

WELL SIP-T26P User manual



About This Guide

Thank you for choosing this Enterprise IP Phone which is especially designed for power users in the office environment. It features fashionable and sleek design, abundant telephony applications, broad interoperability with the popular 3rd party VoIP products, fulfilling the VoIP deployment needs from enterprise and ITSP.

In this User Guide, you will find everything you need to quickly use your new phone. Be sure to verify with your system administrator that your network is prepared for configuring your IP phone. As well, be sure to read the Packing List section in this guide before you set up and use the phone.

Declaration of Conformity



Hereby, it's declared that this phone is in conformity with the essential requirements and other relevant provisions of the CE, FCC.

CE Mark Warning

This is a class B device, in a domestic environment; this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning



To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and

have to collect such WEEE separately.

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Trouble Shooting 1

Getting Started

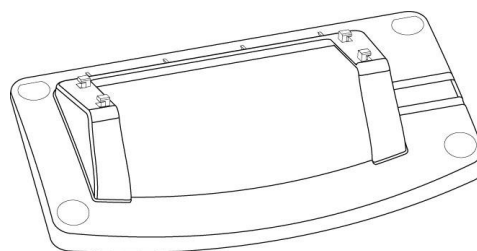
Packing List

The following components are included in your package:

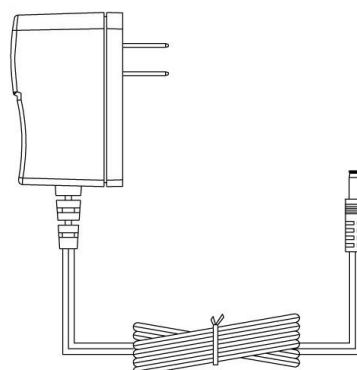
- Enterprise IP Phone



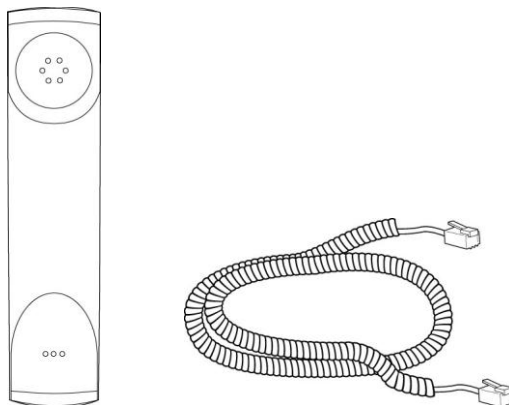
- Phone Stand



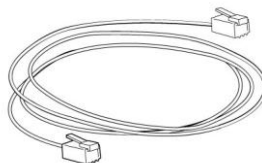
- Power Adapter



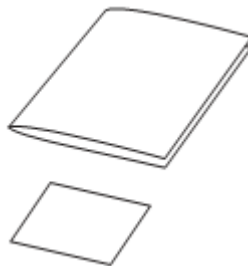
- Handset & Handset Cord



- Ethernet Cable



- Quick Installation Guide & Quick Reference



- CD Content



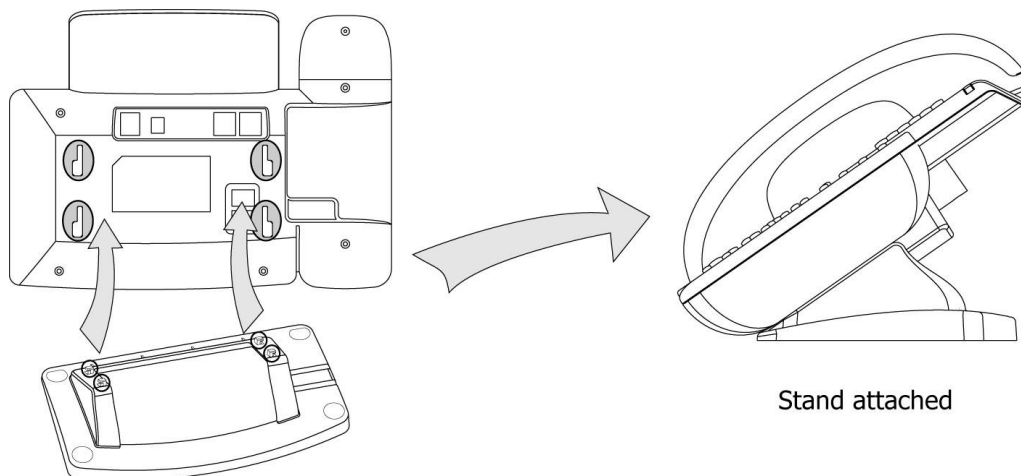
Check this list before installation to ensure that you have received each item. If you are missing any items, contact your IP phone reseller.

Assembling the Phone

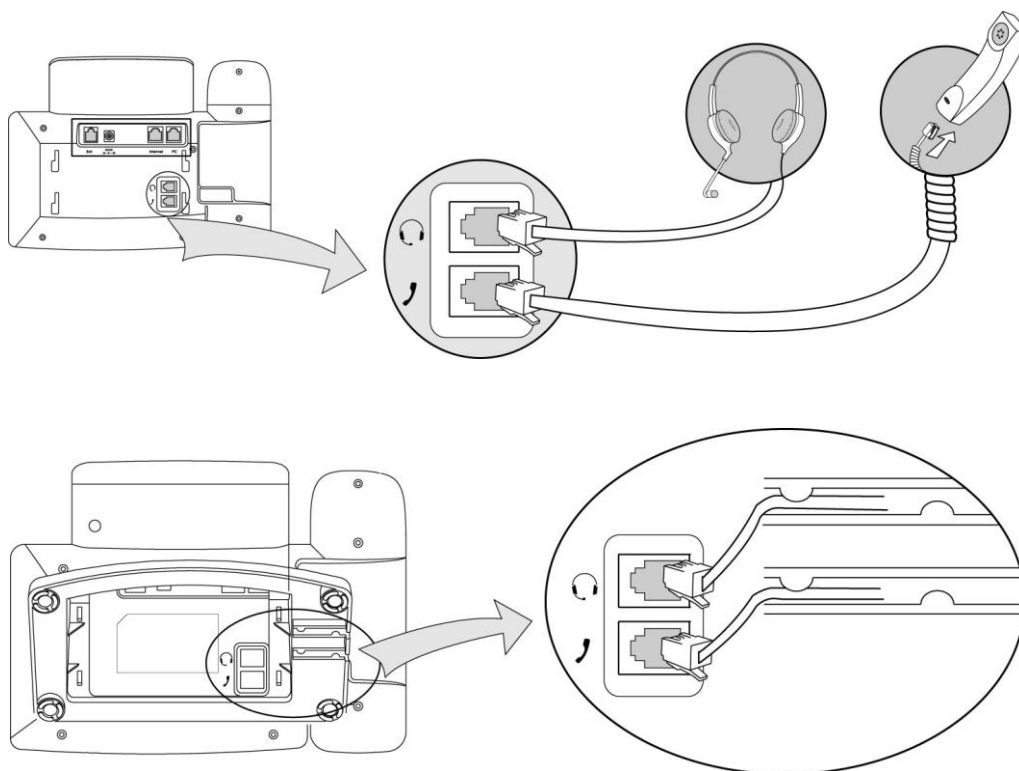
This section introduces how to assemble the phone with the components in the packing list :

- Attach the stand;
- Connect Handset and Headset;
- Connect Network and Power.

1) Attach the Stand, as shown below:



2) Connect Handset and Headset, as shown below:

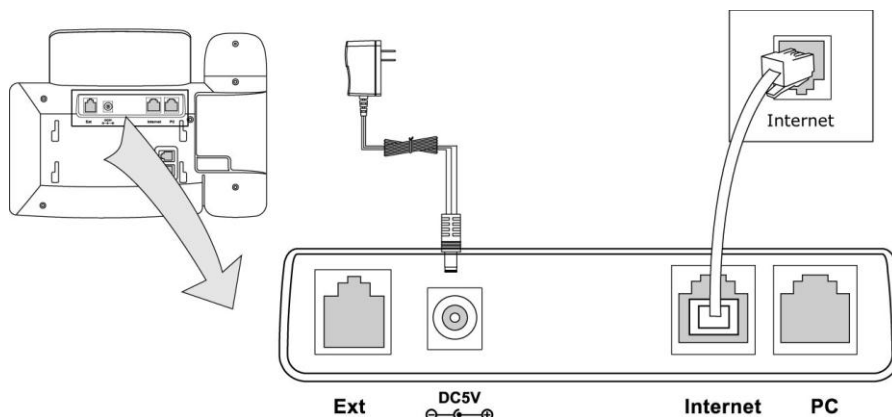


Note:

Headset is not provided in the packing list. Please contact your distributor for more information.

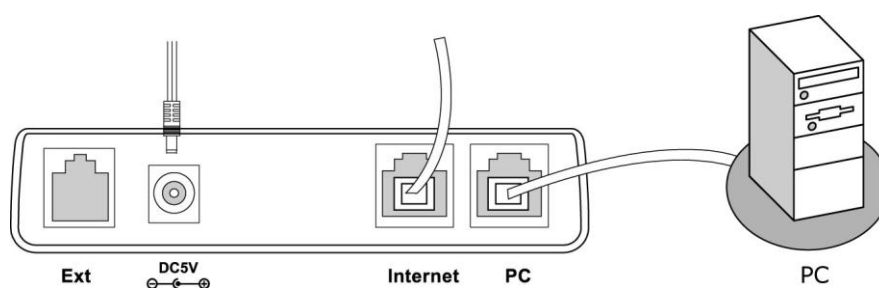
3) Connect Network and Power

There are two ways for network and power source connections. You can either connect the phone to the AC Power directly using the power adapter or to a PoE compliant switch or hub. Your system administrator will advise you on which one to use.

**Note:**

1. If inline power is provided, do not install AC adapter. Make sure the Ethernet cable and switch/hub is PoE compliant.
2. The Internet Port can be also connected to Hub/Switch/IP PBX or other internet devices.

The phone can also share the network connection with other network devices such as PC. Connect the phone's PC port and computer's Network Port together using an Ethernet cable, shown as below:



Configuration and Registration

If you are administrator, you need to do some simple configuration to make the phone work. If not, please contact your system administrator or service provider for more details.

Configuring via Web Page

Press **OK** button on the keypad of the phone to enter the status page and find out the IP address of IP phone. Enter it (for example http://192.168.3.35) into the address bar of web browser. The default administrator's login name and password are **admin/admin**. The default user's login name and password are **user/user**.

Note:

Please locate your PC in the same network segment of IP phone (192.168.3.X) to access the web configuration page. Please consult your system administrator for help.

Network Settings

Click on **Network-> Internet Port (WAN)**

DHCP: Under the default situation the phone attempts to connect a DHCP Server in your network in order to obtain its valid network settings, e.g. IP address, sub mask, gateway, DNS server, etc.

Static IP Address: If your phone cannot contact a DHCP Server for any reason, you need to enter the network settings manually via Static IP Address. Please contact your internet administrator for more details.

PPPoE: If you are using the xDSL Modem, you can connect your phone to the internet via PPPoE mode. Please contact your ITSP for the **User Name** and **Password** for internet access.

Note:

Using the wrong network parameters may result in inaccessibility of your phone and may also have an impact on your network performance. Please contact your system administrator.

Account Settings

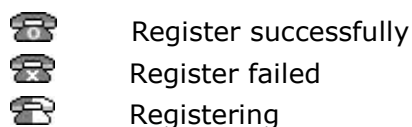
The phone attempts to register the SIP server using the account data produced by the automatic or manual initialization.

Click on **Account**, you will find the following parameters:

Field	Description
<i>Register Status</i>	It shows the register status of the phone.
<i>Account Active</i>	You can choose on/off to enable/disable the account respectively.
<i>Label</i>	The name showing on the LCD of current device.
<i>Display Name</i>	The local phone name showing on the other phone when calling.
Field	Description
<i>Register Name</i>	Register name provided by ITSP.
<i>User Name</i>	User account information, provided by ITSP.

<i>Password</i>	Account password provided by ITSP.
<i>SIP Server</i>	SIP server address provided by ITSP.

When you have finished the Network and Account Setting configuration, the Register Status Icons will show in the idle screen:



When all accounts register fail, phone will display "No Service" by default.

When the phone reboot, it will register automatically. If many phones register at the same time, this will affect the server, the users can set the register power up time so that the phone will random register automatically within the set time.

Setting the power up time via web interface:

Click on **Network->Advanced->Registration random**, enter the time in the field.

Note:

Should the IP PBX (SIP registrar) require an authentication, you will be prompted to enter the correct password. Make sure you are using the appropriate input method or enter the password via the web interface.

Configuring via keypad

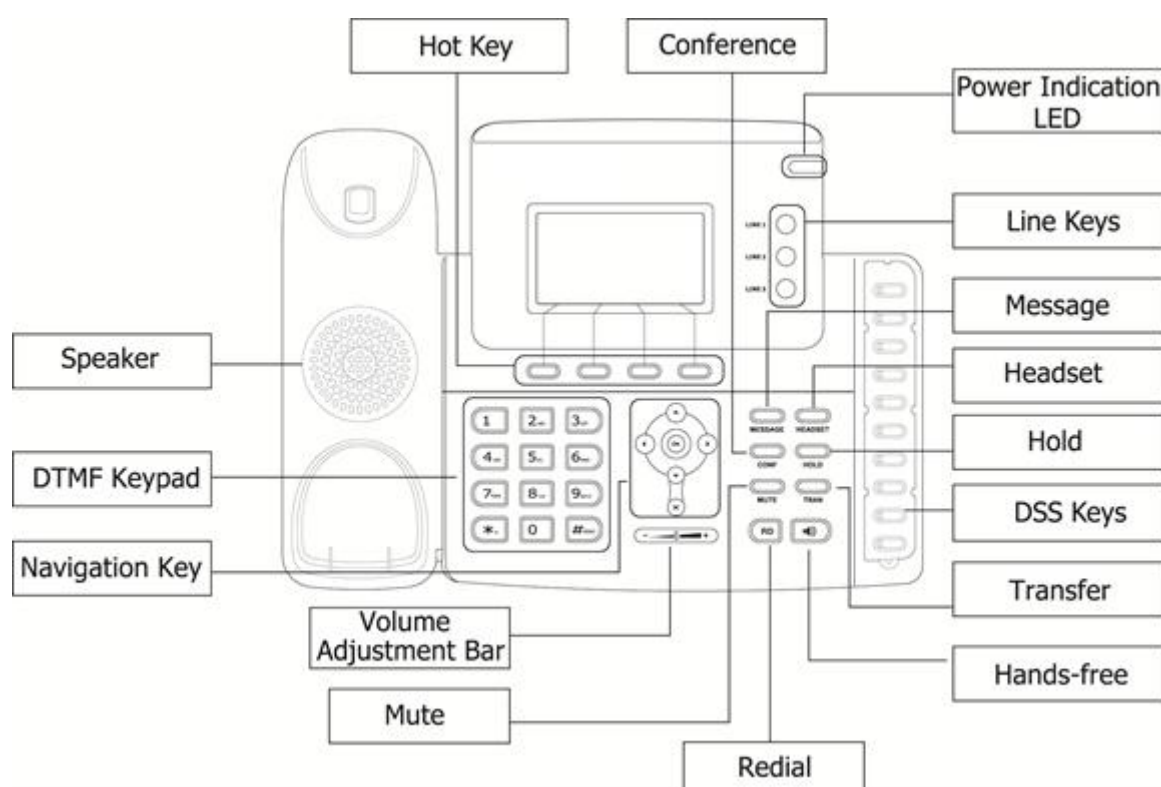
Network Settings: Press **Menu->Settings->Advanced Settings**, enter the password, and select **Network ->WAN Port /PC Port /VLAN/Webserver Type/802.1x Settings/VPN** Option to enter the internet relating configuration page.

Account Settings: Press **Menu->Settings->Advanced Settings**, enter the password, and select **Accounts** to configure the account settings.

You can refer to the above "Configuring via Web Page" for the parameter detail.

Overview

Keypad Instruction



You can check the following list which introduces the IP phone's keypad in details:

Power Indication LED

It will show the power status, it will be on if the phone is powered, off if the phone is not powered, and blink when someone calls in or there is a call on mute.

Hot Keys

The screen will display labels for these keys, to identify their context-sensitive functions and you can custom hot keys under different status.

Line Keys

These buttons are used to active up to the three user accounts.

DSS Keys


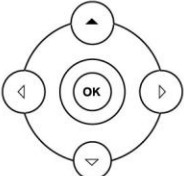
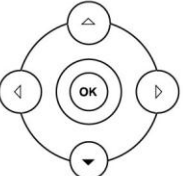
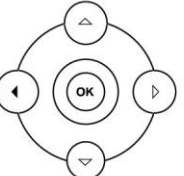
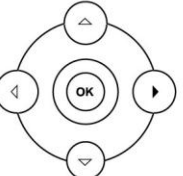

These keys are used for various functionalities such as call /Line appearance Button, Speed dial, Intercom, Pickup, Hold, Transfer, etc. When the assigned function is enabled, the corresponding LED will be on.

DTMF Keypad

Use the DTMF hard keys to enter numbers, letters and special characters. Depending on the selected input mode, you can enter digits, lower / upper case or special characters.


Navigation Keys


Use the navigation keys to navigate in the display menus and confirm and cancel actions.


Cancel	Up	Down	Left	Right	Confirm
					

Audio Device Control Keys

Use the audio device control keys to perform the following actions depending on your phone type:


 : Adjust the volume of the handset, headset, speaker , ring tone and signal tone;


 : Allows hands-free communication during calls;
Press it again to switch to the Group Listening mode.


 : Place and receive calls through an optionally connected headset. The LED will be on when the phone is in Headset mode;

 : Mute audio transmission locally during calls;

Hard Feature Keys

 : Allow users to access the voicemail interface directly;

 : Enable a setup of conference.

 : Place a call on hold or resume it;

 : During a call, press it to transfer the current call to the third party;

When the phone is in the idle status, press it to enter the forward configuration page.



: Press it to enter the Dialed Calls interface and select a record to dial out.

LED Instruction

Table 1 DSS Keys configured as BLF

LED Status	Description
Steady green	The monitored account is active
Blinking red	There is an incoming call to the monitored account
Steady red	The monitored account is on a conversation
Off	It is not active as BLF

Table 2 DSS Keys configured as BLA (Bridged Line Appearances)

LED Status	Description
Steady green	All of the members are in idle status
Steady red	Some part(s) is seizing the line
Blinking green 300ms	Some part(s) is ring-back
Blinking red 300ms	Some part(s) is ringing
Steady orange	Some part(s) is on the phone
Blinking Orange 500ms	Some part(s) is under the public hold status, and all of the members can retrieve the call
Blinking green 500ms	Some part(s) is under the private hold status, and only the initiator can retrieve the call
Blinking red 500ms	Three way conference, all of the parts press hold
Off	It is not active as BLA

Table 3 Line Keys configured as BLF

LED Status	Description
Steady green	The monitored account is in idle status
Fast Blinking green	There is an incoming call to the monitored account
Slow blinking green	The monitored account is on an conversation
Off	It is not active as BLF

Table 4 Line Keys configured as BLA

LED Status	Description
Steady green	All of the members are in idle status
Slow blinking green	Some part(s) is seizing the line/ ring-back/ under the private hold status
Fast blinking green	Some part(s) is ringing/on the phone / under the public hold status or all of the parts press hold
Off	It is not active as BLA

Table 5 Line Keys

LED Status	Description
Steady green	The account is active





















Blinking green	There is an incoming call to the account, or there is a call on hold
Off	The phone is in idle status whatever registered /unregistered








Table 6 Power Indication LED

LED Status	Description
Steady green	Power on
Blinking green	There is an incoming call, or there is a call on mute
Off	Power off

Icon Instruction

The IP Phone displays different kinds of icons on its LCD, you can refer to the following table for their meanings:

Icon	Description
	Flashes when the internet is disconnected
	Account register failed
	Account registering
	Account register successful
	Missed calls
	Call in
	Call out
	Input Method: all letters and numbers
	Input Method: numbers
	Input Method: multi-lingual letters in lower case
	Input Method: multi-lingual letters in upper case
	Call mute
	Call hold
	Voicemail
	SMS
	Call forward
	DND(Do Not Disturb)
	Auto answer
Icon	Description
	In handset mode
	In headset mode

	In speaker mode
	Ring volume is 0
	The recording feature cannot be started
	The recording cannot be stopped
	The recording box is full
	This call cannot be recorded
	The recording session is successfully started

User Interface

There are two ways to customize specific options on your phones:

1. Using keypad and display on the phone;
2. Using Web user interface in an Internet browser from your PC; please refer to "Configuration and Registration" to get into the Web interface.

In many instances, it is feasible to configure the phone via both Phone and Web interfaces; however, some features can only be configured via either Phone or Web interface. Please refer to the following table for differences:

Phone Options	Phone UI	Web UI
Status		
--IP		
--MAC		
--Firmware	√	√
--Network		
--Phone		
--Accounts		
Features		
--Call Forward	√	
--Call Waiting	√	
--DSS Keys	√	
--Key as Send	√	
--Hot Line	√	
--Anonymous Call	√	
--Auto Redial Settings	√	
--DND Code	√	√
--ReDialTone		
--Emergency		
--BusyToneDelay		
--Return code when refuse		
--Return code when DND		
--Intercom	√	
--Call Completion	√	
Basic Phone Functions		
--Language	√	√
--Time & Date	√	√
--Ring Tone	√	√
--Phone Volume	√	
--Logo Customization		√

Overview

Note:
The above table only indicates most of phone functions rather than all of them.
Please refer to the relating parts for more details.

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Customizing Your Phone

General Settings

Phone Status

You can view the status of your phone using the Phone interface or the Web interface.

This option allows you to review:

- Network status: IP Address, Mac Address, Gateway, DNS, WAN, LAN, etc;
- Phone status: Model, Hardware version, Firmware version, Product ID, MAC, etc;
- Accounts: The 3 SIP accounts status;

To check the Phone Status via Phone interface:

- 1) Press **OK** button directly or **Menu** hot key, choose the **Status** option.
- 2) Use the **navigation keys** to scroll through the list and check the specific one.

To view the Phone Status via Web interface:

Open the web browsers and input the IP Address <http://WAN-ip-address>; Enter the account and password (default account and password are both "admin"), Click **Status** directly to check the status.

Language

The default Phone interface language is **English**. The Web interface language will depend on your computer Operation System. It will automatically match the language with your computer and browser.

It also supports Simplified Chinese, Traditional Chinese, French, German, Italian, Polish, Turkish, Portuguese, Spanish, etc. You can change the language for the phone user interface and the web user interface independently from each other.

Note:

All languages may not be available for selection. The available languages depend on the language packs currently loaded to the IP phone. Please contact with your system administrator for more information about loading language packs.

To change the language via Phone interface :

- 1) Press **Menu->Settings->Basic Settings->Language**.
- 2) Scroll through the list of available languages.
- 3)

Press the **Save** hot key when the desired language is highlighted. The language appears on the graphic display will be changed to the one you selected.



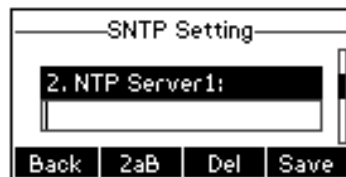
- 4) Press **Save** hot key to save the changes.
- 5) Press **Back** hot key to return to the previous screen.

Time and Date

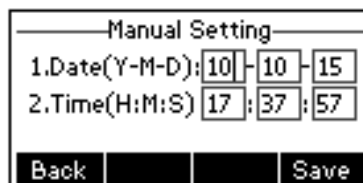
The time and date appears on the idle screen of the IP phone. If the phone cannot obtain a time and date from the SIP server, please contact your system administrator if the time or date is incorrect. You can set the time manually or via the SNTP server which is used to synchronize the time.

To change the Time and Date via the Phone interface:

- 1) Press **Menu->Settings->Basic Settings->Time & Date**.
- 2) If **SNTP Settings** is selected, the phone will automatically get the time from the specific NTP Server. Use the navigation keys to highlight the specific option and the relating changes. You can set the **Time Zone**, **NTP Server1/Server2**, and **Daylight Saving** respectively.



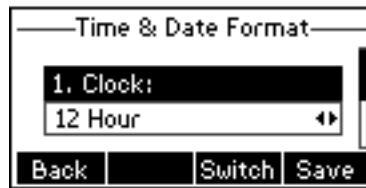
- 3) If **Manual Setting** is selected, the time can be set manually. Use the navigation keys to highlight the option and enter the specific date and time.



- 4) Press **Save** hot key, the time appears on the idle screen will be changed. Press **Back** hot key to return to the previous screen.

To set the time format via the Phone interface:

- 1) Press **Menu->Settings->Basic Settings->Time & Date->Time & Date Format**.



- 2) Use the **Switch** hot key to choose a preferred time format: 12 hour or 24 hour.
- 3) Use the **Switch** hot key to choose a preferred date format, the IP phone can support 7 kind of date display format.
- 4) Press the Save hot key to save the changes and return to the previous screen.

To configure the Time and Date via the Web interface:

Click on **Phone->Preference** to do the relating changes. You can also configure the **Update Interval** which specifies the time frequency that the phone refreshes the time automatically. Please refer to the instruction above for the parameters' detail.

Logout						
Status	Account	Network	Phone	Contacts	Upgrade	Security
Preference Features Softkey Layout DSS Key EXT Key Action URL Voice Ring Tones Dial Plan SMS						
WEB Language	English ?					
DHCP Time	Disabled ?					
Time Zone	+8 China(Beijing) ?					
Primary NTP Server	cn.pool.ntp.org ?					
Secondary NTP Server	cn.pool.ntp.org ?					
Update Interval(seconds)	1000 ?					
Daylight Saving Time	Automatic ?					
Fixed Type	<input type="checkbox"/> By Date <input type="checkbox"/> By Week					
StartTime	Month <input type="text"/> Day <input type="text"/> Hour <input type="text"/>					
EndTime	Month <input type="text"/> Day <input type="text"/> Hour <input type="text"/>					
Offset(minutes)	<input type="text"/>					
Manual Time	Disabled ?					
Time Format	24 Hour ?					
Date Format	WWW MMM DD ?					
Live Dialpad	Disabled ?					
Inter Digit Time(1~14)(seconds)	4 ?					
Flash Hook Time(<800ms)	1 ?					
Backlight Time(seconds)	30 ?					
Keyboard Lock	Disabled ?					
WatchDog	Enabled ?					

NOTE

Time Zone
Choose the time zone you are in.

NTP Server
The server which is used to synchronize the clock of the phone.

Update Interval
Specify the interval at which the unit will refresh the time.

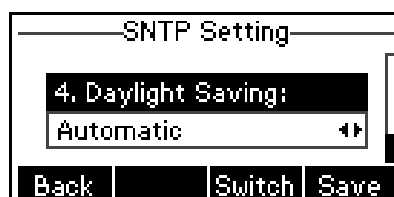
Daylight Saving Time
The parameter used to active the daylight saving time.

Manual Time
Enable or disable to set time manually.

Ring Tone
The upload ringtones must be format of wav whose sampling rate should be 8K, mono, 16-bit U-law compression

To configure the Daylight Saving Time settings via the Phone interface:

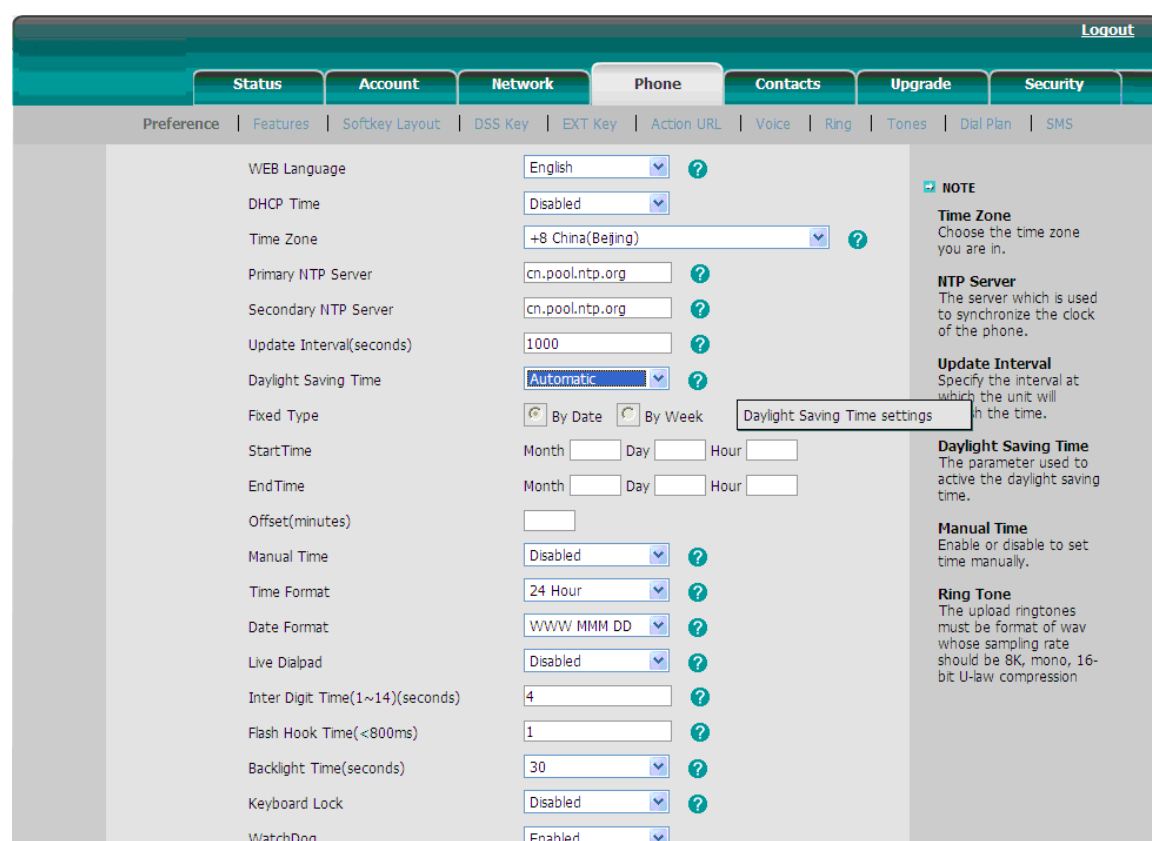
- 1) Press **Menu->Settings->Basic Settings->Time & Date->SNTP Settings->Daylight Saving**.



- 2) Press the **Switch** hot key to choose Off/On/Automatic options.
- 3) Press the **Save** hot key to save the changes.

To change the Daylight Saving Time Settings via the Web interface:

- 1) Click on **Phone->Preference->Daylight Saving Time** to do relating changes.
- 2) Select **Enable** option, then you can set the **Daylight Saving Time**.
- 3) Select **Automatic**. There is a table named as AutoDST.xml has been saved in the configuration file, if the table includes daylight saving time of your time zone, it will show the **Fixed Type: By Date** or **By Week**. And the daylight saving time is unchangeable, unless to update the AutoDST.xml via auto provision.



Note:




By default the time zone is +8 China (Beijing), Daylight Saving Time is Automatic.

Keypad Lock

You can lock the keypad of your phone when you are temporarily not using it. This function helps you to protect your phone from unauthorized use. You can lock the following specific keys:

- Menu Key:* The Menu hot keys cannot be used until unlocked. You cannot access the menu of the phone.
- Function Keys:* The hard function keys cannot be used until unlocked. You cannot access the MESSAGE, HEADSET, CONF, HOLD, MUTE, TRAN, RD Keys. History, Directory, DND, OK, X, navigation Keys, etc.
- All Keys:* All of the keys cannot be used until unlocked. You can only use the phone to answer the incoming calls.
- Lock&Answer:* All the incoming calls will be put through automatically (Auto Answer), but cannot be hung up by your party.

To enable keypad lock via Phone interface:

- 1) Press **Menu->Settings->Advanced Settings**, enter the password, and then press **Confirm** hot key.
- 2) Press **Phone Setting->Lock**.
- 3) Use the **navigation keys** (or press the **Switch** hot key) to highlight the one you want to lock.
- 4) Press **Save** hot key to active the change, or **Back** hot key to return to the previous screen.
- 5) If Keypad Lock is enabled, the icon  will be displayed on the top right corner of the idle screen.
- 6) If you choose **Lock&Answer**, it will show the icon  and  on the user interface.

To unlock the phone via Phone interface:

- 1) Press **Menu** hot key, you are prompted for the password.
- 2) Enter the password, and then press **Enter** hot key, the phone will be unlocked.
- 3) The icon will be disappearing from the idle screen.
- 4) If you select **Lock&Answer**, you have to enter **Menu->Settings->Advanced Settings**, enter the password, and then press **Phone Setting->Lock** to disable this option.

To enable keypad lock via Web interface:

Click on **Phone->Preference-> Keyboard Lock** to do the relating changes. Please refer to the instruction above for the parameters' detail.

The screenshot shows the 'Phone' tab with the 'Preference' sub-tab selected. The 'Keyboard Lock' option is set to 'Disabled'. A tooltip for 'Menu key' is visible, stating 'The menu button is not available'. Other settings include: WEB Language (English), DHCP Time (Disabled), Time Zone (+8 China(Beijing)), Primary NTP Server (cn.pool.ntp.org), Secondary NTP Server (cn.pool.ntp.org), Update Interval(seconds) (1000), Daylight Saving Time (Automatic), Fixed Type (By Date), Start Time (Month, Day, Hour), End Time (Month, Day, Hour), Offset(minutes), Manual Time (Disabled), Time Format (24 Hour), Date Format (WWW MMM DD), Live Dialpad (Disabled), Inter Digit Time(1~14)(seconds) (4), Flash Hook Time(<800ms) (1), Backlight Time(seconds) (30), and WatchDog (Enabled).

Note:

The default password for unlock is **admin**.

Audio Settings

Volume

You can adjust the volume of handset/speaker/headset/Ring.

To adjust the volume when you are not in an active call:

- 1) Press **Menu->Settings->Basic Settings**.
- 2) Scroll to Phone Volume, and press **Enter** hot key, highlight the one you want to adjust the volume, use the **Volume Adjustment Bar** or **navigation keys** to adjust the volume.

The screenshot shows the 'Phone Volume' menu with three options: 1. Handset Volume, 2. Speaker Volume, and 3. Headset Volume. The 'Handset Volume' option is highlighted. At the bottom, there are 'Back' and 'Enter' buttons.


- 3) Press **OK** hot key to save the change or **Back** hot key to cancel.

- 4) And you can also press the **Volume Adjustment Bar** to adjust the ring volume when the phone is in idle status.

To adjust the volume when you are in an active call:

When Handset/Headset/Hands-free mode is activated, press the **Volume Adjustment Bar** to a comfortable level.

Note:

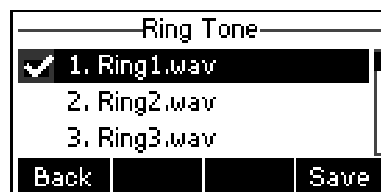
The volume can only be adjusted via Phone interface. When you adjust the ring volume to 0, or press the **Silence** hot key when there is incoming call, the icon  will be shown on the LCD. Press the **Volume Adjustment Bar** to adjust the volume, and the icon will disappear.

Ring Tones

You can adjust the type and volume of the ring tone.

To adjust the Ring Tone Type via Phone interface:

- 1) Press **Menu->Settings->Basic Settings**.
- 2) Scroll to **Ring Tone**, and press **Enter** hot key.
- 3) Use the **navigation keys** to highlight the specific one.



- 4) Press **Save** hot key to save the changes or **Back** hot key to cancel.

To change the Ring Tone Type via Web interface:

Click on **Phone->Preference->Ring Type**, highlight the specific one in the scroll-down menu, click **Confirm** button to save the changes. You can also delete the specific one by click the **Del** button.

The screenshot displays the 'Phone Ringing type settings' web interface. The main configuration area includes the following settings:

- Time Zone: +8 China(Beijing)
- Primary NTP Server: cn.pool.ntp.org
- Secondary NTP Server: cn.pool.ntp.org
- Update Interval(seconds): 1000
- Daylight Saving Time: Automatic
- Fixed Type: ☒ By Date ☐ By Week
- StartTime: Month [] Day [] Hour []
- EndTime: Month [] Day [] Hour []
- Offset(minutes): []
- Manual Time: Disabled
- Time Format: 24 Hour
- Date Format: WWW MMM DD
- Live Dialpad: Disabled
- Inter Digit Time(1~14)(seconds): 4
- Flash Hook Time(<800ms): 1
- Backlight Time(seconds): 30
- Keyboard Lock: Function Keys
- WatchDog: Enabled
- Ring Type: Ring1.wav
- Upload Ringtone: []

Buttons at the bottom include 'Confirm', 'Cancel', 'Del', 'Upload', and 'Cancel'. A 'Phone Ringing type settings' dialog box is visible on the right side of the interface.

Note:

The ring tone file of system cannot be deleted.

To upload the new Ring Tone via Web interface:

- 1) Click on **Phone->Preference->Upload Ringtone**.
- 2) Click on **Browse** button to choose the specific ring tone file.
- 3) Click on **Upload** button to upload the file.

Note:

The ring tone file format must be in 16bits WAV format (via Ulaw Compression), 8K sample rate (monophony). Blank or other special characters cannot be included in the file name.

To specify ring tones for a specific account via Web interface:

Click on **Account->Basic->Ring Type** option, and highlight the preferred one for the chosen account in the scroll-down menu, then click **Confirm** button to save the changes.

Register Status	Registered	
Account Active	<input checked="" type="radio"/> On <input type="radio"/> Off	
Label	55555	?
Display Name	55555	?
Register Name	55555	?
User Name	55555	?
Password	•••••	?
SIP Server	192.168.1.10	Port 5060 ?
Enable Outbound Proxy Server	Disabled	?
Outbound Proxy Server		Port 5060 ?
Transport	UDP	?
Backup Outbound Proxy Server		Port 5060 ?
NAT Traversal	Disabled	?
STUN Server		Port 3478 ?
Voice Mail		?
Proxy Require		?
Anonymous Call	Off	?
On Code		?
Off Code		?
Anonymous Call Rejection	Off	?
On Code		?
Off Code		?
Missed call log	Enabled	?
Auto Answer	Disabled	?
Ring Type	common	?
Codecs >>	<ul style="list-style-type: none"> common Ring1.wav Ring2.wav Ring3.wav Ring4.wav Ring5.wav 	
Advanced >>		

display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.

Advanced
The Advanced parameters for administrator.

Codec Selection

The IP phone supports the following voice codecs: G723_63, G722, G726-16, G726-24, G726-32, G726-40, PCMA, G729, PCMU and G723_53

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

To enable/disable the codecs:

- 1) Click on **Account->Codecs**.

- 2) Use the **navigation keys** to highlight the desired one in the **Enable/Disable** Codecs list, and press the **>>** / **<<** to move to the other list.
- 3) Click **Confirm** to save the change.

Note:

1. Codec selection can only be set via Web interface.
2. If codec G722 is negotiated, the LCD screen prompts call in process with HD voice.

Contact Management

Edit/Add/Delete Contact

You can store a large number of contacts in your phone's directory. You can add, edit, delete, dial, and search for a contact in the contact list.

The directory includes Local Directory, Blacklist, Remote Phonebook and Broadsoft.

To add a Group via Phone interface:

- 1) Press **Dir->Local Directory**
- 2) Press **Group** hot key to enter to the Add Group page.
- 3) Enter the group name and choose the ring.

- 4) Press the **save** hot key to save.



To add a contact via Phone interface:

- 1) Press **Dir->Local Directory**
- 2) Select a group, and press the **Enter** hot key.
- 3) Press **Add** hot key; enter Name, Office/Mobile number, Account and other information of the contact from the keypad. Use the **123** hot key to select between numeric and upper/lower case alphanumeric modes.



- 4) Use the **navigation keys** to select the desired account as Line if you want to assign the contact to a specific account.
- 5) Select and set a special ring tone for the contact.
- 6) Use the navigation keys to select the group which you want to assign.
- 7) Press **Save** hot key to add the contacts or **Back** hot key to cancel the change.

To edit a contact via Phone interface:

- 1) Press **Dir->Local Directory**
- 2) Select a group, and press the **Enter** hot key.
- 3) Use the **navigation key** to highlight the one you want to edit, press **Option->Detail**, Enter to the edit interface.
- 4) And then make the changes, press **Save** hot key to save the changes, or press **Back** hot key to return to the directory.



To delete a contact via Phone interface:

- 1) Press **Dir->Local Directory**
- 2) Select a group, and press the **Enter** hot key.
- 3) Use the **navigation keys** to highlight the one you want to delete, press **Option** hot key, and scroll to **Delete**, press **OK** hot key.
- 4) It will pop up a warning frame asking whether confirm to delete the contact.

- 5) Press **OK** hot key to confirm the operation, or press the **Cancel** hot key to return to the directory.



To move a contact to the Blacklist via Phone interface:

- 1) Press **Dir->Local Directory**
- 2) **Select** a group, and press the **Enter** hot key.
- 3) Use the **navigation keys** to highlight the one you want to move, press **Option** hot key, and scroll to **Move to Blacklist**, press **OK** hot key.
- 4) It will pop up a warning frame asking whether confirm to move the contact.
- 5) Press **OK** hot key to confirm the operation, or press the **Cancel** hot key to return to the directory



Note:

If a contact is moved to the blacklist, then the call from this contact cannot get through.

To move a contact in History to Contacts via Phone interface:

- 1) Press **History** hot key to enter the call history list.
- 2) Use the **navigation keys** to highlight a record, and then press the **Option** hot key to pop up the field, highlight **Add to Contacts** option, then press the **OK** hot key to enter the edit interface.
- 3) Press the **abc** hot key to switch the input mode.
- 4) After the edition, press the **Save** hot key to save the change. Then you can go to **Contacts** interface to check the record.

To search a contact via Phone interface:

- 1) Press **Dir->Local Directory->Search**.
- 2) It will turn to the **Search** interface, and then you can enter the query condition, press the **OK** hot key.

3) Then the phone will show the record which qualified.

Search Contact

Search:

Cancel
abc
Delete
Ok

To add/delete/edit/move the contacts via the Web interface:

Click on **Contacts->Local Phone Book**. Please refer to the instruction above for the parameters' detail.

[Logout](#)

Status
Account
Network
Phone
Contacts
Upgrade
Security

Local PhoneBook
BlackList
Remote PhoneBook
Phone Call Info
LDAP
Broadsoft
Call Log

Contacts
All Contacts

Index	Name	Office Num	Mobile Num	Other Num	Account	Groups
1	12345	326	325	552	Auto	
2	361	361	3122	1245	Account2	
3	?				Auto	
4	Andy	1010			Auto	
5	Jakey	3030			Auto	
6	Kobe				Auto	
7	Peter	1000			Auto	
8	Tom	2525			Auto	
9						
10						

Page: 1 Prev Next Move To BlackList Delete All Del

NOTE
Add Contact/Blacklist
 Fill out the contact information. User shouldn't leave contact name blank.

Delete Contact/Blacklist
 Select the contact you want to delete in the grid, and then press the button Delete to confirm.

Move to Contact/Blacklist
 Choose the contacts you want to move in the grid, and press the button move to Contact/Blacklist to move it.

Import
 Browse the file in XML format.

Export
 Click Export button and create a file with whose name you prefer to export.

Contacts

Name:
 Office Num:
 Mobile Num:
 Other Num:
 Account: Auto
 Ring: Auto
 Groups: N/A

Add
Edit
Search

Group Information

Groups:
 Ring: Auto

Add
Edit
Del
Delete All

Please select the contacts list file

浏览...

Import XML
Export XML

浏览...

Import CSV
Export CSV


☒ Show title

Import/Export Contact list

Import/Export Contact List via Web interface:

- 1) Click on **Contacts->Local Phone Book**.

The screenshot displays the 'Local PhoneBook' interface. At the top, there are tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. The 'Contacts' tab is active, showing a sub-tab for 'Local PhoneBook'. Below this, there's a table of contacts with columns: Index, Name, Office Num, Mobile Num, Other Num, Account, and Groups. The table lists 10 contacts. To the right of the table, there are icons for adding, deleting, and moving contacts. Below the table, there are input fields for Name, Office Num, Mobile Num, Other Num, Account, Ring, and Groups, along with Add, Edit, and Search buttons. On the far right, there's a 'NOTE' section with instructions for adding, deleting, and moving contacts, and an 'Import/Export' section with buttons for Import XML, Export XML, Import CSV, and Export CSV.

- 2) Browse the specific contact list file in .XML format or .csv format, and then click **Import** button. The imported contact lists will be showed in the Contact List.
- 3) Move the mouse to the icon , you will see the notes for parameters. The meanings of this icon on other pages is the same, we will not elaborate it one by one.
- 4) Click the **Export** button to export the contact list.

Note:

Import/Export Contact List can be only set via Web interface.

Blacklist

If you add a contact to blacklist, then the call from this contact cannot get through.

To add contacts to Blacklist via the Web interface:

- 1) Click on **Contacts->Blacklist** to enter the Blacklist interface.
- 2) Enter **Name, Office Num, Mobile Number, Other Num** of the contact and select an account.
- 3) Click **Add** key to add the contact to the blacklist.

You can refer to the configuration of Contact for more information.

The screenshot shows the 'BlackList' management page. At the top, there are tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. Below these are sub-tabs: Local PhoneBook, BlackList (selected), Remote PhoneBook, Phone Call Info, LDAP, Broadsoft, and Call Log. The main content area displays a table with columns: Index, Name, Office Num, Mobile Num, Other Num, and Account. The first row shows Index 1, Name AA, Office Num 0599102, Mobile Num 0599123, and Account Auto. Below the table are navigation buttons: Prev, Next, Move To Contacts, Delete All, and Del. At the bottom, there are input fields for Name, Office Num, Mobile Num, Other Num, and a dropdown for Account (set to Auto), along with Add, Edit, and Search buttons. A 'Logout' link is in the top right corner. A 'NOTE' on the right side says 'blacklistnote'.

Remote Phone Book

The IP phone has directory itself, but in the enterprise applications where there are a need for a common phone book. For the maintenance and the update of it, the common phone book is usually carried out on the server or IPPBX to maintain up-to-date public phone book, terminal users need to have remote phone book function. When the users browse the remote phone book, the terminal will check and download the latest information released on the server in time, and display on the terminal for the user.

To set the Remote Phone Book via Web interface:

- 1) Click on **Contacts->Remote Phone Book**.

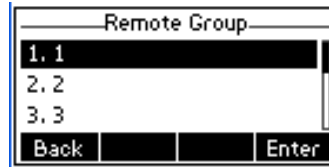
The screenshot shows the 'Remote Phone Book' configuration page. At the top, there are tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. Below these are sub-tabs: Local PhoneBook, BlackList, Remote PhoneBook (selected), Phone Call Info, LDAP, Broadsoft, and Call Log. The main content area displays a table with columns: Index, Phone Book Url, and Phone Book Name. The first row shows Index 1, Phone Book Url HTTP://123....., and Phone Book Name ABC. Below the table are 'Confirm' and 'Cancel' buttons. At the bottom, there are input fields for Name, Office Num, Mobile Num, Other Num, and a dropdown for Account (set to Auto), along with Add, Edit, and Search buttons. A 'Logout' link is in the top right corner. A 'NOTE' on the right side says 'Remote phone book' and explains that this feature allows users to download contact lists from the server by inputting the phonebook URL and renaming the phonebook.

- 2) Input the **Phone book URL** and the **phone book name**, then click **Confirm**

button to save the changes.

To check the Remote Phone Book via Phone interface:

- 1) Press the **Dir->Remote phonebook**.
- 2) Enter to the **Remote Group** page, choose a special one, and press the **Enter** hot key, it will go to the corresponding URL address to download the contact information for you.



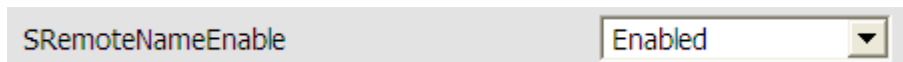
Note:

1. This IP phone can support 5 remote phone books at most.
2. Every contact in the remote phone book can set several phone numbers.

In addition, you can enable or disable the search remote phonebook for caller ID feature via web interface. If it's enabled, the phone displays the name of the caller stored in Remote Phone Book when there is an incoming call.

To enable search remote phone book via Web interface:

- 1) Click on **Phone -> Features**
- 2) Select **Enabled** in the pull-down menu of **SRemoteNameEnable**



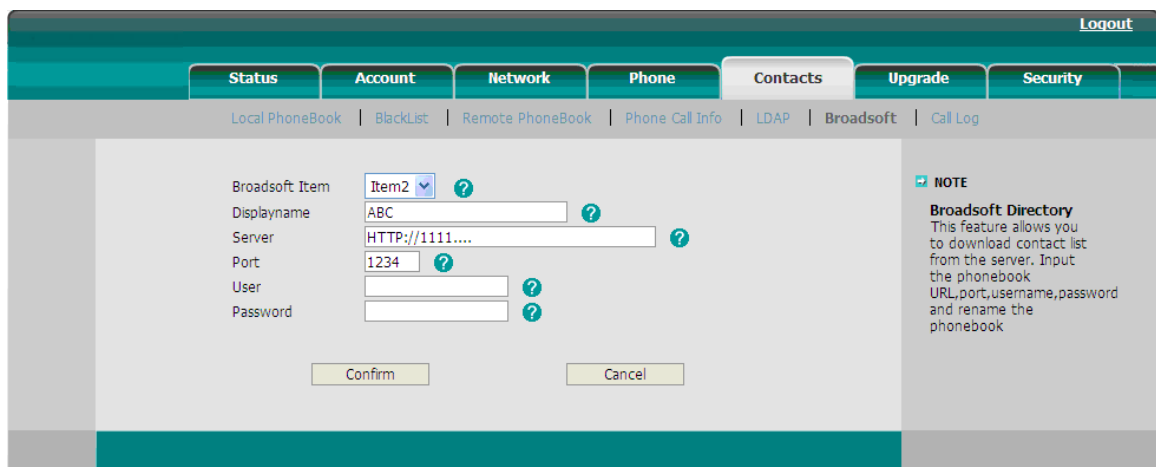
- 3) Click **Confirm** to save the configuration.

Broadsoft

Broadsoft phone book is the same as the Remote Phone Book. This feature allows you to download contact list from the server.

Configure Broadsoft via web interface:

Click on **Contacts->Broadsoft**, Select the **Broadsoft** Item, and then enter the **Server, port, username, password** and rename the phonebook.



Configure Broadsoft via phone interface:

- 1) Press **Menu->Features->Broadsoft Settings Menu->Broadsoft Dir Settings**

- 2) Select **Broadsoft** Item, enter the **Display Name, Server, Port, User Name** and **Password**.
- 3) Press the **Save** hot key to save the changes.

LDAP

LDAP can support these functions:

1. Search the contact: Press the DSS key which is set up as LDAP, enter a number or letter in the new interface, the phone will search the contact in LDAP server which following the Certain rules, and show it in the LCD, user can choose the contact to call out.
2. Search the incoming call: the phone will search the local directory when there is a coming call. If they can't find the contact in the local directory, it will search them through LDAP server, and show the contact name in the LCD. The LDAP Lookup For Incoming Call option can be configured to enable or disable this function via web interface.
3. The function of Dial-up directory: under the dial interface, each time you press a key there are inquiring for a number. It will show on the LCD and let the user to choose. The LDAP Lookup For PreDial/Dial option can be configured to enable or disable this function via web interface.

To set the LDAP via the Web interface:

- 1) Click on **Contacts->LDAP**.
- 2) Configure the corresponding options.

3) Click **Confirm** to save the change.

Other Settings

Key as Send

Users can set a specific button ("#" or "*") to active as the send button.

To set the send key via the IP phone interface:

1) Press **Menu->Features->Key as Send->Enter** to enter the configuration page.

2) Press the **Switch** hot key to choose a button that you want to use as the send key: "#", "*", or disable this option.

3) Press the **Save** hot key to save the changes.

To set the send key via the Web interface:

1) Click on **Phone->Features->Key As Send**.

- Highlight the specific one in the pull-down menu, then click **Confirm** button to save the changes.

Hot Line

To set the hot line number via the IP phone interface:

- Press **Menu->Features->Hot Line->Enter** to enter the configuration page.

- Enter the hot line **Number** and **HotLine Delay** time (for example, 20 seconds), then press the **Save** hot key to save the changes.
- When you pick up the handset or press the speaker button, and it will dial out the number automatically if you do not press any keys for 20 seconds.

To set the Hot Line via the Web interface:

- Click on **Phone-> Features**.

- 2) Enter the Hotline **Number** and **Hotline Delay**, and then click Confirm button to save the changes.

On Code	<input type="text"/>	?	<p>to forward an incoming call to another phone number.</p> <p>Target The number to which the incoming calls will be forwarded.</p> <p>On Code The code that will be sent to PBX when it is switched On.</p> <p>Off Code The code that will be sent to PBX when it is switched Off.</p> <p>Call Waiting This call feature allows your phone to accept other incoming calls during the conversation.</p> <p>Key As Send Select * or # as the send key.</p> <p>Hotline Number When you pick up the phone, it will dial out the hotline number automatically.</p> <p>Upload Logo The picture must be format of dob, it can be black and white, or 2 gray scale.</p>
Off Code	<input type="text"/>	?	
Busy	<input type="radio"/> On <input checked="" type="radio"/> Off	?	
Target	<input type="text"/>	?	
On Code	<input type="text"/>	?	
Off Code	<input type="text"/>	?	
No Answer	<input type="radio"/> On <input checked="" type="radio"/> Off	?	
After Ring Time(seconds)	<input type="text" value="10"/>	?	
Target	<input type="text"/>	?	
On Code	<input type="text"/>	?	
Off Code	<input type="text"/>	?	
General Information:			
Call Waiting	<input type="text" value="Enabled"/>	?	
Call Waiting Tone	<input type="text" value="Enabled"/>	?	
Auto redial	<input type="text" value="Disabled"/>	?	
Key As Send	<input type="text" value="#"/>	?	
Reserve # in User Name	<input type="text" value="Enabled"/>	?	
Button Sound	<input type="text" value="Enabled"/>	?	
Send Sound	<input type="text" value="Enabled"/>	?	
Hotline Number	<input type="text"/>	?	
Hotline Delay	<input type="text" value="4"/>	?	Set hotline number
ReDialTone	<input type="text"/>	?	
Emergency	<input type="text"/>	?	
BusyToneDelay(seconds)	<input type="text" value="0"/>	?	
Ringer Device for Headset	<input type="text" value="Use Speaker"/>	?	
Headset Send Volume (1~53)	<input type="text" value="29"/>		

Headset Prior

Headset prior allows the phone to use headset in priority.

To place a call using Headset Prior, enable the Headset Prior feature, physically connect your headset and press the HEADSET button to activate it for use. Press the desired numeric keys to place a call will now connect to the headset automatically. To receive a call using Headset Prior, enable Headset Prior feature, physically connect your headset and press the HEADSET button to activate it for use, press the **Line** key or the **Answer** soft key to receive a call will now connect to the headset automatically.

To enable Headset Prior via Web interface:

- 1) Click on **Phone -> Features**;
- 2) Select **Enabled** in the pull-down menu of **Headset Prior**;

Headset Prior	<input type="text" value="Enabled"/>
---------------	--------------------------------------

- 3) Click **Confirm** to save the configuration.

Note:

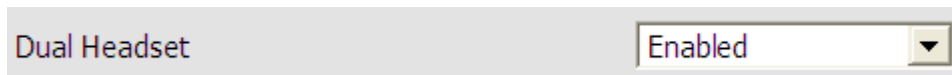
The headset icon on the LCD screen or a headset button in steady green indicates that the headset mode is activated.

Dual Headset

Dual headset allows users to use two headsets. To use this feature, you must connect your headsets to headset jack and handset jack respectively. Once the phone joins in a call, people with the Headset connected to the headset jack has a full-duplex conversation, another people with the headset connected to the handset jack is only allowed listening.

To enable Dual Headset via Web interface:

- 1) Click on **Phone -> Features**;
- 2) Select **Enabled** in the pull-down menu of **Headset Training**;

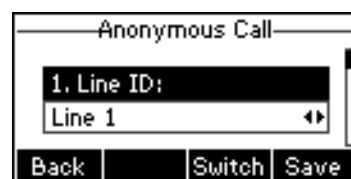


- 3) Click **Confirm** to save the configuration.

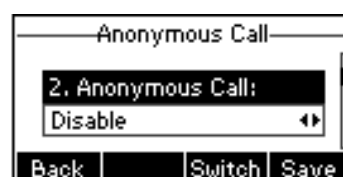
Anonymous call

To set the anonymous call via the IP phone interface:

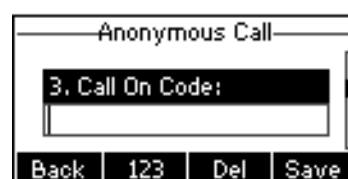
- 1) Press **Menu ->Features-> Anonymous Call ->Enter** to enter the configuration page.



- 2) Press the **Switch** hot key, you can choose the **Line ID** to enable this feature.
- 3) Press the **navigation keys** to enter and select whether to enable the anonymous call function. This feature allows making a call with the display of call identification information which has been blocked.

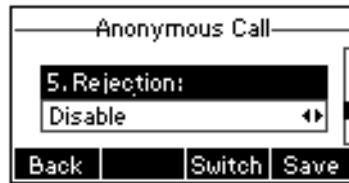


- 4) If you want to realize this function by server, please choose and enter the **Call On Code** and **Call Off Code**. When you choose to enable the **Anonymous call** function on your IP phone, it will send information to the server, and the server will enable/ disable the anonymous call function for your IP phone automatically.



- 5) Press the **navigation keys** to enter and select whether to enable the **Anonymous rejection** function. The feature allows rejecting all calls from

callers who have blocked the display of their calling identification information (calling number and calling name).



- 6) If you want to realize this function by server, please select and enter the **Reject On Code** and **Reject Off Code**. When you enable the anonymous reject function on your IP phone, it will send information to the server, and the server will enable/ disable the anonymous reject function for your IP phone automatically.
- 7) Press the **Save** hot key to save the changes.

Note:

This configuration is only available for the current default account.

To set the anonymous call via the Web interface:

- 1) Click on **Account-> Basic-> Anonymous Call** to do the relating changes. Please refer to the instruction above for the parameters' detail.
- 2) Then click the **Confirm** button to save the changes.

Account		Account 1
Basic >>		
Register Status	Registered	
Account Active	<input checked="" type="radio"/> On <input type="radio"/> Off	
Label	55555	?
Display Name	55555	?
Register Name	55555	?
User Name	55555	?
Password	?
SIP Server	192.168.1.10	Port 5060 ?
Enable Outbound Proxy Server	Disabled	?
Outbound Proxy Server		Port 5060 ?
Transport	UDP	?
Backup Outbound Proxy Server		Port 5060 ?
NAT Traversal	Disabled	?
STUN Server		Port 3478 ?
Voice Mail		?
Proxy Require		?
Anonymous Call	Off	?
On Code		?
Off Code		?
Anonymous Call Rejection	Off	?

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.

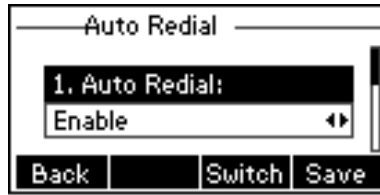
Advanced
The Advanced parameters for administrator.

Auto Redial

Auto redial is a telephone feature that redials a busy number in a fixed number of times before giving up.

To set auto redial via the IP phone interface:

- 1) Press **Menu->Features->Auto Redial ->Enter** to enter the configuration page.



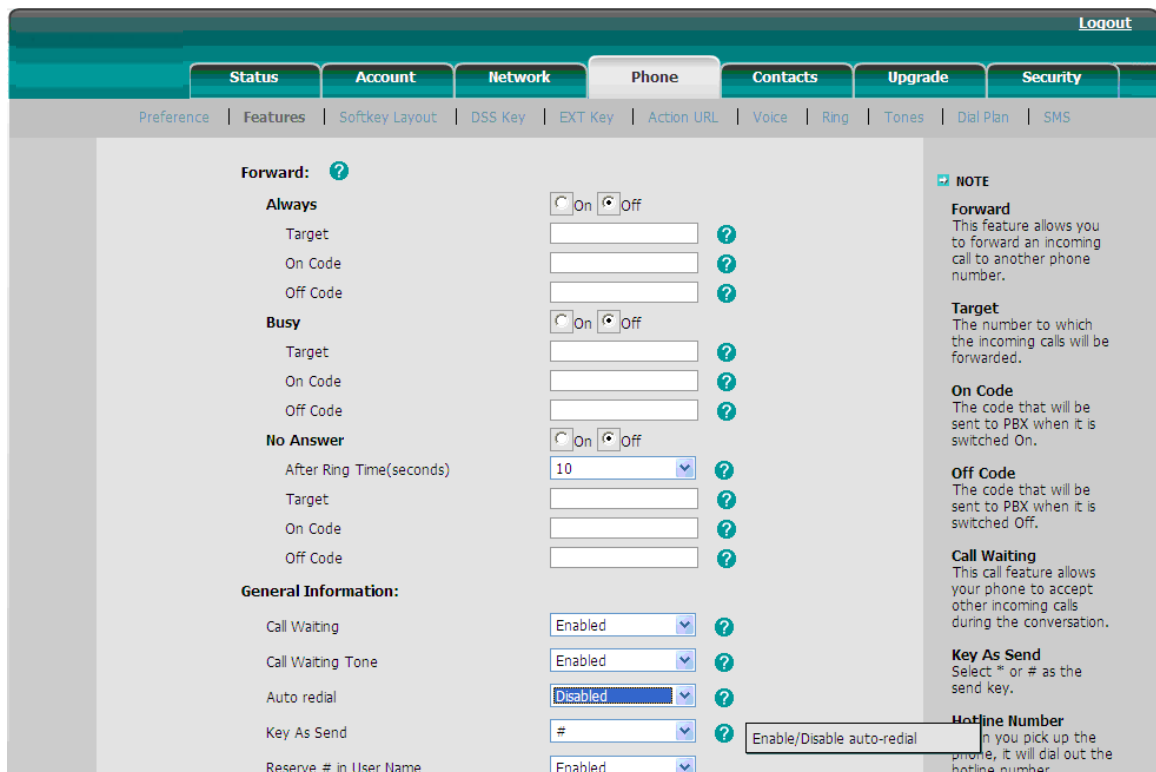
- 2) Press the **Switch** hot key, you can select whether to enable the **Auto redial** function.
- 3) Press the **navigation keys** to select and set the redial interval, It is measured by seconds.
- 4) Press the **navigation keys** to select and set the **redial times**.
- 5) Press the **Save** hot key to save the changes.

Note:

If you enable the auto redial function, after no operations for 5 seconds in the auto redial interface, it will turn to the idle interface automatically.

To configure auto redial via the Web interface:

- 1) Click on **Phone-> Features-> Auto Redial**.
- 2) Select **Enabled** or **Disabled** in the pull-down menu, then click **Confirm** button to save the change.



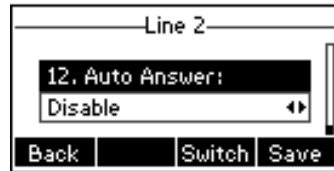
Auto Answer

Auto-answer allows an incoming call to be answered without requiring any action by the user. This is a useful feature for people who have difficulty in using their hands or fingers, who have a visual impairment, or who have a cognitive impairment. You can

set this function to a special account.

To set Auto Answer via the IP phone interface:

- a) Press **Menu->Settings->Advanced Settings**, enter the password and press **enter** hot key.
- b) Then press **Accounts->Line X** to enter the configuration page, use the **navigation keys** to select **Auto Answer** option.



- c) Press the **Switch** hot key to **enable** or **disable** the auto answer function. The default is Disable.
- d) Press the **Save** hot key to save the changes.

To configure Auto Answer via Web interface:

- 1) Click on **Account-> Basic-> Auto Answer**.
- 2) Select **Enabled** or **Disabled** in the pull-down menu, click **Confirm** button to save the change.

SIP Server	192.168.1.10	Port	5060	?
Enable Outbound Proxy Server	Disabled			?
Outbound Proxy Server		Port	5060	?
Transport	UDP			?
Backup Outbound Proxy Server		Port	5060	?
NAT Traversal	Disabled			?
STUN Server		Port	3478	?
Voice Mail				?
Proxy Require				?
Anonymous Call	Off			?
On Code				?
Off Code				?
Anonymous Call Rejection	Off			?
On Code				?
Off Code				?
Missed call log	Enabled			?
Auto Answer	Disabled			?
Ring Type	common			?

Enable/Disable auto-answer

Codecs >> ?
 Advanced >>

Confirm Cancel

Proxy Require
 A special parameter just for Nortel server. If you login to Nortel server, the value should be:
 com.nortelnetworks.firewall

Codecs
 Choose the codecs you want to use.

Advanced
 The Advanced parameters for administrator.

Missed call log

Defines whether to save the missed calls to the call history record This function can only be set via the Web interface:

- 1) Click on **Account-> Basic->Missed call log**.
- 2) Select **Enabled** or **Disabled** in the pull-down menu, click **Confirm** button to save the change.

Register Name	55555	?
User Name	55555	?
Password	?
SIP Server	192.168.1.10	Port 5060 ?
Enable Outbound Proxy Server	Disabled	?
Outbound Proxy Server		Port 5060 ?
Transport	UDP	?
Backup Outbound Proxy Server		Port 5060 ?
NAT Traversal	Disabled	?
STUN Server		Port 3478 ?
Voice Mail		?
Proxy Require		?
Anonymous Call	Off	?
On Code		?
Off Code		?
Anonymous Call Rejection	Off	?
On Code		?
Off Code		?
Missed call log	Enabled	?
Auto Answer	Disabled	?
Ring Type	common	?

[Codecs >>](#) ?

[Advanced >>](#)

Enable/Disable missed call records

User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be:
com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.

Advanced
The Advanced parameters for administrator.

Broadsoft Call Log

This feature allows you to download call history from the server. Enter the URL, port, username, password and rename the phonebook

Configure Call Log via web interface:

- 1) Click on **Contacts->Call Log**
- 2) Select the **Call log** Item.
- 3) Enter the **Display Name, URL, Port, Username** and **Password**.

4) Click the **confirm** button to save the changes.

Logout

Status Account Network Phone Contacts Upgrade Security

Local PhoneBook | BlackList | Remote PhoneBook | Phone Call Info | LDAP | Broadsoft | Call Log

Call Log Item: Call Log1 ?

DisplayName: miss call ?

Server: http://abc... ?

Port: 123 ?

User: 123 ?

Password: ... ?

Confirm Cancel

NOTE

Call log
This feature allows you to download call history from the server. Input the URL, port, username, password and rename the phonebook

To configure Broadsoft Call Log via phone interface:

- 1) Press **Menu->Features->Broadsoft Settings Menu->Call Log Settings**
- 2) Select **Call Log Item**, enter the **Display Name**, **Server**, **Port**, **User Name** and **Password**.
- 3) Press the **Save** soft key to save the changes.

To check Call Log via phone interface:

- 1) Press **Menu->History Type->Network Call Log**
- 2) Select the **Call log Item**, press **Enter** to download the call log.

Logo Customization

You can upload your own logo which would be shown in the idle screen.

Upload the logo via web interface:

- 1) Click on **Phone->Features->Use Logo->Custom Logo** via the Web interface, click on **Browse** button, and select the corresponding file.
- 2) Click **Upload** button to complete the logo customization. You will find the desired logo shown on the idle screen.

Intercom Barge	Enabled	?
Call Completion	Disabled	?
Enable Semi-Attend Transfer	Enabled	?
Blind Transfer OnHook	Enabled	
Attend Trans OnHook	Enabled	
Transfer on Conference Hang up	Disabled	
Feature Key Synchronisation	Disabled	
Time Out for Dial-now Rule	1	
ACD Auto Available	Disabled	
ACD Auto Available Timer(0~120s)	60	
RFC 2543 Hold	Disabled	
Use Outbound Proxy In Dialog	Enabled	
IsDeal180	Enabled	
Logon Wizard	Disabled	
PswPrefix		
PswLength	0	
PswDial	Disabled	
PushXML Server IP		
Use Logo	Custom Logo	?
Upload Logo (The pixel < 132*64)		
	Upload	Cancel
	Confirm	Cancel

Logo settings, you can select System logo, you can also choose Custom logo to upload your own logo

Note:

1. You can also upload the Logo by AUTO PROVISION.
2. By the Web interface, users can set a logo to be a System Logo or a Custom Logo, and the Custom Logo can be deleted.
3. Only support DOB format file, For more information please contact the system administrator.

Programmable Key

The hot-key, navigation keys and function keys on the keypad are editable. Users can customize specific features for these keys according to their actual needs.

This function can only be set via the Web interface:

- 1) Click on **Phone->DSS Key->Programmable Key**.

Key	Type	Line	Extension
SoftKey1	History	Local History	
SoftKey2	N/A	Auto	
SoftKey3	History	Auto	
SoftKey4	DND	Auto	
Up	Menu	Auto	
Down	SwitchAccount	Auto	
Left	SMS	Local History	
Right	NewSMS	Auto	
OK	Forward	Auto	
Cancel	Redial	Auto	
CONF	Call Return	Auto	
Hold	Pick Up	Auto	
Mute	XML Group	Auto	
TRAN	XML PhoneBook	Auto	
	Status	Auto	
	Speed Dial	Auto	
	Local Group	Auto	
	Local PhoneBook	Auto	
	Broadsoft Group	Auto	
	Broadsoft PhoneBook	Auto	
	N/A	Auto	
	Forward	Auto	

- 2) Customize specific features for these keys.
- 3) Click **Confirm** button to save the change.

Softkey Layout

The phone can support 12 kinds of call interface to set up softkey, user can setup different function keys according to his/her own requirement or habit.

To set up softkey via web interface:

- 1) Click on **Phone->Softkey Layout**.
- 2) In the **Custom Softkey** field, select **Enable** in the pull-down menu.
- 3) You can select the corresponding call states which you want to set up the softkey in the **Call States** field.
- 4) Highlight the desired one in the **Unselect Softkeys/Select Softkeys** list, and press the / to move to the other list.
- 5) And you can use / to select the order how to display in the call states.
- 6) Click **Confirm** to save the changes.

- 7) You can also click on the **Reset to Default** button to reset the softkeys interface.

The screenshot shows the 'Phone' configuration page in the Enterprise IP Phone web interface. The 'Custom SoftKey' is set to 'Disabled' and 'Call States' is set to 'Dialing'. The 'Unselected Softkeys' list includes Empty, History, Directory, Call Switch, Line Selection, and Pool. The 'Selected Softkeys (ordered by position)**' list includes Send, IME, Delete, and Cancel. A 'NOTE' section on the right explains the availability of these softkeys based on call state and account status. At the bottom are 'Confirm', 'Cancel', and 'Reset to Default' buttons.

Note:

We can add the Empty key more than once, but others can only choose once time, and can't be repeat. when you set up the Selected Softkeys key more than 4 options, the LCD of phone will transfer the forth key with "More" automatically, which can be use to switch to the next page, and continue to show up the other key.

Live Dialpad

The feature defines whether to dial out the dialed number automatically.

This function can only be set via the Web interface:

- 1) Click on **Phone->Preference->Live Dialpad**.
- 2) Enable or disable it in the pull-down menu.

3) Click **Confirm** button to save the change.

The screenshot displays the 'Phone' configuration page of the Enterprise IP Phone web interface. The page has a teal header with a 'Logout' link and a navigation bar with tabs: Status, Account, Network, Phone (selected), Contacts, Upgrade, and Security. Below the navigation bar is a sub-menu with links: Preference, Features, Softkey Layout, DSS Key, EXT Key, Action URL, Voice, Ring, Tones, Dial Plan, and SMS. The main content area is divided into two columns. The left column lists settings with their current values: WEB Language (English), DHCP Time (Disabled), Time Zone (+8 China(Beijing)), Primary NTP Server (cn.pool.ntp.org), Secondary NTP Server (cn.pool.ntp.org), Update Interval(seconds) (1000), Daylight Saving Time (Automatic), Fixed Type (By Date), StartTime (Month, Day, Hour), EndTime (Month, Day, Hour), Offset(minutes), Manual Time (Disabled), Time Format (24 Hour), Date Format (WWW MMM DD), Live Dialpad (Disabled), Inter Digit Time(1~14)(seconds) (4), Flash Hook Time(<800ms) (1), Backlight Time(seconds) (30), Keyboard Lock (Disabled), and WatchDog (Enabled). The right column contains a 'NOTE' section with explanations for Time Zone, NTP Server, Update Interval, Daylight Saving Time, Manual Time, and Ring Tone. A tooltip for 'Inter Digit Time' states: 'Enable/Disable "dial out automatically" on user dial-up interface'.

Replace Rule

A dial plan establishes the expected number. This includes country codes, access codes, area codes and all combinations of digits dialed. For example if you set the *Prefix* as 0 and *Replace* as 0086 (Chinese country code), when you dial 0 out, the number will be replaced by 0086 automatically.

To set a Dial Plan via the Web interface:

- 1) Click on **Phone->Dial Plan->Replace Rule**.
- 2) Enter the desired **Prefix**, **Replace** and **Account**.
- 3) Click **Add** button to save the changes.
- 4) You can also delete a specific one from the dial plan list.

5) You can select a record to modify, then click **Edit** button to submit.

Replace Rule >> ?

Index	Prefix	Replace	Account
1	01	01234	1
2	125478	1254	1,2
3			
4			
5			
6			
7			
8			
9			
10			

Prefix Replace Account

Dial-now>> ?
Area Code>>
Block Out>> ?

NOTE
Digit 0-9 *
 Identifies a specific digit (do not use # if it is defined as send key).
[digit-digit]
 Identifies any digit dialed that is included in the range.
[digit-digit,digit]
 Specifies a range as a comma separated list.
x
 Matches any single digit/character which is dialed.
.
 Matches an arbitrary number of digits.

Dial Now

Dial-now enables you to define the specific length of any number/letter in advance(for example xxx), next time when users dial 123 whose length matches the Dial-now rule, the phone will dial out 123 in one second without pressing Send button.

To set a Dial Plan via the Web interface:

1) Click on **Phone->Dial Plan->Dial now**.

Replace Rule >> ?
Dial-now>> ?

Index	Dial-now Rule	Account
1	05991245	1
2	1245789633	1
3		
4		
5		
6		
7		
8		
9		
10		

Dial-now Rule Account

NOTE
Digit 0-9 *
 Identifies a specific digit (do not use # if it is defined as send key).
[digit-digit]
 Identifies any digit dialed that is included in the range.
[digit-digit,digit]
 Specifies a range as a comma separated list.
x
 Matches any single digit/character which is dialed.
.
 Matches an arbitrary number of digits.

2) Enter the number in **Dial-now Rule** and **Account**.

- 3) Click **Add** button to save the changes.
- 4) You can select a record to modify, then click **Edit** button to submit.
- 5) You can also delete a specific one from the dial plan list by clicking **Del** button.
- 6) You can also set the **Time Out for Dial-now Rule** via web interface. Choose **Phone->Features->Time Out for Dial-now Rule**, enter the time.

Note:

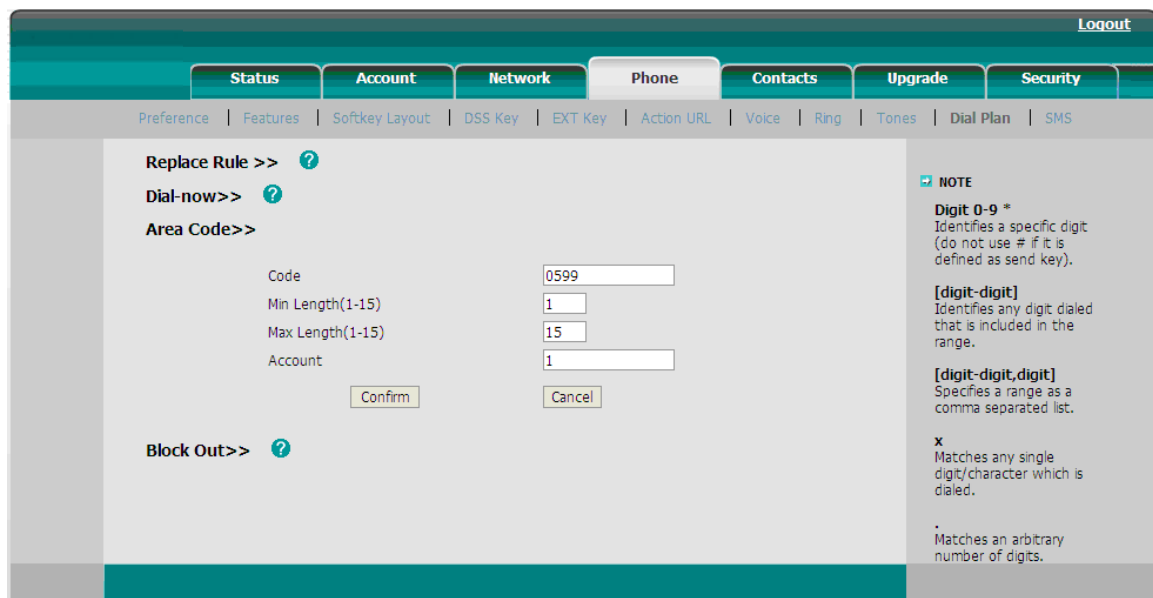
If need to replace the unknown contents, then you can use (.) or (x), "." stand for a string of char, "x" stand for any one char. The content in () stand for a variable, the first variable is expressed by \$ 1, the second variable is expressed by \$ 2, the rest can be done in the same manner. For example: if you want to replace the any input content with the content beginning with 8. Input (.) in Prefix box, and input 8\$1 in Replace box.

Area Code

Area codes are also known as Numbering Plan Areas (NPAs). These are necessary (for the most part) only when dialed from outside the code area and from mobile phones. Area codes usually indicate geographical areas within one country, although the correlation to geographical area is becoming obsolete. For non-geographical numbers, as well as mobile telephones outside of the United States and Canada, the "area code" does not correlate to a particular geographic area.

To add the area code via the Web interface:

- 1) Click on **Phone->Dial Plan->Area Code**.
- 2) Enter the **Code** and **Account**, set the **Min Length** and the **Max Length** option, and then click the **Confirm** button to save.



Area Code

Code: 0599

Min Length(1-15): 1

Max Length(1-15): 15

Account: 1

Confirm Cancel

Block Out

NOTE

Digit 0-9 *
Identifies a specific digit (do not use # if it is defined as send key).

[digit-digit]
Identifies any digit dialed that is included in the range.

[digit-digit,digit]
Specifies a range as a comma separated list.

x
Matches any single digit/character which is dialed.

.
Matches an arbitrary number of digits.

Block Out

The specific phone numbers can be forbidden to be call out from your IP phone.

- 1) Click on **Phone->Dial Plan->Block Out**.
- 2) Enter **Block Out Number** and **Account**, click **Add** button to save the changes,

- or choose the specific one in the list, click **Delete** button to delete the record.
- 3) You can select a record to modify, then click **Edit** button to submit.
 - 4) You cannot dial out the number from your IP phone unless it is removed from the forbidden list.

The screenshot shows the 'Phone' configuration page in the Enterprise IP Phone web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Phone' (selected), 'Contacts', 'Upgrade', and 'Security'. Below this is a sub-navigation bar with 'Preference', 'Features', 'Softkey Layout', 'DSS Key', 'EXT Key', 'Action URL', 'Voice', 'Ring', 'Tones', 'Dial Plan', and 'SMS'. The main content area is titled 'Block Out' and contains a table with columns 'Index', 'Block Out Number', and 'Account'. The table has 10 rows. The first row is filled with '1', '07597894', and '1'. The second row is filled with '2', '0147', and '1'. The remaining rows are empty. To the right of the table are 'Add', 'Edit', and 'Del' buttons. Below the table are input fields for 'Block Out Number' and 'Account'. On the right side of the page, there is a 'NOTE' section with the following text: 'Digit 0-9 * Identifies a specific digit (do not use # if it is defined as send key). [digit-digit] Identifies any digit dialed that is included in the range. [digit-digit,digit] Specifies a range as a comma separated list. x Matches any single digit/character which is dialed. . Matches an arbitrary number of digits.'

Index	Block Out Number	Account
1	07597894	1
2	0147	1
3		
4		
5		
6		
7		
8		
9		
10		

Block Out Number: Account:

Add Edit Del

NOTE
Digit 0-9 *
 Identifies a specific digit
 (do not use # if it is
 defined as send key).
[digit-digit]
 Identifies any digit dialed
 that is included in the
 range.
[digit-digit,digit]
 Specifies a range as a
 comma separated list.
x
 Matches any single
 digit/character which is
 dialed.
.
 Matches an arbitrary
 number of digits.

Note:

1. The numbers set in Emergency cannot use the dial plan rule.
2. In the Account field, you can enter 1,2,3..., "1" represents Account 1, "2" represents Account 2 , if the account box is empty, it mean this rule works for all accounts .

Feature Synchronisation

When enabled the synchronize function, configure the DND/FWD function on device or server, DND/FWD status on device and server will be in correspondence.

To set Feature Key Synchronisation via the Web interface:

- 1) Click on **Phone->Features-> Feature Key Synchronisation**
- 2) There is a pull-down menu in the Type field, choose Intercom from the list.
- 3) Select whether to enable this function from the pull-down menu.

4) Click the **Confirm** to save the change.

Ringer Device for Headset	Use Speaker	?
Headset Send Volume (1~53)	29	
Return code when refuse	486 (Busy here)	?
Return code when DND	480 (Temporarily not available)	?
DND On Code		?
DND Off Code		?
Allow Intercom	Enabled	?
Intercom Mute	Disabled	?
Intercom Tone	Enabled	?
Intercom Barge	Enabled	?
Call Completion	Disabled	?
Enable Semi-Attend Transfer	Enabled	?
Blind Transfer OnHook	Enabled	
Attend Trans OnHook	Enabled	
Transfer on Conference Hang up	Disabled	
Feature Key Synchronisation	Disabled	
Time Out for Dial-now Rule	1	
ACD Auto Available	Disabled	
ACD Auto Available Timer(0~120s)	60	
RFC 2543 Hold	Disabled	
Use Outbound Proxy In Dialog	Enabled	
IsDeal180	Enabled	
Logon Wizard	Disabled	
PswPrefix		

Push XML

Users configure the server's IP address on Web page, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.

To set Push XML via Web interface:

- 1) Click on **Phone->Features-> PushXML Server IP**, enter the server IP in the field.
- 2) Click the **Confirm** button to save the change.







ACD Auto Available Timer(0~120s)	60
RFC 2543 Hold	Disabled
Use Outbound Proxy In Dialog	Enabled
IsDeal180	Enabled
Logon Wizard	Disabled
PswPrefix	
PswLength	0
PswDial	Disabled
PushXML Server IP	10.1.3.145
Use Logo	Disabled

WatchDog

When 'WatchDog' function is 'Enabled', phone will auto reboot after 10 seconds if some important process of phone crash. When disable the function, the phone will not reboot.

Configure watchdog via web interface :

Click on **Phone-> Preference->WatchDog**, in the pull-down menu, select enable or disable this function.




Flash Hook Time(<800ms)	<input type="text" value="1"/>	
Backlight Brightness	<input type="text" value="3"/>	
Backlight Time(seconds)	<input type="text" value="15"/>	
LCD Contrast	<input type="text" value="10"/>	
Keyboard Lock	<input type="text" value="Disabled"/>	
WatchDog	<input type="text" value="Enabled"/>	

Basic Call Functions

Making a call

Call Devices

You can make a phone call via the following devices:

- 1) Pick up the handset, icon  will be showed in the idle screen.
- 2) Press the **Speaker** button,  icon will be showed in the idle screen.
- 3) Press the **Headset** button if the headset is connected to the Headset Port in advance. In the dial-up interface, the icon  will be showed in the idle screen.

You can also dial the number first, and then select the method you will use to speak to the other party.

Call Methods

You can dial the number directly by filling the SIP Server in the registered interface. But the number which you dialed must be in accordance with SIP server.

If you have registered more than one account, you can select a certain account to make your call:

- 1) Press the **Right navigation key** to select a default account when your phone is in idle status.
- 2) In the dial-up interface, press the **Line** hot key to select an account. Then press the **Select** hot key to confirm.
- 3) Press the **three line keys** on the keypad to active the chosen account.

Then

- 1) Dial the number you want to call, or
- 2) In dial-up interface, press **Pool** hot key, use the navigation button to highlight your choice, enter into the corresponding option or
- 3) Press the **RD** button to enter the Dialed Calls interface, then use the **Up/Down navigation keys** to select a record.
- 4) Press the **DSS keys** which have been set as speed dial button.

Then press the Send button or Send hot key to make the call out if necessary.

You can also dial-up via web interface:

- 1) Click on **Contact->Local Phone Book/BlackList**, click on the number which you want to dial out, and then the phone will dial out by default account.

The screenshot displays the 'Local PhoneBook' tab within the 'Contacts' section of the web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Phone', 'Contacts', 'Upgrade', and 'Security'. Below this, a sub-menu shows 'Local PhoneBook', 'BlackList', 'Remote PhoneBook', 'Phone Call Info', 'LDAP', 'Broadsoft', and 'Call Log'. The main content area features a table of contacts with columns for Index, Name, Office Num, Mobile Num, Other Num, Account, and Groups. The table lists 10 contacts, with some numbers highlighted in blue. To the right of the table are buttons for 'Add', 'Edit', 'Del', and 'Delete All'. Below the table is a 'Group Information' section with a 'Groups' dropdown menu and buttons for 'Add', 'Edit', 'Del', and 'Delete All'. At the bottom, there is a 'Please select the contacts list file' section with buttons for 'Import XML', 'Export XML', 'Import CSV', and 'Export CSV'. A 'Show title' checkbox is also present.

Index	Name	Office Num	Mobile Num	Other Num	Account	Groups
1	12345	326	325	552	Auto	
2	361	361	3122	1245	Account2	
3	?				Auto	
4	Andy	1010			Auto	
5	Jakey	3030			Auto	
6	Kobe				Auto	
7	Peter	1000			Auto	
8	Tom	2525			Auto	
9						
10						

Page: 1 | Prev | Next | Move To BlackList | Delete All | Del

Contacts

Name:
Office Num:
Mobile Num:
Other Num:
Account:
Ring:
Groups:
Add Edit Search

Group Information

Groups:
Ring:
Add Edit Del Delete All

Please select the contacts list file

浏览...
Import XML Export XML
 浏览...
Import CSV Export CSV

☒ Show title

NOTE

Add Contact/Blacklist
Fill out the contact information. User shouldn't leave contact name blank.

Delete Contact/Blacklist
Select the contact you want to delete in the grid, and then press the button Delete to confirm.

Move to Contact/Blacklist
Choose the contacts you want to move in the grid, and press the button move to Contact/Blacklist to move it.

Import
Browse the file in XML format.

Export
Click Export button and create a file with whose name you prefer to export.

- 2) Or click on **Contact->Phone Call Info**, enter the number in **Dial a Number**, select the line from the **Outgoing Identity** list. Then click on the **Dial** button to call out.

- 3) Or click on **Contact->Phone Call Info**, click on the number which you want to dial out from the call list, the phone will dial out by corresponding account.

Call Panel

Dial a Number: Dial Hangup

Outgoing Identity: 2105@192.168.1.199

Call List

Dialed List

Index	Date	Time	Local Identity	Name	Number
1	Mon Dec 13	11:09	2105@192.168.1.199		2512@192.168.1.199
2	Mon Dec 13	08:58	2105@192.168.1.199		3205@192.168.1.199
3	Sat Dec 11	11:51	2105@192.168.1.199		2522@192.168.1.199
4	Sat Dec 11	11:50	2105@192.168.1.199		2522@192.168.1.199
5	Sat Dec 11	11:49	2105@192.168.1.199		13159257316@192.168.1.199
6	Sat Dec 11	11:49	2105@192.168.1.199		6203@192.168.1.199

Missed List

Index	Date	Time	Local Identity	Name	Number
1	Sat Dec 11	12:20	2105@192.168.1.199	黄建辉	6203@192.168.1.199
2	Sat Dec 11	10:31	2105@192.168.1.199	Joyce 王丽云	8103@192.168.1.199
3	Fri Dec 10	18:13	2105@192.168.1.199	Receptionist 前台	8888@192.168.1.199
4	Fri Dec 10	18:12	2105@192.168.1.199	Lily 郝丽丽	3207@192.168.1.199
5	Thur Dec 9	18:32	2105@192.168.1.199	黄建辉	6203@192.168.1.199

Received List

Index	Date	Time	Local Identity	Name	Number
1	Mon Dec 13	09:20	2105@192.168.1.199	Vin 徐明	8500@192.168.1.199
2	Mon Dec 13	09:19	2105@192.168.1.199	Vin 徐明	8500@192.168.1.199
3	Sat Dec 11	11:51	2105@192.168.1.199	李秋勇	2522@192.168.1.199
4	Sat Dec 11	11:35	2105@192.168.1.199	李秋勇	2522@192.168.1.199
5	Fri Dec 10	17:09	2105@192.168.1.199	黄彬	2802@192.168.1.199
6	Fri Dec 10	09:45	2105@192.168.1.199	2020	2020@192.168.1.199

- 4) You can click the **Hangup** button to end the call in the web page.

Password dial

When number entered is beginning with the password prefix, the following N numbers after the password prefix will be hidden as *, N stand for the value which you enter in the PswLength field. For example: you set the password prefix is 3, enter the PswLength is 2, then you enter the number 34567, it will display 3**67 on the phone.

To configure password dial via web interface:

- 1) Click on **Phone->Features->PswDial**, in the pull-down menu, select **Enable**.
- 2) Enter the **Password** prefix in the **PswPrefix** field
- 3) Enter the **Length** in the **PswLength** field.

4) Click **Confirm** button to save the changes.

The screenshot displays a configuration page for an Enterprise IP Phone. It features a list of settings on the left and their corresponding values on the right. Most settings are toggleable via dropdown menus, with some having a green question mark icon. At the bottom, there are 'Confirm' and 'Cancel' buttons.

Setting	Value	Help Icon
Intercom Tone	Enabled	?
Intercom Barge	Enabled	?
Call Completion	Disabled	?
Enable Semi-Attend Transfer	Enabled	?
Blind Transfer OnHook	Enabled	
Attend Trans OnHook	Enabled	
Transfer on Conference Hang up	Disabled	
Feature Key Synchronisation	Disabled	
Time Out for Dial-now Rule	1	
ACD Auto Available	Disabled	
ACD Auto Available Timer(0~120s)	60	
RFC 2543 Hold	Disabled	
Use Outbound Proxy In Dialog	Enabled	
IsDeal180	Enabled	
Logon Wizard	Disabled	
PswPrefix		
PswLength	0	
PswDial	Disabled	
PushXML Server IP	Disabled / Enabled	
Use Logo	Disabled	?

Call Completion

Have encountered such a situation? When you call a contact, but the other side is busy on a call. Do you want the server to inform you immediately when the contact ends the call, in order to establish a conversation with each other in time? Call Completion can help you to solve this problem.

To configure Call Completion via phone interface:

- 1) Press the following hot keys: **Menu->Features- ->Call Completion** to enter into the configuration page.

The screenshot shows a phone's internal menu for 'Call Completion'. The title is 'Call Completion'. Below it, the first option is '1. Call Completion:' with a value of 'Disable' and a right arrow. At the bottom, there are three buttons: 'Back', 'Switch', and 'Save'.

- 2) Press the **Switch** hot key, select whether to enable this option.
- 3) Press the **Save** hot key to save your changes.

Answering a call

Answering an incoming call

- 1) If you are not on an active call, lift the handset to answer it using the handset,

or press the **Speaker** button to answer it using the speakerphone, or press the **Headset** button to answer it using the headset.


- 2) If you are on an active call, the LCD will prompt to display: *Incoming Call : xxx*. Press **Answer** hot key to answer the call, or **Reject** hot key to refuse it.

During the conversation, you can alternate between Headset, Handset and Speaker phone by pressing the corresponding buttons or picking up the handset.

Denying an incoming call

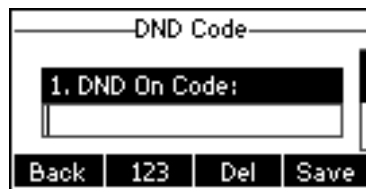
Press **Reject** hot key or **X** button to deny the incoming call directly.

DND

Press **DND** hot key to active DND Mode. Further incoming calls will be rejected and the LCD will display  icon. Press **DND** hot key again to deactivate DND mode. You can find the incoming call record in the Call History.

You can also set DND function by the DND Code:

- 1) Press **Menu->Features->DND Code** to enter the configuration interface.



- 2) Set the DND On Code and the DND Off Code, then press the **Save** hot key to save the changes.
- 3) If the DND mode is not enabled, when you press the **DND** hot key, the phone will send a message to the server, and the server will turn on the DND function. Then any calls to the extension will be rejected by the server automatically. The incoming call record will not be displayed in the **Call History**.

Intercom

Intercom mode is useful in an office environment as a quick access to connect to the operator or the secretary.

To configure Intercom option via phone interface:

- 1) Press the following hot keys: **Menu->Features->Intercom->OK** to enter the configuration page.



- 2) **Intercom Allow**: To set whether to answer the incoming intercom calls.
- 3) **Intercom Mute**: To set whether to mute the incoming intercom calls automatically.

- 4) **Intercom Tone:** To set whether to play ring tones when there is incoming intercom calls to your extension.
- 5) **Intercom Barge:** To set whether to answer the incoming intercom calls during a conversation. If the option is enabled, when there is incoming intercom calls to your extension and you are on an intercom conversation, it will refuse the call automatically; or it will put the current call on hold and put the incoming intercom call through.
- 6) Choose and set the different options by **Navigation keys** and the **Switch** hot key.
- 7) Press the **Save** hot key to save your changes.

To configure Intercom option via web interface:

- 1) Click on **Phone->Features**
- 2) Configure **Allow Intercom, Intercom Mute, Intercom Tone** or **Intercom Barge**. You can refer to the above instruction for more information.

Call Forward

This feature allows you to forward an incoming call to another phone number, such as a cell phone or voice mailbox.

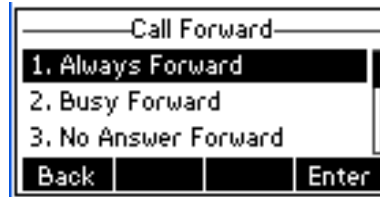
The following call forwarding events can be configured:

- *Always:* Incoming calls are immediately forwarded.
- *Busy:* Incoming calls are immediately forwarded when the phone is busy.
- *No Answer:* Incoming calls are forwarded when the phone is not answered after a
specific period.

To configure Call Forward via Phone interface:

- 1) Press the following hot keys: **Menu->Features ->Call Forward->Enter**. You can also press the **Tran** button to enter the forward setting page directly when the phone is in the idle status.

- 2) There are 3 options: **Always Forward**, **Busy Forward** and **No Answer Forward**.



- 3) Select one of them, enter the phone number you want to forward.



- 4) If you want to realize this function by the server, please enter the **On Code** and **Off Code** option, then when you choose to enable the call forward function via your IP phone, it will send message to the server, and the server will turn on the function immediately. When there is call to the extension, the server will forward it to the set number automatically based on the forward type. The IP phone will not show the record in the **call history**.



- 5) Press **Save** hot key to save the changes.

To configure **Call Forward** via Web interface:

Click on **Phone->Features->Forward** to do the relating changes. Please refer to the above configuration information.

In addition, you can enable or disable the phone to forward an incoming call to international number which begins with the international call prefixes of 00. If you need specify the international call prefix, contact your system administrator for more information.

Forward international is enabled by default.

To configure Forward International via phone interface:


- 1) Press **Menu -> Settings -> Advanced Settings -> Fwd International**.
- 2) Select **Disable or Enable** in the pull-down menu of **Fwd International**.
- 3) Press **Save** to save your changes.

Active Call

Mute

This function allows you to mute the microphone of the active audio device during a call; you cannot be heard by the other party. You can still hear all other parties while mute is enabled. When you press the MUTE button all of the conversation will be muted.

To mute/resume a conversation:

- 1) Press **MUTE** button during a conversation to mute the calls, the icon  will be shown on the LCD.
- 2) Press **MUTE** button again to turn on the microphone.

Call Hold

This call function allows you to place an active call on hold. In this case your IP PBX might play a melody or message to the other party while waiting. Other calls can be received and made while a call is put on hold.

To hold/resume a call via Phone interface:

- 1) Press the **HOLD** button or **Hold** hot key to place your active call on hold.
- 2) During the call, there will be a "dodo" sound for each 30 second, suggesting that the current call is placed on Hold.
- 3) If there is only one call on hold, press the **Resume** hot key or **Hold** button to retrieve the call.
- 4) If there are more than one call on hold, press the **line** button or the **Up/Down** button to highlight the call, then press the **Resume** hot key or **HOLD** button to retrieve the call.

Note:

When you are under the call hold status, putting down the handset, the conversation will go on over the speaker instead of hanging up the call.

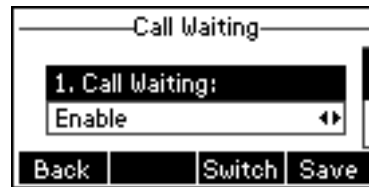
Call Waiting

This call feature allows your phone to accept other incoming calls to the extension no matter under which circumstances.

To enable/disable Call Waiting via Phone interface:

- 1) Press **Menu->Features->Call Waiting->Enter** hot keys.

- 2) Use the **navigation keys** or **Switch** hot key to enable/disable the call waiting option.



- 3) Use the **navigation keys** or **Switch** hot key to enable/disable the **Play Tone** option. This option used to define whether to play ring tones when there is call incoming during an active call.
- 4) Press **Save** hot keys to save the changes, or **Back** hot key to return to the previous menu.

To enable/disable Call Waiting via Web interface:

Click on **Phone->Features->Call Waiting** option to do the relating changes.

Call Transfer

You can customize your phone so that incoming calls are transferred directly to the third party such as another extension, mobile phone number, etc. There are three ways to transfer the call: **Blind Transfer**, **Attended Transfer** and **Semi-Attended Transfer**. If there are calls connected at present, A can choose the extensions to be transferred.

To Blind Transfer via phone interface:

- 1) A and B is on an conversation, A press **TRAN** Button or **Transfer** hot key to place B on hold, then A can dial the third telephone number C (or press **Directory** hot key to enter the contact list, then choose a record) and press the **Transfer** hot key to call out.
- 2) A shows on hold status for a short while, A disconnect from the call.
- 3) C answer the call, Call is established between B and C.
- 4) If C refused to answer the call, it will prompt A that the transfer operation is failed. If the current mode is speaker, it will ring up; if the current mode is handset or headset, it will play ring tones for every five seconds. Pressing any function keys to exit the prompt interface. This function should be supported by server.

To configure Attended Transfer via phone interface:

- 1) A and B is on an conversation, A press **TRAN** Button or **Transfer** hot key to put B on hold, then A can dial the telephone number C (or press **Directory** hot key to enter the contact list, then choose a record) and press the **OK** or **Send** button to dial out.
- 2) After C answered it, A and C can have a private conversation without B hearing, then A press the **Tran** button or **Transfer** hot key to complete the transfer.
- 3) A will be disconnected from the call. Call is established between B and C.

To configure Semi-Attended Transfer via phone interface:

- 1) A and B is on an conversation, A press the **TRAN** button or **Transfer** hot key to place B on hold, then A can dial the new number of C (or press **Directory** hot key to enter the contact list, then choose a record) and press the **OK** or **Send** button to dial out.
- 2) While C is ringing, A hang up or press the **Transfer** hot key. Then A will turn to the hold status, and the LCD will display as Transferred.
- 3) A disconnect from the call, when C answer the call, Call is established between B and C.

Note:

Make sure that the SIP server you have registered supports this function.

3-way Conference

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

To establish a conference:

- 1) Press the **Conference** hot key or **Conf** button during an active call.
- 2) The first call is placed on hold. Enter the number to conference in (or press **Directory** hot key to enter the contact list, then select a contact to conference in) , then press the **Send** hot key.
- 3) When the call is answered, User can have a private conversation at first. And then press the **CONF** button or **Conference** hot key, the conference is established between user and the other two parties.

- 4) During the conference, press the **Split** hot key to split the conference into two hold lines, and press the **Resume** hot key to resume the selected call respectively.
- 5) When you press the **Hold** key, the conference will be on hold.
- 6) User Hang up to disconnect all parties.

Network Conference

If you want to make a conference with more than three people, you can enable the function of network conference. This function needs the server's support.

If you enabled this function, you can put the meeting conference on the server.

To enable network conference via web interface:

- 1) Click on **Account->Account X->Advance->Conference Type**, there is a pull-down menu, choose **Network** from the list.
- 2) Enter the **Conference URI**.
- 3) Click **Confirm** button to save the changes.

Subscribe Register	Disabled	?
Subscribe for MWI	Disabled	?
MWI Subscription Period(Scope:0~84600)(seconds)	3600	
Caller ID Header	FROM	?
Use Session Timer	Disabled	?
Session Timer(seconds)		?
Refresher	Uac	?
Use user=phone	Disabled	?
Voice Encryption (SRTP)	On Off	?
ptime(ms)	20	?
BLF List URI		?
BLF List Code	*97	
Shared Line	Disabled	?
Dialog-Info Call Pickup	Disabled	?
BLA Number		?
BLA Subscription Period(Scope:60~7200)	300	?
SIP Send MAC	Disabled	?
SIP Send Line	Disabled	?
SIP Registration Retry Timer(Scope:0~1800)(seconds)	30	?
Enable Signal Encode	Disabled	
Signal Encode Key		
Conference Type	Local	
Conference URI		
ACD Subscription Period(120~3600)		

Confirm Cancel

To establish a conference:

- 1) Press the **Conference** hot key or **Conf** button during an active call.
- 2) Enter the number to conference in, then press the **Send** hot key
- 3) When the call is answered, press the **CONF** button or **Conference** hot key.
- 4) After starting a three way conference, press **Conf** button to enter Conference dialing interface and invite another party to participate in conference.
- 5) After starting conference, press **Hold** key to place local call on hold without

influencing others in conference.

HuaWei ATS Conference

To enable this function can establish Multi-Party Conference; you can add or delete any attenders, also you can have a private chat with any member. This function needs the server's support.

To enable ATS conference via web interface:

- 1) Click on **Account->Account X->Advance->Conference Type**, there is a pull-down menu, select **ATS** from the list.
- 2) Enter the **Conference URI**.
- 3) Click **Confirm** button to save the changes.

Subscribe Register	Disabled	?
Subscribe for MWI	Disabled	?
MWI Subscription Period(Scope:0~84600)(seconds)	3600	
Caller ID Header	FROM	?
Use Session Timer	Disabled	?
Session Timer(seconds)		?
Refresher	Uac	?
Use user=phone	Disabled	?
Voice Encryption (SRTP)	On Off	?
ptime(ms)	20	?
BLF List URI		?
BLF List Code	*97	
Shared Line	Disabled	?
Dialog-Info Call Pickup	Disabled	?
BLA Number		?
BLA Subscription Period(Scope:60~7200)	300	?
SIP Send MAC	Disabled	?
SIP Send Line	Disabled	?
SIP Registration Retry Timer(Scope:0~1800)(seconds)	30	?
Enable Signal Encode	Disabled	
Signal Encode Key		
Conference Type	ATS	
Conference URI		
ACD Subscription Period(120~3600)	3600	

Confirm Cancel

To establish a conference:

- 1) Press the **Conference** hot key or **Conf** button during an active call.
- 2) Dial the number to conference in, then press the Send hot key
- 3) When the call is answered, press the **CONF** button or **Conference** hot key.
- 4) After the conference is established, initiator can continually add conference members, press Add hot key to enter into dial-up interface, enter the number, press **Send** button to dial out, when people answer the call, he have been attended to the conference.
- 5) Initiator can have a private chart with any member, choose any members in phone member list, and then press **PriChat** hot key, both sides enter into private chart mode, if there only 3 CONF members, both sides begin to private

chart and the third party enter into a Hold mode. If there are more than 3 attender, and then the other attenders keep on talking. If you want to end the private chat, press **ExitPri** hot key.

- 6) Initiator can remove any CONF members, choose any members in phone member list , and then press **Remove** hot key again.
- 7) Initiator Hangs up, the conference is over.

Message

The phone supports SMS (Short Messaging Service) and Voicemail, if you want to use them, please make sure that your VoIP telephony system supports this functionalities and your accounts' message has been enabled.

Voicemail

Voice mailbox messages, which are usually stored on a media server of local or hosted VoIP telephony system, it can be retrieved from phone.

New voice messages can be indicated both acoustically and visually as described below:

- The idle screen will indicate the new voice messages coming:
- The **MESSAGE** button will be lighted.

To configure the Voicemail code via Phone interface:

- 1) Press **Menu->Messages->Voice Mail->Set Voice Mail**.
- 2) Use the **navigation keys** to highlight the line for which you want to set, enter the code which the phone used to connect to your system. Press **123** hot key to select the proper input method.
- 3) Press **Save** to save the change, press **Back** to return to the previous menu.

Note:

Please contact your system administrator for the connecting code. Different systems have different codes.

To configure Subscribe for MWI via the web interface:

- 1) Click on **Account->Advanced->Subscribe for MWI**.

- 2) Select **Enable** in the pull-down menu.

Codecs >> ?	
Advanced >>	
UDP Keep-alive Message	Enabled ?
UDP Keep-alive Interval(seconds)	30
Login Expire(seconds)	3600 ?
	5060 ?
RPort	Disabled ?
SIP Session Timer(seconds) T1	0.5 ?
SIP Session Timer(seconds) T2	4
SIP Session Timer(seconds) T4	5
Subscribe Period(seconds)	1800 ?
DTMF Type	RFC2833 ?
How to INFO DTMF	Disabled
DTMF Payload(Scope:96~255)	101
100 reliable retransmission	Disabled ?
Enable Precondition	Disabled ?
Subscribe Register	Disabled ?
Subscribe for MWI	Disabled ?
MWI Subscription Period(Scope:0~84600)(seconds)	3600
Caller ID Header	FROM ?
Use Session Timer	Disabled ?
Session Timer(seconds)	
Refresher	Uac ?
Use user=phone	Disabled ?
Voice Encryption (SRTP)	<input type="radio"/> On <input checked="" type="radio"/> Off ?
ptime(ms)	20 ?
BLF List URI	

The device sends a Subscribe packet to the server to subscribe Message waiting, the device will send a Subscribe packet to the server after registration

display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

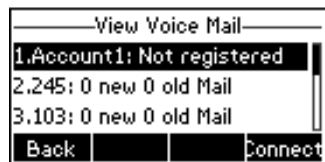
Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be:
com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.

Advanced
The Advanced parameters for administrator.

To view the voicemail via the Phone interface:

- 1) Press **Menu->Messages->Voice Mail->View Voice Mail**.



- 2) You can view the amount of the voice mail that includes new or old voice mail.
- 3) Select the account and press the **Connect** button, then you are able to listen to your new and old messages.

To retrieve the new voicemail via the Phone interface:

- 1) Press the **Voicemail** hot key or the **Message** button, if you have already set the Voicemail number.
- 2) User may be prompted to enter the password which is needed to connect to your VoIP telephony system. It depends on your system.
- 3) Voice mailbox is enabled and Users are able to listen to your new and old messages.

Note:

Before retrieving the new voicemail, please make sure that the connecting code has been set on the phone.

SMS

You can retrieve the SMS in the same way as Voicemail.

To retrieve the SMS via Phone interface:

- 1) Press **Menu->Messages->Text Message** to enter the configuration interface.
- 2) Use the **navigation keys** to highlight the options. You can read the message in the **Inbox/ Sentbox / Outbox/ Draftbox**.



- 3) Press **View** hot key to open and read the specific message.

To reply the SMS via Phone interface:

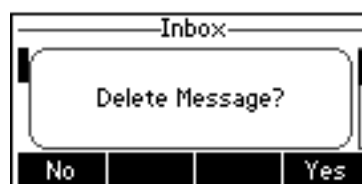
- 1) After retrieving the specific message, press **Reply** hot key, use the **abc** hot key to change the input method.
- 2) Press **Send** hot key, Users are required to select an account which is used to send out the message and the number you want to send to.
- 3) Press **Send** hot key to send out the message, or **Back** hot key to return to the previous menu.

To edit a new message via the Phone interface:

- 1) Press **Menu->Messages->Text Message->New Message** to enter.
- 2) Edit the new message; use the **abc** hot key to change the Input Method.
- 3) Press **Send** hot key to send out the message, or **Back** hot key to return to the previous menu.

To delete the message via the Phone interface:

- 1) After retrieving the specific message, press **Delete** hot key.
- 2) You are prompted to confirm the Delete, press **Yes** hot key to delete the message, press **No** hot key to return to previous menu.



To edit the message via Web interface:

- 1) Click on **Phone->SMS**.
- 2) Select the **Account**, and enter **Number**, **Message** content.

- 3) Click **Send** button to send out the message, or **Cancel** button to cancel the operation.

The screenshot displays the 'Phone' tab in the configuration menu. The 'SMS' sub-tab is active, showing fields for 'Account' (55555), 'Number' (2105), and 'Message' (dsadadsadsadzc). A 'Send' button and a 'Cancel' button are at the bottom. A 'NOTE' box on the right states: 'SMS Number Input the phone number which you are going to send message to.'

Note:

You cannot perform the messaging via Web interface except edit Message.

Advanced Phone Functions

Account Settings

Please refer to the previous part "Configuration and Registration" for the basic Account setting information. The following table lists the instruction of the field about the advanced Account Setting.

Field Name	Description
<i>UDP Keep-alive Message</i>	Defines whether to active the phone UDP Keep-alive mechanism. The default is Enabled.
<i>UDP Keep-alive Interval(seconds)</i>	This parameter specifies how often the phone will send a packet to the SIP server. Default is 30 seconds.
<i>Login Expire(seconds)</i>	This parameter specifies the time frequency that phone refreshes its registration. The default interval is 3600 seconds.
<i>Local SIP Port</i>	Local SIP port. The default value is 5060.
<i>RPort</i>	The parameter allows you configuring the proxy to send responses back to a particular address and port. The default is disabled.
<i>SIP Session Timer</i>	This document defines an extension to the Session Initiation Protocol (SIP). This extension allows for a periodic refresh of SIP sessions through a re-INVITE or UPDATE request. The refresh allows both user agents and proxies to determine if the SIP session is still active.
<i>Subscribe Period(seconds)</i>	This parameter could set the period of the subscription. The default value is 3600.
<i>DTMF Type</i>	Select the DTMF type.

You can only configure these settings via Web interface.

- 1) Click on **Account**.
- 2) Select the desired account.

3) Click on **Advanced** to do the relating settings.

Account Account 1

Basic >>

Codecs >> ?

Advanced >>

UDP Keep-alive Message	Enabled	?
UDP Keep-alive Interval(seconds)	30	
Login Expire(seconds)	3600	?
Local SIP Port	5060	?
RPort	Disabled	?
SIP Session Timer(seconds) T1	0.5	?
SIP Session Timer(seconds) T2	4	
SIP Session Timer(seconds) T4	5	
Subscribe Period(seconds)	1800	?
DTMF Type	RFC2833	?
How to INFO DTMF	Disabled	?
DTMF Payload(Scope:96~255)	101	
100 reliable retransmission	Disabled	?
Enable Precondition	Disabled	?
Subscribe Register	Disabled	?
Subscribe for MWI	Disabled	?
MWI Subscription Period(Scope:0~84600)(seconds)	3600	

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.


Advanced
The Advanced parameters for administrator.

Note:

You can consult your system administrator for more information.

TLS

TLS(Transport Layer Security), an IETF standards track protocol(RFC 5246), was based on the earlier SSL specifications developed by Netscape Corporation.

If you make a call based on TLS, the IP phone UI will display the icon  ring back interface.

Click on **Account->Basic**, select **Transport** option, in the pull-down menu, you can choose the **TLS** option, and then click the **Confirm** button to save the change.

The screenshot shows the 'Account Basic' configuration page. The 'Transport' dropdown menu is open, showing 'TLS' as the selected option. A tooltip explains the Transport options: 'There are UDP, TCP, TLS three options. The registered packet protocol is UDP, TCP or TLS, TLS (Transport Layer Security) is encrypted'. The page also includes a 'NOTE' section with definitions for Display Name, Register Name, User Name, NAT Traversal, and Proxy Require.

Account	Account 1
Basic >>	
Register Status	Registered
Account Active	<input checked="" type="checkbox"/> On <input type="checkbox"/> Off
Label	55555
Display Name	55555
Register Name	55555
User Name	55555
Password
SIP Server	192.168.1.10
Enable Outbound Proxy Server	Disabled
Outbound Proxy Server	
Transport	TLS
Backup Outbound Proxy Server	
NAT Traversal	Disabled
STUN Server	
Voice Mail	
Proxy Require	
Anonymous Call	Off
On Code	
Off Code	
Anonymous Call Rejection	Off

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Advanced
Advanced parameters for administrator.

DNS-SRV

If the SIP server cannot be used, the phone will be connected on the server which is available.

To set DNS-SRV via web interface:

Click on **Account->Basic**, select **Transport** option, in the pull-down menu, you can select the **DNS-SRV** option, and then click the **Confirm** button to save the change.

Account Account 1

Basic >>

Register Status	Registered	
Account Active	<input checked="" type="radio"/> On <input type="radio"/> Off	
Label	55555	?
Display Name	55555	?
Register Name	55555	?
User Name	55555	?
Password	?
SIP Server	192.168.1.10	Port 5060 ?
Enable Outbound Proxy Server	Disabled	?
Outbound Proxy Server		Port 5060 ?
Transport	TLS	?
Backup Outbound Proxy Server	UDP TCP TLS DNS-SRV	Port 5060 ?
NAT Traversal		?
STUN Server		Port 3478 ?
Voice Mail		?

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Codecs
Choose the codecs you want to use.

Advanced
The Advanced parameters for

Network Setting

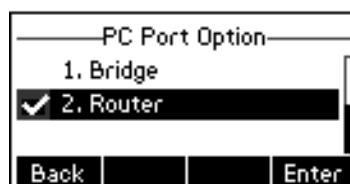
PC Port Setting

Please refer to the previous part "Configuration and Registration" for the basic Network WAN setting information. The following table lists the instructions of the field about the Network PC Port Setting.

Field Name	Description
<i>As Bridge</i>	If you select the Bridge mode, then the two Fast Ethernet ports will be transparent.
<i>As Router</i>	If you select the Router mode, the SIP phone will work as a router
<i>IP address</i>	User could configure the PC port IP address.
<i>DHCP Server</i>	If you set the DHCP server on, the device connected to the PC port will get the IP address automatically between the start IP address and the end IP address. But if you select the bridge mode, the DHCP server cannot work.
Field Name	Description
<i>Start IP Address</i>	Indicate the range of the IP address
<i>End IP Address</i>	Indicate the range of the IP address

To configure PC Port settings via Phone interface:

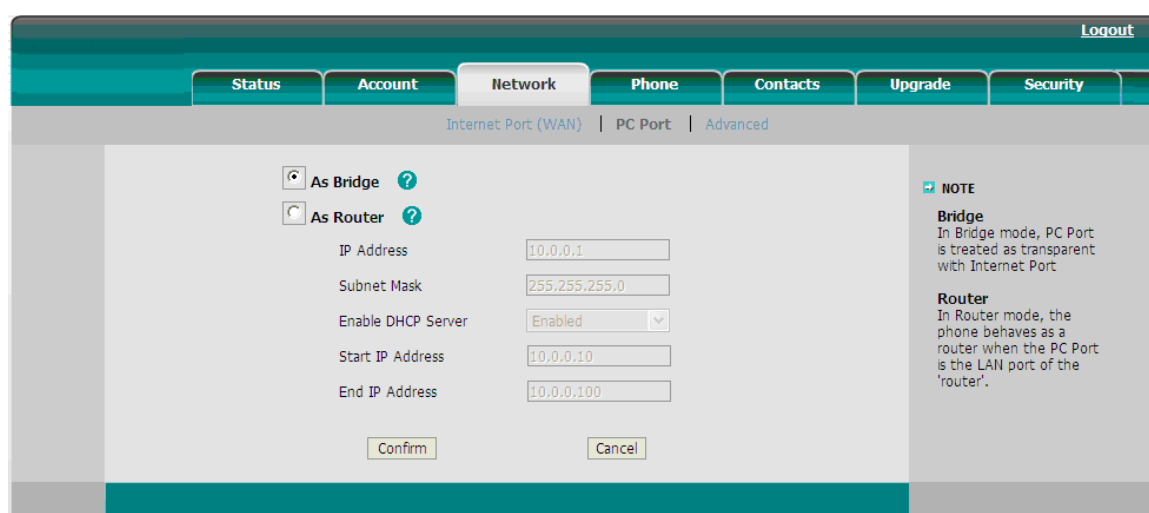
- 1) Press **Menu->Settings->Advanced Settings**.
- 1) Enter the password required, scroll to **Network** option, press **Enter** hot key, select **PC port** option, then press **Enter** hot key.



- 2) If you select **Bridge**, it will save and return to the previous menu.
- 3) If you select **Router**, you will be prompted to enter the **IP Address**, **Subnet Mask**, **DHCP Server Disable/Enable**.etc.
- 4) Press **Save** hot key to save the changes, or **Back** hot key to return to the previous menu.

To configure PC Port settings via Web interface:

Click on **Network-> PC Port** to do the relating configuration. Consult your system administrator for more information.



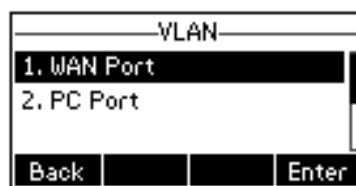
VLAN Setting

VLAN is a group of hosts with a common set of requirements that communicate as if they were attached to the Broadcast domain, regardless of their physical location. The following table lists the instruction of the field about the VLAN Setting.

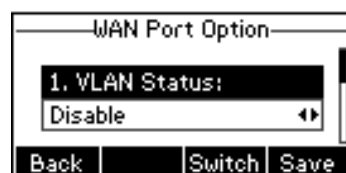
Field Name	Description
QoS	When the network capacity is insufficient, QoS could provide priority to users by setting the value.
Local RTP Port	Define the port for voice transmission.
WebServer	Users can select the WebServer type: Disable, HTTP, HTTPS, or HTTPS & HTTP.

To configure VLAN settings via Phone interface:

- 1) Press **Menu->Settings->Advanced Settings**.
- 2) Enter the password required, scroll to Network option, then press **Enter** hot key, select **VLAN** and press **Enter** hot key.



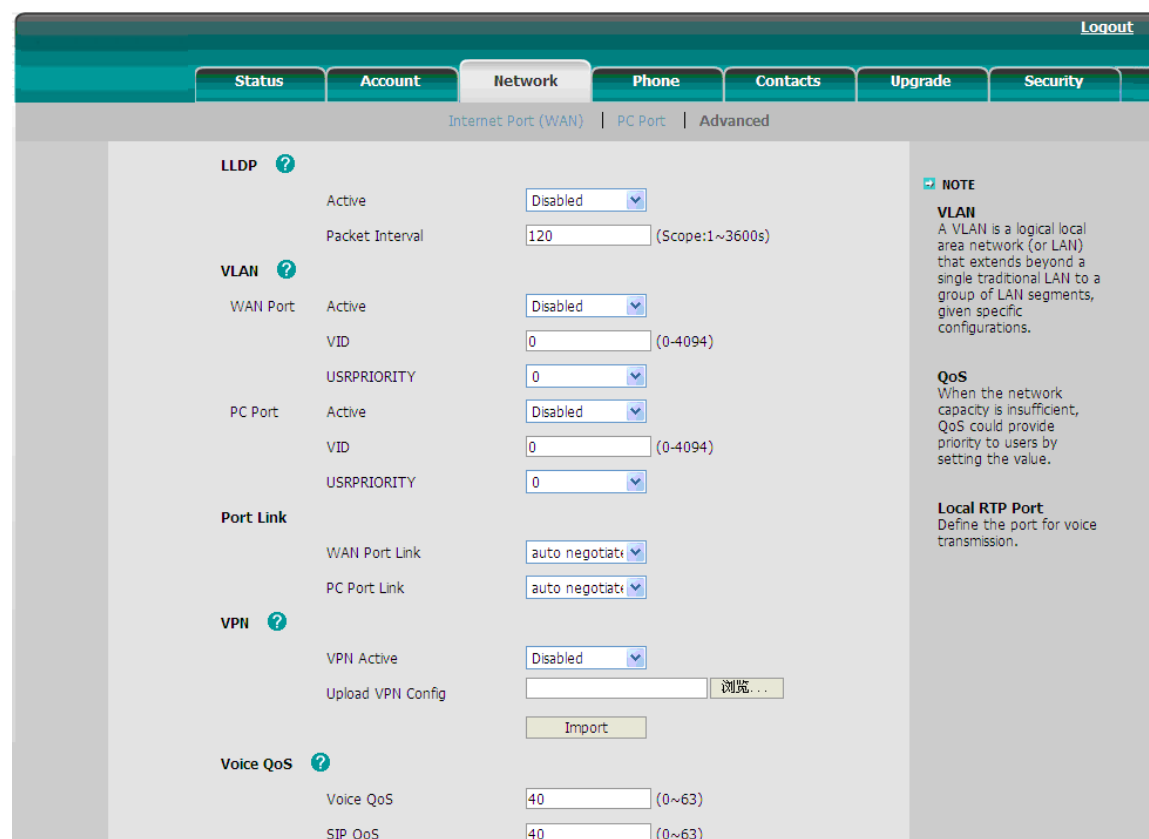
- 3) Select **WAN Port**, press **Enter** hot key.



- 4) Set the **VLAN Status**, and enter the **VID Number, Priority**.
 5) Press **Save** hot key to save the settings, or **Back** hot key to return to VLAN menu.
 6) Follow the same way to set the **PC Port** option.

To configure VLAN settings via Web interface:

Click on **Network->Advanced** to do the relating configuration. You can consult your system administrator for more information.



LLDP

The Link Layer Discovery Protocol (LLDP) is a vendor-neutral Layer 2 protocol that allows a network device to advertise its identity and capabilities on the local network.

Enable LLDP function; the phone will go to switch to get related VLAN parameters automatically. (Synchronous with VLAN in switch)

To configure LLDP via Web interface:

- 1) Click on **Network->Advanced->LLDP->Active** option, in the pull-down menu, choose **Enable**.
- 2) Then enter the corresponding Packet Interval in **Packet Interval** field.
- 3) User can also disable this function when select **Disable** in active field.
- 4) Click the **Confirm** button to save the changes.

The screenshot displays the 'Advanced' configuration page for the IP Phone's Network settings. The 'LLDP' section is expanded, showing the 'Active' status set to 'Disabled' and the 'Packet Interval' set to '120'. Below this, the 'VLAN' section shows 'WAN Port' and 'PC Port' both set to 'Active'. The 'Port Link' section shows 'WAN Port Link' and 'PC Port Link' both set to 'auto negotiate'. The 'VPN' section shows 'VPN Active' set to 'Disabled'. The 'Voice QoS' section shows 'Voice QoS' and 'SIP QoS' both set to '40'. A 'NOTE' box on the right explains VLAN and QoS. Buttons for 'Import' and 'Upload VPN Config' are also visible.

HTTPS

This IP phone can support HTTPS (Hypertext Transfer Protocol over Secure Socket Layer). Adding SSL layer under HTTP, in short, it is a security version of HTTP.

To configure HTTPS settings via Phone interface:

- 1) Press **Menu->Settings->Advanced Settings**.
- 2) Enter the password required, scroll to **Network** option, press **Enter** hot key, select **Webserver Type** option, and then press **Enter** hot key again.

The screenshot shows a 'Webserver Type' selection screen. It has a title bar 'Webserver Type' and a list box containing '1. Webserver Type: HTTP & HTTPS'. Below the list box are three buttons: 'Back', 'Switch', and 'Save'.

- 3) Press the **navigation keys** or **Switch** hot key to select the transmission mode.
- 4) Press **Save** hot key to save the settings, or **Back** hot key to return to Webserver Type menu.

To configure HTTPS settings via Web interface:

Click on **Network->Advanced**, select **WebServer** option, in the pull-down menu of **Type** field, select the transmission mode, and then click the **Confirm** button to save the changes.

PORT LINK

WAN Port Link: auto negotiat...

PC Port Link: auto negotiat...

VPN ?

VPN Active: Disabled

Upload VPN Config: [Browse...] Import

Voice QoS ?

Voice QoS: 40 (0~63)

SIP QoS: 40 (0~63)

Local RTP Port ?

MaxRTPPort: 11800 (0~65535)

MinRTPPort: 11780 (0~65535)

WebServer ?

HTTP port: 80 (1~65535)

HTTPS port: 443 (1~65535)

Type: HTTP&HTTPS

802.1x ?

802.1X Mode: [Dropdown]

Identity: [Text]

MD5 Password: [Text]

Span to PC port

Span to PC port: Disabled

Registration random

Define the port for voice transmission.

Note:

1. For more details of the HTTPS, you can consult with your system administrator.
2. IP phone also support Internet Protocol Version 6.

Maintenance Tasks

Administrator Mode

The phone allows two modes to configure the phone:

- User Mode
- Administrator Mode

Administrator mode grants unlimited access to the phone configuration on both Web and Phone interface. User Mode is not able to access the settings on the Phone interface such as: Advance settings of accounts, Advance settings of Network, Reset to Factory, and other advance phone settings.

Administrator/User Password

Administrator mode grants unlimited access to the phone configuration on both web and phone user interface. The administrator/user password is used to access:

- Web interface.
- The advanced settings of the phone such as Network, Account via the Web and Phone interface.

The default administrator password is **admin**. Meanwhile the user name for Web interface access is **admin**.

To change the administrator password via Phone interface:

- 1) Press **Menu-> Settings-> Advanced Settings**.
- 2) You are prompted to enter the required password, the default one is **admin**.
- 3) Scroll to **Set Password**, press **Enter** hot key.
- 4) You are prompted to enter the **Current, New** and **Confirm password**, press **abc** hot key to change the input method.
- 5) Press **Save** hot key to confirm the change, or **Back** hot key to return to the previous menu.

To change the administrator password via Web interface:

Click on **Security->Password->admin**, enter the **Current, New** and **Confirm password**, click **Confirm** button to save the changes, or **Cancel** button to cancel the changes.

To logout via Web interface:

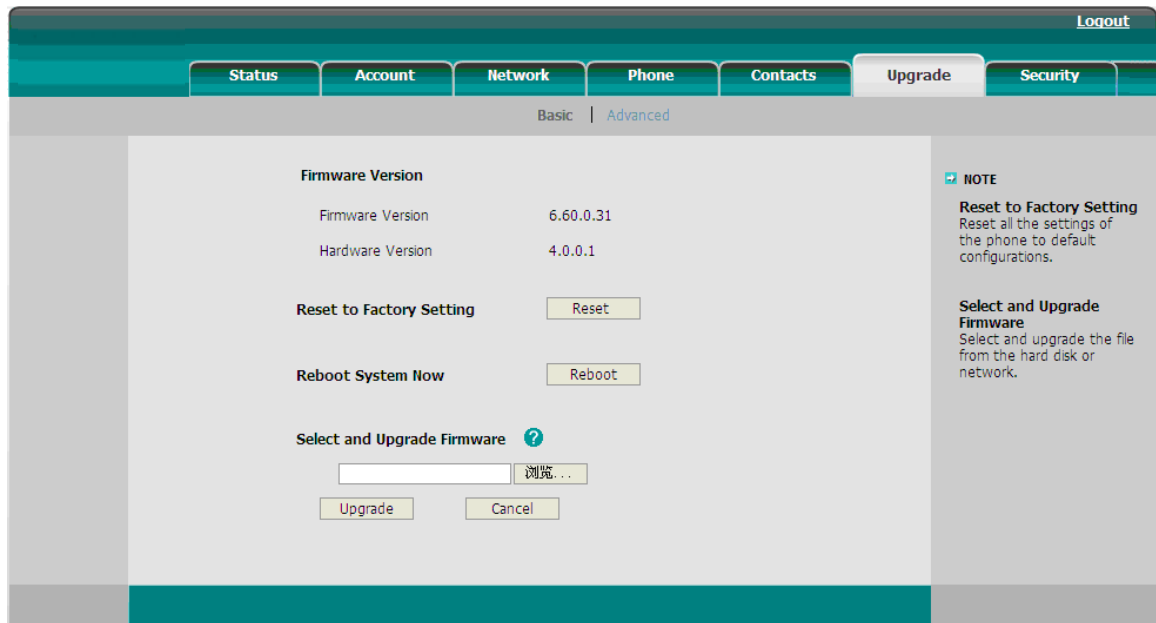
Click the **Logout** button in the top right corner.

Reboot

You should reboot the phone when you are challenged, e.g. after applying changes to the phone configuration.

To reboot via Web interface:

- 1) Click on **Upgrade->Basic**.



- 2) Click **Reboot** button.
- 3) You are prompted to confirm the change, press **OK** to confirm the changes, press **Cancel** to cancel the operation.

Note:

Please do not power off during reboot, or it will cause the flash memory error.

Reset to Factory

You should reset the phone only in this case: the phone configuration was changed and the phone is not functioning anymore. To maintain the configuration of the phone, you need your system administrator or service provider's advice.

To reset to factory via phone interface:

- 1) Press the **Menu->Settings->Advanced Settings**.
- 2) You are prompted to enter the required password, the default one is **admin**.
- 3) Scroll to **Reset to factory** option, and press **Enter** hot key to enter the configuration interface.
- 4) You are prompted to confirm the change, press **OK** hot key to reset to factory settings, or press **Cancel** hot key to return to previous menu.
- 5) It will take a few minutes to reset, please do not power off during resetting, or it will cause flash memory error.

Note:

The reset to factory option must in the admin mode.

To reset to factory via Web interface:

- 1) Click on **Upgrade->Basic**.
- 2) Click **Reset** button.
- 3) You are prompted to confirm the change, press **OK** to confirm the changes, press **Cancel** to cancel the operation.

Note:

If you confirm Reset to Factory, contact list, call history, account settings, etc will be lost. You need to export the configuration first if you still want to import the old configurations after reset. Or your phone must be configured manually unless mass provisioning is used!

To Export/Import the old configuration file via Web interface:

- 1) Click on **Upgrade->Advanced**, select **Export/Import Config**, click on **Export** button to export the file to your local computer.
- 2) Click on **Upgrade->Advanced**, select **Export /Import Config**, click on **Browse** button, select the specific configuration file in your local computer, click on **Import** button.
- 3) It will take a few minutes to reset, please do not power off during resetting, or it will cause flash memory error.

The screenshot shows the 'Upgrade' tab in the web interface. The 'Advanced' sub-tab is selected. The main area contains a form for 'Custom Option(128 ~ 254)' with various input fields and buttons. On the right, there is a 'NOTE' section with instructions for Custom Option, AES Key, and Export/Import Config.

Firmware Update

The phone is delivered with pre-installed firmware which allows operating your phone flawlessly. If you require updating the phone's firmware please contact your system administrator. You can only update the firmware via Web interface.

Warning:

Please do not power off or unplug the Ethernet cable during the updating.

To update the firmware manually via Web interface:

- 1) Click on **Upgrade->Basic**, select the firmware file in your local computer.
- 2) Click on **Upgrade** button to update the new firmware.

To update the firmware automatically via Web interface:

- 1) Click on **Upgrade->Advanced**, configure the relating settings: **Custom Option**, **Custom Option Type**, **URL**, **Account**, **Password**, **Common AES Key** and **MAC-Oriented AES Key**, **PNP config** and **Check New Config**.
- 2) Click **Confirm** button, the phone will check the server for a new firmware in a

specific time, and it updates automatically if there is new firmware.

- 3) Users can also update the firmware immediately by pressing **Auto-provision** button.

Configure Auto Provision via phone interface:

- 1) Press **Menu->Settings->Advanced Settings-> Auto Provision**.
- 2) Enter the **URL, User Name** and **Password**.
- 3) Click the **Save** hot key to save the changes.

The parameters of the Auto-provision :

Parameter	Description
<i>Update Protocol</i>	The phone can be updated via TFTP, FTP or HTTP.
<i>TFTP Server</i>	If you choose TFTP as protocol TFTP, you need to enter the TFTP server IP address and port.
<i>Check new config</i>	You can specific the period that your phone checks the new firmware from the server: Power on, Repeatedly, Weekly, Power on + Repeatedly, Power on + Weekly and Disabled.
<i>Scheduling</i>	You can specific the period in days which the phone checks and updates the new firmware, the range is 1-30 days.

Note:

1. Any power interruption during the following process will most likely lead to a flash memory error. As a result the system cannot boot up anymore.
2. The upgrade priority is PNP, Custom Option, URL by descending.
3. Users can also delete some configuration options by Auto-provision, for example, to delete the admin password.
4. Automatic firmware upgrade via the keypad is available only under the circumstance of all accounts unregistered or unsuccessfully registered.
5. If the automatic firmware upgrade via the keypad is necessary, please contact your system administrator for the specific function code.

Decryption

This IP phone can support y000000000004.cfg and mac.cfg files encryption and decryption for user authentication to realize security usage. If there are any encrypted y000000000004.cfg or mac.cfg files on the server, users can open the webpage of your IP phone.

Click on **Upgrade->Advanced**, select and fill in the Common AES Key (for y000000000004.cfg, should be 16 bit) and MAC-Oriented AES Key (for mac.cfg, should be 16 bit) option, then click the **Confirm** button to decryption the files and upgrade to the new version. Shown as below:

Note:

You can consult your system administrator for the decrypt password.

Set AES Key via phone interface:

- 1) Press **Menu->Settings->Advanced Settings->Set AES Key**.

- 2) Enter the **Common AES** and the **MAC-oriented** option.
- 3) Press the **Save** hot key to save the changes.

Zero-sp-touch

Zero-sp-touch this function can help users to configure AUTOP and network parameters quickly. Enable this function, when the power is on or press the corresponding DSSKEY, the phone will turn to the zero-sp-touch interface.

Configure Zero-sp-touch when the power is on via web interface:

- 1) Click on **Upgrade->Advanced->Zero Active**, in the pull-down menu, select **Enable** to enable this function.
- 2) Enter the time in the **Wait Time** field.

3) Click **Confirm** button to save the changes.

Enter into zero-sp-touch interface, first a countdown interface come into view.

- 1) No operation or press **Cancel** hot key, it will enter idle status.
- 2) Press **Status** key to enters phone's **Status** interface and press **OK** button to return.
- 3) Press **OK** key, enter **Network Setting** interface, press next key to enter an **AutoP** setting interface, enter the corresponding contents; press **OK** key to save the settings or **Back** key return to previous menu.

System Log Export

If there are any errors happened in your phone, you can export the system log and send to your system administrator for diagnosis.

To export the System Log:

- 1) Click on **Upgrade->Advanced**, select **Export System Log** type, if the type is **Local**, it will export the syslog directly; if the type is **Server**, it will export the syslog to the specified server.
- 2) Click **Export** button to export the file

Logout

Status Account Network Phone Contacts Upgrade Security

Basic | Advanced

Custom Option(128 ~ 254) ?

Custom Option Type String ?

URL ?

Account ?

Password ?

Common AES Key ?

MAC-Oriented AES Key ?

ForbidZero Enabled ?

WaitTime 5

PNP Config Enabled ?

Check New Config Disabled ?

Click this button to auto provision immediately Auto provision ?

Export / Import Config 浏览... ?

Import Export

Export System Log Local ?

Export

PCAP Trace Start Stop Export ?

Confirm Cancel

NOTE

Custom Option
Specify the DHCP Option that you want to use for provisioning. Refer to Auto Provision Manual for details about provisioning.

AES Key
It is provided by ISP.

Click this button to auto provision immediately
Click this button to auto provision immediately.

Export/Import Config
Export the configuration files to backup the settings, and could import all the settings after reset.

System Log
There are two methods to export the system log, Local or Server.

Click Export to export current phone's configuration; choose the import configuration file, click Import, it will import the configuration to the current phone and reboot

PCAP Trace Export

The PCAP Trace used to record the data transport of your IP phone. If there are any errors happened in your phone, you can export the PCAP trace and send to your system administrator for diagnosis.

To export the PCAP Trace:

Click on **Upgrade->Advanced** to enter the interface, select **PCAP Trace** option, click **Start** button to begin to capture the trace, and click **Stop** to stop capture the trace, and then click **Export** to export the file to your local computer.

802.1X

IEEE 802.1X is an IEEE Standard for port-based Network Access Control (PNAC). It is part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN, either establishing a point-to-point connection or preventing it if authentication fails. It is used for securing wireless 802.1X access points and is based on the Extensible Authentication Protocol (EAP).

This IP phone can support 802.1X. For the details, please consult your system administrator for more details.

DSS keys Configuration

Enter into the **DSS key** interface, there are three types of key: **MEMERORY key**、**LINE key**、**Program key**. The configuration of **LINE key** and **MEMERORY key** are almost the same. The phone has 10 **Memory keys** and 3 **Line keys** which are able to configure 35 features per key. The following list shows the valid type on the **Memory key** or **LINE key** and provides a description of each feature. The Memory key's default value for each key is N/A which indicates the DSS key hasn't been configured for any features. The Line key's default value for each key is Line.

- N/A
- Line
- Speed Dial
- BLF
- BLF List
- Voice Mail
- Pick Up
- Group Pickup
- Call Park
- Intercom
- DTMF
- Prefix
- Local Group
- XML Group
- XML Browser
- LDAP
- Broadsoft Group
- Conference
- Forward
- Transfer
- Hold
- DND
- Redial
- Call Return
- SMS
- Record
- URL Record
- Multicast Page
- Group Listening
- Public Hold
- Private Hold
- Shared Line
- Hot Desking
- ACD

- Zero-sp-touch
- URL

Note:

1. Quick access features like Intercom and Voicemail must first be configured on your PBX in order to work on your phone. See your system administrator for more information.
2. Users can also connect EXP38/EXP39 to extend the DSS keys to 38 or more.

Line

This key function is used to accept incoming calls and allows placing active calls on hold or resuming them. It can also be used to make a call using a specific extension

Configure the key as **Line**:

- 1) Click on **Phone->DSS Key->Memory Key**.
- 2) **Line** is selected from the pull down menu "Type".
- 3) This functionality will be applied to the line selected from the pull down menu "Line". When the default value is **Auto**, press **Line** DSS key in the idle state to make a call, the phone will select an available extension as the outgoing extension
- 4) Click **Confirm** button to save the changes.

Speed Dial

This key function allows you to speed up dialing numbers often used or hard to remember. Optionally, you can also configure a speed dial key to dial prefix numbers. With this option, the prefix numbers automatically dial when you press the DSS key.

Configure the key as **Speed Dial**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Speed Dial** is selected from the pull down menu "Type".
- 3) Enter the number you want to dial out in the **Value** field.
- 4) This functionality will be applied to the line selected from the pull down menu "Line".
- 5) Click **Confirm** button to save the changes.

BLF

You can configure the key for Busy Lamp Field (BLF) which allows you to monitor the status (idle, ringing, or busy) of other SIP accounts. User can dial out on a BLF configured key.

To assign the key as BLF:

- 1) Select Phone->DSS Key->Memory Key or Line Key, select one of the keys you want to make the assignment, there is a pull-down menu in the Type field, and select BLF from the list.
- 2) Enter the number you want to monitor in the Value field,
- 3) In the "Line" field, select a line for which to apply this key.
- 4) And then enter the feature codes in the extension field.

- 5) Press Confirm button to save the changes.

Please refer to "LED Instruction" for more details about the LED status in different situation.

Note:

In the Web interface, you can enter pickup code in the **Extension** field to enable the pickup function. For example, if you enter 212 in the **value** field, and enter the pickup code *83 in the **Extension** field, when there is an incoming call to 212, press the BLF DSS key, the call is picked up by the phone.

BLF List

This key function can monitor the list status, You can add accounts to the monitored list, the server sends messages to the phone to decide which account BLF list to be monitored.

Configure BLF List key via web interface:

- 1) Click on **Account->Advanced-> BLF List URI**, enter the **BLF List URI**.
- 2) Enter the BLF List Code in the **BLF List Code** field.
- 3) Click **Confirm** button to save the changes.

To configure **BLF List** key via web interface:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**
- 2) Select **BLF List** from the pull down menu of **Type**.
- 3) In the **Line** field, select a line for which to apply this key.
- 4) Click **Confirm** button to save the configuration.

Voice Mail

This key function is configured as Voicemail, which allows you to retrieve voicemail quickly by pressing this key.

Configure the key as **Voice Mail** via web interface:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**, choose one of the keys you want to make the assignment, there is a pull-down menu in the **Type** field, choose Voice Mail from the list.
- 2) Enter the number you want to set as the voice mail box in the **Value** field.
- 3) In the **Line** field, select a line for which to apply this key.
- 4) Click **Confirm** button to save the changes.

Pick Up

This key function allows you to specify extension that you want to monitor. when the monitored extension receives a call, you can press this DSS key to pick up the incoming calls.

Configure the key as Pick Up:

- 1) Choose **Phone->DSS Key->Memory Key** or **Line Key**, choose one of the keys you want to make the assignment, there is a pull-down menu in the **Type** field, choose **Pick Up** option from the list.

- 2) Enter the feature code (for example, input *78345, *78 is the feature code and the 345 is the extension number you want to pickup) in the **Value** field.
- 3) In the **Line** field, select a line for which to apply this key.
- 4) Click **Confirm** button to save the changes.

Group Pick up

This key function allows you to specify a group you want to monitor. You can add accounts to the monitored group, when the monitored group receives a call, you can press this key to pick up the incoming call. If the group receives multiple calls simultaneously, you will pick up the specific one the server assigns to you.

Configure the key as **Group Pick Up**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Group Pick Up** is selected from the pull down menu "Type".
- 3) Enter the feature code (for example,*79) in the **Value** field.
- 4) This functionality will be applied to the line selected from the pull down menu "Line".
- 5) Click **Confirm** button to save the changes.

Call Park

This key function allows you to place a call on hold at one phone and then retrieve the parked call from any other phone. When the conversation which is monitored is transferred to an unused extension, you can press this key to retrieve the call.

Configure the key as **Call Park**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Call Park** is selected from the pull down menu "Type".
- 3) Enter the feature code(for example, enter *681234, *68 is the call park code and the 1234 is the extension you want to park) in the **Value** field.
- 4) This functionality will be applied to the line selected from the pull down menu "Line".
- 5) Click **Confirm** button to save the changes.
- 6) Retrieve the parked call by dialing the call park retrieval code and the parked extension from any other phone.

Intercom

This key function is useful in an office environment as a quick access to connect the operator or the secretary. Before configure the Intercom DSS key, enable allow Intercom via Web interface. You can refer to Basic Call Function/Intercom.

Configure the key as **Intercom**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Intercom** is selected from the pull down menu "Type".
- 3) Enter the extension number you want to intercom in the **Value** field.

- 4) This functionality will be applied to the line selected from the pull down menu "Line".
- 5) Click **Confirm** button to save the changes.

Note:

Intercom feature is not available on all servers. Please contact your system administrator for more details.

DTMF

This key function allows you to send out the desired DTMF number during the conversation. The number needs to be set in advance.

Configure the key as **DTMF**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **DTMF** is selected from the pull down menu "Type".
- 3) Enter the specific number in the **Value** field.
- 4) Click **Confirm** button to save the changes.

Prefix

This key function allows you to dial a call with the prefix number. When press the DSS key in the idle state, the phone will be ready to make a new call, and show up the prefix number which your set previously on the dial interface. You can enter other digit and then dial out

Configure the key as **Prefix**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Prefix** is selected from the pull down menu "Type".
- 3) Enter the specific number in the **Value** field.
- 4) Click **Confirm** button to save the changes

Local Group

This key function allows you to access Local Group quickly. Press the DSS key in the idle state, you can enter the Local Group interface.

Configure the key as **Local Group**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Local Group** is selected from the pull down menu "Type".
- 3) Select a group from the pull down menu "Line".
- 4) Click **Confirm** button to save the changes.

XML Group

This key function allows you to access the XML Group quickly. Press the DSS key in the idle state, you can enter into the XML Group interface.

Configure the key as **XML Group**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **XML Group** is selected from the pull down menu "Type".
- 3) The default value in the pull down menu "Line" is N/A.
- 4) If you pre-configure the **remote phone book** on the **Contacts**, then you can select a remote phone book to specify the group in the pull down menu "Line".
- 5) Click **Confirm** button to save the changes.

XML Browser

Xml browser is a simple browser feature, which is based on xml language and http/https service, you can use tools such as php, javascript, etc., accordance with the established syntax to generate server-side functions dynamically to meet the need of the user xml file, and then download it to phone by http/https. In order to realize a simple browser feature, Using xml browser, customers can achieve the customized features, such as weather information, stock information, date of inquiry, access to address book, google search, news browsing, playing music etc.

Configure the key as **Xml Browser**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Xml Browser** is selected from the pull down menu "Type".
- 3) Enter the **URL** address you want to connect to in the **Value** field.
- 4) Click **Confirm** button to save the changes.

LDAP

This key function allows you to access LDAP quickly, you can refer to **Contact Management->LDAP** for more details.

Configure the key as **LDAP**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **LDAP** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.
- 4) When you press this key in the idle state, the phone will enter into the contact search interface.

Broadsoft Group

This key function allows you to access Broadsoft Group. Press the DSS key under the idle state, you can enter into the Broadsoft Group interface.

Configure the key as **Broadsoft Group**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Broadsoft Group** is selected from the pull down menu "Type".
- 3) The default value in the pull down menu "Line" is N/A.
- 4) If you pre-configure the **Broadsoft Group** on the **Contacts**, then you can select a Broadsoft Group to specify the group in the pull down menu "Line".
- 5) Click **Confirm** button to save the changes.

Conference

This key function allows you to conference another party during a conversation.

Configure the key as **Conference**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Conference** is selected from the pull down menu "Type".
- 3) Enter the number you want to conference in the **Value** field.
- 4) Click **Confirm** button to save the changes.

Forward

This key function allows you to forward a call to other account. Press the DSS key in the idle state, the phone will turn to the Always forward configure interface, then you can configure the Forward to number, when there is any call to the extension will be forwarded to the configure number automatically.

Configure the key as **Forward**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**
- 2) **Forward** is selected from the pull down menu "Type".
- 3) Enter the extension you want to forward to in the **Value** field.
- 4) If you leave the **Value** field blank, the DSS key acts as a Forward button.
- 5) Click **Confirm** button to save the changes.

Transfer

This key function allows you to perform the Blind/Attended/Semi-Attended Transfer during a conversation.

Configure the key as **Transfer**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Transfer** is selected from the pull down menu "Type".
- 3) Enter the extension you want to forward to in the **Value** field.
- 4) During a conversation, press this key, the phone will Blind transfer the call.
- 5) If you leave the **Value** field blank, the DSS key acts as a **Transfer** button.
- 6) Click **Confirm** button to save the changes.

Hold

This key function allows you to hold or retrieve a call during the conversation.

Configure the key as **Hold**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Hold** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

DND

This key function allows you to activate the DND function immediately when you

press the DSS key. Press it again to deactivate DND mode.

Configure the key as **DND**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **DND** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

Redial

This key function allows you to access the **Dialed Calls** interface when press the DSS key in the idle state, then you can select a call to dial out or add it to the contacts etc.

Configure the key as **Redial**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**
- 2) **Redial** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

Call Return

This key function allows you to call return the latest call you received.

Configure the key as **Call Return**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Call Return** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

SMS

This key function allows you to access the **Text Message** interface quickly when press it in the idle state.

Configure the key as **SMS**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **SMS** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

Record

This key function allows you to record during the conversation. Using this feature, please pay attention to the maximum record time and frequency in advance.

Generally, it's only a few minutes.

Configure the key as **Record**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Record** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

URL Record

This key function allows you to achieve the call recording capability during the conversation when follows the voice prompts.

- 1) During a conversation, press the DSS key to start recording.
- 2) Enable the **Recording** function, the recording icon will be flashing on the LCD.
- 3) Press the DSS key again to disable the recording function, the flashing recording icon will be disappeared simultaneously.
- 4) Follow the voice prompts to listen to the recording.

Configure the key as **URL Record**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **URL Record** is selected from the pull down menu "Type".
- 3) Enter the server URL in the **Value** field.
- 4) Click **Confirm** button to save the changes.

Note:

During a conversation, press this type of DSS key to start the recording process; if the other party hung up, your phone will turn to the idle status.

Multicast Paging

Multicast is the delivery of a message or information to a group of destination simultaneously in a single transmission from the source creating copies automatically in other network elements.

A multicast paging is essentially a predefined broadcast address that the phone is programmed to listen to. Each phone can be configured to listen to as many as 10 different multicast IP addresses. The priority of the IP Addresses is equivalent to the stream priority, 1 is the highest, 10 is the lowest. Streams with higher priorities will overlap those with lower priorities.

Users can configure the phone to send (receive) a Real Time Transport Protocol (RTP) stream to (from) pre-configured multicast address without involving SIP signaling.

To configure a programmable key for sending Multicast RTP stream via web interface:

- 1) Click on **Phone -> DSS Key -> Memory Key**. Then choose the desired key you want to configure as multicast Paging.
- 2) Select the key Type as **Multicast Paging** in the pull-down menu;
- 3) Enter the **multicast address(es)** (IP: port) in the **Value** field.

DSS Key 1	Multicast Paging	239.0.1.15:10000
-----------	------------------	------------------

- 4) Click **Confirm** to save the configuration.

Note:

The multicast address ranges from 224.0.0.0 to 239.255.255.255.

Users can also configure the phone to use a default codec for sending Multicast RTP stream via web interface:

- 1) Click on **Phone** -> **Features**.
- 2) Select a codec (i.e., **PCMU**) from the pull-down menu of the **MulticastCodec** configuration item.



MulticastCodec	PCMU
----------------	------

- 3) Click **Confirm** to save the configuration.

To configure Listening Address(es) for receiving Multicast RTP stream via web interface:

- 1) Click on **Contacts** -> **MulticastIP**. Enter the **Listening Address(es)** which is preconfigured as multicast address (IP: port) in the **Listening Address** field.
- 2) Enter the **Label**(String) which will appear on the LCD screen when receiving RTP from the corresponding Listening Address.

	Listening Address	Label	Priority
1 IP Address	239.0.1.15:10000	T28	1

- 3) Click **Confirm** to save the configuration .

After pressing a Multicast Paging key on the phone, the Phone sends RTP to a preconfigured multicast address(es) (IP:port). Any phone in the local network then listens for RTP on the preconfigured multicast address(es) (IP:port). For both sending and receiving of the multicast RTP there is no sip signaling involved. The phone receiving RTP will display the preconfigured listening multicast label (Label) to the user.

Note:

1. Multicast RTP is one way only- from sender to the multicast address(es) (receiver).
For outgoing RTP multicasts, all other existing calls on the phone will be put on hold.

Group Listening

This key function allows you to enable the Speakerphone and Handset/Headset mode at the same time. It is suitable for the group conversation which has more than one person at one side. You are able to speak and listen by using handset/headset; meanwhile the others nearby can listen by using speakerphone. You can get back to the previous mode by pressing the key again. (If the current mode is handset or headset, users can press the speaker button to open or close the group listening function)

Configure the key as **Group Listening**:

- 1) Click on **Phone**->**DSS Key**->**Memory Key** or **Line Key**.
- 2) **Group Listening** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

Public Hold

This key function allows particular BLA group to hold or retrieve a call during a conversation

Configure the key as **Public Hold**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**, Press **Confirm** button to save the changes.
- 2) **Public Hold** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

Private Hold

This key function allows all members belonging to that particular BLA group to hold the call, but only the initiator can retrieve the call during a conversation.

Configure the key as **Private Hold**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Private Hold** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes.

Shared Line

Shared Line Appearances (SLA, which is also named as BLA) feature allows subscribers to share SIP lines and also provides status monitoring of the shared line. When a user places an outgoing call using such an appearance, all members belonging to that particular SLA group are notified of this status and are blocked from using this line appearance until the line goes back to idle or the call is placed on hold.

Similarly all members of the SLA group are notified of an incoming call and the call can be picked up on a line appearance associated with the SLA extension.

Configure the key as **Shared Line**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Shared Line** is selected from the pull down menu "Type".
- 3) Enter the primary account number in the **Value** field.
- 4) This functionality will be applied to the line selected from the pull down menu "Line".
- 5) click **Confirm** button to save the changes.
- 6) Please refer to "LED Instruction" for more details about the LED status in different situation.

Hot Desking

Hot desking is a phone feature that allows accounts to login or logout in an IP phone. After a certain account login, the corresponding configuration of the account will be applied to the phone.

Our phone can support two kinds of Hot desking: Base Mode and Advanced Mode.

- **Base Mode:** Press the DSS key which is configured Hot Desking when the phone is idle to activate the Base Mode. In this mode, the phone will clear all of the account registered on the phone, then register the login account in the first line.
- **Advanced Mode:** In this mode, after registering the new account, the phone will connect the URL to download the Xml file and configuration information of the new account.

Configure the key as **Hot Desking**:

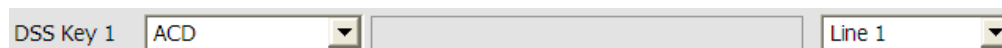
- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**
- 2) **Hot Desking** is selected from the pull down menu "Type".
- 3) Enter the URL address in the **Value** field to activate the advanced mode. Or you can leave it blank to activate base mode.
- 4) Click **Confirm** button to save the changes.

Automatic Call Distribution (ACD)

The Phone supports an Automatic Call Distribution (ACD) feature for specific servers. The ACD feature allows the ACD system to distribute calls from a queue to registered IP phone users (agents). To use the ACD feature on the phone, the user should pre-configure a programmable key as ACD key.

To configure an ACD key via Web interface:

- 1) Click on **Phone -> DSS Key -> Memory Key(or Line Key)**.
- 2) Select **ACD** in the pull-down menu of **Key Type**.
- 3) Select the right **line** for ACD.

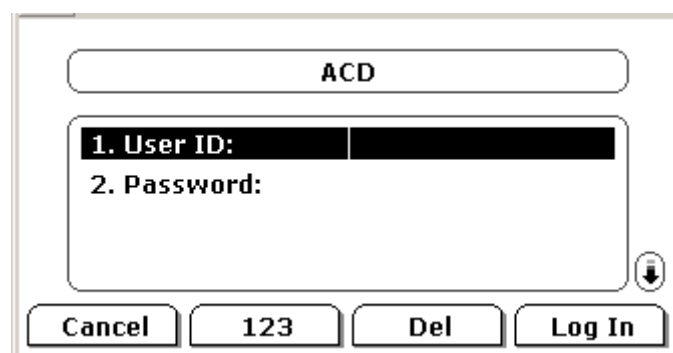


DSS Key 1 ACD Line 1

- 4) Click **Confirm** to save the configuration.

Once the user wants to subscribe to a queue (in order to receive incoming calls), the user presses the ACD key. The LCD screen prompts the user to specify the following information:

- **User ID:** the phone number(s) used to login into the queue.
- **Password:** the password used to login to the queue.



ACD

1. User ID: []

2. Password: []

Cancel 123 Del Log In

When ready to receive calls from the server, the phone user logs into a queue and changes the phone status to "available" manually using the IP phone UI (press DSS Key again to enter to the ACD Status page, then select the phone status as **Available/ Unavailable**). If the phone is set to "available", then the server begins to distribute calls to this phone.

Note:

ACD feature is not available on all servers. Please contact your system administrator for more information.

Zero-sp-touch

This key function allows phone to complete configuration just need to provide the server's user name and password.

Configure the key as **Zero-sp-touch**:

- 1) Click on **Phone->DSS Key->Memory Key** or **Line Key**.
- 2) **Zero-sp-touch** is selected from the pull down menu "Type".
- 3) Click **Confirm** button to save the changes

URL

This key function allows you to send HTTP requests to a web server.

Configure the key as **URL**:

- 1) Click on **Phone->DSS Key->Memory Key**.
- 2) **URL** is selected from the pull down menu "Type".
- 3) Enter the URL in the **Value** field.
- 4) Click **Confirm** button to save the changes.

SNMP

Simple Network Management Protocol (SNMP) is an Internet-standard protocol for managing devices in IP networks. It is used mostly in network management systems to monitor network-attached devices for conditions that warrant administrative attention. SNMP exposes management data in the form of variables on the managed systems, which describe the system configuration. These variables can then be queried (and sometimes set) by managing applications. For further information, you can refer to <http://en.wikipedia.org/wiki/SNMP>.

By default, the phone does not accept SNMP requests. This is necessary because SNMP might introduce unwanted security threats. The phone only supports the GET request of SNMP. SET or other requests are not supported. To enable SNMP, you must specify what IP addressed may send SNMP requests to the phone.

To configure SNMP via Web interface:

- 1) Click on **Network -> Advanced**.
- 2) Select **Enabled** in the pull-down menu of **SNMP Active**.
- 3) Enter the **Port** number and **Trusted Address**.

SNMP ?

Active

Port (0~65535)

Trusted Address

4) Click **Confirm** to save your configuration.

Note:

1. You may use several trusted IP addresses separated by space.
For example, "192.168.3.10 192.168.3.78" will allow accesses from the two listed IP addresses.
2. Wildcards are not supported here

The table below shows part of the available object identifiers (OID) on the phone:

OID	Description	MAX-ACCESS
1.3.6.1.2.1.37459.2.1.1.0	The textual identification of the contact person for this managed node, together with information on how to contact this person. If no contact	read-only
1.3.6.1.2.1.37459.2.1.2.0	An administratively-assigned name for this managed node. By convention, this is the node's fully-qualified domain name. If the name is unknown, the value is the zero-length string.	read-only
1.3.6.1.2.1.37459.2.1.3.0	The physical location of this node (e.g., `telephone closet, 3rd floor'). If the location is unknown, the value is the zero-length string.	read-only
1.3.6.1.2.1.37459.2.1.4.0	The time (in hundredths of a second) since the network management portion of the system was last re-initialized.	read-only
1.3.6.1.2.1.37459.2.1.5.0	The IP Phone's Firmware version	read-only
1.3.6.1.2.1.37459.2.1.6.0	The IP Phone's Hardware version	read-only
1.3.6.1.2.1.37459.2.1.7.0	Model of machine	read-only
OID	Description	MAX-ACCESS
1.3.6.1.2.1.37459.2.1.8.0	The IP Phone's MacAddress	read-only
1.3.6.1.2.1.37459.2.1.9.0	The IP Phone's IPAddress	read-only
1.3.6.1.2.1.37459.2.1.10.0	The IP Phone's LastUpVersion	read-only

We recommend a SNMP test suite which is available at <http://www.net-snmp.org/>

The phone supports only SNMP v2c. Basic GET requests can be run in order to retrieve information from an SNMP-capable device. For example:

```
>snmpget v2c -c public 192.168.3.10 1.3.6.1.2.1.37459.2.1.5.0
```

Action URL

Action URLs are HTTP GET Requests allowing the phone to interact with web server applications. To use this feature, you should specify a HTTP URL corresponding to the predefined event. In case the predefined event has taken place, a HTTP GET to the specified URL is performed.

HTTP GET requests may contain variable names and variable values, which are separated by "=". Each variable value starts with \$ in the query part of the URL (e.g. `http://192.168.10.1/help.xml? MAC=$mac`).

Action URLs can be triggered only by predefined events (e.g. Log on).

To configure Action URL via web interface:

- 1) Click on **Phone->Action URL**
- 2) Enter a **HTTP URL** in the blank field.

Setup Completed	<input type="text" value="http://192.168.10.1/help.xml? model=\$model"/>	?
Log On	<input type="text" value="http://192.168.10.1/help.xml? Logon=\$active_user"/>	?
Log Off	<input type="text" value="http://192.168.10.1/help.xml? Logoff=\$active_ur"/>	?
Register Failed	<input type="text"/>	?
Off hook	<input type="text"/>	?

- 3) Click **Confirm** to save the Configuration.

Note:

Action URL are allowed to be HTTPS URL, for more details contact with the system administrator.

The following table provides Variables can be used in the Action URL:

Parameter	Descriptions
\$mac	MAC Address of the phone
\$ip	IP Address of the phone
\$model	Device model name of the phone
\$firmware	Firmware version of the phone
\$active_url	SIP URI of the active user
\$active_user	User part of the SIP URI of the active user
\$active_host	Host part of the SIP URI of the active user
\$local	SIP URI of the caller
\$remote	SIP URI of the callee
\$display_local	Display name of the caller
\$display_remote	Display name of the callee
\$call_id	Call id
\$duration	Call duration
\$callDirection	Indication of the call as sender or recipient




Action URI

Opposite to Action URL, Action URI allows the phone to interact with web server applications by receiving and handling HTTP GET requests. To simply verify this feature, you can enter the following string into the browser address bar and press **Enter**:

`http://192.168.10.5/cgi-bin/ConfigManApp.com?key=DNDOn`

Then the phone with the IP Address of 192.168.10.5, if existed, will turn on DND. Be aware that only predefined phone events are supported by the phone.

The following table provides the predefined phone events:

Phone Events	Descriptions
key=OK/key=ENTER	Press the OK button or Enter soft key
key=SPEAKER	Press the  button
key=F_TRANSFER	Press the transfer button
key=VOLUME_UP	Increase the volume
key=VOLUME_DOWN	Reduce the volume
key=MUTE	Press the  button
key=F_HOLD	Hold/Resume the call
key=X	Press the  button
key=0-9/*/POUND	Enter the DTMF number(include Numeric , * or # keys)
key=L1-L6	Press a line key
key=D1-D10	Press a DSS key
key=F_CONFERENCE	Press the Conference button
key=F1-F4	Press a soft key
key=MSG	Press the Message button
key=HEADSET	Press the Headset button
key=RD	Press the RD button
key=UP/ DOWN/ LEFT/ RIGHT	Press the navigation key
key=Reboot	Reboot the phone
key=AutoP	Check the Auto provision
key=DNDOOn	Enable DND
key=DNDOOff	Disable DND
key=ATrans=XXXX	Attended Transfer to extension XXXX
key=BTrans=XXXX	Blind Transfer to extension XXXX
key=CALLEND	Press the Cancel or Reject soft keys.

Music on Hold

Music on hold (MOH) is the business practice of playing recorded music to fill the silence that would be heard by telephone callers who have been placed on hold. To use this feature, you should specify a SIP URI pointing to the Music on Hold Server account. When a call is put on hold, the phone will invite this SIP URI to call the held phone to play music on hold. The Music on Hold Server is an application acting as a SIP client which automatically answers to SIP INVITE messages and immediately plays audio from any source located anywhere (LAN, Internet).

To configure Music on Hold via Web interface:

- 1) Click on **Account** -> **Advanced**;
- 2) Enter the value of **Music on hold server**;

Music on hold server

sip:201@192.168.1.10

- 3) Click **Confirm** to save your configuration or **Cancel** to cancel.

Note:

1. Syntax of the Music on hold server is the same as which of SIP URI.
2. All involved parties can not use encrypted RTP.

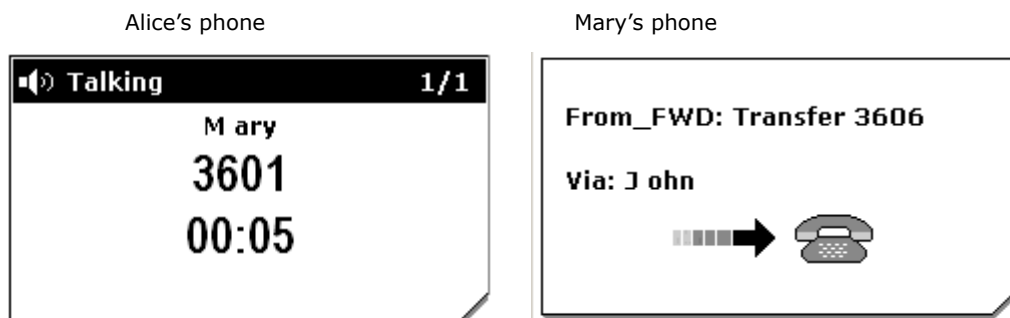
SIP Diversion Display

When an outgoing call from the phone is being diverted to another party (i.e., via call forward), the phone will display the Caller ID of the new party. Similarly, the new party will display the Caller ID of the original caller and the redirected call information indicated by DIVERSION or HISTORY-INFO header.

Call Diversion example:

- 1) Alice calls John.
- 2) John's phone is busy.
- 3) John's phone forwards the incoming call to another party (Mary).
- 4) Alice's phone displays the Caller ID of the new party (Mary)
- 5) Mary's phone receives a call and displays the caller ID of the initiator (Alice) and the diverted party (John).

See example of the phones' LCD below:



Alice@ 3606 calls John's phone which is busy and diverts to Mary.

Mary @ 3601 receives the incoming call from Alice diverted from John.

Note:

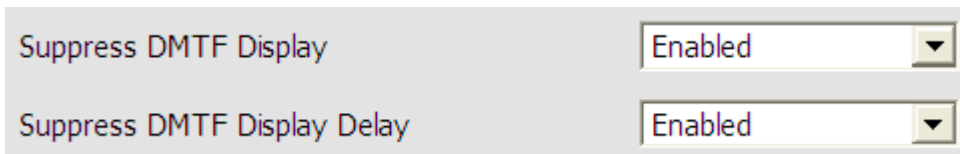
1. Diversion is not available on all servers. Please contact your system administrator for more information.
2. Redirected call information is sent by SIP message in the Diversion header, per draft-levy-sip-diversion-08, or the HISTORY-INFO header, per RFC 4244. It depends on the server.
3. Multiple diversions may take place on the phone. The phone receiving a diverted call (i.e., Mary's phone in example above) with the Diversion header will display the information of the last diverted party (last encountered); the phone receiving a diverted call (i.e., Mary's phone in example above) with the History-Info header will display the information of the first diverted party (first encountered).

Suppress DTMF Display

This feature allows the suppression of DTMF display when a number is dialed during an active call. When the Suppress DTMF Display is disabled, and you press number keys during an active call, the IP phone displays each digit as dialed in the LCD interface. When the Suppress DTMF Display is enabled, the IP phone will display “*” instead of dialed digits immediately in the LCD interface during the call. In addition, you can enable the Suppress DTMF Display Delay to display “*” for a dialed digit with a 400ms delay.

To configure Suppress DTMF Display via Web interface:

- 1) Click on **Phone -> Features**
- 2) Select **Enabled** in the pull-down menu of both **Suppress DTMF Display** and **Suppress DTMF Display Delay**



The screenshot shows a web interface with two configuration items. The first item is 'Suppress DTMF Display' with a pull-down menu set to 'Enabled'. The second item is 'Suppress DTMF Display Delay' with a pull-down menu also set to 'Enabled'.

- 3) Click **Confirm** to save your configuration.

Tone Settings

You can use the country tone, or if you don't want to use the default one, you can custom it by yourself.

You can define the frequency and time period of all the following tones :

- Dial
- Ring Back
- Busy
- Congestion
- Call Waiting
- Dial Recall
- Record
- Info
- Stutter
- Message
- Auto Answer

To edit the tone filed via Web interface:

- 1) Click on **Phone->Tones**.
- 2) Enter the **frequency** and **time period**(in ms) as the following format:
Frequency /Time Period (for example 400/200).
- 3) Press **Confirm** button to save the changes, or **Cancel** to cancel the change.

Note:

1. Please contact your system administrator for more information about the frequency and time period parameters. You can enter up to 8 groups for each tone.
2. If the frequency is set as 0, it means silence.

Voice

To edit the Voice filed via Web interface:

- 1) Click on **Phone->Voice**.
- 2) Set the following parameters shown in the table.

Parameter	Description
<i>Echo canceller</i>	Defines whether to enable the echo canceller.
<i>VAD</i>	Voice activity detection (VAD), also known as speech activity detection or speech detection, is a technique used in speech processing in which the presence or absence of human speech is detected.
<i>CNG</i>	A comfort noise generator (CNG) is a program used to generate background noise for voice communications during periods of silence that occur during the course of conversation.
Parameter	Description
<i>JITTER BUFFER</i>	It is a shared data area where voice packets can be collected, stored, and sent to the voice processor in evenly.
<i>Type</i>	To select the type of JITTER BUFFER, adaptive or Fixed.
<i>Delay</i>	To set the Min Delay, Max Delay and Normal Delay parameter.

The screenshot shows the 'Phone' tab selected in the top navigation bar. Below it, the 'Voice' sub-tab is active. The main content area is divided into two sections: 'Echo Cancellation' and 'JITTER BUFFER'. The 'Echo Cancellation' section has three dropdown menus: 'Echo canceller' (set to 'Enabled'), 'VAD' (set to 'Disabled'), and 'CNG' (set to 'Enabled'). The 'JITTER BUFFER' section has a 'Type' dropdown (set to 'Adaptive'), and three text input fields for 'Min Delay' (0), 'Max Delay' (300), and 'Normal Delay' (120). At the bottom of this section are 'Confirm' and 'Cancel' buttons. On the right side, there is a 'NOTE' section with three items: 'VAD' (Voice Activity Detection), 'CNG' (Comfort Noise Generation), and 'JITTER BUFFER' (It is a shared data area where voice packets can be collected, stored, and sent to the voice processor in evenly).

- 3) Press **Confirm** button to save the changes, **Cancel** to cancel the changes.

Ring

Users can group your contacts, and then set the ringing tone for each group.

To edit the Ring option via Web interface:

- 1) Click on **Phone->Ring**.
- 2) **Internal Ringer Text:** To set group name. For example, family.
- 3) **Internal Ringer File:** To select a special ring tone for the group.

4) Click the **Confirm** button to save the changes.

Logout

Status

Account

Network

Phone

Contacts

Upgrade

Security

PreferenceFeaturesSoftkey LayoutDSS KeyEXT KeyAction URLVoiceRingTonesDial PlanSMS

1	Internal Ringer Text	<input type="text"/>	<div>NOTE</div>
	Internal Ringer File	<div>Ring1.wav</div>	
2	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	
3	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	
4	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	
5	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	
6	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	
7	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	
8	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	
9	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	
10	Internal Ringer Text	<input type="text"/>	
	Internal Ringer File	<div>Ring1.wav</div>	

Confirm

Cancel

Trouble Shooting

I can not register to the server?

- 1) Check the IP address. If you set your WAN port in DHCP mode, please make sure that your DHCP server is on.
- 2) Check your gateway.
- 3) Check your DNS server.
- 4) Make sure your account information is the same as you have got from your ISP.
- 5) Check whether the SIP server is on.
- 6) Check the SIP register port, the default value is 5060.

I can't get the IP address?

- 1) Make sure you have plugged the Ethernet cable into the WAN port.
- 2) Make sure that the DHCP server is on, and there are available IP addresses in the server.
- 3) Try to set your WAN port to static IP client mode.

During a call, I cannot hear any voice?

- 1) Make sure your handset is tightly connected with the phone.
- 2) Check whether you have muted the conversation or not.
- 3) Consult the outbound server details with your ITSP.

Have DTMF problem?

- 1) Check which kind of DTMF you are using, and whether it is compatible with the server.
- 2) Consult the payload value with your ITSP.

How to change the time?

Select the time zone or enter the time information manually on the webpage or the phone.

How to answer the incoming calls during a call?

If a call comes in when you are in a conversation, press the Answer hot key to answer the call, or press the Reject hot key to refuse it.

How to refuse incoming calls during a call?

You can turn off the function of call waiting, and then our phone will refuse all the incoming calls when you are in a conversation.

How to send SMS?

You could edit the SMS in the Menu-> Messages->Text Messages.

Note:

Make sure that the SIP server you have registered supports SMS function.

How to update the firmware?

- 1) Enter the webpage of your phone, go to Upgrade, then you can find the option "Select and Upgrade Firmware" at the bottom of the page.
- 2) Select the file to update, then click the Upgrade button.

Note:

Make sure the firmware you select is provided by your service provider, or the device will probably crash after the update.

The manual is only for reference; WELL reserve the right to improve or change the product and the user guide without notice.

You can download the latest user manuals from our website:

<http://www.well.cz>

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