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Declaration of Conformity



Hereby, Yealink Network Technology CO., LTD. declares that this phone is in conformity with the essential requirements and other relevant provisions of the CE, FCC.

CE Mark Warning

This device is marked with the CE mark in compliance with EC Directives 2006/95/EC and 2004/108/EC.

Part 15 FCC Rules

This device is compliant with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference.

2. This device must accept any interference received, including interference that may cause undesired operation.

Class B Digital Device or Peripheral

Note: This device is tested and complies with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

1. Reorient or relocate the receiving antenna.
2. Increase the separation between the equipment and receiver.
3. Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
4. Consult the dealer or an experienced radio/TV technician for help.

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.

GNU GPL INFORMATION

Yealink SIP-T41P firmware contains third-party software under the GNU General Public License (GPL). Yealink uses software under the specific terms of the GPL. Please refer to the GPL for the exact terms and conditions of the license.

The original GPL license, source code of components licensed under GPL and used in Yealink products can be downloaded from Yealink web site:

<http://www.yealink.com/GPLOpenSource.aspx?BaseInfoCatId=293&NewsCatId=293&CatId=293>.

About This Guide

Thank you for choosing the SIP-T41P IP Phone, an ultra-elegant gigabit IP Phone which is especially designed for power users in the office environment. This unit provides business telephony features such as Call Hold, Call Transfer, Busy Lamp Field and Conference.

This guide provides 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 Packaging Contents and Regulatory Notices sections in this guide before you set up and use the SIP-T41P IP phone.

If this is your first time using the SIP-T41P IP phone, we recommend that you first refer to the *Quick Installation Guide* and *Quick Reference Guide*, available online:

<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Note

The Shared Line, Busy Lamp Field List, Network Directory and Network Call Log features are hidden for IP phones in neutral version, which are designed for BroadWorks environment. Please contact your system administrator for more information.

In This Guide

This guide is based on firmware version 71. The topics provided in this guide include:

- Chapter 1 [Overview](#)
- Chapter 2 [Getting Started](#)
- Chapter 3 [Customizing Your Phone](#)
- Chapter 4 [Basic Call Features](#)
- Chapter 5 [Advanced Phone Features](#)

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Overview

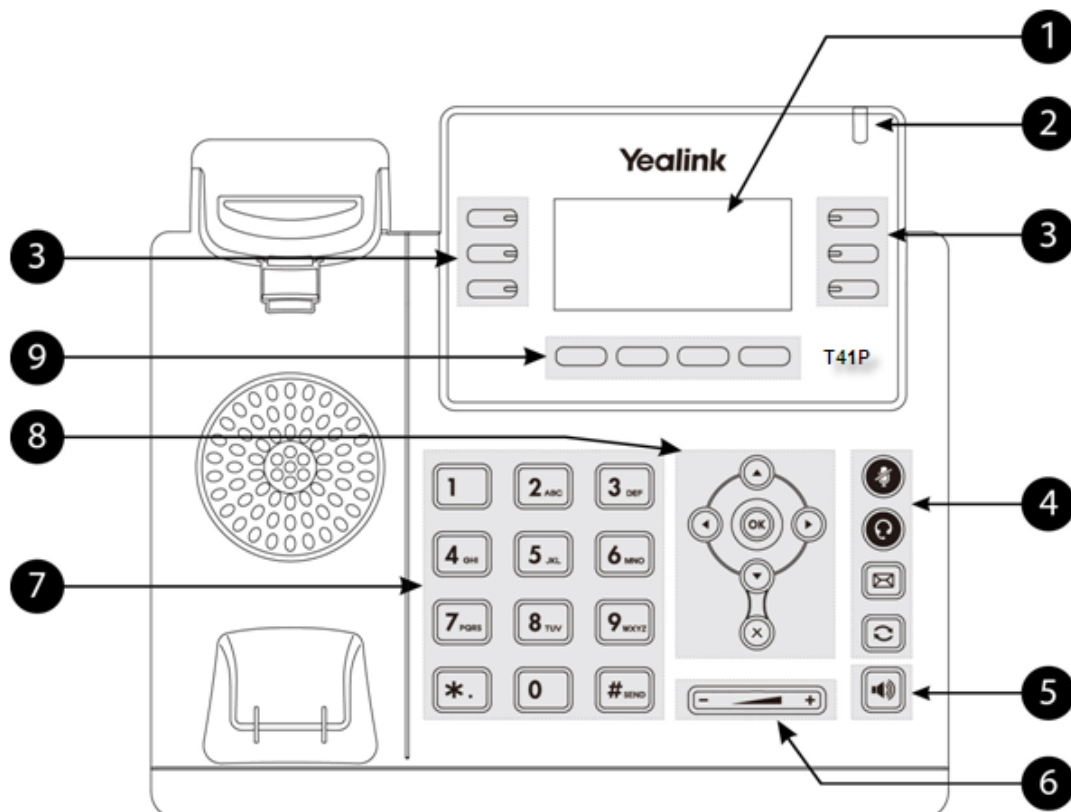
This chapter provides the overview of the SIP-T41P IP phone. The topics include:

- [Hardware Component Instructions](#)
- [Icon Instructions](#)
- [LED Instructions](#)
- [User Interfaces](#)
- [Documentations](#)




If you require additional information or assistance with your new phone, contact your system administrator.

Hardware Component Instructions

The main hardware components of the SIP-T41P IP phone are the LCD screen and the keypad.
























Hardware component instructions of the SIP-T41P IP phone are:

	Item	Description
①	LCD Screen	Shows information about calls, messages, soft keys, time, date and other relevant data: <ul style="list-style-type: none"> • Call information—caller ID, call duration • Icons (for example, DND) • Missed call text or second incoming caller information • Prompt text (for example, "Saving file!") • Date and time
②	Power Indicator LED	Indicates phone power and some feature status.
③	Line Keys	Use these keys to activate up to three accounts and assign various features.
④	Mute Key	Toggles and indicates the mute feature.
	Headset Key	Toggles and indicates the headset mode.
	Message Key	Accesses voice mails.
	Redial Key	Redials a previously dialed number.
⑤	Speakerphone Key	Toggles the hands-free speakerphone mode.
⑥	Volume Key	Adjusts the volume of the handset, headset, speaker, and ringer.
⑦	Keypad	Provides the digits, letters, and special characters available in context-sensitive applications.
⑧		Scroll through the displayed information.
		Confirms actions or answers incoming calls.
		Cancels actions or rejects incoming calls.
⑨	Soft Keys	Label automatically to identify their context-sensitive features.

Icon Instructions

Icons appear on the phone LCD screen are described in the following table:

Icon	Description
	Network is unavailable
	Registered successfully
	Registration failed
	Registering
	Hands-free speakerphone mode
	Handset mode
	Headset mode
	Voice Mail
AA	Auto Answer
DND	Do Not Disturb
	Call Hold
	Call Mute
	Ringer volume is 0
	Keypad Lock
	Call Forward/Forwarded Calls
	Missed Calls
	Received Calls
	Placed Calls
	Recording box is full

Icon	Description
	A call cannot be recorded
	Recording starts successfully
	Recording cannot be started
	Recording cannot be stopped

LED Instructions

Power Indicator LED

LED Status	Description
Solid red	The phone is initializing.
Fast flashing red	The phone is ringing. The network is disconnected.
Slow flashing red	The phone receives a voice mail.
Off	The phone is powered off.

Line key LED

LED Status	Description
Solid green	The line is seized. The line is in conversation.
Fast flashing green	The line receives an incoming call.
Slow flashing green	The call is placed on hold.
Off	The line is inactive.

Line key LED (configured as BLF key)

LED Status	Description
Solid red	The monitored user is busy.
Fast flashing red	The monitored user receives an incoming call.
Off	The monitored user is idle. The monitored user does not exist.

Line key LED (used as page switch key)

LED Status	Description
Solid red	There is a parked call on the line key of the non-current page.
Solid green	There is a call on the line key of the non-current page.
Fast flashing green	The line receives an incoming call on the non-current page.
Fast flashing red	The monitored user receives an incoming call on the non-current page.
Off	Line keys are idle.

User Interfaces

There are two ways to customize specific configurations on your SIP-T41P IP phone:

- Using the user interface on the IP phone.
- Using the user interface in an internet browser window from your PC.

The hardware components keypad and LCD screen constitute the phone user interface, which allows the user to execute all call operation tasks and basic configuration changes directly on the phone. In addition, you can use the web user interface to access all configuration settings. In many instances, it is possible to use both the phone user interface and the web user interface to operate the phone and change settings. However, in some instances, it is only possible to use the phone user interface or the web user interface.

Phone User Interface

You can customize your phone by pressing the Menu soft key to access the phone user interface. The Advanced Settings option is only accessible to an administrator, and the default administrator password is admin (case-sensitive). For more information on customizing your phone using the available options from the phone user interface, refer to [Customizing Your Phone](#) on page 17.

Web User Interface

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

The options you can use to customize the IP phone via phone user interface and/or via web user interface are listed in the following table:

Options	Phone User Interface	Web User Interface
Status		
--IPv4		
--MAC		
--Firmware	√	√
--Network		
--Phone		
--Accounts		
Basic Call Features		√

Options	Phone User Interface	Web User Interface
--Auto Answer	√	
--Auto Redial	√	
--Call Forward	√	
--Call Waiting	√	
--Call Park	√	
--Call Pickup	√	
--Call Return	√	
--Conference		
--Anonymous Call	√	
--Anonymous Call Rejection	√	
--DND	√	
Basic Phone Settings		
--Language	√	
--Time & Date	√	√
--Administrator Password	√	√
--Ring Tones	√	√
--Backlight	√	√
--Call Waiting	√	√
--Phone Volume	√	
--Call History Management	√	
--Contact Management		√
--Directory		√
--Local Directory	√	√
--Blacklist	√	√
--Remote Phonebook		√
--Dial Plan		√
--DSS Keys	√	√
--Key as Send	√	√
--Hot Line	√	√
--Live Dialpad		√
Advanced Phone Features		√
--BLF	√	
--Call Recording	√	
--Intercom	√	
--Multicast Paging		
--Music on Hold		
--ACD	√	
--Messages	√	
SIP Account		
--User Options		
--Register Status	√	

Options	Phone User Interface	Web User Interface
--Account Active	√	√
--Label	√	
--Display Name	√	
--Register Name	√	
--User Name	√	
--Password	√	
--SIP Server	√	
--Server Option		
--Registrar Port		
--Outbound Status	√	
--Outbound Proxy	√	
--NAT Traversal		
--STUN Status	√	
--STUN Server	√	

Note The table above lists most of the feature options. Please refer to the relevant sections for more information.

Documentations

The following table shows documentations available for the SIP-T41P IP phone.

Name	Contents	Where found	Format/ Language
Quick Installation Guide	Basic setup of the phone	In the package	PDF/English
Quick Reference Guide	Basic call features and phone customizations	In the package	PDF/English
User Guide	Phone/Web user interface settings Basic call features and advanced phone features	CD attached in the package	PDF/English

Note You can also download the latest documentations from website:
<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Getting Started

This chapter provides basic installation instructions and information for obtaining the best performance with the SIP-T41P IP phone. The topics include:

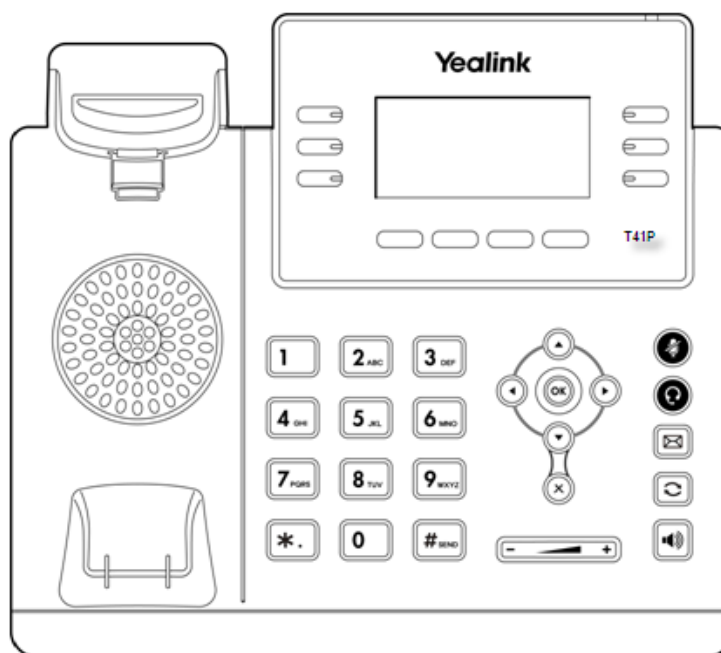
- [Packaging Contents](#)
- [Phone Installation](#)
- [Phone Initialization](#)
- [Phone Status](#)
- [Basic Network Settings](#)
- [Registration](#)
- [Idle Screen](#)

If you require additional information or assistance with your new phone, contact your system administrator.

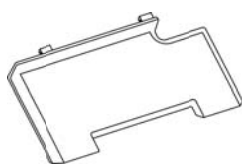
Packaging Contents

The following components are included in your SIP-T41P IP phone package:

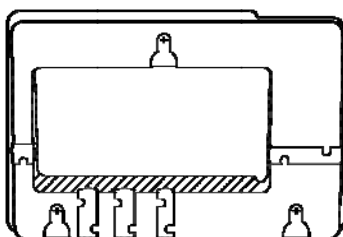
- **SIP-T41P IP Phone**



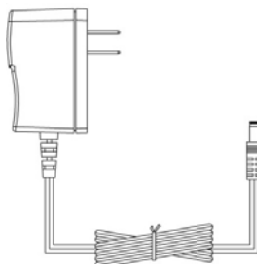
- Phone Stand



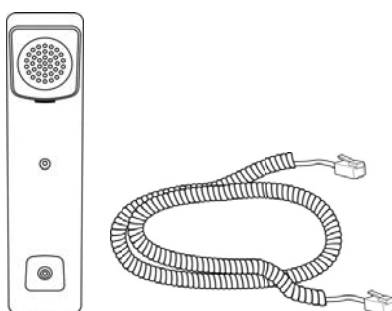
- Wall Mount Bracket (Optional)



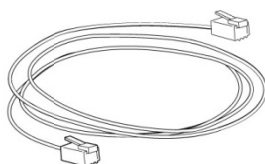
- Power Adapter (Optional)



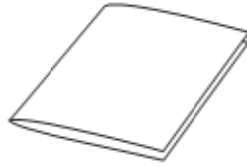
- Handset & Handset Cord



- Ethernet Cable



- Quick Installation Guide & Quick Reference Guide



- CD-ROM



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

Phone Installation

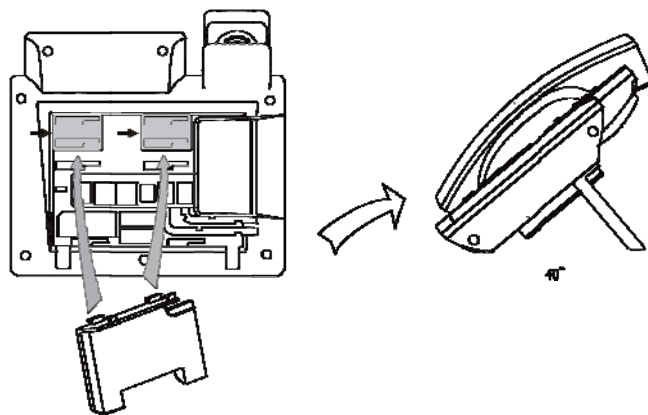
If your phone is already installed, proceed to [Phone Initialization](#) on page 14.

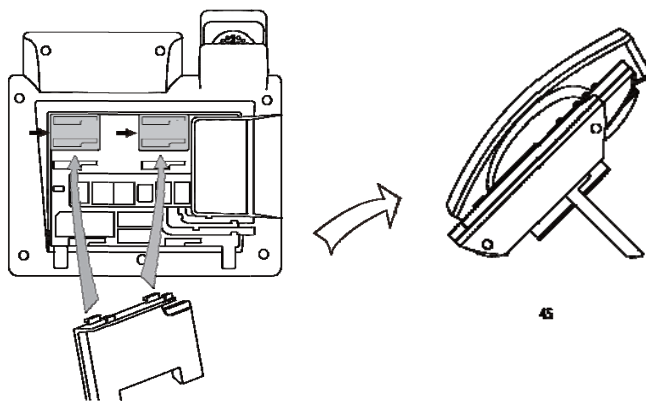
This section introduces how to install the phone with the components in the packaging contents:

- Attach the stand or the optional wall mount bracket
- Connect the handset and optional headset
- Connect the network and power

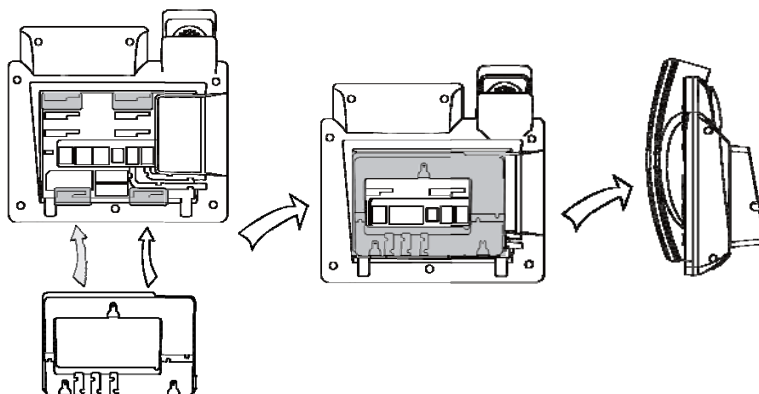
1) Attach the stand or the optional wall mount bracket

Desk Mount Method



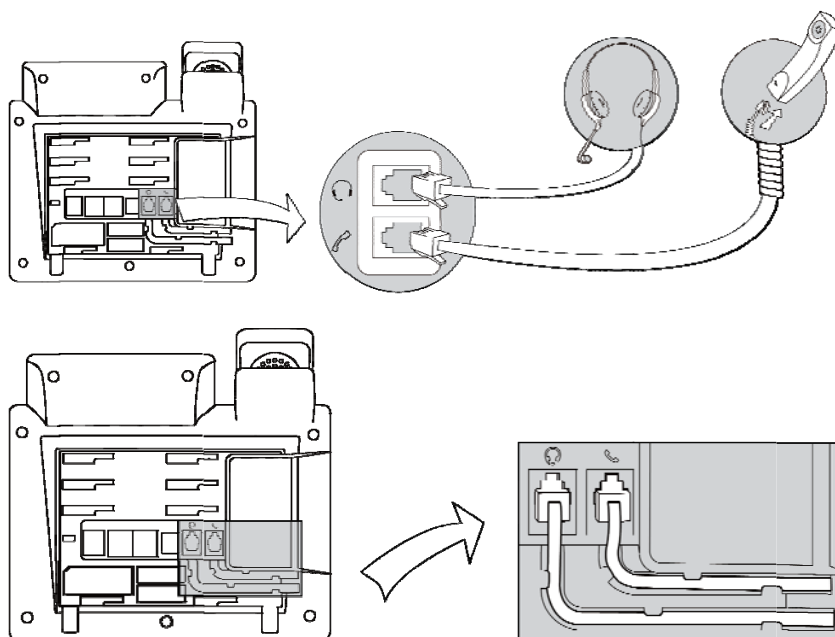


Wall Mount Method (Optional)



Note The headset is not provided in the packaging contents. For more information on how to attach the wall mount bracket, refer to *Yealink Wall Mount Quick Installation Guide for T4X IP Phones*.

2) Connect the handset and optional headset



Note

The headset is not provided in the packaging contents. Contact your system administrator for more information.

3) Connect the network and power

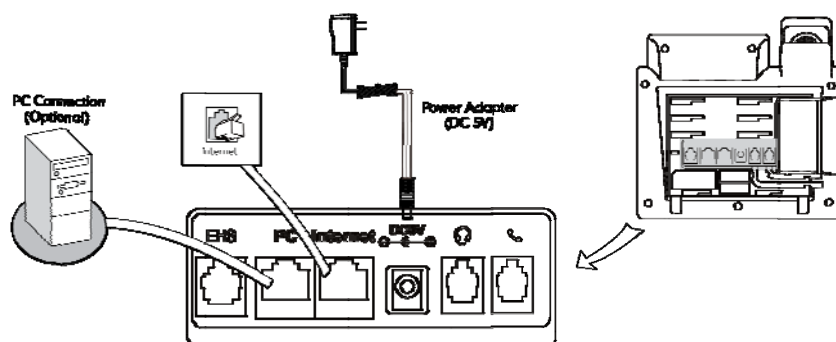
You have two options for power and network connections. Your system administrator will advise you which one to use.

- AC power (Optional)
- Power over Ethernet (PoE)

AC Power (Optional)

To connect the AC power:

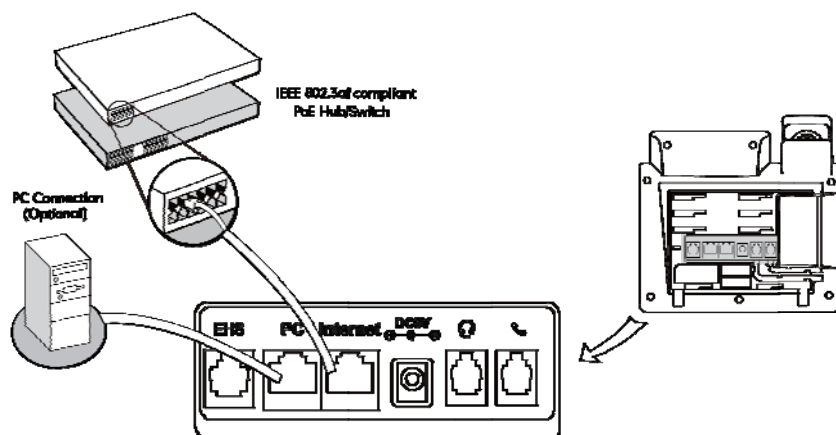
1. Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.
2. Connect the supplied Ethernet cable between the Internet port on the phone and the Internet port in your network or switch/hub device port.

**Power over Ethernet**

Using a regular Ethernet cable, the SIP-T41P IP phone can be powered from a PoE (IEEE 802.3af) compliant switch or hub.

To connect the PoE:

1. Connect the Ethernet cable between the Internet port on the phone and an available port on the in-line power switch/hub.



Note If in-line power is provided, you don't need to connect the AC adapter. Make sure the Ethernet cable and switch/hub is PoE compliant.

The phone can also share the network with other network devices such as a PC (personal computer). It is an optional connection.

Important! Do not unplug or remove power to the phone while it is updating firmware and configurations.

Phone Initialization

After your phone has been powered on, the system boots up and performs the following steps:

Automatic Phone Initialization

The phone finishes the initialization process by loading the saved configuration. The phone LCD screen will display "Initializing...please wait" during the initialization.

DHCP (Dynamic Host Configuration Protocol)

By default the phone attempts to contact a DHCP server in your network in order to obtain the valid IPv4 network settings, e.g., IP address, subnet mask, default gateway address and DNS address.

Note If your network does not use DHCP, proceed to [Basic Network Settings](#) on page 15.

Phone Status




You can view phone status via phone user interface or web user interface.

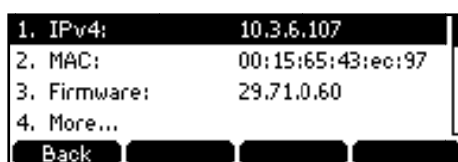
Available information of phone status includes:

- Network status including IPv4 status, IP address mode, MAC address, LAN type, LAN IP and LAN mask.

- Phone status including device model, hardware version, firmware version, product ID and MAC address).
- Account status including register status of SIP accounts.

To view the phone status via phone user interface:

1. Press , or press **Menu->Status**.
2. Press  or  to scroll through the list and view the specific information.



To view the phone status via web user interface:



1. Open the web browser of your computer.
2. Enter the IP address in the browser's address bar, and then press the **Enter** key.
3. Enter the user name (admin) and password (admin) in the login page.
4. Click **Confirm** to login.

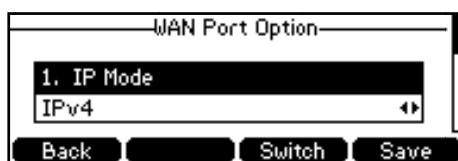
The first page shows the available information of phone status.

Basic Network Settings

If your phone cannot contact a DHCP server for any reason, you need to configure network settings manually. IP phones support the IPv4 address only, the IPv6 address only or both IPv4 and IPv6 addresses.

To configure the IP address mode via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) **->Network->WAN Port**.
2. Press  or  to select **IPv4**, **IPv6** or **IPv4 & IPv6** from the **IP Mode** field.



3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

To configure a static IPv4 address via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) **->Network->WAN Port->IPv4->Static IP Client**.
2. Enter the desired values in the **IPv4**, **Subnet Mask**, **Default Gateway**, **Pri DNS** and **Sec DNS** fields respectively.



3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

To configure a static IPv6 address via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) **->Network->WAN Port->IPv6->Static IPv6 Client**.
2. Enter the desired values in the **IPv6 IP**, **IPv6 IP Prefix**, **IPv6 Gateway**, **IPv6 Pri.DNS** and **IPv6 Sec.DNS** fields respectively.



3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Note

Using the wrong network parameters may result in inaccessibility of your phone and may also have an impact on your network performance. For more information about these parameters, contact your system administrator.

Registration

Generally, your phone will be deployed with multiple other phones. In this case, your system administrator will configure the phone parameters beforehand, so that after you start up your phone, the phone will be registered and ready to use. If your phone is not registered, you may have to register it. For more information on how to register your phone, refer to [Account Management](#) on page 50.

Idle Screen

If the phone has successfully started up, the idle LCD screen is shown as below.



The idle screen shows the label of current account, time and date, and four soft keys.

Customizing Your Phone

You can customize your SIP-T41P IP phone by configuring the backlight, time & date, ring tone and so on. You can add contacts to the phone's local directory manually or from call history. You can also handle incoming calls from different contacts in different ways.

This chapter provides basic operating instructions for customizing your phone. The topics include:

- [General Settings](#)
- [Audio Settings](#)
- [Contact Management](#)
- [Call History Management](#)
- [System Customizations](#)

If you require additional information or assistance with your new phone, contact your system administrator.

General Settings

Backlight

You can configure the backlight feature to adjust the brightness of the phone LCD screen. Backlight status on the LCD screen can be configured as the following types:

- **Always on:** Backlight is on permanently.
- **1min, 2min, 5min, 10min, 30min:** Backlight is turned off when the phone has been inactive for approximately 1min, 2min, 5min, 10min or 30min.

To configure the backlight via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Phone Settings**->**Backlight**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Backlight Time** field.



3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure the backlight via web user interface at the path **Settings->Preference**.



Language

The default language of the phone user interface is English. The phone will detect and use the same language as which of your internet browser for the web user interface. If the language of your internet browser is not supported by the phone, the web user interface will use English by default. You can change the language for the phone user interface and the web user interface respectively from each other.

Note

Not all of the supported languages are available for selection. The available languages depend on the language packs currently loaded to the IP phone. Please contact your system administrator for more information about the available languages of your new phone.

To change the language for the phone user interface:

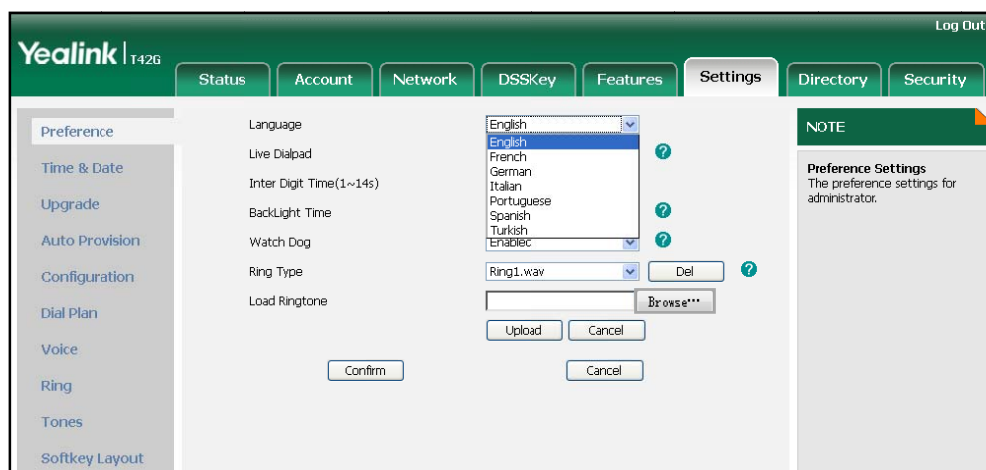
1. Press **Menu->Settings->Basic Settings->Language**.
2. Press  or  to select the desired language.



3. Press the **Save** soft key to accept the change.
Text on the LCD screen will change to the selected language.

To change the language for the web user interface:

1. Click on **Settings->Preference**.
2. Select the desired language from the pull-down list of **Language**.



3. Click **Confirm** to accept the change.
Text on the web user interface will change to the selected language.

Time & Date

The time and date display on the LCD screen when the phone is idle. You can configure the phone to obtain the time and date from the Simple Network Time Protocol (SNTP) server automatically, or configure the date and time manually. If the phone cannot obtain the time and date from the SNTP server, contact your system administrator for more information.

To configure the SNTP settings via phone user interface:

1. Press **Menu**->**Settings**->**Basic Settings**->**Time & Date**->**SNTP Settings**.
2. Press **←** or **→**, or the **Switch** soft key to select the time zone that applies to your area from the **Time Zone** field.
The default time zone is "+8 China(Beijing)".
3. Enter the domain names or IP addresses in the **NTP Server 1** and **NTP Server 2** fields respectively.
4. Press **←** or **→**, or the **Switch** soft key to select the desired value from the **Daylight Saving** field.



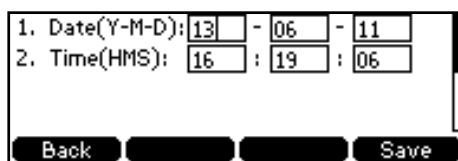
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Note

Please refer to [Appendix A - Time Zones](#) for the list of available time zones on the IP phone.

To configure the time and date manually via phone user interface:



1. Press **Menu**->**Settings**->**Basic Settings**->**Time & Date**->**Manual Settings**.
2. Enter the specific time and date.





3. Press the **Save** soft key to accept the change.

The time and date on the LCD screen will change accordingly.

To configure the time and date format via phone user interface:

1. Press **Menu Settings->Basic Settings->Time & Date->Time & Date Format**.
2. Press  or , or the **Switch** soft key to select the desired time format (12 Hour or 24 Hour) from the **Clock** field.



3. Press  or , or the **Switch** soft key to select the desired date format from the **Date Format** field.
4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

There are 7 available date formats. For example, for the date format "WWW DD MMM"; "WWW" represents the abbreviation of the week, "DD" represents two-digit day, and "MMM" represents the first three letters of the month.

The date formats you need to know:

Date Format	Example (2013-2-27)
WWW MMM DD	Wed Feb 27
DD-MMM-YY	27-Feb-13
YYYY-MM-DD	2013-02-27
DD/MM/YYYY	27/02/2013
MM/DD/YY	02/27/13
DD MMM YYYY	27 Feb 2013
WWW DD MMM	Wed 27 Feb

You can also configure the date and time settings via web user interface at the path **Settings->Time & Date**.

Administrator Password

The Advanced Settings option is only accessible to an administrator. The default administrator password is admin. For security reasons, you should change the default administrator password as soon as possible.

To change the administrator password via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) **->Set Password**.
2. Enter the current password in the **Current PWD** field.
3. Enter the new password in the **New PWD** field.

4. Enter the new password again in the **Confirm Password** field.



5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

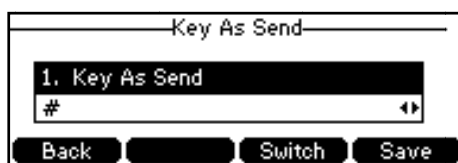
You can also configure the administrator password via web user interface at the path **Security->Password**.

Key as Send

You can set the "#" or "*" to perform as a send key while dialing.

To configure key as send via phone user interface:

1. Press **Menu->Features->Key as send**.
2. Press \leftarrow or \rightarrow , or the **Switch** soft key to select # or * from the **Key As Send** field, or select **Disable** to disable this feature.



3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.


You can also configure key as send via web user interface at the path **Features->General Information**.

Audio Settings

Volume


You can press the volume key to adjust the ringer volume when the phone is idle. You can also press the volume key to adjust the receiver volume of engaged audio devices (handset, speakerphone or headset), when the phone is in the dialing interface or during a call.

To adjust the volume when the phone is idle:

1. Press  to adjust the ringer volume.




To adjust the volume when the phone is during a call:

1. Press  to adjust the volume of currently used audio device (handset, speakerphone or headset).





Note

If the ringer volume is adjusted to minimum, the  icon will appear on the LCD screen.

Ring Tones

Ring tones are used to indicate incoming calls. You can select different ring tones to distinguish the different accounts registered on your phone, or to distinguish your phone from your neighbor's.

To select a ring tone for the phone via phone user interface:

1. Press **Menu**->**Settings**->**Basic Settings**->**Ring Tones**.
2. Press  or  to select the desired ring tone.

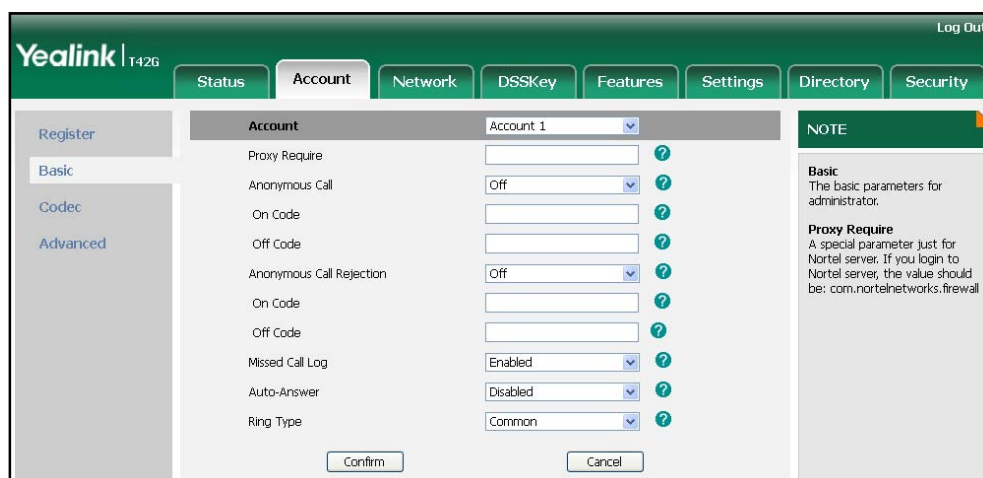


3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

To select a ring tone for the account via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Basic**.
4. Select the desired ring tone from the pull-down list of **Ring Type**.

If **Common** is selected, this account will use the ring tone selected for the phone at the path of the web user interface **Settings->Preference**.

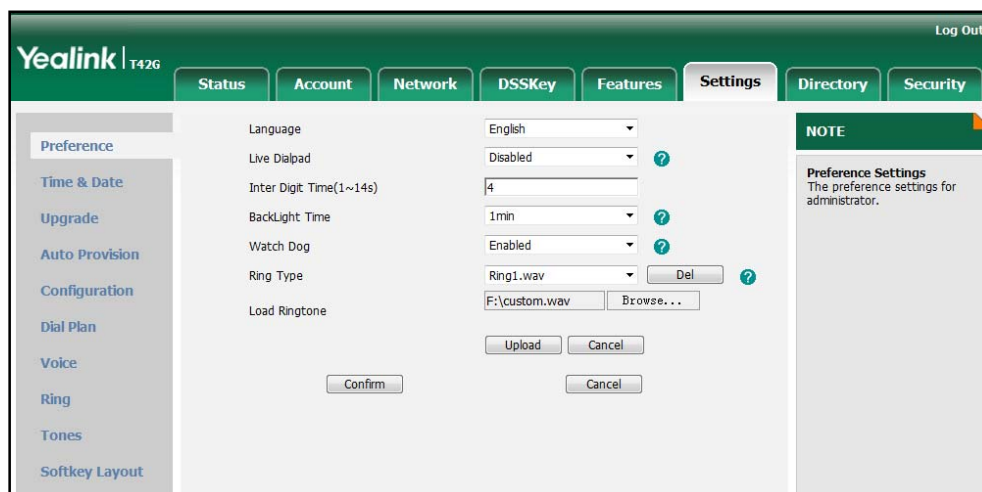


5. Click **Confirm** to accept the change.

Note You can select a ring tone for the account via web user interface only.

To upload a custom ring tone for your phone via web user interface:

1. Click on **Settings->Preference**.
2. Click **Browse** to locate a ring tone (the file format must be *.wav) file from your local system.



3. Click **Upload** to upload the file.

Note The ring tone for an incoming call on the phone may be different. For example, when the phone receives an incoming call from a contact stored in the local directory, it will play the ring tone assigned to the contact (if not assigned, play the ring tone assigned to the contact group) in the local directory (refer to [Adding Contacts](#) in the [Contact Management](#) section). Otherwise, the phone will play the ring tone assigned to the account. If both the contact ring tone and the account ring tone are not assigned, then the phone will play the ring tone assigned for the phone.

Contact Management





This section provides the operating instructions for managing contacts. The topics include:

- [Directory](#)
- [Local Directory](#)
- [Blacklist](#)
- [Remote Phonebook](#)

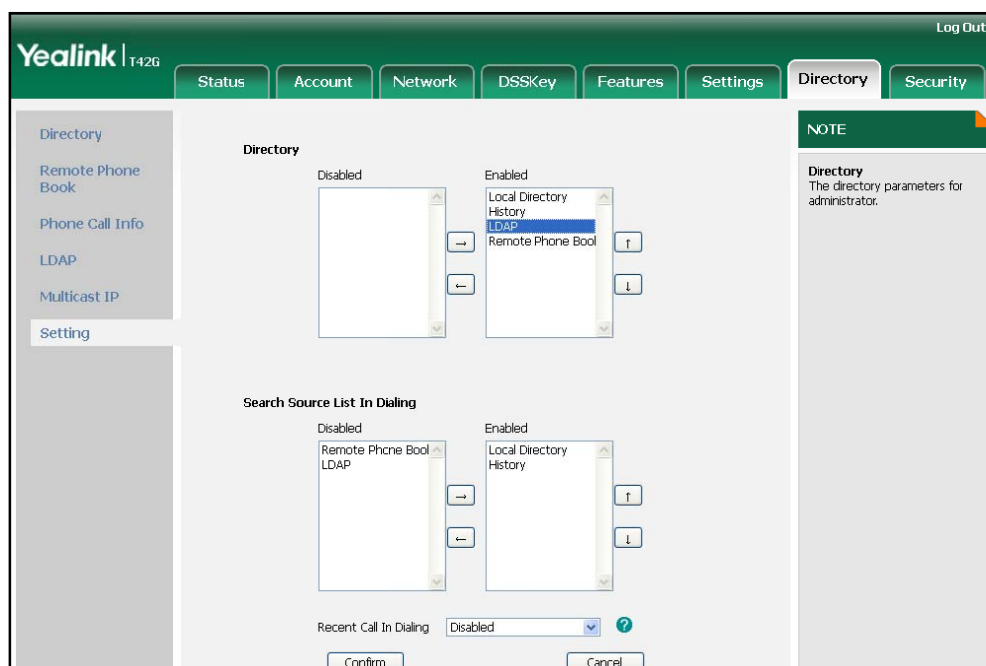
Directory

Directory provides easy access to the lists you use frequently. The lists can be Local Directory, History, Remote Phone Book and LDAP.

To configure the directory via web user interface:

1. Click on **Directory->Setting**.
2. In the **Directory** block, select the desired list from the **Disabled** column and then click  .
The selected list appears in the **Enabled** column.
3. Repeat the step 2 to add more lists to the **Enabled** column.
4. To remove a list from the **Enabled** column, select the desired list and click  .
5. To adjust the display order of enabled lists, select the desired list and click  or  .

The phone LCD screen displays the list(s) in the adjusted order.



6. Click **Confirm** to accept the change.

Note

You can configure the directory via web user interface only.

To check the directory via phone user interface:

1. Press the **Directory** soft key when the phone is idle.

The phone LCD screen displays the enabled list(s) in the directory.



If there is only one list in the directory, pressing the **Directory** soft key will enter this list directly.

Note

If the remote phonebook and LDAP features are not configured in advance, you cannot see the remote phonebook and LDAP lists on the phone user interface. For more information about the remote phonebook, refer to [Remote Phonebook](#) on page 35. For more information about the LDAP, contact your system administrator.

Local Directory

The built-in phone directory can store the names and phone numbers of your contacts. You can store up to 1000 contacts in your phone's local directory. There are four default groups in the local directory: All Contacts, Company, Family and Friend. You can add new groups and add new contacts to different groups in the local directory. You can edit, delete or search for a contact in the local directory. You can also call a contact from the local directory.

Adding Groups



To add a group to the local directory:

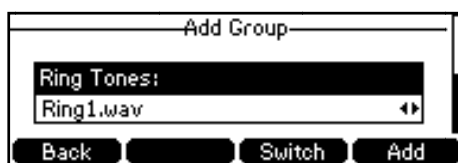
1. Press the **Directory** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

2. Press the **Group** soft key.
3. Enter the desired group name in the **Name** field.
4. Press  or  to select the desired ring tone from the **Ring Tones** field.



5. Press the **Add** soft key to accept the change or the **Back** soft key to cancel.

You can also edit or delete the Company, Family and Friend contact groups.

Editing Groups

To edit a group in the local directory:

1. Press the **Directory** soft key.



The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group.
3. Press the **Option** soft key and then select **Detail** from the prompt list.



4. Press  or  to highlight the group information and then edit.
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Deleting Groups

To delete a group from the local directory:

1. Press the **Directory** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.

If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group.
3. Press the **Option** soft key and then select **Delete** from the prompt list.

The phone LCD screen prompts the following warning:



4. Press the **OK** soft key to confirm the deleting or the **Cancel** soft key to cancel.

Adding Contacts

You can add contacts to the local directory in the following ways:

- Manually
- From call history
- From remote phonebook

Adding Contacts Manually

To add a contact to the local directory manually:

1. Press the **Directory** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.

If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group and press the **Enter** soft key.
3. Press the **Add** soft key.
4. Enter the name and the office, mobile or other numbers in the corresponding fields.



5. Press **◀** or **▶**, or the **Switch** soft key to select the desired account from the **Account** field.

If **Auto** is selected, the phone will use the first available account when placing calls to the contact from the local directory.

6. Press **◀** or **▶**, or the **Switch** soft key to select the desired ring tone from the **Ring Tones** field.
7. Press the **Add** soft key to accept the change or the **Back** soft key to cancel.

Note

If the contact has existed in the directory, the LCD screen will prompt "ContactName existed!".

Adding Contacts from Call History

To add a contact to the local directory from the call history:

1. Press the **History** soft key.
2. Press \uparrow or \downarrow to highlight the desired entry.
3. Press the **Option** soft key, and then select **Move to Contacts** from the prompt list.



4. Enter the contact name.
5. Press the **Save** soft key to accept the change.
The entry is successfully saved in the local directory.

Adding Contacts from Remote Phonebook

To add a contact to the local directory from the remote phonebook:

1. Press **Menu->Directory->Remote Phone Book**.
If Remote Phonebook is added to the directory, press **Directory->Remote Phone Book** to enter the remote phonebook.
2. Select the desired remote group and press the **Enter** soft key.
3. Press \uparrow or \downarrow to highlight the desired entry.
4. Press the **Option** soft key, and then select **Move to Contacts** from the prompt list.
5. Press the **Save** soft key to save the contact to the local directory.

If the contact has already existed in the local directory, the LCD screen will prompt "Overwrite the original contact?". Press the **OK** soft key to overwrite the original contact in the local directory or the **Cancel** soft key to cancel.



For more information on remote phonebook operating, refer to [Remote Phonebook](#) on page 35.

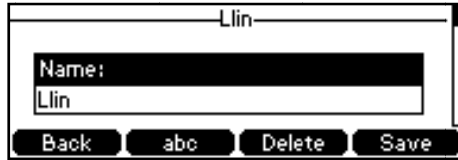
Editing Contacts



To edit a contact in the local directory:

1. Press the **Directory** soft key.
The IP phone enters the local directory directly as there is only Local Directory in the directory by default.
If Local Directory is removed from the directory, press **Menu->Directory->Local**

Directory to enter the local directory.

2. Select the desired contact group and press the **Enter** soft key.
3. Press  or  to highlight the desired contact.
4. Press the **Option** soft key, and then select **Detail** from the prompt list.



5. Press  or  to highlight the contact information and then edit.
6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.



Deleting Contacts

To delete a contact from the local directory:

1. Press the **Directory** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.

If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group and press the **Enter** soft key.
3. Press  or  to highlight the desired contact.
4. Press the **Option** soft key, and then select **Delete** from the prompt list.

The phone LCD screen prompts the following warning:



5. Press the **OK** soft key to confirm the deleting or the **Cancel** soft key to cancel.

You can also delete all contacts by pressing the **Option** soft key, and then select **Delete All**.

You can also manage contacts via web user interface at the path **Directory->Local Directory**.



Placing Calls to Contacts

To place a call to a contact from the local directory:

1. Press **Directory** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.

If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group and press the **Enter** soft key.
 3. Press  or  to highlight the desired contact.
 4. Press the **Dial** soft key.
- If only one number of the contact is stored in the local directory, press the **Dial** soft key to dial out the number.
 - If multiple numbers of the contact are stored in the local directory, press the **Dial** soft key to display a list of numbers.

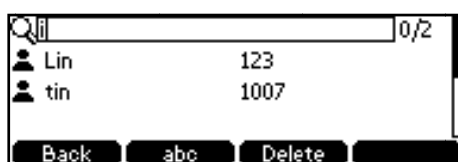
Press  or  to highlight the desired number.

Press the **Dial** soft key to dial out the number.

Searching for Contacts

To search for a contact in the local directory:

1. Press the **Directory** soft key.
The IP phone enters the local directory directly as there is only Local Directory in the directory by default.
If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.
2. Press the **Search** soft key.
3. Enter a few continuous characters of the contact name or continuous numbers of the contact phone number using the keypad.




The contacts whose name or phone number matches the characters entered will appear on the LCD screen. You can dial from the result list.




Search Source List in Dialing

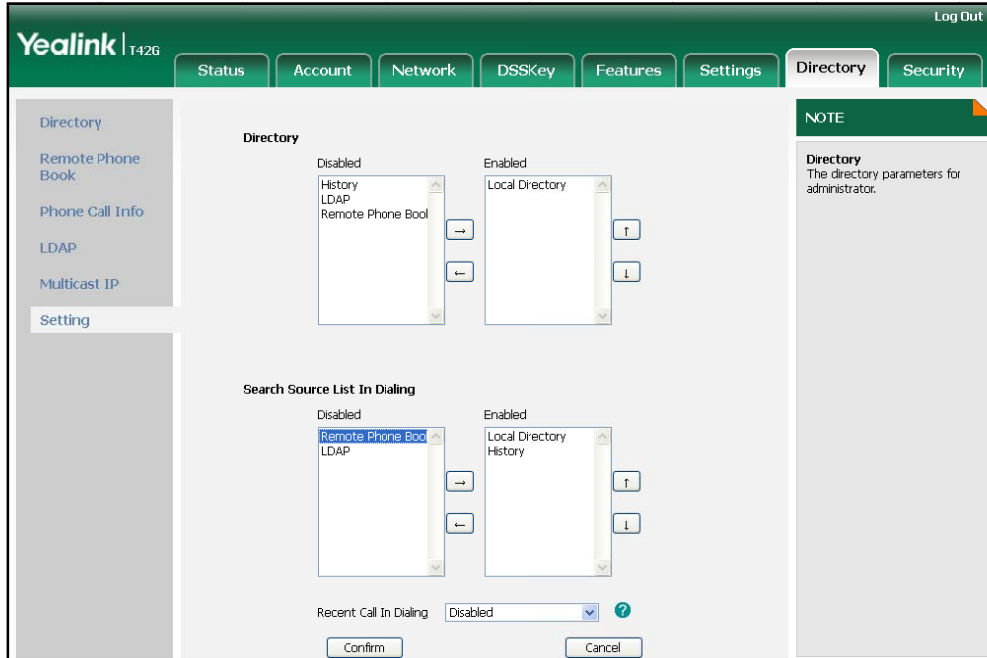
You can search a contact from the desired lists when the phone is in the dialing interface. The lists can be Local Directory, History, Remote Phonebook and LDAP.

To configure search source list in dialing via web user interface:

1. Click on **Directory->Setting**.
2. In the **Search Source List In Dialing** block, select the desired list from the **Disabled** column and click  .

The selected list appears in the **Enabled** column.

3. Repeat the step 2 to add more lists to the **Enabled** column.
 4. To remove a list from the **Enabled** column, select the desired list and click .
 5. To adjust the display order of lists, select the desired list and click  or .
- The phone LCD screen displays the search results in the adjusted order.





6. Click **Confirm** to accept the change.

Note You can configure search source list in dialing via web user interface only.

To search for a contact in the enabled search source lists:

1. Pick up the handset, press the speakerphone or press the line key.
2. Enter a few continuous characters of the entry's name or the entry's phone number using the keypad.

The entries in the enabled search source lists whose name or phone number matches the characters entered will appear on the LCD screen. You can press  or  to scroll to the desired contact, and then place a call to the contact.

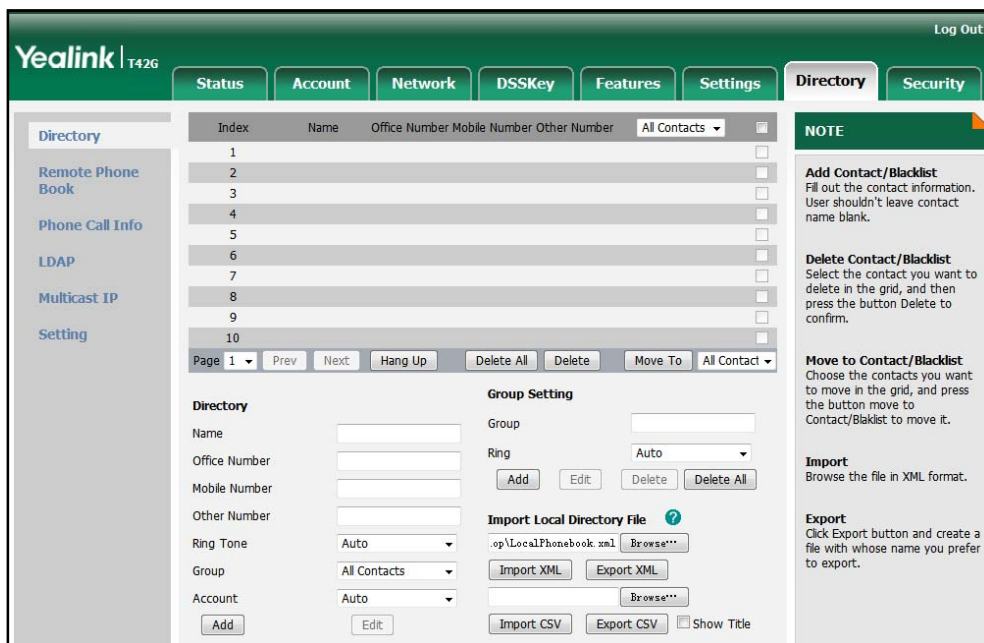


Importing/Exporting Contact Lists

You can manage your phone's local directory via phone user interface or web user interface. But you can only import or export the contact list via web user interface.

To import an XML file of contact list via web user interface:

1. Click on **Directory->Local Directory**.
2. Click **Browse** to locate a contact list file (file format must be .xml) from your local system.
3. Click **Import XML** to import the contact list.



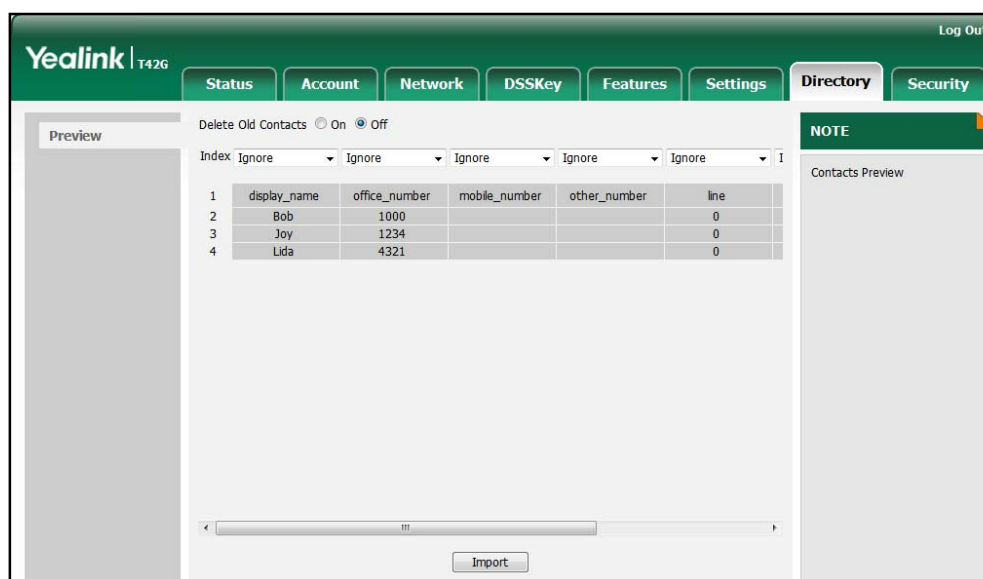
The web user interface prompts "The original contact will be covered, Continue?".

4. Click **OK** to complete importing the contact list.

To import a CSV file of contact list via web user interface:

1. Click on **Directory->Local Directory**.
2. Click **Browse** to locate a contact list file (file format must be .csv) from your local system.
3. (Optional.) Check the **Show Title** checkbox.
It will prevent importing the title of the contact information which is located in the first line of the CSV file.
4. Click **Import CSV** to import the contact list.
5. (Optional.) Mark the **On** radio box in the **Delete Old Contacts** field.
It will delete all existing contacts while importing the contact list.

- (Optional.) Select the contact information you want to import into the local directory from the pull down list of **Index**.



- Click **Import** to complete importing the contact list.

To export a contact list via web user interface:

- Click on **Directory->Local Directory**.
- Click **Export XML** (or **Export CSV**).
- Click **Save** to save the contact list to your local system.

Note You can import/export contact lists via web user interface only.



Blacklist

The built-in phone directory can store the names and phone numbers of the blacklist. You can store up to 30 contacts in your phone's blacklist directory. You can add, edit, delete or search for a contact in the blacklist directory. You can also call a contact in the blacklist directory, but an incoming call from the contact in the blacklist directory will be rejected automatically.

To add a contact to the blacklist directory manually:

- Press **Menu->Directory->Blacklist**.
- Press the **Add** soft key.
- Enter the name and the office, mobile or other numbers in the corresponding fields.



4. Press  or , or the **Switch** soft key to select the desired account from the **Account** field.

If **Auto** is selected, the phone will use the first available account when placing calls to the contact from the local directory.



5. Press the **Add** soft key to accept the change or the **Back** soft key to cancel.

To move a contact to the blacklist directory:

1. Press **Directory** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.

If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group and press the **Enter** soft key.
3. Press  or  to highlight the desired contact.
4. Press the **Option** soft key and then select **Add to Blacklist** from the prompt list.

The phone LCD screen prompts "Move selected to Blacklist?".



5. Press the **OK** soft key to confirm the setting.

Operating instructions of editing blacklists, deleting blacklists, placing calls to blacklists and searching for a contact in the blacklist, refer to the operating instructions of [Local Directory](#) on page 26.

Remote Phonebook

You can access your corporate directory through your phone. You can add contact to the local directory from the corporate directory. You can also call a contact from the corporate directory.

You can configure your new phone to access up to 5 corporate directories. All remote phonebooks must be less than 5MB in size. Contact your system administrator for the access URL of the corporate directory.

To configure an access URL for remote phonebook via web user interface:

1. Click on **Directory->Remote Phone Book**.
2. Enter the access URL in the **Remote URL** field.

3. Enter the name in the **Display Name** field.

Index	Remote URL	Display Name
1	http://10.3.6.153/phonebook.xml	Branch 1
2		
3		
4		
5		

Sremote Name: Disabled

Sremote Name Flash Time(Seconds): 3600

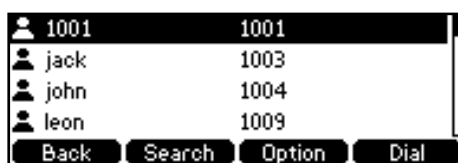
NOTE: Remote Phone Book. This feature allows you to download contact list from the server. Input the phonebook URL and rename the phone book.

4. Click **Confirm** to accept the change.

Note You can configure an access URL for remote phone book via web user interface only.

To access your corporate directory via phone user interface:

1. Press **Menu->Directory->Remote Phone Book**.
If Remote Phone Book is added to the directory, press **Directory->Remote Phone Book** to enter the remote phonebook.
2. Select the desired remote group, and then press the **Enter** soft key.
The phone connects to load the corporate directory, and then displays the corporate contacts on the LCD screen.

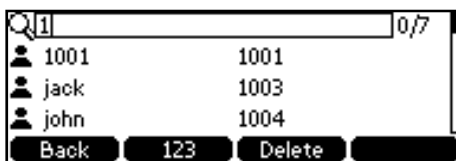


3. Press the **Back** soft key to back to the previous interface.

To search for a contact in the corporate directory:

1. Press **Menu->Directory->Remote Phone Book**.
If Remote Phone Book is added to the directory, press **Directory->Remote Phone Book** to enter the remote phonebook.
2. Select the desired remote group, and then press the **Enter** soft key to load the corporate directory.
3. Press the **Search** soft key.

4. Enter a few continuous characters of the contact name or continuous numbers of the contact phone number using the keypad.



The contacts whose name or phone number matches the characters entered will appear on the LCD screen. You can place a call from the result list.

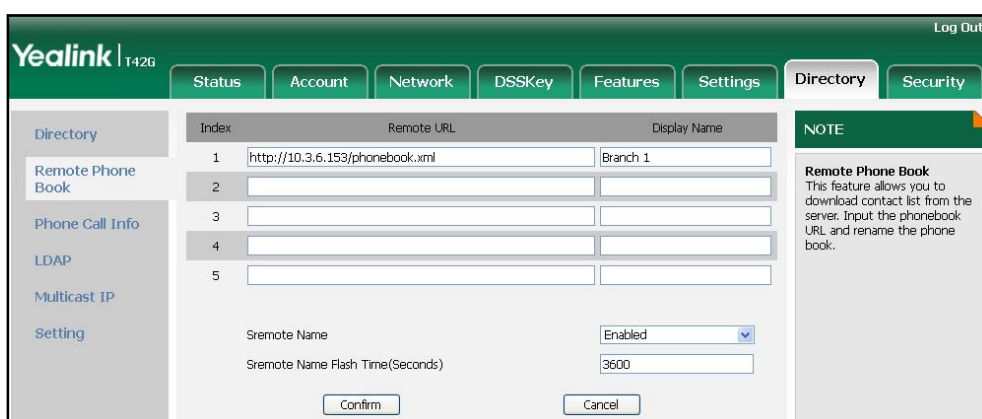
To place a call from the corporate directory:

1. Press **Menu->Directory->Remote Phone Book**.
If Remote Phone Book is added to the directory, press **Directory->Remote Phone Book** to enter the remote phonebook.
2. Select the desired remote group, and then press the **Enter** soft key to load the corporate directory.
3. Select the desired contact in the corporate directory.
4. Press the **Dial** soft key.

In addition, you can enable the phone to present the caller identity stored in the corporate directory when receiving a call.

To enable the presentation of caller identity stored in the remote phonebook:

1. Click on **Directory->Remote Phone Book**.
2. Select **Enabled** from the pull-down list of **Sremote Name**.
3. Enter the refresh period in the **Sremote Flash Time (Seconds)** field.
The default value is 3600 seconds.





4. Click **Confirm** to accept the change.

Call History Management

The SIP-T41P IP phone maintains call history lists of Placed Calls, Received Calls, Missed Calls and Forwarded Calls. The call history list supports up to 400 entries in all. You can check the call history, dial a call, add a contact or delete an entry from the call history list. To save the call history, you should enable the history record feature in advance.





To enable the history record feature via phone user interface:

1. Press **Menu->Features->History Setting**.
2. Press  or , or the **Switch** soft key to select **Enable** from the **History Record** field.







3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.





To check the call history:

1. Press the **History** soft key.
The LCD screen displays the all call list.
2. Press  or  to switch between All, Placed, Received, Missed and Forwarded call lists.
3. Press  or  to select the desired entry.
4. Press the **Option** soft key, and then select **Detail** from the prompt list.
The detailed information of the entry appears on the LCD screen.

To dial a call from the call history list:

1. Press the **History** soft key.
2. Press  or  to switch between All, Placed, Received, Missed and Forwarded call lists.
3. Press  or  to select the desired entry.
4. Press the **Send** soft key.

To add a contact to the local directory from the call history list:

1. Press the **History** soft key.
2. Press  or  to switch between All, Placed, Received, Missed and Forwarded call lists.
3. Press  or  to select the desired entry.

4. Press the **Option** soft key, and then select **Move to Contacts** (or **Add to Blacklist**) from the prompt list.



5. Enter the desired values in the corresponding fields.
6. Press the **Save** soft key to accept the change.

For more information, refer to [Contact Management](#) on page 25.

To delete an entry from the call history list:

1. Press the **History** soft key.
2. Press or to switch between All, Placed, Received, Missed and Forwarded call lists.
3. Press or to select the desired entry.
4. Press the **Delete** soft key.

To delete all entries from the call history list:

1. Press the **History** soft key.
2. Press or to switch between All, Placed, Received, Missed and Forwarded call lists.
3. Press the **Option** soft key, and then select **Delete All** from the prompt list.
4. Press the **OK** soft key.

The phone LCD screen prompts "Delete all records?".



5. Press the **OK** soft key to confirm the deletion or the **Cancel** soft key to cancel.

System Customizations

Headset Use

If you want to use a headset, physically connect your headset and activate the headset mode for use. For more information on physically connecting a headset, refer to [Phone Installation](#) on page 11.

Headset Mode Activation/Deactivation

To activate the headset mode:

1. Press  on the phone.

The headset key LED illuminates solid green when the headset mode is activated. Press the line key or the **Answer** soft key to answer an incoming call, the call will be connected to your headset automatically.

Enter the desired number and then press the **Send** soft key, the phone will place a call using the headset automatically. For more information about using the headset to place a call, refer to [Placing Calls](#) on page 61.

To deactivate the headset mode:

1. Press  again on the phone.

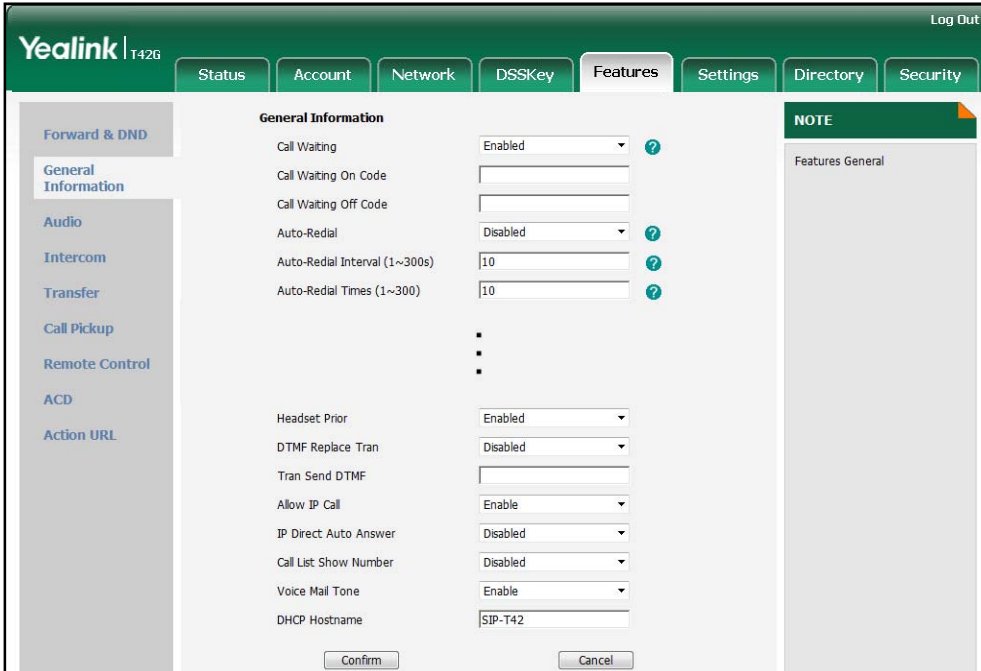
The headset key LED turns off when the headset mode is deactivated.

Headset Prior

You can use headset in priority when enabling the headset prior feature. This feature is especially useful for permanent or full-time headset users.

To enable the headset prior feature via web user interface:

1. Click on **Features->General Information**.
2. Select **Enabled** from the pull-down list of **Headset Prior**.




The screenshot shows the Yealink T42G web user interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Features' tab is selected. On the left, a sidebar lists various feature categories: Forward & DND, General Information, Audio, Intercom, Transfer, Call Pickup, Remote Control, ACD, and Action URL. The 'General Information' section is active, displaying a list of settings. The 'Headset Prior' setting is set to 'Enabled'. Other settings include Call Waiting (Enabled), Auto-Redial (Disabled), and various intervals and times. A 'NOTE' box on the right indicates 'Features General'. At the bottom, there are 'Confirm' and 'Cancel' buttons.

Setting	Value
Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto-Redial	Disabled
Auto-Redial Interval (1~300s)	10
Auto-Redial Times (1~300)	10
Headset Prior	Enabled
DTMF Replace Tran	Disabled
Tran Send DTMF	
Allow IP Call	Enable
IP Direct Auto Answer	Disabled
Call List Show Number	Disabled
Voice Mail Tone	Enable
DHCP Hostname	SIP-T42

3. Click **Confirm** to accept the change.

To use the headset prior feature, you should activate the headset mode in advance:

1. Physically connect the headset.
2. Press  to activate the headset mode.

Note

If the headset prior feature is enabled, the headset mode will not be deactivated until you press the **Headset** key again.

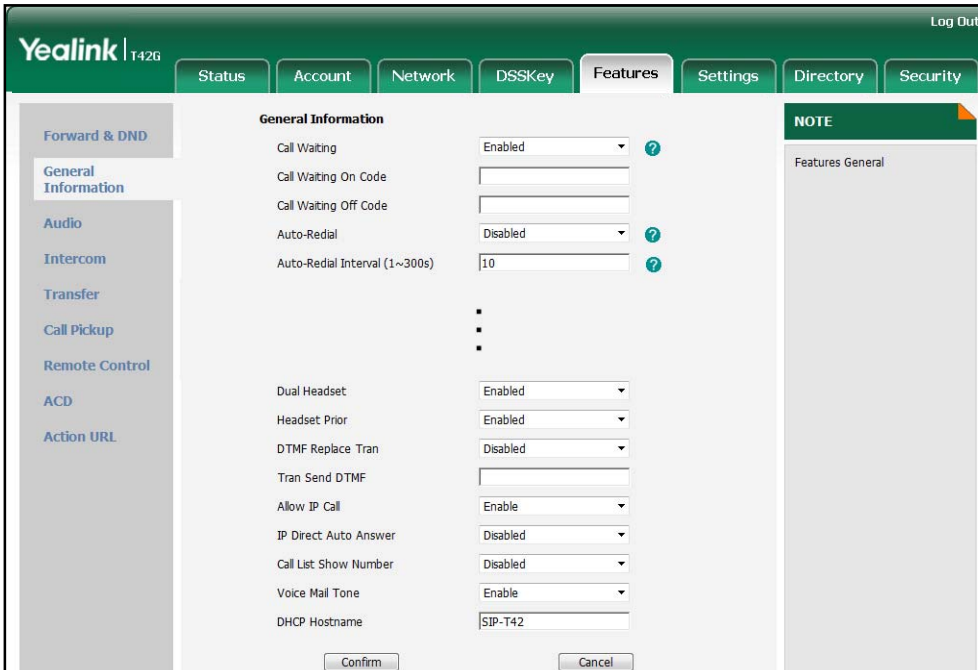
You can configure headset prior via web user interface only.

Dual Headset

You can use two headsets when enabling the dual headset feature. To use this feature, you must physically connect headsets to the 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 to listen to.

To enable the dual headset feature via web user interface:

1. Click on **Features->General Information**.
2. Select **Enabled** from the pull-down list of **Dual Headset**.



The screenshot shows the Yealink T42G web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. The 'Dual Headset' setting is set to 'Enabled'. Other settings include Call Waiting (Enabled), Call Waiting On Code, Call Waiting Off Code, Auto-Redial (Disabled), Auto-Redial Interval (1~300s) set to 10, and various other call management options like DTMF Replace Tran, Tran Send DTMF, Allow IP Call, IP Direct Auto Answer, Call List Show Number, Voice Mail Tone, and DHCP Hostname (SIP-T42). A 'NOTE' box on the right indicates 'Features General'. 'Confirm' and 'Cancel' buttons are at the bottom.

3. Click **Confirm** to accept the change.

Note

You can configure dual headset via web user interface only.

DSS Keys

There are two types of DSS keys: Line Keys and Programmable Keys. The details will be introduced in the following. The SIP-T41P IP phone supports 15 line keys.

Line Keys

You can assign predefined functionalities to the line keys. You can also define a label for the line key feature which will appear on the LCD screen. Line keys allow you to quick access features such as call return and voice mail. The line key LED will indicate the monitoring status when the keys are assigned specific features, such as BLF. The default key type of line key 1-3 is Line. The default key type of line key 4-15 is N/A, which indicates that this line key provides no functionality.

To assign functionality to the line key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired line key, and then press the **Enter** soft key.
3. Select the desired key type from the **Type** field.
4. (Optional.) Select the desired key event type from the **Key Type** field.
5. (Optional.) Select the desired line from the **Account ID** field.
6. (Optional.) Enter a string that will appear on the LCD screen in the **Label** field.
7. (Optional.) Enter the corresponding value in the **Value** field.
8. (Optional.) Enter the corresponding value in the **Extension** field.
9. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

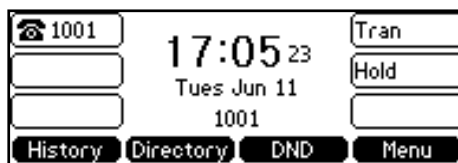
You can also configure the line key via web user interface at the path **DSSKey->Line Key**.

Note

When the phone is idle, you can also long press the line key to configure it directly on the phone.

You can assign functionalities to 15 line keys. These 15 line keys are displayed on three pages. When you assign functionality to line keys located in line key 1-6, the labels of the line keys display on the first page.

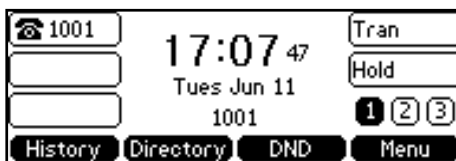
The following figure shows an example:



When you assign functionality to line key located in line key 7-15, the labels of the line key 6-10 display on the second page, and the labels of the line key 11-15 display on the

third page. The right bottom line key is used to switch pages.

The following figure shows an example:



The following table lists the page icon status:

Icons	Description
1	Indicates the current page of the line keys.
2 3	Indicates the non-current page of the line keys.

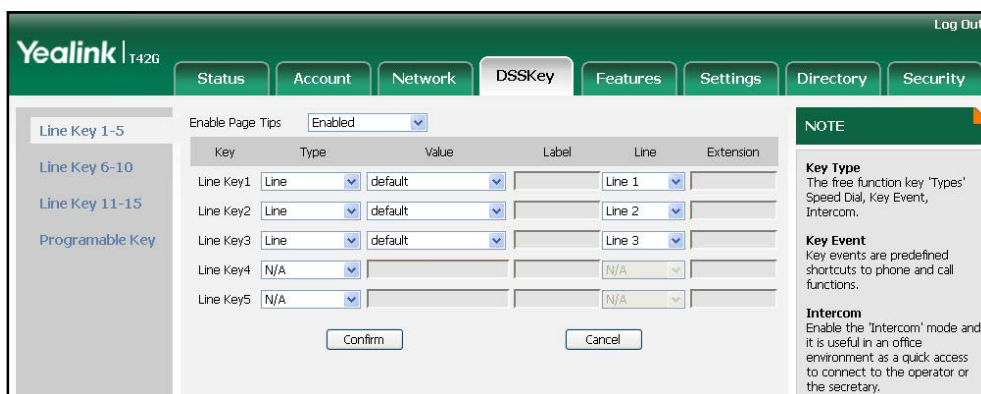
You can configure the page icon and page switch key LED to indicate the different status. This feature is disabled by default. For more information on the page switch key LED status, refer to [LED Instructions](#) on page 5.

The following table lists the page icon status for different states:

Icons Status	Description
Fast flashing	<p>The BLF monitored user receives an incoming call on the non-current page.</p> <p>The line receives an incoming call on the non-current page.</p> <p>There is a hold call on the line on the on the non-current page.</p>

To configure the page icon to indicate status via web user interface:

1. Click on **DSSKey->Line Key**.
2. Select **Enabled** in the **Enable Page Tips** field.



3. Click **Confirm** to accept the change.

The line key features are explained in the following subchapters in detail:

- Line
- Speed Dial
- Voice Mail
- Directed Pickup
- Group Pickup
- DTMF
- Prefix
- Local Group
- XML Group
- LDAP
- Conference
- Forward
- Transfer
- Hold
- DND
- Group Listening
- Zero Touch
- URL
- Directory

For more information, contact your system administrator.

Line

You can use this key feature to accept the incoming calls, place active calls on hold or resume a held call. It performs the same as a hard line key.

Dependencies: *Type (Line)*

Account ID (the account this feature will be applied to)

Value (Default/Lock/Float)

Usage: When receiving an incoming call, the DSS key LED flashes green:

1. Press the DSS key to accept the incoming call.
2. Press the DSS key to place the active call on hold.
3. Press the DSS key again to resume the held call.

If you select **Lock** in the **Value** field, the line keys located in the same position of the other page will be automatically assigned the line feature with **Lock** filled in the **Value** field. Then you can use this key on every page. This feature is only applicable to line key 1-3.

If you select **Float** in the **Value** field, the phone will automatically switch to the page the line key is located when the line key receives an incoming call.

Speed Dial

You can use this key feature to speed up dialing the numbers often used or hard to remember.

Dependencies: *Type (Speed Dial)*

Account ID (the account this feature will be applied to)

Label (key label displayed on the LCD screen)

Value (the number you want to dial out)

Usage: Press the DSS key to dial out the number specified in the **Value** field, using the account selected from the **Account ID** field.

Voice Mail

You can use this key feature to connect voice mail quickly. For more information, refer to [Voice Mail](#) on page 103.

Dependencies: *Type (Key Event)*

Key Type (Voice Mail)

Account ID (the account this feature will be applied to)

Label (key label displayed on the LCD screen)

Value (the voice mail access code)

Usage: Press the DSS key to dial out the voice mail access code, you can follow the voice prompt to listen to the voice mails.

Directed Pickup

You can use this key feature to answer someone else's incoming call on the phone.

Dependencies: *Type (Key Event)*

Key Type (Direct Pickup)

Account ID (the account this feature will be applied to)

Label (key label displayed on the LCD screen)

Value (the direct pickup code followed by the specific phone number)

Usage: Press the DSS key on your phone when the specific phone number receives an incoming call. The call is answered on your phone.

Group Pickup

You can use this key feature to answer incoming calls in a group that is associated with their own group.

Dependencies: *Type (Key Event)*

Key Type (Group Pickup)

Account ID (the account this feature will be applied to)

Label (key label displayed on the LCD screen)

Value (the group pickup feature code)

Usage: Press the DSS key on your phone when a phone number in the group receives an incoming call. The call is answered on your phone.

DTMF

You can use this key feature to send the specification of arbitrary key sequences via DTMF.

Dependencies: *Type (Key Event)*

Key Type (DTMF)

Label (key label displayed on the LCD screen)

Value (DTMF sequence)

Note

DTMF sequence can only contain "0-9", "*", "#" and "A-D".

Usage: Press the DSS key during an active call to send the key sequence specified in the **Value** field.

Prefix

You can use this key feature to add a specified prefix number before the dialed number.

Dependencies: *Type (Key Event)*

Key Type (Prefix)

Label (key label displayed on the LCD screen)

Value (the prefix number)

Usage: Press the DSS key when the phone is idle, the phone enters into the dialing screen and displays the prefix number which you specified in the **Value** field, enter other digits and then dial out.

Local Group

You can use this key feature to access the contact group in the local directory quickly. For more information, refer to [Local Directory](#) on page 26.

Dependencies: *Type (Key Event)*

Key Type (Local Group)

Local Group (the contact group name you want to access)

Label (key label displayed on the LCD screen)

Usage: Press the DSS key to access the contact group specified in the **Local Group** field.

XML Group

You can use this key feature to access the remote group in your corporate directory

quickly. You should configure the remote phonebook in advance. For more information, refer to [Remote Phonebook](#) on page 35.

Dependencies: *Type (Key Event)*

Key Type (XML Group)

XML Group (the remote group name you want to access if the remote phonebook is configured)

Label (key label displayed on the LCD screen)

Usage: Press the DSS key to access the remote group specified in the **XML Group** field.

LDAP

You can use this key feature to access the LDAP search interface quickly.

Dependencies: *Type (Key Event)*

Key Type (LDAP)

Label (key label displayed on the LCD screen)

Usage:

1. Press the DSS key to access the LDAP search interface.
2. Enter a few continuous characters of the contact name or continuous numbers of the contact phone number using the keypad.

The contacts whose name or phone number matches the characters entered will appear on the LCD screen.

Conference

You can use this key feature to set up a conference call. For more information, refer to [Conference](#) on page 79.

Dependencies: *Type (Key Event)*

Key Type (Conf)

Label (key label displayed on the LCD screen)

Value (the number you want to add to the conference)

Usage: Press the DSS key during an active call to set up a conference with the number specified in the **Value** field.

Forward

You can use this key feature to forward an incoming call to someone else. For more information, refer to [Call Forward](#) on page 73.

Dependencies: *Type (Key Event)*

Key Type (Forward)

Label (key label displayed on the LCD screen)

Value (the number you want to forward to)

Usage: Press the DSS key to forward an incoming call to the number specified in the **Value** field.

Transfer

You can use this key feature to handle the call differently depending on the transfer mode on DSS key when there is an active call on the phone.

Dependencies: *Type (Key Event)*

Key Type (Tran)

Label (key label displayed on the LCD screen)

Value (the number you want to transfer to)

Usage:

- When the transfer mode on DSS key is **Blind Transfer**, press the DSS key to complete the blind transfer to the number specified in the **Value** field.
- When the transfer mode on DSS key is **Attended Transfer**, press the DSS key to dial out the number specified in the **Value** field, and then you can perform the attended or semi-attended transfer.
- When the transfer mode on DSS key is **New Call**, press the DSS key to place a new call to the number specified in the **Value** field.

Note

You can configure the transfer mode on DSS key via web user interface at the path **Features->Transfer**. For more information on how to configure the transfer mode on DSS key, refer to [Busy Lamp Field \(BLF\)](#) on page 89.

Hold

You can use this key feature to place an active call on hold or retrieve a held call.

Dependencies: *Type (Key Event)*

Key Type (Hold)

Label (key label displayed on the LCD screen)

Usage:

1. Press the DSS key during an active call to place the call on hold.
2. Press the DSS key again to retrieve the held call.

DND

You can use this key feature to activate or deactivate the DND mode.

Dependencies: *Type (Key Event)*

Key Type (DND)

Label (key label displayed on the LCD screen)

Usage:

1. Press the DSS key to activate the DND mode.
2. Press the DSS key again to deactivate the DND mode.

Note

When the DND mode is activated, the incoming calls will be rejected automatically.

Group Listening

You can use this key feature to activate 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 through the handset/headset, meanwhile the others nearby can only listen through the speaker.

Dependencies: *Type (Key Event)*

Key Type (Group Listening)

Label (key label displayed on the LCD screen)

Usage:

1. During a call, press the DSS key to activate the group listening mode.
You can then speak and listen through the handset/headset, other people at your side can listen through speaker at the same time.
2. Press the DSS key again to deactivate the group listening mode.

Zero Touch

You can use this key feature to configure auto provision and network parameters quickly.

Dependencies: *Type (Key Event)*

Key Type (Zero Touch)

Label (key label displayed on the LCD screen)

Usage:

1. Press the DSS key to access the zero touch screen.
2. Press the **OK** soft key within a few seconds.
3. Configure the network parameters in the corresponding fields.
4. Press the **Next** soft key.
5. Configure the auto provision parameters in the corresponding fields.
6. Press the **OK** soft key.

The phone will reboot to update configurations.

URL

You can use this key feature to trigger the phone to send a HTTP GET request contained a specific URL.

Dependencies: *Type (URL)*

Label (key label displayed on the LCD screen)

URL (the URL contained in the HTTP GET request)

Usage: Press the DSS key to trigger the phone to send a HTTP GET request contained the URL specified in the **URL** field.

Directory

You can use this key feature to easy access to the lists you use frequently. For more

information, refer to [Directory](#) on page 25.

Dependencies: *Type (Key Event)*

Key Type (Directory)

Label (key label displayed on the LCD screen)

Usage: Press the DSS key to immediately access to the lists you use frequently.

Programmable Keys

You can customize the soft keys, navigation keys and function keys.

To customize the programmable keys via web user interface:

1. Click on **DSSKey->Programmable Key**.
2. Customize specific features for these keys.

Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local Call Log			
SoftKey 2	Directory	N/A			
SoftKey 3	DND	N/A			
SoftKey 4	Menu	N/A			
Up	History	Local Call Log			
Down	Directory	N/A			
Left	Switch Account	N/A			
Right	Switch Account	N/A			
OK	Status	N/A			
Cancel	N/A	N/A			
MUTE	N/A	N/A			

3. Click **Confirm** to accept the change.

Note

You can configure the programmable keys via web user interface only.

Account Management

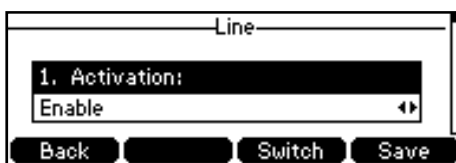
You can register one or multiple accounts on the SIP-T41P IP phone. You can also configure each line key associated with an account or configure multiple line keys associated with an account.

Account Registration

To register an account via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) **->Accounts**.
2. Select the desired account and then press the **Enter** soft key.

3. Select **Enable** from the **Activation** field.



4. Enter the desired values in **Label**, **Display Name**, **Register Name**, **User Name**, **Password** and **SIP Server1** fields respectively. Contact your system administrator for more information.
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.
6. Repeat steps 2 to 5 until you have registered all accounts.

The following figures show single or multiple accounts registered on the phone:

Single account:



Multiple accounts:



To disable an account via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Accounts**.
2. Select the desired account and then press the **Enter** soft key.
3. Select **Disable** from the **Activation** field.
4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also register or disable an account via web user interface at the path **Account**->**Register**.

Multiple Line Keys per Account

You can configure multiple line keys associated with an account. This enhances call visualization and simplifies call handling.

If this is the case, the LCD screen resembles the following figure:



Incoming calls to this line will be distributed evenly among the available line keys. Similarly, outgoing calls will be distributed.

Your phone can be configured to have a combination of lines with a single line key and lines with multiple line keys.

Dial Plan

Dial plan is a string of characters that governs the way your SIP-T41PIP phone processes the inputs received from your phone keypad. The SIP-T41P IP phone supports the following dial plan features:

- [Replace Rule](#)
- [Dial-now](#)
- [Area Code](#)
- [Block Out](#)

The basic expression syntax you need to know:

Syntax	Description
.	The dot "." can be used as a placeholder or multiple placeholders for any character. Example: "12." would match "123", "1234", "12345", "12abc", etc.
x	An "x" can be used as a placeholder for any character. Example: "12x" would match "121", "122", "123", "12a", etc.
[]	The square brackets "[]" can be used as a placeholder for a single character which matches any of a set of characters. Example: "91[5-7]1234" would match "9151234", "9161234", "9171234", etc.
()	The parentheses "()" can be used to group together patterns, for instance, to logically combine two or more patterns. Example: "([1-9])([2-7])3" would match "923", "153", "773", etc.
\$	The "\$" should be followed by the sequence number of a parenthesis. The "\$" plus the sequence number means the whole character or characters placed in the parenthesis. The number directs to the right parenthesis when there are more than one. Example: A replace rule configuration: Prefix: "9([5-7])(.)", Replace: "5\$2". When you enter "96123" to dial out on your phone, the number will be replaced as "5123" and then dialed out. "\$2" means the characters in the second parenthesis, that is, "123".

Note

You can configure the dial plan rules via web user interface only.

Replace Rule

You can configure one or more replace rules to remove the specified string and replace it with another string. You can configure a pattern with wildcards (expression syntax refer to the table above), so that any string that matches the pattern will be replaced. This feature is convenient for you to dial out a long number. For example, a replace rule is configured as "Prefix: 1" and "Replace: 1234567", when you try to dial out the number "1234567", you just need to enter "1" on the phone and then press the Send soft key.

To add a replace rule via web user interface:

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Enter the string (e.g., 2xx) in the **Prefix** field.
3. Enter the string (e.g., 12345) in the **Replace** field.
4. Enter the desired line ID in the **Account** field or leave it blank.

Index	Prefix	Replace	Account	<input type="checkbox"/>
1				<input type="checkbox"/>
2				<input type="checkbox"/>
3				<input type="checkbox"/>
4				<input type="checkbox"/>
5				<input type="checkbox"/>
6				<input type="checkbox"/>
7				<input type="checkbox"/>
8				<input type="checkbox"/>
9				<input type="checkbox"/>
10				<input type="checkbox"/>

Prefix Replace Account

NOTE

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

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

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

x
Matches any single digit/character which is dialed.

.
Matches an arbitrary number of digits.

5. Click **Add** to add the replace rule.

When you enter the number "234" using the keypad and then dial out, the phone will dial out "12345" instead.

Note

The valid values of **Account** field can be one or more digits among 1, 2, and 3. Each digit must be separated by a ",". For example: when you enter the value "1, 2" in the **Account** field, this replace rule will apply to account1 and account2.

If you leave the **Account** field blank, the replace rule will apply to all accounts.

To edit a replace rule via web user interface:

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Select the desired replace rule by checking the checkbox.

3. Edit the values in the **Prefix** and **Replace** fields.
4. Enter the desired line ID in the **Account** field or leave it blank.
5. Click **Edit** to edit the replace rules.

To delete one or more replace rules via web user interface:

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Select the one or more replace rules by checking the checkbox(es).
3. Click **Del** to delete the replace rule(s).

Dial-now

You can configure one or more dial-now rules on your phone. When the dialed out number matches the dial-now string, the number will be dialed out automatically. For example, a dial-now rule is configured as "2xx", then entering any three-digit string begins with 2 will be dialed out automatically on the phone.

To add a dial-now rule via web user interface:

1. Click on **Settings->Dial Plan->Dial-now**.
2. Enter the desired value (e.g., 1234) in the **Rule** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

For more information about the valid values of **Account** field, refer to [Replace Rule](#) on page 53.

Index	Dial-now Rule	Account	<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>
10			<input type="checkbox"/>

Rule: 1001 Account: 2

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.

4. Click **Add** to add the dial-now rule.

When you enter the number "1234" using the keypad, the phone will dial out "1234" automatically without pressing any key.

Note

You can also edit or delete the dial-now rule, refer to [Replace Rule](#) on page 53 for more information.

Time Out for Dial-Now Rule

You can configure the interval for dial-now rules. That is, you can configure your phone to dial out the phone number automatically, which matches a dial-now rule, after the specified delay time.

To configure the time out for dial-now rule via web user interface:

1. Click on **Features->General Information**.
2. Enter the time within 1-14 (in seconds) in the **Time-Out For Dial-Now Rule** field.

The screenshot shows the Yealink web user interface with the 'Features' tab selected. The 'General Information' section is active, displaying various settings. The 'Time-Out For Dial-Now Rule' field is set to 2. A 'NOTE' section on the right provides details for 'Call Waiting', 'Key As Send', and 'Hotline Number'.

Setting	Value
Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#
Reserve # in User Name	Enabled
Hotline Number	
Hotline Delay(0~10s)	4
Busy Tone Delay (Seconds)	0
Return Code When Refuse	496 (Busy Here)
Return Code When DND	480 (Temporarily Not Av)
Time-Out For Dial-Now Rule	2

NOTE

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.

3. Click **Confirm** to accept the change.

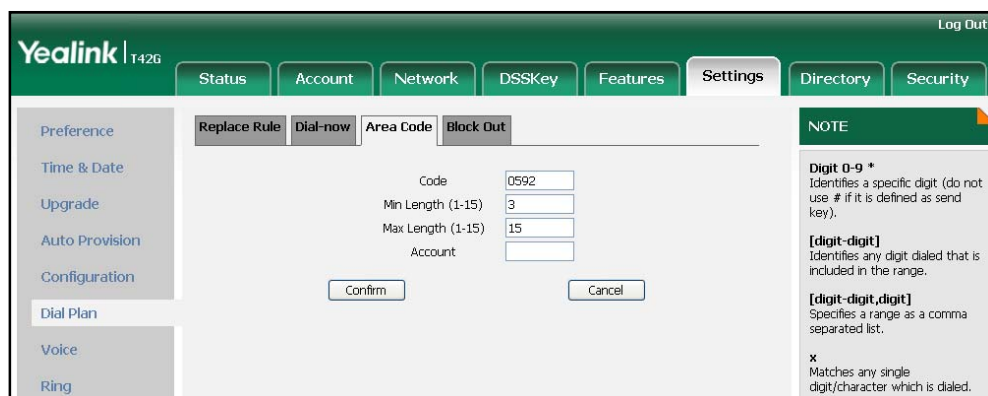
Area Code

Area codes are also known as Numbering Plan Areas (NPAs). They usually indicate geographical areas in a country. This feature is necessary only when dialing the number outside the code area. For example, area code is configured as "Code: 011, Min Length: 4, Max Length: 11". When you dial out the number "56789" (the length of the number is between 4 and 11), the phone will add the area code and dial out the number "01156789".

To configure the area code and lengths via web user interface:

1. Click on **Settings->Dial Plan->Area Code**.
2. Enter the desired values in the **Code**, **Min Length (1-15)** and **Max Length (1-15)** fields.
3. Enter the desired line ID in the **Account** field or leave it blank.

For more information about the valid values of **Account** field, refer to [Replace Rule](#) on page 53.



4. Click **Confirm** to accept the change.

Note

The default values of minimum and maximum length are 1 and 15 respectively.

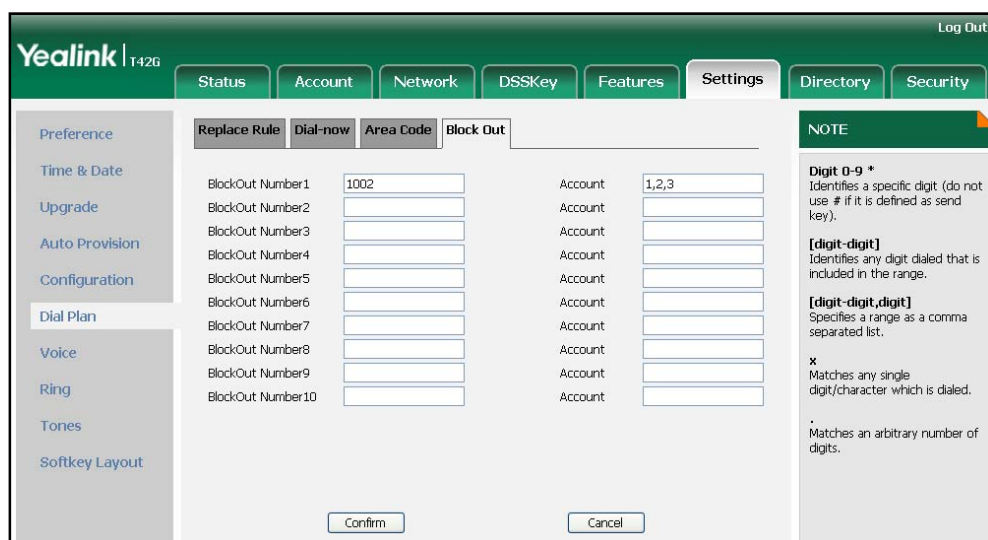
Block Out

You can block some specific numbers from being dialed on your phone. When you dial a block out number on your phone, the dialing will fail and the LCD screen will prompt "Forbidden Number".

To add a block out number via web user interface:

1. Click on **Settings->Dial Plan->Block Out**.
2. Enter the desired value in the **BlockOut Number** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

For more information about the valid values of the **Account** field, refer to [Replace Rule](#) on page 53.



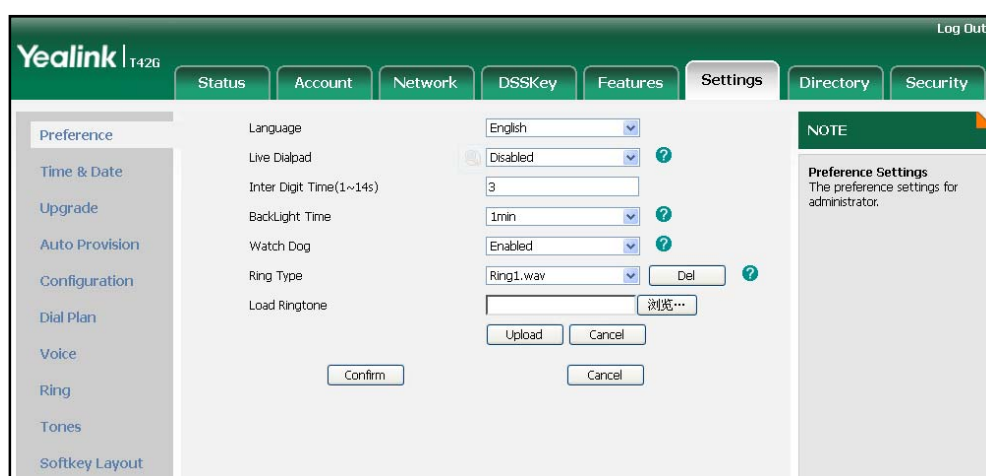
4. Click **Confirm** to add the block out number.

Live Dialpad

You can enable the live dialpad feature on the SIP-T41P IP phone, which enables the IP phone to automatically dial out the phone number without pressing any other key. You can also configure a period of delay time to dial out the phone number. The phone will dial out the phone number automatically after the specified delay time.

To enable the live dialpad via web user interface:

1. Click on **Settings->Preference**.
2. Select **Enabled** from the pull-down list of **Live Dialpad**.
3. Enter the desired delay time (in seconds) in the **Inter Digit Time (1~14s)** field.



4. Click **Confirm** to accept the change.

Note

The default delay time of live dialpad feature is 4s.
You can configure live dialpad via web user interface only.

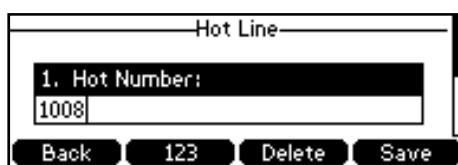
Hotline

You can dial a hotline number immediately once you lift the handset, press the speakerphone key or press a line key. You can also configure a period of delay time to dial out the hotline number, the phone will dial out the hotline number automatically after the specified delay time.

To configure the hotline number via phone user interface:

1. Press **Menu->Features->Hot Line**.
2. Enter the desired number in the **Hot Number** field.
3. Enter the delay time (in seconds) in the **HotLine Delay** field.

The valid values range from 0 to 10.



4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure hotline via web user interface at the path **Features->General Information**.

Basic Call Features

The SIP-T41P IP phone is designed to be easily used like a regular phone on a public switched telephone network (PSTN). You can place calls, answer calls, transfer a call to someone else, or conduct a conference call.

This chapter provides basic operating instructions for the SIP-T41P IP phone. The topics include:

- [Placing Calls](#)
- [Answering Calls](#)
- [Ending Calls](#)
- [Redialing Numbers](#)
- [Recent Call In Dialing](#)
- [Auto Answer](#)
- [Auto Redial](#)
- [Call Return](#)
- [Call Mute](#)
- [Call Hold/Resume](#)
- [Do Not Disturb \(DND\)](#)
- [Call Forward](#)
- [Call Transfer](#)
- [Call Waiting](#)
- [Conference](#)
- [Call Park](#)
- [Call Pickup](#)
- [Anonymous Call](#)
- [Anonymous Call Rejection](#)

If you require additional information or assistance with your new phone, contact your system administrator.

Placing Calls

You can place a call in three ways using your SIP-T41P IP phone:

- Using the handset
- Using the speakerphone
- Using the headset

You can also dial the number first, and then choose the way you want to speak to the other party.



You can search the call history, the local contact directory or the remote phonebook and dial from the search results. For more information, refer to [Contact Management](#) on page 25 and [Call History Management](#) on page 38.

During a call, you can alternate between Speakerphone, Headset, or Handset mode by pressing the Speakerphone key, the Headset key, or picking up the handset.

The call duration of active calls is visible on the LCD screen. In the figure below, the call to the number "1007" has lasted 15 seconds.



To place a call using the handset:

1. Pick up the handset.
2. Enter the desired number using the keypad.
3. Press , , or the **Send** soft key.

By default, the # key is configured as a send key. You can set the * key as the send key or set neither of them as send keys. For more information, refer to the [Key as Send](#) on page 22.




Note

You can also dial using the SIP URI or IP address. To obtain the IP address of a phone, press the **OK** key. The maximum SIP URI or IP address length is 32 characters. For example, SIP URI: 3606@sip.com, IP: 192.168.1.15.

Your phone may not support direct IP dialing. Contact your system administrator for more information.

To place a call using the hands-free speakerphone mode:

Do one of the following:


- With the handset on-hook, press  or the line key to obtain a dial tone. Enter the desired number using the keypad. Press , , or the **Send** soft key.

- With the handset on-hook, enter the desired number using the keypad.

Press  ,  ,  or the **Send** soft key.



To place a call using the headset:


Do one of the following:

- With the optional headset connected, press  to activate the headset mode.



Press the line key to obtain a dial tone.

Enter the desired number using the keypad.

Press  ,  , or the **Send** soft key.

- With the optional headset connected, press  to activate the headset mode.

Enter the desired number using the keypad.

Press  ,  , or the **Send** soft key.

Note



To permanently use the headset mode, refer to [Headset Prior](#) on page 40.

To place multiple calls:

You can have more than one call on your SIP-T41P IP phone. To place a new call during an active call, do one of the following:

- Press the line key. The active call is placed on hold.



Enter the desired number using the keypad.



Press  ,  , or the **Send** soft key.

- Press the **Hold** soft key to place the original call on hold.

Press the **NewCall** soft key.

Enter the desired number using the keypad.

Press  ,  , or the **Send** soft key.

You can press  or  to switch between the calls, and then press the **Resume** soft key to retrieve the desired call.

Answering Calls

When you are not in another call, you can answer a call in three ways:

- Using the handset
- Using the speakerphone
- Using the headset

Note

You can ignore incoming calls by pressing the **X** key, the **Reject** soft key or the **Silence** soft key. You can also activate Do Not Disturb mode to ignore the incoming calls without ring on your phone. For more information, refer to [Do Not Disturb \(DND\)](#) on page 70.

You can forward incoming calls to someone else by pressing the **Fwd** soft key. For more information, refer to [Call Forward](#) on page 73.

Answering When Not in Another Call


In all cases, call duration and destination of the active call will appear on the LCD screen.

To answer a call using the handset:

1. Pick up the handset.


To answer a call using the hands-free speakerphone mode:

Do one of the following:

- Press  .
- With the handset on-hook and the headset mode deactivated, press the **Answer** soft key.
- With the handset on-hook and the headset mode deactivated, press the line key (the line key LED flashes green).

To answer a call using the headset:

Do one of the following:



- Press  .
- With the headset mode activated, press the **Answer** soft key.
- With the headset mode activated, press the line key (the line key LED flashes green).

Answering When in Another Call

If you have an active call, and an incoming call arrives on the phone, do one of the following:

- Press the **Answer** soft key.

The incoming call is answered and the original call is placed on hold.


- Press  to access the new call.
- Press  or the **Answer** soft key.

The incoming call is answered and the original call is placed on hold.

Ending Calls

To end a call:

Do one of the following:

- If you are using the handset, press the **Cancel** soft key or hang up the handset.
- If you are using the headset, press the **Cancel** soft key.
- If you are using the speakerphone, press  or the **Cancel** soft key.

Note





To end a call placed on hold, you can press the **Cancel** soft key to end the call directly, or press the **Resume** soft key to resume the call before ending it.

Redialing Numbers

To redial the last dialed number from your phone:

1. Press  twice.
A call to your last dialed number is attempted.

To redial a previously dialed number from your phone:

1. Press  when the phone is idle.
2. Press  or  to select the desired entry from the placed calls list, and then press  or the **Send** soft key.

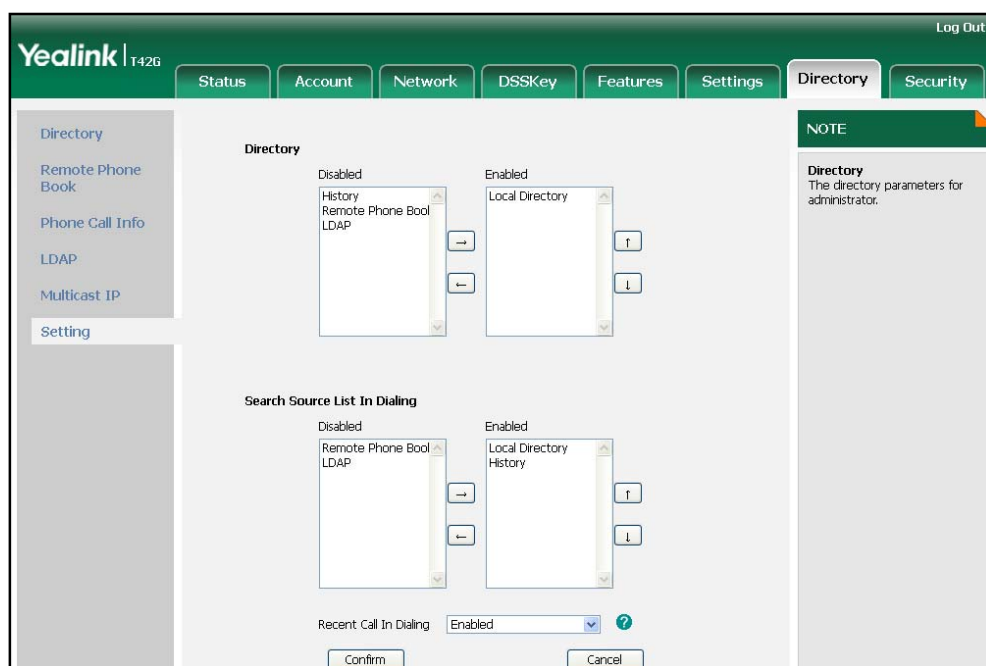
Recent Call In Dialing

You can view the placed calls list when the phone is in the pre-dialing interface. You should enable the recent call in dialing feature in advance.

To enable the recent call in dialing feature via web user interface:

1. Click on **Contacts->Setting**.

2. Select **Enabled** from the pull-down list of **Recent Call In Dialing**.



3. Click **Confirm** to accept the change.

Note

You can enable the recent call in dialing feature via web user interface only.

To view placed calls list when the phone is in the dialing interface:

1. Pick up the handset, press the speakerphone or press the line key.
The phone LCD screen displays the placed calls list.



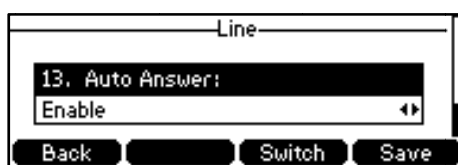
Auto Answer

The auto answer feature allows the phone to automatically answer an incoming call on the line. You can enable the auto answer feature on a per-line basis.

To configure auto answer via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) ->**Accounts**.
2. Select the desired account and then press the **Enter** soft key.

- Press  or , or the **Switch** soft key to select **Enable** from the **Auto Answer** field.



- Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

The **AA** icon appears on the phone LCD screen.

You can also configure auto answer via web user interface at the path **Account->Basic**.



Note

The auto answer feature is only applicable when there is no other call in progress on the phone.

Auto Redial

You can enable the auto redial feature to redial the phone number automatically when the called party is busy. You can also configure the times of the auto redial and the time to wait between redial attempts.

To configure auto redial via phone user interface:

- Press **Menu->Features->Auto Redial**.
- Press  or , or the **Switch** soft key to select **Enable** from the **Auto Redial** field.



- Enter the desired time (in seconds) in the **Redial Interval** field.
The default time interval (in seconds) is 10.
- Enter the desired times in the **Redial Times** field.
The default value is 10.
- Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure auto redial via web user interface at the path **Features->General Information**.

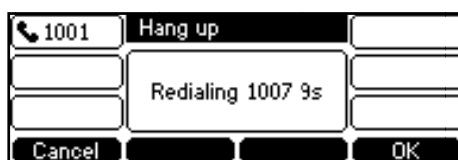
To use auto redial:

When the called party is busy, the phone LCD screen prompts the following:



1. Press the **OK** soft key to activate auto redial.

The phone LCD screen prompts the following:







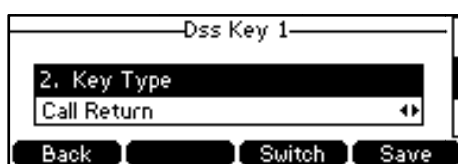
2. Wait for a period of time or press the **OK** soft key to redial the phone number. The phone will retry as many times as you configured until the called party is idle.

Call Return

You can press a call return key to place a call back to the last incoming call.

To configure a call return key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select a desired DSS key.
3. Press  or , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press  or , or the **Switch** soft key to select **Call Return** from the **Key Type** field.
5. (Optional.) Enter a string that will appear on the LCD screen in the **Label** field.



6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure a call return key via web user interface at the path **DSSKey->Line key**.

Call Mute

You can mute the microphone of the active audio device during an active call, and then the other party cannot hear you.


To mute a call:

1. Press  during an active call.

The phone LCD screen indicates that the call is on mute.



To un-mute a call:

1. Press  again to un-mute the call.

The mute key LED turns off.

Call Hold/Resume

You can place an active call on hold. At any time, at most one active call can be in progress on your phone, other calls can be received and made while placing the original call on hold. When placing a call on hold, your IP PBX might play a melody to the other party while waiting.

To place a call on hold:

1. Press the **Hold** soft key during a call.

The phone LCD screen shows the call is on hold and the line key LED flashes green.





Note

The phone will beep softly every 30 seconds to remind you that you still have a call on hold.

To resume a held call:

1. Press the **Resume** soft key.

Multiple Calls on Hold:

If multiple calls are placed on hold, press  or  to switch between the calls, and then press the **Resume** soft key to retrieve the desired call.

If more than two calls are placed on hold, an indication appears on the LCD screen, for example "2/3", indicating that this is the second call out of three calls.

If multiple calls are placed on hold on more than one line key, you can view the details of the calls by pressing the corresponding line key, and then press the **Resume** soft key to retrieve the call.

Do Not Disturb (DND)

You can use the DND feature to reject incoming calls automatically on the phone and the callers can hear a busy signal or a message depending on your server.

You can enable/disable the DND feature for the phone system, or you can customize the DND feature for each account or all accounts. The following describes the DND modes:

- **Phone** (default): DND in phone mode means that the DND feature is effective for the phone system.
- **Custom**: DND in custom mode means that you can configure the DND feature for each account or all accounts.

You can also configure the phone to receive incoming calls from some authorized numbers when the DND feature is enabled.

To configure the DND mode via web user interface:

1. Click on **Features->Forward & DND**.
2. In the **DND** block, mark the desired radio box in the **Mode** field.

The screenshot shows the 'Forward & DND' configuration page in the Yealink T426 web interface. The page has a green header with the Yealink logo and 'T426' on the left, and 'Log Out' on the right. Below the header are navigation tabs: Status, Account, Network, DSSKey, Features (selected), Settings, Directory, and Security. On the left side, there is a sidebar menu with options: Forward & DND (selected), General Information, Audio, Intercom, Transfer, Call Pickup, Remote Control, ACD, and Action URL. The main content area is divided into two sections: 'Forward' and 'DND'. The 'Forward' section has a 'Mode' field with radio buttons for 'Phone' (selected) and 'Custom', an 'Account' dropdown menu set to '1006', and an 'Always Forward' section with radio buttons for 'On' and 'Off' (selected). Below this are three input fields for 'Target', 'On Code', and 'Off Code'. The 'DND' section has a 'DND Emergency' dropdown menu set to 'Disabled', a 'DND Authorized Numbers' input field, a 'Mode' field with radio buttons for 'Phone' (selected) and 'Custom', an 'Account' dropdown menu set to '1006', a 'DND Status' section with radio buttons for 'On' and 'Off', and two input fields for 'DND On Code' and 'DND Off Code'. At the bottom of the page are 'Confirm' and 'Cancel' buttons. On the right side, there is a 'NOTE' box with the text 'Features Forward And DND'.

3. Click **Confirm** to accept the change.

Note

You can configure the DND mode via web user interface only.

To activate DND in phone mode:

1. Press the **DND** soft key when the phone is idle.

The **DND** icon on the idle screen indicates that the DND feature is enabled.

Incoming calls will be rejected automatically and "**n Missed Call(s)**" ("n" indicates the number of the missed calls) will prompt on the LCD screen.



To activate DND in custom mode for a specific account:

1. Press the **DND** soft key when the phone is idle.

The LCD screen displays a list of accounts registered on the phone.



2. Press \uparrow or \downarrow to select the desired account and then press the **Enter** soft key.
3. Press \leftarrow or \rightarrow , or the **Switch** soft key to select **Enable** from the **DND Enables** field.
4. Press the **Save** soft key to accept the change.

The **DND** icon appears on the idle screen and the associated line icon changes to \ominus .

Incoming calls on the specific line will be rejected automatically, and the phone LCD screen prompts "**n Missed Call(s)**" ("n" indicates the number of the missed calls).



To activate DND in custom mode for all accounts:

1. Press the **DND** soft key when the phone is idle.

The LCD screen displays a list of accounts registered on the phone.



2. Press the **All On** soft key to activate DND for all accounts.
3. Press the **Save** soft key to accept the change.

The **DND** icon appears on the idle screen and all line icons change to \ominus .

Incoming calls will be rejected automatically and "**n Missed Call(s)**" ("n" indicates the number of the missed calls) will be prompted on the LCD screen.



Note

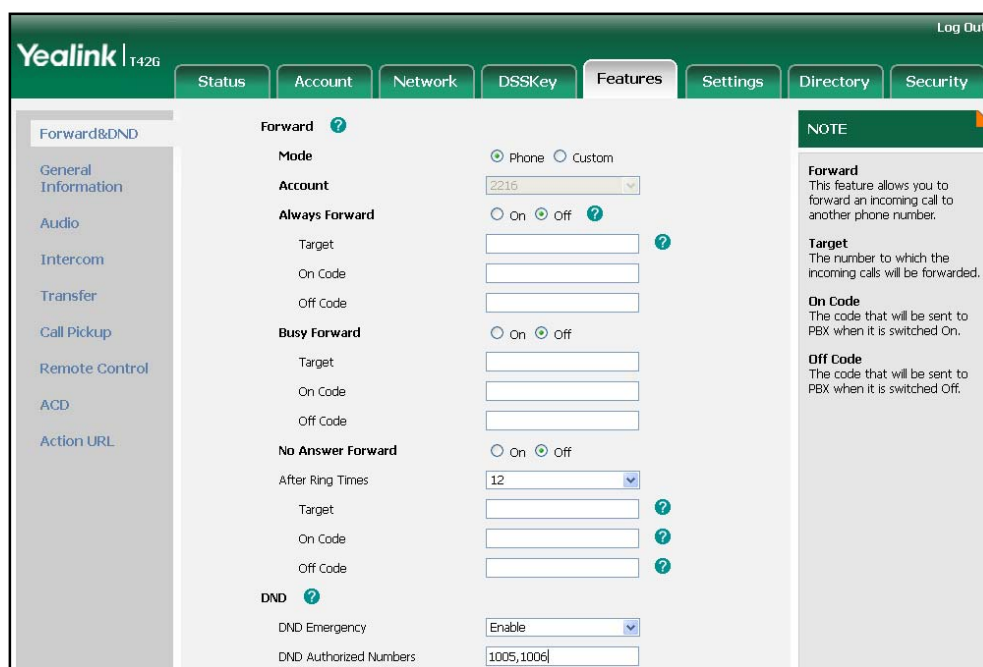
The prompt message displays only if the Missed Call Log feature for the line is enabled. The missed call log feature can be configured via web user interface at the path **Account->Basic**.

The Do Not Disturb feature is local to the phone, and may be overridden by the server settings. For more information, contact your system administrator.

To configure the DND authorized numbers via web user interface:

1. Click on **Features->Forward & DND**.
2. Select **Enabled** from the pull-down list of **DND Emergency**.
3. Enter the numbers in the **DND Authorized Numbers** field.

For multiple numbers, enter a comma between each number.



4. Click **Confirm** to accept the change.

When the DND feature is enabled on the phone, the phone can still receive incoming calls from the numbers specified in the **DND Authorized Numbers** field.

Call Forward

You can configure your phone to forward incoming calls to another party by the static forwarding. You can also forward calls while your phone is ringing, refer to the dynamic forwarding.

Static Forwarding

You can configure three types of call forward:

- **Always Forward:** Incoming calls are immediately forwarded.
- **Busy Forward:** Incoming calls are immediately forwarded if the phone is busy.
- **No Answer Forward:** Incoming calls are forwarded if not answered after a period of time.

You can enable/disable the call forward feature for the phone system, or you can customize the call forward feature for each account or all accounts. The following describes the call forward modes:

- **Phone:** Call forward in phone mode means that the call forward feature is effective for the phone system.
- **Custom:** Call forward in custom mode means that you can configure the call forward feature for each account or all accounts.

To configure the call forward mode via web user interface:

1. Click on **Features->Forward & DND**.
2. In the **Forward** block, mark the desired radio box in the **Mode** field.

The screenshot shows the 'Forward & DND' configuration page in the Yealink web user interface. The page is titled 'Forward' and has a 'NOTE' section on the right. The configuration options include:

- Mode:** Radio buttons for 'Phone' (selected) and 'Custom'.
- Account:** A dropdown menu showing '2216'.
- Always Forward:** Radio buttons for 'On' and 'Off' (selected). Below are 'Target', 'On Code', and 'Off Code' input fields.
- Busy Forward:** Radio buttons for 'On' and 'Off' (selected). Below are 'Target', 'On Code', and 'Off Code' input fields.
- No Answer Forward:** Radio buttons for 'On' and 'Off' (selected). Below are 'After Ring Times' (set to 12), 'Target', 'On Code', and 'Off Code' input fields.

The **NOTE** section on the right contains the following text:

Forward
This feature allows you 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.

3. Click **Confirm** to accept the change.



Note

You can configure the call forward mode via web user interface only.

To enable call forward in phone mode:

1. Press **Menu->Features->Call Forward**.
2. Press  or  to select the desired forwarding type, and then press the **Enter** soft key.
3. Depending on your selection:

a.) If you select **Always Forward**:

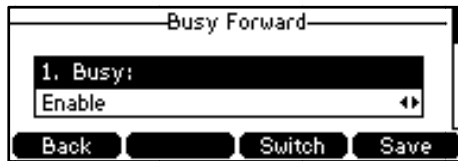
- 1) Press  or , or the **Switch** soft key to select **Enable** from the **Always** field.



- 2) Enter the destination number you want to forward all incoming calls to in the **Forward To** field.
- 3) (Optional.) Enter the always forward on code or off code respectively in the **On Code** or **Off Code** field.

b.) If you select **Busy Forward**:

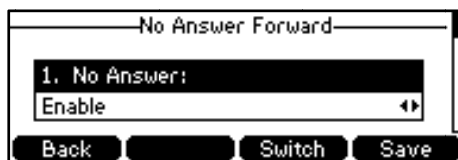
- 1) Press  or , or the **Switch** soft key to select **Enable** from the **Busy** field.





- 2) Enter the destination number you want to forward all incoming calls to when the phone is busy in the **Forward To** field.
- 3) (Optional.) Enter the busy forward on code or off code respectively in the **On Code** or **Off Code** field.


c.) If you select **No Answer Forward**:

- 1) Press  or , or the **Switch** soft key to select **Enable** from the **No Answer** field.









- 2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward To** field.
- 3) Press  or , or the **Switch** soft key to select the ring time to wait before forwarding in the **After Ring Times** field.

The default value is 12.

- 4) (Optional.) Enter the no answer forward on code or off code respectively in the **On Code** or **Off Code** field.
4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.
The  icon on the idle screen indicates the call forward is enabled.



To enable call forward in custom mode:

1. Press **Menu->Features->Call Forward**.
2. Press  or  to select the desired account, and then press the **Enter** soft key.
3. Press  or  to select the desired forwarding type, and then press the **Enter** soft key.
4. Depending on your selection:
 - a.) If you select **Always Forward**, you can enable it for a specific account.
 - 1) Press  or , or the **Switch** soft key to select **Enable** from the **Always** field.

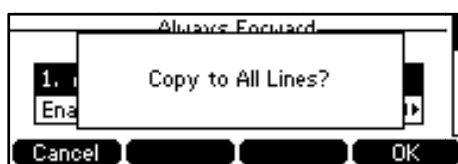


- 2) Enter the destination number you want to forward all incoming calls to in the **Forward To** field.
- 3) (Optional.) Enter the always forward on code or off code respectively in the **On Code** or **Off Code** field.



You can also enable the always forward for all accounts. After the always forward was enabled for a specific account, do as below:

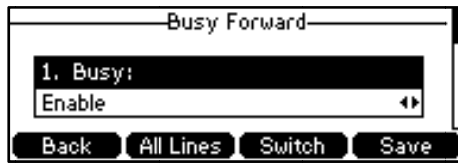
- 1) Press  or  to highlight the **Always** field.
- 2) Press the **All Lines** soft key.

The phone LCD screen prompts "Copy to All Lines?".





- 3) Press the **OK** soft key to accept the change or the **Cancel** soft key to cancel.
- b.) If you select **Busy Forward**, you can enable it for a specific account.

- 1) Press  or , or the **Switch** soft key to select **Enable** from the **Busy** field.

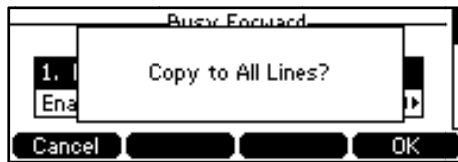




- 2) Enter the destination number you want to forward incoming calls to when the phone is busy in the **Forward To** field.
- 3) (Optional.) Enter the busy forward on code or off code respectively in the **On Code** or **Off Code** field.

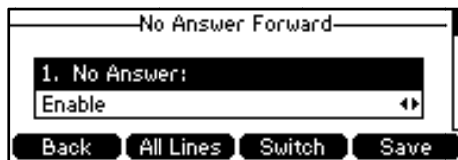
You can also enable the busy forward for all accounts. After the busy forward was enabled for a specific account, do as below:



- 1) Press  or  to highlight the **Busy** field.
- 2) Press the **All Lines** soft key.

The phone LCD screen prompts "Copy to All Lines?".





- 3) Press the **OK** soft key to accept the change or the **Cancel** soft key to cancel.
- c.) If you select **No Answer Forward**, you can enable it for a specific account.
- 1) Press  or , or the **Switch** soft key to select **Enable** from the **No Answer** field.

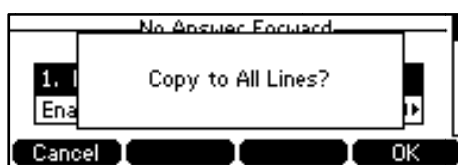


- 2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward To** field.
- 3) Press  or , or the **Switch** soft key to select the ring time to wait before forwarding in the **After Ring Times** field.
The default value is 12.
- 4) (Optional.) Enter the no answer forward on code or off code respectively in the **On Code** or **Off Code** field.

You can also enable the no answer forward for all accounts. After the no answer forward was enabled for a specific account, do as below:



- 1) Press  or  to highlight the **No Answer** field.
- 2) Press the **All Lines** soft key. The phone LCD screen prompts "Copy to All

Lines?".



- 3) Press the **OK** soft key to accept the change or the **Cancel** soft key to cancel.

5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

The  icon appears on the idle screen, and the associated line icon changes to .





You can also configure forward feature via web user interface at the path **Features->Forward & DND**.

Note







You can also enter the SIP URL or IP address in the **Forward To** field. For more information on using the SIP URL or IP address, refer to [Placing Calls](#) on page 61.

Call forward feature is local to the phone, and may be overridden by the server settings. Call forward on code or off code may be different between servers. For more information, contact your system administrator.

To disable call forward in phone mode:

1. Press **Menu->Features->Call Forward**.
2. Press  or  to select the desired forwarding type, then press the **Enter** soft key.
3. Press  or , or the **Switch** soft key to select **Disable** to disable the call forward.
4. Press the **Save** soft key to accept the change.

To disable call forward in custom mode for a specific account:

1. Press **Menu->Features->Call Forward**.
2. Press  or  to select the desired account, then press the **Enter** soft key.
3. Press  or  to select the desired forwarding type, then press the **Enter** soft key.
4. Press  or , or the **Switch** soft key to select **Disable** to disable the call forward.
5. Press the **Save** soft key to accept the change.

Dynamic Forwarding

To forward an incoming call to another party:

1. When the phone is ringing, press the **Fwd** soft key.

- Enter the number you want to forward the incoming call to.



- Press , , or the **Send** soft key.

The phone LCD screen prompts a call forward message.

Call Transfer

You can transfer a call to another party in one of the three ways:

- **Blind Transfer:** Transfer a call directly to another party without consulting.
- **Semi-Attended Transfer:** Transfer a call when the target phone is ringing.
- **Attended Transfer:** Transfer a call with prior consulting.

To perform a blind transfer:



- Press the **Tran** soft key during a call.
- Enter the number you want to transfer the call to.





- Press the **Tran** soft key to complete the transfer.

Then the call is connected to the number to which you are transferring.

To perform a semi-attended transfer:

- Press the **Tran** soft key during a call.
- Enter the number you want to transfer the call to.
- Press  or  to dial out.
- Press the **Tran** soft key to complete the transfer when receiving ringback.

To perform an attended transfer:

- Press the **Tran** soft key during a call.
- Enter the number you want to transfer the call to.
- Press  or  to dial out.
- After the party answers the call, press the **Tran** soft key to complete the transfer.



If you are using a handset, the transfer can be completed by hanging up the handset.

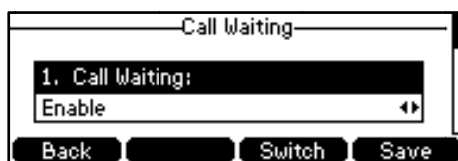
You can cancel the transfer before the call is connected by pressing the **Cancel** soft key.



Call Waiting

You can enable or disable the call waiting feature on the phone. If the call waiting feature is enabled, you can receive another call when there is an active call on the phone. Otherwise, another incoming call is automatically rejected by the phone with a busy message when there is an active call on the phone. You can also enable or disable the phone to play a warning tone when receiving another call.

To configure call waiting via phone user interface:

1. Press **Menu->Features->Call Waiting**.
2. Press  or , or the **Switch** soft key to select **Enable** from the **Call Waiting** field.



3. Press  or , or the **Switch** soft key to select **Enable** from the **Play Tone** field.
4. (Optional.) Enter the call waiting on code or off code respectively in the **CW On Code** or **CW Off Code** field.
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure the call waiting feature via web user interface at the path **Features->General Information**.

Conference

You can create a conference with two parties using the phone's local conference feature. You can create a conference between an active call and a call on hold by pressing the **Conf** soft key. The SIP-T41P IP phone also supports the network conference feature.

Note

The network conference feature is not available on all servers. For more information, contact your system administrator.



Local Conference

The SIP-T41P IP phone supports up to 3 parties (including yourself) in a conference call. This is called Local Conference and is the default method of conference.

To set up a local conference call:



1. Place a call to the first party.
2. When the first party answers the call, press the **Conf** soft key to place a new call.

The active call is placed on hold.




3. Enter the number of the second party and press , , or the **Send** soft key.
4. When the second party answers the call, press the **Conf** soft key again to join all parties in the conference.



To join two calls in a conference:

1. Place two calls using two different accounts on the phone (for example, place the first call using account 1, and then place the second call using account 2).
2. Press  or  to select the call for conference and make sure the call is active (for example, select the call on account 1).
3. Press the **Conf** soft key to join the two calls in the conference on account 1.

During the conference call, you can do the following:

- Press the **Hold** soft key to place the conference on hold.
- Press the **Split** soft key to split the conference call into some individual calls.
- Press the **Manage** soft key, and then press  or  to select the desired party:
 - Press the **Far Mute** soft key to forbid the party speaking.
 - Press the **Remove** soft key to remove the party from the conference call.
 - Press the **Back** soft key to return to the previous interface.
- Press  to mute the conference call.
- Press the **Cancel** soft key to drop the conference call.

Network Conference

You can use network conference feature on the SIP-T41P IP phone to conduct a conference with multiple participants.

This feature allows you to perform the following:

- Join two calls together into a conference call.
- Invite another party into an active conference call.
- Remove a specific conference party.

To use this feature, contact your system administrator for the network conference-URI in advance, and configure the network conference via web user interface.

To configure the network conference via web user interface:

1. Click on **Account->Advanced**.
2. Select **Network Conference** from the pull-down list of **Conference Type**.
3. Enter the conference URI (e.g., conference@example.com) in the **Conference URI** field.

The screenshot shows the Yealink T42G web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' section is active, showing 'Account 1'. The 'Advanced' tab is selected. The configuration table is as follows:

Parameter	Value
Keep Alive Type	Default
Keep Alive Interval (Seconds)	30
Local SIP Port	5062
Conference Type	Network Conference
Conference URI	conference@example.com
ACD Subscrip Period(120~3600s)	3600
Early Media	Disabled
SIP Server Type	Default
Music Server URI	
Directed Call Pickup Code	
Group Call Pickup Code	
Distinctive Ring Tones	Disabled
Unregister When Reboot	Disabled
Out Dialog BLF	Disabled

Buttons for 'Confirm' and 'Cancel' are at the bottom. A 'NOTE' box on the right says: 'Advanced: The Advanced parameters for administrator.'

4. Click **Confirm** to accept the change.

To set up a network conference call:

1. Place a call to the first party.
2. Press the **Conf** soft key to create a new call.
The active call is placed on hold.
3. Enter the number of the second party and press **OK**, **# SEND**, or the **Send** soft key.
4. When the second party answers the call, press the **Conf** soft key to add the second party to the conference.
5. Press the **Conf** soft key to create a new call.
The conference is placed on hold.
6. Enter the number of the new party and then press **OK**, **# SEND**, or the **Send** soft key.
7. When the new party answers the call, press the **Conf** soft key to add the new party to the conference.
8. Repeat steps 5 to 7 until you have added all intended parties.

The procedures to set up a network conference call for specific servers may be different from introduced above. Contact your system administrator for more information.





Call Park

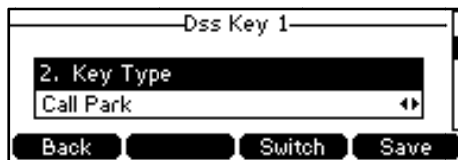
You can use call park feature to place a call on hold, and then retrieve the call from another phone in the system (for example, a phone in another office or conference room). You can park an active call by pressing the call park key on the phone. If the call is parked successfully, the response is either a voice prompt confirming that the call was parked, or a visible indication on the LCD screen.



Note

Call park feature is not available on all servers. Contact your system administrator for more information.

To configure a call park key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**.
2. Select the desired DSS key.
3. Press  or , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press  or , or the **Switch** soft key to select **Call Park** from the **Key Type** field.



5. Press  or , or the **Switch** soft key to select the desired line from the **Account ID** field.
6. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
7. Enter the call park feature code in the **Value** field.
8. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure a call park key via web user interface at the path **DSSKey->Line Key**.

To use the call park feature:

1. User on phone A places a call to phone B.
2. User on phone A wants to take the call in a conference room for privacy, then presses the call park key on phone A.
3. User on phone A walks to an available conference room where the phone is designated as phone C. The user dials the call park retrieve feature code to retrieve the parked call.

The system establishes call between phone C and B.

Note

The call park feature code and call park retrieve feature code are predefined on the system server. Contact your system administrator for more information.

If the parked call is not retrieved within a period of time assigned by the system, the phone performing call park feature will receive call back.

Call Pickup

You can use call pickup feature to answer someone else's incoming call on the phone. The SIP-T41P IP phone supports the directed call pickup and group call pickup features. Directed call pickup is used for picking up a call that is ringing at a specific phone number. Group call pickup is used for picking up a call that is ringing at any phone number in the group. The pickup group should be predefined, contact your system administrator for more information.

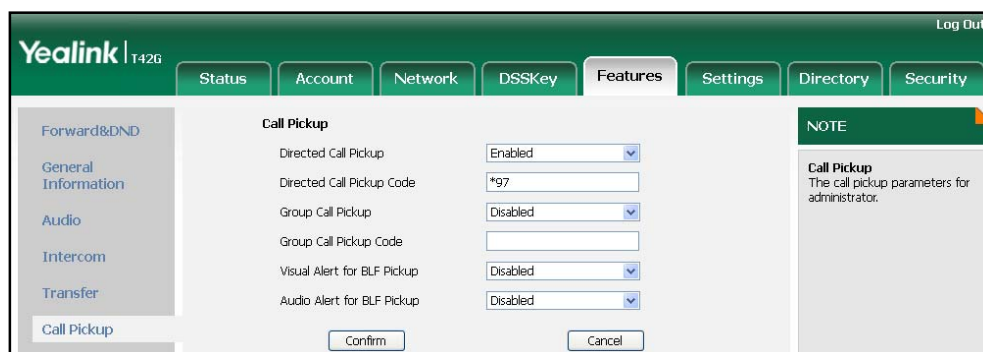
You can pick up an incoming call by using the **DPickup/GPickup** soft key. To use the call pickup feature, you need to configure the call pickup code beforehand on a global or per-line basis via web user interface.

Note If there are many incoming calls at the same time, pressing the **GPickup** soft key on the phone will pick up the call that rings first.

Directed Call Pickup

To enable directed call pickup and configure the directed call pickup code on a global basis via web user interface:

1. Click on **Features->Call Pickup**.
2. Select **Enabled** from the pull-down list of **Directed Call Pickup**.
3. Enter the directed call pickup code in the **Directed Call Pickup Code** field.

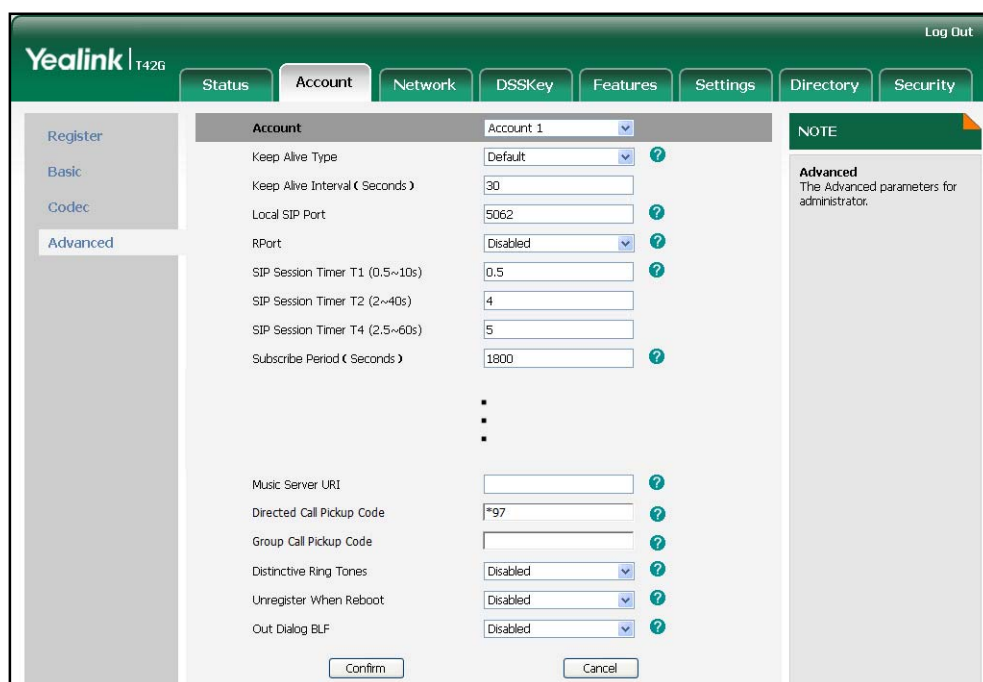


4. Click **Confirm** to accept the change.

To configure the directed call pickup code on a per-line basis via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.

- Enter the directed call pickup code in the **Directed Call Pickup Code** field.

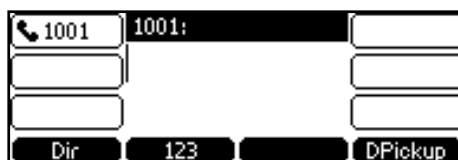


- Click **Confirm** to accept the change.

To pick up a call directly:

- Pick up the handset.

The **DPickup** soft key appears on the LCD screen.



- Press the **DPickup** soft key on your phone when a phone receives an incoming call.
- Enter the phone number which receives an incoming call.
- Press the **DPickup** soft key again.

The call is answered on your phone.

You can also configure a DSS key as a directed pickup key via phone user interface or web user interface. You can pick up a call by pressing the directed pickup key directly.

Group Call Pickup

To enable the group call pickup and configure the group call pickup code on a global basis via web user interface:

- Click on **Features->Call Pickup**.
- Select **Enabled** from the pull-down list of **Group Call Pickup**.

- Enter the group call pickup code in the **Group Call Pickup Code** field.

The screenshot shows the 'Call Pickup' configuration page in the Yealink T42G web interface. The 'Features' tab is active. The configuration includes:

- Directed Call Pickup: Disabled
- Directed Call Pickup Code: (empty field)
- Group Call Pickup: Enabled
- Group Call Pickup Code: *98
- Visual Alert for BLF Pickup: Disabled
- Audio Alert for BLF Pickup: Disabled

Buttons for 'Confirm' and 'Cancel' are visible at the bottom. A 'NOTE' box on the right states: 'Call Pickup: The call pickup parameters for administrator.'

- Click **Confirm** to accept the change.

To configure the group call pickup code on a per-line basis via web user interface:

- Click on **Account**.
- Select the desired account from the pull-down list of **Account**.
- Click on **Advanced**.
- Enter the group call pickup code in the **Group Call Pickup Code** field.

The screenshot shows the 'Account' configuration page in the Yealink T42G web interface. The 'Account' dropdown is set to 'Account 1'. The 'Advanced' tab is active. The configuration includes:

- Keep Alive Type: Default
- Keep Alive Interval (Seconds): 30
- Local SIP Port: 5062
- RPort: Disabled
- SIP Session Timer T1 (0.5~10s): 0.5
- SIP Session Timer T2 (2~40s): 4
- SIP Session Timer T4 (2.5~60s): 5
- Subscribe Period (Seconds): 1800
- Music Server URI: (empty field)
- Directed Call Pickup Code: (empty field)
- Group Call Pickup Code: *98
- Distinctive Ring Tones: Disabled
- Unregister When Reboot: Disabled
- Out Dialog BLF: Disabled

Buttons for 'Confirm' and 'Cancel' are visible at the bottom. A 'NOTE' box on the right states: 'Advanced: The Advanced parameters for administrator.'

- Click **Confirm** to accept the change.

To pick up a call in the group:

- Pick up the handset.

The **GPickup** soft key appears on the LCD screen.



2. Press the **GPickup** soft key on your phone when a phone in the group receives an incoming call.

The call is answered on your phone.

You can also configure a DSS key as group pickup key via phone user interface or web user interface. You can pick up a call by pressing the group pickup key directly.

Note

The directed call pickup code and group call pickup code are predefined on the system server. Contact your system administrator for more information.

The call pickup code configured on a per-line basis takes precedence over that configured on a global basis.

Anonymous Call

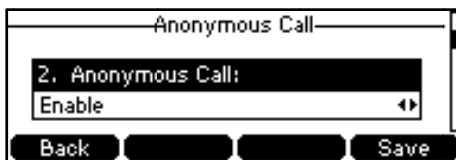
You can use anonymous call feature to block the identity and phone number from showing up to the called party when you call someone. For example, you want to call to consult some of the services, but you don't want to be harassed. Anonymous call is configurable on a per-line basis.

Note

The anonymous call feature is not available on all servers. Contact your system administrator for the anonymous call on code and off code.

To configure anonymous call via phone user interface:

1. Press **Menu->Features->Anonymous Call**.
2. Select the desired line.
3. Press **◀** or **▶**, or the **Switch** soft key to select **Enable** from the **Anonymous Call** field.



4. (Optional.) Enter the anonymous call on code in the **Call On Code** field.
5. (Optional.) Enter the anonymous call off code in the **Call Off Code** field.
6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure anonymous call via web user interface at the path

Account->Basic.

To place an anonymous call:

1. Using the specific line on the phone to place a call to phone B.





The LCD screen of phone B prompts an incoming call from anonymity.



Anonymous Call Rejection

You can use anonymous call rejection feature to reject incoming calls from anonymous callers. Anonymous call rejection automatically rejects incoming calls from callers who deliberately block their identities and numbers from showing up. Anonymous call rejection is configurable on a per-line basis.

To configure anonymous call rejection via phone user interface:

1. Press **Menu->Features->Anonymous Call**.
2. Select the desired line.
3. Press  or  to scroll to the **Rejection** field.
4. Press  or , or the **Switch** soft key to select **Enable** from the **Rejection** field.



5. (Optional.) Enter the anonymous call rejection on code in the **Reject On Code** field.
6. (Optional.) Enter the anonymous call rejection off code in the **Reject Off Code** field.
7. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure anonymous call rejection via web user interface at the path **Account->Basic**.

Advanced Phone Features

This chapter provides operating instructions for the advanced features of the SIP-T41P IP phone. The topics include:

- [Busy Lamp Field \(BLF\)](#)
- [Call Recording](#)
- [Hot Desking](#)
- [Intercom](#)
- [Multicast Paging](#)
- [Music on Hold](#)
- [Automatic Call Distribution \(ACD\)](#)
- [Messages](#)

If you require additional information or assistance with your new phone, contact your system administrator.

Busy Lamp Field (BLF)



You can use the BLF feature to monitor a specific user for status changes on the phone. For example, you can configure a BLF key on the phone for monitoring the status of a friend's phone (busy or idle). When the friend makes a call, the BLF key (line key) LED illuminates solid red on your phone to indicate that the friend's phone is in use and busy. For more BLF key LED indications, refer to [LED Instructions](#) on page 5.

You can press a BLF key to dial out the monitored phone number when the monitored phone is idle. You can receive a visual alert and an audio alert when the monitored phone number receives an incoming call. You can also pick up a call directly by pressing the BLF key when the monitored phone number receives an incoming call. For more information, contact your system administrator.

To configure a BLF key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press **◀** or **▶**, or the **Switch** soft key to select **BLF** from the **Type** field.



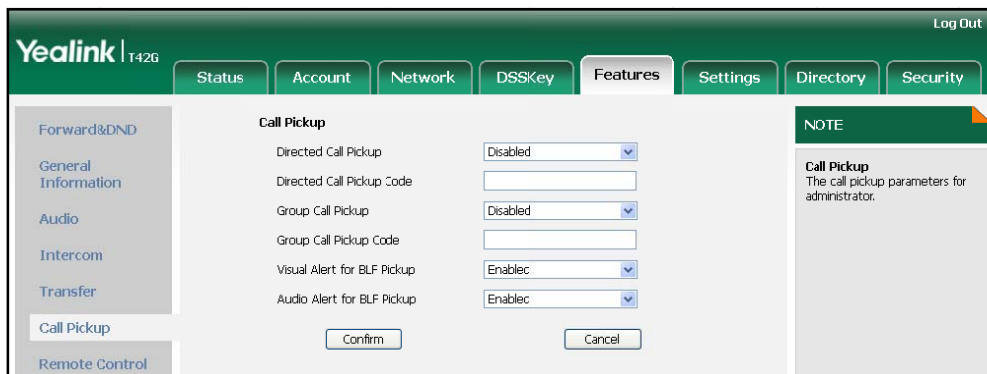
4. Press  or , or the **Switch** soft key to select the desired line from the **Account ID** field.
5. (Optional.) Enter a string that will appear on the LCD screen in the **Label** field.
6. Enter the phone number or extension you want to monitor in the **Value** field.
7. (Optional.) Enter the pickup code in the **Extension** field.
8. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure the BLF key via web user interface at the path **DSSKey->Line Key**.

You can enable audio alert for BLF pickup feature on the phone. This allows the monitoring phone to play a warning tone, when the monitored phone number receives an incoming call. You can also enable visual alert for BLF pickup feature on the phone. This allows the LCD screen of the monitoring phone to display the caller ID, when the monitored phone number receives an incoming call.

To enable the visual and audio alert feature via web user interface:

1. Click on **Features->Call Pickup**.
2. Select **Enabled** from the pull-down list of **Visual Alert for BLF Pickup**.
3. Select **Enabled** from the pull-down list of **Audio Alert for BLF Pickup**.



4. Click **Confirm** to accept the change.

When the monitored phone number receives an incoming call, the following occurs on the monitoring phone:

- The phone plays a warning tone (if enabled).
- The BLF key LED flashes red.
- The caller ID appears on the LCD screen (if enabled).

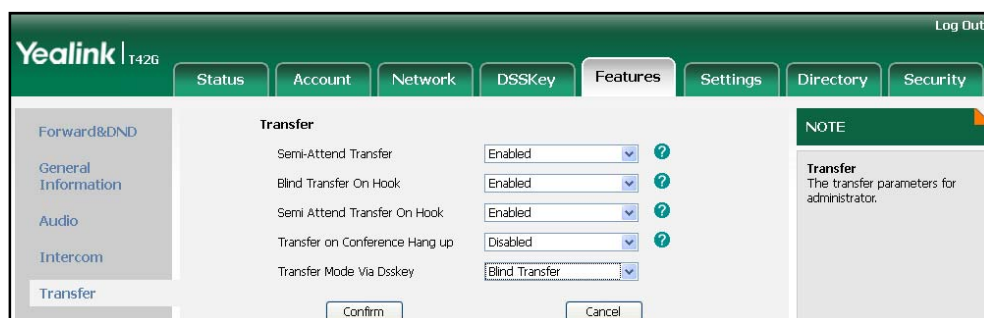
In the following figure, the phone LCD screen shows an incoming call from 1001 on the monitored line.



When there is an active call on the IP phone, you can transfer the active call to the monitored phone number directly by pressing the BLF key. The phone transfers the active call differently depending on the transfer mode on DSS key. For more information on performing call transfer, refer to [Call Transfer](#) on page 78.

To configure the transfer mode on DSS key via web user interface:

1. Click on **Features->Transfer**.
2. Select the desired transfer mode from the pull-down list of **Transfer Mode Via Dsskey**. Depending on your selection:
 - If you select **Blind Transfer**, press the BLF key to blind transfer to the monitored phone number.
 - If you select **Attended Transfer**, press the BLF key to dial out the monitored phone number and then you can perform the attended or semi-attended transfer.
 - If you select **New Call**, press the BLF key to place a new call to the monitored user.



3. Click **Confirm** to accept the change.

Note

You can configure the transfer mode on DSS key via web user interface only.

Call Recording





You can record calls by pressing a record key on the SIP-T41P IP phone. There are 2 different ways of configuring call recording and they even work differently:

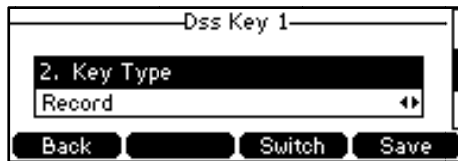
- **Record:** The phone sends SIP INFO message containing a specific header "Record: on/off" to trigger a recording.
- **URL Record:** The phone sends HTTP URL request to trigger a recording. Contact your system administrator for the predefined URL.

Note

The record feature is not available on all servers. Contact your system administrator for more information.



To configure a Record key via phone user interface:

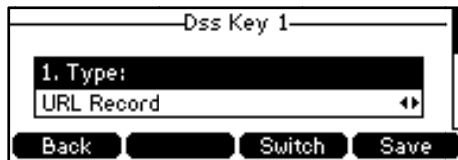
1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press  or , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press  or , or the **Switch** soft key to select **Record** from the **Key Type** field.



5. (Optional.) Enter a string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

To configure a URL Record key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press  or , or the **Switch** soft key to select **URL Record** from the **Type** field.



4. (Optional.) Enter a string that will appear on the LCD screen in the **Label** field.
5. Enter the URL (e.g., <http://10.1.2.224/phonerecording.cgi>) in the **Value** field.
6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure the Record or URL Record keys via web user interface at the path **DSSKey->Line Key**.

The Record/URL Record keys control the recording and are available:

- During an active call
- When calls are on hold or mute
- During a blind or attended transfer
- During a conference call
- When the phone prompts you to answer an incoming call






The Record/URL Record keys are not available:

- When there are no connected calls on your phone
- When you place a new call

To record a call:

1. Press the Record or URL Record key during a call.
If the recording starts successfully, the recording icon appears on the LCD screen and the Record or URL Record key LED flashes green.
2. Press the Record or URL Record key again to stop recording.
The recording icon disappears from the LCD screen and the Record or URL Record key LED turns off.

Recording status indications you need to know:

Circumstance	Icons on the LCD screen
A recording is started	 appears on the LCD screen
A recording cannot be started	 appears for 1 second
A recording cannot be	 appears for 1 second, then goes back
The recording box is full	 appears for 1 second
The cannot be recorded	 appears for 1 second

You can listen to the recordings stored on your server system. For example, you can dial an access code to listen to the recordings.

Note

The way in which you listen to the recordings may be different on different servers. Contact your system administrator for more information.

Hot Desking

Hot Desking originates from the definition of being the temporary physical occupant of a work station or surface by a particular employee. A primary motivation for Hot Desking is cost reduction. This feature is regularly used in places where not all the employees are in the office at the same time, or not in the office for very long at all,

which means actual personal offices would be often vacant, consuming valuable space and resources.

You can use Hot Desking on the SIP-T41P IP phone to log out the existing accounts and then log in a new account. As a result, many users can share the phone resource in different time. To use this feature, you need to configure a Hot Desking key in advance.

Note

The hot desking feature is not available on all servers. Contact your system administrator for more information.

To configure a hot desking key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press **◀** or **▶**, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press **◀** or **▶**, or the **Switch** soft key to select **Hot Desking** from the **Key Type** field.
5. (Optional.) Enter a string that will appear on the LCD screen in the **Label** field.



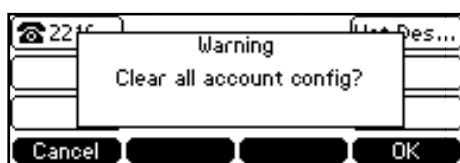
6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure a hot desking key via web user interface at the path **DSSKey->Line Key**.

To use hot desking:

1. Press the hot desking key when the phone is idle.

The phone LCD screen prompts the following warning:



2. Press the **OK** soft key, registration configurations of all accounts on the phone will be cleared immediately.

The login wizard will be shown as below:



3. Enter the login information in each field.

Intercom



Intercom is a useful feature in an office environment to quickly connect with the operator or the secretary. You can press the intercom key to automatically connect with a remote extension for outgoing intercom calls, and the remote extension will automatically answer the incoming intercom calls.

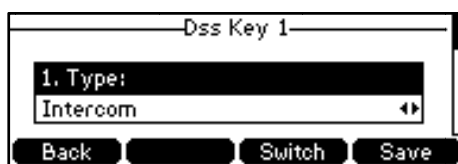
Note

The intercom feature is not available on all servers. Contact your system administrator for more information.

Outgoing Intercom Calls

To configure an intercom key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press  or , or the **Switch** soft key to select **Intercom** from the **Type** field.



4. Select the desired line from the **Account ID** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the remote extension number in the **Value** field.
7. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure an intercom key via web user interface at the path **DSSKey->Line Key**.

To place an intercom call:

1. Press the intercom key when the phone is idle.
The phone is automatically connected to the extension specified in the **Value** field.
2. Press the intercom key again or the **Cancel** soft key to end the intercom call.

Incoming Intercom Calls

By default, the SIP-T41P IP phone supports answering an incoming intercom call automatically. The phone automatically plays a warning tone when it receives an incoming intercom call. In addition, you can enable the phone to mute the microphone when it automatically answers an incoming intercom call. You can also

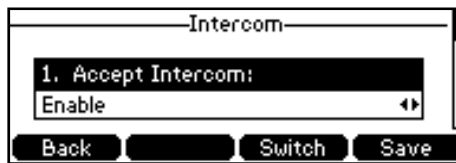
enable the phone to automatically answer an incoming intercom call while there is already an active call on the phone, the active call is placed on hold.

Intercom features you need to know:

Intercom features	Description
Accept Intercom	Allows you to enable or disable the IP phone to automatically answer an incoming intercom call.
Intercom Mute	Allows you to enable or disable the microphone on the IP phone for Intercom calls.
Warning Tone	Allows you to enable or disable the IP phone to play a warning tone when the IP phone receives an incoming intercom call.
Intercom Barge	Allows you to enable or disable the IP phone to automatically answer an incoming intercom call while there is already an active call on the phone.

To configure intercom features via phone user interface:

1. Press **Menu->Features->Intercom**.
2. Make the desired changes.



3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure these specific parameters via web user interface at the path **Features->Intercom**.

Accept Intercom

You can enable or disable the phone to automatically answer an incoming intercom call. If Accept Intercom is enabled, the phone automatically answers an incoming intercom call. If Accept Intercom is disabled, the phone rejects incoming intercom calls and sends a busy signal to the caller. Accept Intercom is enabled by default.

Note Your administrator can set a time period delay before the phone automatically answers. Contact your system administrator for more information.

Intercom Mute

You can mute or un-mute the microphone on the phone for intercom calls automatically. If Intercom Mute is enabled, the microphone is muted for intercom calls. If Intercom Mute is disabled, the microphone is un-muted for incoming calls. Intercom Mute is disabled by default.

Intercom Tone

You can enable or disable the phone to play a warning tone when receiving an intercom call. If Intercom Tone is enabled, the phone plays a warning tone to alert you before answering the intercom call. If Intercom Tone is disabled, the phone automatically answers the intercom call without warning. Intercom Tone is enabled by default.

Intercom Barge





You can enable or disable the phone to automatically answer an incoming intercom call while there is already an active call on the phone. If Intercom Barge is enabled, the phone automatically answers the intercom call and places the active call on hold. If Intercom Barge is disabled, the phone handles an incoming intercom call like a waiting call. Intercom Barge is disabled by default.

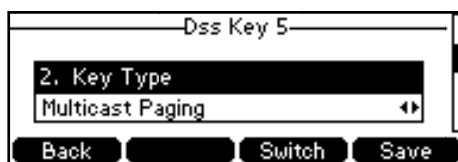
Multicast Paging

You can use multicast paging to quickly and easily forward time sensitive announcements out to people within the multicast group. You can configure a multicast paging key on the phone, which allows you to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address(es) without involving SIP signaling. You can configure the phone to allow it to receive an RTP stream from pre-configured multicast listening address(es) without involving SIP signaling. You can specify up to 10 multicast listening addresses.

Sending RTP Stream

To configure a multicast paging key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**.
2. Select the desired DSS key.
3. Press  or , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press  or , or the **Switch** soft key to select **Multicast Paging** from the **Key Type** field.



5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the multicast IP address and port number (e.g., 224.5.6.20:10008) in the **Value** field.

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

7. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

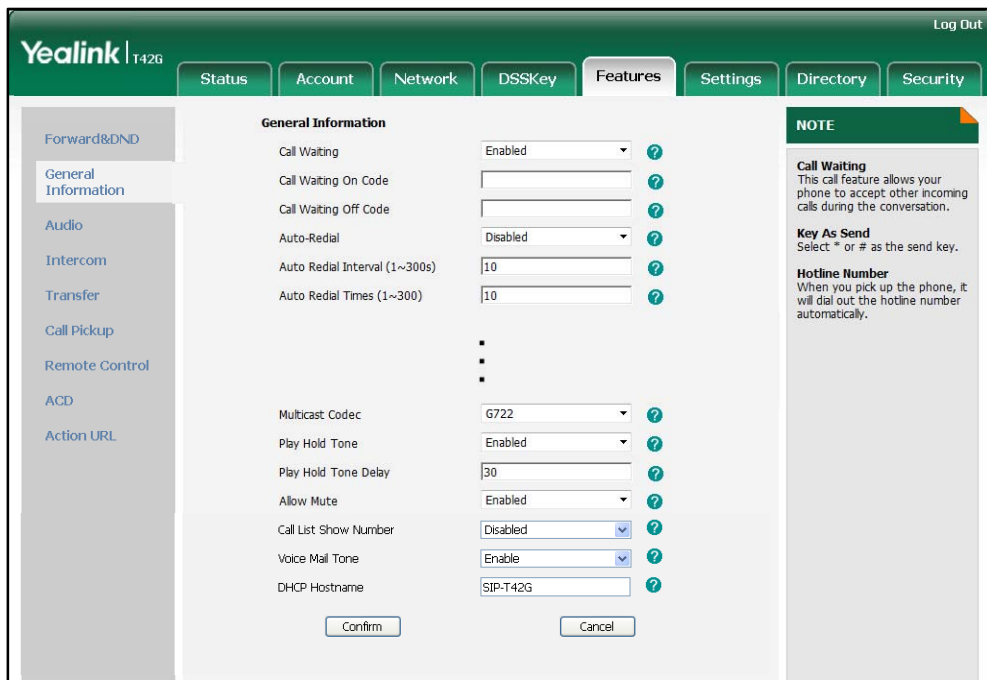
You can also configure a multicast paging key via web user interface at the path **DSSKey->Line Key**.

You can also configure the phone to use a default codec for sending multicast RTP stream via web user interface.

To configure a default codec for multicast paging via web user interface:

1. Click on **Features->General Information**.
2. Select the desired codec from the pull-down list of **Multicast Codec**.

The default codec is G722.



3. Click **Confirm** to accept the change.

Note If G722 codec is used for multicast paging, the phone LCD screen displays the **HD** icon to indicate that is providing high definition voice.

You can configure a default codec for multicast paging via web user interface only.

To send RTP stream:

1. Press the multicast paging key when the phone is idle.
- The phone sends RTP to a preconfigured multicast address (IP: Port). Any phone in the local network then listens to the RTP on the preconfigured multicast address (IP: Port). For both sending and receiving of the multicast RTP there is no sip signaling involved.
- The multicast paging key LED illuminates solid green.

The following figure shows a multicast RTP session on the phone:



2. Press the **Hold** soft key to place the current multicast RTP session on hold.
3. Press the **Cancel** soft key to cancel the multicast RTP session.

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 placed on hold.

Receiving RTP Stream

You can configure the phone to receive a Real Time Transport Protocol (RTP) stream from the pre-configured multicast address(es) without involving SIP signaling. You can specify up to 10 multicast addresses that the phone listens to on the network.

You can also change the behavior of how the phone handles incoming multicast paging calls by configuring specific parameters via web user interface. The specific parameters are: Paging Barge and Paging Priority Active.

Paging Barge

You can use the paging barge feature to define the priority of the voice call in progress. If the priority of an incoming multicast paging call is lower than that of the active call, then it will be ignored automatically. If Disabled is selected from the pull-down list of Paging Barge, the voice call in progress shall take precedence over all incoming multicast paging calls. The valid values in the Paging Barge field are:

- **1 to 10:** Define the priority of the active call, 1 is the highest priority, 10 is the lowest priority.
- **Disabled:** The voice call in progress shall take precedence over all incoming paging calls.

Paging Priority Active

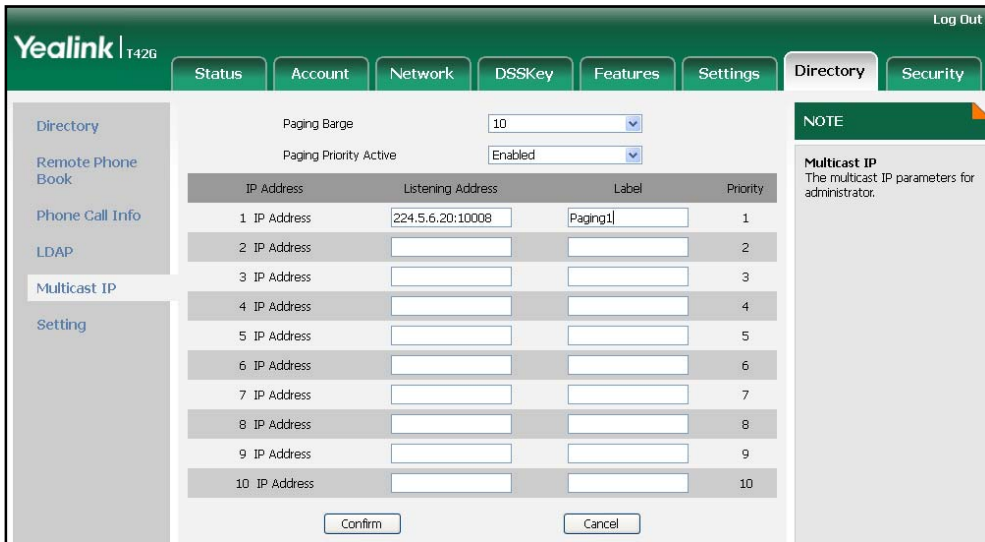
You can enable or disable this feature to decide how the phone handles the incoming multicast paging calls, when there is already a multicast paging call on the phone. If enabled, the phone will ignore the incoming multicast paging call with a lower priority, otherwise, the phone will play the incoming multicast RTP and place the previous multicast paging call on hold. If disabled, the phone will automatically ignore all incoming multicast paging calls.

To configure multicast listening addresses via web user interface:

1. Click on **Directory->Multicast IP**.

2. Select the desired value from the pull-down list of **Paging Barge**.
3. Select the desired value from the pull-down list of **Paging Priority Active**.
4. Enter the multicast IP address(es) and port number (e.g., 224.5.6.20:10008) which the phone listens for incoming RTP multicast in the **Listening Address** field.
5. Enter the label in the **Label** field.

Label will appear on the LCD screen when receiving the multicast RTP stream.



6. Click **Confirm** to accept the change.

Note

The multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

The priorities of listening addresses are predefined: 1 with the highest priority, 10 with the lowest priority.

Both the multicast paging sender and the receiver will play a warning tone when pressing the multicast paging key.

Music on Hold

Music on hold (MoH) is the business practice of playing recorded music to fill the silence that would be heard by the callers who have been placed on hold. To use this feature, you should specify a SIP URI pointing to a Music on Hold Server account, when placing a call on hold, the phone will invite this SIP URI to the Music on Hold Server account. The Music on Hold account automatically answers to the SIP INVITE messages and immediately plays audio from some source located anywhere (LAN, Internet) to the held party. Contact your system administrator for the SIP URI.

To configure music on hold server via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.

- Click on **Advanced**.
- Enter the SIP URI (e.g., sip:moh@sip.com) in the **Music Server URI** field.

The screenshot shows the Yealink T42G web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Music Server URI' field is populated with 'sip:moh@sip.com'. Other configuration options include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5062), 'RPort' (Disabled), 'SIP Session Timer T1 (0.5~10s)' (0.5), 'SIP Session Timer T2 (2~40s)' (4), 'SIP Session Timer T4 (2.5~60s)' (5), and 'Subscribe Period (Seconds)' (1800). The 'Music Server URI' field is highlighted in the instructions. A 'NOTE' box on the right states: 'Advanced: The Advanced parameters for administrator.'

- Click **Confirm** to accept the change.

When you have placed a call on hold, the held party can hear the music.

Note

All involved parties cannot use encrypted RTP (SRTP).

You can configure the music on hold server via web user interface only.

Automatic Call Distribution (ACD)

ACD is often used in offices for customer service, such as call center. The ACD system handles large volumes of incoming calls from callers who have no need to talk to a specific person but who require assistance from any of multiple persons at the earliest opportunity. The ACD feature on the SIP-T41P IP phone allows the ACD system to distribute large volumes of incoming calls to the registered IP phone users. To use this feature, you should configure an ACD key in advance.

Note

Make sure the ACD feature is enabled on your IP phone. For more information about enabling the ACD feature, contact your system administrator.

To configure an ACD key via phone user interface:

- Press **Menu->Features->DSS Keys**.
- Select the desired DSS key.

- Press  or , or the **Switch** soft key to select **ACD** from the **Type** field.



- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

You can also configure an ACD key via web user interface at the path **DSSKey->Line Key**.

To log in the ACD system:

- Press the ACD key when the phone is idle.

The phone LCD screen prompts you the following information:

User ID: the identity used to log in the queue.

Password: the password used to log in the queue.



- Press the **Login** soft key to log in.

Note

Contact your system administrator for the User ID and Password to access the ACD system.

After configuring an ACD key, you can press the ACD key to log in the ACD system. After logging in, you are ready to receive calls from the ACD system. You can press the ACD key to show your current phone status. You can press the **Available/Unavailable** soft key to change your phone status. The system server monitors your phone status. When you set the phone status to available, the ACD key LED illuminates solid green, and then the server begins distributing calls to your phone. When you set the phone status to unavailable, the ACD key LED flashes green, and then the server temporarily stops distributing calls to your phone. To log out the ACD system, press the **Logout** soft key.

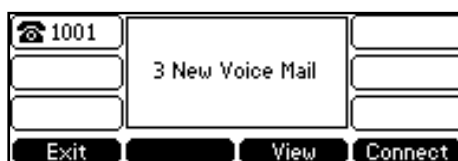
Note

It is recommended you configure no more than one ACD key per phone. At any time, at most one ACD key can be in progress on your phone.

Messages

Voice Mail

You can leave voice mails for someone else on the SIP-T41P IP phone. You can also listen to the voice mails stored in a centralized location. When receiving a new voice mail, the phone will play a warning tone, and the power indicator LED will slow flash red. The phone LCD screen will display a prompt message and appear an icon.



Note

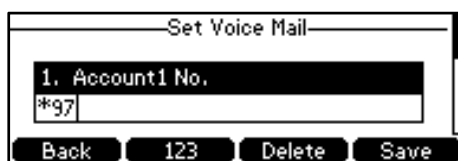
Voice mail feature is not available on all servers, contact your system administrator for more information.

To leave a voice mail:

You can leave a voice mail for someone else when he/she is busy or inconvenient to answer the call. Follow the voice prompt from the system server to leave the voice mail, and then hang up after completing the voice mail.

To configure voice mail access codes via phone user interface:

1. Press **Menu->Messages->Voice Mail->Set Voice Mail**.
2. Press the navigation keys to highlight the account which you want to set.
3. Press the **123** soft key to select the proper input mode and then enter the voice mail access code (e.g., *97).



4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Note

Voice mail access code must be predefined on the system server. Contact your system administrator for the more information.

To listen to the voice mails:

1. When the phone user interface prompts receiving new voice mails and the power indicator LED slow flashes red, press  or the **Connect** soft key to dial out the voice mail access code.

- Follow the voice prompt to listen to voice mails.

Note

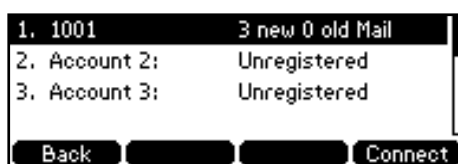
Before listening to the voice mails, make sure the voice mail access code has been configured.

When all new voice mails are retrieved, the power indicator LED will stop flashing.

To view the voice mail via phone user interface:

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

The phone LCD screen displays the amount of the voice mails that includes new or old voice mails.



- Select an account and then press the **Connect** soft key to listen to the voice mails.

Message Waiting Indicator (MWI)

The SIP-T41P IP phone supports MWI feature when receiving a new voice message. If someone leaves you a voice mail, you will receive a message waiting indicator. MWI will be indicated in three ways: a warning tone, the power indicator LED slow flashes red and an indicator message (including a voice mail icon) appears on the LCD screen. This is cleared only when you retrieve all voice mails or delete them.

For some particular servers, the MWI service is unsolicited. So the SIP-T41P IP phone just needs to handle the MWI messages sent from the server. But for some servers, the MWI service is solicited. In this case, the SIP-T41P IP phone must enable the MWI Subscription for MWI messages.

Note

MWI service is not available on all servers. Contact your system administrator for more information.

The MWI subscription parameters you need to know:

Options	Description
Subscribe for MWI	Enable or disable a subscription for MWI service.
MWI Subscription Period	Period of MWI subscription. The IP phone re-sends a SUBSCRIBE request before initial SUBSCRIBE expiration.
Subscribe MWI to Voice Mail	Enable or disable a subscription to the voice mail number for MWI service.

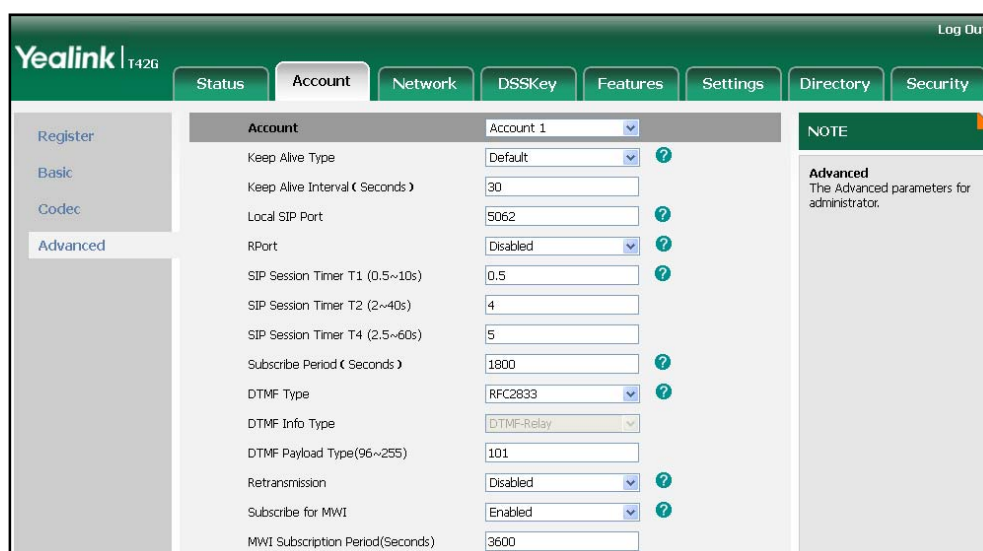
Options	Description
	To use this feature, you should also configure the voice mail number.

Note

Whether the phone subscribes the MWI messages to the account or the voice number MWI service depends on the server. Contact your system administrator for more information.

To enable subscribe for MWI via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select **Enabled** from the pull-down list of **Subscribe for MWI**.
5. Enter the period time in the **MWI Subscription Period (Seconds)** field.



The screenshot shows the Yealink T42G web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Account' dropdown is set to 'Account 1'. The 'Subscribe for MWI' dropdown is set to 'Enabled', and the 'MWI Subscription Period(Seconds)' field is set to 3600. A note on the right side of the page reads: 'NOTE: Advanced: The Advanced parameters for administrator.'

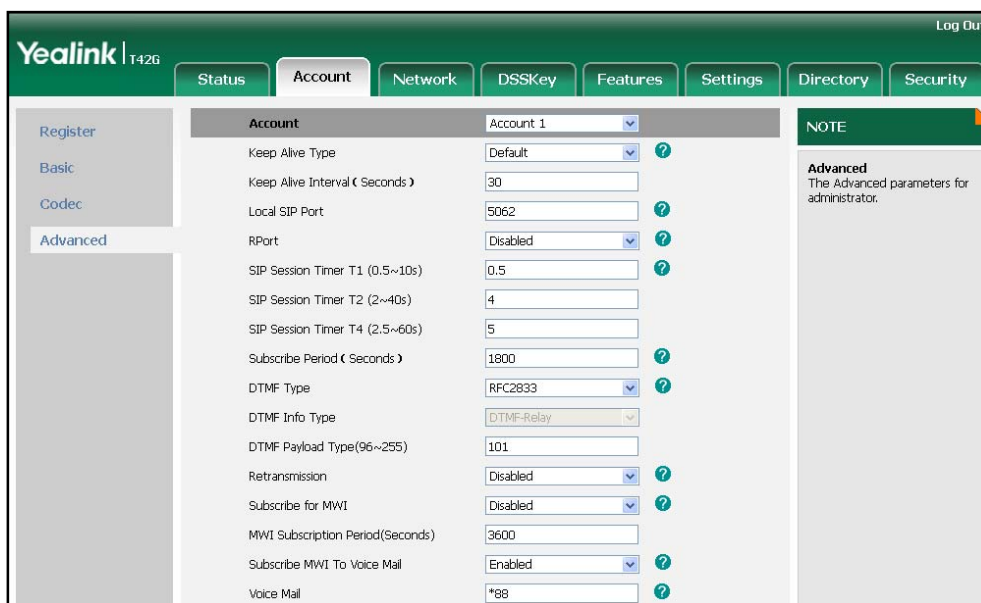
6. Click **Confirm** to accept the change.

The IP phone will subscribe to the account number for MWI service by default.

To enable the Subscribe MWI to VM feature via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select **Enabled** from the pull-down list of **Subscribe MWI To Voice Mail**.

5. Enter the desired voice number in the **Voice Mail** field.



6. Click **Confirm** to accept the change.

Troubleshooting

This chapter provides general troubleshooting information to help you solve the problems you might encounter when using your SIP-T41P IP phone.

If you require additional information or assistance with your new phone, contact your system administrator.

Why is the phone LCD screen blank?

- Ensure that the phone is properly plugged into a functional AC outlet.
- Ensure that the phone isn't plugged into a plug controlled by a switch that is off.
- If the phone is plugged into a power strip, try plugging it directly into a wall outlet instead.
- If the phone is powered from PoE, ensure you use a PoE compliant switch or hub, or contact your system administrator for more information.

Why does the phone display "Network Unavailable"?

- Ensure that the Ethernet cable is plugged into the Internet port on the phone and the Ethernet cable is not loose.
- Ensure that the switch or hub in your network is operational.

Why doesn't the phone display time and date correctly?

Check if you have configured your phone to obtain the time and date from the SNTP server automatically. If your phone is unable to access the SNTP server, configure the time and date manually.

How do I find the basic information of the IP phone?

Press the OK key when the IP phone is idle to check the basic information of the IP phone, such as IP address and firmware version. For more the basic information, refer to [Phone Status](#) on page 14.

How to obtain the MAC address of a phone when the phone is not powered on?

There are three ways to obtain the MAC address of a phone:

- You can ask your supplier for shipping information sheet which includes MAC addresses according to the corresponding PO.
- You can find the MAC address in the label of carton box.

- You can also find the MAC address from the phone's bar code on the back of the phone.

Why can't I get a dial tone?

Check for any loose connections and that the phone has been installed properly. For more on installation instructions, refer to [Phone Installation](#) on page 11.

Check whether dial tone is present on one of the audio modes.

- Switch between the Handset, Headset (if present) or Hands-Free Speakerphone to check whether dial tone is present for one of the audio modes.
- If the dial tone exists on another audio mode, connect a different handset or headset to isolate the problem.

Why doesn't the phone ring?

Check the ringer volume on your phone. To adjust the ringer volume setting, press the Volume key when the phone is on-hook and idle. For more information, refer to [Volume](#) on page 22.

Why can't I receive calls?

- Check the SIP registration with your system administrator.
- Check that DND (Do Not Disturb) mode is deactivated on your phone. Refer to [Do Not Disturb \(DND\)](#) on page 70.
- Check that call forward feature is disabled on the phone. Refer to [Call Forward](#) on page 73.
- Check whether the caller number is stored in the blacklist directory. Refer to [Blacklist](#) on page 34.

Why is my handset not working?

Check that the handset cord is fully connected to both the handset jack on the phone and handset. Refer to [Phone Installation](#) on page 11.

Why is my headset not working?

- Check that the headset cord is fully connected to the headset jack on the phone. Refer to [Phone Installation](#) on page 11.
- Check that the headset mode is activated. Refer to [Headset Use](#) on page 39.
- Check that the headset volume is adjusted to an appropriate level. Refer to [Volume](#) on page 22.

What is the difference between user name, register name and display name?

Both user name and register name are defined by the server. A user name is used to

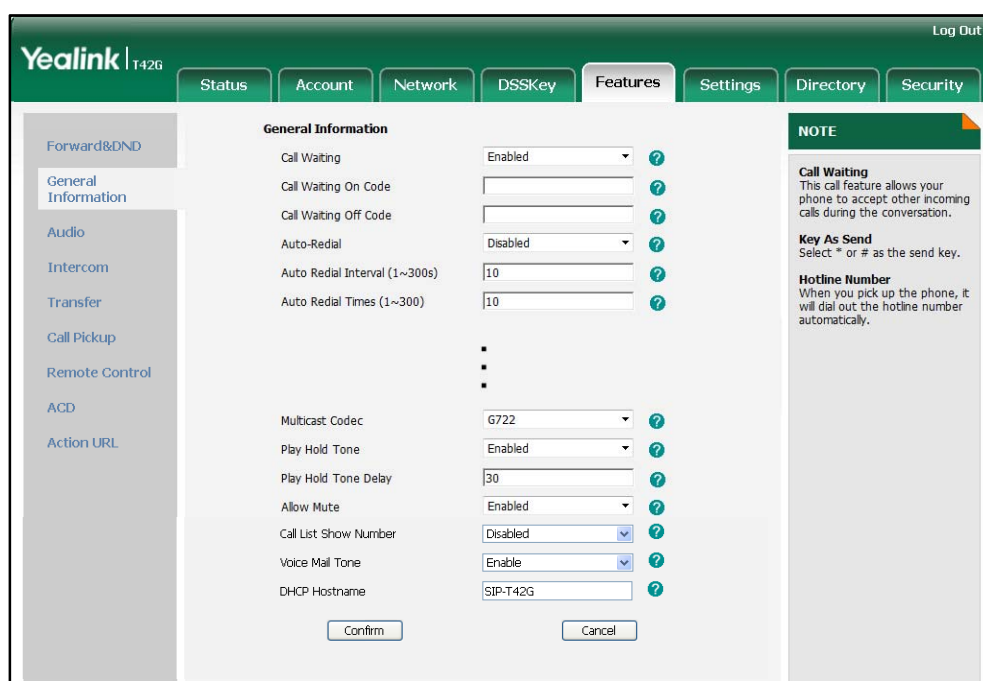
identify the account, while a register name matched with a password is used for authentication if the server requires. Display name is the caller ID that will display on the callee's phone LCD screen. Server configuration may override the local configuration.

Why does the phone play a tone when hold? How to disable it?

When the phone is hold, it will play a hold tone every 30 seconds to prompt that the phone is being hold status. The call hold tone feature is enabled by default. You can disable it or change the interval to play a hold tone via web user interface only.

To configure the call hold tone and call hold tone delay via web user interface:

1. Click on **Features->General Information**.
2. Select **Enabled** or **Disabled** from the pull-down list of **Play Hold Tone**.
3. Enter the desired time in the **Play Hold Tone Delay** field.



4. Click **Confirm** to accept the change.

How to change the user password?

To change the user password via web user interface:

1. Click on **Security->Password**.
2. Select **user** from the pull-down list of **User Type**.
3. Enter the new user password in the **New Password** field and **Confirm Password** field.

The screenshot shows the 'Security' tab in the Yealink T426 web interface. The 'User Type' dropdown menu is set to 'User'. Below it, there are three password fields: 'Old Password' (empty), 'New Password' (masked with dots), and 'Confirm Password' (masked with dots). There are 'Confirm' and 'Cancel' buttons at the bottom. A 'NOTE' box on the right states: 'User Type: Select your type. If you log in as user, you can only change your own password. If you login as an administrator, you can modify both the user's and admin's passwords.'

4. Click **Confirm** to accept the change.

You can also contact your system administrator for help.

Note

If logging in the web user interface of the phone with the user credential, you need to enter the current user password in the **Old Password** field.

How to make a call using SRTP?

You can enable SRTP to encrypt the audio stream(s) of phone calls. The parties participating in the call should enable SRTP. You can enable SRTP on per-line basis.

To enable SRTP via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select the desired type from the pull-down list of **RTP Encryption (SRTP)**.

The screenshot shows the 'Account' tab in the Yealink T426 web interface. The 'Account' dropdown menu is set to 'Account 1'. The 'Advanced' sub-tab is selected. The 'RTP Encryption (SRTP)' dropdown menu is set to 'Force SRTP'. Other settings visible include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5062), 'PTime(ms)' (20), 'ACD Subscrip Period(120~3600s)' (3600), 'Early Media' (Disabled), 'SIP Server Type' (Default), 'Music Server URI', 'Directed Call Pickup Code', 'Group Call Pickup Code', 'Distinctive Ring Tones' (Disabled), 'Unregister When Reboot' (Disabled), and 'Out Dialog BLF' (Disabled). There are 'Confirm' and 'Cancel' buttons at the bottom. A 'NOTE' box on the right states: 'Advanced: The Advanced parameters for administrator.'

5. Click **Confirm** to accept the change.

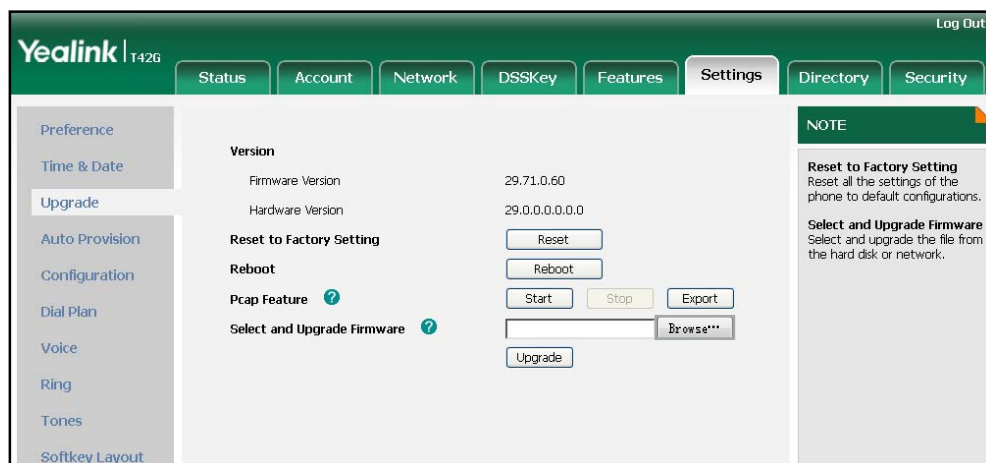
Note

The SRTP feature is not available on all servers. Contact your system administrator for more information.

How to reboot the phone?

To reboot the phone via web user interface:

1. Click on **Settings->Upgrade**.



2. Click **Reboot** to reboot the IP phone.

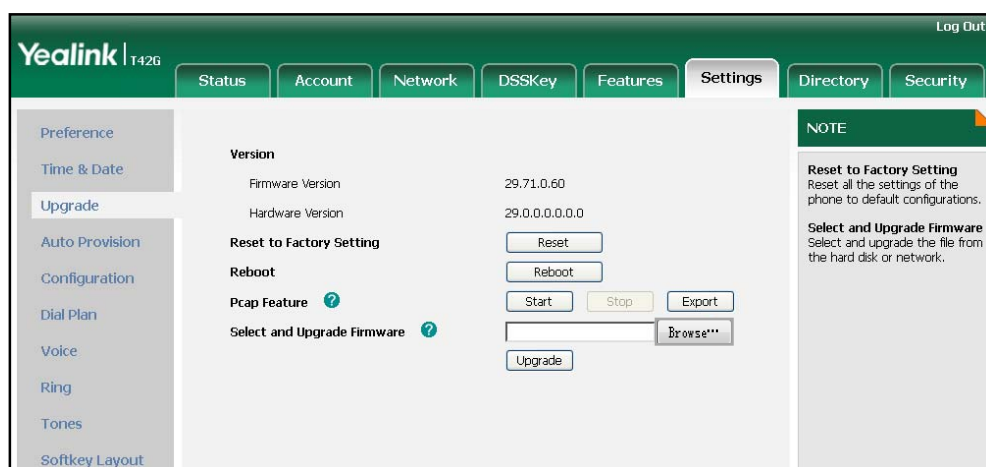
Note Any reboot of the phone may take a few minutes.

How to export PCAP Trace?

We may need you to provide a PCAP trace to help analyze your problem.

To export a PCAP trace via web user interface:

1. Click on **Settings->Upgrade**.
2. Click **Start** to start the SIP trace capture.
3. Recreate the error to be documented in the trace.
4. Click **Stop** to stop the capture.



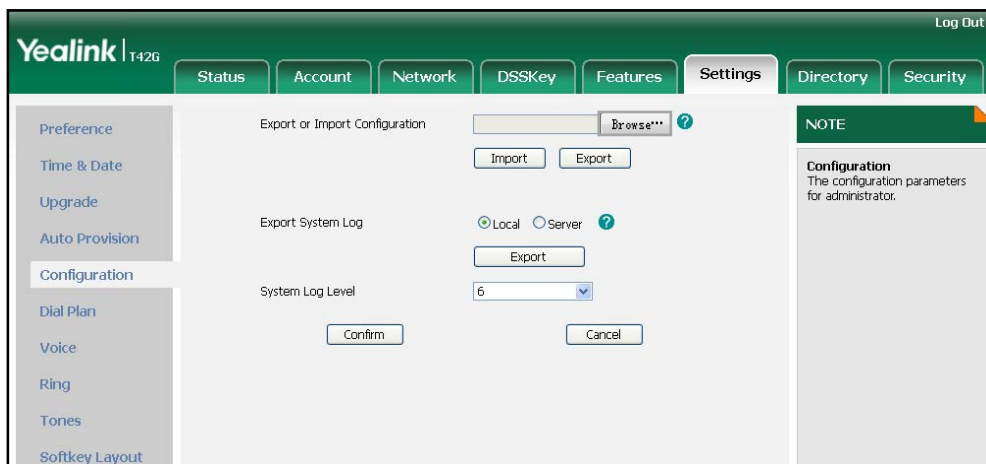
5. Click **Export** to save the file to your local system.

How to export system log?

We may need you to provide debug log information to help analyze your problem. By default, you can export the debug system log information to local PC.

To export the system log to local PC via web user interface:

1. Click on **Settings->Configuration**.
2. Select **6** from the pull-down list of **System Log Level**.
The default system log level is 3.
3. Click **Confirm** to accept the change.
The web user interface prompts "Do you want to restart your machine?". The configuration will take effect after reboot.
4. Click **OK** to reboot the phone.
After rebooting the phone, the system log level is set as 6, the debug level.
5. Mark the **Local** radio box in the **Export System Log** field.



6. Click **Export** to download and save the system log file to your local PC system.

You can also export the system log to a syslog server, contact your system administrator for more information.

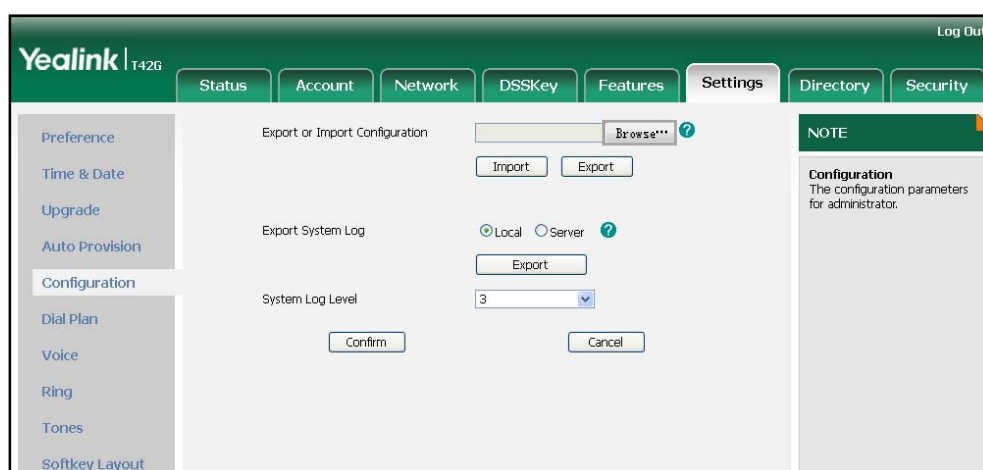
How to export/import the phone configurations?

We may need you to provide your phone configurations to help analyze your problem. In some instance, you may need to import configurations to your phone.

To export the phone configurations via web user interface:

1. Click on **Settings->Configuration**.

2. Click **Export** to save the configuration file to your local system.



To import the phone configurations via web user interface:

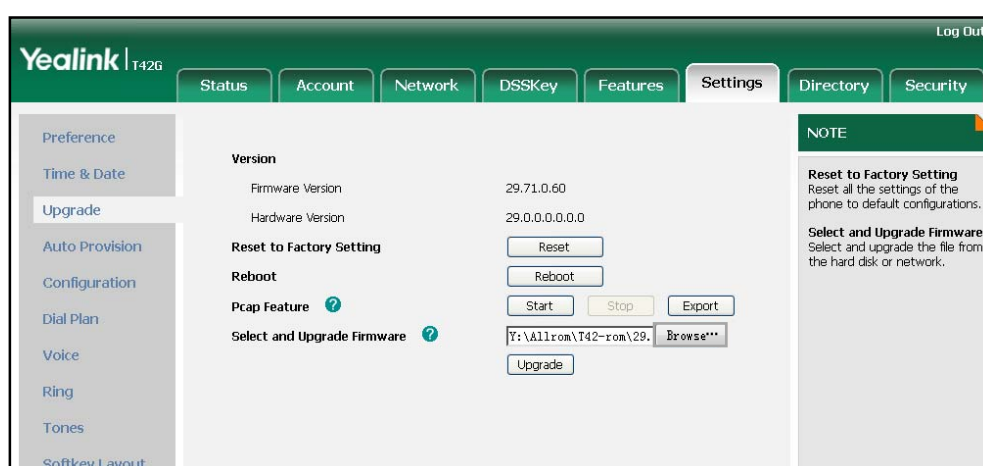
1. Click on **Settings->Configuration**.
2. Click **Browse** to select a configuration file from your local system.
3. Click **Import** to import the configuration file.

Note The file format of configuration file must be “.bin”.

How to upgrade firmware?

To upgrade firmware via web user interface:

1. Click on **Settings->Upgrade**.
2. Click **Browse** to select the firmware from your local system.



3. Click **Upgrade** to upgrade the firmware.

The web user interface prompts “Firmware of the SIP Phone will be updated. It will take 5 minutes to complete. Please don't power off!”.

4. Click **OK** to confirm upgrading.

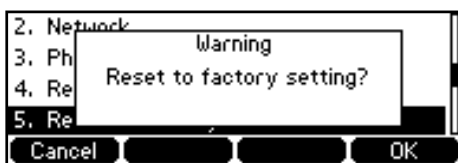
How to reset the phone?

Reset the phone to factory configurations after you have tried almost all troubleshooting suggestions but do not solve the problem. You need to note that all customized settings will be overwritten after reset.

To reset the phone via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) ->**Reset to Factory**.
2. Press the **Enter** soft key.

The phone LCD screen prompts the following warning:



3. Press the **OK** soft key.

The phone LCD screen prompts the following:



The phone will be reset to factory successfully after startup.

Note

Reset of the phone may take a few minutes. Do not power off until the phone starts up successfully.

Regulatory Notices

Service Agreements

Contact your Yealink Authorized Reseller for information about service agreements applicable to your product.

Limitations of Liability

TO THE FULL EXTENT ALLOWED BY LAW, YEALINK EXCLUDES FOR ITSELF AND ITS SUPPLIERS ANY LIABILITY, WHETHER BASED IN CONTRACT OR TORT (INCLUDING NEGLIGENCE), FOR INCIDENTAL, CONSEQUENTIAL, INDIRECT, SPECIAL, OR PUNITIVE DAMAGES OF ANY KIND, OR FOR LOSS OF REVENUE OR PROFITS, LOSS OF BUSINESS, LOSS OF INFORMATION OR DATA, OR OTHER FINANCIAL LOSS ARISING OUT OF OR IN CONNECTION WITH THE SALE, INSTALLATION, MAINTENANCE, USE, PERFORMANCE, FAILURE, OR INTERRUPTION OF ITS PRODUCTS, EVEN IF YEALINK OR ITS AUTHORIZED RESELLER HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES, AND LIMITS ITS LIABILITY TO REPAIR, REPLACEMENT, OR REFUND OF THE PURCHASE PRICE PAID, AT YEALINK'S OPTION. THIS DISCLAIMER OF LIABILITY FOR DAMAGES WILL NOT BE AFFECTED IF ANY REMEDY PROVIDED HEREIN SHALL FAIL OF ITS ESSENTIAL PURPOSE.


Safety Instructions

Save these instructions. Read these safety instructions before use!

The following basic safety precautions should always be followed to reduce risk of fire, electrical shock, and other personal injury.

General Requirements

- Before you install and use the device, read the safety instructions carefully and observe the situation during operation.
- During the process of storage, transportation, and operation, please always keep the device dry and clean.
- During the process of storage, transportation, and operation, please avoid collision and crash of the device.
- Please attempt not to dismantle the device by yourself. In case of any discrepancy, please contact the appointed maintenance center for repair.
- Without prior written consent, no organization or individual is permitted to make any change to the structure or the safety design of the device. Yealink is under no circumstance liable to consequences or legal issues caused by such changes.
- Please refer to the relevant laws and statutes while using the device. Legal rights of others should be respected as well.

 **Environmental Requirements**

- Place the device at a well-ventilated place. Do not expose the device under direct sunlight.
- Keep the device dry and free of dusts.
- Place the device on a stable and level platform.
- Please place no heavy objects on the device in case of damage and deformation caused by the heavy load.
- Keep at least 10 cm between the device and the closest object for heat dissipation.
- Do not place the device on or near any inflammable or fire-vulnerable object, such as rubber-made materials.
- Keep the device away from any heat source or bare fire, such as a candle or an electric heater.
- Keep the device away from any household appliance with strong magnetic field or electromagnetic field, such as a microwave oven or a refrigerator.

 **Operating Requirements**

- Do not let a child operate the device without guidance.
- Do not let a child play with the device or any accessory in case of accidental swallowing.
- Please use the accessories provided or authorized by the manufacturer only.
- The power supply of the device shall meet the requirements of the input voltage of the device. Please use the provided surge protection power socket only.
- Before plugging or unplugging any cable, make sure that your hands are completely dry.
- Do not spill liquid of any kind on the product or use the equipment near water, for example, near a bathtub, washbowl, kitchen sink, wet basement or near a swimming pool.
- Do not tread on, pull, or over-bend any cable in case of malfunction of the device.
- During a thunderstorm, stop using the device and disconnect it from the power supply. Unplug the power plug and the Asymmetric Digital Subscriber Line (ADSL) twisted pair (the radio frequency cable) to avoid lightning strike.
- If the device is left unused for a rather long time, disconnect it from the power supply and unplug the power plug.
- When there is smoke emitted from the device, or some abnormal noise or smell, disconnect the device from the power supply, and unplug the power plug immediately. Contact the specified maintenance center for repair.
- Do not insert any object into equipment slots that is not part of the product or auxiliary product.
- Before connecting a cable, connect the grounding cable of the device first. Do not disconnect the grounding cable until you disconnect all other cables.

 **Cleaning Requirements**

- Before cleaning the device, stop using it and disconnect it from the power supply.

- Use a piece of soft, dry and anti-static cloth to clean the device.
- Keep the power plug clean and dry. Using a dirty or wet power plug may lead to electric shock or other peril

Appendix A - Time Zones

Time Zone	Time Zone Name
-11:00	Samoa
-10:00	United States-Hawaii-Aleutian
-10:00	United States-Alaska-Aleutian
-09:00	United States-Alaska Time
-08:00	Canada(Vancouver, Whitehorse)
-08:00	Mexico(Tijuana, Mexicali)
-08:00	United States-Pacific Time
-07:00	Canada(Edmonton, Calgary)
-07:00	Mexico(Mazatlan, Chihuahua)
-07:00	United States-Mountain Time
-07:00	United States-MST no DST
-06:00	Canada-Manitoba(Winnipeg)
-06:00	Chile(Easter Islands)
-06:00	Mexico(Mexico City, Acapulco)
-06:00	United States-Central Time
-05:00	Bahamas(Nassau)
-05:00	Canada(Montreal, Ottawa, Quebec)
-05:00	Cuba(Havana)
-05:00	United States-Eastern Time
-04:30	Venezuela(Caracas)
-04:00	Canada(Halifax, Saint John)
-04:00	Chile(Santiago)
-04:00	Paraguay(Asuncion)
-04:00	United Kingdom-Bermuda(Bermuda)
-04:00	United Kingdom(Falkland Islands)
-04:00	Trinidad&Tobago
-03:30	Canada- New Foundland(St.Johns)
-03:00	Denmark-Greenland(Nuuk)
-03:00	Argentina(Buenos Aires)
-03:00	Brazil(no DST)
-03:00	Brazil(DST)
-02:00	Brazil(no DST)
-01:00	Portugal(Azores)
0	GMT
0	Greenland
0	Denmark-Faroe Islands(Torshavn)
0	Ireland(Dublin)
0	Portugal(Lisboa, Porto, Funchal)
0	Spain-Canary Islands(Las Palmas)
0	United Kingdom(London)
0	Morocco
+01:00	Albania(Tirane)
+01:00	Austria(Vienna)
+01:00	Belgium(Brussels)
+01:00	Caicos
+01:00	Chad
+01:00	Croatia(Zagreb)
+01:00	Czech Republic(Prague)
+01:00	Denmark(Kopenhagen)
+01:00	France(Paris)
+01:00	Germany(Berlin)
Time Zone	Time Zone Name

+01:00	Hungary(Budapest)
+01:00	Italy(Rome)
+01:00	Luxembourg(Luxembourg)
+01:00	Macedonia(Skopje)
+01:00	Netherlands(Amsterdam)
+01:00	Namibia(Windhoek)
+02:00	Estonia(Tallinn)
+02:00	Finland(Helsinki)
+02:00	Gaza Strip(Gaza)
+02:00	Greece(Athens)
+02:00	Israel(Tel Aviv)
+02:00	Jordan(Amman)
+02:00	Latvia(Riga)
+02:00	Lebanon(Beirut)
+02:00	Moldova(Kishinev)
+02:00	Russia(Kaliningrad)
+02:00	Romania(Bucharest)
+02:00	Syria(Damascus)
+02:00	Turkey(Ankara)
+02:00	Ukraine(Kyiv, Odessa)
+03:00	East Africa Time
+03:00	Iraq(Baghdad)
+03:00	Russia(Moscow)
+03:30	Iran(Teheran)
+04:00	Armenia(Yerevan)
+04:00	Azerbaijan(Baku)
+04:00	Georgia(Tbilisi)
+04:00	Kazakhstan(Aktau)
+04:00	Russia(Samara)
+04:30	Afghanistan
+05:00	Kazakhstan(Aqtobe)
+05:00	Kyrgyzstan(Bishkek)
+05:00	Pakistan(Islamabad)
+05:00	Russia(Chelyabinsk)
+05:30	India(Calcutta)
+06:00	Kazakhstan(Astana, Almaty)
+06:00	Russia(Novosibirsk, Omsk)
+07:00	Russia(Krasnoyarsk)
+07:00	Thailand(Bangkok)
+08:00	China(Beijing)
+08:00	Singapore(Singapore)
+08:00	Australia(Perth)
+09:00	Korea(Seoul)
+09:00	Japan(Tokyo)
+09:30	Australia(Adelaide)
+09:30	Australia(Darwin)
+10:00	Australia(Sydney, Melbourne, Canberra)
+10:00	Australia(Brisbane)
+10:00	Australia(Hobart)
+10:00	Russia(Vladivostok)
+10:30	Australia(Lord Howe Islands)
+11:00	New Caledonia(Noumea)
+12:00	New Zealand(Wellington, Auckland)
+12:45	New Zealand(Chatham Islands)
+13:00	Tonga(Nukualofa)

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