



ES290 IP Innovative VoIP Phone User Manual



Escene Communication Co.Ltd

www.escene.cn/en/

Escene Communication

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1. Getting Started

About

Escene ES290 is a highly innovative based VoIP phone, It can perfectly satisfy all kinds of businessmen's communications. ES290 integrates with 132x64 pixel graphic LCD, a large resolution display, elegant and intuitionist user interface, which you can perfectly experience.ES290 delivers HD audio quality with HD handset, HD speaker and HD codec (G.722), it rich features can effectively improve the communications in working. ES290 quite meets the demands of SMEs, Home Office and ISP applications.

Feature Highlights

a) Multi-Language

The LCD display supports Multi-Language.

b) HD Voice

Special voice processing technology, high-fidelity voice quality, HD encoding, HD Handset, ensure clear, realistic smooth communication.

c) Senior Calling Ability

2 lines with double color(GREEN & RED) LEDs, Synchronously control or manage 2 calls, Call queue, Switch between lines. Multi-parties conference, call transfer.

d) All kinds of Phone Book

It supports XML Personal Phone Book\LDAP\Enterprise Phone Book etc. This feature satisfies customer's phone book requirements.

- e) Support HTTP\TFTP\Auto-Provision.
- f) Support POE(Remark: ES290-PN) and Power Adapter.
- g) 2-angle adjustable bracket, wall-mountable

2. Set up the Phone

a) Packing List:

Check the packing list before installation, if you find anything missing, contact your system administrator.

- 1*ES290 IP Phone
- 1*Handset
- 1* Handset Cord
- 1*Ethernet Cable
- 1*Phone Bracket

- 1*Quick Setup Guide
- 1*Manufacturer Certification

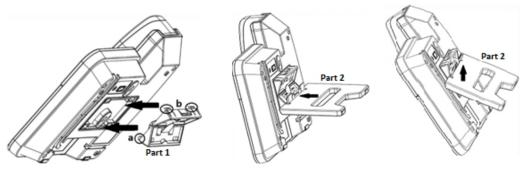
b) Phone Installation:

This section introduces how to install the phone with the components in the packing list:

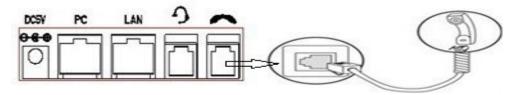
- Attach the Bracket
- Connect the Handset and optional Headset
- Connect the Network and Power

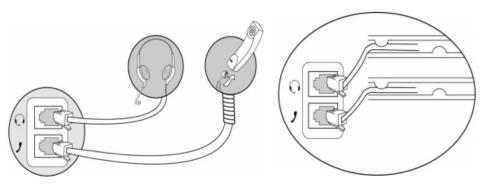
Attach the Bracket

Pls follow the following step "a" to "b", firstly let the part 1 of the bracket join to the phone, and then let the part 2 of the bracket join to the lower holder.



Connect the Handset and optional Headset



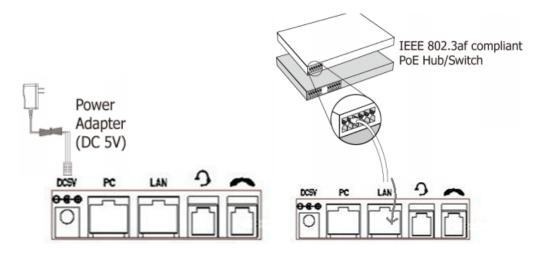


Connect the Network and Power

You have two options for power supply. Your system administrator will advise you which one to use.

- AC power adapter
 - POE(Power over Ethernet) IEEE802.3af

NOTES: Pls make sure your phone support POE feature. You can check the label on the back of the phone , for example "Model: ES290-PN Rev:2.1.0", the P means it support POE feature.

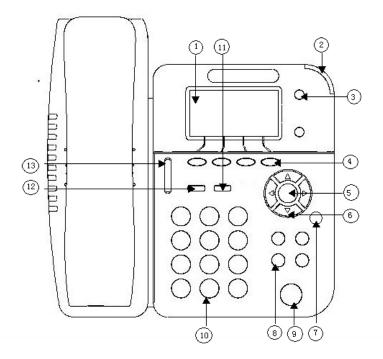


Note: If POE works, the phone doesn't need to connect to the AC power adapter. Make sure the Ethernet cable and switch/hub is POE compliant.

3. Phone User Interface

3.1 Hardware Component Instructions

The main hardware components of the ES290 IP Phone are the LCD screen and the keypad.



	ITEM	DESCRIPTION		
1	LCD Screen	Displayed information about calls, messages, soft keys, time, date and		
		other relevant data:		
		• Call information — caller ID, call duration		
		• Icons (e.g. DND)		
		 Missed calls or second incoming caller's information 		
		•Time and date		
2	Light Status	Red-Flashing: There is an incoming call or be Hold.		
		Red-Steady: Hook-off. or be in an usual conversation		
3	Line Key	Green-Steady: There is a conversation making on the line		
		Red-Flashing: There is call coming in		
		Green-Flashing: The line is on hold		
		Dark: Accounts are idle		
4	Soft Key	Labels automatically to identity their context-sensitive features		
5	ОК Кеу	Confirm the action		
6	Navigation Key	Scroll through the displayed information, and in the idle feature:		
		UP: Open the "All CONTACT LOG"		
		DOWN: Open the "MISSED CALL"		
		RIGHT: Open the "RECEIVED CALL"		
		LEFT: Open the "DAIL CALL"		
7	С Кеу	Cancels actions or rejects an incoming call, and the other feature:		
		In the idle: Open the "Phone Status".		
		Diagnosis: Press and hold 3 second to open "Hardware Diagnosis".		
		MUTE: "MUTE" feature is enabled if you press it while the conversation		
		making on the phone.		
8	Functions Key	Conference\Redial\Transfer\Hold		
9	Speaker Key	Toggles the hands-free speaker phone mode.		
10	Keypad	Provides the digits, letters and special characters in context-sensitive		
		applications.		
11	Headset Key	Toggles and indicates the headset mode.		
12	Message Key	Indicates and accesses voice messages.		
13	Volume Key	Adjusts the volume of the handset, headset, speaker and ringer		

Hardware component instructions of the ES290 IP Phone are:

3.2 Phone Screen Display Features

If the phone has successfully started up and after using, the idle LCD display will show information as below:

スロッやいてあるの言言	X (1) (2) (4) 🗸 🛏 (3) (4) (6) (1) (4)
☎5207 🚍	🏠 5207 🛛 🗖 🗖 🛣
12 :0 9 12 📼	12:11 or 📼
01-08 THU 🚍	1 MissedCall 🛛 🛣
Menu Los DND Dir	Menu Log DND Dir

	ITEM	DESCRIPTION	
1	TIME & DATE	TIME & DATE display in the middle of the screen.	
2	Auto-Answer icon	Enable this feature, it will display "AA" at the top right corner.	
3	Missed Call	Missed Call under the TIME in the middle of screen	
4	Line Status	There are four status as below:	
		a. LAN:Disconnect:Disconnect the network	
		b. Eacount failed to register	
		c. Account successfully registered	
		d. Account successfully registered and DND feature is enabled.	
		The DND icon also will display at the top right corner.	
5	Soft Key Area	Labels automatically to identity their context-sensitive features	
6	Screen Top Icon	The Screen Top Icon from left to right is:	
		Handset Hand on status	
		:Speaker Hand on status	
		:Headset Hand on status	
		Call MUTE	
		Missed Call	
		Call Forward	
		E:Text Message	
		Keypad Lock	
		Retwork is unavailable	

3.3 Basic Network Settings

DHCP Settin	g		
Feature	Operating Steps		
DHCP	Press OK or MENU> System	Press OK or MENU> System Settings> Advanced Settings> Password(Default is	
	Empty)> Network> LAN Po	rt	
	•	Press LAN Port to login in to the menu	
	•	Select "DHCP" mode	
	•	Press " Enter " key	
	•	Set the DNS\web port\telnet port	
	•	Press " Save " key to make it work	
		Tips "Network is changed, press OK	
	reboot "		

The phone supports Three Modes of Network Setting. Include PPPoE\Static IP\DHCP.

Static IP Setting

Feature	Operating Steps	
Static IP	Press OK or MENU> System S	ettings> Advanced Settings> Password(Default is
	Empty)> Network> LAN Port	
	•	Press LAN Port to login in to the menu
	•	Select " Static " mode
	•	Press " Enter " key
	•	Set the IP\Mask\GW\DNS\web
	port\telnet port	
	•	Press " Save " key to make it work
	•	Tips "Network is changed, press OK
	reboot "	

PPPoE Setting

Feature	Operating Steps	
PPPoE	Press OK or MENU> System S	ettings> Advanced Settings> Password(Default is
	Empty)> Network> LAN Port	
	•	Press LAN Port to login in to the menu
	•	Select " PPPoE " mode
	•	Press " Enter " key
	•	Set the User Name\Password\web
	port\telnet port	
	•	Press "Save" key to make it work
		Tips "Network is changed, press OK
	reboot "	

3.4 SIP Account Settings

ES290 IP phone makes calls based on sip accounts, It can support Single account or Multi-account, Each account can be configured to the different SIP server.

If you want to	Then
Create an SIP account	1) Select "System setting" > "Advanced setting";
	2) Enter the password required (The default is empty);
	3) Select "SIP" > "Account sip";
	4) Select one of the account you want to setting, you can configure
	the following parameters
	-Enable account*: Select Enable
	-Number of lines: Default is 2
	-Description: description of this account
	-Display Name: The name displayed on the screen
	-Authentication user: the Authenticated users are matched with
	the SIP server. (The default With the same account)
	-Account*: the account matches with the SIP server.(extension
	number)
	-User pass word*: the user password matches with the SIP server
	-SIP Server*: The primary SIP server, all calls through this server
	-Out Bound Server: The out bound SIP server
	-STUN Type: Enable/Disable STUN feature
	-STUN: Input STUN URL -Auto Answer: Enable/Disable this account auto answer feature
	* Note: When you finish the setting, you can press Save to make it
	work, and then you can see the status icon in the LCD idle.
	The parameters with the * mark must be set.
Disable sip account	1) Select "System setting" > "Advanced setting";
	2) Enter the password required (The default is empty);
	3) Select "SIP" > "Account sip";

4) Select "Enable account" > "Disable";
5) Select "Save" to saves settings

3.5 Basic Features

3.5.1 Making a Call

If you want to		Then
Place a call using	Pick up the handset	1) You can hear dial tone; 2) Enter a number;
the handset		3) Press # button (default),
Place a call using a	Press Speaker button	 -or wait 5s (default), then it send the number automatically.
speakerphone		
Place a call using a	Put on your headset,	
headset	active Headset button so	
	that the status light is	
	Red 问 , and then do as	
	using speakerphone	

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

3.5.2 Anonymous Call

You can use anonymous call feature to block the identity and phone number from showing up to the called party when you call someone. E.g, you want to call to consult some of the services, but you don't want to be harassed.

Enable Anonymous Call	Press OK or MENU> Function Setting> Anonymous
	Press Enter or OK button ,
	-You can select which Account want to use, enable/disable this
	feature and enable/disable reject anonymous

3.5.3 Redial

Redial	Press REDIAL button to dial the last number
	-or press Navigation button-Left > "Dialed number", select a
	number, and press Dial

To redial the last placed call from your phone

3.5.4 Call Log

Dial from a call log	1) Press MENU or OK button > "Call history", you can select "All
	Calls", "Missed calls", "Received calls" and "Dialed numbers",
	- or press Navigation button (in Standby interface) > select "All
	Calls"(up) "Missed calls"(down), "Received calls"(left) and "Dialed
	numbers" (right)
	2) Then press Dial button.
	NOTE: You also can press the "log" to login this menu when in the
	idle.

3.5.5 Making Calls to Contact

You can also dial a contact from the Personal Phone Book.

Placing Contacts	Calls	to	1) Press MENU or OK button > "Phone Book", you can select
Contacts			"Personal Phone Book", "Enterprise Phone Book", "LDAP" and
			"Black List",
			- or press Navigation button (in Standby interface) > select the
			desired contact.
			2) Then press Dial button.
			NOTE: You also can press the "DIR" to login this menu when in the
			idle.

3.5.6 Multi-lines to Answer the Call

Multi-lines to Answer the Call	1) Another Line button is Red 💗 and flashing, Light strip is Red and
	flashing;
	2) Press the flashing \bigcirc Line button to answer (at this time, the
	original call will be hold.)

3.5.7 Auto-Answer

You can set the phone	and let it auto-answer	the coming call.

Auto-Answer Coming Call	the	1) Enable the Auto-Answer feature.
coming can		2) Auto-Answer mode you can set in the MENU>Function Setting>
		Auto Answer >Device
		• Speaker
		 Handset
		 Headset
		When you use the Handset mode, at this time you need to hands up
		the handset and then it can work at this status.
		3)Filter Groups
		Auto-answer the coming call in this special groups.

3.5.8 Ending a Call

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

Hang up while using the	Return the handset to its cradle,
Handset	-or press End
Hang up while using the	Press Speaker button that is Red

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	

3.5.9 Using Hold and Recover (Switch Calling Line)

You can hold and resume calls. You can take a call in one line at anytime, 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,
	-or press soft key Hold
Hold a line and switch to	Press another Line button for the appropriate line
another line	
Resume a call on current	Press Line button,
line	
Release a call on different	Select the line want to release hold, press the line, so recovery;
line	

NOTES

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

• A held call is indicated by the Yellow-green 💛 and flashing Line button or Hold in the LCD.

3.5.10 Transferring Calls

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

Talk to the transfer	1) Press TRANSFER button or press XFER;
recipient before	2) Enter number;

transferring a call	3) press " #" (default) ,
(consult transfer)	-or press Send then transfer the call,
(00000000000000000000000000000000000000	-or wait five seconds(default)then transfer the call
Transferred to idle	1) Press TRANSFER button or XFER;
lines or other	2) Press Blind;
numbers without	3) Enter number;
talking to the transfer	4) Press " # " (default)
recipient	-or press Send, then transfer the call;
(Blind transfer)	-or wait five seconds(default)then transfer the call
Blind transfer to the	1) Press TRANSFER button or press XFER;
held line	2) Press the Line button of held line

3.5.11 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 \circ

Toggle Mute on	Press ${f C}$ button, then the screen top and left will have a MUTE
	icon
Toggle Mute off	Press C button again, then the button light off

3.5.12 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.).

Enable global DND	1) Press DND ;
	2) All enabled line on the phone would changes to 🔤 status. and
	the icon is DND .
Enable DND on a	Press MENU or OK button > "Function setting" > "DND" > (select line)
single line	"Enable"
Disable DND	Global DND enabled, press DND to disable global DND;
	Line DND enabled, press twice DND ,

-or press MENU or OK button > "Function setting" > "DND" >(select
line) "Disable"

3.5.13 3-way Conference

You can enable a three-party conference, during the conversation three phone parties can communicate with every party.

If you want to	Then
Invite the transfer	1) When the transfer recipient answer the call, press CONFERCENCE
recipient into a	button or "CONF" on your phone;
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 "CONFERENCE" button or "CONF" in an active call;
into a conference in a	2) Enter the third party number;
active call	3) After connected the third party, press "CONFERENCE" button or
	"CONF" again
establish a conference	1) when one phone line is holding on and the other line is busy;
with held line	2) Press "CONFERENCE" button,
	-or Press "CONF" Soft key
	3)Press the held line's programmable button, the 3-way Conference
	is enable.

3.5.14 Voice Mail

When the Phone get a voice mail from server. it will light up the voice mail button——.		
Voice Mail	1) Press the Voice Mail button(There has Voic	e Mail
	icon—,without is—)	
	2) Enter the User Password	
	3) It will login into the voice mail server. You need to follow the	ne IVR to
	do it.	

3.6 Advanced Settings

3.6.1 Using the phone book

Enterprise Phone Book

Search the Contacts	1) Press DIR in the idle status,
from Enterprise	-or press " MENU" or "OK" button > "Phone book">"Enterprise
Phone Book	Phone Book",
	2) Select "Enterprise Phone Book", press " OK" button;
	3) Press "Find" and input the name who you want to search.
Call the Contact	1) Press "DIR" in the idle,
from Enterprise	-or press " MENU" or "OK" button > "Phone book">"Enterprise
Phone Book	Phone Book",
	2) Select "Enterprise Phone Book", press " OK" button;
	3) Press "Find" and input the name who you want to search.
	4) When you search the person, you can dial it.

Personal Phone Book

Add Contacts	1) Press Phone Book,
	-or press " MENU" button > "Phone book">"Personal phone
	book>View All",
	-or press " OK" button > "Phone book">"Personal phone book>View
	All";
	2) Select "Add contact", press " OK" button;
	3) Use the navigation keys to select content, press "OK" button to set
	and modify:
	-Name: set the name of contact,

	-Office Number: Setting the contact Office Number
	-Mobile Phone Number: Setting the contact Mobile Phone
	Number
	-Others Number: Setting the contact Others Number
	-SIP Account: Setting the contact call SIP account
	-Group: the contacts be divided into different user's groups
	4) Press " Save" soft key to complete
Add group	1) Press "DIR" soft key,
	-or press " MENU" button > "Phone book">"Personal phone
	book>View All",
	-or press " OK" button > "Phone book">"Personal phone book>View
	All";
	2) Select the "add group" then press OK button;
	3) Use the navigation keys to select content, press OK button to set
	and modify:
	-Group name: name of the group
	4) Press " Save "soft key to complete
Modify group	1) Press "DIR" soft key,
	-or press "MENU" button > "Phone book">"Personal phone
	book>View All",
	-or press " OK" button > "Phone book">"Personal phone book>View
	All";
	2) Select the "Modify group" then press " OK" button ;
	3) Select the group you want to modify, press the "OK" button to
	set and modify, press " Save" to save the change
Delete group	1) Press "DIR" soft key,
	-or press " MENU" button > "Phone book">"Personal phone
	book>View All",
	-or press " OK" button > "Phone book">"Personal phone book>View
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All";
2) Select the "Delete group" or OK button;
3) Select a group you want to delete, press OK button

LDAP

Search the Contacts	1) Press "DIR" in the idle,
from LDAP	-or press " MENU" or OK "button" > "Phone book">"LDAP",
	2) Select "LDAP", press " OK" button;
	3) Press "Find" and input the name or number who you want to find
	search from the LDAP server.
Call the Contact	1) Press "DIR" in the idle,
from LDAP	-or press " MENU" or "OK" button > "Phone book">"LDAP",
	2) Select "LDAP", press " OK" button;
	3) Press "Find" and input the name or number who you want to find
	from the LDAP server.
	4) When you search the person, you can dial it.

Black List

Add the Contacts	1) Press "DIR" in the idle,
	-or press " MENU" or "OK" button > "Phone book">"Black List",
	2) Select "Black List", press " OK" button;
	3) Press "Add" and input the name\office number\mobile
	number\other number\SIP account who what you want to add into
	the Black List.
View the Contact	1) Press "DIR" in the idle,
from Black List	-or press " MENU" or "OK" button > "Phone book">"Black List",

2) Select "Black List", press " OK" button;
3) Press "RUN" to view someone who what you want to find.
4)If you want to move or change it, you can follow the RUN to do.

3.6.2 Using Call Logs

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

View call logs	1) Press "MENU" or "LOG" button > "All Call" > "Missed Calls",
	"Received Calls", or "Dialed numbers"
	2) Use the navigation keys to view the call record information.
Delete/Save Call	1) Login in to the Call Logs
Logs	2) Use the navigation keys to view the call record or select DEL key.
	3) Use the navigation keys to view the call record or select Save key.

3.6.3 Peer-to-Peer

When all of the phone accounts were disable or not register. it will be show this mode in the idle. It can use by when the new workstation isn't have SIP server.

Peer-to-Peer	Disable all of accounts or not register.	
Make Call with	1) Press OK or MENU button> System Setting> Advanced Setting> SIP	
Peer-to-Peer	Account;	
	2) Disable all of accounts or un-register;	
	3) Turn back the phone idle, you can call someone use IP address.	

3.7 Keypad Setting

SayHi series IP Phone can through two ways configuration it, one is setting in MENU, another is setting in website. Here just description in MENU.

NOTES: When you want to input the IP address like ".", it was replaced by the "*".

3.7.1 Language Setting

Switch the Language	1) Press OK or MENU button> System Setting> Phone Setting>	
between Chinese and	Language	
English	2) Here you can select	
	English\French\Italian\Polish\Protuguese\Runssian\Spanish\T	
	sh\Chinese	
	3) After you finish select, press Save to make it work.	

ES290 IP Phone support Multi-Language setting, as below is an example.

3.7.2 Message

ES290 have Message feature. It will display in the LCD when it has a New Message.

Create a Message	1) Press OK or MENU button;	
	2) Select "Messaging"	
	3) Voice Message: Setting the Voice Message code in here.	
	Text Message: Write down the Text Message in here.	
	4) Select Text Message> New Message.	
	5) Input the receiver and write down message body, and then press	
	send to finish.	
Message Inbox	1) Select Message Inbox.	
	2) Select which one you want to check.	
	3) You can press Enter to read or press Del to delete.	

3.7.3 Time & Date

SNTP	1) Press OK or MENU button;		
	2) Press OK or MENU button> System Setting> Phone Setting> Time & Date>		
	Time and Date setting> SNTP		
	3) SNTP		
	-Time Zone: Setting the time zone		
	-NTP Server 1: NTP server address 1		
	-NTP Server 2: NTP server address 2		
	-DayLight: Enable/Disable Day Light		
SIP Server	1) Press OK or MENU button;		
	2) Press OK or MENU button> System Setting> Phone Setting> Time & Date>		
	Time and Date setting>SIP Server		

	3) Press Save to make it work
Manual Setting	Press OK or MENU button;
	2) Press OK or MENU button> System Setting> Phone Setting> Time & Date>
	Time and Date setting> Manual Setting
	3) Manual Setting
	 Manual Setting: Year\Month\Days\Hours\Minutes\Seconds
Time Display	1) Press OK or MENU button;
	2) Press OK or MENU button> System Setting> Phone Setting> Time & Date>
Format	Time Display Format
	3) Time Mode: 24hour\12hour
	Date mode:
	DDMMWWW\MMDDWWW\WWWDDMMM\DDMMMYY\YYYYMMDD\DDM
	MYYYY\MMDDYY\DDMMMYYYY\WWWDDMMM etc.

3.7.4 Ring Tone and Volume Setting

1) Press OK or MENU button;	
2) Press OK or MENU button> System Setting> Phone Setting>Ring Type	
3) Select the ring type from 1 to 8 or custom ring, and then press Save to	
make it work.	
1) Press OK or MENU button;	
2) Press OK or MENU button> System Setting> Phone Setting> Volume	
Setting	
3) Volume Setting: Handset\Speaker\Headset\Ring volume	
4) Press Enter to adjust the volume and press Save to make it work	

NOTES: For the Custom Ring Type you need to upload it from website.

3.7.5 Searching Phone Book

Accurate Search	1) Press MENU or OK button > "Function Setting", you can select " Accurate Search "		
	2) Then press Enable/Disable and Save.		
	3) When you back to idle, you can use the digital keypad to search the contact.		
T9 Search	1) Press MENU or OK button > "Function Setting", you can select " T9		
	search"		
	2) Then press Enable/Disable and Save.		
	3) When you back to idle, you can use the digital keypad to search the		
	contact.		

NOTES: The Search Phone Book setting default is Accurate Search.

3.7.6 Cannot Set the Features with Keypad

As below features are cannot setting with the keypad:

1) Dial Plan.

- 2) Custom Ring Type
- 3) SNTP Server and Time & Date
- 4) Update the Firmware or Backup.

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,

<u> </u>			🥞 Administrator Logout
ES ENE			Please Select Language:
			English(English)
Current Lo	ocation: Phone Status		
Phone Status Phone S	status		
Network			Note
Network	System Run Time	1 Days17 Hours22 Minutes17 Seconds	Register status:
SIP Account	Register status 🔞		It shows the Register Status.
	Account1	Registered	
Phone Setting	Account2	Unregister	Network Status:
	Network Status 🚱		It shows the information of LAN port
PhoneBook	LAN Connection	Dynamic	and PC port.
	MAC Address	00:26:8b:01:c4:3c	
Phone Maintenance	LAN IP Address	192.168.0.205	System Info:
	Netmask	255.255.255.0	It shows the version of firmware
Password	Gateway	192.168.0.1	
	Primary DNS	210.21.4.130	
	Secondary DNS	192.168.1.200	
	VPN IP Address		
	PC IP Address		
	PC Netmask		
	Device Type	Bridge	
	DHCP Server	off	
	System Info 🔞		
	Phone Model	ES290N	
	Software Version	V1.0.8.8-3974	
	Hardware Version	V2.x.x	
	Hardware ID	1	
	Kernel Version	V2.6.4	
	AutoProvision Server URL	voip.autoprovision.com	
	TFTP Server IP	voip.autoprovision.com	

ITEM	DESCRIPTION	
System Run Time	The phone system normal running time.	
Register Status	The status with Account 1~3.	
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)	
	Static IP 🕜	
	IP Address	192.168.0.200
	Netmask	255.255.255.0
	Gateway	192.168.0.1
	O PPPoE 🕜	
	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 Mode Network Connection Mode has DHCP, Static IP, PPPoE.	
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	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on	
Server IP		*
Port	1080 *	
Anonymous Login	\checkmark	
Username]
Password		
Paging Setting		
Paging 1	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ 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.	
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.

e Bridge @		
🔿 Router 🕜		
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)

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	The vlan ID you want the phone or pc to join
[LAN/PC Port]	

5 SIP Account

5.1Basic

	-
Enable	
Account Mode	VOIP 🗸
Amount Of Line Account Used	1 (Default: 2)
Display Name	•
Username	5207 * 🕜
Authenticate Name	5207
Password	••••
Label	0
SIP Server	192.168.0.7
Secondary server	0
OutboundProxy Server	•
Secondary OutboundProxy Server	0
Polling Interval Time Of Registration	32 s Default Value: 32s, Range: 20s~~60s
NAT Traversal	Disable 🗸 🕜
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	DECSRIPTION	
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	The line key of account used, default is 2	
Account Used		
Display Name	It is showed as Caller ID when making a phone call	

his secondary
ise this

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	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on
Allow-events	${\ensuremath{\overline{\odot}}}$ off ${\ensuremath{\overline{\odot}}}$ on
Registered NAT	⊖ off ● on
UDP Keep-alive Message	${\ensuremath{ \bullet }}$ off ${\ensuremath{ \circ }}$ on
UDP Keep-alive Interval	30 (15-60s)

ITEM	DECSRIPTION
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

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

ITEM	DECSRIPTION	
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
Keyboard Lock	Disabled V
Hot Line Function	● off ◯ Delay 5 s (0-30)
Hot Number	0
Auto Answer	\odot 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)
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 🗸

ITEM	DECSRIPTION
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	When select on, then you can capture the phone's packet use notebook
Mirroring	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 $oldsymbol{O}$
Time Format	● 24 Hour 〇 12 Hour 🚱
Date Format	DD MM WWW 🗸 🚱
Time Zone- GMT	GMT+08:00 Beijing V
Manual Setting	2000 Year Month 1 Days 0 Hours 0 Minutes 0 Seconds

ITEM	DECSRIPTION	
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	\bigcirc off \odot 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	○ off ● on
Transfer Code	● off ○ on Number:
Conference Exit Result	 Disconnect All O 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)
Pound Send Mothod	● # ○ %23
RFC 2833 PayLoad	101
P-Asserted-Identity	\bigcirc off \odot on
SIP Session Timer(seconds) T1	0.5
SIP Session Timer(seconds) T2	4 🕜
SIP Session Timer(seconds) T4	5 🕜
Local SIP port	5060 (Default: 5060)
RTP Port Range	10000 10128
Affiliated Port	\bigcirc off \odot on
Headset Mode	 Normal O Seat Mode
Ring Type On Seat Mode	Headset O Speaker

ITEM	DECSRIPTION
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 keyEnable/Disable the DND feature.Play Hang-up ToneThe tone with hang up in busy.Transfer CodeThe code with transfer.Conference Exit ResultConference originator hang up the phone, hang up two ways of it.Return Code WhenSelect the code feedback to the server when you reject the call.RefusePactor of the code feedback to the server when you open DND function.Flash Hook Time(<800ms)The time with the flash hook.Called No Answer TimeWhen it has coming call and enable this feature, the caller will be request time out in the stipulated time.Pound Send MethodWhen you to use the code, such as: #28#123 or %23123, you need to se this feature.RFC 2833 Play LoadDefault is 101, RTP Payload for DTMF Digits, Telephony Tones and Telephony SignalsP-Asserted-IdentityEnable/Disable the P-Asserted-Identity feature.SIP Session Timer T1The SIP Session Timer setting.SIP Session Timer T2The SIP Session Timer setting.SIP Session Timer T4The SIP Session Timer setting.SIP PortThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.Ring Type On Seat ModeSelect ring type mode with headset or speaker.	·	
Transfer CodeThe code with transfer.Conference Exit ResultConference originator hang up the phone, hang up two ways of it.Return Code When RefuseSelect the code feedback to the server when you reject the call.Return Code When DNDSelect the code feedback to the server when you open DND function.Flash Hook Time(<800ms)	DND Soft key	Enable/Disable the DND feature.
Conference Exit ResultConference originator hang up the phone, hang up two ways of it.Return Code When RefuseSelect the code feedback to the server when you reject the call.Return Code When DNDSelect the code feedback to the server when you open DND function.Flash Hook Time(<800ms)	Play Hang-up Tone	The tone with hang up in busy.
Return Code When RefuseSelect the code feedback to the server when you reject the call.Return Code When DNDSelect the code feedback to the server when you open DND function.Flash Hook Time(<800ms)	Transfer Code	The code with transfer.
RefuseSelect the code feedback to the server when you open DND function.Flash Hook Time(<800ms)	Conference Exit Result	Conference originator hang up the phone, hang up two ways of it.
Flash Hook Time(<800ms)The time with the flash hook.Called No Answer TimeWhen it has coming call and enable this feature, the caller will be request time out in the stipulated time.Pound Send MethodWhen you to use the code, such as: #28#123 or %23123, you need to se this feature.RFC 2833 Play LoadDefault is 101, RTP Payload for DTMF Digits, Telephony Tones and Telephony SignalsP-Asserted-IdentityEnable/Disable the P-Asserted-Identity feature.SIP Session Timer T1The SIP Session Timer setting.SIP Session Timer T2The SIP Session Timer setting.SIP Session Timer T4The port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.		Select the code feedback to the server when you reject the call.
Called No Answer TimeWhen it has coming call and enable this feature, the caller will be request time out in the stipulated time.Pound Send MethodWhen you to use the code, such as: #28#123 or %23123, you need to se this feature.RFC 2833 Play LoadDefault is 101, RTP Payload for DTMF Digits, Telephony Tones and Telephony SignalsP-Asserted-IdentityEnable/Disable the P-Asserted-Identity feature.SIP Session Timer T1The SIP Session Timer setting.SIP Session Timer T2The SIP Session Timer setting.SIP Session Timer T4The SIP Session Timer setting.Local SIP PortThe port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	Return Code When DND	Select the code feedback to the server when you open DND function.
request time out in the stipulated time.Pound Send MethodWhen you to use the code, such as: #28#123 or %23123, you need to se this feature.RFC 2833 Play LoadDefault is 101, RTP Payload for DTMF Digits, Telephony Tones and Telephony SignalsP-Asserted-IdentityEnable/Disable the P-Asserted-Identity feature.SIP Session Timer T1The SIP Session Timer setting.SIP Session Timer T2The SIP Session Timer setting.SIP Session Timer T4The SIP Session Timer setting.Local SIP PortThe port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	Flash Hook Time(<800ms)	The time with the flash hook.
this feature.RFC 2833 Play LoadDefault is 101, RTP Payload for DTMF Digits, Telephony Tones and Telephony SignalsP-Asserted-IdentityEnable/Disable the P-Asserted-Identity feature.SIP Session Timer T1The SIP Session Timer setting.SIP Session Timer T2The SIP Session Timer setting.SIP Session Timer T4The SIP Session Timer setting.Local SIP PortThe port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	Called No Answer Time	-
Telephony SignalsP-Asserted-IdentityEnable/Disable the P-Asserted-Identity feature.SIP Session Timer T1The SIP Session Timer setting.SIP Session Timer T2The SIP Session Timer setting.SIP Session Timer T4The SIP Session Timer setting.Local SIP PortThe port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	Pound Send Method	When you to use the code, such as: #28#123 or %23123, you need to set this feature.
SIP Session Timer T1The SIP Session Timer setting.SIP Session Timer T2The SIP Session Timer setting.SIP Session Timer T4The SIP Session Timer setting.Local SIP PortThe port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	RFC 2833 Play Load	
SIP Session Timer T2The SIP Session Timer setting.SIP Session Timer T4The SIP Session Timer setting.Local SIP PortThe port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	P-Asserted-Identity	Enable/Disable the P-Asserted-Identity feature.
SIP Session Timer T4The SIP Session Timer setting.Local SIP PortThe port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	SIP Session Timer T1	The SIP Session Timer setting.
Local SIP PortThe port range setting with SIP, default is 5060.RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	SIP Session Timer T2	The SIP Session Timer setting.
RTP Port RangeThe port range with RTPAffiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	SIP Session Timer T4	The SIP Session Timer setting.
Affiliated PortEnable/Disable the affiliated port feature.Headset ModeSelect headset mode with normal or seat.	Local SIP Port	The port range setting with SIP, default is 5060.
Headset Mode Select headset mode with normal or seat.	RTP Port Range	The port range with RTP
	Affiliated Port	Enable/Disable the affiliated port feature.
Ring Type On Seat Mode Select ring type mode with headset or speaker.	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	${\small \bullet} {\small \ } {\rm off} {\textstyle \bigcirc} {\small \ } {\rm on}$	Number:	0
If Busy	${\small \bullet} {\small \ }$ off ${\displaystyle \bigcirc} {\small \ }$ on	Number:	0
If No Answer	${\small \bullet} {\small \ } {\rm off} {\small \bigcirc} {\small \ } {\rm on}$	Number:	0
Ring Frequency	15	Seconds (Default: 15s, Max: 15s)	

ITEM	DECSRIPTION
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	DECSRIPTION	
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	DECSRIPTION
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	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 🕜	$\begin{tabular}{ c c c c c c } \hline Up & G723 & << & G722 \\ G711U & & \\ G729A \\ iLBC & & \\ G726_32 & & \\ \hline \end{tabular} \end{tabular} \end{tabular} \end{tabular} \end{tabular} \end{tabular}$
Jitter Buffer 🔞	
Туре	Adaptive O 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	DECSRIPTION
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	1

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

6.3 Line Keys

	Mode	Account	Name	Number
Key1:	Line N	🖌 Account1 🗸		
Key2:	Line 💉	🖌 Account1 🗸		
Key3:	Line 💉	🖌 Account1 🗸		

line keys >> Mode Account Name Number Account1 🗸 Key1: Line Speed Dial Key2: Account1 🗸 Speed Dial Prefix DTMF Key3: Account1 V BLF Paging Call Park Submit Intercom BLA Function Kove 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. You can use multicast paging to quickly and easily forward time sensitive Paging 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 www.escene.cn/en/

	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

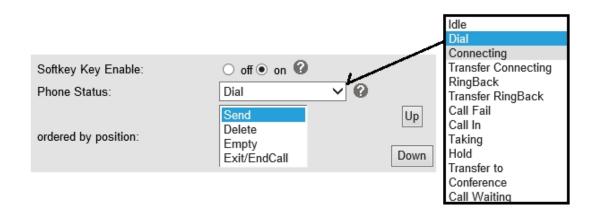
Function Keys: If you do not like the default setting with the function keys feature. You can change to whatever you like.

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	25	
	No Dial Timeout		5		
ID	Operation	Prefix	IP Address	Description	
	Add Rule	Delete All Rule]		

ITEM	DECSRIPTION
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.)		

www.escene.cn/en/

ITEM	DECSRIPTION
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 V	Description	test2
Group Name	test2	Ring Type	Ring2 🗸
		Submit Cancel	

	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	
ar	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 \overline{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 Number	1 🗸		
First Name	test	Last Name	test
Mobile Number	1111	Office Number	1111
OtherNumber	1111	Account	Account1 🗸
Group1	test 🗸	Group2	None 🗸



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 \overline{m} , 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 <a>Image. You can download and save this contact to PC after you select <a>Image.

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 🔾 on 💿 off 🕜 LDAP 0 LDAP Name Filter (sn=%s) 0 LDAP Number Filter (telephoneNumber=%) 0 Server Address 192.168.0.65 0 Cwmp Port 389 DC=Idap,DC=escene, 0 Base 0 bb@ldap.escene.com Username Password escene 2012 0 0 Max. Hits(1~32000) 50 0 LDAP Name Attributes 1 sn LDAP Name Attributes 2 cn LDAP Name Attributes 3 0 LDAP Number Attributes 1 telephoneNumber LDAP Number Attributes 2 LDAP Number Attributes 3 Version2 • Version3 Protocol 0 Search Delay(ms)(0~2000) 0 💿 on 🔾 off 🚱 LDAP Lookup For Incoming Call 💿 on 🔿 off 🚱 LDAP Lookup For PreDial/Dial

7.4 Ban List

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

Serial Number	1 🗸	Descr	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	
				Account
ID Operation 1 Image: Constraint of the second sec	Name test3 testc	Phone Number1:3333 Number2:3333 Number3:3333	Description 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 📧.

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 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.

FTP Upgrade >>	
Server IP	
Filename	
Username	
Password	
Software Upgrade	Upgrade
Kernel Upgrade	Kernel Upgrade
Note: It's no necessary to	input filename when 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 Upgrade
Note: It's no necessary to input filename w	hen 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

I

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.

_og	>>		
		○ No Record	
		 Call 	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	● on ○ off
Option:	66 (Default :66, Min:1, Max:254)
Protocol	TFTP 🗸
Software Server URL	voip.autoprovision.com
Username	
Password	
Auto Download Software	v
Auto Download Kernel	\checkmark
Auto Download Config File	\checkmark
Auto Download Expansion	v
Auto Download Enterprise Phonebook	v
Auto Download Personal Phonebook	v
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 V
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 server
Phonebook	
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.

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 User
root

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