

WELL SIP-T20/T20P 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 fashion 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 configure 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 2

Getting Started

Packing List

The following components are included in your package:

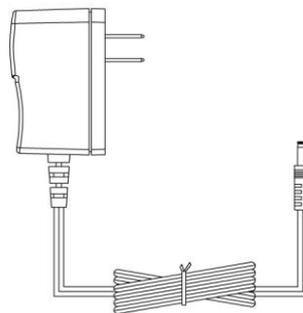
- Enterprise IP Phone



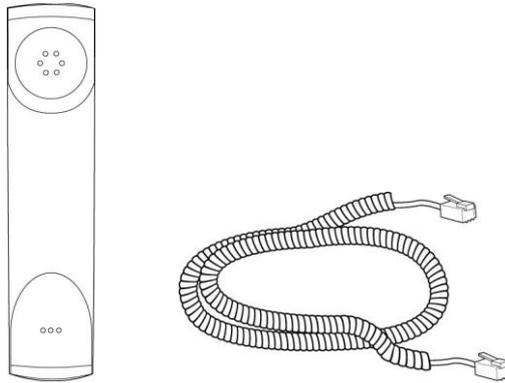
- Phone Stand



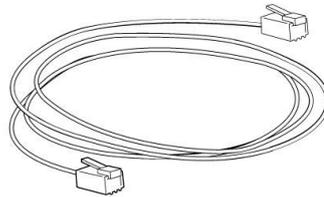
- Power Adapter



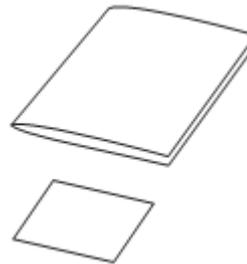
- Handset & Handset Cord



- Ethernet Cable



- Quick Installation Guide & Quick Reference



- CD Content



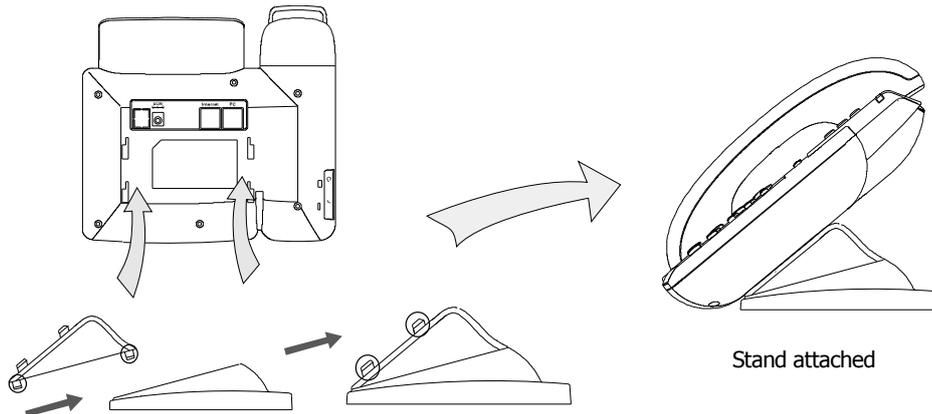
Check this list before installation to ensure that you have received each item. If you are missing any items, contact your IP phone reseller.

Assembling the Phone

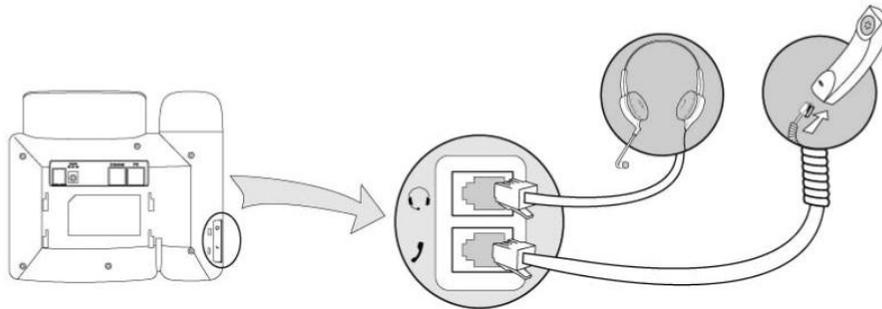
This section introduces how to assemble the phone with the components in the packing list :

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

1) Attach the Stand, as shown below:



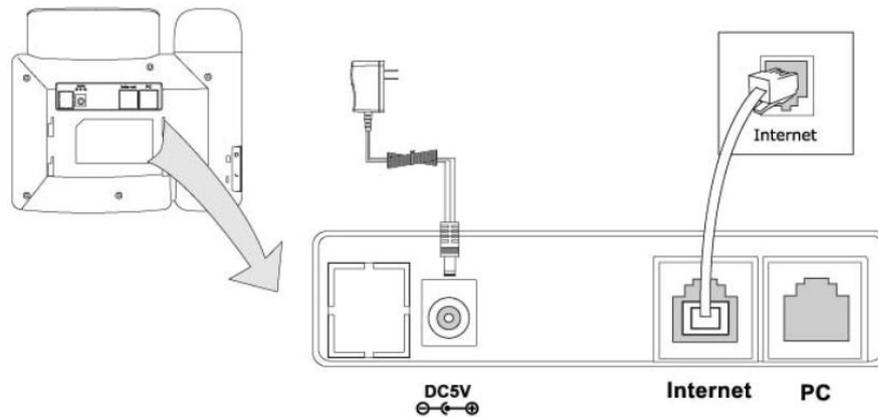
2) Connect Handset and Headset, as shown below:

**Note:**

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

3) Connect Network and Power

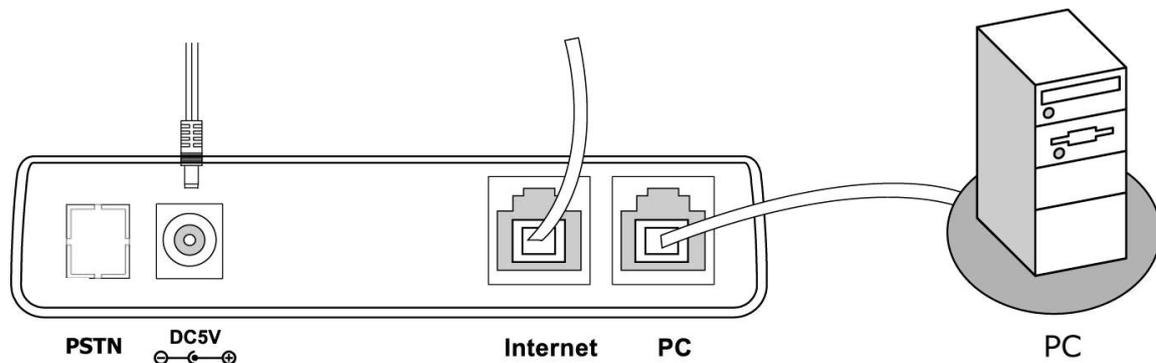
There are two ways for network and power source connections. Users can either connect the phone to the AC Power directly using the power adapter or to a PoE compliant switch or hub. Your system administrator will advise you on which one to use.



Note:

1. If inline power is provided, do not install AC adapter. Make sure the Ethernet cable and switch/hub are 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 with your system administrator or system administrator 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://192.168.3.28>) 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 Portn (WAN)**.

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

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

PPPoE: If you are using the xDSL Modem, users 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 network administrator.

Account Settings

The phone attempts to register to the SIP server using the account data provided 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>	Users can select 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 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 configure the register power up time so that the phone will random register automatically within the configured time.

To configure the power up time via web interface:

Select **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 user interface.

Configuring via keypad

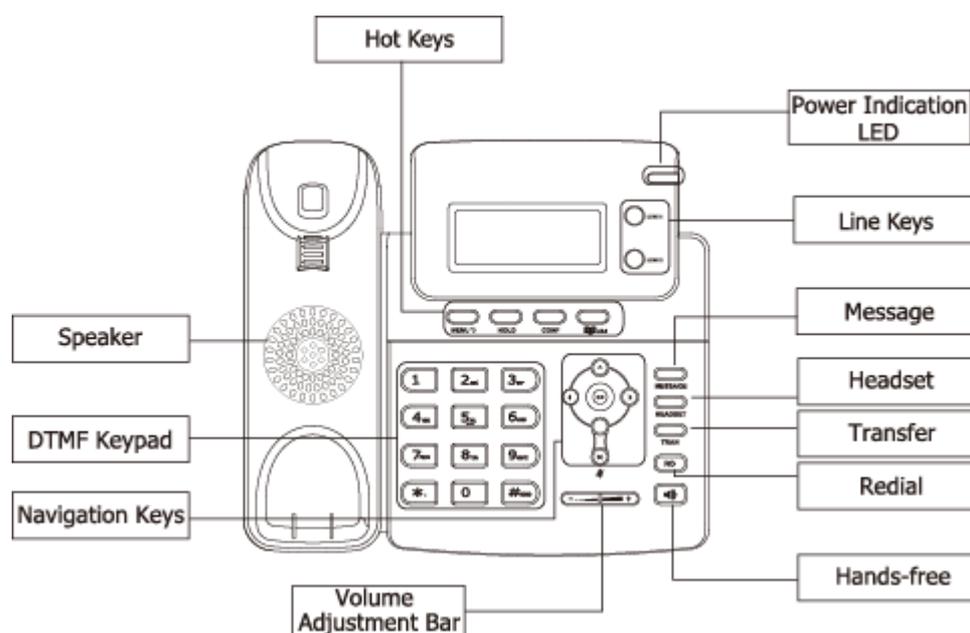
Network Settings: Press **MENU->Settings->Advanced**, enter the password, and select **Network->WAN Port/PC Port /VLAN/Web Type/802.1x Settings** to enter the internet relating configuration page.

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

Users can refer to "**Configuring via Web Page**" for the parameter details.

Overview

Keypad Instruction



Users 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

In different interfaces, different functions of the hot keys.

Line Keys

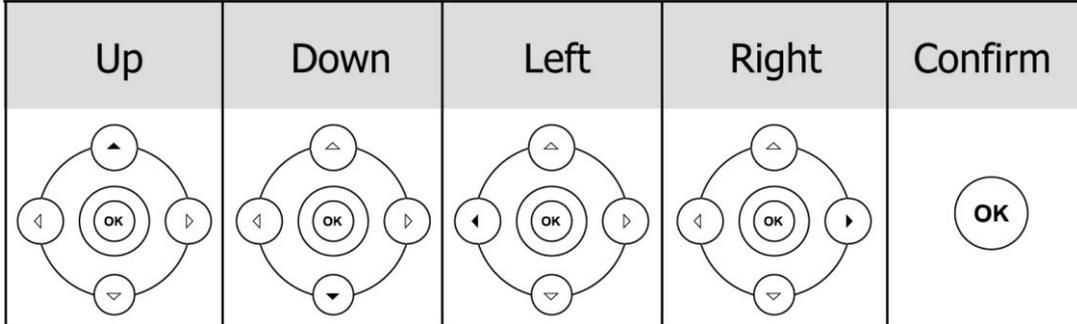
This buttons are used to active up to the two user accounts.

DTMF Keypad

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

Navigation Keys

Use the navigation keys to navigate in the display Menus and Confirm/cancel actions.



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 and ring tone;

: Allows for hands-free communication during calls;

Press to switch to the Group Listening mode.

: Place and receive calls through an optionally connected headset;

: Mute audio transmission locally during calls;

Hard Feature Keys

: Allow users to access the voicemail directly;

: Forward the current call to the third party;

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

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

LED Instruction

Table 1 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 2 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 a conversation
Off	It is not active as BLF

Table 3 Power Indication LED

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

Icon Instruction

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

Icon	Description
	Flashes when the internet is disconnected
	Missed calls
	Call in
	Call out
2aB	Input Method: all letters and numbers
123	Input Method: numbers
abc	Input Method: multi-lingual letters in lower case
ABC	Input Method: multi-lingual letters in upper case
	Call mute
HOLD	Call hold
	Voicemail
Icon	Description

	Call forward
	Auto answer
DND	DND
	Keypad Lock
	In handset mode
	In headset mode
	In speaker mode

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 possible to use both the user interfaces to operate the phone and change settings; some, however, are only possible via a phone or web user interface. Please refer to the following table for differences:

Phone Options	Phone UI	Web UI
Status		
--IP		✓
--MAC		✓
--Firmware	✓	✓
--Network		✓
--Phone		
--Accounts		
Call Features		
--Forward	✓	
--Call Waiting	✓	
--DSS Keys	✓	
--Key as Send	✓	
--Hot Line	✓	
--Anonym Call	✓	
--Auto Redial	✓	
--DND	✓	
--ReDialTone		✓
--Emergency		
--BusyToneDelay		
--Return code when refuse		
--Return code when DND		
--Dial Plan		
--Tones		
--Intercom	✓	
--Call Completion	✓	

Enterprise IP Phone**Overview**

Basic Phone Functions --Language --Time & Date --Ring Tone --Phone Volume	 √ √ √ √	 √
Advanced Phone Functions --Accounts --Network --Keypad Lock --Reset to Factory --Set AES Key --Set admin PWD --Voice --Upgrade	 √ √ √ √ √ √	 √
Other Features --Messages --History --Directory	 √	 √ √ √

Note:

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

The default administrator password is **admin**.

Customizing Your Phone

General Settings

Phone Status

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

This option allows you to review:

- IP;
- MAC;
- Firmware;
- Network: MAC, WAN, LAN, Gateway, DNS, etc;
- Phone: Model, Hardware, Firmware, Product ID and MAC;
- Accounts: The 2 SIP accounts status;

To check the Phone Status via Phone interface:

- 1) Press **OK** button directly to check the IP.
- 2) Use the **navigation keys** to check the other information.

I . I P: 192 . 168 . 0 . 120

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 **admin/admin**), select **Status** directly to check the status.

Language

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

It also supports French, German, Italian, Polish, Turkish, Portuguese, Spanish, etc. Users 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->Language**.


 A screenshot of a phone's language selection menu. It shows two options: "1. English" and "2. Deutsch". The first option is highlighted with a white background.

- 2) Scroll through the list of available languages.
- 3) Press **OK** button when the desired language is highlighted. The language appears on the graphic display will be changed to the one you select.
- 4) Press **MENU** 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 call server, please contact your system administrator if the time or date is incorrect. Users can configure 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->Time & Date.**


 A screenshot of the "Time & Date" settings menu. It shows two options: "1. SNTP" and "2. Manual". The first option is highlighted with a white background.

- 2) If **SNTP** is selectn, the phone will automatically get the time from the specific NTP Server. Use the navigation keys to highlight the specific options and do the relating changes. Users can configure the **Time Zone, NTP Server1/Server2,** and **DST** respectively.
- 3) If **Manual** is selectn, the time can be configured manually. Use the **navigation keys** to highlight the option and enter the specific date and time.


 A screenshot of the manual date entry screen. It shows the text "1. Date [D- M- Y] :" followed by the date "18-10-10". The date is highlighted with a white background.

- 4) Press **OK** button, the time appears on the idle screen will be changed. Or press **Menu** key to return to the previous screen.

To configure the time format via the Phone interface:

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


 A screenshot of the time format selection menu. It shows the text "1. Clock:" followed by two options: "12 Hour" and "24 Hour". The "24 Hour" option is highlighted with a white background.

- 2) Use the **Left/Right navigation keys** to select a preferred time format: 12 hour or 24 hour.
- 3) Press **OK** button to save the changes and return to the previous screen.

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

- 1) Click on **Phone->Preference->Daylight Saving Time** to do the relating changes.
- 2) Select **Enable** option, then users can configure 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.

The screenshot displays the 'Phone' configuration page. The left sidebar contains tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. The main content area is divided into a settings list on the left and a 'NOTE' section on the right. The settings list includes: 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 (1000), Daylight Saving Time (Automatic), Fixed Type (Automatic), Start Time (Month, Day, Hour), End Time (Month, Day, Hour), Offset (minutes), Manual Time (Disabled), Time Format (24 Hour), Live Dialpad (Disabled), Inter Digit Time (4), Flash Hook Time (1), Keyboard Lock (Disabled), WatchDog (Enabled), Ring Type (Ring1.wav), and Upload Ringtone. The 'NOTE' section contains the following information:

- 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 activate 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.

Note:

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

Keypad Lock

Users 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. Users can lock the following specific keys:

Menu Key: The Menu keys can not be used until unlocked. Users can not access the Menu of the phone.

Function Keys: The hard function keys can not be used until unlocked. Users can not access the MESSAGE, HEADSET, TRAN, RD, Speaker, CONF, HOLD, Navigation Keys, OK button, Mute button, Volume Adjustment button, etc.

All Keys: All of the keys can not be used until unlocked. Users can only use the phone to answer the incoming calls.

Lock&Answer: Users can only use the phone to answer the incoming calls (But can not hung up the call by your party) . Or enter the Menu pages to do some configurations.

To enable keypad lock via Phone interface:

1) Press **MENU->Settings->Advanced**, enter the password, and then press **OK** button.

- 2) Select **Keypad Lock**.
- 3) Use the **navigation keys** to highlight the one you want to lock.



- 4) Press **OK** button to active the change, or **Menu** to return to the previous screen.
- 5) The icon  will be displayed on the top right corner of the idle screen.
- 6) If you select **Lock&Answer**, it will show the icon  and AA on the user interface.

To unlock the phone via Phone interface:

- 1) Press **Menu** key, you are prompted for the password.
- 2) Enter the password, and then press **OK** button, the phone is unlocked.
- 3) If you select **Lock&Answer**, you have to enter **MENU->Settings->Advanced-> Keypad Lock** interface 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 instructions above for the parameters' detail.

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 checked="" 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	?	
Live Dialpad	Disabled	?	
Inter Digit Time(1~14)(seconds)	4	?	
Flash Hook Time(<800ms)	1	?	
Keyboard Lock	Disabled	?	
WatchDog	Disabled		
Ring Type	Menu Key		
Upload Ringtone	Function Keys		
	All Keys		
	Lock&Answer		
	Del	?	
	浏览...		
	Upload	Cancel	
	Confirm	Cancel	

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

Note:

1. The default password for unlock is **admin**.
2. Users can make emergency calls when the phone is locked.

Audio Settings

Volume

Users can adjust the volume of handset/speaker/headset.

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

- 1) Press **Menu->Settings->Basic**.
- 2) Scroll to **Phone Volume**, and press **OK** button, highlight the one you want to adjust the volume, use the **Volume Adjustment Bar** or **navigation keys** to adjust the volume.
- 3) Press **OK** button to save the contrast change or **Menu** key to cancel.
- 4) And users 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.

Ring Tones

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

To adjust the Ring Tone Type via Phone interface:

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

* 1. Ri ng 1. wav
2. Ri ng 2. wav

- 4) Press **OK** button to save the contrast change or **Menu** to cancel.

To adjust the volume via Phone interface:

- 1) Press **Menu ->Settings->Basic**.
- 2) Press **Phone Volume->Ring Volume**, use the **Volume Adjustment Bar** or **navigation keys** to adjust the volume.
- 3) Press **OK** button to save the contrast change or **Menu** 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 change. Users can also delete the specific one by clicking the **Delete** button.

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 checked="" 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	?
Live Dialpad	Disabled	?
Inter Digit Time(1~14)(seconds)	4	?
Flash Hook Time(<800ms)	1	?
Keyboard Lock	Disabled	?
WatchDog	Enabled	
Ring Type	Ring1.wav	Del ?
Upload Ringtone	<input type="text"/> <input type="button" value="浏览..."/> <input type="button" value="Cancel"/>	

Confirm

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 activate 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

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 **Browse** button to select the specific ring tone file.
- 3) Click **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 can not 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 selectn account in the scroll-down menu, then click **Confirm** button to update the change.

The screenshot shows a configuration page for an IP phone. The 'Ring Type' dropdown menu is open, displaying a list of ringtone options. The 'common' option is currently selected. Below the dropdown, there are buttons for 'Confirm' and 'Cancel'. To the right of the configuration fields, there is a sidebar with additional information and sections for 'NAT Traversal', 'Proxy Require', 'Codecs', and 'Advanced'.

Register Name	378	?
User Name	378	?
Password	•••	?
SIP Server	10.1.4.45	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 >>

Confirm **Cancel**

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

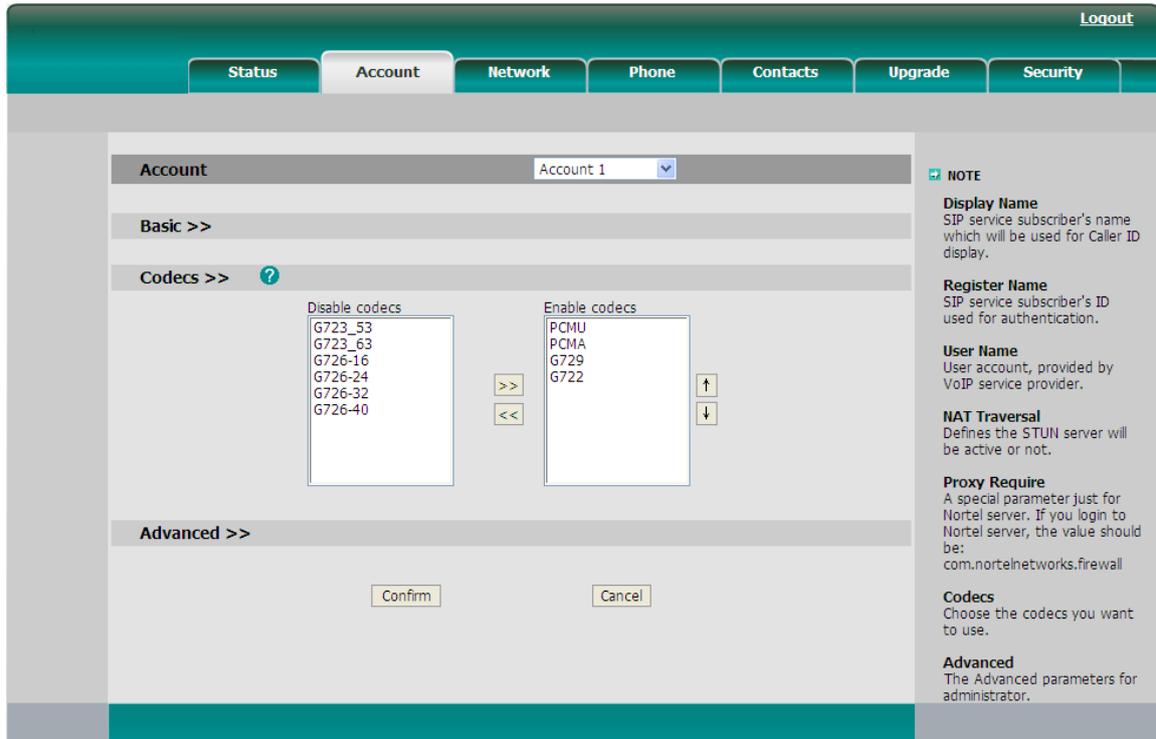
The IP phone supports the following voice codecs:

G722, G723_53, G723_63, G726-16, G726-24, G726-32, G726-40, G729, PCMU and PCMA

Users 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** button to save the change.

Note:

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

Contact Management

Configure/Add/Delete Contact

Users can store a large number of contacts in your phone's directory. Users can add, configure, delete, dial, or search for a contact in this directory.

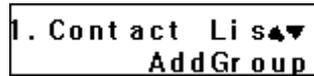
The directory includes Local Directory and Blacklist.

1. Local Directory
2. Blacklist

To add a Group via Phone interface:

- 1) Press **Menu->Directory->Local Directory**
- 2) Use **left/right navigation keys** to select **AddGroup**, press **OK** button to enter to the **Add Group** page.
- 3) Enter the group name and select the ring.

- 4) Press **OK** button to save.



To add a contact via Phone interface:

- 1) Press **Menu->Directory->Local Directory**.
- 2) Select a group, and press **OK** button.
- 3) Press **Up/Down navigation key** to ADD page. Press **OK** button to start adding a new contact, enter **Name, phone number** of the contact by the keypad. Use the **2aB** to select between numeric and upper/lower case alphanumeric modes.
- 4) Select and configure a special ring tone for the contact.
- 5) Use the **navigation keys** to select the group which you want to configure.
- 6) Press **OK** button to add the contact, or **Menu** key to cancel the change.

To configure/delete a contact via Phone interface:

- 1) Press **Menu->Directory->Local Directory**.
- 2) Select a group, and press **OK** button.
- 3) Use the **Up/Down navigation key** to highlight the contact you want to configure/delete, then press **left/right navigation keys** to **Configure/Del** page.



- 4) Make the desired changes, press **OK** button, or press **Menu** return to previous menu.

To move a contact to the Blacklist via Phone interface:

- 1) Press **Menu->Directory->Local Directory**.
- 2) Select a group, and press **OK** button.
- 3) Use the **Up/Down navigation keys** to highlight the contact you want to move, then press **left/right navigation keys** to **M2B** page.
- 4) Make the desired changes, press **OK** button, or press **Menu** return to previous menu.
- 5) It will pop up a warning frame asking whether **Confirm** to move the contact.
- 6) Press **OK** button to confirm the operation, or press the **Menu** button to return to the directory.

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

- 1) Press **MENU->History**.
- 2) Use the **navigation keys** to highlight a record, then press the **CONTACT** hot key to enter the configure page.

To add/delete/configure the contact list via the Web interfaces:

Click on **Contacts** and then do the relating changes. Please refer to the instruction above for the parameters' detail.

The screenshot shows the 'Contacts' management interface. At the top, there are navigation tabs: Status, Account, Network, Phone, **Contacts**, Upgrade, and Security. Below the tabs, there are links for 'Local Phone Book', 'BlackList', and 'Phone Call Info'. The main area features a table of contacts with the following data:

Index	Name	Office Num	Mobile Num	Other Num	Account	Group
1	aa	147852			Auto	
2	qq	123			Auto	
3	ww	0599123			Auto	
4						
5						
6						
7						
8						
9						
10						

Below the table, there are controls for 'Page: 1', 'Prev', 'Next', 'Move To BlackList', 'Delete All', and 'Del'. There are also sections for 'Contacts' (Name, Number, Mobile Num, Other Num, Account, Ring, Group) and 'Group Information' (Group, Ring). At the bottom, there are buttons for 'Add', 'Edit', and 'Search'. On the right side, there is a 'NOTE' section with instructions for 'Add Contact', 'Delete Contact', 'Import', and 'Export'.

To search a contact via Phone interface:

- 1) Press the **CONTACT** hot key to enter the contact list.
- 2) Press the digital number on the keypad, it will turn to the search interface automatically, and search the record in **Contact**.

Import/Export Contact list

Import/Export Contact List via Web interface:

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

The screenshot shows the 'Local Phone Book' interface. At the top, there are tabs for 'Status', 'Account', 'Network', 'Phone', 'Contacts', 'Upgrade', and 'Security'. The 'Contacts' tab is active. Below the tabs, there are links for 'Local Phone Book', 'BlackList', and 'Phone Call Info'. The main content area is titled 'Contacts' and shows a table with the following data:

Index	Name	Office Num	Mobile Num	Other Num	Account	Group
1	aa	147852			Auto	
2	qq	123			Auto	
3	ww	0599123			Auto	
4						
5						
6						
7						
8						
9						
10						

Below the table, there are buttons for 'Page: 1', 'Prev', 'Next', 'Move To BlackList', 'Delete All', and 'Del'. To the right of the table, there is a 'NOTE' section with the following text:

Add Contact
Put in the informations about contact. User shouldn't leave contact name blank.

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

Import
Browse the file in XML format.

Export
Click Export button and create a file with the name you like.

Below the table, there are input fields for 'Name', 'Number', 'Mobile Num', 'Other Num', 'Account', 'Ring', and 'Group', along with 'Add', 'Edit', and 'Search' buttons. There are also sections for 'Group Information' and 'Please select the contacts list file' with file selection buttons and 'Import XML', 'Export XML', 'Import CSV', and 'Export CSV' buttons.

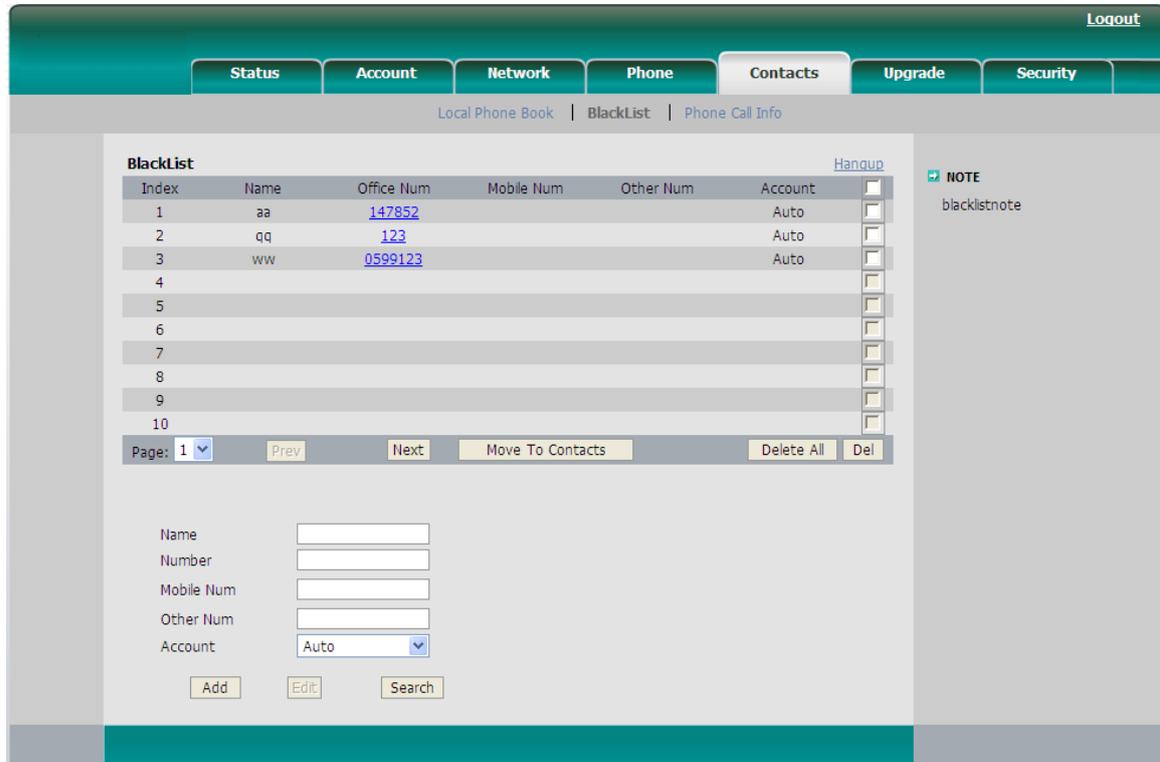
- 2) Browse the specific contact list file in .XML format or .csv format, and then click **Import** button. The imported contact lists will be shown in the directory.
- 3) Move the mouse to the icon ; you will see the notes for parameters. The meanings of this icon on other pages is the same, we will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

Note:

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

Blacklist

If you add a contact to blacklist, then the call from this contact cannot get through. The operation of blacklist is the same as contact.



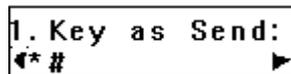
Other Settings

Key as Send

Users can configure a specific button (“#” or “*”) to active as the send button.

To configure the send key via the IP phone interface:

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



- 2) Press the **Left/Right navigation keys** to select a button that you want to use as the send key: “#”, “*”, or disable this option.
- 3) Press **OK** button to save the changes.

To configure the send key via the Web interface:

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

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

The screenshot displays the 'Phone' configuration page in a web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Phone', 'Contacts', 'Upgrade', and 'Security'. Below this, a secondary menu shows 'Preference', 'Features', 'DSS Key', 'Action URL', 'Voice', 'Ring', 'Tones', and 'Dial Plan'. The main content area is divided into several sections:

- Forward:** Includes sub-sections for 'Always', 'Busy', and 'No Answer'. Each sub-section has 'On' and 'Off' radio buttons, a 'Target' text field, and 'On Code' and 'Off Code' text fields.
- General Information:** Contains dropdown menus for 'Call Waiting' (Enabled), 'Call Waiting Tone' (Enabled), 'Auto redial' (Disabled), 'Key As Send' (set to '#'), 'Reserve # in User Name' (Enabled), 'Button Sound' (Enabled), and 'Send Sound' (Enabled).
- NOTE:** A sidebar on the right provides explanatory text for 'Forward', 'Target', 'On Code', 'Off Code', 'Call Waiting', 'Key As Send', and 'Hotline Number'.

A tooltip is visible over the 'Key As Send' dropdown, stating: 'Set keys as call out function'.

Hot Line

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

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

1. Hot Number :
1 2 3

2) Enter the **hot line number** and configure the **HotLine Delay** (for example, 20 seconds), then press **OK** button to save the changes.

3) 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 configure the Hot Line via the Web interface:

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

2) Input the **Hotline Number** and **Hotline Delay**, and then click **Confirm** button to save the change.

On Code	<input type="text"/>	?	
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)	10	?	
Target	<input type="text"/>	?	
On Code	<input type="text"/>	?	
Off Code	<input type="text"/>	?	
General Information:			
Call Waiting	Enabled	?	
Call Waiting Tone	Enabled	?	
Auto redial	Disabled	?	
Key As Send	#	?	
Reserve # in User Name	Enabled	?	
Button Sound	Enabled	?	
Send Sound	Enabled	?	
Hotline Number	<input type="text"/>	?	
Hotline Delay	4	?	Set hotline number
ReDialTone	<input type="text"/>	?	
Emergency	<input type="text"/>	?	
BusyToneDelay(seconds)	0	?	
Ringer Device for Headset	Use Speaker	?	
Headset Send Volume (1~53)	29		

to forward an incoming call to another phone number.

Target
The number to which the incoming calls will be forwarded.

On Code
The code that will be sent to PBX when it is switched On.

Off Code
The code that will be sent to PBX when it is switched Off.

Call Waiting
This call feature allows your phone to accept other incoming calls during the conversation.

Key As Send
Select * or # as the send key.

Hotline Number
When you pick up the phone, it will dial out the hotline number automatically.

Upload Logo
The picture must be format of dob, it can be black and white, or 2 gray scale.

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	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**;

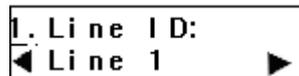


- 3) Click **Confirm** to save the configuration.

Anonymous call

To configure the anonymous call via the IP phone interface:

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



- 2) By the **navigation keys**, users can select the **Line ID**.
- 3) Press the **navigation keys** to select whether to enable the anonymous call function. This feature allows the subscriber to make a call with the display of their calling identification information blocked.
- 4) If you want to realize this function by server, please select and enter the **Anonym On Code** and **Anonym Off Code**. When you select 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 the subscriber to reject 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 select to enable the Rejection option on your IP phone, it will send information to the server, and the server will enable/ disable the rejection anonymous call function for your IP phone automatically.

Note:

This configuration is only available for the current default account.

- 7) Press **OK** button to save the changes.

To configure 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) Click **Confirm** button to save the changes.

The screenshot shows the 'Account' configuration page for 'Account 1'. The 'Basic' tab is active, displaying various configuration fields. The 'Anonymous Call' field is set to 'Off'. A tooltip 'Enable/Disable anonymous call' is shown over the 'Anonymous Call' dropdown menu. On the right side, there is a 'NOTE' section with several informational items: 'Display Name', 'Register Name', 'User Name', 'NAT Traversal', 'Proxy Require', 'Codecs', and 'Advanced'.

Field	Value	Port	Help
Register Status	Registered		
Account Active	<input checked="" type="checkbox"/> On <input type="checkbox"/> Off		
Label	378		?
Display Name	378		?
Register Name	378		?
User Name	378		?
Password	...		?
SIP Server	10.1.4.45	5060	?
Enable Outbound Proxy Server	Disabled		?
Outbound Proxy Server		5060	?
Transport	UDP		?
Backup Outbound Proxy Server		5060	?
NAT Traversal	Disabled		?
STUN Server		3478	?
Voice Mail			?
Proxy Require			?
Anonymous Call	Off		?
On Code			?
Off Code			?
Anonymous Call Rejection	Off		?

Auto Redial

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

To configure auto redial via the IP phone interface:

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

1. Auto Redial:
◀ * Disable ▶

- 2) By the **navigation keys**, users can select whether to enable the **Auto Redial** function.
- 3) Press the **navigation keys** to select and configure the **Redial Interval**. It is measured by seconds.
- 4) Press the **navigation keys** to select and configure the redial times.
- 5) Press **OK** button to save the changes.

Note:

If you enable the auto redial function, then without 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.

The screenshot shows the configuration page for 'Auto Redial'. The 'Auto redial' dropdown menu is set to 'Disabled'. A tooltip points to this menu with the text 'Enable/Disable auto-redial'. The page is divided into sections: 'Busy', 'No Answer', 'General Information', and a 'Target' section on the right. The 'Target' section contains definitions for 'Target', 'On Code', 'Off Code', 'Call Waiting', 'Key As Send', and 'Hotline Number'. The 'Auto Answer' section is also visible at the bottom of the page.

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. Users can configure this function to a special account.

To configure Auto Answer via the IP phone interface:

- 1) Press **Menu->Settings->Advanced**, enter the password and press **OK** button
- 2) Then press **Accounts->Line X** to enter the configuration page, use the **navigation keys** to select Auto Answer option.

12. Auto Answer :
 ◀ * Disable ▶

- 3) Use the **navigation keys** to enable or disable the auto answer function. The default is Disable.
- 4) Press **OK** button 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.
- 3) Click **Confirm** button to save the change.

Register Name	<input type="text" value="378"/>	?	
User Name	<input type="text" value="378"/>	?	
Password	<input type="password" value="•••"/>	?	
SIP Server	<input type="text" value="10.1.4.45"/>	Port <input type="text" value="5060"/>	?
Enable Outbound Proxy Server	<input type="button" value="Disabled"/>	?	
Outbound Proxy Server	<input type="text"/>	Port <input type="text" value="5060"/>	?
Transport	<input type="button" value="UDP"/>	?	
Backup Outbound Proxy Server	<input type="text"/>	Port <input type="text" value="5060"/>	?
NAT Traversal	<input type="button" value="Disabled"/>	?	
STUN Server	<input type="text"/>	Port <input type="text" value="3478"/>	?
Voice Mail	<input type="text"/>	?	
Proxy Require	<input type="text"/>	?	
Anonymous Call	<input type="button" value="Off"/>	?	
On Code	<input type="text"/>	?	
Off Code	<input type="text"/>	?	
Anonymous Call Rejection	<input type="button" value="Off"/>	?	
On Code	<input type="text"/>	?	
Off Code	<input type="text"/>	?	
Missed call log	<input type="button" value="Enabled"/>	?	
Auto Answer	<input type="button" value="Disabled"/>	?	
Ring Type	<input type="button" value="common"/>	?	<input type="button" value="Enable/Disable auto-answer"/>
Codecs >> ?			
Advanced >>			
<input type="button" value="Confirm"/>		<input type="button" value="Cancel"/>	

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.

Missed call log

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

- 1) Click on **Account->Basic->Missed call log**.
- 2) Select **Enabled** or **Disabled** in the pull-down menu.

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

Register Status	Registered	
Account Active	<input checked="" type="radio"/> On <input type="radio"/> Off	
Label	<input type="text" value="378"/>	?
Display Name	<input type="text" value="378"/>	?
Register Name	<input type="text" value="378"/>	?
User Name	<input type="text" value="378"/>	?
Password	<input type="password" value="•••"/>	?
SIP Server	<input type="text" value="10.1.4.45"/>	Port: <input type="text" value="5060"/> ?
Enable Outbound Proxy Server	<input type="button" value="Disabled"/>	?
Outbound Proxy Server	<input type="text"/>	Port: <input type="text" value="5060"/> ?
Transport	<input type="button" value="UDP"/>	?
Backup Outbound Proxy Server	<input type="text"/>	Port: <input type="text" value="5060"/> ?
NAT Traversal	<input type="button" value="Disabled"/>	?
STUN Server	<input type="text"/>	Port: <input type="text" value="3478"/> ?
Voice Mail	<input type="text"/>	?
Proxy Require	<input type="text"/>	?
Anonymous Call	<input type="button" value="Off"/>	?
On Code	<input type="text"/>	?
Off Code	<input type="text"/>	?
Anonymous Call Rejection	<input type="button" value="Off"/>	?
On Code	<input type="text"/>	?
Off Code	<input type="text"/>	?
Missed call log	<input type="button" value="Enabled"/>	?
Auto Answer	<input type="button" value="Disabled"/>	? Enable/Disable missed call records
Ring Type	<input type="button" value="common"/>	?
Codecs >> ?		
Advanced >>		

which will be used for call 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.

Logo Customization

Users can configure text logo which shown in the idle screen

- 1) Click on **Phone**-> **Features**-> **Use Logo** via the Web interface, select **Enable** or **Disable** in the pull-down menu.

The screenshot displays a configuration page for various phone features. The 'Use Logo' setting is highlighted in blue and set to 'Enabled'. Below it is a text input field for 'Text Logo'. A tooltip box on the right side of the page provides additional information: 'Logo settings, you can select System logo, you can also choose Custom logo to upload your own logo'. At the bottom of the page, there are 'Confirm' and 'Cancel' buttons.

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	Disabled	
IsDeal180	Enabled	
Logon Wizard	Disabled	
PswPrefix		
PswLength	0	
PswDial	Disabled	
Use Logo	Enabled	?
Text Logo		

- 2) If you select **Enable**, and then enter the text logo in the **Text Logo** field.
- 3) Click **Confirm** button to save the change.

Programmable Key

The hot-key, navigation keys and function keys on the keypad are configurable. Users can customize specific features for these keys according to their actual needs. (The programmable keys can only be available when the phone is idle.)

This function can only be configured via the Web interface:

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

The screenshot shows the 'Programmable Key' configuration page. At the top, there are tabs for 'Status', 'Account', 'Network', 'Phone', 'Contacts', 'Upgrade', and 'Security'. Below these are sub-tabs for 'Preference', 'Features', 'DSS Key', 'Action URL', 'Voice', 'Ring', 'Tones', and 'Dial Plan'. The main content area is titled 'Line Key >>' and 'Programmable Key >>'. A table with columns 'Key', 'Type', 'Line', and 'Extension' is shown. A dropdown menu is open for the 'Type' column, listing various features like History, N/A, Directory, DND, Menu, SwitchAccount, Forward, Redial, Call Return, Pick Up, Remote Phone Book, Status, Speed Dial, Local Group, Local PhoneBook, and Forward. At the bottom, there are 'Confirm', 'Cancel', and 'Reset to Default' buttons.

Key	Type	Line	Extension
Up	History	Auto	
Down	N/A	Auto	
Left	Directory	Auto	
Right	History	Auto	
OK	DND	Auto	
Cancel	Menu	Auto	
CONF	SwitchAccount	Auto	
Hold	Forward	Auto	
TRAN	Redial	Auto	
	Call Return	Auto	
	Pick Up	Auto	
	Remote Phone Book	Auto	
	Status	Auto	
	Speed Dial	Auto	
	Local Group	Auto	
	Local PhoneBook	Auto	
	Forward	Auto	

- 2) Select and customize specific features for these keys.
- 3) Click **Confirm** button to save the changes.

Live Dialpad

Defines whether to dial out the dialed number automatically.

This function can only be configured 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 configuration page for an Enterprise IP Phone, specifically the 'Phone' tab. The interface includes a navigation bar with tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. Below this, there are sub-tabs for Preference, Features, DSS Key, Action URL, Voice, Ring, Tones, and Dial Plan. The main configuration area lists various settings:

- 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 By Week
- StartTime: Month [] Day [] Hour []
- EndTime: Month [] Day [] Hour []
- Offset(minutes): []
- Manual Time: Disabled
- Time Format: 24 Hour
- Live Dialpad: Disabled
- Inter Digit Time(1~14)(seconds): 4
- Flash Hook Time(<800ms): 1
- Keyboard Lock: Disabled
- WatchDog: Enabled
- Ring Type: Ring1.wav
- Upload Ringtone: []

On the right side, there is a 'NOTE' section with several explanatory text blocks:

- 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 activate 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, and 11.025K, 16-bit, mono, 11.025K, 16-bit, stereo, 11.025K, 16-bit, stereo, 22.05K, 16-bit, stereo, 22.05K, 16-bit, stereo, 44.1K, 16-bit, stereo, 44.1K, 16-bit, stereo, 48K, 16-bit, stereo, 48K, 16-bit, stereo, 96K, 16-bit, stereo, 96K, 16-bit, stereo, 192K, 16-bit, stereo, 192K, 16-bit, stereo.

A tooltip is visible over the 'Live Dialpad' setting, stating: '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 configure 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 configure a Dial Plan via the Web interface:

- 1) Click on **Phone->Dial Plan->Replace Rule**.

The screenshot shows the 'Replace Rule' configuration page. At the top, there are navigation tabs: Status, Account, Network, Phone (selected), Contacts, Upgrade, and Security. Below these are sub-tabs: Preference, Features, DSS Key, Action URL, Voice, Ring, Tones, and Dial Plan. The main content area is titled 'Replace Rule >>' and contains a table with the following data:

Index	Prefix	Replace	Account
1	05	0599123456	
2	1	123456	1
3			
4			
5			
6			
7			
8			
9			
10			

Below the table, there are input fields for 'Prefix', 'Replace', and 'Account', and buttons for 'Add', 'Edit', and 'Del'. To the right, a 'NOTE' section explains symbols: 'Digit 0-9 *' identifies a specific digit; '[digit-digit]' identifies any digit dialed in a range; '[digit-digit,digit]' specifies a range as a comma-separated list; 'x' matches any single digit/character; and '.' matches an arbitrary number of digits.

- 2) Enter the desired **Prefix**, **Replace** and **Account**.
- 3) Click **Add** button to save the changes.
- 4) Users can also delete a specific one from the dial plan list by clicking **Del** button.
- 5) Users can select a record to modify, then click **Configure** button to submit.

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 out the 123 whose length matches the Dial-now rule, the phone will dial out 123 immediately without pressing Send button.

To configure a Dial Plan via the Web interface:

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

The screenshot shows the 'Dial Plan' configuration page. At the top, there are tabs for 'Status', 'Account', 'Network', 'Phone', 'Contacts', 'Upgrade', and 'Security'. Under the 'Phone' tab, there are sub-tabs for 'Preference', 'Features', 'DSS Key', 'Action URL', 'Voice', 'Ring', 'Tones', and 'Dial Plan'. The main content area is titled 'Replace Rule >>' and 'Dial-now >>'. It contains a table with the following data:

Index	Dial-now Rule	Account
1	0599147	1
2	059914785239	2
3		
4		
5		
6		
7		
8		
9		
10		

Below the table, there are input fields for 'Dial-now Rule' and 'Account', and buttons for 'Add', 'Edit', and 'Del'. To the right, there is a 'NOTE' section with the following text:

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) Users can select a record to modify, then click **Configure** button to submit.
 - 5) Users can also delete a specific one from the dial plan list by clicking **Del** button.
 - 6) Users can also configure the **Time Out for Dial-now Rule** via web interface.
- Click on **Phone->Features->Time Out for Dial-now Rule**, enter the time.

Note:

1. If need to replace the unknown contents, then users 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**.



The screenshot shows a web interface for configuring phone settings. The top navigation bar includes tabs for Status, Account, Network, Phone (selected), Contacts, Upgrade, and Security. Below the navigation bar, there are sub-tabs for Preference, Features, DSS Key, Action URL, Voice, Ring, Tones, and Dial Plan. The main content area is titled 'Area Code>>' and contains the following fields:

Code	<input type="text" value="05989"/>
Min Length(1-15)	<input type="text" value="1"/>
Max Length(1-15)	<input type="text" value="15"/>
Account	<input type="text" value="2"/>

Below the fields are 'Confirm' and 'Cancel' buttons. There is also a 'Block Out>>' section with a '查看详细' button. A 'NOTE' section on the right provides definitions for various symbols used in dial plans:

- 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 **Code** and **Account**, configure the **Min Length** and the **Max Length**, and then click **Confirm** button to save.

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 the **phone number** and **Account** and click **Add** button to save the changes, or select the specific one in the list, click **Del** button to delete the record.
- 3) Users can select a record to modify, then click **Configure** button to submit.

4) Users cannot dial out the number from your IP phone unless it is removed from the forbidden List.

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.

Index	Block Out Number	Account
1	0147	1
2	014852	2
3		
4		
5		
6		
7		
8		
9		
10		

Block Out Number Account

Note:

1. In the Account field, users 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 configure 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, select **Intercom** from the list.
- 3) Select whether to enable this function from the pull-down menu.

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

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

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 checked="" 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	?
Live Dialpad	Disabled	?
Inter Digit Time(1~14)(seconds)	4	?
Flash Hook Time(<800ms)	1	?
Keyboard Lock	Disabled	?
WatchDog	Enabled Disabled Enabled	
Ring Type		
Upload Ringtone	<input type="text"/> 浏览...	?

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 activate 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

Basic Call Functions

Making a call

Call Devices

Users can make a phone call via the following devices:

- 1) Pick up the handset,  icon will be shown 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.

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

Call Methods

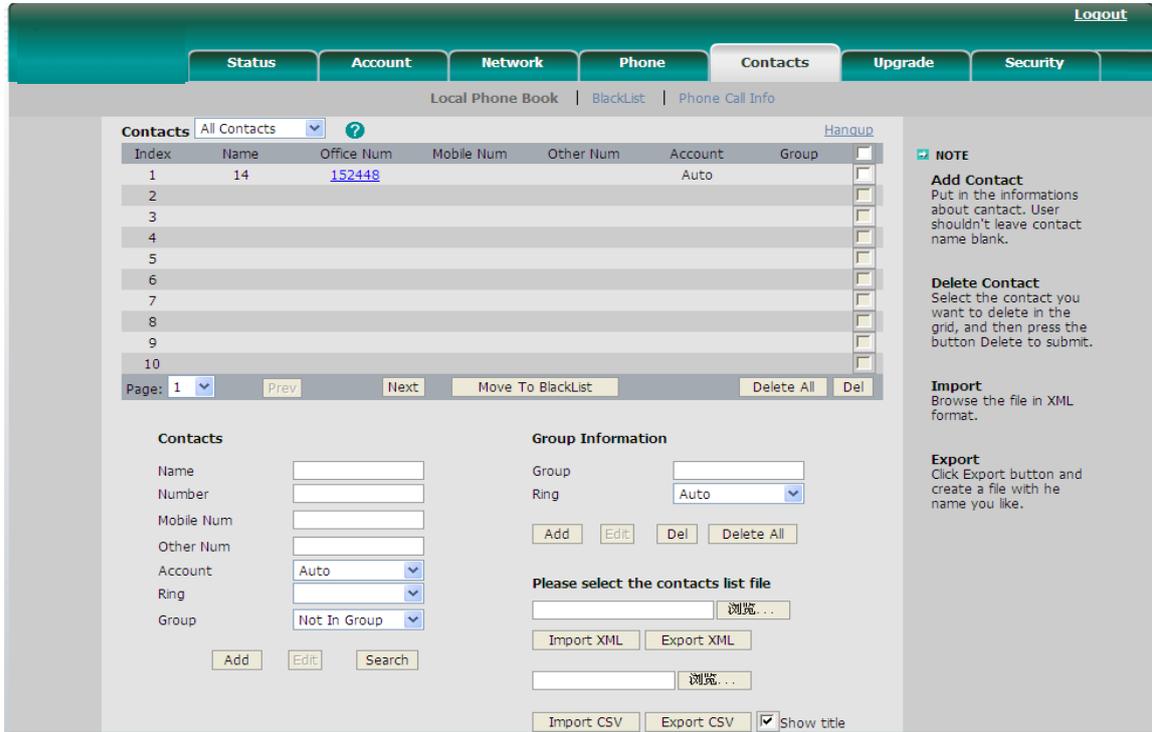
Users can dial the number directly if you didn't register a account but fill the SIP Server in the registered interface. But the number which you dial must be in the same SIP server.

Users can press an available line button if there is more than one account, then

- 1) Dial the number you want to call, or
- 2) Go to **Directory**, use the **navigation keys** to highlight your choice, or
- 3) Press the **Up navigation key** to enter the call history interface, then use the navigation keys to highlight your choice (press **Left/Right navigation keys** to select **All Calls**, **Missed Calls**, **Dialed Calls**, **Received Calls** and **Forwarded Calls**) or
- 4) Press the **RD** button to enter the **Dialed Calls** interface, and then select a record to dial out.
- 5) Then press the **SEND** button to dial out if necessary.

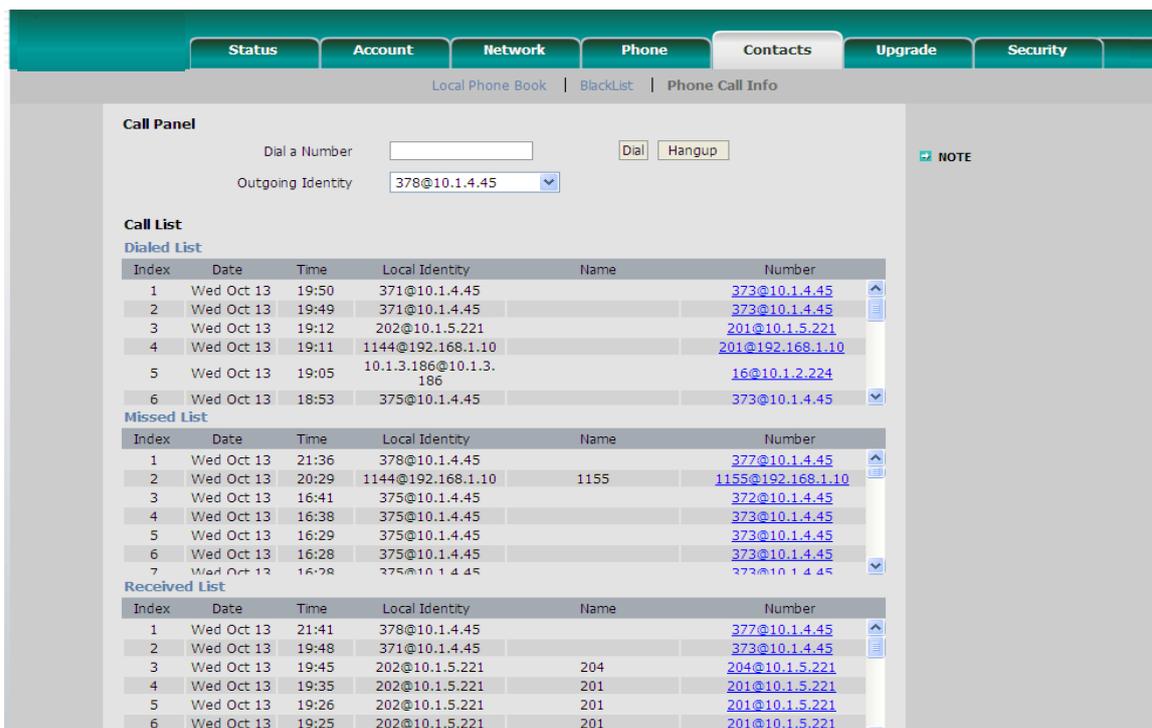
And users can also dial-up via web interface:

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



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

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



- 4) Users 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 configure the password prefix as 3, enter the PswLength as 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.

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		
PswLength	0	
PswDial	Disabled	
PushXML Server IP		
Use Logo	System Logo	?

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 end 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 **MENU->Features->Call Completion** to enter the configuration page.

- 2) By the **Left/Right navigation keys**, select whether to enable this option.
- 3) Press **OK** button to save your changes.

Answering a call

Answering an incoming call

- 1) If you are not on an active call, lift the handset using, 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, press the corresponding **line key** or **OK** button to answer it.

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

Denying an incoming call

Press the **X** key to deny the incoming call directly.

DND

If users enable the DND function, all the incoming calls will be rejected and the display shows: ^{DND} icon; Users can find the incoming call record in the Call History.

To configure the DND function via Phone interface:

- 1) Press **Menu->Features->DND** to enter the configuration interface.
- 2) Use the navigation keys to select **Enable/Disable**.
- 3) Press **OK** button to save the changes.
- 4) Select **DND** option again to deactivate DND mode.

Users can also configure DND function by the DND Code:

- 1) Press **Menu->Features->DND** to enter the configuration interface.
- 2) Configure the **DND On Code** and the **DND Off Code**, then press **OK** button to save the changes.
- 3) When you enable the **DND** function, 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. And the incoming call record will not be displayed in the Call History.

Call Forward

This feature allows you to forward an incoming call to another phone number e.g. 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 ring times.

To configure Call Forward via Phone interface:

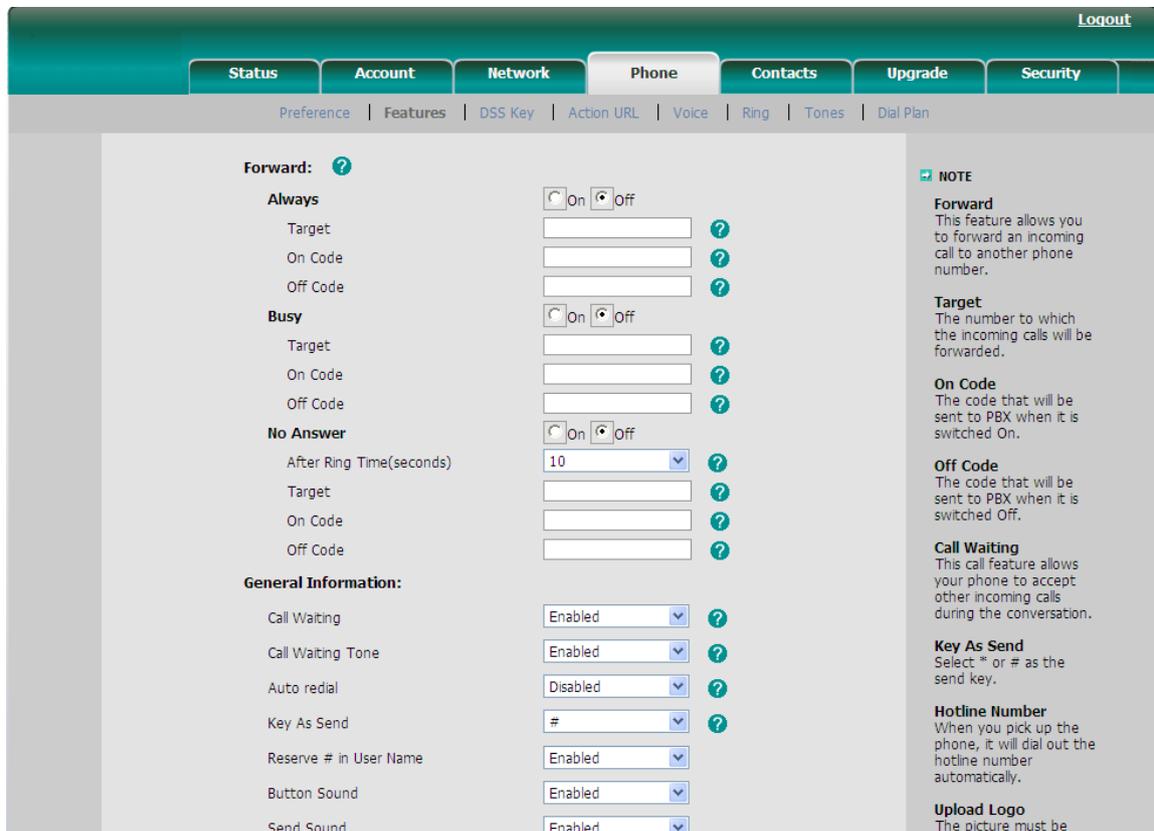
1) Press **Menu->Features->Forward**. Users can also press the **Tran** button or the corresponding **DSS key** to enter the forward setting page directly when the phone is idle.

- 1. Always
- 2. Busy

- 2) There are 3 options: **Always, Busy** and **No Answer**.
- 3) If you select one of them, enter the phone number you want to forward.
- 4) If you want to realize this function by server, please enter the **On Code** and **Off Code**, then when you select 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 configured number automatically based on the forward type. And the IP phone will not show the record in the call history anymore.
- 5) Press **OK** button to save the changes.

To configure Call Forward via Web interface:

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

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 **Menu**->**Features**->**Intercom** to enter the configuration page.
- 2) **Intercom Allow**: To configure whether to answer the incoming intercom calls.
- 3) **Intercom Mute**: To configure whether to mute the incoming intercom calls automatically.
- 4) **Intercom Tone**: To configure whether to play ring tones when there is incoming intercom calls to your extension.
- 5) **Intercom Barge**: To configure whether to answer the incoming intercom calls during a conversation. If the option is enabled, when there is incoming intercom calls to your extension, if 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) Select and configure the different options by **navigation keys**.
- 7) Press **OK** button to save your changes.

During an Active Call

Mute

This function allows you to mute the microphone of the active audio device during a call; users can not be heard by the other party. Users can still hear all other parties while mute is enabled.

To mute/resume the conversation:

Press  button during a conversation to mute the current call, the icon  will be shown on the LCD, and the power indication LED will blink. Press  again to get the microphone return to normal conversation. When you press the MUTE button all of the conversations will be muted

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 having a call on hold.

To hold/resume a call:

- 1) Press the **HOLD** button to put your active call on hold.
- 2) During the call, there will be a "dodo.." sound for each 30 second, suggesting that there is a current call in hold state.
- 3) If there is only one call on hold, press the **HOLD** button again to retrieve the call.
- 4) If there is more than one call on hold, press the **Up/Down navigation keys** or the corresponding **Line key** to highlight the call, then press the **HOLD** button again 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**.
- 2) Use the **navigation keys** to active/inactive **Call Waiting**.
- 3) Use the **navigation keys** to enable/disable the **Play Tone** option. This option used to define whether to play ring tones when there is call incoming during a call.
- 4) Press **OK** button to save the changes, or **Menu** to return to the previous menu.

To enable/disable Call Waiting via Web interface:

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

The screenshot shows the 'Call Waiting' configuration page in the IP Phone web interface. The page is titled 'Call Waiting' and is part of the 'Phone' features section. It includes sections for 'Forward' (Always, Busy, No Answer), 'General Information', and a 'NOTE' section. The 'Call Waiting' option is set to 'Enabled'. A tooltip is visible over the 'Key As Send' dropdown, which is set to '#'. The tooltip text reads: 'Busy waiting for the new incoming call switches * or # as the key.'

Users can also configure whether to open the **Call Waiting Tone** option in this page.

Call Transfer

Users 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-Attend Transfer.

To Blind Transfer via phone interface:

- 1) A and B is on an conversation, A press **TRAN** Button to put B on hold, then A can dial the third telephone number C and press the **TRAN** Button to call out. A will turn to hold status for a short while and then turn to the idle status.
- 2) After C answered it, or A press the **Menu** key to complete the transfer.
- 3) A will be disconnected from the call. B can talk to 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 Attended Transfer via phone interface:

- 1) A and B is on an conversation, A press **TRAN** Button to put B on hold, then A can dial the third telephone number C and press the **OK** or **SEND** button to call

out.

- 2) After C answered it, A and C can have a private conversation without B hearing it, then A press the **TRAN** button to complete the transfer.
- 3) A will be disconnected from the call. B can talk to C.

To Semi-Attend Transfer via phone interface:

- 1) A and B is on an conversation, A press the **TRAN** button to put B on hold, then A can dial a new number C and press the **OK** or **SEND** button to call out.
- 2) While C is ringing, A hang up or press the **TRAN** button. A will turn to hold status for a short while and then turn to the idle status.
- 3) You will be disconnected from the call, when C pick up, B can talk to C.

Note:

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

3-way Conference

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

To establish a conference:

- 1) Press the **CONF** button during an active call.
- 2) The first call is placed on hold. You will hear a dial tone. Dial the number to conference in, then press the **SEND** key.
- 3) Press the **Menu** key to stop the conference in operation before the call is answered.
- 4) When the call is answered, users can have a private conversation at first. And then press the **CONF** button, the conference call will now include you and the other two parties.
- 5) Hang up to disconnect all parties.

Network Conference

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

If you enabled this function, users 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, select **Network** from the list.
- 2) Enter the **Conference URI**.

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

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 type="radio"/> Off	?
ptime(ms)	20	?
BLF List URI		?
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	Local Network	

To establish a conference:

- 1) Press the **Conf** hot key 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.
- 4) After starting a three way conference, press **Conf** button to enter conference dialing interface and invite another party to participate in teleconference.
- 5) After starting conference, press **Hold** key to hold local call without influencing others in conference.

Voicemail

Your voice mailbox messages, which are usually stored on a media server of your local

or hosted VoIP telephony system, can be accessed from your 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 you want to configure, enter the code which the phone uses to connect to your system. Press **123** to select the proper input method.
- 3) Press **OK** button to save the change, or press **Menu** to return to the previous

menu.

Note:

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

Want to see amount of Voice mail via phone interface, must enable the Subscribe for MWI via the web interface at first.

- 1) Click on **Account->Advanced-> Subscribe for MWI**.
- 2) Select **Enable** in the pull-down menu.

The screenshot shows a configuration page for SIP settings. The 'Advanced' section is expanded, and the 'Subscribe for MWI' option is highlighted in blue. The value is currently set to 'Disabled'. A tooltip box points to this option with the text: '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'. Other settings include 'Subscribe Register' (Disabled), 'Subscribe Period(seconds)' (1800), and 'MWI Subscription Period(seconds)' (3600). The right sidebar contains information about 'Register Name', 'User Name', 'NAT Traversal', 'Proxy Require', 'Codecs', and 'Advanced' parameters.

To view the voicemail via the Phone interface:

- 1) Press **Menu->Messages->Voice Mail->View Voice Mail**.
- 2) Users 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 **MESSAGE** button directly, if you already configured the Voicemail number.
- 2) You may be prompted to enter the password which is needed to connect to your VoIP telephony system. It depends on your system.
- 3) Your voice mailbox is called and you are able to listen to your new and old voicemails.

Note:

Before retrieving the new voicemail, please make sure that the connecting code has

been set on the phone.

Advanced Phone Functions

Account Setting

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 configure the period of the subscription. The default value is 3600.
<i>DTMF Type</i>	Select the DTMF type.

Users 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.

Users can consult your system administrator for more information.

TLS

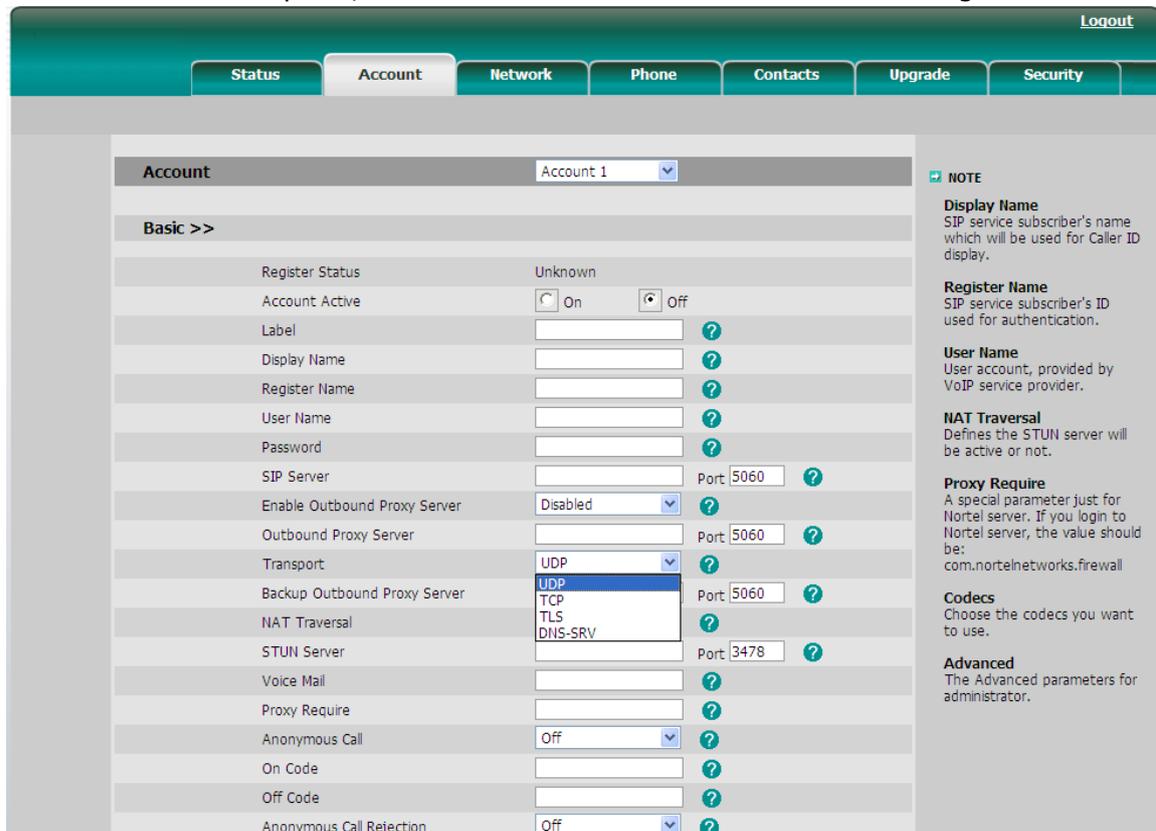
TLS(Transport Layer Security), an IETF standards track protocol(RFC 5246), was based on the earlier SSL specifications developed by Netscape Corporation.

Enterprise IP Phone

Advanced Phone Functions

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

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



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

Codexes
Choose the codecs you want to use.

Advanced
The 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 configure DNS-SRV via web interface:

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

The screenshot shows the configuration interface for an Enterprise IP Phone. The 'Account' tab is selected, and the 'Basic' sub-tab is active. The 'Transport' dropdown menu is open, showing 'DNS-SRV' as the selected option. A tooltip explains that there are three options: UDP, TCP, and TLS, and that the registered packet protocol is UDP, TCP, or TLS, TLS (Transport Layer Security) is encrypted. The interface includes various fields for SIP configuration, a 'NOTE' section on the right, and a 'Logout' button at the top right.

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>Enable DHCP Server</i>	If you configure 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 can not work.
<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**.
- 2) Enter the password required, select **Network option**, press **OK** button and select **PC Port** option, then press **OK** button to enter.

1. Bridge
* 2. Router

- 3) If you select **Bridge**, it will save and return to the previous menu.
- 4) If you select **Router**, you will be prompted to enter the **IP, Subnet Mask, DHCP Server Disable/Enable** etc.
- 5) Press **OK** button to save the changes, or **Menu** to return to the previous menu.

To configure PC Port via Web interface:

Click on **Network->PC Port** to do the relating configuration, users can configure the starting and end **IP address** only via Web interface. Users can 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
<i>Voice QoS</i>	When the network capacity is insufficient, QoS could provide priority to users by setting the value.
<i>Local RTP Port</i>	Define the port for voice transmission.
<i>WebServer</i>	Users can select the WebServer type: Disable, HTTP, HTTPS, or HTTPS & HTTP.

To configure VLAN settings via Phone interface:

- 1) Press **Menu->Settings->Advanced**.
- 2) Enter the password required, scroll to **Network** option, press **OK** button and select **VLAN**, then press **OK** to enter.
- 3) Select **WAN Port** and press **OK** button to enter.
- 4) Use the navigation keys to select and configure the **VLAN** status, input the **VID Number, Priority**.
- 5) Press **OK** button to save the settings, or **Menu** to return to **VLAN** menu.
- 6) Follow the same way to configure the **PC Port** option.

To configure VLAN settings via Web interface:

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

The screenshot shows the 'Advanced' configuration page for the Network section. The 'LLDP' section is currently set to 'Disabled' for the 'Active' field, with a 'Packet Interval' of 120. The 'VLAN' section is configured for both 'WAN Port' and 'PC Port' to be 'Active', with 'VID' set to 0 and 'USRRIORITY' set to 0. The 'Port Link' section is set to 'auto negotiat' for both 'WAN Port Link' and 'PC Port Link'. The 'Voice QoS' section has 'Voice QoS' and 'SIP QoS' both set to 40. The 'Local RTP Port' section has 'MaxRTPPort' set to 11800 and 'MinRTPPort' set to 11780. The 'WebServer' section is also visible at the bottom. A 'NOTE' section on the right provides information about VLAN and QoS.

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 settings via Web interface:

- 1) Click on **Network->Advanced->LLDP->Active**, in the pull-down Menu, select **Enable**.
- 2) Then enter the corresponding packet interval in **Packet Interval** field.
- 3) Users can also disable this function when you select **Disable** in **active** field.

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

The screenshot displays the configuration page for an Enterprise IP Phone, specifically the 'Network' tab. The page is divided into several sections for configuration:

- LLDP**: Active (Disabled), Packet Interval (120, Scope: 1~3600s).
- VLAN**: WAN Port Active (Disabled), VID (0, 0-4094), USRPRIORITY (0); PC Port Active (Disabled), VID (0, 0-4094), USRPRIORITY (0).
- Port Link**: WAN Port Link (auto negotiat), PC Port Link (auto negotiat).
- Voice QoS**: Voice QoS (40, 0~63), SIP QoS (40, 0~63).
- Local RTP Port**: MaxRTPPort (11800, 0~65535), MinRTPPort (11790, 0~65535).
- WebServer**: (configurable).

On the right side, there are two informational sections:

- NOTE**: A VLAN is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.
- 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.

The top navigation bar includes tabs for Status, Account, Network (selected), Phone, Contacts, Upgrade, and Security. A Logout link is visible in the top right corner.

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. Users can configure this transmission mode via web page.

To configure HTTPS settings via Web interface:

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

The screenshot displays the configuration interface for an IP phone. The settings are organized into several sections:

- VLAN**: WAN Port (Active, Disabled), VID (0), USRPRIORITY (0); PC Port (Active, Disabled), VID (0), USRPRIORITY (0).
- Port Link**: WAN Port Link (auto negotiate), PC Port Link (auto negotiate).
- Voice QoS**: Voice QoS (40), SIP QoS (40).
- Local RTP Port**: MaxRTPPort (11800), MinRTPPort (11780).
- WebServer**: HTTP port (80), HTTPS port (443), Type (HTTP&HTTPS).
- 802.1x**: 802.1X Mode (HTTP&HTTPS), Identity, MD5 Password.

On the right side, there are informational text blocks for **QoS** and **Local RTP Port**.

Note:

1. For more details of the HTTPS, users 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, 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 advance 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->Set PWD**.
- 2) You are prompted to enter the **Current PWD, New PWD** and **Confirm PWD**. Press **abc** to change the input method.
- 3) Press **OK** button to confirm the change, or **Menu** to return to 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 system administrator's advice.

To configure reset to factory via phone interface:

- 1) Press **Menu->Settings->Advanced**.
- 2) You are prompted to enter the required password, the default one is **admin**.
- 3) Select **Reset Factory**, then press **OK** button to enter.

- 4) You are prompted to confirm the change, press **OK** to reset to factory settings, or **Menu** 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.

To configure reset to factory via Web interface:

- 1) Select **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 all current setting changes including 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 a new manually unless mass provisioning is used!

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

- 1) Click on **Upgrade->Advanced**, select **Import/Export Config**, click **Export** button to export the file to your local computer.
- 2) Click on **Upgrade->Advanced**, select **Import/Export Config**, click **Browse** button, select the specific configuration file in your local computer, click **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.

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. Users 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->Browse**, select the firmware file in your local computer.
- 2) Click **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 **Autoprovision** button.

To configure Auto Provision via phone interface:

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

The parameters of the Autoprovision :

Parameter	Description
<i>Update Protocol</i>	The phone can be updated via TFTP, FTP or HTTP.
<i>TFTP Server</i>	If you select TFTP as protocol TFTP, you need to enter the TFTP server IP address and port.
<i>Check new config</i>	Users 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>	Users 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 y000000000007.cfg and mac.cfg files encryption and decryption for user authentication to realize security usage. If there are any encrypted y000000000007.cfg or mac.cfg files on the server, users can open the webpage of your IP phone.

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Click on **Upgrade->Advanced**, select and fill in the **Common AES Key** (for y000000000007.cfg and **MAC-Oriented AES Key** (for mac.cfg), then click **Confirm** button to decryption the files and upgrade to the new version. Shown as below:

Custom Option(128 ~ 254) ?

Custom Option Type ?

URL ?

Account ?

Password ?

Common AES Key ?

MAC-Oriented AES Key ?

ForbidZero ?

WaitTime ?

PNP Config ?

Check New Config ?

Click this button to auto provision immediately ?

Export / Import Config 浏览... ?

Export System Log ?

PCAP Trace ?

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.

Note:

Users can ask your system administrator for the decrypt password.

To configure AES Key via phone interface:

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

1. Common: 2 a B

- 2) Enter the **Common AES** and the **MAC-Oriented**.
- 3) Press **OK** button 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.

To turn on 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 turn on this function.

- 2) Enter the time in the **Wait Time** field.
- 3) Click **Confirm** to save.

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

- 1) Not any operation or press **Cancel** hot key, will enter idle interface.
- 2) Press **OK** key, enter a network setting interface, press next key enter an AutoP setting interface, enter the corresponding contents; press **OK** key to save the settings. Press **Menu** key return to previous menu.

System Log Export

If there are any errors happened in your phone, users 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

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, users can export the PCAP trace and send to your system administrator for diagnosis.

To export the PCAP Trace:

Click on **Upgrade->Advanced** to enter, select **PCAP Trace** option, click **Start** button began 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.

DSS keys Configuration

The phone has 2 line keys which are programmable. The following list shows the functions users can configure on the DSS keys and provides a description for each function. The default configuration for each key is Line.

- Line
- Speed Dial
- BLF
- BLF List
- Voice Mail
- Pick Up
- Group Pickup
- Call Park
- Intercom
- DTMF
- Prefix
- Local Group
- Conference
- Forward
- Transfer
- Hold
- DND
- Redial
- Call Return
- Record
- URL Record
- Multicast page
- Group Listening
- Public Hold
- Private Hold
- Shared Line
- ACD
- Zero-sp-touch

Note:

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.

Line

Users can configure these keys as line keys to active up to the six user accounts.

To configure the key as Line:

Click on **Phone->DSS Key->Line Key**, select one of the link key you want to make the configuration, there is a pull-down menu in the **Type** field, select Line from the list, press **Confirm** button to save the changes.

Speed Dial

Users can configure the key as a simplified speed dial key. This key function allows you to easily access the most frequently dialed numbers.

To configure the key as Speed Dial:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Speed Dial** from the list.
- 2) Enter the number you want to perform **Speed Dial** 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.

BLF

Users 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 configure the key as BLF:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, 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) Click **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, users can also configure the pickup number to active the pickup function. For example, if you set the BLF number as 212, and the pickup number is *83, then when there is an incoming call to 212, press the BLF key, it will call out the *83 automatically to pick up the incoming call on 212.

BLF List

BLF list is a function which can monitor the group status, it is not one to one monitoring, but the information feedback from the server to decide which BLF list will

monitor which account.

To configure BLF List via web interface:

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

To configure the key as BLF List:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, and select **BLF List** from the list.
- 2) In the **Line** field, select a line for which to apply this key.
- 3) Press **Confirm** button to save the changes.

Voice Mail

When the key is configured as Voicemail key you are allowed to access voicemail quickly by pressing this key.

To configure the key as Voice Mail:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Voice Mail** from the list.
- 2) Enter the number you want to configure 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

When you configure a Pick Up key, you specify the extension that you want to monitor. Then, when the monitored extension receives a call, users can press this key to pick up the incoming calls.

To configure the key as Pick Up:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **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

When you configure a Group Pick Up key, you specify the extension group that you want to monitor. Then, when the monitored group receives a call, users 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 configures to you.

To configure the key as Group Pick Up:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, and select **Group Pick Up** from the list.
- 2) Enter the feature code (for example,*78) 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.

Call Park

Call Park is a feature that allows a person to put a call on hold at one telephone configuration and continue the conversation from any other telephone configuration. The "call park" feature is activated by pressing a preprogrammed button or a special sequence of buttons. When the conversation which is monitored was transferred to an unused extension number, users can press this key to retrieve the call.

To configure the key as Call Park:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Call Park** from the list.
- 2) Enter the number you want to park 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.

Intercom

Users can configure the key for Intercom mode and is useful in an office environment as a quick access to connect to the operator or the secretary.

To configure the key as Intercom:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Intercom** from the list.
- 2) Enter the extension number you want to intercom in the **Value** field.
- 3) In the **Line** field, select a line for which to apply this key, the default one is Line 1.

- 4) 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

You are allowed to send out the desired DTMF number during the conversation. The number needs to be configured in advance.

To configure the key as DTMF:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **DTMF** from the list.
- 2) In the **Value** field, enter the specific number.
- 3) Click **Confirm** button to save the changes.

Prefix

When you enable the function of prefix, press this key, the phone will be ready to make a new call, and show up the content which you configured previously on the dial interface. And you could enter other figure and call out

To configure the key as Prefix:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, and select **Prefix** from the list, and enter the number you want to show up on the dial interface in the **Value** field.
- 2) Click **Confirm** button to save the changes

Local Group

The keys can be configured as Local Group key. Then pressing this key under the idle status, users can enter the Local Group interface.

To configure the key as Local Group:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Local Group** from the list.
- 2) In the **Line** field, select a group.
- 3) Click **Confirm** button to save the changes.

Conference

You are allowed to configure the DSS key to be used as a conference key while remaining in the current call. This key allows a user on a call to conference another party while remaining in the conference.

To configure the key as Conference:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Conference** from the list.
- 2) Click **Confirm** button to save the changes.

Forward

If the key is configured as Forward key, press this key under the idle status, the IP phone will turn to the forward page, and users can configure the Forward to number, then when there is any call to the extension number will be forwarded to the configured number automatically.

To configure the key as Forward:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Forward** from the list.
- 2) Enter the extension number you want to forward to in the **Extension** field.
- 3) Click **Confirm** button to save the changes.

Transfer

You are able to configure the key as a transfer key to perform the Blind/Attended/Semi-Attended Transfer.

To configure the key as Transfer:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Transfer** from the list.
- 2) Enter the Number in the **Value** field, when you are on a conversation, press this key, the phone will Blind transfer to the number. Or users can leave it blank to configure as the **Transfer** button.
- 3) Click **Confirm** button to save the changes.

Hold

The key can be configured as a hold key. Users can use this key to hold and retrieve a call during the conversation.

To configure the key as Hold:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Hold** from the list.
- 2) Click **Confirm** button to save the changes.

DND

If the key is configured as DND key, you are allowed to active the DND function immediately when you press it. Press it again to deactivate DND mode.

To configure the key as DND:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **DND** from the list.
- 2) Click **Confirm** button to save the changes.

Redial

If the key is configured as Redial key, press this key under the idle status, it will enter the Dialed Calls interface, then users can select a special line to call out by pressing the line keys.

To configure the key as Redial:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Redial** from the list.
- 2) Click **Confirm** button to save the changes.

Call Return

When the key is configured as Call Return key you are allowed to dial out the last phone call you received.

To configure the key as Call Return:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Call Return** from the list.
- 2) Click **Confirm** button to save the changes.

Record

Call recording is a phone function to record the conversation in the process of dialogue. Using this feature, please pay attention to the maximum recording time and frequency in advance. Generally, it maybe a few minutes.

To configure the key as Record:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Record** from the list.
- 2) Click **Confirm** button to save the changes.

URL Record

During the conversation, pressing the type of DSS key, and then follow the voice prompts to achieve the call recording capability.

- 1) When you are on the conversation, pressing the **DSS key** to start the recording process in the current call.
- 2) Enable the **Recording** function, the recording icon will be flashing for the ongoing recording process.

- 3) Pressing the **DSS key** again to disable the recording function, and the flashing recording icon will be disappeared simultaneously.
- 4) Follow the voice prompts to listen to the recording.

To configure a DSS key as URL Record:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **URL Recorder** from the list.
- 2) Enter the condition code in the **Value** field.
- 3) 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	<input type="text" value="239.0.1.15:10000"/>	<input type="text" value="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:

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

When the key is configured as Group Listening key, you are allowed 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 using handset/headset; meanwhile the others nearby can listen using speakerphone. Users 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)

To configure the key as Group Listening:

- 1) Click on **Phone**->**DSS Key**->**Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, and select **Group Listening** from the list.
- 2) Click **Confirm** button to save the changes.

Public Hold

The key can be configured as a public hold key. During a conversation, all members belonging to that particular BLA group can use this key to hold a call.

To configure the key as Public Hold:

- 1) Click on **Phone**->**DSS Key**->**Line Key**, select one of the keys you want to make

Enterprise IP Phone

Advanced Phone Functions

the configuration, there is a pull-down menu in the **Type** field, select **Public Hold** from the list.

- 2) Click **Confirm** button to save the changes.

Private Hold

The key can be configured as a private hold key. During a conversation, all members belonging to that particular BLA group can use this key to hold the call, but only the initiator can retrieve the call.

To configure the key as Private Hold:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Private Hold** from the list.
- 2) Click **Confirm** button to save the changes.

Shared Line

The 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 usage and are blocked from using this line appearance until the line goes back to idle state or when 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.

To configure the key as Shared Line:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, select **Shared Line** from the list.
- 2) Enter the condition code in the **Value** field.
- 3) In the **Line** field, select a line for which to apply this key, the default one is Line 1.
- 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.

The screenshot shows a configuration interface with two dropdown menus. The first dropdown is labeled 'DSS Key 1' and has 'ACD' selected. The second dropdown is labeled 'Line 1' and has a downward arrow indicating it is open.

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.

The screenshot shows a login screen titled 'ACD'. It has two input fields: '1. User ID:' and '2. Password:'. Below the fields are four buttons: 'Cancel', '123', 'Del', and 'Log In'. There is also a small downward arrow icon on the right side of the input area.

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

Users can also press the DSSKey which has been configured as the Zero-sp-touch. Then press the DSSKey, the phone will turn to the Zero-sp-touch interface.

To configure the key as Zero-sp-touch:

- 1) Click on **Phone->DSS Key->Line Key**, select one of the keys you want to make the configuration, there is a pull-down menu in the **Type** field, and select **Zero-sp-touch** from the list.
- 2) Click **Confirm** button to save the changes

URL

If the key is configured as URL, then pressing this key, you can send HTTP requests to a web server.

To assign the key as URL:

- 1) Click on **Phone->DSS Key->Memory Key**, choose one of the keys you want to

make the assignment, there is a pull-down menu in the **Type** field, and choose **URL** from the list.

- 2) Enter the number you want to set as URL in the **Value** field.
- 3) Press **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: Enabled

Port: 161 (0~65535)

Trusted Address: 192.168.1.10

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
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 is 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=DNDOOn

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

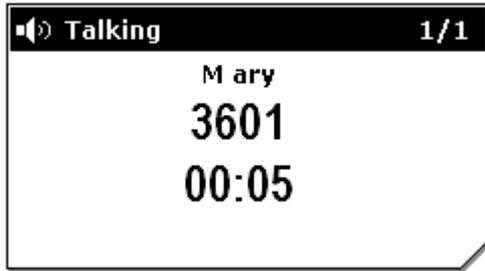
When an outgoing call from the phone is being diverted to another party (i.e., via call forward), the phone will display the Caller ID of the new party, Similarly, the new party will display the Caller ID of the original caller and the redirected call information indicated by DIVERSION or HISTORY-INFO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

Call Diversion example:

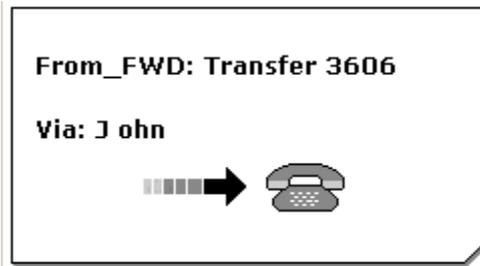
- 1) Alice calls John.
- 2) John's phone is busy.
- 3) John's phone forwards the incoming call to another party (Mary).
- 4) Alice's phone displays the Caller ID of the new party(Mary)
- 5) Mary's phone receives a call and displays the caller ID of the initiator (Alice) and the diverted party (John).

See example of the phones' LCD below:

Alice's phone



Mary's phone



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:

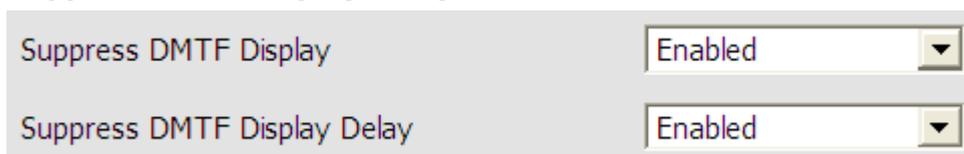
1. Diversion is not available on all servers. Please contact your system administrator for more information.
2. Redirected call information is sent by SIP message in the Diversion header, per draft-levy-sip-diversion-08, or the HISTORY-INFO header, per RFC 4244. It depends on the server.
3. Multiple 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**



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

Tone Settings

Users 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 configure 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) Click **Confirm** button to save the changes, or **Cancel** to cancel the changes.

Note:

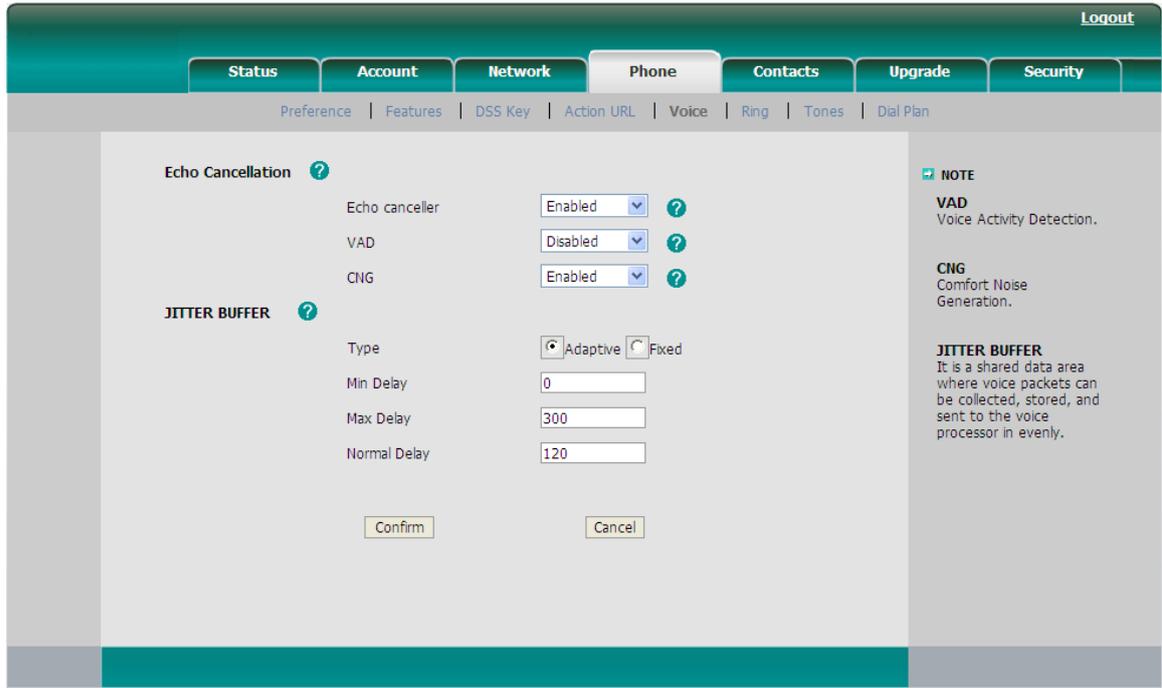
1. Please contact your system administrator for more information about the frequency and time period parameters. Users can enter up to 8 groups for each tone.
2. If the frequency is configured as 0, it means silence.

Voice

To configure the Voice filed via Web interface:

- 1) Click on **Phone->Voice**.
- 2) Configure 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.
<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.
Parameter	Description
<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.



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

Ring

Users can group your contacts, and then configure the ringing tone for each group. To configure the Ring option via Web interface:

- 1) Click on **Phone->Ring**.

The screenshot shows a web interface for configuring ring tones. The interface has a top navigation bar with tabs: Status, Account, Network, Phone, Contacts, Upgrade, and Security. The 'Phone' tab is selected. Below the navigation bar, there are sub-tabs: Preference, Features, DSS Key, Action URL, Voice, Ring, Tones, and Dial Plan. The 'Ring' sub-tab is selected. The main content area displays a table with 10 rows. Each row has two columns: 'Internal Ringer Text' and 'Internal Ringer File'. The 'Internal Ringer File' column contains a dropdown menu with 'Ring1.wav' selected. At the bottom of the table, there are 'Confirm' and 'Cancel' buttons. On the right side of the interface, there is a 'NOTE' icon.

- 2) **Internal Ringer Text:** To configure group name. For example, family.
- 3) **Internal Ringer File:** To select a special ring tone for the group.
- 4) Click **Confirm** button to save the changes.



1-2-3. Spojeno.

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Trouble Shooting

I can not register to the server ?

- 1) Check the IP address. If you configure 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 configure your WAN port to static IP client mode.

During a call, I can not 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 key to answer the incoming call.

How to refuse incoming calls during a call?

Users 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 update the firmware?

- 1) Enter the webpage of your phone, go to Upgrade, then users 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 system administrator, or the device will probably crash after the update.

How to auto provision?

Consult the auto provision server address with your ITSP.

The manual is only for reference; please take the object as the standard.

WELL reserve the right to improve or change the product and the user guide without notice.

Users can download the latest user manuals from our website:

<http://www.well.cz/>

V61.0