Yealink
easy VoIP

Ultra-elegant Gigabit IP Phone SIP-T42G User Guide

Version 72.2

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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, and
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 experience 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.

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Yealink SIP-T42G IP phone 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 online:
http://www.yealink.com/GPLOpenSource.aspx?BaseInfoCateId=293&NewsCateId=293&CateId=293.

About This Guide

Thank you for choosing the SIP-T42G IP Phone, an ultra-elegant gigabit IP Phone which is exquisitely designed to provide business telephony features, such as Call Hold, Call Transfer, Busy Lamp Field, Multicast Paging and Conference over an IP network.

This guide provides everything you need to quickly use your new phone. First, verify with your system administrator that the IP network is ready for phone configuration. Also be sure to read the Packaging Contents and Regulatory Notices sections in this guide before you set up and use the SIP-T42G IP phone.

If this is your first time using the SIP-T42G IP phone, we recommend that you first refer to the *Quick Installation Guide* and *Quick Reference Guide*, available online: http://www.yealink.com/SupportDownloadfiles_detail.aspx?CateId=312&flag=142.

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

In This Guide

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

Summary of Changes

This section describes the changes to this guide for each release and guide version.

Changes for Release 72, Guide Version 72.2

Major updates have occurred to the following sections:

- Documentations on page 8
- Packaging Contents on page 9

Changes for Release 72, Guide Version 72.1

The following section is new:

- Call Completion on page 74

Major updates have occurred to the following sections:

- LED Instructions on page 4
- Backlight on page 19
- Anonymous Call on page 95

Changes for Release 71, Guide Version 71.120

Major updates have occurred to the following sections:

- Icon Instructions on page 3
- DSS Keys on page 47
- Dial Plan on page 59
- Anonymous Call on page 95
- Troubleshooting on page 115

Changes for Release 71, Guide Version 71.90

Major updates have occurred to the following sections:

- Phone Status on page 14
- Directory on page 29
- Remote Phone Book on page 40

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Overview

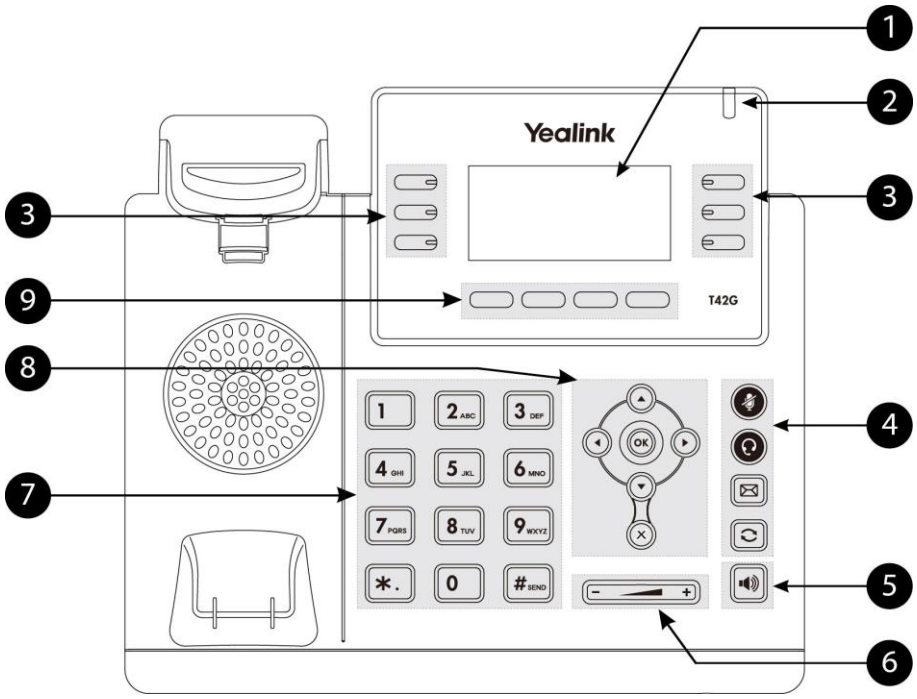
This chapter provides the overview of the SIP-T42G IP phone. Topics include:

- Hardware Component Instructions
- Icon Instructions
- LED Instructions
- User Interfaces
- Documentations

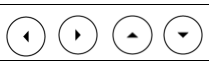
If you require additional information or assistance with your new phone, contact your system administrator.

Hardware Component Instructions

The main hardware components of the SIP-T42G IP phone are the LCD screen and the keypad.



Hardware component instructions of the SIP-T42G IP phone are:

	Item	Description
①	LCD Screen	Shows information about calls, messages, soft keys, time, date and other relevant data: • Call information—caller ID, call duration • Icons (for example, DND) • Missed call text or second incoming caller information • Prompt text (for example, "Saving config file!") • Time and date
②	Power Indicator LED	Indicates phone power and some feature statuses.
③	Line Keys	Use these keys to activate up to three accounts and assign various features.
④	Mute Key	Toggles and indicates 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 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 identity their context-sensitive features.

Icon Instructions

Icons appearing on the LCD screen are described in the following table:

Icon	Description
	Network is unavailable
	Registered successfully
	Register 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
2aB	Alphanumeric input mode
123	Numeric input mode
abc	Multi-lingual lowercase letters input mode
ABC	Multi-lingual uppercase letters input mode
Abc	Multi-lingual uppercase and lowercase letters input mode
	Call Forward/Forwarded Calls

Icon	Description
	Missed Calls
	Received Calls
	Placed Calls
	Recording box is full
	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 (300ms)	The phone is ringing.
Slow flashing red (1s)	The phone receives a voice mail.
Off	The phone is powered off. The phone is idle. The phone is busy. The call is placed on hold or is held. The call is mute.

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 a BLF key)

LED Status	Description
Solid green	The monitored user is idle.
Fast flashing green (200ms)	The monitored user receives an incoming call.
Slow flashing green (500ms)	The monitored user is dialing. The monitored user is talking. The monitored user's conversation is placed on hold.
Slow flashing green (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user does not exist.

Line key LED (used as a 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
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 on every page are idle.

Note The above introduces the default LED status. The statuses of the power indicator LED and BLF key are configurable via web user interface. For more information, refer to *Yealink_SIP-T4X_IP_Phone_Family_Administrator_Guide*.

User Interfaces

Two ways to customize configurations of your SIP-T42G IP phone:

- The user interface on the IP phone.
- The user interface in a web browser on 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 cases, it is possible to use either the phone user interface and/or the web user interface interchangeably. However, in some cases, it is only possible to use one or the other interface to operate the phone and change settings.

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 the administrator, and the default administrator password is "admin" (case-sensitive). For more information on customizing your phone with the available options from the phone user interface, refer to Customizing Your Phone on page 19.

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		√
--Auto Answer	√	
--Auto Redial	√	
--Call Completion		
--Call Forward	√	
--Call Waiting	√	
--Call Park	√	
--Call Pickup	√	
--Call Return	√	
--Conference		
--Anonymous Call	√	
--Anonymous Call Rejection	√	
--DND	√	
Basic Phone Settings		
--Language	√	√

Options	Phone User Interface	Web User Interface
--Time & Date	√	√
--Administrator Password	√	√
--Ring Tones	√	√
--Backlight	√	√
--Keypad Lock	√	√
--Phone Volume	√	
--Call History Management	√	
--Contact Management		√
--Local Directory	√	√
--Blacklist	√	√
--Remote Phone Book		√
--LDAP		√
--Dial Plan		√
--DSS Keys	√	√
--Key as Send	√	√
--Hotline	√	√
--Live Dialpad		√
--Emergency Number		√
Advanced Phone Features		√
--BLF	√	
--Call Recording	√	
--Intercom	√	
--Multicast Paging		
--Music on Hold		
--ACD	√	
--Messages	√	
SIP Account		√
--User Options		
--Register Status	√	
--Account Status	√	
--Label	√	
--Display Name	√	
--Register Name	√	
--User Name	√	
--Password	√	
--SIP Server	√	
--Server Option		
--Server Port		
--Outbound Status	√	
--Outbound Proxy	√	
--NAT Traversal		

Options	Phone User Interface	Web User Interface
--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-T42G IP phone.

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

Note You can also download the latest documentations online: http://www.yealink.com/SupportDownloadfiles_detail.aspx?CateId=312&flag=142.

Getting Started

This chapter provides the following basic installation instructions and information for obtaining the best performance with the SIP-T42G IP phone. Topics include:

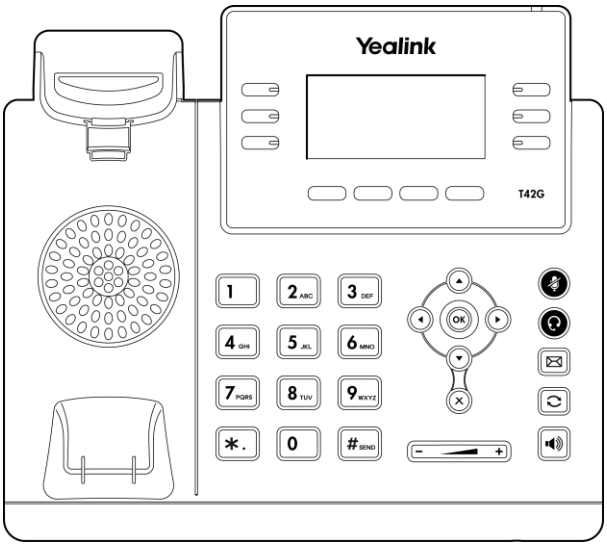
- Packaging Contents
- Phone Installation
- Phone Initialization
- Phone Status
- Basic Network Settings
- Registration
- Idle Screen

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

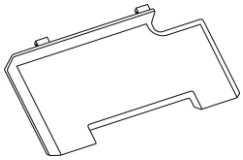
Packaging Contents

The following components are included in your SIP-T42G IP phone package:

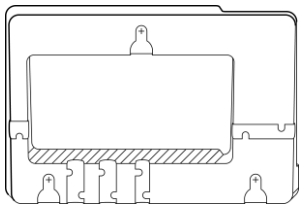
- **SIP-T42G IP Phone**



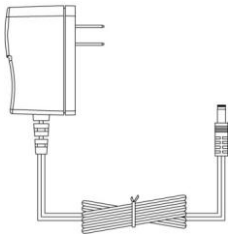
- **Phone Stand**



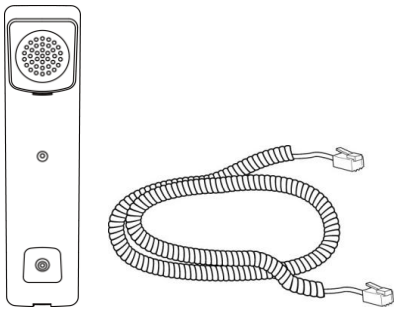
- **Wall Mount Bracket (Optional)**



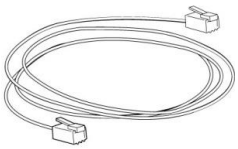
- **Power Adapter (Optional)**



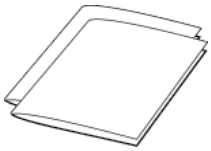
- **Handset & Handset Cord**



- **Ethernet Cable**



- **Quick Installation Guide & Quick Reference Guide**



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

Phone Installation

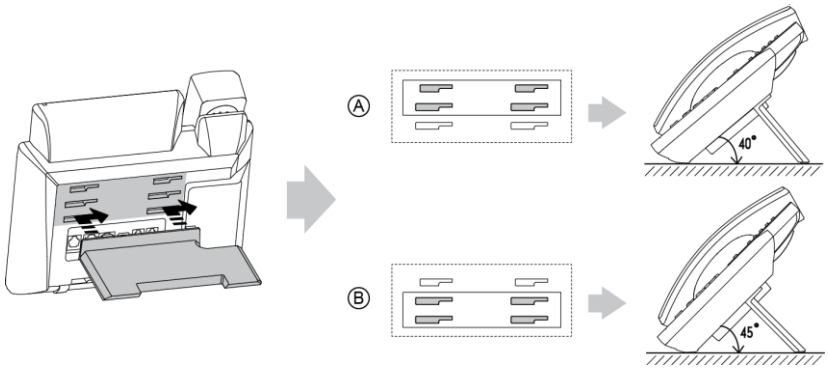
If your phone is already installed, proceed to Phone Initialization on page 13.

This section introduces how to install the phone:

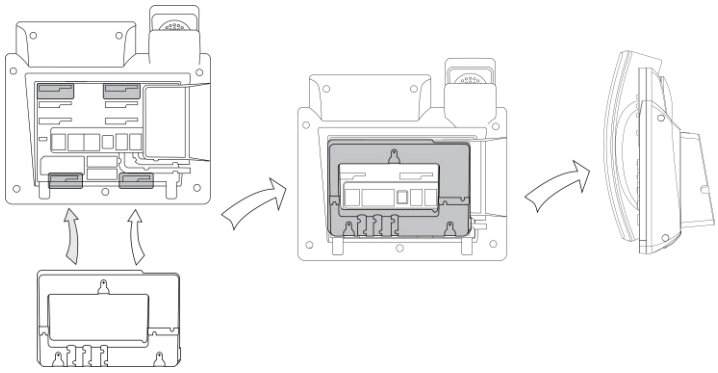
1) Attach the stand or the optional wall mount bracket
2) Connect the handset and optional headset
3) Connect the network and power

1) Attach the stand or the optional wall mount bracket

Desk Mount Method

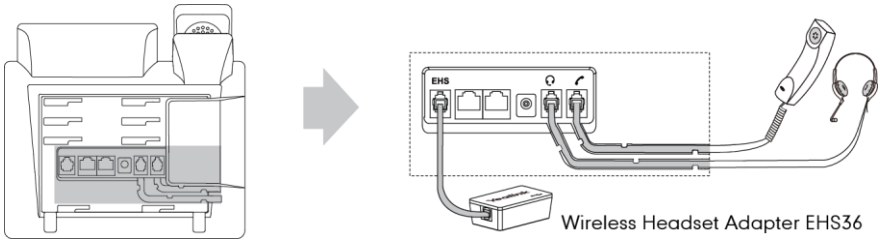


Wall Mount Method (Optional)



Note The wall mount bracket should be purchased separately. For more information on how to attach the wall mount bracket, refer to *Yealink Wall Mount Quick Installation Guide for SIP-T4X IP Phones*.

2) Connect the handset and optional headset



Note The headset and wireless headset adapter EHS36 should be purchased separately.

EHS36 is used for connecting the phone and a compatible wireless headset. For more information on how to use the EHS36, refer to *Yealink EHS36 User Guide*.

3) Connect the network and power

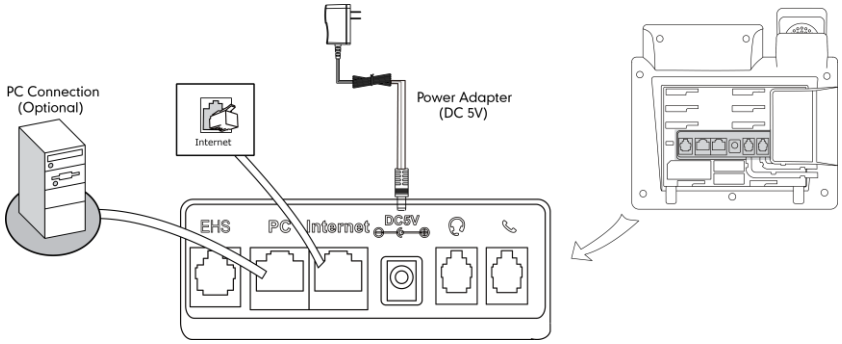
You have two options for power and network connections. Your system administrator will advise you which one to use.

- 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 included or a standard Ethernet cable between the Internet port on the phone and the one on the wall or switch/hub device port.

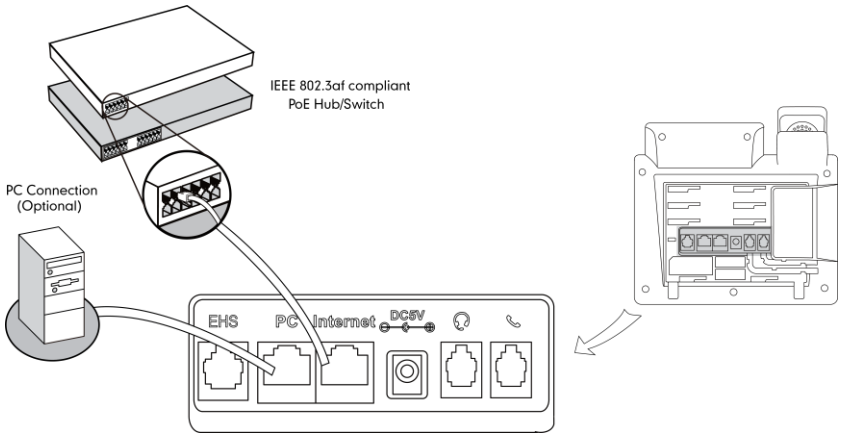


Power over Ethernet

With the included or a regular Ethernet cable, the SIP-T42G IP phone can be powered from a PoE-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 switch/hub is provided, you don't need to connect the phone to the power adapter. Make sure the switch/hub is PoE-compliant.

The phone can also share the network with another network device 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 is powered on, the system boots up and performs the following steps:

Automatic Phone Initialization

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

DHCP (Dynamic Host Configuration Protocol)

The phone attempts to contact a DHCP server in your network to obtain valid IPv4 network settings (e.g., IP address, subnet mask, default gateway address and DNS address) by default.

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 (e.g., IPv4 status, IP address mode, MAC address, LAN type, LAN IP and LAN mask).
- Phone status (e.g., device model, hardware version, firmware version, product ID and device certificate status).
- Account status (e.g., register status of SIP accounts).

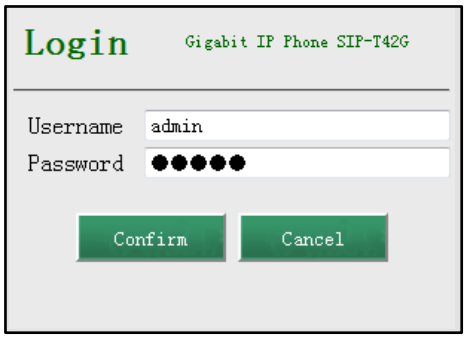
Note You can view device certificate status via phone user interface only.

To view the phone status via phone user interface:

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

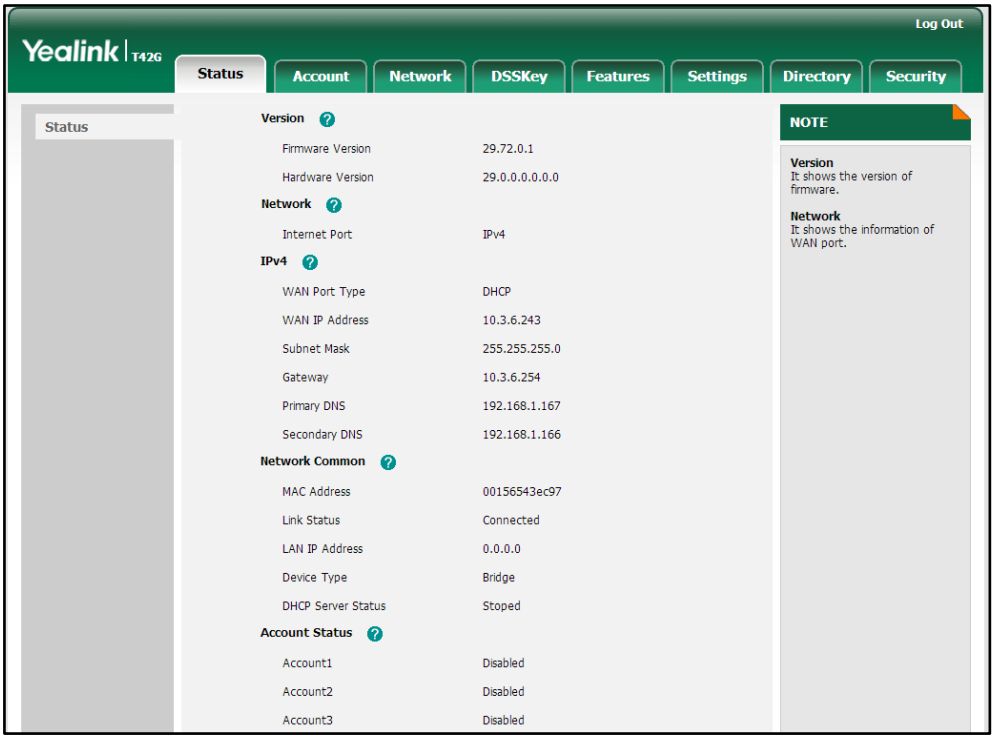
To view the phone status via web user interface:

1. Open the web browser on 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 phone status is displayed on the first page of the web user interface.

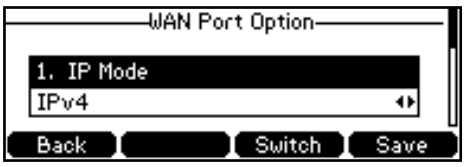


Basic Network Settings

If your phone cannot contact a DHCP server for any reason, you need to configure network settings manually. The IP phone can support either 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 ►, or the **Switch** soft key 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**.
2. Press ▼ to select **IPv4** and press the **Enter** soft key.
3. Press ▼ to select **Static IPv4 Client** and press the **Enter** soft key.

4. Enter the desired values in the **IPv4 Address**, **Subnet Mask**, **Default Gateway**, **Primary DNS** and **Secondary DNS** fields respectively.

5. 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**.
2. Press (▾) to select **IPv6** and press the **Enter** soft key.
3. Press (▾) to select **Static IPv6 Client** and press the **Enter** soft key.
4. Enter the desired values in the **IP Address**, **Prefix**, **Default Gateway**, **Primary DNS** and **Secondary DNS** fields respectively.

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

Note Wrong network settings may result in inaccessibility of your phone and may also have an impact on your network performance. For more information on 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 for 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 57.

Idle Screen

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

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

Customizing Your Phone

You can customize your SIP-T42G IP phone by personally configuring certain settings, for example, contrast, time & date and ring tones. You can add contacts to the phone's local directory manually or from call history. You can also personalize different ring tones for different callers.

This chapter provides basic operating instructions for customizing your phone. 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

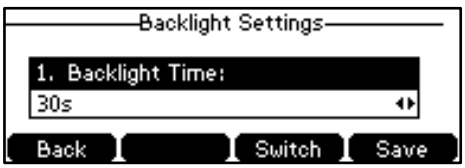
Backlight

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

- **Always On:** Backlight is on permanently.
- **15, 30, 60, 120, 300, 600** or **1800:** Backlight is turned off when the phone is inactive after the specified time (in seconds).

To configure the backlight via phone user interface:

1. Press **Menu**->**Settings**->**Basic Settings**->**Display**->**Backlight Settings**.
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.

Backlight is configurable 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 web browser for the web user interface. If the language of your web 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.

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 on the available languages of your new phone.

To change the language for the phone user interface:

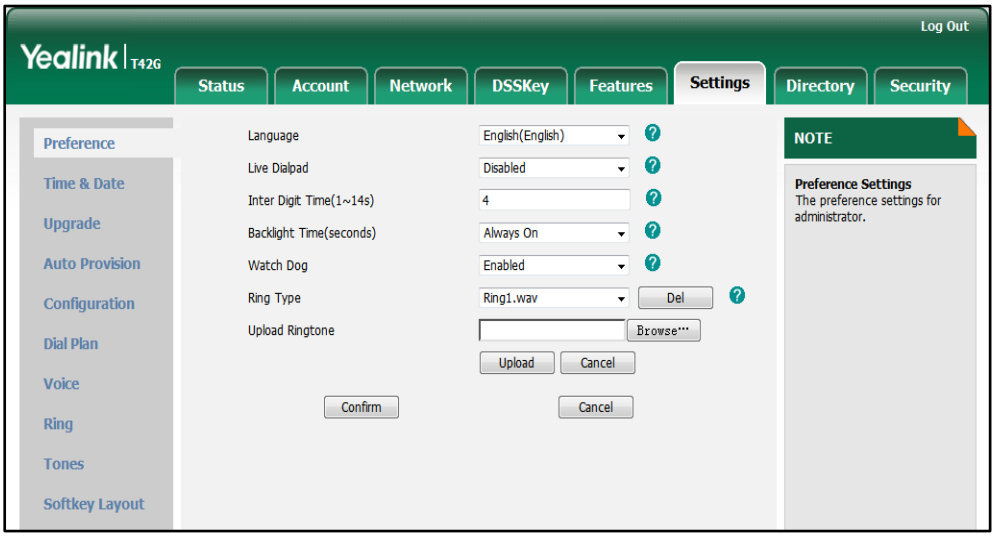
1. Press **Menu->Settings->Basic Settings->Language**.
2. Press ▲ or ▼ to select the desired language.

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

   Text displayed on the phone user interface 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 displayed on the web user interface will change to the selected language.

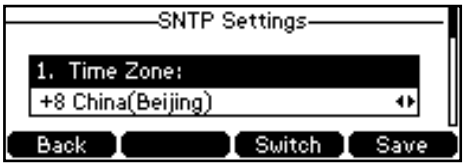
Time & Date

The time and date are displayed 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 Server1** and **NTP Server2** fields respectively.
4. Press ◄ or ► , or the **Switch** soft key to select the desired value from the **Daylight Saving** field.

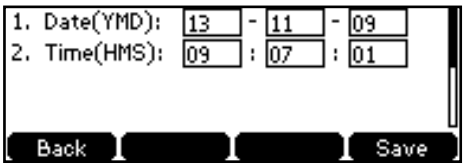


5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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.

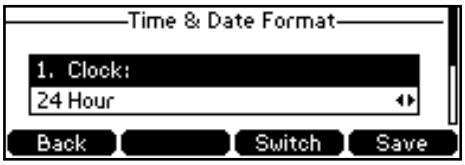


3. Press the **Save** soft key to accept the change.

   The date and time displayed 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 the 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

Time and date are configurable via web user interface at the path **Settings**->**Time & Date**.

Administrator Password

The Advanced Settings option is only accessible to the 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 PWD** field.

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

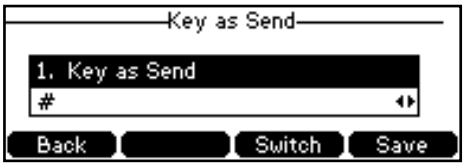
Administrator password is configurable via web user interface at the path **Security->Password**.

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 ◄ or ► , 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.

Key as send is configurable via web user interface at the path **Features->General Information**.

Keypad Lock

You can lock the keypad of your phone temporarily when you are not using it. This feature helps to protect your phone from unauthorized use.

Keypad lock consists of the following:

Menu Key: The Menu soft key is locked. You cannot access the menu of the phone until unlocked.

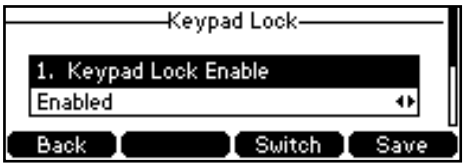
Function Keys: The function keys are locked. You cannot use the MESSAGE, REDIAL, MUTE, OK, X, navigation keys, soft keys and line keys until unlocked.

All Keys: All keys are locked, except the Volume, Headset, Speakerphone and digit keys. You are only allowed to dial emergency numbers, reject or answer incoming calls, and cancel a call.

Note The emergency number setting, if desired, must be made before lock activation. For more information, refer to Emergency Number on page 64.

To activate the keypad lock via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Phone Settings**->**Keypad Lock**.
2. Press ◄ or ► , or the **Switch** soft key to select **Enabled** from the **Keypad Lock Enable** field.
3. Press ◄ or ► , or the **Switch** soft key to select the desired type from the **Keypad Lock Type** field.



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

   Long press # to lock the keypad immediately when the phone is idle.

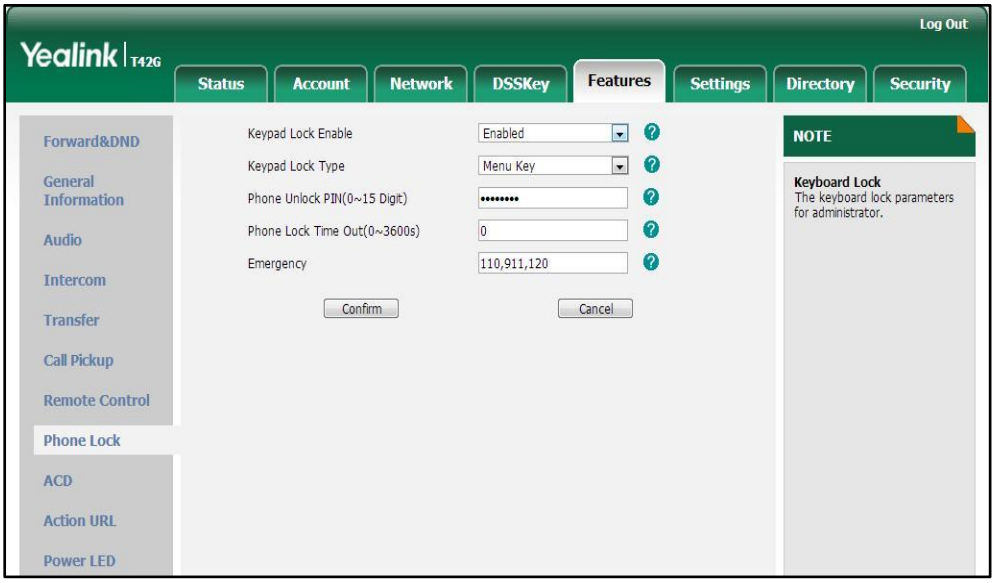
   The LCD screen prompts "Keypad Locked!" and displays the icon .

You can configure the interval (in seconds) to automatically lock the keypad instead of long pressing # .

To configure the interval of automatic keypad lock via web user interface:

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

2. Enter the desired time in the **Phone Lock Time Out (0~3600s)** field.



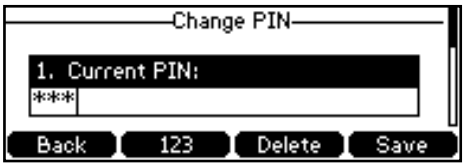
3. Click **Confirm** to accept the change.

Note The default timeout is 0 second, that is, you can long press # to lock the keypad only. Interval of automatic keypad lock is configurable via web user interface only.

To unlock the keypad, you must know the keypad unlock PIN of the phone. The default keypad unlock PIN is "123".

To change the keypad unlock PIN via phone user interface:

1. Press **Menu**->**Settings**->**Basic Settings**->**Phone Unlock PIN**.
2. Enter the desired values in the **Current PIN**, **New PIN** and **Confirm PIN** fields respectively.



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

Note The unlock PIN length must be within 15 digits.

To unlock the keypad via phone user interface:

1. Press any locked key, the LCD screen prompts "Please Enter PIN".

2. Enter the PIN in the **PIN** field.
3. Press the **Confirm** soft key to unlock the keypad.

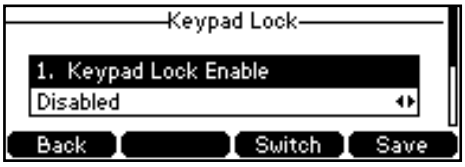
   The icon disappears from the LCD screen.

   You can long press # or wait for a period of time (if configured) to lock the keypad again.

Note You can also unlock the keypad by entering administrator password. When you enter the administrator password to unlock the keypad, the phone will turn to the **Reset Phone PIN** interface.

To deactivate the keypad lock via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Phone Settings**->**Keypad Lock**.
2. Press or , or the **Switch** soft key to select **Disabled** from the **Keypad Lock Enable** field.



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

Keypad lock is configurable via web user interface at the path **Features->Phone Lock**.

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 currently engaged audio devices (handset, speakerphone or headset) when the phone is in use.

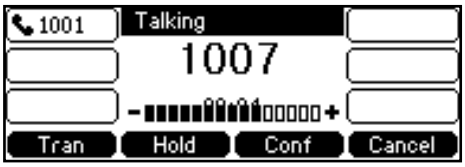
To adjust the volume when the phone is idle:

1. Press to adjust the ringer volume.

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

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

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

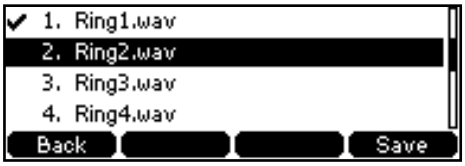


Ring Tones

Ring tones are used to indicate incoming calls. You can select different ring tones to distinguish 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.



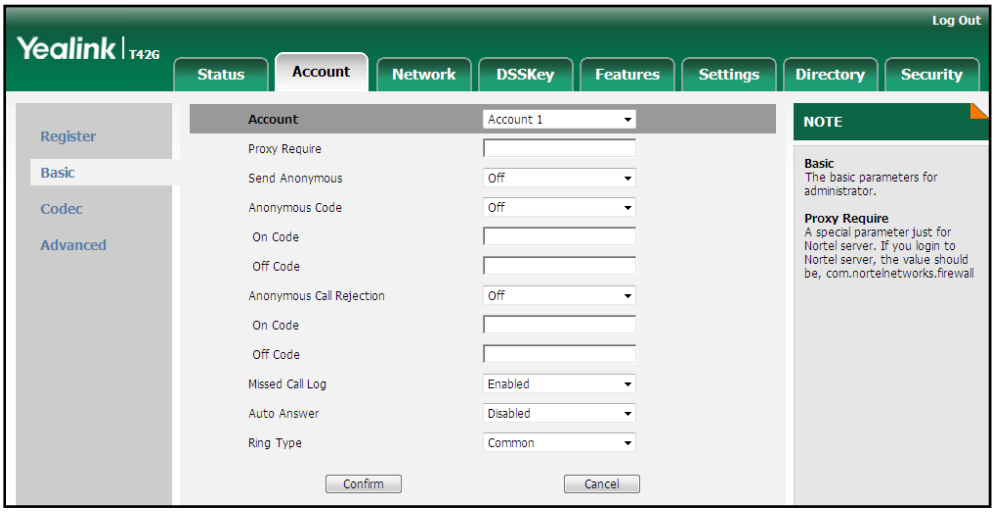
3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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. Refer to the above instruction.

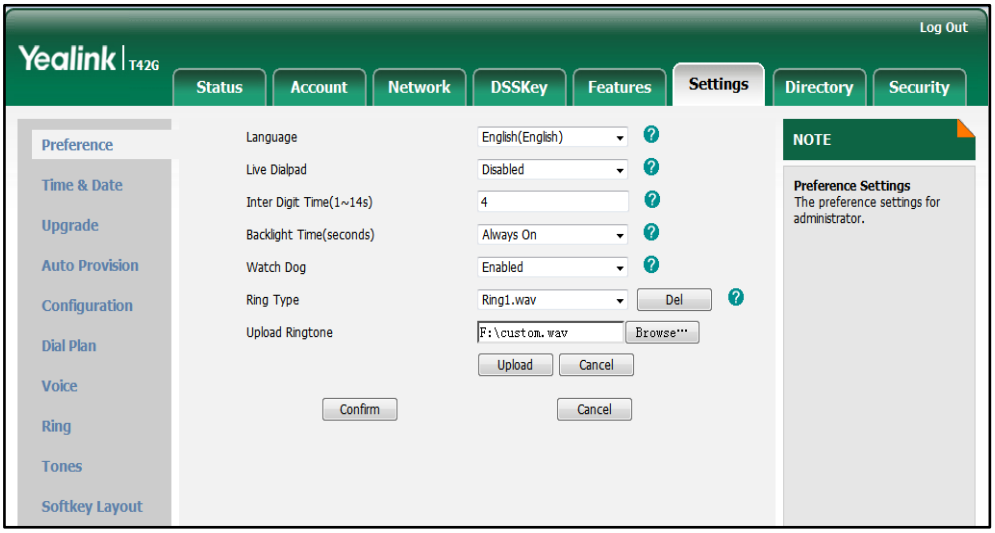


5. Click **Confirm** to accept the change.

Note A ring tone for the account is configurable 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 file (the file format must be *.wav) 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 in the local directory (refer to Adding Contacts). If no ring tone is assigned to the contact, the phone will play the ring tone assigned to the associated group (refer to Adding Groups). Otherwise, the phone will play the ring tone assigned to the account. If no ring tone is assigned to the contact and account, the phone will play the ring tone assigned to the phone.

All custom ring tone files must be within 100KB. Uploading custom ring tones for your phone is configurable via web user interface only.

Contact Management

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

- Directory
- Local Directory
- Blacklist
- Remote Phone Book

Directory

Directory provides easy access to frequently used lists. The lists may contain Local Directory, History, Remote Phone Book and LDAP.

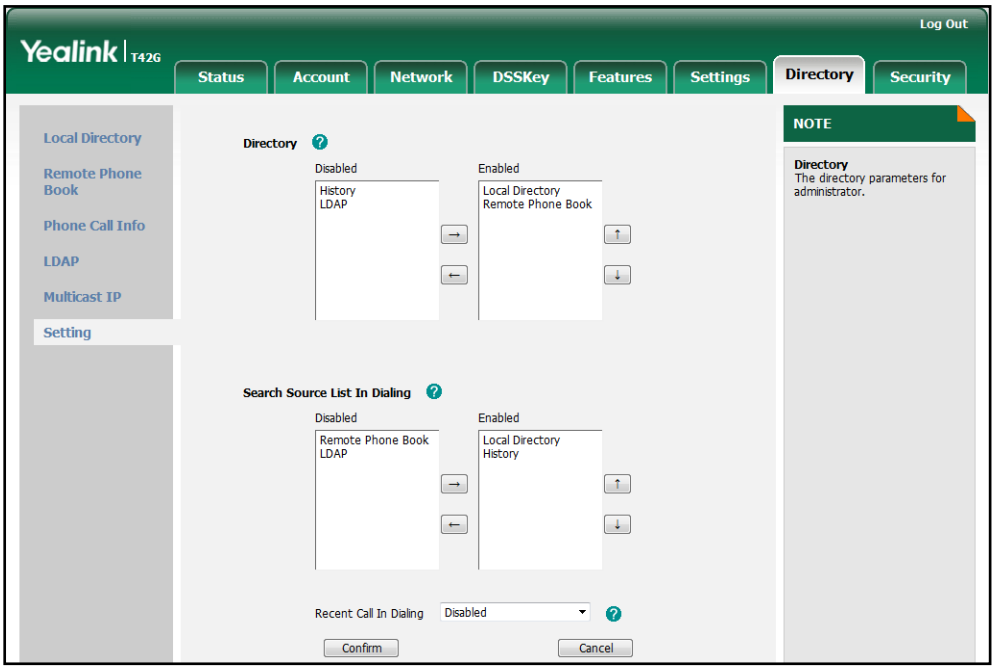
To configure the directory via web user interface:

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



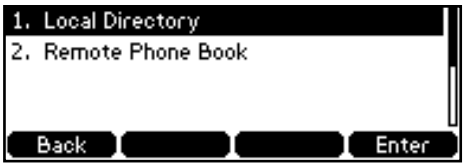
6. Click **Confirm** to accept the change.

Note Directory is configurable via web user interface only.

To check directory via phone user interface:

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

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



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

Note If remote phone book and LDAP are not configured in advance, you cannot see remote phone book and LDAP lists on the phone user interface. For more information on remote phone book, refer to Remote Phone Book on page 40. For more information on the LDAP, refer to *Yealink_SIP-T4X_IP_Phone_Family_Administrator_Guide*.

Local Directory

The built-in phone directory can store the names and phone numbers of your contacts. You can store up to 1000 contacts and 50 groups 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 contacts; edit, delete or search for a contact; or simply dial a contact number from the local directory.

Adding Groups

To add a group to the local directory:

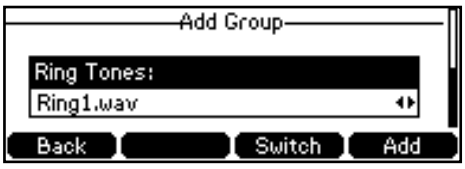
1. Press the **Directory** soft key.

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

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

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

   If **Auto** is selected, this group will use the ring tone assigned to the account. For more information on the ring tone for the account, refer to Ring Tones on page 27.

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:

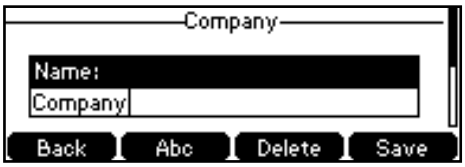
1. Press the **Directory** soft key.

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

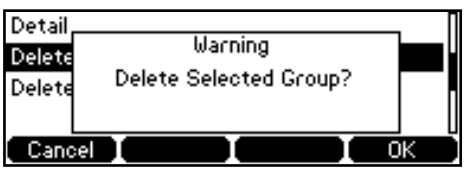
To delete a group from the local directory:

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 LCD screen prompts the following warning:



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

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

Adding Contacts

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

- Manually
- From call history
- From remote phone book

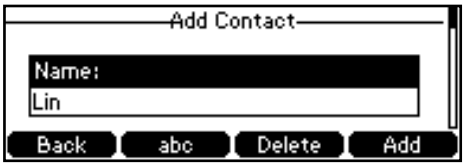
Adding Contacts Manually

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.

   If **Auto** is selected, the phone will play the ring tone assigned to the group when receiving incoming calls from this contact.
7. Press the **Add** soft key to accept the change or the **Back** soft key to cancel.

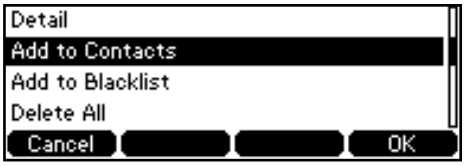
Note If the contact has existed in the directory, the LCD screen will prompt "Contact name existed!".

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 ▲ or ▼ to highlight the desired entry.

3. Press the **Option** soft key, and then select **Add 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 to the local directory.

Adding Contacts from Remote Phone Book

To add a contact to the local directory from the remote phone book:

1. Press **Directory**->**Remote Phone Book**.

   If Remote Phone Book is removed from the directory, press **Menu**->**Directory**->**Remote Phone Book** to enter remote phone book.
2. Select the desired remote group and press the **Enter** soft key.
3. Press ( ) or ( ) to highlight the desired entry.
4. Press the **Option** soft key, and then select **Add 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 phone book operating, refer to Remote Phone Book on page 40.

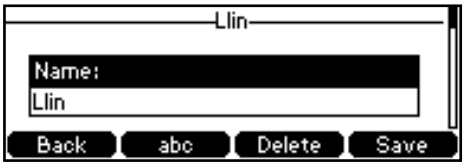
Editing Contacts

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 LCD screen prompts the following warning:

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

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

Placing Calls to Contacts

To place a call to 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. Do one of the following:

   - 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.

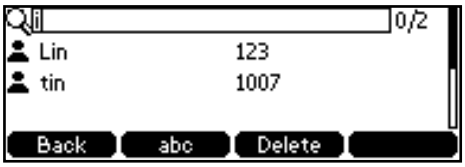
Searching for Contacts

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 the contact 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

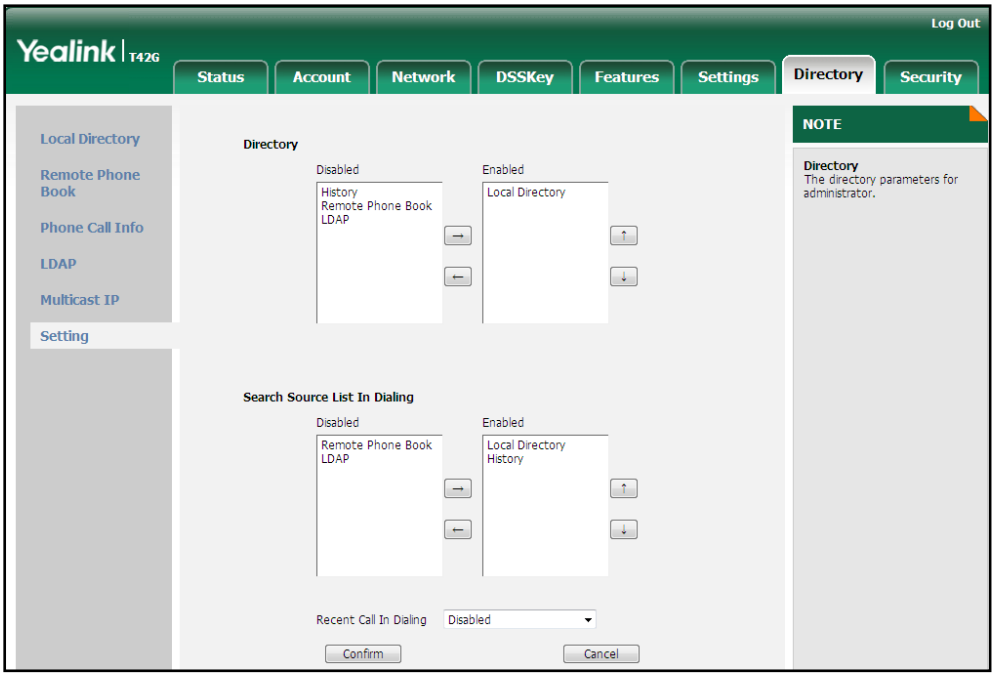
You can search for a contact from the desired lists when the phone is in the dialing screen. The lists can be Local Directory, History, Remote Phone Book and LDAP.

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 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 the enabled lists, select the desired list and click ↑ or ↓ .

The LCD screen displays the search results in the adjusted order.



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

Note Search source list in dialing is configurable via web user interface only.

To search for an entry 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 entry, and then place a call to the entry.

Importing/Exporting Contact Lists

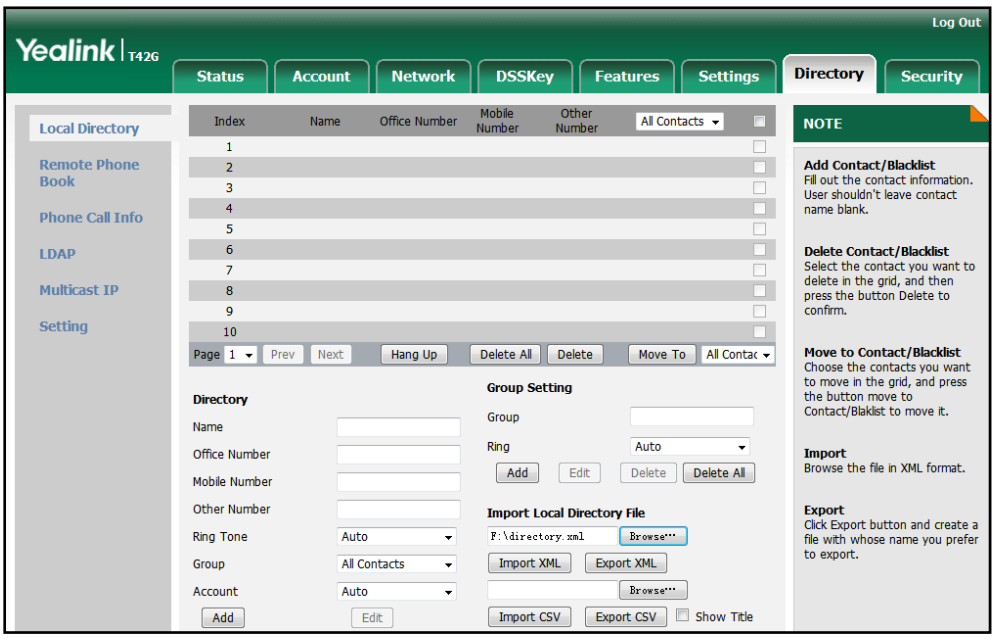
You can manage your phone's local directory via phone user interface or web user interface. But you can only import or export the contact list via web user interface.

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.

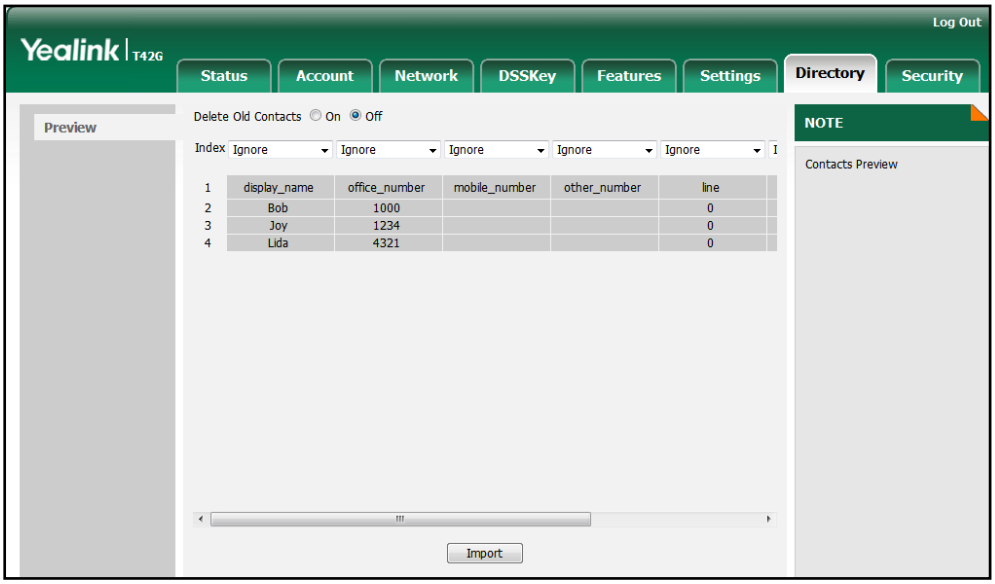
To import a CSV file of contact list via web user interface:

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.

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



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

To export a contact list via web user interface:

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

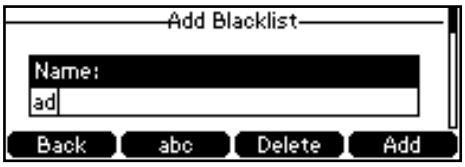
Note Importing/exporting contact lists is available via web user interface only.

Blacklist

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

To add a contact to the blacklist directory manually:

1. Press **Menu->Directory->Blacklist**.
2. Press the **Add** soft key.
3. 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 blacklist directory.

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

To add a contact to the blacklist directory 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 **Option** soft key and then select **Add to Blacklist** from the prompt list.

   The LCD screen prompts "Move selected to blacklist?".

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

For operating instructions on editing, deleting, placing calls to and/or searching for contacts in the blacklist, refer to the operating instructions of Local Directory on page 30.

Remote Phone Book

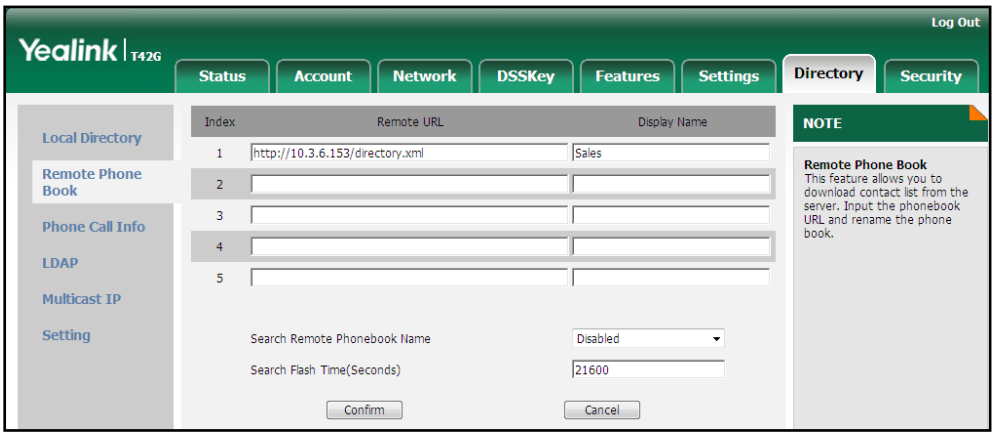
You can add local contacts, search for a contact, or simply dial a contact number from the corporate directory.

You can configure your new phone to access up to 5 corporate directories. The phone supports up to 5000 corporate directory entries. For the access URL of the corporate directory, contact your system administrator.

To configure an access URL for remote phone book 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.



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

Note An access URL for remote phone book is configurable via web user interface only.

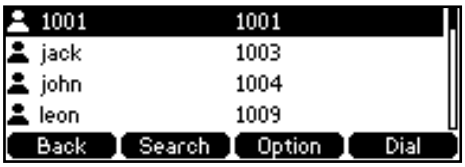
To access your corporate directory via phone user interface:

1. Press **Directory->Remote Phone Book**.

   If Remote Phone Book is removed from the directory, press **Menu->Directory->Remote Phone Book** to enter remote phone book.

2. Select the desired remote group, and then press the **Enter** soft key.

   The phone connects to load the corporate directory, and then the corporate contact list appears on the LCD screen.



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

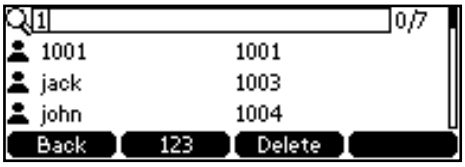
To search for a contact in the corporate directory:

1. Press **Menu->Directory->Remote Phone Book**.

   If Remote Phone Book is added to the directory, press **Directory->Remote Phone Book** to enter remote phone book.

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 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**.

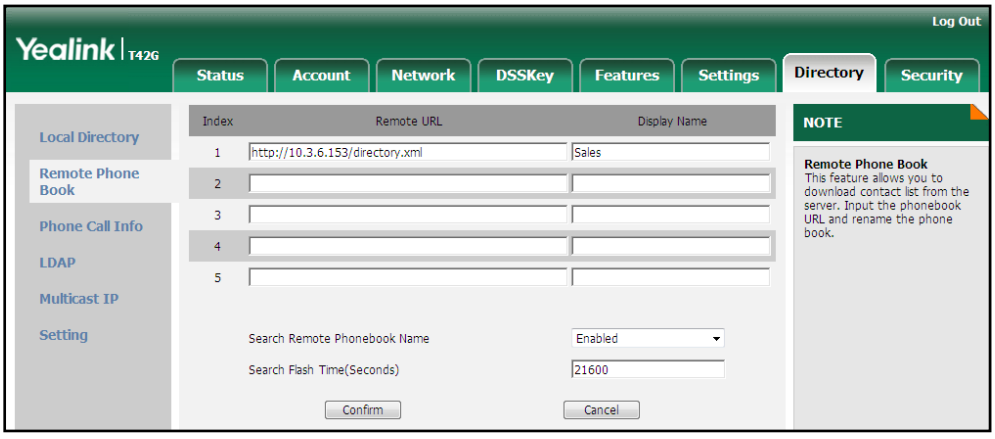
   If Remote Phone Book is added to the directory, press **Directory**->**Remote Phone Book** to enter remote phone book.
2. Select the desired remote group, and then press the **Enter** soft key to load the corporate directory.
3. Select the desired contact in the corporate directory.
4. Press the **Dial** 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 remote phone book via web user interface:

1. Click on **Directory**->**Remote Phone Book**.
2. Select **Enabled** from the pull-down list of **Search Remote Phonebook Name**.
3. Enter the refresh period in the **Search Flash Time (Seconds)** field.

   The default value is 21600 seconds.



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

Call History Management

The SIP-T42G IP phone maintains call history lists of Placed Calls, Received Calls, Missed Calls and Forwarded Calls. Call history lists support 400 entries in all. You can view the call history, place a call, add a contact or delete an entry from the call history list. You should enable history record in advance.

To enable history record 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 view the call history:

1. Press the **History** soft key.

   The LCD screen displays all call records.
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.

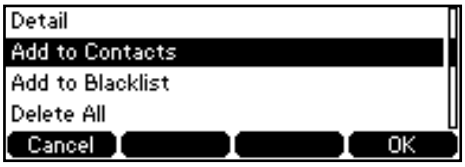
To place a call from the call history list:

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 (or blacklist 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 **Add 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 29.

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 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 press the **Send** soft key, then the phone will place a call using the headset automatically. For more information on using the headset to place a call, refer to Placing Calls on page 68.

To deactivate the headset mode:

1. Press again on the phone.

   The headset key LED goes out when the headset mode is deactivated.

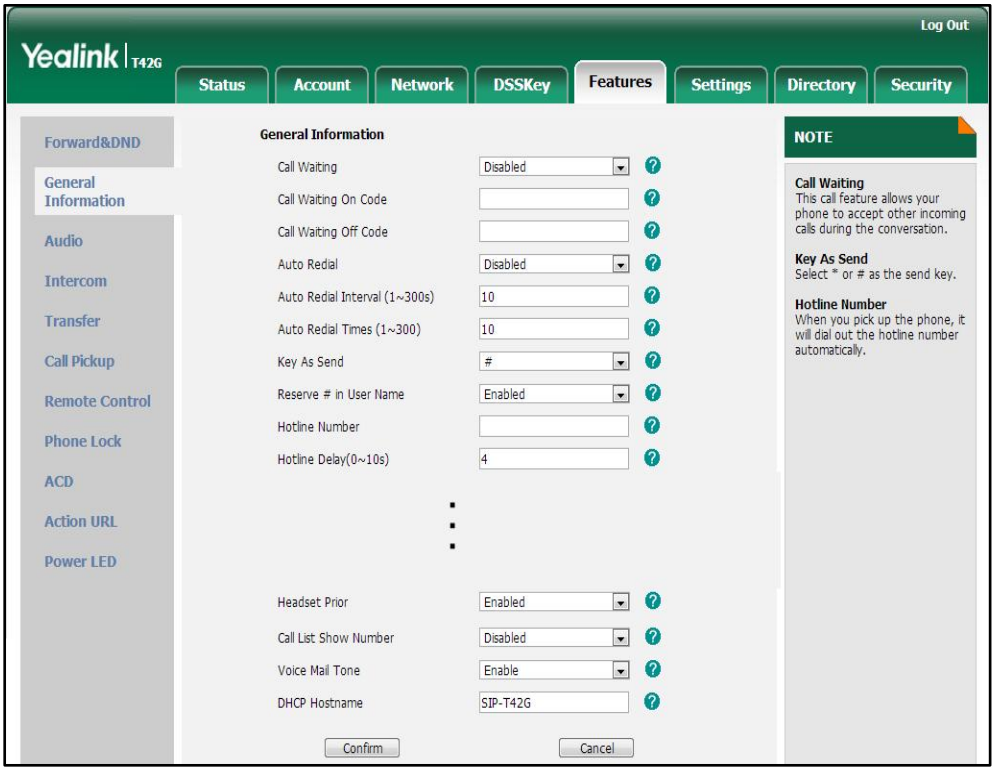
Headset Prior

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

To enable headset prior via web user interface:

1. Click on **Features**->**General Information**.

2. Select **Enabled** from the pull-down list of **Headset Prior**.



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

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

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

Note

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

Headset prior is configurable via web user interface only.

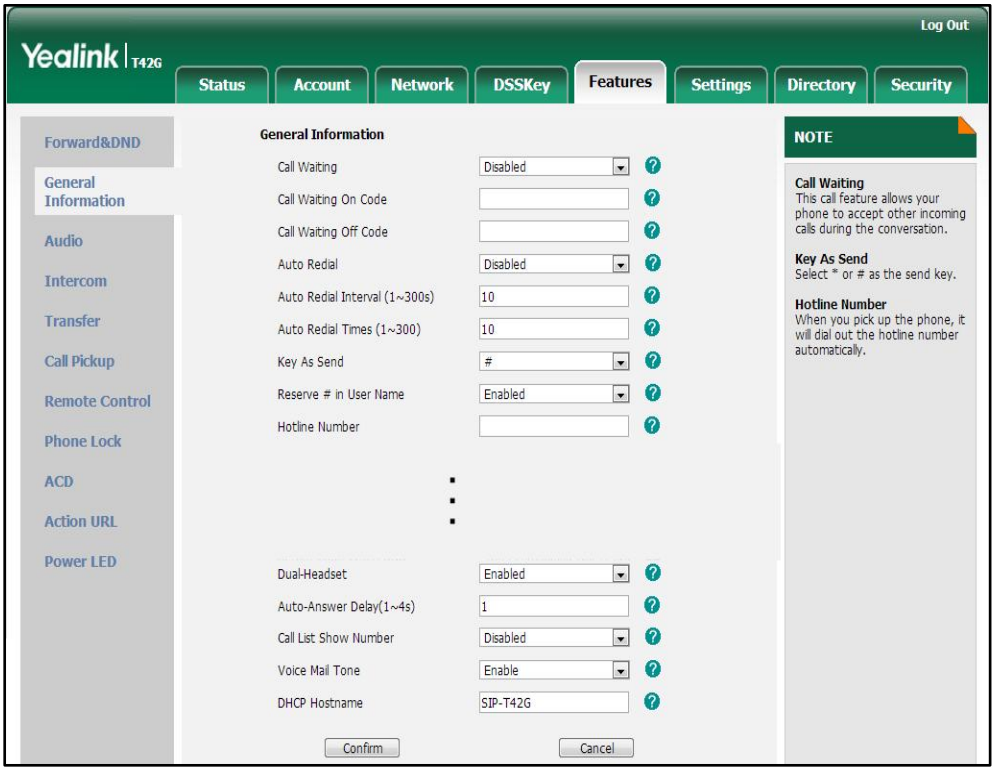
Dual Headset

You can use two headsets when enabling dual headset. To use this feature, you must physically connect headsets to the headset jack and handset jack respectively. Once the phone connects to a call, the headset connected to the headset jack will have full-duplex capabilities, while the one connected to the handset jack will only be able to listen.

To enable dual headset via web user interface:

1. Click on **Features**->**General Information**.

2. Select **Enabled** from the pull-down list of **Dual-Headset**.



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

Note Dual headset is configurable via web user interface only.

DSS Keys

Two types of DSS keys: Line Keys and Programmable Keys. Details will be introduced in the following. The SIP-T42G IP phone supports 15 line keys.

Line Keys

You can assign predefined functionalities to 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 quickly access features such as call return and voice mail. The line key LEDs will indicate the monitored status when line keys are assigned with particular 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 until configuration.

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.

Line key is configurable 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 indicated on three pages. When you assign functionality to line key 1-6, labels of line keys are indicated on the first page.

The following figure shows an example:

When you assign functionalities to line key 7-15, labels of the line key 6-10 are indicated on the second page, and labels of the line key 11-15 are indicated on the third page. The right bottom line key is used to switch pages.

The following figure shows an example:

You can configure the page icon and page switch key LED to indicate different statuses. This feature is disabled by default.

The following table lists the page icon and status:

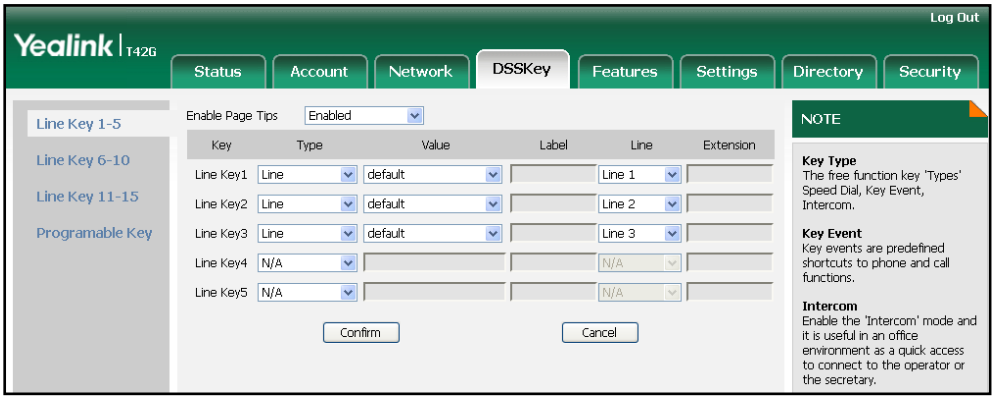
Icons/Status	Description
1	Indicates the current page of line keys.
2 3	Indicates the non-current page of line keys.
Fast flashing	The BLF monitored user receives an incoming call on the non-current page. The line on the non-current page receives an incoming call.

	There is a hold call on the line on the non-current page.

For more information on the page switch key LED status, refer to LED Instructions on page 4.

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

1. Click on **DSSKey**->**Line Key**.
2. Select **Enabled** from the pull-down list of **Enable Page Tips**.



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

Note The page icon status is configurable via web user interface only.

Line key features are explained in the following subchapters in detail:

- Account
- Speed Dial
- Voice Mail
- Directed Pickup
- Group Pickup
- DTMF
- Prefix
- Local Group
- XML Group
- XML Browser
- LDAP
- Conference
- Forward
- Transfer
- Hold

- DND
- Group Listening
- Zero Touch
- URL
- Keypad Lock
- Directory

For more information, contact your system administrator.

Account

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 (Account)***

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** from the **Value** field, the line keys located in the same position of the other page will be automatically assigned the line key 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** from the **Value** field, the phone will automatically switch to the page the line key 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 a voice mail quickly. For more information, refer to Voice Mail on page 111.

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 (Directed Pickup)

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

Label (key label displayed on the LCD screen)

Value (directed 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, then the phone will enter into the dialing screen and display the prefix number which you specified in the **Value** field. You can 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 30.

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 remote phone book in advance. For more information, refer to Remote Phone Book on page 40.

Dependencies: ***Type (Key Event)***

Key Type (XML Group)

PhoneBook Name *(the remote group name you want to access if the remote phone book 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.

XML Browser

You can use this key feature to access the XML browser quickly. The XML browser allows you to create custom services which meet your functional requirements on the server. You can customize practical applications, such as weather report, stock information, Google search, etc.

Dependencies: ***Type (Key Event)***

Key Type (XML Browser)

Label *(key label displayed on the LCD screen)*

Value *(the access URL for XML browser)*

Usage: Press the DSS key to access the XML browser specified in the **Value** 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 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 87.

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.

Note When leaving the **Value** field blank, the DSS key performs the same as the **Conf** soft key during a call.

Forward

You can use this key feature to forward an incoming call to someone else. For more information, refer to Call Forward on page 80.

Dependencies: ***Type (Key Event)***

Key Type (Forward)

Label *(key label displayed on the LCD screen)*

Value *(the number you want to forward to)*

Usage:

1. Press the DSS key to forward an incoming call to the number specified in the **Value** field.
2. When call forward is in phone mode, press the DSS key to activate call forward for

the phone.

3. When call forward is in custom mode, press the DSS key to access the custom call forward interface. You can activate or deactivate call forward for one or all accounts.

Note If the **Value** field is left blank, the DSS key performs the same as the **Fwd** soft key when receiving an incoming call.

When call forward is activated, the forward key LED will illuminate solid green.

Transfer

You can use this key feature to handle the call differently depending on the transfer mode via 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 via 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 via DSS key is **Attended Transfer**, press the DSS key to dial out the number specified in the **Value** field, and then perform the attended or semi-attended transfer.
- When the transfer mode via DSS key is **New Call**, press the DSS key to place a new call to the number specified in the **Value** field.

Note Transfer mode via DSS key is configurable via web user interface at the path **Features->Transfer->Transfer Mode Via Dsskey**.

When leaving the **Value** field blank, the DSS key performs the same as the **Tran** soft key during a call.

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. You can also use this key feature to access the custom DND interface. For more information, refer to Do Not Disturb (DND) on page 77.

Dependencies: ***Type (Key Event)***

Key Type (DND)

Label *(key label displayed on the LCD screen)*

Usage:

When DND is in phone mode:

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

When DND is in custom mode:

1. Press the DSS key to access the custom DND interface. You can activate or deactivate DND for one or all accounts.

Note When DND is activated, the DND key LED will illuminate solid green, and 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 an HTTP GET request containing a specific URL.

Dependencies: ***Type (URL)***

Label *(key label displayed on the LCD screen)*

Value *(the URL contained in the HTTP GET request)*

Usage: Press the DSS key to trigger the phone to send an HTTP GET request containing the URL specified in the **Value** field.

Keypad Lock

You can use this key feature to immediately lock the keypad of your phone instead of long pressing #SEND. For more information, refer to Keypad Lock on page 23.

Dependencies: ***Type (Key Event)***

Key Type (Keypad Lock)

Label *(key label displayed on the LCD screen)*

Usage: Press the DSS key to immediately lock the keypad of your phone instead of long pressing #SEND.

Directory

You can use this key feature to easily access frequently used lists. For more information, refer to Directory on page 29.

Dependencies: ***Type (Key Event)***

Key Type (Directory)

Label *(key label displayed on the LCD screen)*

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

Note The DSS key performs the same function as the **Directory** soft key when the phone is idle.

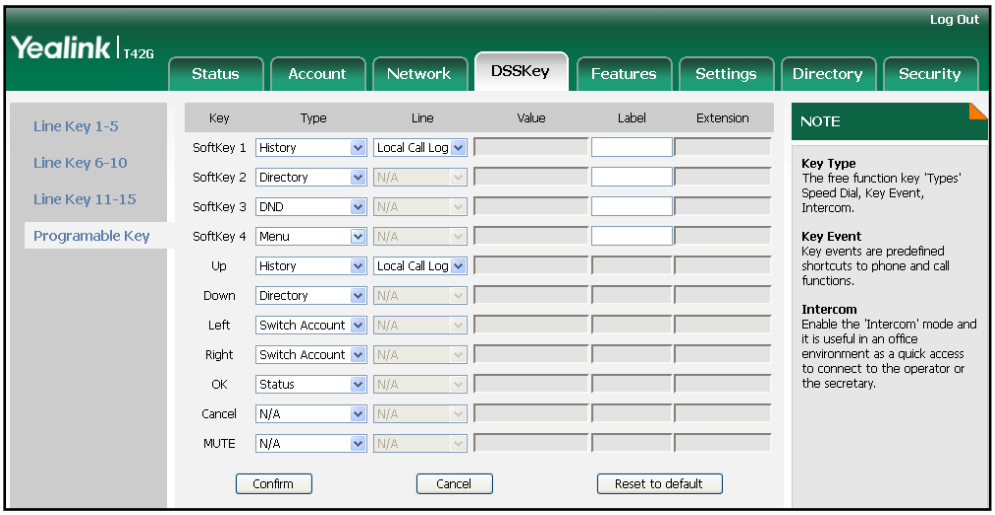
Programmable Keys

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.



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

You can click **Reset to default** to reset custom settings to defaults.

Note Programmable keys are configurable via web user interface only.

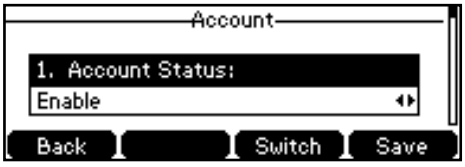
Account Management

You can register one or multiple accounts on the SIP-T42G IP phone. You can also configure each line key to associate with an account or configure multiple line keys to associate 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 **Account Status** field.



4. Enter the desired values in **Label**, **Display Name**, **Register Name**, **User Name**, **Password** and **SIP Server1/2** 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.

You can repeat steps 2 to 5 to register more 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 **Account Status** field.
4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Registering an account is configurable via web user interface at the path **Account**->**Register**.

Multiple Line Keys per Account

You can configure multiple line keys to associate 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 account will be distributed evenly among the available line keys. Outgoing calls will be distributed similarly.

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

Dial Plan

Dial plan is a string of characters that governs the way your SIP-T42G IP phone processes the inputs received from your phone keypad. The SIP-T42G IP phone supports the following dial plan features:

- Replace Rule
- Dial-now
- Area Code
- Block Out

Basic expression syntaxes you need to know:

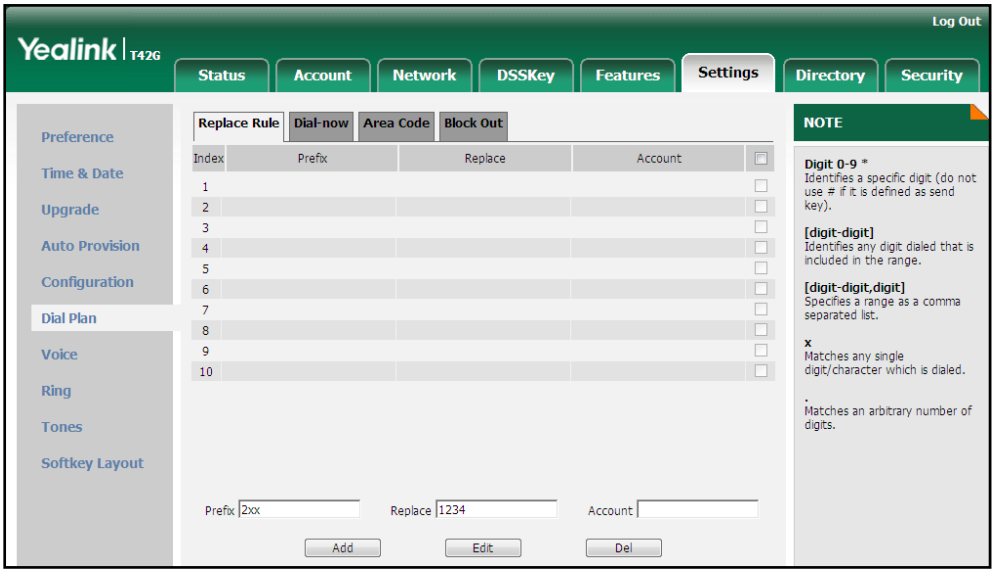
Syntax	Description
.	The dot "." can be used as a placeholder or multiple placeholders for any character. Example: "12." would match "123", "1234", "12345", "12abc", etc.
x	An "x" can be used as a placeholder for any character. Example: "12x" would match "121", "122", "123", "12a", etc.
[]	The square 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 "91**5**1234", "91**6**1234", "91**7**1234", 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 "**92**3", "**15**3", "**77**3",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”.

Replace Rule

You can configure one or more replace rules (up to 100) 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., 1234) in the **Replace** field.
4. Enter the desired line ID in the **Account** field or leave it blank.



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

   When you enter the number "234" using the keypad and then press the **Send** soft key, the phone will dial out "1234" instead.

Note

Valid values of **Account** field can be one or more digits among 1, 2, and 3. Every two digits must be separated by a comma. For example, when you enter the value "1, 2" in the **Account** field, this replace rule will apply to account 1 and account 2.

If you leave the **Account** field blank or enter 0, 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 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 accept the change.

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).

Note Replace rule is configurable via web user interface only.

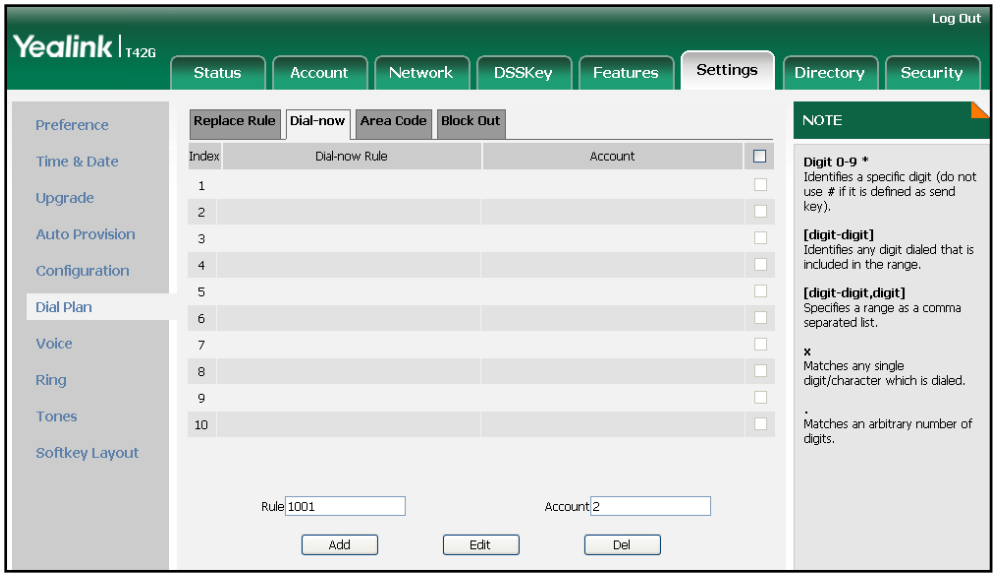
Dial-now

You can configure one or more dial-now rules (up 100) on your phone. When the dialed number matches the dial-now string, the number will be dialed out automatically. For example, a dial-now rule is configured as "2xx", then any entered three-digit string beginning 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., 1001) in the **Rule** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

   For more information on the valid value of the **Account** field, refer to Replace Rule on page 59.



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

   When you enter the number “1001” using the keypad, the phone will dial out “1001” automatically without pressing any key.

Note You can also edit or delete the dial-now rule, refer to Replace Rule on page 59 for more information.

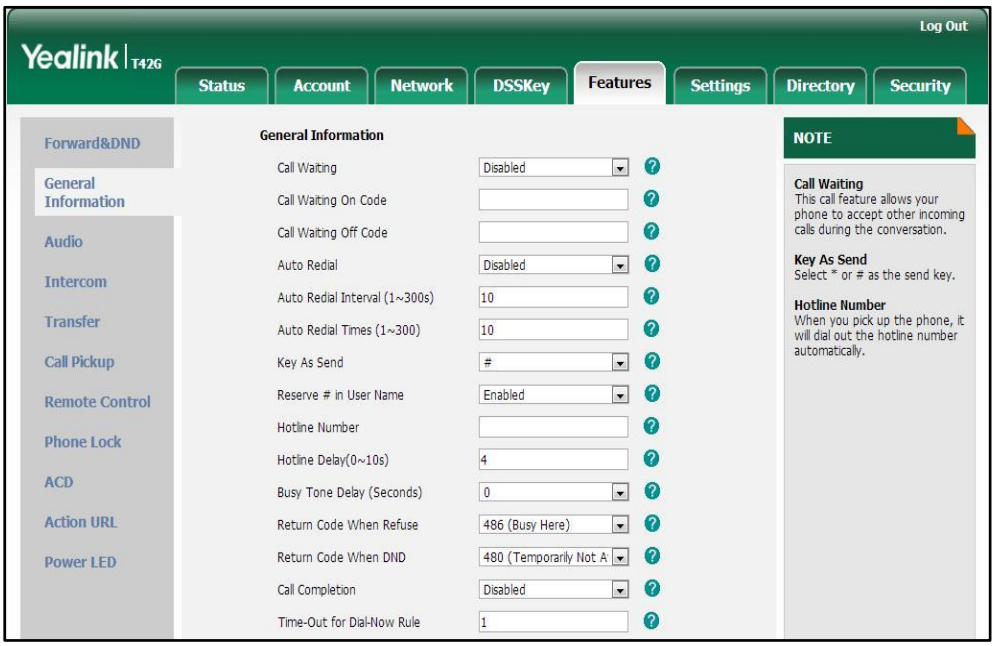
Dial-now rule is configurable via web user interface only.

Delay Time 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 period of time.

To configure the delay time 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.



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

Note Delay time for dial-now rule is configurable via web user interface only.

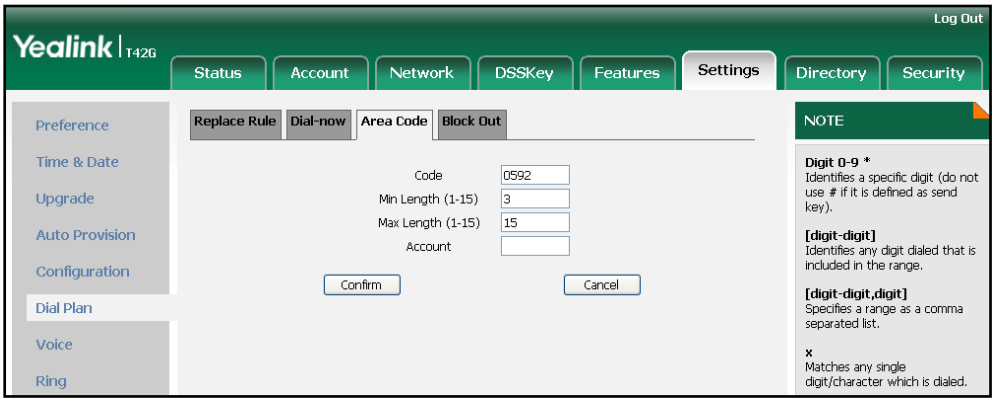
Area Code

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 on the valid value of the **Account** field, refer to Replace Rule on page 59.



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

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

Area code is configurable via web user interface only.

Block Out

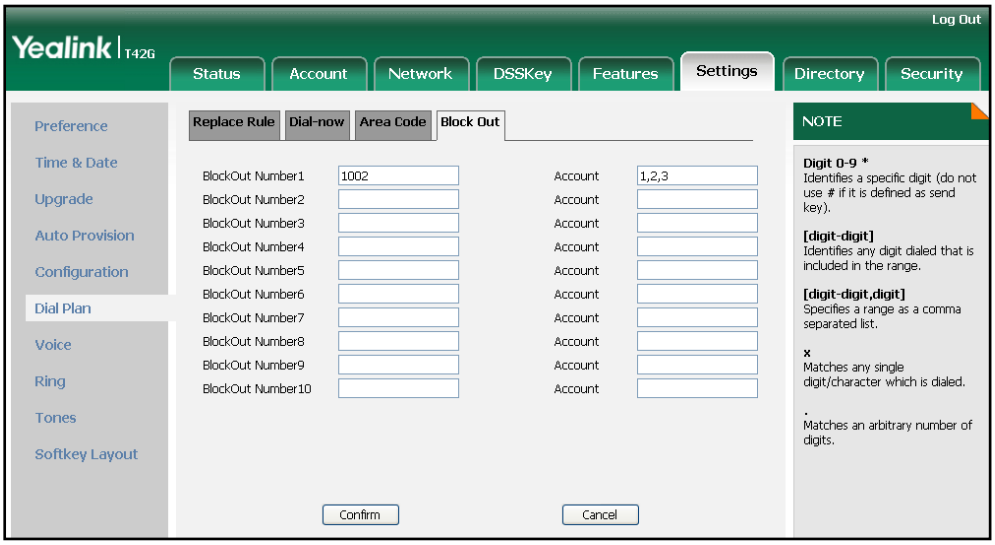
You can block specific numbers (up 10) from being dialed on your phone. When you dial a block out number on your phone, the dialing will fail and the LCD screen will prompt "Forbidden Number".

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 on the valid value of the **Account** field, refer to Replace Rule on page 59.



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

Note Block out number is configurable via web user interface only.

Emergency Number

Public telephone networks in countries around the world have a single emergency telephone number (emergency services number), that allows a caller to contact local emergency services for assistance when necessary. The emergency telephone number may differ from country to country. It is typically a three-digit number so that it can be easily remembered and dialed quickly. Some countries have a different emergency number for each of the different emergency services.

You can specify the emergency telephone numbers on the IP phone for contacting the emergency services in an emergency situation.

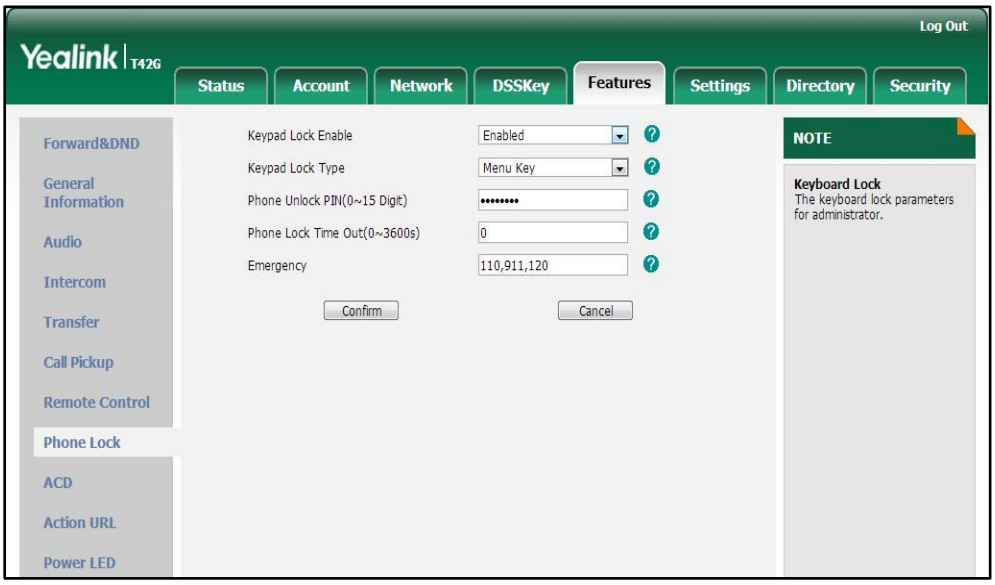
Note Contact your local phone service provider for available emergency numbers in your area.

To specify emergency numbers via web user interface:

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

2. Enter the emergency number in the **Emergency** field.

   For multiple emergency numbers, enter a comma between every two numbers. The default emergency numbers are 110, 911 and 120.



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

Note Emergency number is configurable via web user interface only.

Live Dialpad

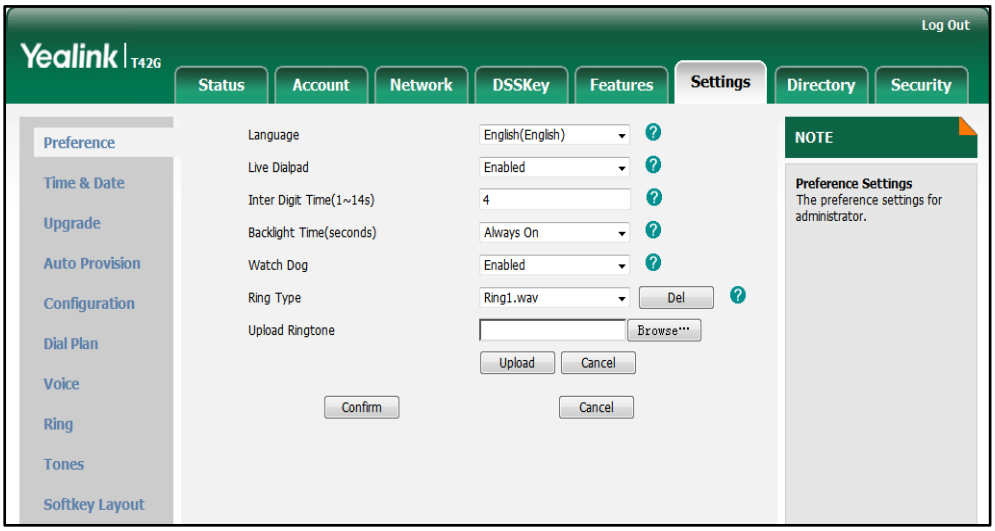
You can enable live dialpad on the SIP-T42G IP phone, which enables the IP phone to automatically dial out the phone number without pressing any other key. You can also configure a delay, where the phone will dial out the phone number automatically after the specified period of time.

To enable 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 the **Inter Digit Time (1~14s)** field.

   The default delay time is 4s.



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

Note Live dialpad is configurable via web user interface only.

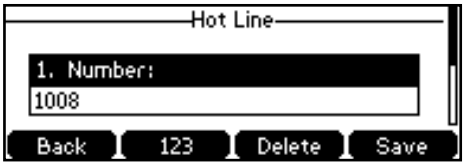
Hotline

You can dial a hotline number immediately upon lifting the handset, pressing the speakerphone key or the line key. You can also configure a delay, where the phone will dial out the hotline number automatically after the specified period of time.

To configure the hotline number via phone user interface:

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

   The valid values range from 0 to10.



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

Hotline is configurable via web user interface at the path **Features**->**General Information**.

Basic Call Features

The SIP-T42G 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-T42G IP phone. Topics include:

- Placing Calls
- Answering Calls
- Ending Calls
- Redialing Numbers
- Recent Call In Dialing
- Auto Answer
- Auto Redial
- Call Completion
- 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-T42G 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 also search and dial a contact from call history, local directory or remote phone book. For more information, refer to Contact Management on page 29 and Call History Management on page 43.

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 OK , #SEND , or the **Send** soft key.

The # key is configured as a send key by default. You can also set the * key as the send key, or set neither. For more information, refer to the Key as Send on page 23.

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 length of 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 OK , #SEND 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 45.

To place multiple calls:

You can have more than one call on your SIP-T42G 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 **New Call** soft key.

  Enter the desired number using the keypad.

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

You can press or to switch between 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 77.

You can forward incoming calls to someone else by pressing the **Fwd** soft key. For more information, refer to Call Forward on page 80.

Answering When Not in Another Call

Call duration and destination will always appear on the LCD screen for the active call.

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 (OK) 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.

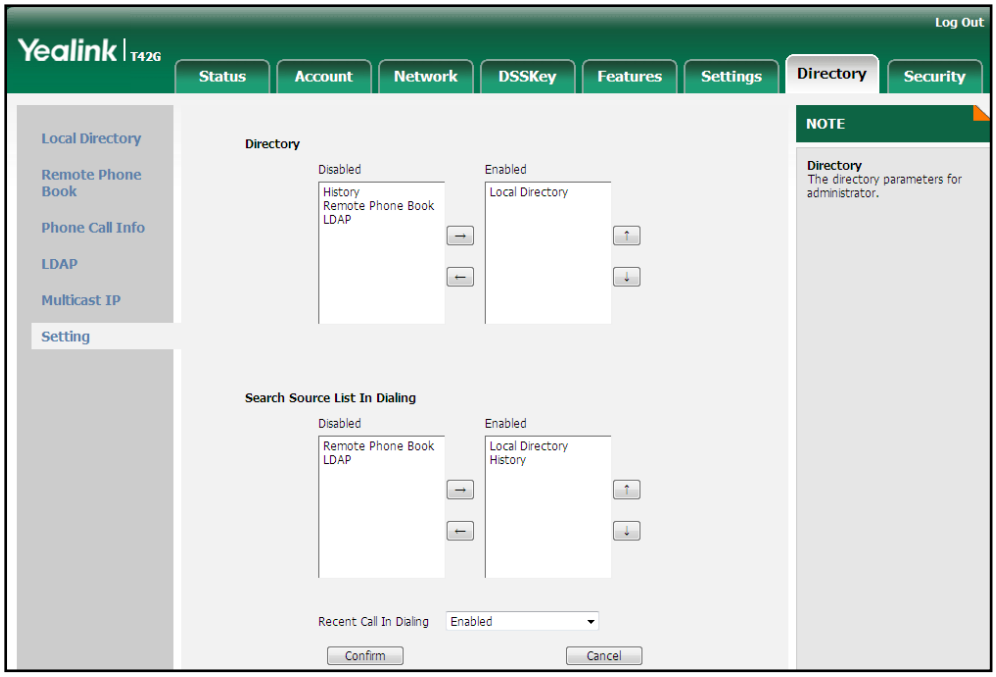
Recent Call In Dialing

To view the placed calls list when the phone is in the pre-dialing interface, you should enable recent call in dialing in advance.

To enable recent call in dialing via web user interface:

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

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



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

Note Recent call in dialing is configurable via web user interface only.

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

1. Pick up the handset, press the speakerphone or press the line key.

   The LCD screen displays the placed calls list.

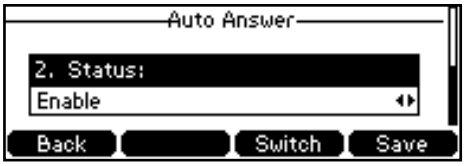
Auto Answer

You can use auto answer to automatically answer an incoming call on a line. Auto answer is configurable on a per-line basis.

To configure auto answer via phone user interface:

1. Press **Menu->Features->Auto Answer**.
2. Press ◄ or ► , or the **Switch** soft key to select the desired line from the **Account** field.

3. Press ◄ or ► , or the **Switch** soft key to select **Enable** from the **Status** field.



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

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

Auto answer is configurable via web user interface at the path **Account->Basic**.

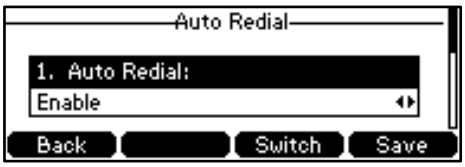
Note Auto answer is only applicable when there is no other call in progress on the phone.

Auto Redial

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

To configure auto redial via phone user interface:

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



3. Enter the desired time (in seconds) in the **Redial Interval** field.

   The default time interval is 10 seconds.

4. Enter the desired times in the **Redial Times** field.

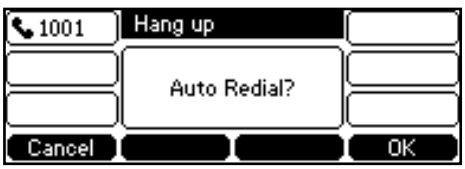
   The default value is 10.

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

Auto redial is configurable via web user interface at the path **Features->General Information**.

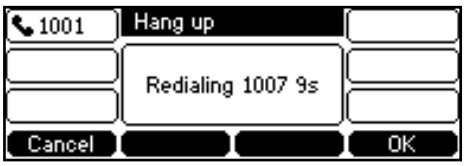
To use auto redial:

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



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

   The LCD screen prompts the following:



2. Wait for a period of time or press the **OK** soft key to redial the phone number.

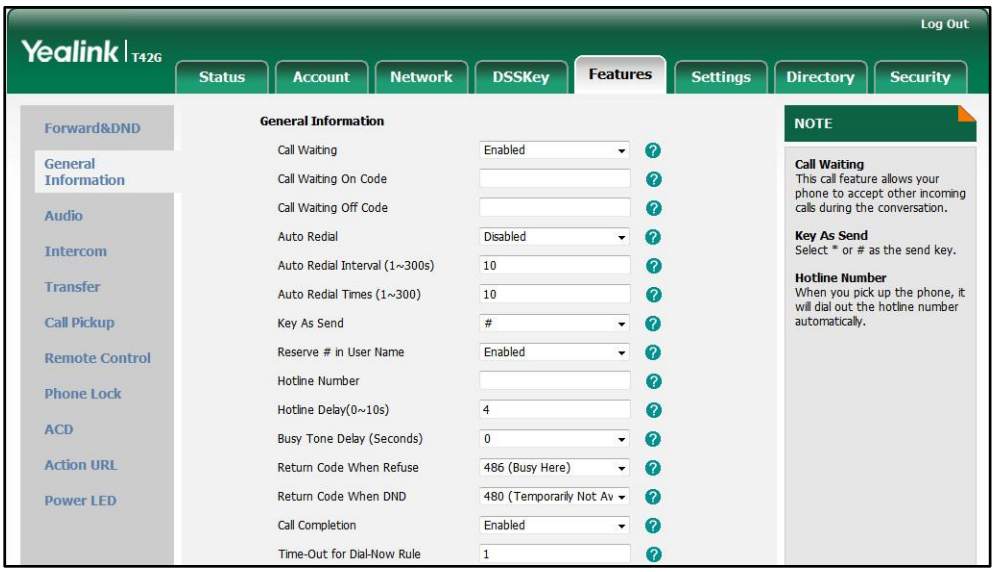
The phone will retry as many times as configured until the called party is idle.

Call Completion

You can use call completion to notify the caller who failed to reach a desired callee when the callee becomes available to receive a call.

To configure call completion via web user interface:

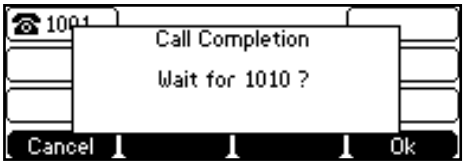
1. Click on **Features->General Information**.
2. Select **Enable** from the pull-down list of **Call Completion**.



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

To use call completion:

When the called party is busy, the following prompt will appear on the LCD screen of the phone:



1. Press the **OK** soft key, the phone returns to the idle screen and call completion is activated.

   When the called party becomes idle, the following prompt appears on the LCD screen of the phone:

2. Press the **OK** soft key to redial the number.

Note Call completion is not available on all servers. For more information, contact your system administrator.

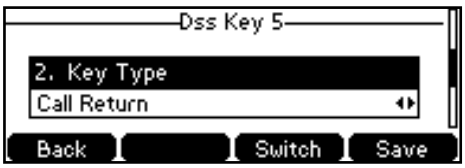
Call completion is configurable via web user interface only.

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.

Call return key is configurable 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 mute key LED illuminates solid red.

   The 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 goes out.

Call Hold/Resume

You can place an active call on hold. Only one active call can be in progress at any time. Other calls can be made and received while placing the original call on hold. When you place a call on hold, your IP PBX may play music on hold to the other party while waiting.

To place a call on hold:

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

   The line key LED flashes green. The LCD screen indicates that the call is on hold.

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, do one of the following:

- Press ▲ or ▼ to switch between the calls, and then press the **Resume** soft key to retrieve the desired call.
- Press the corresponding line key to retrieve the call.

If more than one call is placed on hold, a numbered prompt appears on the LCD screen, for example "2/3", indicating that this is the second call out of three calls.

Do Not Disturb (DND)

You can use DND to reject incoming calls automatically on the phone. Callers will receive a busy message.

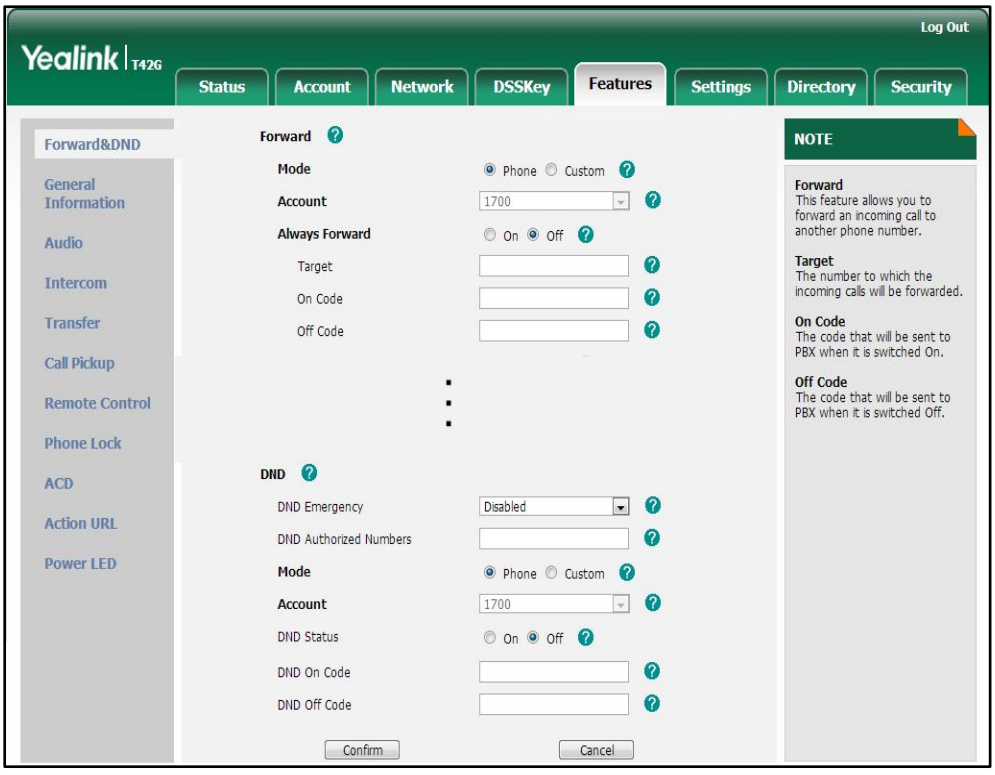
You can enable/disable DND for the phone system, or you can customize DND for each or all accounts. Two DND modes:

- **Phone** (default): DND is effective for the phone system.
- **Custom**: DND can be configured for each or all accounts

You can receive incoming calls from authorized numbers when DND 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.



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

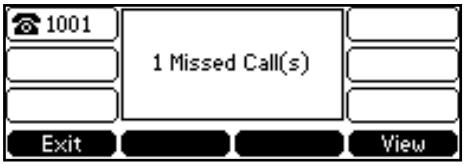
Note DND mode is configurable via web user interface only.

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 DND is enabled.

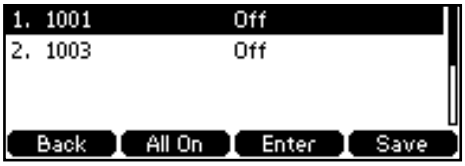
   Incoming calls will be rejected automatically and "**n Missed Call(s)**" ("n" indicates the number of the missed calls) will prompt on the LCD screen.



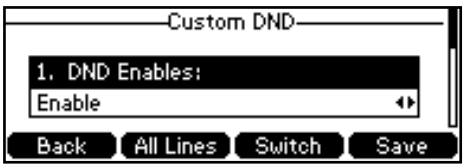
To activate DND in custom mode for a specific account:

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

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



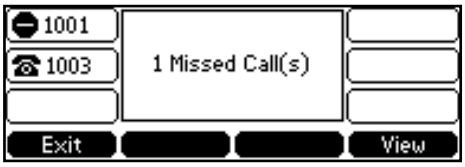
2. Press ▲ or ▼ to select the desired account and then press the **Enter** soft key.
3. Press ◄ or ► , or the **Switch** soft key to select **Enable** from the **DND Enables** field.



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 ⊖ .

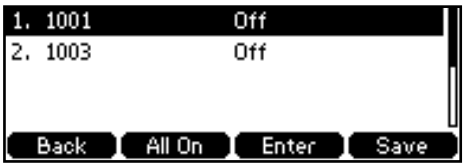
   Incoming calls on the specific line will be rejected automatically, and the LCD screen prompts "**n Missed Call(s)**" ("n" indicates the number of the missed calls).



To activate DND in custom mode for all accounts:

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

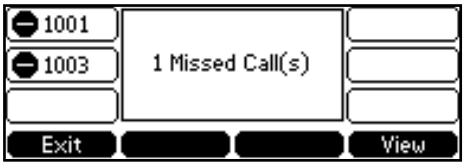
   The LCD screen displays a list of accounts registered on the phone.



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 ⛔ .

   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 Missed Call Log for the line is enabled. Missed call log is configurable via web user interface at the path **Account->Basic**.

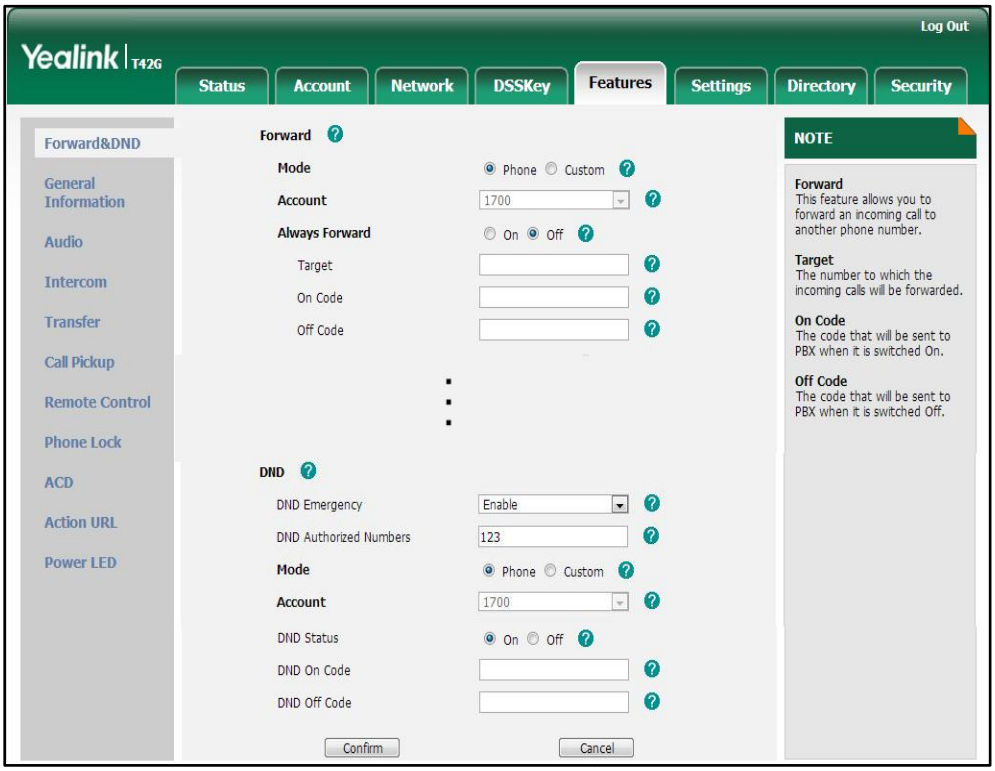
Do not disturb is local to the phone, and may be overridden by the server settings. For more information, contact your system administrator.

To configure the DND authorized numbers via web user interface:

1. Click on **Features**->**Forward & DND**.
2. Select **Enable** 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 every two numbers.



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

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

Note DND authorized number is configurable via web user interface only.

Call Forward

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

Static Forwarding

Three types of static forwarding:

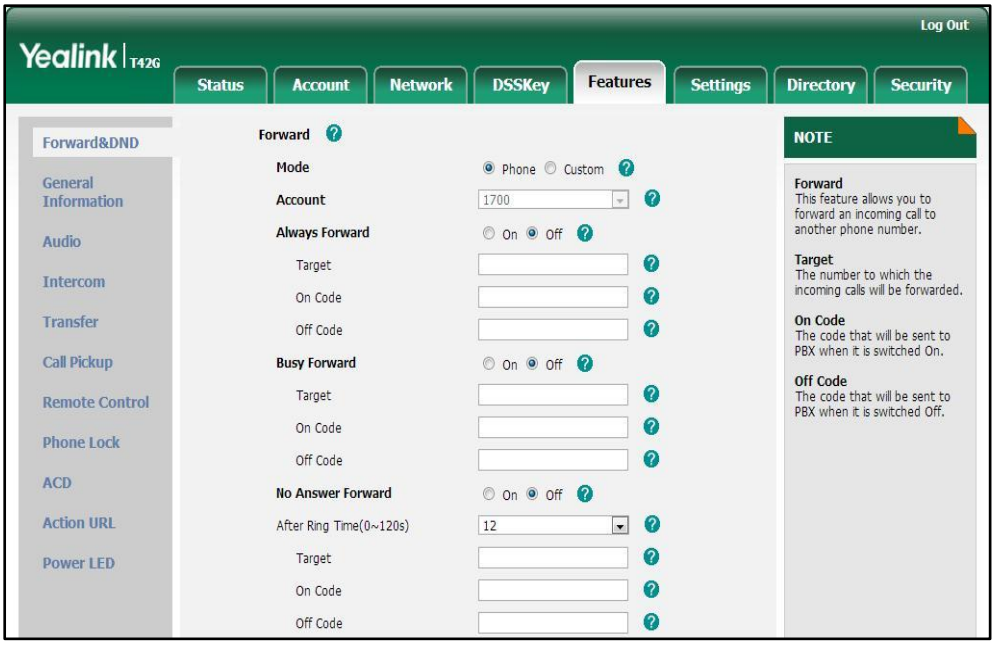
- **Always Forward**: Incoming calls are immediately forwarded.
- **Busy Forward**: Incoming calls are immediately forwarded if the phone is busy.
- **No Answer Forward**: Incoming calls are forwarded if not answered after a period of time.

You can enable/disable call forward for the phone, or you can customize call forward for each or all accounts. Two call forward modes:

- **Phone** (default)**:** Call forward is effective for the phone system.
- **Custom:** Call forward can be configured for each 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.



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

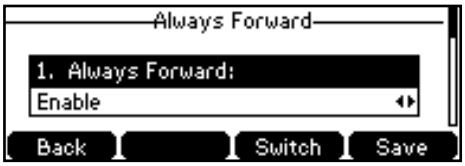
Note Call forward mode is configurable via web user interface only.

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 Forward** field.

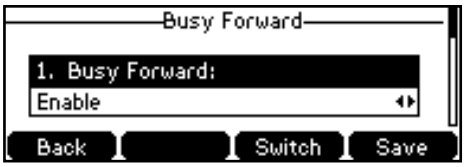


2) Enter the destination number you want to forward all incoming calls to in the **Forward to** field.

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 Forward** field.

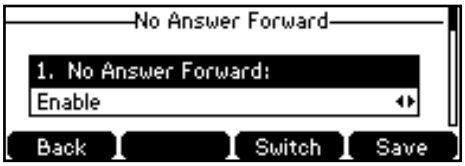


2) Enter the destination number you want to forward all incoming calls to when the phone is busy in the **Forward to** field.

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 Forward** 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 from the **After Ring Time** field.

The default ring time is 12 seconds.

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.

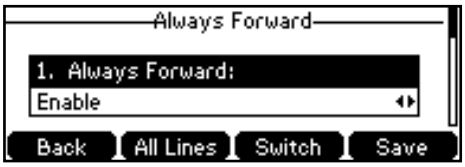
To enable call forward in custom mode:

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 Forward** field.



2) Enter the destination number you want to forward all incoming calls to in the **Forward to** field.

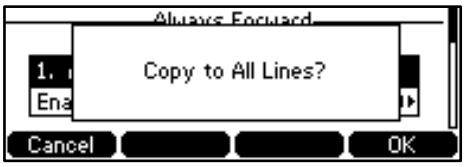
3) (Optional.) Enter the always forward on code or off code respectively in the **On Code** or **Off Code** field.

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

1) Press ▲ or ▼ to highlight the **Always Forward** field.

2) Press the **All Lines** soft key.

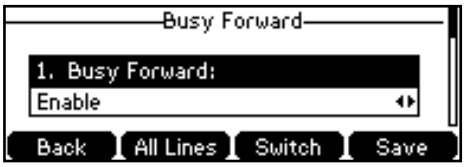
The LCD screen prompts "Copy to All Lines?".



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 Forward** field.



2) Enter the destination number you want to forward incoming calls to when the phone is busy in the **Forward to** field.

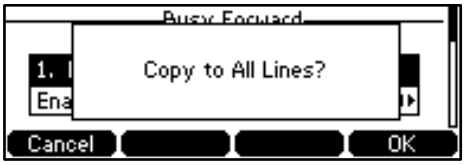
3) (Optional.) Enter the busy forward on code or off code respectively in the **On Code** or **Off Code** field.

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

1) Press ▲ or ▼ to highlight the **Busy Forward** field.

2) Press the **All Lines** soft key.

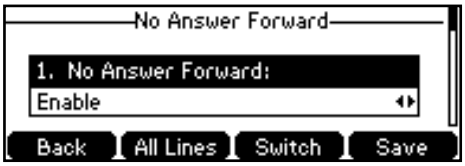
The LCD screen prompts "Copy to All Lines?".



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 Forward** 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 from the **After Ring Time** field.

The default ring time is 12 seconds.

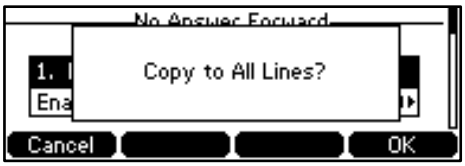
4) (Optional.) Enter the no answer forward on code or off code respectively in the **On Code** or **Off Code** field.

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

1) Press ▲ or ▼ to highlight the **No Answer Forward** field.

2) Press the **All Lines** soft key.

The 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 ↱ .

Call forward is configurable 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 68.

Call forward 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, and 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, and then press the **Enter** soft key.
3. Press or to select the desired forwarding type, and 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.
2. Enter the number you want to forward the incoming call to.

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

   The 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:

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

2. Enter the number you want to transfer the call to.

3. 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:

1. Press the **Tran** soft key during a call.
2. Do one of the following:
   - Enter the number you want to transfer the call to.
   - Press the **Directory** soft key, and then select **Local Directory**. Select the desired group and search for the contact (Directory should be configured in advance. Refer to Directory on page 29 for more information).
   - Press the **Directory** soft key, and then select **History**. Select the desired list and press ▲ or ▼ to select the entry (Directory should be configured in advance. Refer to Directory on page 29 for more information).
   - Press the **Directory** soft key, and then select **Remote Phone Book**. Select the desired group and search for the contact (Directory and Remote Phone Book should be configured in advance. Refer to Directory on page 29 and Remote Phone Book on page 40 for more information).
3. Press OK or # to dial out.
4. Press the **Tran** soft key to complete the transfer when receiving ringback.

To perform an attended transfer:

1. Press the **Tran** soft key during a call.
2. Do one of the following:
   - Enter the number you want to transfer the call to.
   - Press the **Directory** soft key, and then select **Local Directory**. Select the desired group and search for the contact (Directory and Remote Phone Book should be configured in advance. Refer to Directory on page 29 for more information).
   - Press the **Directory** soft key, and then select **History**. Select the desired list and press ▲ or ▼ to select the entry (Directory should be configured in advance. Refer to Directory on page 29 for more information).
   - Press the **Directory** soft key, and then select **Remote Phone Book**. Select the desired group and search for the contact (Directory and Remote Phone Book should be configured in advance. Refer to Directory on page 29 and Remote

Phone Book on page 40 for more information).

3. Enter the number you want to transfer the call to.
4. Press OK or #SEND to dial out.
5. 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.

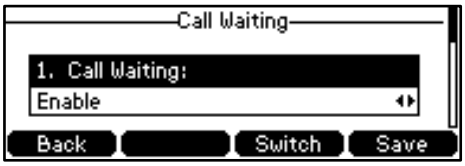
You can cancel the transfer before the call is connected by pressing the **Cancel** soft key.

Call Waiting

You can enable or disable call waiting on the phone. If call waiting 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.

Call waiting is configurable 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. You can create a conference between an active call and a call on hold by pressing the Conf soft key. The SIP-T42G IP phone also supports network conference.

Note Network conference is not available on all servers. For more information, contact your system administrator.

Local Conference

The SIP-T42G IP phone supports up to 3 parties (including yourself) in a conference call. This is the default method of conference called Local 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 OK , #SEND , 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 two 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 from speaking.
  - Press the **Remove** soft key to remove the party from the conference call.
  - Press the **New Call** soft key to place a new 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 on the SIP-T42G IP phone to conduct a conference with multiple participants.

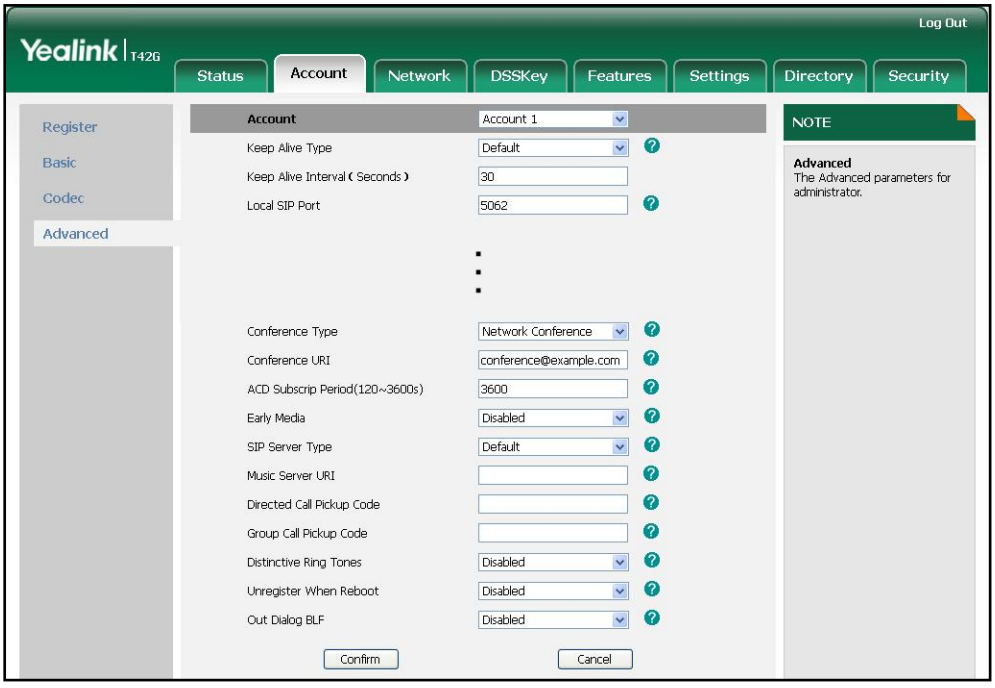
This feature allows you to perform the following:

- 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.

To configure 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.



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

Note Network conference is configurable via web user interface only.

To set up a network conference call:

1. Place a call to the first party.
2. 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 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 place 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 on specific servers may be different from that introduced above. Contact your system administrator for more information.

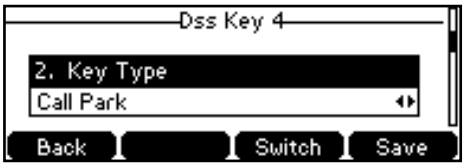
Call Park

You can use call park 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 prompt on the LCD screen.

Note Call park 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 code in the **Value** field.
8. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Call park key is configurable via web user interface at the path **DSSKey->Line Key**.

To use call park:

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 code to retrieve the parked call.

   The system establishes call between phone C and B.

Note

The call park code and call park retrieve 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 to answer someone else's incoming call on the phone. The SIP-T42G IP phone supports directed call pickup and group call pickup. 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 call pickup, you need to configure the call pickup code beforehand on a global or per-line basis via web user interface.

Note

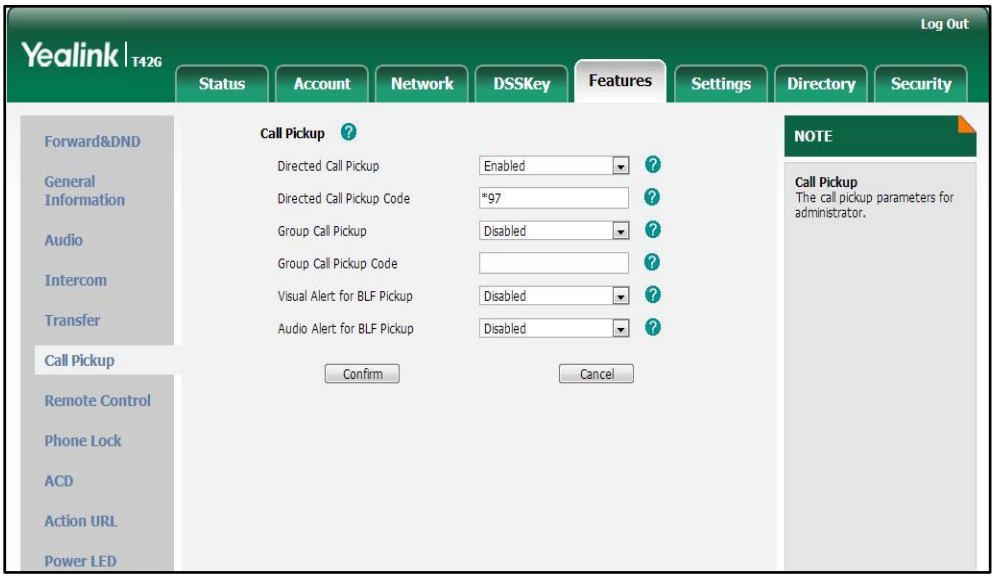
If there are many incoming calls at the same time, pressing the **GPickup** soft key on the phone will pick up the call that rings first.

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**.

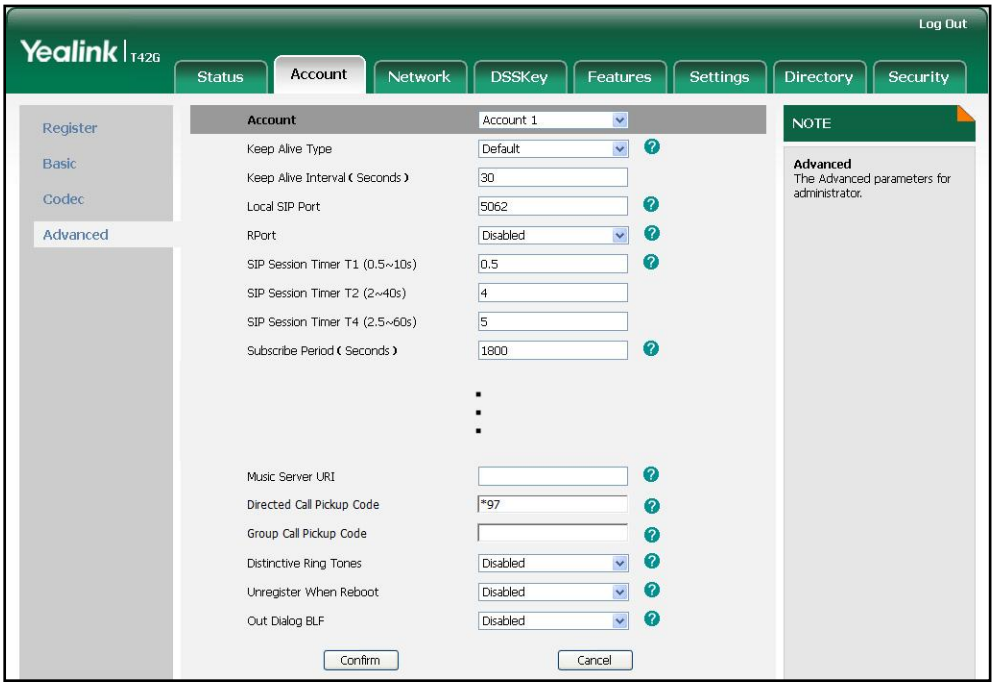
3. Enter the directed call pickup code in the **Directed Call Pickup Code** field.



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



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

To pick up a call directly:

1. Pick up the handset.

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

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

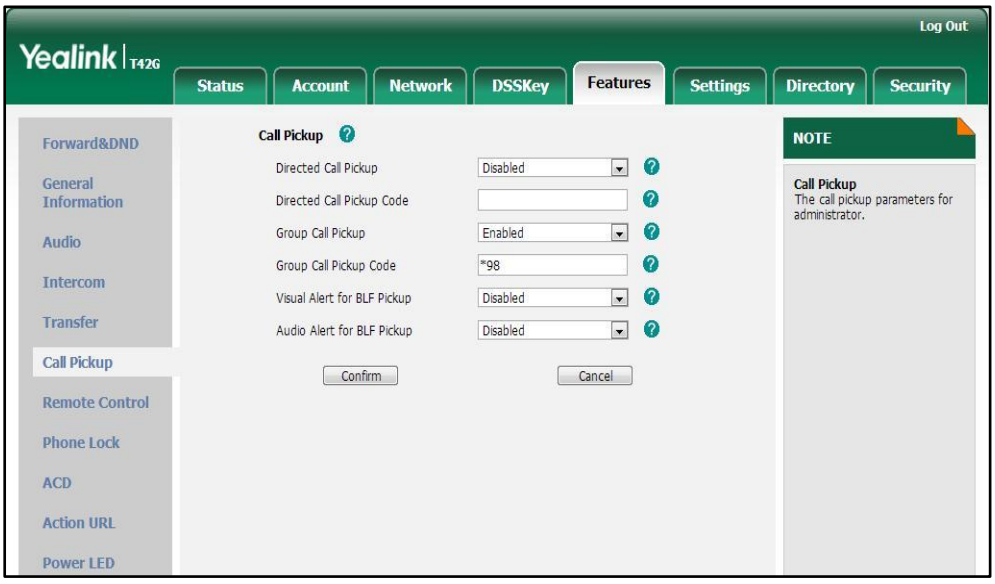
   The call is answered on your phone.

You can also configure a DSS key as 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:

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

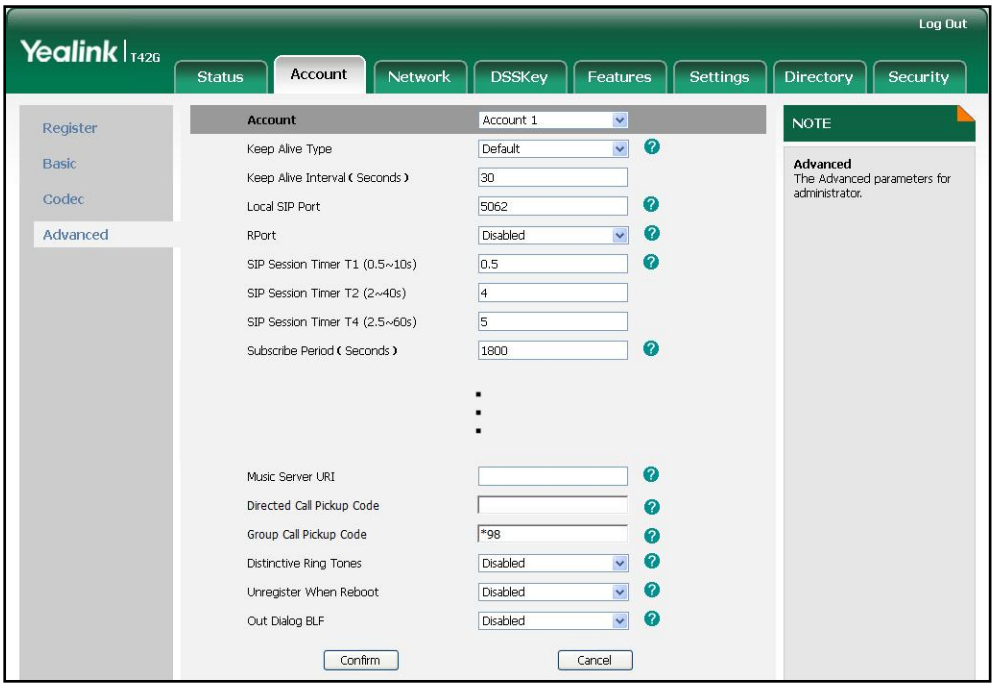


4. Click **Confirm** to accept the change.

To configure the group 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**.
4. Enter the group call pickup code in the **Group Call Pickup Code** field.



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

To pick up a call in the group:

1. 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.

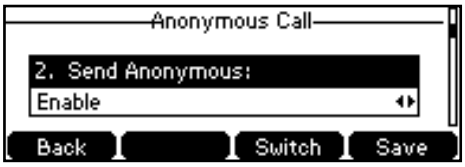
Anonymous Call

You can use anonymous call 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 services, but don't want to be harassed. Anonymous call is configurable on a per-line basis. You can also configure the phone to send anonymous on/off code to the server to activate/deactivate anonymous call on the server side.

Note Anonymous call 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. Press ◄ or ► , or the **Switch** soft key to select the desired line from the **Line ID** field.
3. Press ◄ or ► , or the **Switch** soft key to select **Enable** from the **Send Anonymous** field.



4. (Optional.) Press ◄ or ► , or the **Switch** soft key to select the desired value from the **Anonymous Code** field.
5. (Optional.) Enter the anonymous call on code in the **Call On Code** field.
6. (Optional.) Enter the anonymous call off code in the **Call Off Code** field.
7. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Anonymous call is configurable 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.

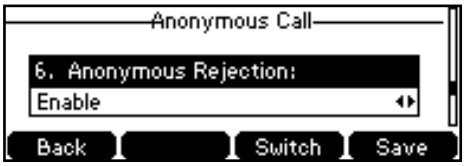
   The LCD screen of phone B prompts an incoming call from anonymity.

Anonymous Call Rejection

You can use anonymous call rejection 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. Press or , or the **Switch** soft key to select the desired line from the **Line ID** field.
3. Press or to scroll to the **Anonymous Rejection** field.
4. Press or , or the **Switch** soft key to select **Enable** from the **Anonymous 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.

Anonymous call rejection is configurable via web user interface at the path **Account**->**Basic**.

Advanced Phone Features

This chapter provides operating instructions for the advanced features of the SIP-T42G IP phone. 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.

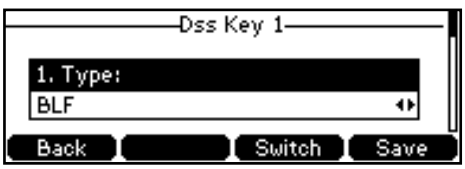
Busy Lamp Field (BLF)

You can use BLF to monitor a specific line 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 line (busy or idle). The BLF key LED illuminates solid red when the friend's line is in use. For more information on BLF key LED indications, refer to LED Instructions on page 4.

You can press a BLF key to dial out the monitored phone number when the monitored line is idle. You can receive a visual and an audio alert (if enabled), and also pick up the call when the monitored line 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.

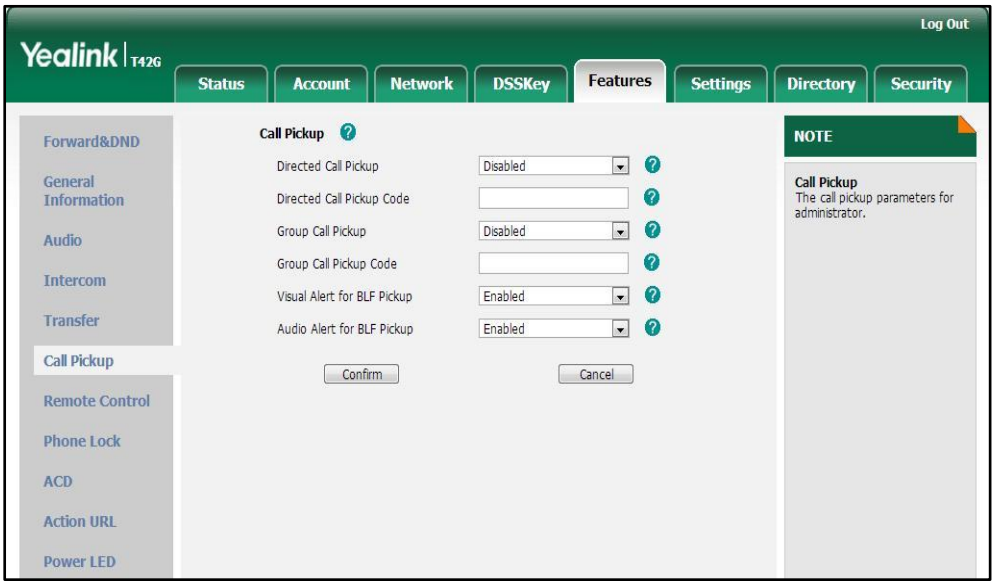
5. (Optional.) Enter a string that will appear on the LCD screen in the **Label** field.
6. Enter the phone number or extension you want to monitor in the **Value** field.
7. (Optional.) Enter the pickup code in the **Extension** field.
8. Press the **Save** soft key to accept the change or the **Back**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

BLF key is configurable via web user interface at the path **DSSKey**->**Line Key**.

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

To enable visual and audio alert 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**.



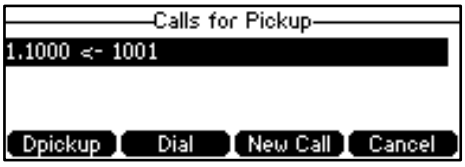
4. Click **Confirm** to accept the change.

Note Visual and audio alerts are configurable via web user interface only.

When the monitored line receives an incoming call, the followings occur on the 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 LCD screen shows an incoming call from 1001 on the monitored line.



Note If your phone is locked and the type of the keypad lock is configured as **Function Keys** or **All keys**, you cannot use the **Dpickup**, **Dial**, **New Call** and **Cancel** soft keys until unlocked.

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 the transfer mode on DSS key, refer to transfer in DSS Keys section on page 47.

Call Recording

You can record calls by pressing a record key on the phone. The SIP-T42G IP phone supports record and URL record.

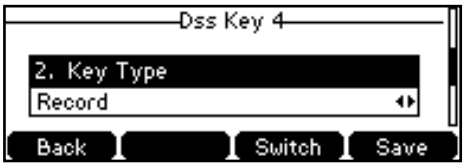
Two ways of call recording:

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

Note Call record is not available on all servers. Contact your system administrator for more information.

To configure a record key via phone user interface:

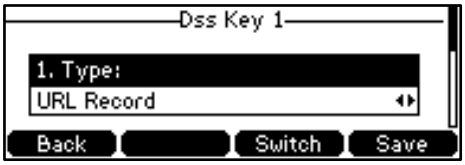
1. Press **Menu**->**Features**->**DSS Keys**.
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select **Record** from the **Key Type** field.



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.

Record and URL Record keys are configurable via web user interface at the path **DSSKey**->**Line Key**.

The Record and 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 and URL Record keys are not available when:

- There are no connected calls on your phone
- 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 will appear on the LCD screen and the Record or URL Record key LED will flash 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 goes out.

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 stopped	∅ appears for 1 second, then goes back

The recording box is full	appears for 1 second
The call 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

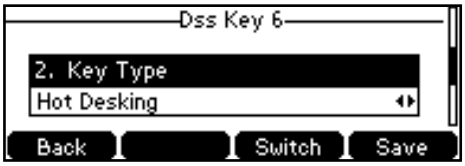
Hot desking originates from the definition of being the temporary physical occupant of a work station or surface by a particular employee. A primary motivation for hot desking is cost reduction. This feature is regularly used in places where not all the employees are in the office at the same time, or not in the office for very long, which means actual personal offices would be often vacant, consuming valuable space and resources.

You can use hot desking on the SIP-T42G IP phone to log out of the existing accounts and then log into 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 Hot desking 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.



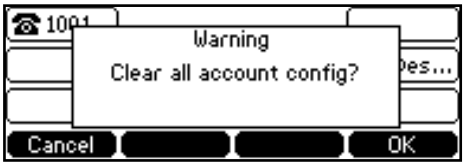
6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Hot desking key is configurable 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 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.
4. Press the **Save** soft key to login or the **Back** soft key to cancel.

Intercom

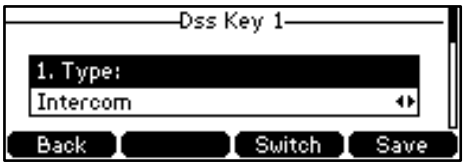
Intercom is a useful feature in an office environment to quickly connect with the operator or the secretary. You can press the intercom key to automatically connect with a remote extension for outgoing intercom calls, and the remote extension will automatically answer the incoming intercom calls.

Note Intercom 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.

Intercom key is configurable 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

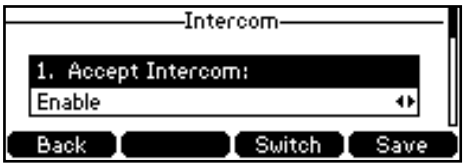
The SIP-T42G IP phone supports automatically to answer an incoming intercom call by default. The phone automatically plays a warning tone when it receives an incoming intercom call. In addition, you can enable the phone to mute the microphone when it automatically answers an incoming intercom call. You 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 Feature	Description
Accept Intercom	Enable or disable the IP phone to automatically answer an incoming intercom call.
Intercom Mute	Enable or disable the microphone on the IP phone for intercom calls.
Intercom Tone	Enable or disable the IP phone to play a warning tone when it receives an incoming intercom call.
Intercom Barge	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.

These specific parameters are configurable 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 message to the caller. Accept Intercom is enabled by default.

Note Your administrator can set a period of delay time before the phone automatically answers intercom calls. 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 works for intercom 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 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.

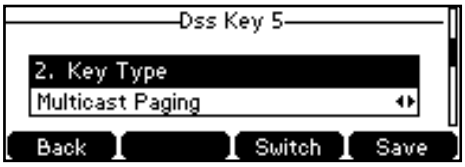
Multicast Paging

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 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.

Multicast paging key is configurable via web user interface at the path **DSSKey**->**Line Key**.

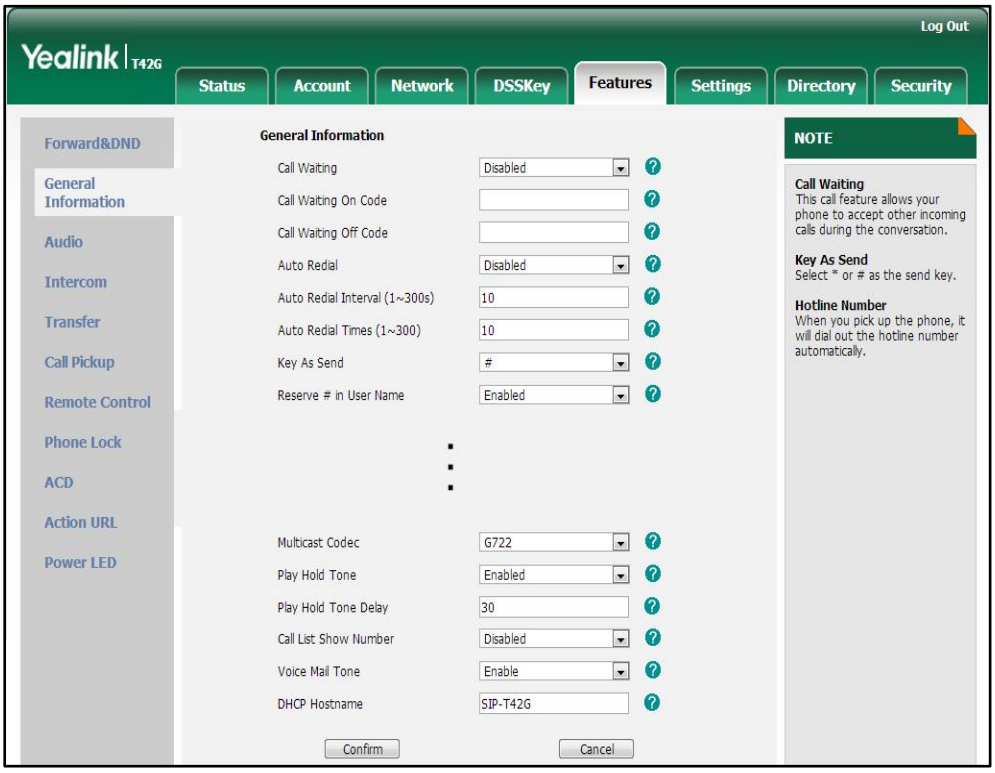
You can also configure the phone to use a default codec for sending multicast RTP stream via web user interface.

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 LCD screen displays the HD icon to indicate that is providing high definition voice.

Default codec for multicast paging is configurable 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.

How the phone handles incoming multicast paging calls depends on Paging Barge and Paging Priority Active parameters configured via web user interface.

Paging Barge

Paging barge parameter defines 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, it will be ignored automatically. If Disabled is selected from the pull-down list of Paging Barge, the voice call in progress takes precedence over all incoming multicast paging calls. Valid values in the Paging Barge field:

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

Paging Priority Active

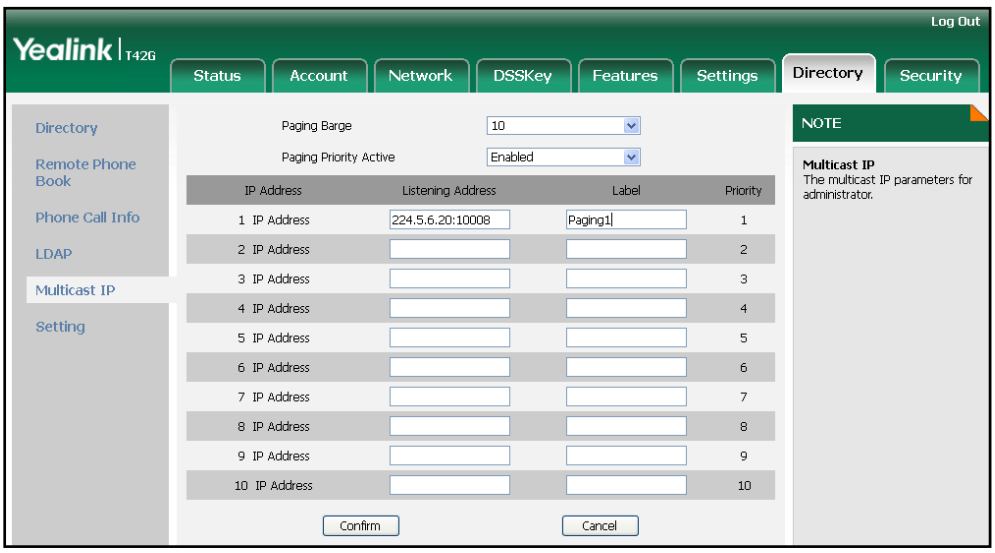
Paging priority active parameter decides how the phone handles incoming multicast paging calls when there is already a multicast paging call on the phone. If enabled, the phone will ignore incoming multicast paging calls with lower priorities, otherwise, the phone will answer incoming multicast paging calls automatically 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 priorities of listening addresses are predefined: 1 with the highest priority, 10 with the lowest.

Both the multicast paging sender and the receiver play a warning tone when establishing a multicast paging call.

Multicast listening addresses are configurable via web user interface only.

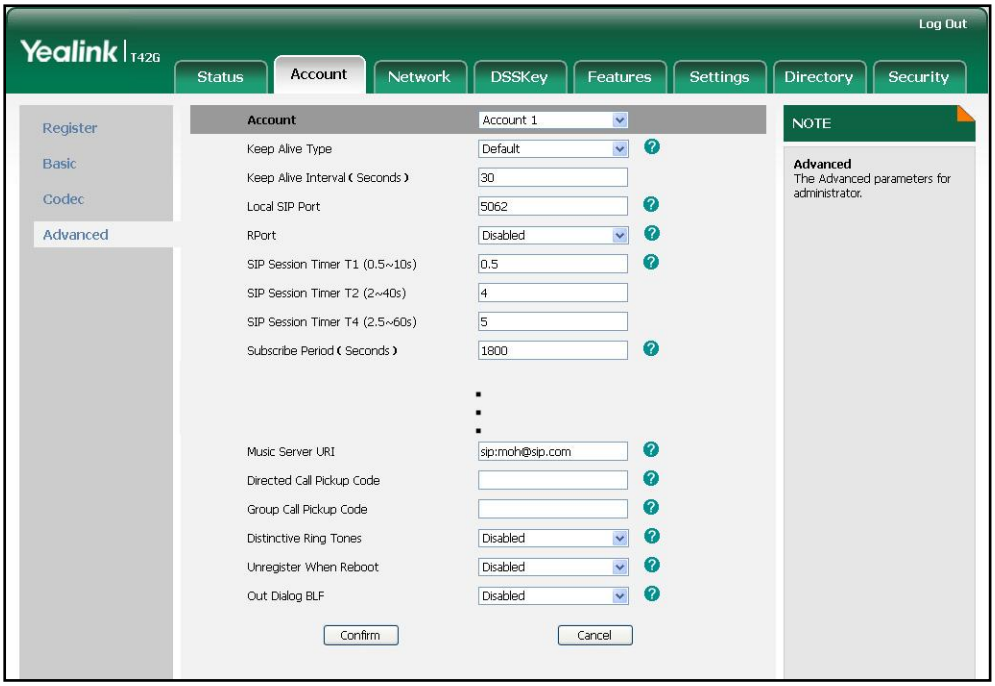
Music on Hold

Music on hold (MoH) is the business practice of playing recorded music to fill the silence that would be heard by the party placed on hold. To use this feature, you should specify a SIP URI pointing to a Music on Hold Server account. When a call is placed on hold, the phone will send a SIP INVITE message to the Music on Hold Server account. The Music on Hold Server account automatically responds 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**.
3. Click on **Advanced**.

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



5. 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).

Music on hold server is configurable via web user interface only.

Automatic Call Distribution (ACD)

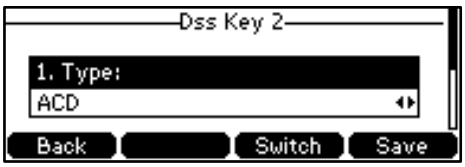
ACD is often used in offices for customer service, such as call center. The ACD system handles large volumes of incoming calls from callers who have no need to talk to a specific person but who require assistance from any of multiple persons at the earliest opportunity. ACD on the SIP-T42G IP phone allows the ACD system to distribute large volumes of incoming calls to the registered ACD users. To use this feature, you should configure an ACD key in advance.

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

To configure an ACD 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 **ACD** from the **Type** field.



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

ACD key is configurable via web user interface at the path **DSSKey->Line Key**.

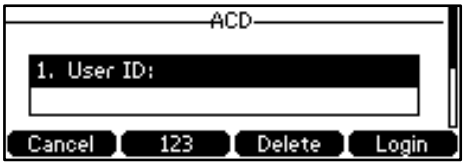
To log into the ACD system:

1. Press the ACD key when the phone is idle.

   The LCD screen prompts you the following information:

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

   Password: the password used to log into the queue.



2. 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 into 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 ACD user status. You can press the **Available/Unavailable** soft key to change your ACD user status. ACD user status synchronizes on both IP phone and ACD system. When you set the ACD user status to be available, the ACD key LED illuminates solid green, and then the server begins distributing calls to your phone. When you set the ACD user status to be unavailable, the ACD key LED flashes green, and then the server temporarily stops distributing calls to your phone. To log out of 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-T42G 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 LCD screen will display a prompt message and appear an icon.

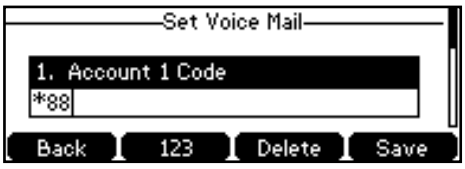
Note Voice mail is not available on all servers. Contact your system administrator for more information.

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 a voice mail, and then hang up.

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., *88).



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 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.

2. Follow the voice prompt to listen to voice mails.

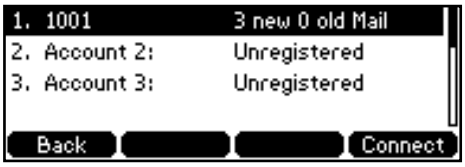
Note Before listening to voice mails, make sure the voice mail access code has been configured.

When all new voice mails are retrieved, the power indicator LED will go out.

To view the voice mail via phone user interface:

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

   The LCD screen displays the amount of new and old voice mails.



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

Message Waiting Indicator (MWI)

The SIP-T42G 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, an indicator message (including a voice mail icon) on the LCD screen, and the power indicator LED slow flashes red. This will be cleared when you retrieve all voice mails or delete them.

The MWI service is unsolicited for some servers, so the SIP-T42G IP phone only handles the MWI messages sent from the server. But for other servers, the MWI service is solicited, so the SIP-T42G IP phone must enable subscription for MWI.

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

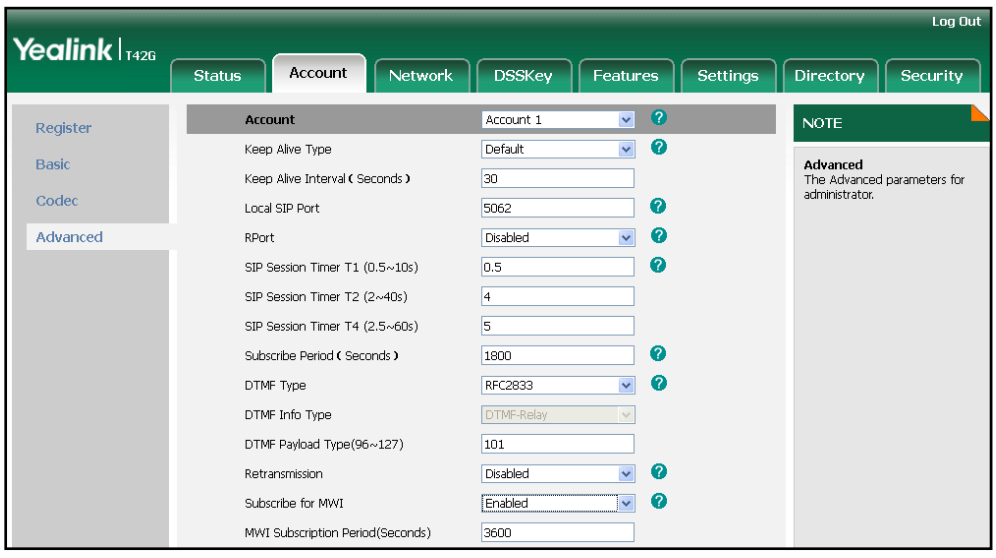
The MWI subscription parameters you need to know:

Options	Description
Subscribe for MWI	Enable or disable a subscription for MWI service.
MWI Subscription Period	Period of MWI subscription. The IP phone sends a refresh SUBSCRIBE request before initial SUBSCRIBE expiration.
Subscribe MWI To Voice Mail	Enable or disable a subscription to the voice mail number for MWI service. To use this feature, you should also configure the voice mail number.

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

To configure 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.

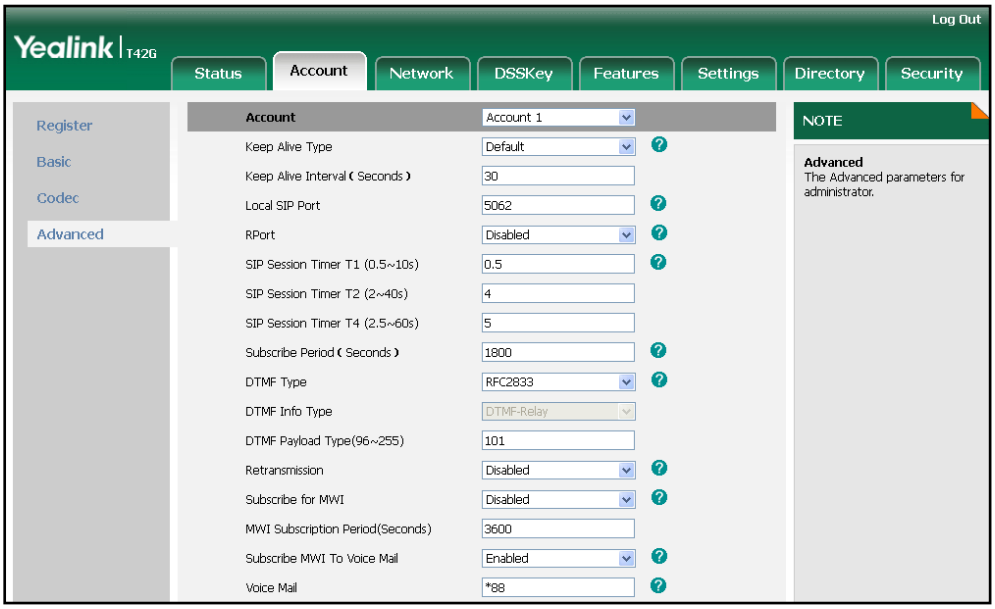


6. Click **Confirm** to accept the change.

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

To enable Subscribe MWI to Voice Mail 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 mail number in the **Voice Mail** field.



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

   The IP phone will subscribe to the voice mail number for MWI service using Subscribe MWI to Voice Mail.

Note MWI subscription is configurable via web user interface only.

Troubleshooting

This chapter provides general troubleshooting information to help you solve the problems you might encounter when using your SIP-T42G IP phone.

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

Why is the LCD screen blank?

- Ensure that the phone is properly plugged into a functional AC outlet.
- Ensure that the phone is plugged into a socket controlled by a switch that is on.
- If the phone is plugged into a power strip, try to plug it directly into a wall outlet instead.
- If the phone is powered from PoE, ensure you use a PoE-compliant switch or hub.

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 the phone to obtain the time and date from the SNTP server automatically. If the phone is unable to access the SNTP server, you need to 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 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?

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 (Purchase Order).
- 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 the installation instructions, refer to Phone Installation on page 11.
- Switch between the Handset, Headset (if present) or Hands-Free Speakerphone to check whether the 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 the 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 26.

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 77.
- Check that call forward is disabled on the phone. Refer to Call Forward on page 80.
- Check whether the caller number is stored in the blacklist directory. Refer to Blacklist on page 39.

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 45.
- Check that the headset volume is adjusted to an appropriate level. Refer to Volume on page 26.

What is the difference between user name, register name and display name?

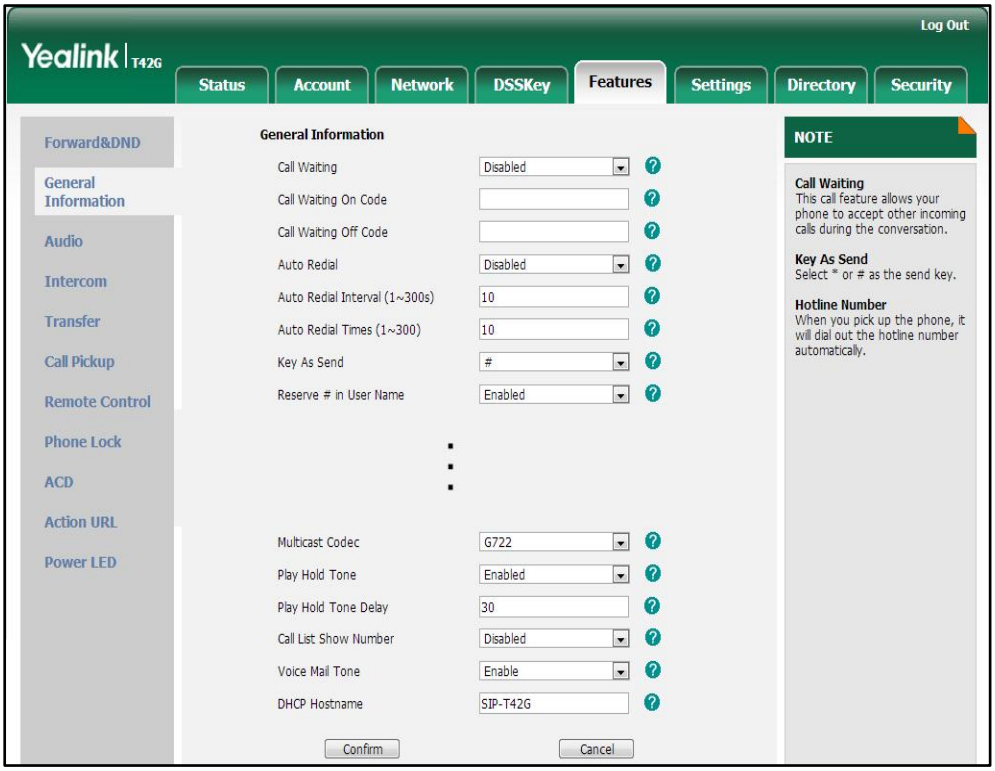
Both user name and register name are defined by the server. A user name is used to identify the account, while a register name matched with a password is used for authentication if the server requires. Display name is the caller ID that will be displayed on the callee's LCD screen. Server configuration may override the local configuration.

Why does the phone play a tone when hold? How to disable it?

When there is a call is on hold, the phone will play a hold tone every 30 seconds. Play hold tone is enabled by default. Play hold tone and the interval of playing a hold tone are configurable 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.



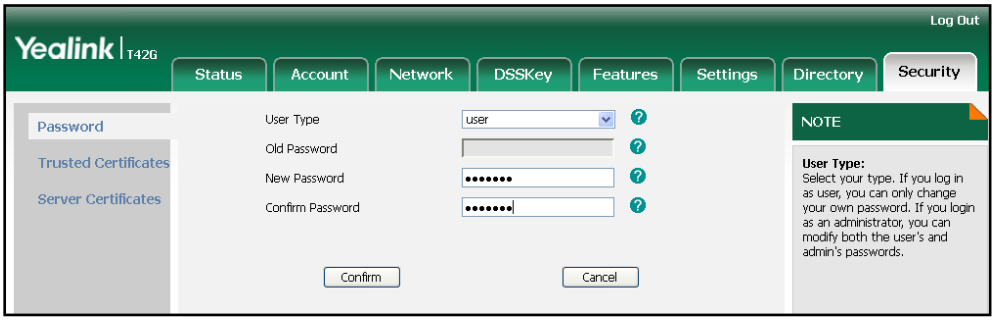
4. Click **Confirm** to accept the change.

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.



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

You can also contact your system administrator for help.

Note If logging into the web user interface of the phone with the user credential, you need to enter the current user password in the **Old Password** field.

User password is configurable via web user interface only.

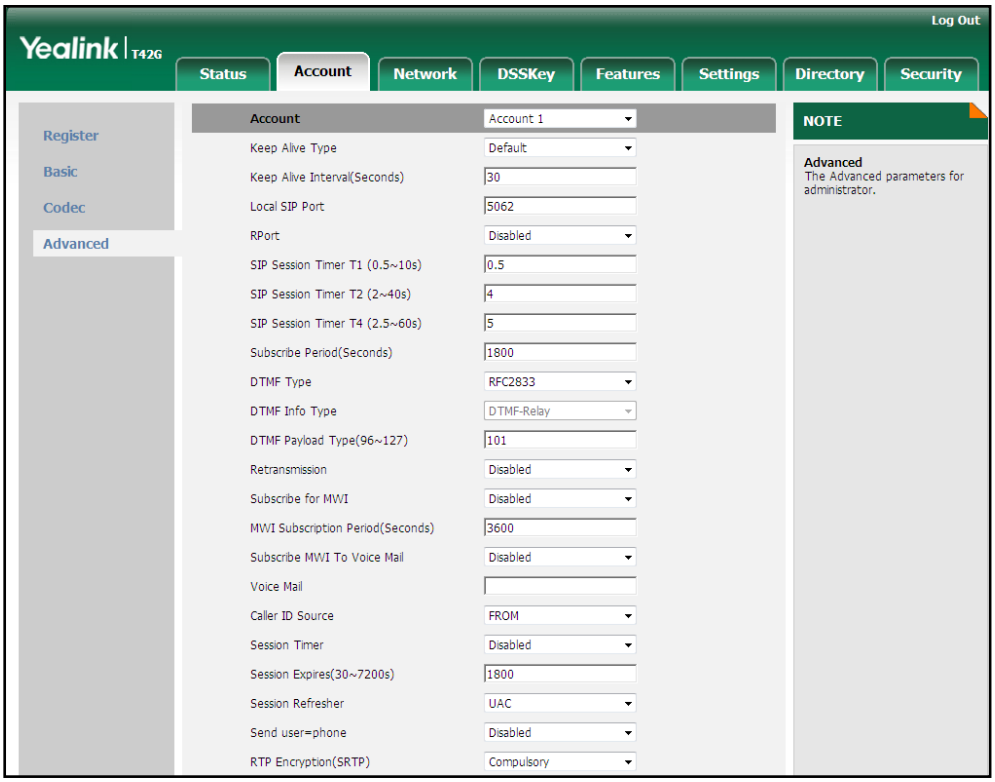
How to make a call using SRTP?

You can enable SRTP to encrypt the audio stream(s) of phone calls. The parties participating in the call should enable SRTP. You can configure 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)**.



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

Note

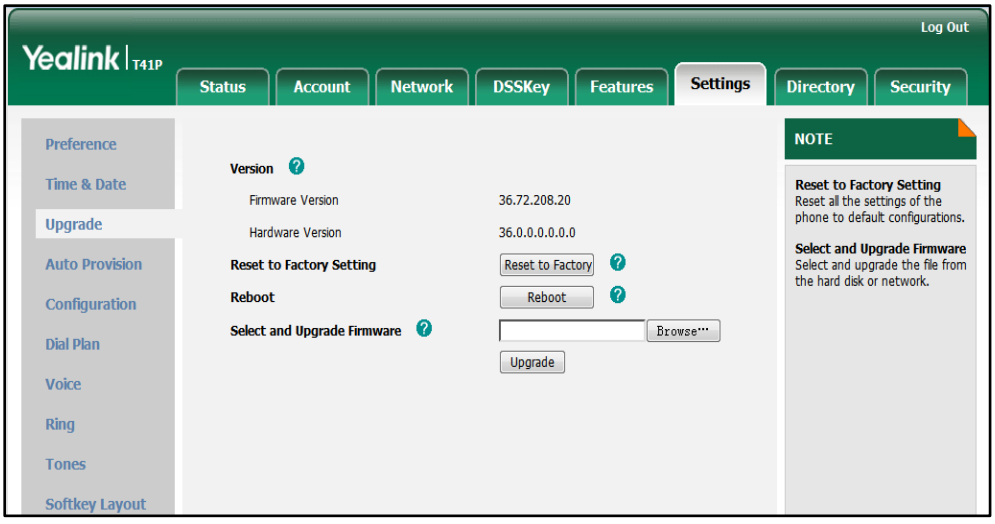
SRTP is not available on all servers. Contact your system administrator for more information.

SRTP is configurable via web user interface only.

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.

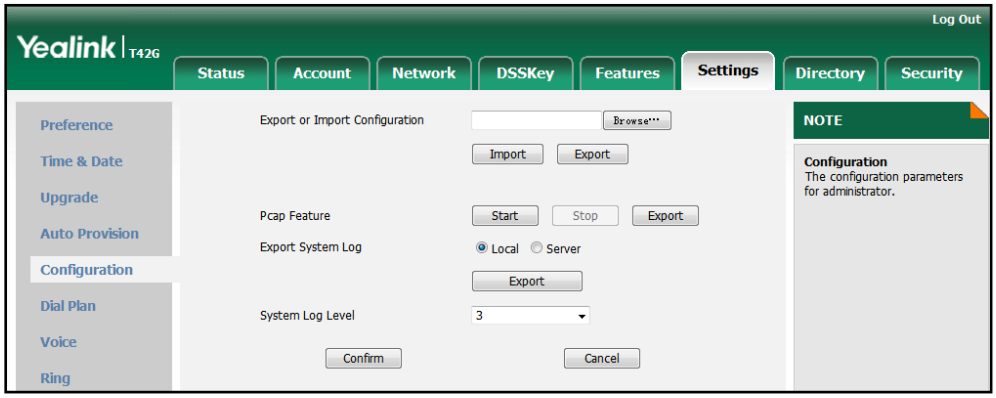
Note Any reboot of the phone may take a few minutes.

How to export PCAP trace?

We may need you to provide a PCAP trace to help analyze problem.

To export a PCAP trace via web user interface:

1. Click on **Settings->Configuration**.
2. Click **Start** to begin capturing signal traffic.
3. Recreate the error to be documented in the trace.
4. Click **Stop** to stop the capture.



5. Click **Export** to open file download window, and then save the file to your local system.

How to export system log?

We may need you to provide the phone configurations to help analyze problem. In some instance, you may need to import configurations to your phone.

To export the system log to local PC via web user interface:

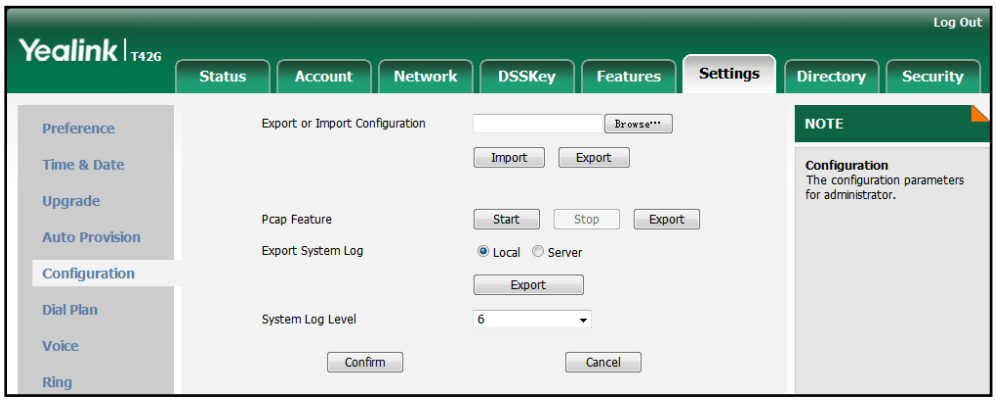
1. Click on **Settings->Configuration**.
2. Select **6** from the pull-down list of **System Log Level**.

   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 a reboot.
4. Click **OK** to reboot the phone.

   After the IP phone finishes reboot, 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 open file download window, and then save the system log file to your local system.

You can also export the system log to a syslog server, contact your system administrator for more information.

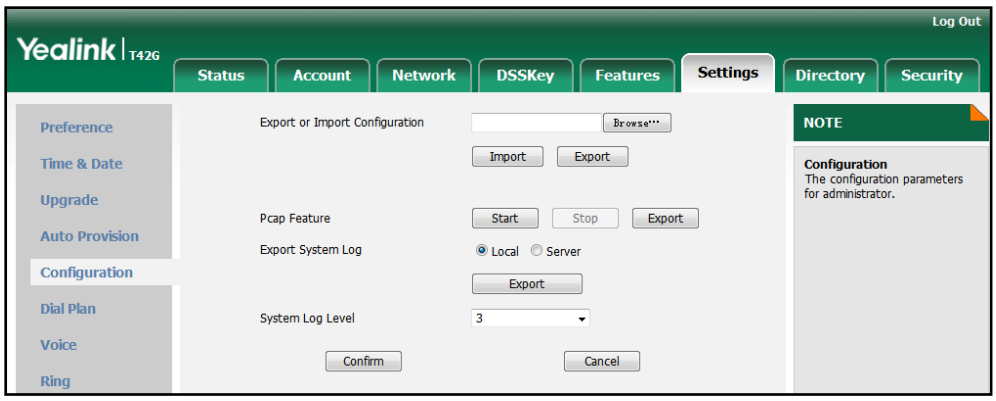
Note It is recommended to reset the syslog level to 3 after exporting the system syslog.

How to export/import the phone configurations?

We may need you to provide the phone configurations to help analyze 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 open file download window, and then save the file to your local system.



To import the phone configurations via web user interface:

1. Click on **Settings->Configuration**.
2. Click **Browse** to locate a configuration file from your local system.

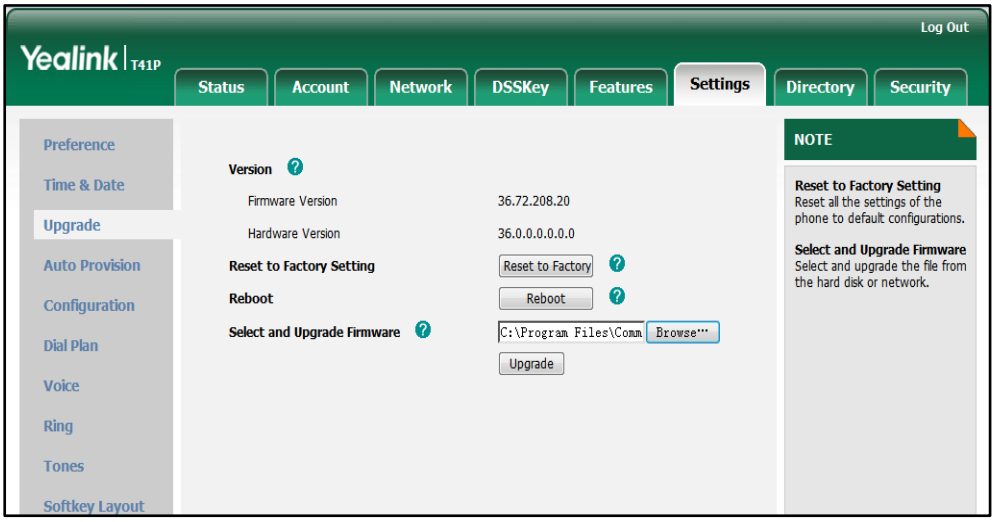
3. Click **Import** to import the configuration file.

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 locate 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.

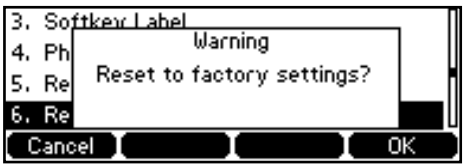
How to reset the phone?

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.

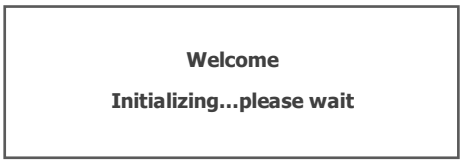
   The LCD screen prompts the following warning:



3. Press the **OK** soft key.

   The LCD screen prompts the following:

   The LCD screen prompts the following:



   The phone will be reset to factory sucessfully 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 on service agreements applicable to your product.

Limitations of Liability

TO THE FULL EXTENT ALLOWED BY LAW, YEALINK EXCLUDES FOR ITSELFAND ITS SUPPLIERS ANY LIABILITY, WHETHER BASED IN CONTRACT OR TORT (INCLUDINGNEGLIGENCE), FOR INCIDENTAL, CONSEQUENTIAL, INDIRECT, SPECIAL, OR PUNITIVE DAMAGES OFANY KIND, OR FOR LOSS OF REVENUE OR PROFITS, LOSS OF BUSINESS, LOSS OF INFORMATION ORDATA, 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 SUCHAMAGES, AND LIMITS ITS LIABILITY TO REPAIR, REPLACEMENT, OR REFUND OF THE PURCHASERICE PAID, AT YEALINK'S OPTION. THIS DISCLAIMER OF LIABILITY FOR DAMAGES WILL NOT BEAFFECTED 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 damageand 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. Pleaseuse 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 perils.

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	Spain(Madrid)
+01:00	Croatia(Zagreb)
+01:00	Czech Republic(Prague)
+01:00	Denmark(Kopenhagen)
+01:00	France(Paris)

Time Zone	Time Zone Name
+01:00	Germany(Berlin)
+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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