <li>● # ○ %23</li> </ul>
Called No AnswerTime Pound Send Mothod RFC 2833 PayLoad	● # ○ %23 101
Called No AnswerTime Pound Send Mothod RFC 2833 PayLoad P-Asserted-Identity	● # ○ %23 101 ○ off ● on
Called No AnswerTime Pound Send Mothod RFC 2833 PayLoad P-Asserted-Identity SIP Session Timer(seconds) T1	● # ○ %23 101 ○ off ● on 0.5
Called No AnswerTime Pound Send Mothod RFC 2833 PayLoad P-Asserted-Identity SIP Session Timer(seconds) T1 SIP Session Timer(seconds) T2	<ul> <li># \@ %23</li> <li>101</li> <li>0 off (Initial States)</li> <li>0 off (Initial States)</li></ul>
Called No AnswerTime Pound Send Mothod RFC 2833 PayLoad P-Asserted-Identity SIP Session Timer(seconds) T1 SIP Session Timer(seconds) T2 SIP Session Timer(seconds) T4	<ul> <li># 0 %23</li> <li>101</li> <li>off • on</li> <li>0.5</li> <li>4</li> <li>6</li> <li>7</li> </ul>
Called No AnswerTime Pound Send Mothod RFC 2833 PayLoad P-Asserted-Identity SIP Session Timer(seconds) T1 SIP Session Timer(seconds) T2 SIP Session Timer(seconds) T4 Local SIP port	<ul> <li># %23</li> <li>101</li> <li>off • on</li> <li>0.5</li> <li>6</li> <li>4</li> <li>6</li> <li>5</li> <li>6</li> <li>(Default: 5060)</li> </ul>
Called No AnswerTime Pound Send Mothod RFC 2833 PayLoad P-Asserted-Identity SIP Session Timer(seconds) T1 SIP Session Timer(seconds) T2 SIP Session Timer(seconds) T4 Local SIP port RTP Port Range	<ul> <li># %23</li> <li>101</li> <li>off on</li> <li>0.5</li> <li>2</li> <li>4</li> <li>2</li> <li>5</li> <li>2</li> <li>5060 (Default: 5060)</li> <li>10000 10128</li> </ul>

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.

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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

Always	$\odot$ off $\bigcirc$ on	Number:	0
If Busy	$\odot$ off $\bigcirc$ on	Number:	0
If No Answer	$\odot$ off $\bigcirc$ on	Number:	0
Ring Frequency	15	Seconds (Default: 15s, Max: 15s)	

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

SIP Qos	26	(0-63)
Voice Qos	46	(0-63)

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

Tone 🕜		
Select Country	United States 🗸	
Ring Volume(0~9)	3	
Output Volume(1~9)		
Handset Volume	5	
SpeakerPhone Volume	5	
Headset volume	3	
Intput Volume(1~7)		
Handset Mic Volume	3	
SpeakerPhone Mic Volume	3	
Headset Mic Volume	3	

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	The speaker volume default is Lv5, the range is 1~9.		
Volume			

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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	The speaker MIC volume default is Lv3, the range is 1~7		
Volume			
Headset MIC Volume	The headset MIC volume default is Lv3, the range is 1~7		

#### 6.2.2 Advanced

Ring 🕜			
Ring Type	Ring1 V Delete		
Uploading Ring Tone	浏览		
	Upload Cancel		
	(Please upload a ring tone with G711A audio coding, Maximum 10 rings and the total sizes must less than 150k.)		
Audio Codecs 🕜	$ \begin{array}{ c c c c c } \hline Up & G723 & << & G722 \\ G711U & & G729A \\ \hline Down & & & \\ \end{array} \\ \hline \end{array} \\ \begin{array}{ c c c c c c c c c c c c c c c c c c c$		
Jitter Buffer 🕜			
Туре	● Adaptive ○ Fixed		
Min Delay	60		
Max Delay	150		
Normal Delay	120		
Other			
Payload Length	30 🗸 ms		
High Rate of G723.1			
VAD			
Echo Suppression Mode			
SideTone			

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			

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Туре	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	Enable/Disable Echo Suppression Mode feature.		
Mode			
Side Tone	Enable/Disable Side Tone feature.		

### 6.3 Line Keys

	Mode	Account	Name	Number
Key1:	Line 🗸	Account1 🗸		
Key2:	Line 🗸	Account1 🗸		
Key3:	Line 🗸	Account1 🗸		

line keys	>>
-----------	----

	Mode	Account	Name	Number
Key1:	Line	Account1 🗸		
Key2:	Speed Dial Speed Dial Prefix	Account1 V		
Key3:	DTMF BLF	Account1 🗸		
	Paging Call Park			
	Intercom	Submit		
Function Kove >>	BLA			
ITEMS	DESCRIBES			
Line	The default val	ue.		
Speed Dial	You can use th	is key feature to speed	up dialing the nu	mbers often used
	or hard to reme	ember.		
Speed Dial	You can use t	his key feature to spee	ed up dial a cal	I with a specified
Prefix	prefix number.			
DTMF	You can use the	nis key feature to send	the specificatio	n of arbitrary key
	sequences via	DTMF.		
BLF	You can use	the BLF feature to me	onitor a specific	c user for status
	changes on the	e phone.		
Paging	You can use	multicast paging to c	quickly and eas	sily forward time
	sensitive anno	uncements out to people	e within the mult	ticast 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.

	Operation	Account	Name	Number
Up:	Contacts 🗸	Account1 🗸		
Down:	Redial 🗸	Account1 🗸		
Left:	Default 🗸	Account1 🗸		
Right:	Default	Account1 🗸		
OK:	Redial DND	Account1 🗸		
Conference:	Contacts Enterprise Phonebook	Account1 🗸		
Redial:	LDAP Dir	Account1 🗸		
Transfer:	Speed Dial	Account1 V		
Hold:	Call List Missed Calls	Account1 🗸		
Service:	Received Calls Dialed Calls	Account1 🗸		
Diretories:	Menu SMS	Account1 🗸		
Menu:	New SMS	Account1 🗸		
Mute:	Call Forward View Status	Account1 🗸		
Message:	Call Forward 🗸 🗸	Account1 🗸		

### 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"

✓	Send Key		○*●#	
	Dial Length		25	
	No Dial Timeout		5	
ID	Operation	Prefix	IP Address	Description
	Add Rule	Delete All Rule	]	

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.

ID	1 🗸	Description	
IP		Port(Default 5060)	5060
Prefix			
Called Insert Number	Disable V	Called Delete Number	Disable V
Position		Position	
Number		Length	
	(Note: When you want to add co first, after that base on the numb delete code.)		

ITEM	DECSRIPTIO
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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

IP Stra	tegy ◉ off ○ on			
ID	Operation	IP Address	Description	Account

## 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.

ID	2 🗸		Description	test2
Group Name	test2		Ring Type	Ring2 🗸
		Submit	Cancel	

		Click the groupname you	can modify or delete the me	mber of the group	
ID	Operation	Group Name	Group Member	Description	Ring Type
1	N 📅	test	0	test	Ring1
		k Delete Group of Delete A	ll Group',the member of group ca	n not within a group,pieas	se click the group
ar	nd delete the 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  $click^{m}$ , 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 Num	nber	1 🗸				
First Name		test	test		test	
Mobile Number		1111		Office Number	1111	
OtherNumber		1111		Account	Account1 V	']
Group1		test 🗸		Group2	None 🗸	
			Submit	Cancel		
Delete	ID	Operation	Name	Phone	0	Group
	1	/ 📅 🖪 🔶	test test	Number1:1111 Number2:1111		test

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

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

Add Contact

You can edit an existed Contact by click 4.

You can delete an existed Contact by click  $\overline{m}$ , if you want to delete all Contacts, you just ought to click 'Delete All Contact'.

Delete All Contact

Number3:1111

You can edit or move this contact to Ban List after you select <a>!</a>. You can download and save this contact to PC after you select <a>!</a>.

### 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=Idap,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=Idap,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
Search Delay(ms)(0~2000)	0
LDAP Lookup For Incoming Call	$\odot$ on $\bigcirc$ 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 .

Serial Number	1 🗸	Desci	iption	test3
First Name	test3	Last	Name	testc
Mobile Number	3333			
Home Number	3333			
Office Number	3333			
Account	Auto Account1 Account2 Account3	Submit Canc	el	
ID Operation	Name	Phone	Descripti	on Account
1 2 🖻 🕼	test3 testc	Number1:3333 Number2:3333 Number3:3333	test3	Auto
Add BanList Delete All BanList				

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  $\overline{m}$ , 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 III.

### **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.

HTTP Upgrade >>		
Select a File		Browse
Software Upgrade	Upgrade	
Kernel Upgrade	Kernel Upgrade	
Configuration	Upload Download	
XML PhoneBook	Upload Download	
Vcard	Upload Download	
EXT Module	Upload Download	
Log	Download	
All Config File	Download	

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	Used for upgrading the software of the phone			
Upgrade				
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	Used for uploading/downloading the XML phonebook of the phone			
Book				
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.

FTP Upgrade >>		
Server IP		
Filename		
Username		
Password		
Software Upgrade	Upgrade	
Kernel Upgrade	Kernel Upg	rade
Note: It's no necessary to in	put filename wh	nen backup.
Configuration	Update	Backup
Phone Book	Update	Backup
EXT Module	Update	Backup

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.

TFTP Upgrade >>				
Server IP				
Filename				
Software Upgrade	Upgrade			
Kernel Upgrade	Kernel Upg	grade		
Note: It's no necessary to input filename wh	en backup.			
Configuration	Update	Backup		
Phone Book	Update	Backup		
EXT Module	Update	Backup		

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.

Default Setting >>

When click this button this equipment will restore to the default status

Pay Attention: It will take effect on next reboot.

Reset to Factory Setting

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.

Reboot	>> Attention: When click this button this equipment will be reboot, web service will be interred, please connect again.
	Reboot

### 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.

Log	>>		
		○ No Record	
		<ul> <li>Call</li> </ul>	Error Level
		○ SIP	Warning Level Record Level
		○ LCD	Debugging Level
		Log send to server	● off ○ on
		Log Server Address	: 514
		Capture Packet	Start End Download

### 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

Auto Provision >>	
Auto Provision	$\odot$ on $\bigcirc$ off
Option:	66 ( Default :66, Min:1, Max:254)
Protocol	TFTP 🗸
Software Server URL	voip.autoprovision.com
Username	
Password	
Auto Download Software	✓
Auto Download Kernel	$\checkmark$
Auto Download Config File	<b>v</b>
Auto Download Expansion	✓
Auto Download Enterprise Phonebook	<b>v</b>
Auto Download Personal Phonebook	<b>v</b>
Booting Checked	$\checkmark$
Disable the phone while booting checking	● off ○ on
Auto Provision Frequency	168 Hour (Default :7 days, Max:30 days )
Auto Provision Time	None 🗸
Auto Provision Next Time	Thu Aug 8 12:24:00 2013 Reset Timing
AES Enable	● off ○ on
AES Key	
	Auto Provision Now

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	Used for auto download Enterprise Phonebook from	
Phonebook	server	
Auto Download Personal	Used for auto download personal phonebook from server	
Phonebook		
Booting Checked	Used for checking the auto provision when phone booting	
Disable the phone while	Enable/Disable the booting checking feature.	
booting checking		
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.	

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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.

	Administrator O User
Username	root
Old Password	
New Password	
Confirm Password	

# **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.

🕵 Administrator | Logout

### 10.2 Multi-Language

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

Please Select Language: English(English) Chinese(Chinese) Russian(Russian) Polish(Polish) Portuguesa(Portuguesa) French(French) Brasil(Brasil) Turkish(Turkish)

### 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.

Note
 Register status:
 It shows the Register Status.

Network Status: It shows the information of LAN port and PC port.

System Info: It shows the version of firmware