



Enterprise IP Phone SIP-T23 & T23P & T23G User Guide

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Part 15 FCC Rules

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

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

1. This device may not cause harmful interference.
2. This device must accept any interference received, including interference that may cause undesired operation.

Class B Digital Device or Peripheral

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

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

WEEE Warning



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

Customer Feedback

We are striving to improve our documentation quality and we appreciate your feedback. Email your opinions and comments to DocsFeedback@yealink.com.

GNU GPL INFORMATION

Yealink SIP-T23(P/G) 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?BaseInfoCatId=293&NewsCatId=293&CatId=293>.

About This Guide

Thank you for choosing the SIP-T23(P/G) IP phone, exquisitely designed to provide business telephony features, such as Call Hold, Call Transfer, Busy Lamp Field, Multicast Paging and Conference over an IP network. The difference among the SIP-T23, T23P and T23G IP phones is that SIP-T23P and SIP-T23G support PoE, while SIP-T23 does not, and only SIP-T23G supports kilomega network interface.

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-T23(P/G) IP phone.

Note

Shared Line, Network Directory and Network Call Log features are hidden for IP phones in neutral firmware version, which are designed for the 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](#)

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Overview

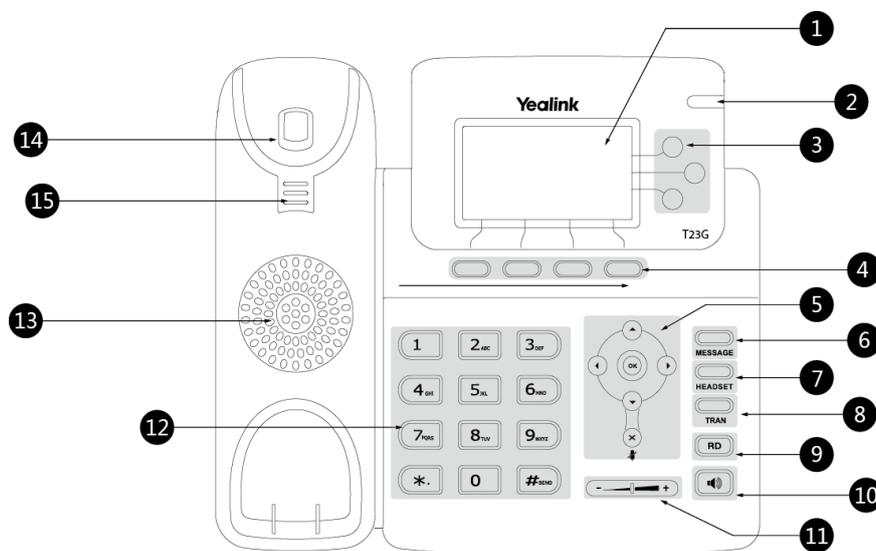
This chapter provides the overview of the SIP-T23(P/G) IP phone. Topics include:

- [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-T23(P/G) IP phone are the LCD screen and the keypad.



Hardware component instructions of the SIP-T23(P/G) IP phone are:

	Item	Description
①	LCD Screen	Shows information about calls, messages, soft keys, time, date and other relevant data: <ul style="list-style-type: none"> • Call information — caller ID, call duration • Icons (for example, ) • 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 status and phone status.
③	Line Keys	Use these keys to activate up to three accounts and assign various features.
④	Soft Keys	Label automatically to identify their context-sensitive features.
⑤		Scroll through the displayed information.
		Confirms actions or answers an incoming call.
		Cancels actions, rejects incoming calls, mutes or un-mutes calls.
⑥	MESSAGE Key	Indicates and accesses voice mails.
⑦	HEADSET Key	Toggles the headset mode.
⑧	TRAN Key	Transfers a call to another party.
⑨	RD Key	Redials a previously dialed number.
⑩	Speakerphone Key	Toggles the hands-free speakerphone mode.
⑪	Volume Key	Adjusts the volume of the handset, headset, speaker, and ringer.
⑫	Keypad	Provides the digits, letters and special characters in context-sensitive applications.
⑬	Speaker	Provides ringer and hands-free (speakerphone) audio output.
⑭	Hookswitch Tab	Secures the handset in the handset cradle when the IP phone is mounted vertically.
⑮	Hookswitch	Picking up the handset from the handset cradle, the hookswitch bounces and the phone connects to the line, laying the handset down on the handset cradle, the phone disconnects from the line.

Icon Instructions

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

Icon	Description
	Network is unavailable
	Private line registers successfully
	Register failed/Registering
	Hands-free speakerphone mode
	Handset mode
	Headset mode
	Multi-lingual lowercase letters input mode
	Multi-lingual uppercase letters input mode
	Alphanumeric input mode
	Numeric input mode
	Multi-lingual uppercase and lowercase letters input mode
	Voice Mail
	Text Message
	Auto Answer
	Do Not Disturb
	Call Forwarded/Forwarded Calls
	Call Hold
	Call Mute
	Ringer volume is 0
	Keypad Lock

Icon	Description
	Received Calls
	Placed Calls
	Missed 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	The phone receives a text message or voice mail.
Off	<p>The phone is powered off.</p> <p>The phone is idle.</p> <p>The phone is busy.</p> <p>The call is muted.</p> <p>The call is placed on hold or held.</p>

Line key LED

LED Status	Description
Solid green	<p>The line is in conversation.</p> <p>The line is seized.</p> <p>The call is muted</p>
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 or BLF List key)

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

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 IP Phone Family Administrator Guide.

User Interfaces

Two ways to customize configurations of your SIP-T23(P/G) 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, 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 17.

Web User Interface

In addition to the phone user interface, you can also customize your phone via web user interface. In order to access the web user interface, you need to know the IP address of your new phone. To obtain the IP address, press the OK key on the phone when the

phone is idle. Enter the IP address (e.g., http://192.168.0.10 or 192.168.0.10) in the address bar of a 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		
--IP Address		
--MAC		
--Firmware	√	√
--Network		
--Phone		
--Accounts		
Basic Phone Settings		
--Contrast		
--Backlight	√	
--Language	√	
--Time & Date	√	
--Administrator Password	√	
--Key as Send	√	
--Keypad Lock	√	
--Ring Tones	√	
--Contact Management		
--Directory		
--Local Directory	√	√
--Blacklist	√	
--Remote Phone Book		
--Call History Management	√	
--Logo Customization		
--DSS Keys	√	
--Account Registration	√	
--Dial Plan		
--Emergency Number		
--Live Dialpad		
--Hotline	√	
Basic Call Features		
--Recent Call In Dialing		
--Auto Answer	√	
--Auto Redial	√	
--Call Completion	√	
--Call Return	√	
--Do Not Disturb (DND)	√	√

Options	Phone User Interface	Web User Interface
--Call Forward	√	
--Call Transfer	√	
--Call Waiting	√	
--Conference		
--Call Park	√	
--Call Pickup	√	
--Anonymous Call	√	
--Anonymous Call Rejection	√	
Advanced Phone Features		
--Busy Lamp Field (BLF)	√	
--BLF List		
--Call Recording	√	
--Hot Desking	√	
--Intercom	√	√
--Multicast Paging		
--Music on Hold		
--Automatic Call Distribution (ACD)	√	
--Messages	√	
SIP Account		
--User Options		
--Activation	√	
--Label	√	
--Display Name	√	
--Register Name	√	
--User Name	√	
--Password	√	
--SIP Server1/2	√	
--Server Option		
--Registrar Port		
--Outbound Status	√	
--Outbound Proxy	√	
--NAT Status	√	√

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 SIPT23(P/G) IP phone.

Name	Contents	Where found	Language
Quick Start 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

Getting Started

This chapter provides basic installation instructions and information for obtaining the best performance with the SIP-T23(P/G) IP phone. Topics include:

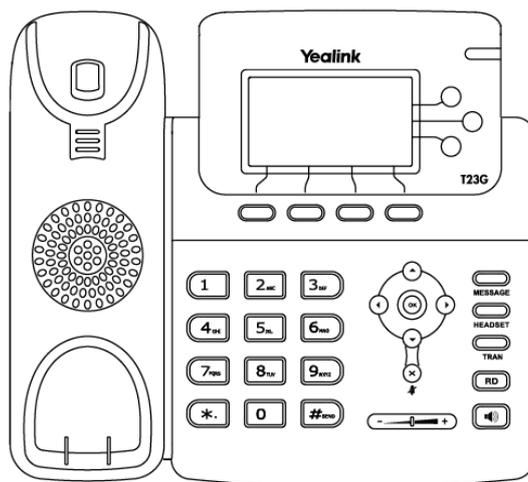
- [Packaging Contents](#)
- [Phone Installation](#)
- [Phone Initialization](#)
- [Phone Status](#)
- [Basic Network Settings](#)
- [Registration](#)
- [Idle Screen](#)

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

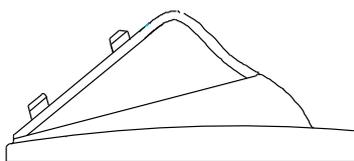
Packaging Contents

The following components are included in your SIP-T23(P/G) IP phone package:

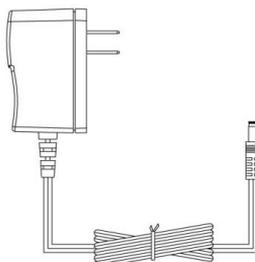
- **SIP-T23(P/G) IP phone**



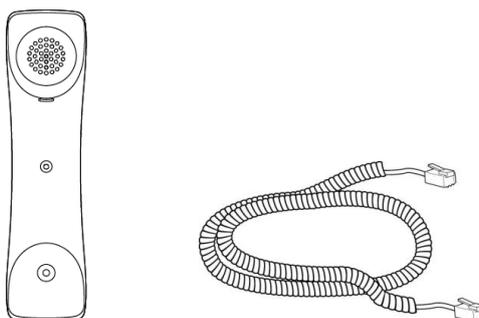
- **Phone Stand**



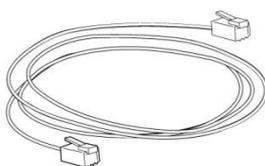
- **Power Adapter (Optional)**



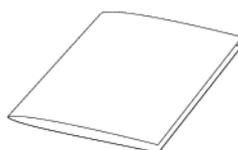
- **Handset & Handset Cord**



- **Ethernet Cable**



- **Quick Start Guide**



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

Phone Installation

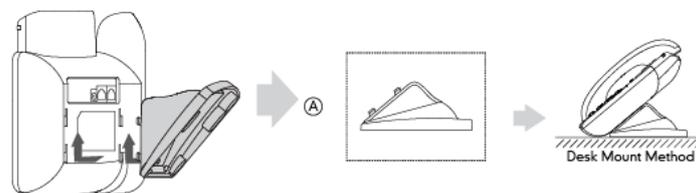
If your phone is already installed, proceed to [Phone Initialization](#) on page 13.

This section introduces how to install the phone:

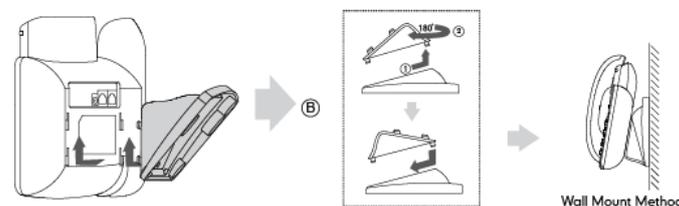
- 1) Attach the stand
- 2) Connect the handset and optional headset
- 3) Connect the network and power

1) Attach the stand

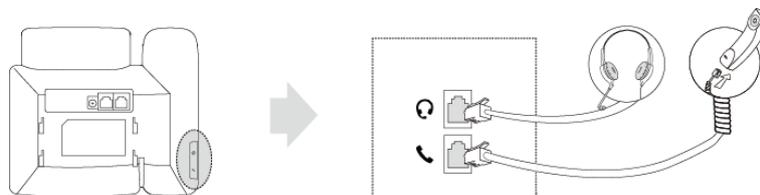
a) Desk Mount Method:



b) Wall Mount Method:



2) Connect the handset and optional headset



Note

A headset is not included in the packaging contents. Contact your system administrator for more information.

3) Connect the network and power

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

- AC power
- Power over Ethernet (PoE)

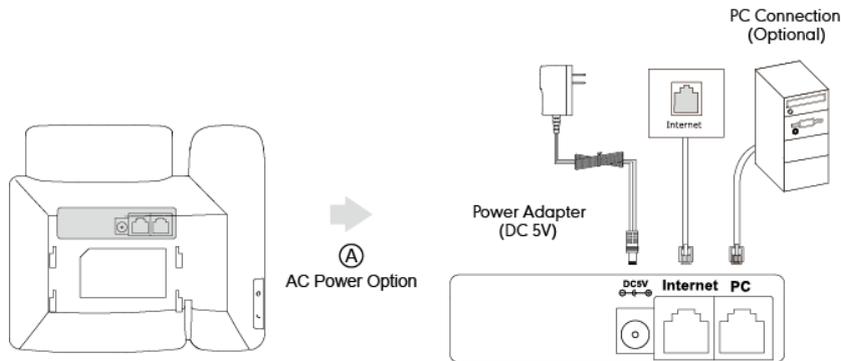
Note

PoE is not applicable to the SIP-T23 IP phone. A power adapter should be purchased separately to power it on.

AC Power

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.

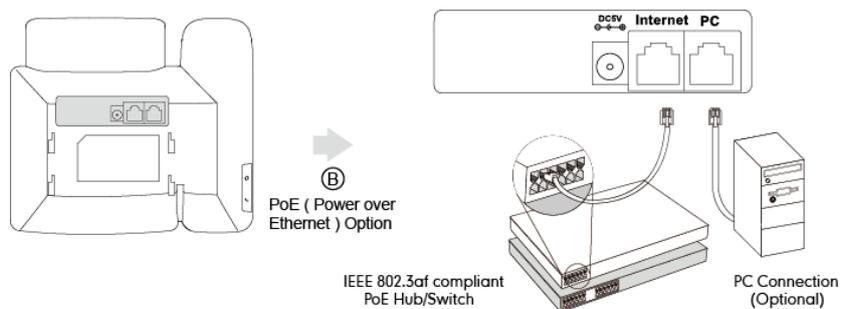


Power over Ethernet

With the included or a regular Ethernet cable, the SIP-T23/T23G IP phone can be powered from a PoE-compliant switch or hub. PoE is not applicable to the SIP-T23 IP phone.

To connect the PoE for the SIP-T23P/T23G IP phone:

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



Note

If in-line power is provided, you don't need to connect the 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). This is an optional connection.

Important! Do not 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 “Welcome 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 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 14.

Phone Status

You can view the system status of your phone via phone user interface or web user interface.

Available information of phone status includes:

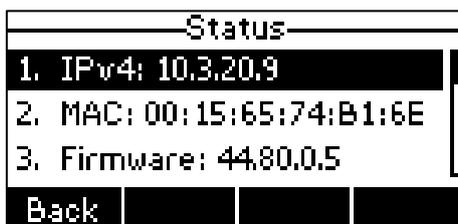
- Network status (e.g., IPv4 status, IP mode, MAC address).
- Phone status (e.g., device model, hardware version, firmware version, product ID, MAC address and device certificate status).
- Account status (e.g., the register status of SIP accounts).

Note

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

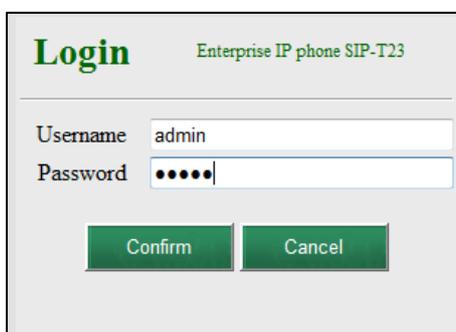
To view the phone status via phone user interface:

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



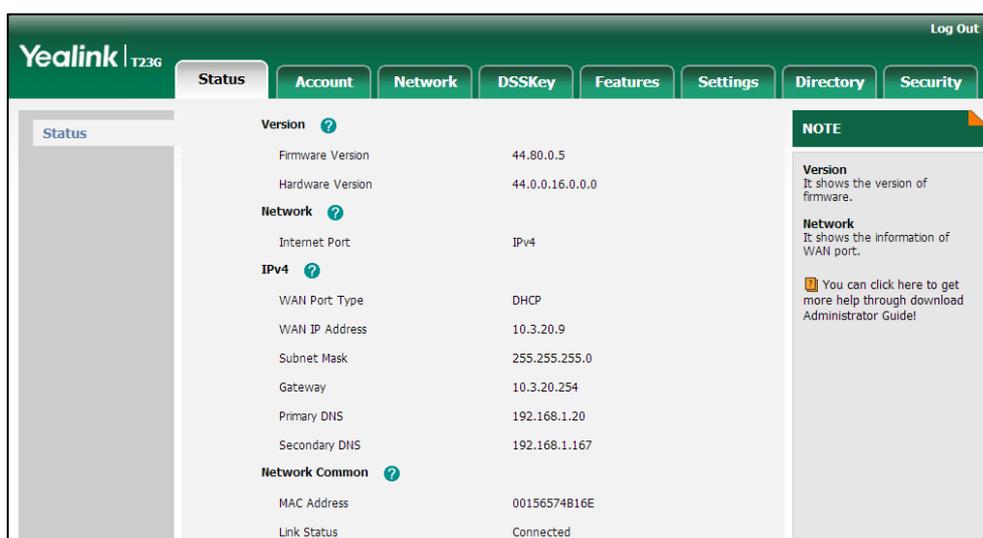
To view the phone status via web user interface:

1. Open a web browser on your computer.
2. Enter the IP address in the browser's address bar, and then press **Enter**.
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.



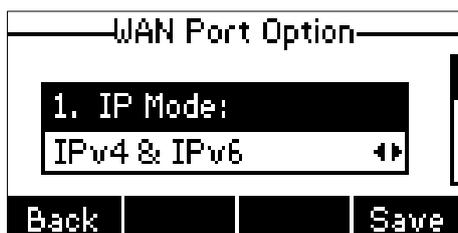
Basic Network Settings

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** (default password: admin)
->**Network->WAN Port**.

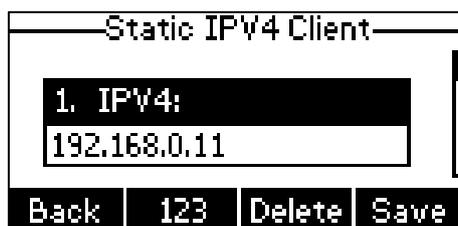
- Press  or  to select **IPv4**, **IPv6** or **IPv4 & IPv6** from the **IP Mode** field.



- 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:

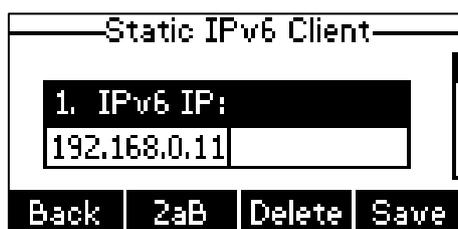
- Press **Menu->Settings->Advanced Settings** (default password: admin)
->**Network->WAN Port**.
Make sure that the IP address mode is configured as **IPv4** or **IPv4 & IPv6**.
- Press  to select **IPv4** and press the **Enter** soft key.
- Press  to select **Static IPv4 Client** and press the **Enter** soft key.
- Enter the desired values in the **IPv4**, **Subnet Mask**, **Default Gateway**, **IPv4 Pri DNS** and **IPv4 Sec DNS** fields respectively.



- 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:

- Press **Menu->Settings->Advanced Settings** (default password: admin)
->**Network->WAN Port**.
Make sure that the IP address mode is configured as **IPv6** or **IPv4 & IPv6**.
- Press  to select **IPv6** and press the **Enter** soft key.
- Press  to select **Static IPv6 Client** and press the **Enter** soft key.
- Enter the desired values in the **IPv6 IP**, **IPv6 IP Prefix**, **IPv6 Default Gateway**, **IPv6 Pri.DNS** and **IPv6 Sec.DNS** fields respectively.



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

If you are using an xDSL modem for IPv4 network connection, you can connect your phone to the Internet via PPPoE mode. Set the WAN port as a PPPoE port. The PPPoE port will perform a PPP negotiation to obtain the IP address. Contact your system administrator for the PPPoE user name and password.

To configure PPPoE via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (default password: admin) **->Network->WAN Port**.
2. Press  to select **IPv4** and press the **Enter** soft key.
3. Press  to select **PPPoE IP Client** and press the **Enter** soft key.



4. Enter the user name and password in the corresponding fields.
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Note

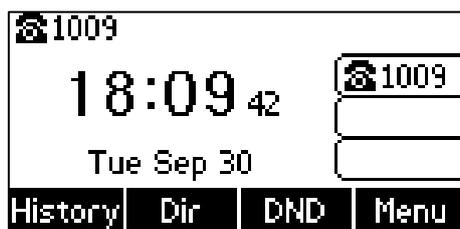
Wrong network settings may result in inaccessibility of your phone and may also have an impact on your network performance. For more information on these parameters, contact your system administrator.

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 screen will be displayed as below:



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

Customizing Your Phone

You can customize your SIP-T23(P/G) IP phone by personally configuring certain settings, for example, backlight, 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

Contrast

You can configure the contrast of the LCD screen to a comfortable level.

To configure the contrast via phone user interface:

1. Press **Menu->Settings->Basic Settings->Display->Contrast**.
2. Press  or , or the **Switch** soft key to increase or decrease the intensity of contrast.

The default contrast level is 6.



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

Contrast is configurable via web user interface at the path **Settings->Preference**.

Backlight

You can configure the backlight status on the LCD screen from the following options:

- **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 designated time (in seconds).

To configure the backlight via phone user interface:

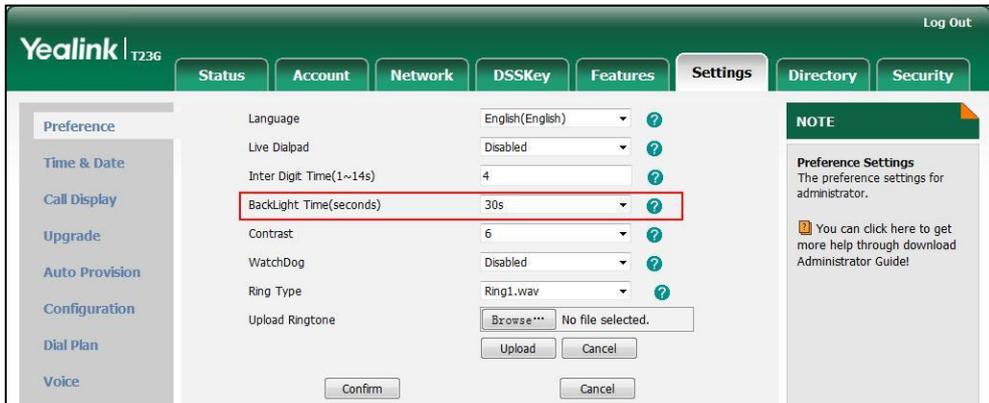
1. Press **Menu->Settings->Basic Settings->Display>BackLight**.



2. Press  or , or the **Switch** soft key to select the desired type from the **Backlight Time** field.
3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

To configure the backlight via web user interface:

1. Click on **Settings->Preference**.
2. Select the desired type from the pull-down list of **Backlight Time (seconds)**.



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

Language

The default language of the phone user interface is English. The phone will detect and use the same language as that 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. You can change the language for the phone user interface and the web

user interface respectively.

To change the language for the phone user interface:

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

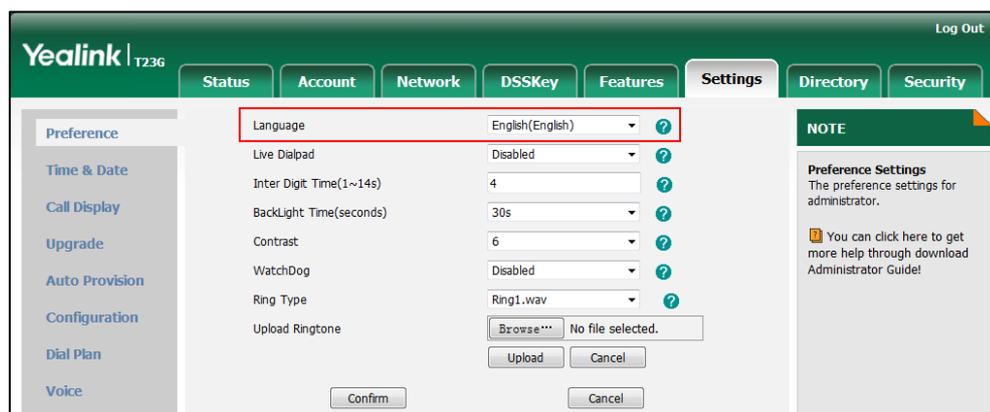


3. Press the **Save** soft key to accept the change.

Text displayed on the LCD screen will change to the selected language.

To change the language for the web user interface:

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



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

Text 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. If the phone cannot obtain the time and date from the Simple Network Time Protocol (SNTP) server, you need to configure the time and date manually. For more information on the SNTP server, contact your system administrator.

To configure the SNTP settings via phone user interface:

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



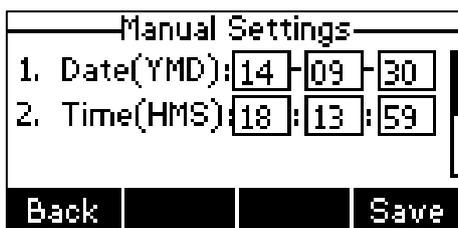
2. Press **◀** or **▶**, or the **Switch** soft key to select the time zone that applies to your area from the **Time Zone** field.
The default time zone is "+8 China(Beijing)".
3. Enter the domain names or IP addresses in the **NTP Server 1** and **NTP Server 2** fields respectively.
4. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Daylight Saving** field.
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Note

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

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

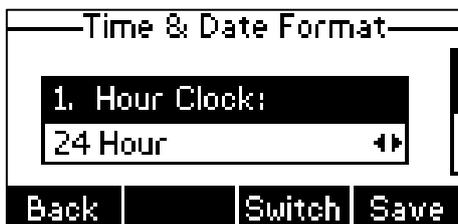
1. Press **Menu->Settings->Basic Settings->Time & Date->Manual Settings**.
2. Enter the specific date in the **Date** field.
3. Enter the specific time in the **Time** field.



4. 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 from the **Hour 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 weekday, “DD” represents the two-digit day, and “MMM” represents the first three letters of the month.

The date formats available:

Date Format	Example(2014-10-26)
WWW MMM DD	Sun Oct 26
DD-MMM-YY	26-Oct-14
YYYY-MM-DD	2014-10-26
DD/MM/YYYY	26/10/2014
MM/DD/YY	10/26/14
DD MMM YYYY	26 Oct 2014
WWW DD MMM	Sun 26 Oct

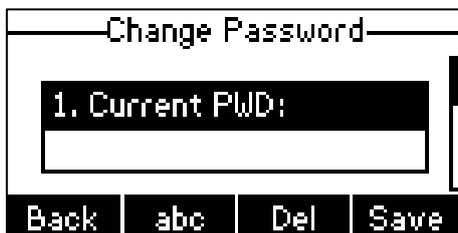
Time and date is configurable via web user interface at the path **Settings->Time & Date**.

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** (default password: admin) ->**Set Password**.



2. Enter the old password in the **Current PWD** field.
3. Enter the new password in the **New PWD** field.
4. Re-enter the new password in the **Confirm PWD** field.
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

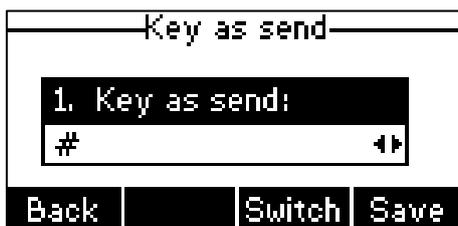
Administrator password is configurable via web user interface at the path **Security->Password**.

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 you to protect your phone from unauthorized use.

Keypad lock consists of the following:

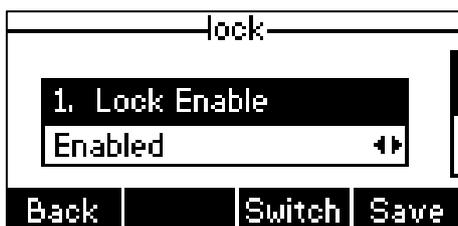
- Menu Key:** The **Menu** soft key is locked. You cannot access the menu of the phone until unlocked.
- Function Keys:** The function keys are locked. You cannot use the MESSAGE, RD, TRAN, OK, X, navigation keys, soft keys and line keys until unlocked.
- All Keys:** All keys are locked except the volume key. You are only allowed to dial emergency numbers, reject incoming calls by pressing the X key, answer incoming calls by lifting the handset, pressing the Speakerphone key or the HEADSET key, and end the call by hanging up the handset, pressing the Speakerphone key .

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** (default password: admin)->**Keypad Lock**.
2. Press  or  , or the **Switch** soft key to select **Enabled** from **Lock Enable** field.



3. Press  or  to highlight the Lock type.



4. Press  or  , or the **Switch** soft key to select the desired lock type.
5. Press the **Save** soft key to accept the change.
6. Long press  to lock the keypad immediately when the phone is idle.

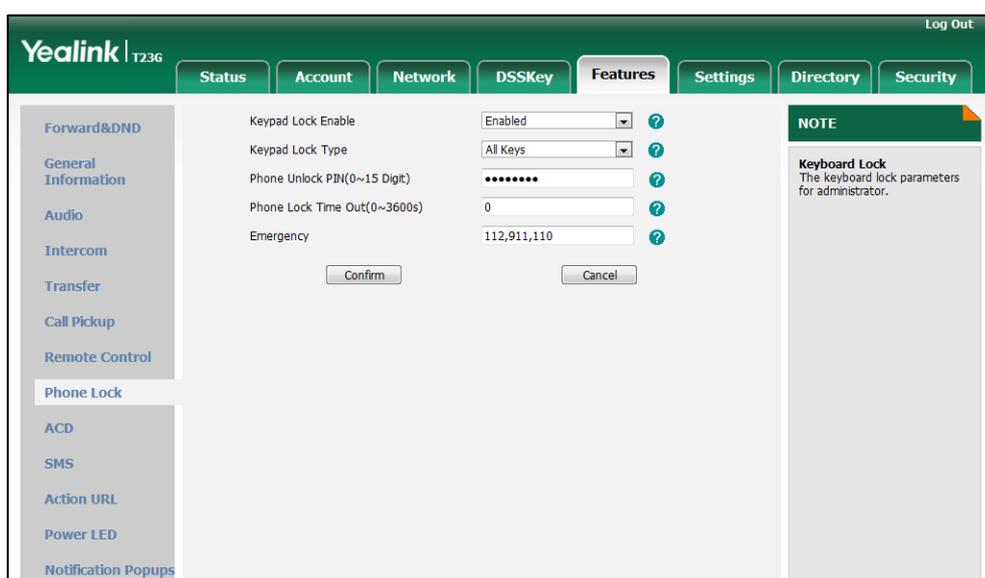
The LCD screen prompts “Keypad locked.” and displays the icon .



You can specify 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 interval in the **Phone Lock Time Out (0~3600s)** field.



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

Note The default interval is 0 second, that is, you need to long press  to lock the keypad. The 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 phone unlock PIN via phone user interface:

1. Press **Menu->Settings->Basic Settings** (default password: admin) ->**Phone Unlock PIN**.



2. Enter the old PIN in the **Current PIN** field.
3. Enter the new PIN in the **New PIN** field.
4. Re-enter the new password in the **Confirm PIN** field.
5. Press the **Save** soft key to accept the change 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 **OK** soft key to unlock the keypad.

The  icon disappears from the LCD screen.

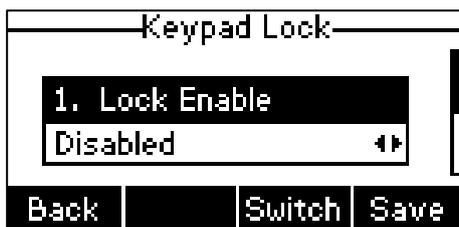
You can long press  or wait for a period of time (if configured) to lock the keypad again.

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

To deactivate the keypad lock via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (default password: admin) ->**Keypad Lock**.

- Press  or , or the **Switch** soft key to select **Disabled** from **Lock Enable** field.



- 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:

- Press  to adjust the ringer volume.

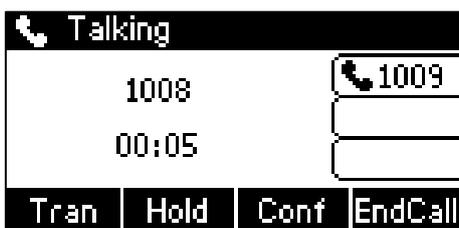


Note

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:

- 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->Sound->Ring Tones->Common**.
2. Press  or  to select the desired ring tone.

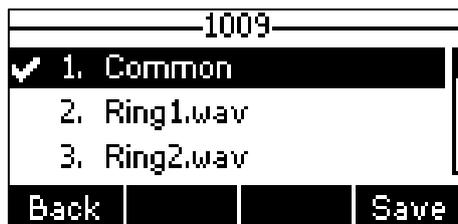


3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

A ring tone for the phone is configurable via web user interface at the path **Settings->Preference->Ring Type**.

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

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

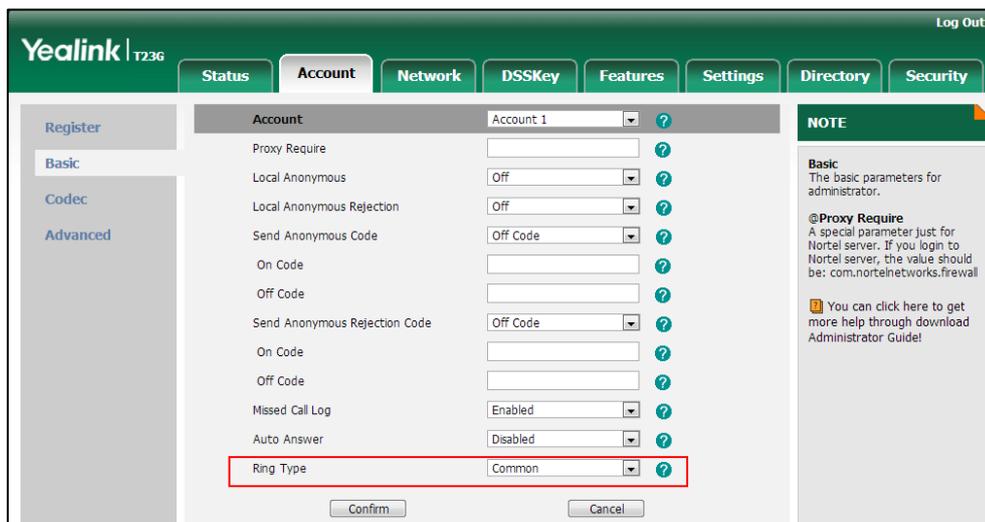


4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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.



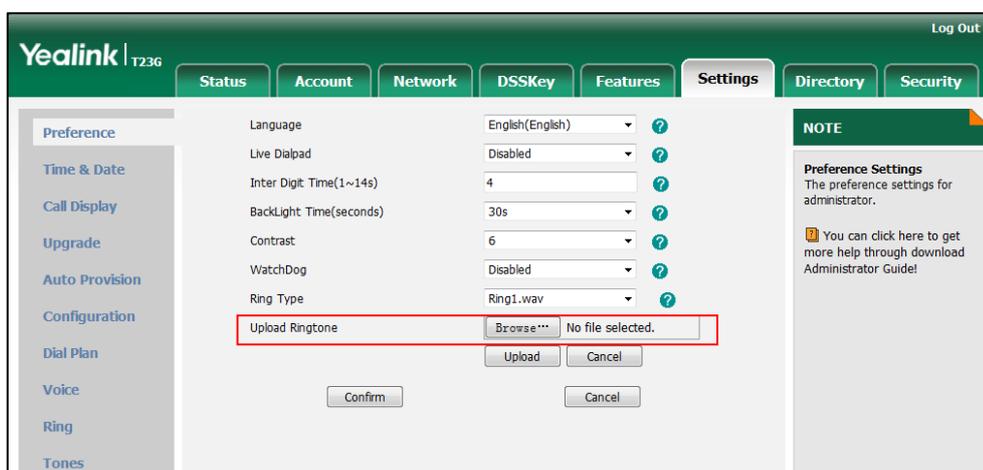
5. Click **Confirm** to accept the change.

Note

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 contact 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, the associated group or account, the phone will play the ring tone assigned to the phone.

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

All custom ring tone files must be no greater than 100KB.

Custom ring tones for your phone can be uploaded 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](#)

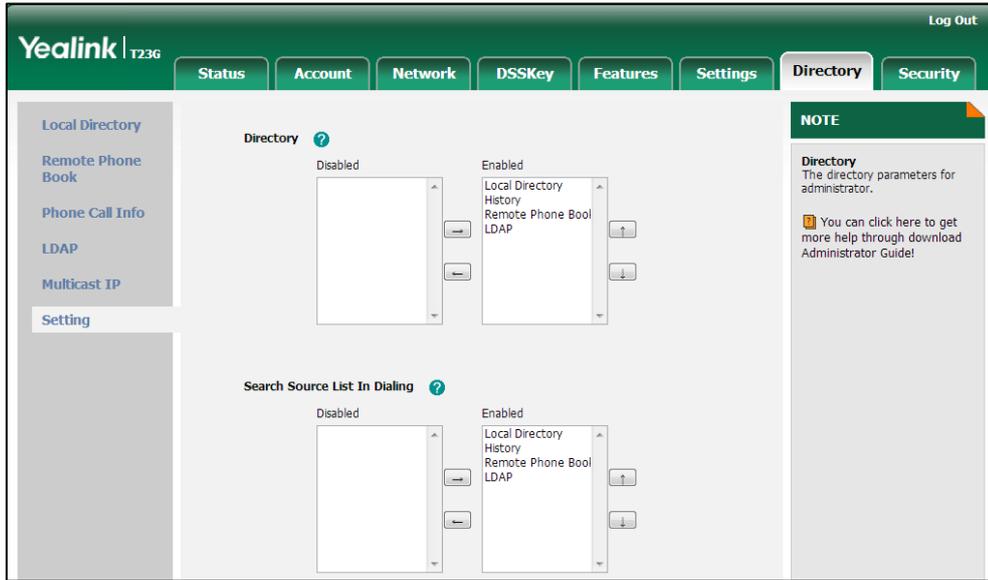
Directory

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 step 2 to add more lists to the **Enabled** column.
4. To remove a list from the **Enabled** column, select the desired list and then click  .
5. To adjust the display order of list, select the desired list and then click  or  .

The LCD screen will display the list(s) in the adjusted order.



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

Note Directory is configurable via web user interface only.

To view the directory via phone user interface:

1. Press the **Dir** soft key when the phone is idle.
The LCD screen displays the list(s) in the directory.



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

Note If the 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 42. For more information on the LDAP, refer to Yealink 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 48 groups in your phone's local directory. 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 **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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 **AddGroup** soft key.
3. Enter the desired group name in the **Name** field.
4. Press **◀** or **▶** to select the desired group ring tone from the **Ring Tones** field.

If **Auto** is selected, this group will use the ring tone specified for the contact. If no ring tone is specified for the contact, it will then play the ring tone specified for the account. For more information on 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.

Editing Groups

To edit a group in the local directory:

1. Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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  or  to select the desired group (e.g., Test).
3. Press the **Option** soft key, and then select **Detail** from the prompt list.



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

Deleting Groups

To delete a group from the local directory:

1. Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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 group (e.g., Test).
3. Press the **Option** soft key, and then select **Delete** from the prompt list.

The LCD screen prompts “Delete selected group?”.



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**. For more information, refer to the above steps.

Adding Contacts

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

- Manually
- From call history
- From a remote phone book

Adding Contacts Manually

To add a contact to the local directory manually:

1. Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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 **Enter** soft key.

If the groups have been added to the local directory, select the desired group and then 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 \leftarrow or \rightarrow , 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 \leftarrow or \rightarrow , or the **Switch** soft key to select the desired ring tone from the **Ring** field.
If **Auto** is selected, this contact will use the ring tone assigned to the group.
7. Press the **Add** soft key to accept the change or the **Back** soft key to cancel.

Note

If the contact already exists 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 call history:

1. Press the **History** soft key.
2. Press \uparrow or \downarrow to select 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 a Remote Phone Book

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

1. Press **Menu->Directory->Remote Phone Book**.
If Remote Phone Book is added to the directory, press **Dir->Remote Phone Book** to

enter the remote phone book.

2. Press \uparrow or \downarrow to select the desired contact.
3. Press the **Option** soft key, and then select **Add to Contacts** from the prompt list.
4. Press the **Save** soft key to save the contact in the local directory.

If the contact already exists 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 42.

Editing Contacts

To edit a contact in the local directory:

1. Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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 **Enter** soft key.
If the groups have been added to the local directory, select the desired group and then press the **Enter** soft key.
3. Press \uparrow or \downarrow to select the desired contact.
4. Press the **Option** soft key, and then select **Detail** from the prompt list.
5. Press \uparrow or \downarrow to switch between different items and then edit.



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

Deleting Contacts

To delete a contact from the local directory:

1. Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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 **Enter** soft key.

If the groups have been added to the local directory, select the desired group and then press the **Enter** soft key.

3. Press **▲** or **▼** to select the desired contact.

4. Press the **Option** soft key, and then select **Delete** from the prompt list.

The LCD screen prompts "Delete selected item?".



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

Placing Calls to Contacts

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

1. Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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 **Enter** soft key.

If the groups have been added to the local directory, select the desired group and then press the **Enter** soft key.

3. Press  or  to select the desired contact.

4. Do one of the following:

- If only one number for the contact is stored in the local directory, press the **Send** soft key to dial out the number.

- If multiple numbers for the contact are stored in the local directory, press the **Send** soft key to display a list of numbers.

Press  or  to select the desired number.

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

Searching for Contacts

To search for a contact in the local directory:

1. Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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  or  to select the **All Contacts** field.

3. Press the **Search** soft key.

4. Enter a few continuous characters of the contact name or continuous numbers of the contact number (office, mobile or other 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.

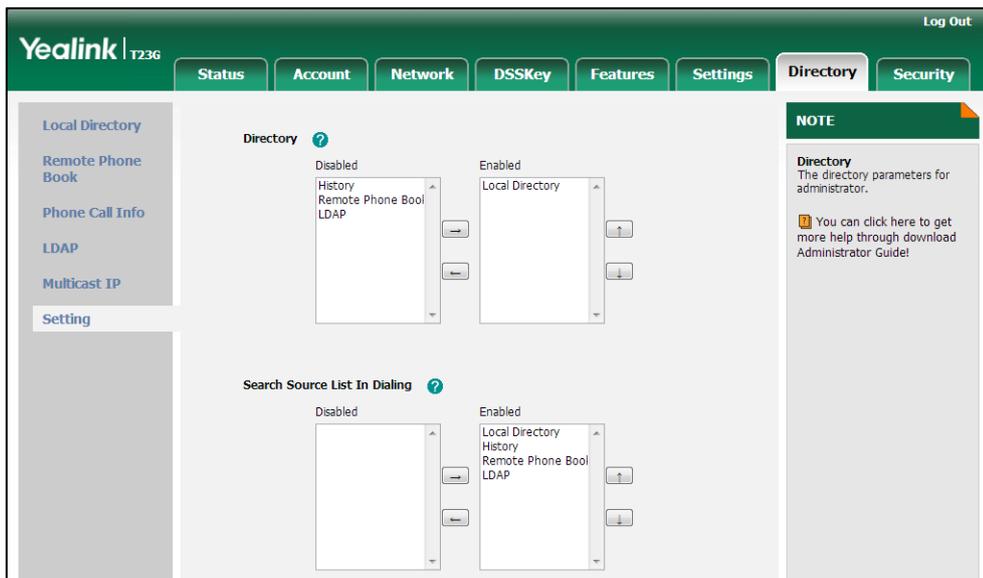
Search Source List in Dialing

You can search for a contact in your desired lists when the phone is in the dialing screen. The lists may contain 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 then 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 then click  .
5. To adjust the display order of search results, select the desired list and then click  or  .

The LCD screen will display search results in the adjusted order.



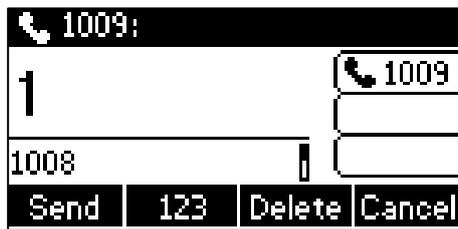
6. Click **Confirm** to accept the change.

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

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

1. Pick up the handset, press the speakerphone or press the line key.
2. Enter a few continuous characters of the contact name or continuous numbers of the contact number (office, mobile or other number) using the keypad.

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

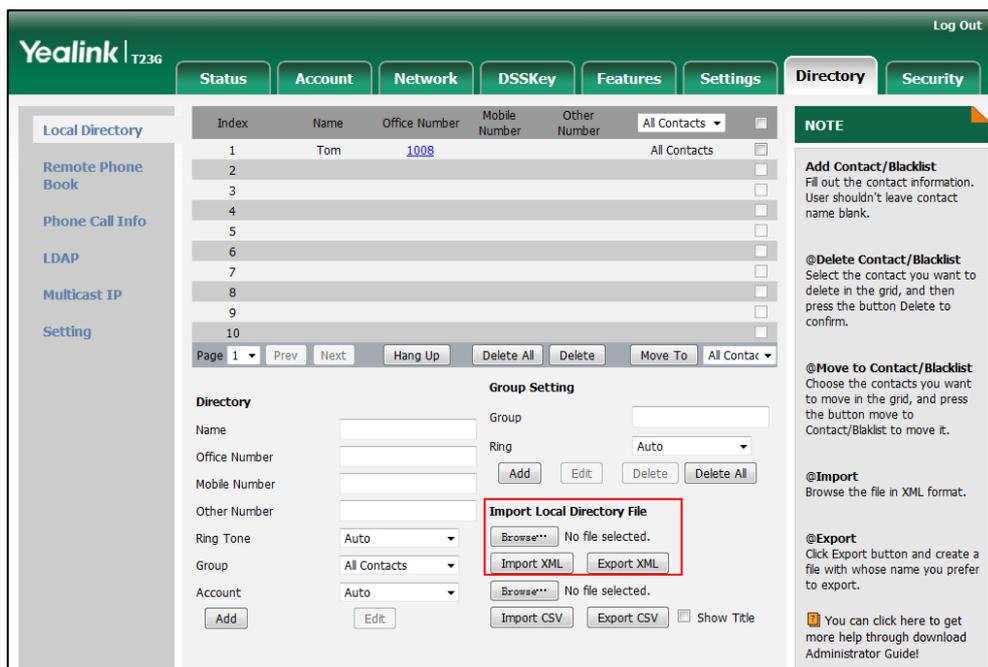


Importing/Exporting Contact Lists

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

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

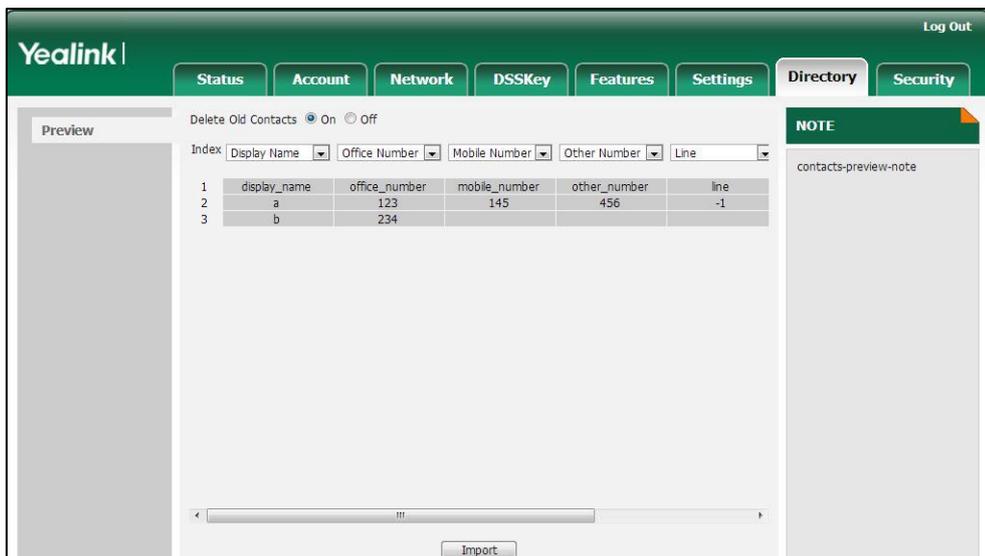
1. Click on **Directory->Local Directory**.
2. Click **Browse** to locate a contact list file (the file format must be *.xml) from your local system.



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 contact list file via web user interface:

1. Click on **Directory**->**Local Directory**.
2. Click **Browse** to locate a contact list file (the 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**.
At least one row information should be selected to be imported into the local directory.



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

To export 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

Contact lists can be imported/exported via web user interface only.

Blacklist

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

To add a contact to the blacklist 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.

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

To add a contact to the blacklist from the local directory:

1. Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory enabled 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 **Enter** soft key.

If the groups have been added to the local directory, select the desired group and then press the **Enter** soft key.

3. Press  or  to select 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 adding, editing, deleting, placing calls to and/or searching for contacts in the blacklist, refer to the operating instructions of [Local Directory](#) on page 30.

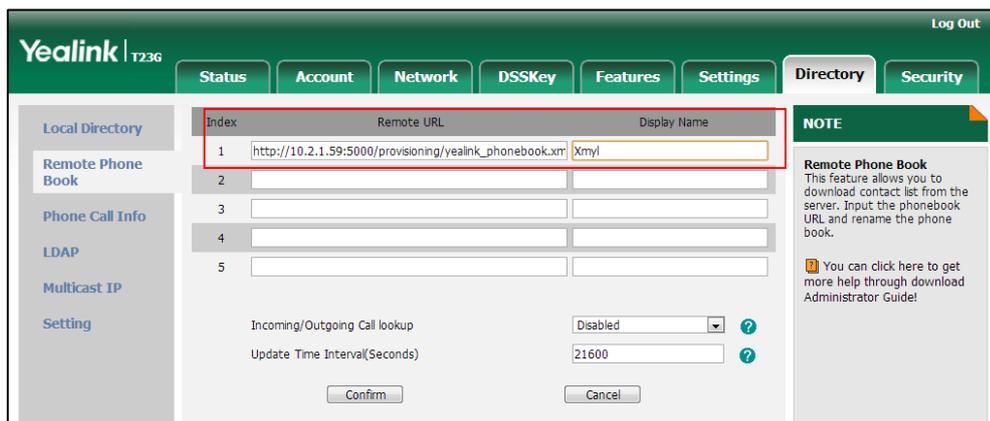
Remote Phone Book

You can access your remote phone book through your phone. You can add new contacts to the local directory, search for a contact, or simply dial a contact number from the remote phone book.

You can configure your new phone to access up to 5 remote phone books. The phone supports up to 2500 remote phone book entries. For the access URL of the remote phone book, contact your system administrator.

To configure an access URL for a 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 a remote phone book is configurable via web user interface only.

To access your remote phone book via phone user interface:

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

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



2. Press  or  to select the desired remote group, and then press the **Enter** soft key.

The phone then connects to the remote phone book and proceeds to load it. The contacts in the remote phone book are displayed on the LCD screen.

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

To search for a contact in the remote phone book:

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

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

2. Select the desired remote group, and then press the **Enter** soft key to load the remote phone book.
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 dial from the result list.

To place a call from the remote phone book:

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

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

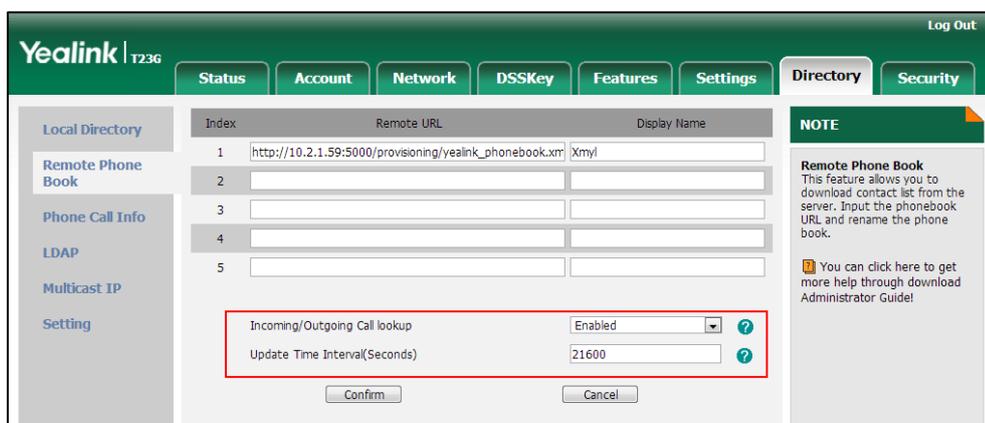
2. Press  or  to select the desired remote group, and then press the **Enter** soft key to load the remote phone book.

3. Select the desired contact in the remote phone book.
4. Press the **Send** soft key.

In addition, you can enable the phone to present the caller identify stored in the remote phone book when receiving a call.

To enable the presentation of caller identify stored in a remote phone book via web user interface:

1. Click on **Directory->Remote Phone Book**.
2. Select **Enabled** from the pull-down list of **Incoming/Outgoing Call lookup**.
3. Enter the desired refresh period in the **Update Time Interval (Seconds)** field.
The default value is 21600 seconds.



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

Call History Management

The SIP-T23(P/G) 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 call history, place a call, add a contact or delete an entry from the call history list. History record feature is enabled by default. If you don't want to save the call history, you can disable the feature.

To disable history record via phone user interface:

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



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

To view call history:

1. Press the **History** soft key.
The LCD screen displays **All Calls** list.
2. Press \leftarrow or \rightarrow to switch between **All Calls**, **Missed Calls** ,**Placed Calls**, **Received Calls**, and **Forwarded Calls**.
3. Press \uparrow or \downarrow 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 \leftarrow or \rightarrow to switch between **All Calls**, **Missed Calls**, **Placed Calls**, **Received Calls**, and **Forwarded Calls**.
3. Press \uparrow or \downarrow to select the desired entry.
4. Press the **Send** soft key.

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

1. Press the **History** soft key.
2. Press \leftarrow or \rightarrow to switch between **All Calls**, **Missed Calls** ,**Placed Calls**, **Received Calls**, and **Forwarded Calls**.
3. Press \uparrow or \downarrow 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, and then press the **Save** soft key.

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 Calls**, **Missed Calls**, **Placed Calls**, **Received Calls**, and **Forwarded Calls**.
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 Calls**, **Missed Calls**, **Placed Calls**, **Received Calls**, and **Forwarded Calls**.
3. Press the **Option** soft key, and then select **Del All** from the prompt list.
4. Press the **OK** soft key.
The LCD screen prompts "Delete all the call records?".
5. Press the **OK** soft key to confirm the deletion or the **Cancel** soft key to cancel.

System Customizations

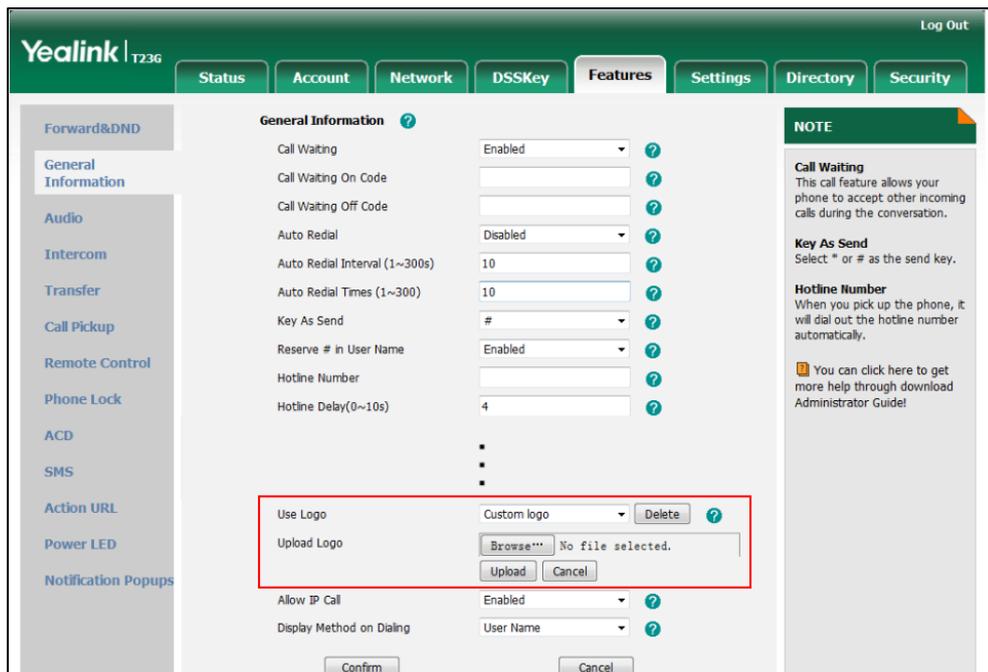
Logo Customization

You can upload your custom logo which will be displayed on the idle screen.

To upload a custom logo via web user interface:

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

2. Select **Custom Logo** from the pull-down list of **Use Logo**.



3. Click **Browse** to select the logo file from your local system.
4. Click **Upload** to upload the file.
5. Click **Confirm** to accept the change.

Note The logo file format must be *.dob, contact your system administrator for more information.
A custom logo can be uploaded via web user interface only.

Headset Use

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:

1. Press  on the phone.
The headset icon on the idle screen indicates that the headset mode is activated. Press the line key or the **Answer** soft key to answer a call. The call will connect to your headset automatically. Enter the desired number and press the **Send** soft key, the phone will then place a call using the headset automatically. For more information on using the headset to place a call, refer to [Placing Calls](#) on page 67.

To deactivate the headset:

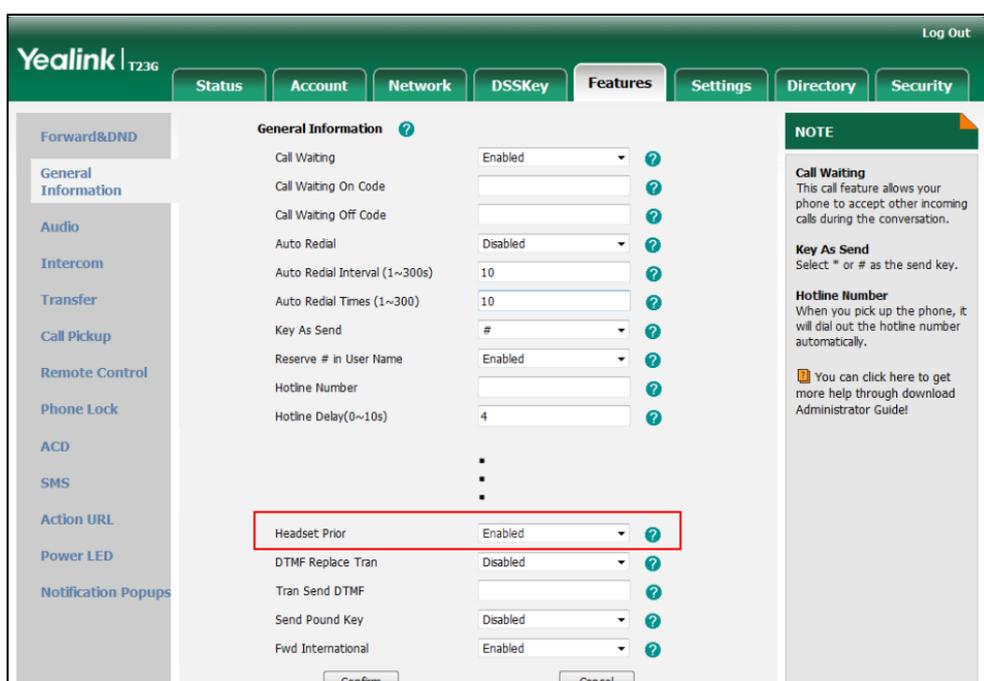
1. Press  again on the phone.
The headset icon disappears from the idle screen when the headset mode is deactivated.

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 activate the headset mode:

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

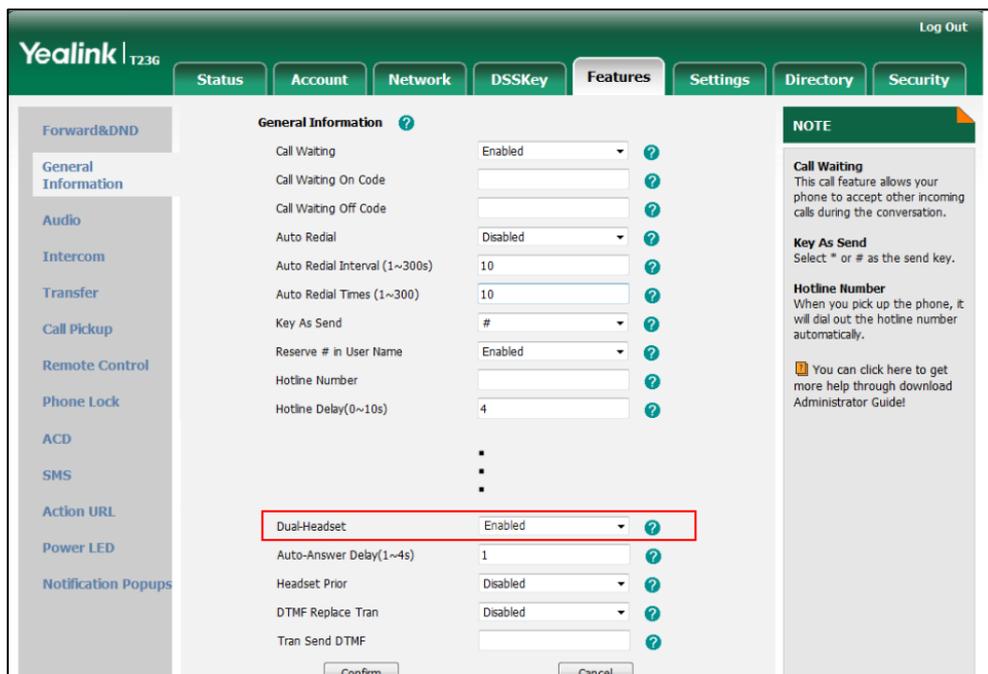
Note If headset prior is enabled, the headset mode will not be deactivated until you press the **HEADSET** key again.
Headset prior is configurable via web user interface only.

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

There are two types of DSS keys: Line Keys and Programmable Keys. The details will be introduced in the following. The SIP-T23(P/G) IP phone supports 3 line keys.

Line Keys

You can assign predefined functionalities to the line keys to have quick access to features such as call return and voice mail. The line key LEDs will indicate status when the keys are assigned with particular features, such as BLF. The default key type of each line key is Line.

To configure a 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 from the **Key Type** field.
5. (Optional.) Select the desired account from the **Account ID** field.
6. (Optional.) Enter the 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.

Common used line key features are explained in the following subchapters in detail:

- Speed Dial
- Voice Mail
- Directed Pickup
- Group Pickup
- DTMF
- Prefix
- Local Group
- XML Group
- XML Brower
- LDAP
- Conf
- Forward
- Tran
- Hold
- DND
- SMS
- Group Listening
- Zero Touch
- Keypad Lock

- Dir

For more information, contact your system administrator.

Speed Dial

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

Dependencies: *Type (Speed Dial)*

Value (the number you want to dial out)

Label (key label displayed on the touch screen)

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

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 quickly connect voice mail. For more information, refer to [Voice Mail](#) on page 121.

Dependencies: *Type (Key Event)*

Key Type (Voice Mail)

Value (the voice mail access code)

Label (key label displayed on the touch screen)

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

Usage: Press the DSS key to dial out the voice mail access code. Then follow the voice prompt to listen to 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 (DPickup)

Value (the directed pickup code followed by the target phone number)

Label (key label displayed on the touch screen)

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

Usage: Press the DSS key on your phone when the target phone number receives an incoming call. The call is then 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 (GPickup)

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

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)

Value (DTMF sequence)

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

Note

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

Prefix

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

Dependencies: *Type (Key Event)*

Key Type (Prefix)

Value (the prefix number)

Usage: Press the DSS key when the phone is idle, the phone will then enter the pre-dialing screen and display the prefix number that you specified in the **Value** field. You can enter the remaining digits and then dial out.

Local Group

You can use this key feature to quickly access a group in the local directory. For more information, refer to [Local Directory](#) on page 30.

Dependencies: *Type (Key Event)*

Key Type (Local Group)

Local Group (the group you want to access)

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

XML Group

You can use this key feature to quickly access a remote group in your remote phone book. You should configure a remote phone book in advance. For more information, refer to [Remote Phone Book](#) on page 42.

Dependencies: *Type (Key Event)*

Key Type (XML Group)

Phone Book Name (the remote group you want to access if the remote phone book is configured)

Usage: Press the DSS key to access the remote group specified in the **Phone Book Name** field.

XML Browser

You can use this key feature to quickly access a XML browser. 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 (XML Browser)*

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 quickly access a LDAP search screen.

Dependencies: *Type (Key Event)*

Key Type (LDAP)

Usage:

1. Press the DSS key to access the LDAP search screen.
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.

Conf

You can use this key feature to set up a conference call. For more information, refer to [Conference](#) on page 89.

Dependencies: *Type (Key Event)*

Key Type (Conf)

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

If the **Value** field is left blank, the DSS key performs the same function 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 81.

Dependencies: *Type (Key Event)*

Key Type (FWD)

Value (the number you want to forward to)

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

Note

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

Tran

When there is an active call on the phone, you can use this key feature to handle the call differently depending on the transfer mode assigned to the DSS key.

Dependencies: *Type (Key Event)*

Key Type (Tran)

Value (the number you want to transfer to)

Usage:

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

Note

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

If the **Value** field is left blank, the DSS key performs the same function as the **TRAN** key or the **Tran** soft key during a call. For more information, refer to [Call Transfer](#) on page 87.

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)

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

Dependencies: *Type (Key Event)*

Key Type (DND)

Usage:

When the DND is in phone mode:

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

When the DND is in custom mode:

1. Press the DSS key to access the custom DND screen.
2. Activate or deactivate DND for one or all accounts.

Note

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

SMS

You can use this key feature to quick access text message. For more information, refer to [Short Message Service \(SMS\)](#) on page 118.

Dependencies: *Type (Key Event)*

Key Type (SMS)

Usage: Press the DSS key when the phone is idle to access text message.

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 conversations which have more than one person present at one end. You are able to speak and listen through the handset/headset, while the others nearby can only listen through the speaker.

Dependencies: *Type (Key Event)*

Key Type (Group Listening)

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, while other people at your side can only listen through the 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 quickly configure auto provision and network parameters.

Dependencies: *Type (Key Event)*

Key Type (Zero Touch)

Usage:

1. Press the DSS key to access the zero touch screen.
2. Press the **OK** soft key in 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.

Keypad Lock

You can use this key feature to immediately lock the keypad of your phone instead of long pressing . For more information, refer to [Keypad Lock](#) on page 22.

Dependencies: *Type (Keypad Lock)*

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

Directory

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

Dependencies: *Type (Dir)*

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

Note

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

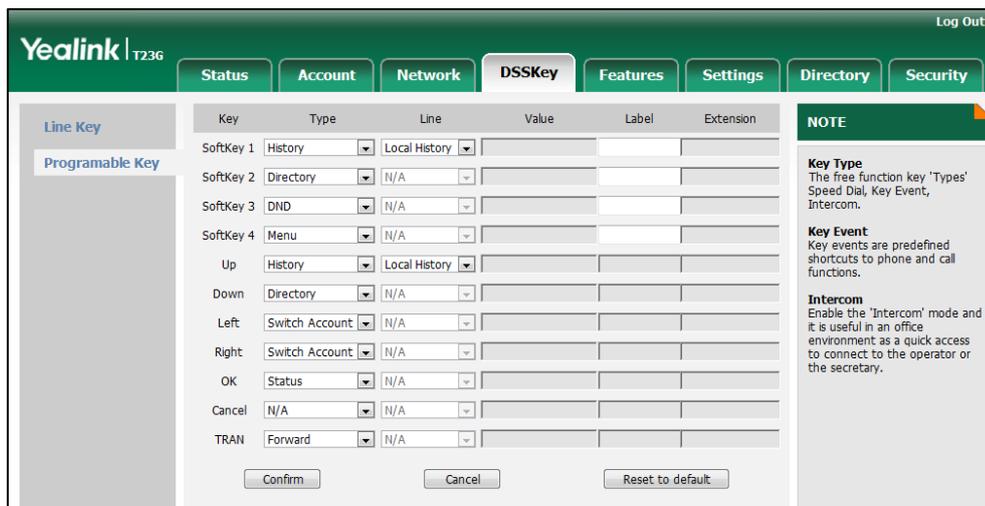
Programmable Keys

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

To customize 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.

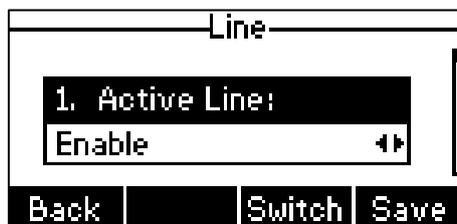
Account Management

You can register one or multiple accounts on the SIP-T23(P/G) 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** (default password: admin) ->**Accounts**.
2. Select the desired account and then press the **Enter** soft key.
3. Select **Enable** from the **Active Line** field.



4. Enter the desired values in the **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.

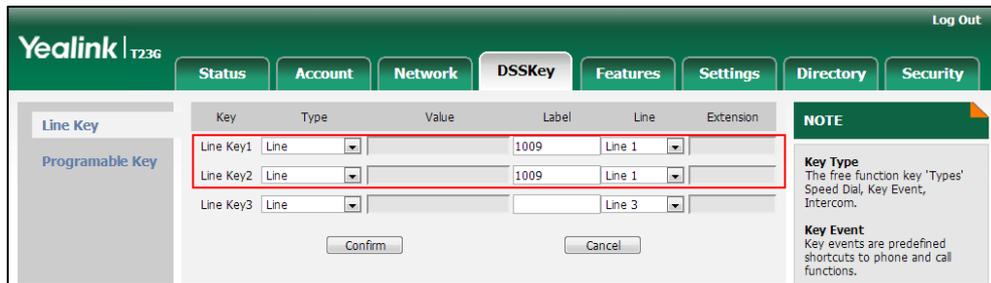
To disable an account via phone user interface:

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

Account registration 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 will resemble 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 the 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-T23(P/G) IP phone processes the inputs received from your phone keypad. The SIP-T23(P/G) IP phone supports the following dial plan features:

- [Replace Rule](#)
- [Dial-now](#)

- [Area Code](#)
- [Block Out](#)

The basic expression syntax you need to know:

.	The dot "." can be used as a placeholder or multiple placeholders for any character. Example: "12." would match "123", "1234", "12345", "12abc", etc.
x	An "x" can be used as a placeholder for any character. Example: "12x" would match "121", "122", "123", "12a", etc.
[]	The square brackets "[]" can be used as a placeholder for a single character which matches any of a set of characters. Example: "91[5-7]1234" would match "9151234", "9161234", "9171234".
()	The parentheses "()" can be used to group together patterns, for instance, to logically combine two or more patterns. Example: "91([5-7])1(x)" would match "91511", "91618", "91715", 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: "001(xxx)45(xx)", Replace: "9001\$145\$2". When you dial out "0012354599" on your phone, the IP phone will replace the number with "9001 235 45 99 ". "\$1" means 3 digits in the first parenthesis, that is, "235". "\$2" means 2 digits in the second parenthesis, that is, "99".

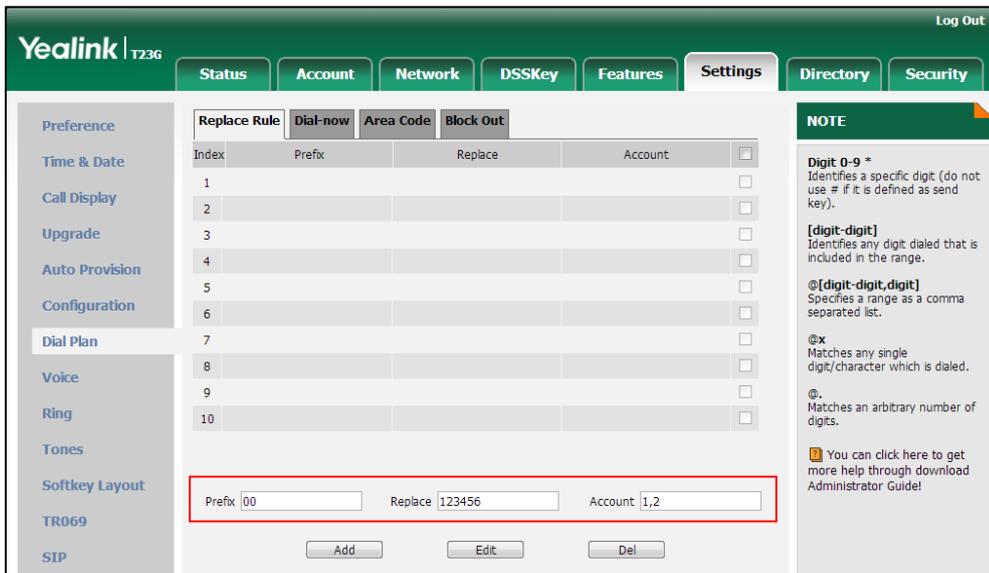
Replace Rule

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 (refer to the expression syntax in 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: 00" and "Replace: 123456", when you try to dial out the number "123456", you just need to enter "00" 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., 00) in the **Prefix** field.
3. Enter the string (e.g., 123456) 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 “00” using the keypad and then press the **Send** soft key, the phone will dial out “123456” instead.

Note

The valid values for the **Account** field can be one or more digits among 1, 2 and 3. Each digit 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 account1 and account2.

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 check box.
3. Edit the values in the **Prefix** and **Replace** fields.
4. Enter the desired line ID in the **Account** field or leave it blank.
5. Click **Edit** to accept the change.

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

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Select one or more replace rules by checking the check box(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 to 100) on your phone. When the dialed out number matches the dial-now string, the number will be dialed out automatically. For example, a dial-now rule is configured as “1xx”, any entered three-digit string beginning with 1 will then 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., 123) in the **Rule** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

For more information on the valid values for the **Account** field, refer to [Replace Rule](#) on page 59.

The screenshot shows the Yealink T236 web interface. The 'Settings' menu is open, and 'Dial Plan' is selected. Under 'Dial Plan', the 'Dial-now' tab is active. A table lists 10 dial-now rules. The 'Rule' field is set to '123' and the 'Account' field is set to '1,2'. A red box highlights these two fields. Below the table are 'Add', 'Edit', and 'Del' buttons. On the right, a 'NOTE' section explains symbols: [digit-0-9] for a specific digit, [digit-digit] for a range, @[digit-digit,digit] for a list, @x for a single digit/character, and @ for an arbitrary number of digits.

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

When you enter the number “123” using the keypad, the phone will dial out “123” automatically without the pressing of 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.

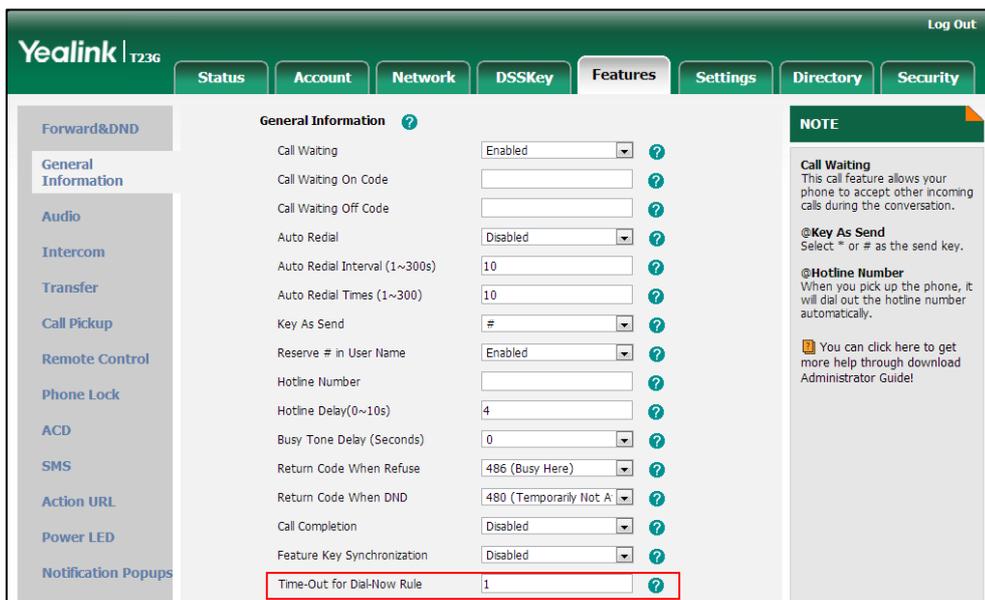
Delay Time for Dial-now Rule

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

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

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

2. Enter the time between 1 and 14 (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 when dialing a phone number outside the code area. For example, an area code is configured as "Code: 0592, Min Length: 1, Max Length: 15". When you dial out the number "56789" (the length of the number is between 1 and 15), the phone will add the area code and dial out the number "059256789". You can only configure one area code rule on your phone.

To configure the area code 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 values for the **Account** field, refer to [Replace Rule](#) on page 59.

The screenshot shows the Yealink T236 web interface. The 'Settings' menu is open, and the 'Block Out' tab is selected. The configuration form contains the following fields:

Code	0592
Min Length (1-15)	1
Max Length (1-15)	15
Account	1

Below the form are 'Confirm' and 'Cancel' buttons. To the right, a 'NOTE' section explains various digit patterns: Digit 0-9, [digit-digit], @[digit-digit,digit], @x, and @.

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.

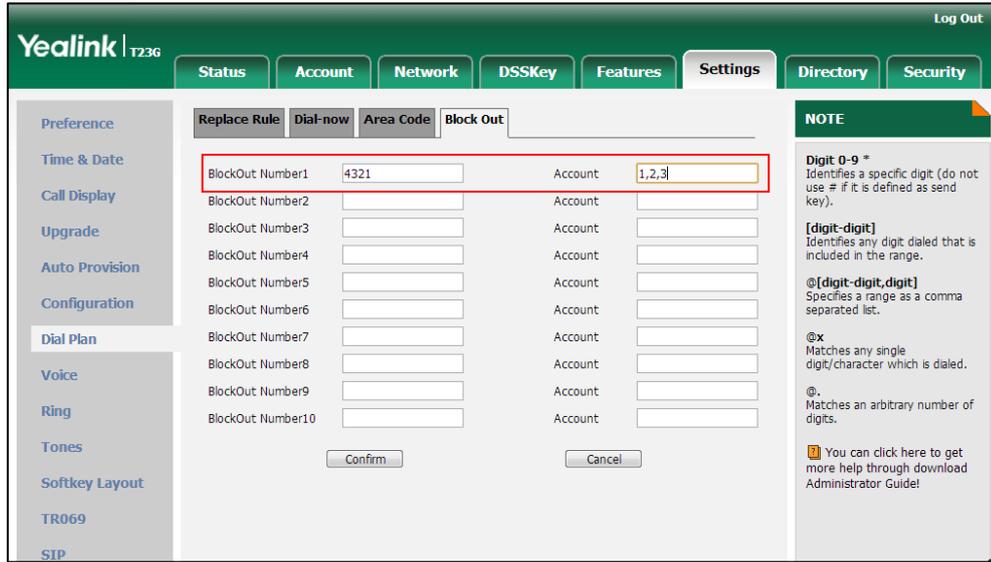
Block Out

You can block specific numbers 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". You can add 10 block out rules at most on your phone.

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 values for 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 different emergency numbers for different emergency services.

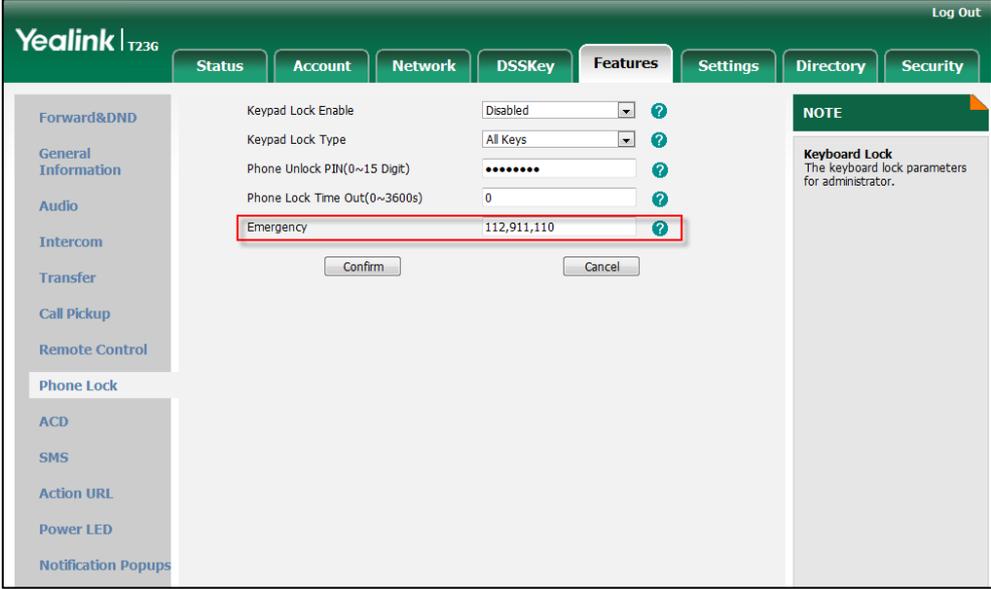
You can specify the emergency telephone number on the IP phone for contacting the emergency services in an emergency situation.

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 numbers, enter a comma between every two emergency numbers. The default emergency numbers are 112, 911 and 110.



The screenshot shows the Yealink T236 web interface. The 'Features' tab is selected, and the 'Emergency' field is highlighted with a red border. The field contains the text '112,911,110'. Other fields in the 'Features' section include 'Keypad Lock Enable' (set to 'Disabled'), 'Keypad Lock Type' (set to 'All Keys'), 'Phone Unlock PIN(0~15 Digit)' (masked with dots), and 'Phone Lock Time Out(0~3600s)' (set to '0'). A 'NOTE' box on the right side of the interface reads: 'NOTE: Keyboard Lock. The keyboard lock parameters for administrator.'

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

Note Emergency number is configurable via web user interface only.

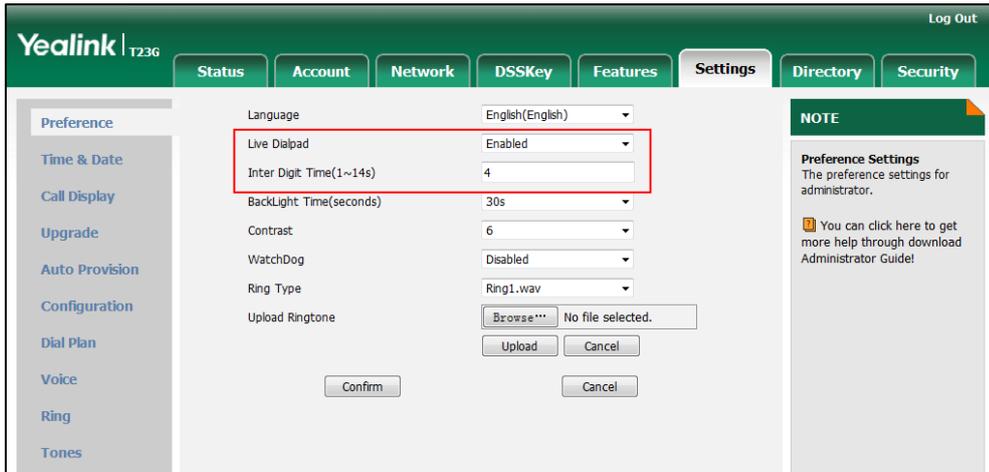
Live Dialpad

You can enable live dialpad on the SIP-T23(P/G) IP phone, which enables the IP phone to automatically dial out a phone number without the pressing of the send key. You can also configure a delay, where the phone will dial out the hotline number automatically after the designated 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.

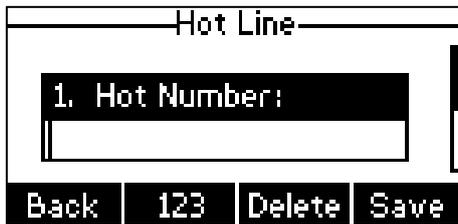
Note Live dialpad is configurable via web user interface only.

Hotline

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

To configure hotline number via phone user interface:

1. Press **Menu->Features->Hot Line**.
2. Enter the desired number in the **Hot Number** field.



3. Enter the delay time (in seconds) in the **HotLine Delay** field.
The valid values range from 0 to 10.
4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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

Basic Call Features

The SIP-T23(P/G) 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-T23(P/G)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 one of three ways using your SIP-T23G/SIP-T23P/T23 IP phone:

- Using the handset

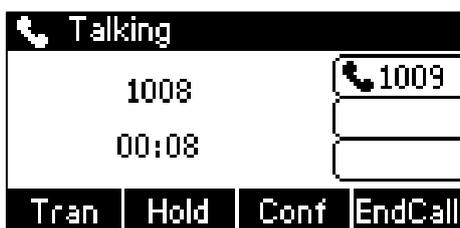
- Using the speakerphone
- Using the headset

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 44.

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

The call duration of the active call is visible on the LCD screen. In the figure below, the call has lasted 25 seconds.



To place a call using the handset:

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

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 [Key as send](#) on page 22.

Note

You can also dial using the SIP URI or IP address. To obtain the IP address of the phone, press the **OK** key when the phone is idle. The maximum SIP URI or IP address length is 32 characters. For example, SIP URI: 2210@sip.com, IP: 192.168.1.15.

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

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

Do one of the following:

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

To place a call using the headset:

Do one of the following:

- With the optional headset connected, press  to activate the headset mode.
Press the line key to obtain a dial tone.
Enter the desired number using the keypad.
Press , , or the **Send** soft key.
- With the optional headset connected, press  to activate the headset mode.
Enter the desired number using the keypad.
Press , , or the **Send** soft key.

Note

To permanently activate the headset mode, refer to [Headset Prior](#) on page 48.

The SIP-T23G IP phone can handle multiple calls at a time. However, only one active call (the call that has audio associated with it) can be in progress at any given time. The SIP-T23G IP phone can handle a maximum of 6 calls at one time.

To place multiple calls:

You can have more than one call on your SIP-T23(P/G) IP phone. To place a new call during an active call, do one of the following:

- Press the line key. The active call is placed on hold.
Enter the desired number using the keypad.
Press , , or the **Send** soft key.
- Press the **Hold** soft key to place the original call on hold.
Press the **NewCall** soft key.
Enter the desired number using the keypad.
Press , , or the **Send** soft key.

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

Answering Calls

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

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

Note

You can ignore incoming calls by pressing the **Reject** soft key or the **Silence** soft key. You can also activate Do Not Disturb mode to ignore all incoming calls without ringing 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 81.

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

To answer a call using the headset:

Do one of the following:

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

Answering When in Another Call

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

- Press the **Answer** soft key.
The incoming call is answered and the original call is placed on hold.
- Press  to access the new call.
Press  or the **Answer** soft key.
The incoming call is answered and the original call is placed on hold.

Ending Calls

To end a call:

Do one of the following:

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

- If you are using the speakerphone, press  or the **EndCall** soft key.

Note

To end a call placed on hold, you can press the **EndCall** 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:

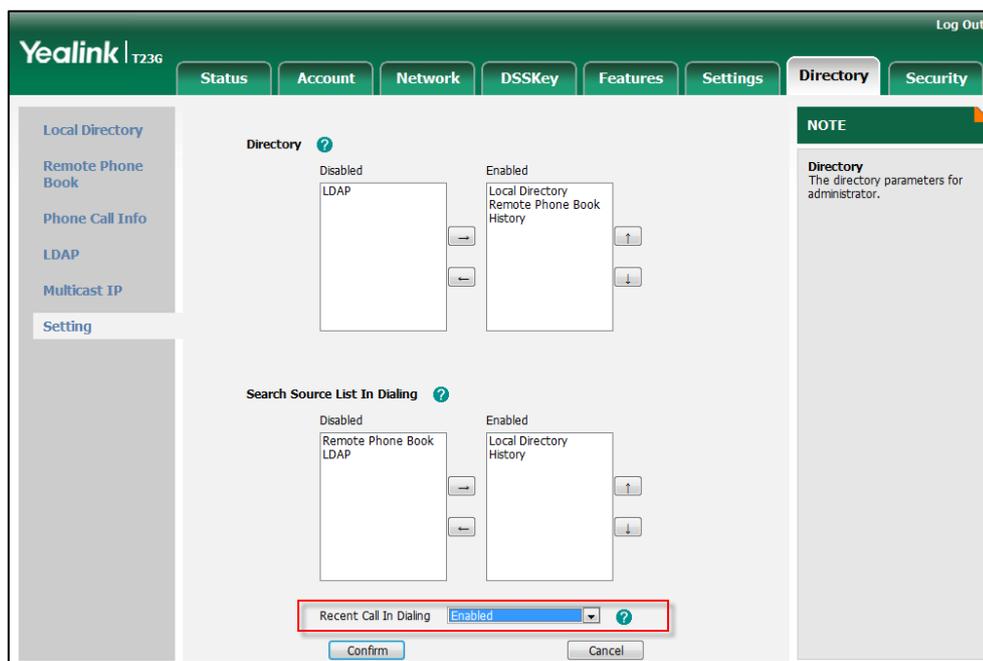
1. Press  when the phone is idle.
2. Press  or  to select the desired entry from the placed call list, and then press  or the **Send** soft key.

Recent Call In Dialing

You can view the placed calls list when the phone is in the pre-dialing screen. To do this, 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**.



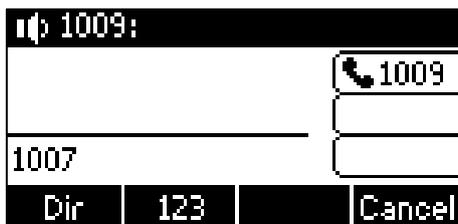
3. Click **Confirm** to accept the change.

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 screen:

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

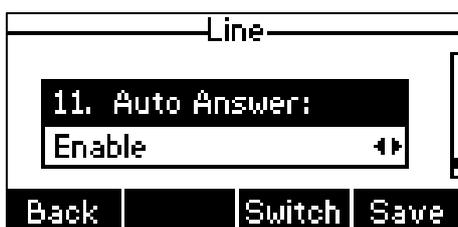


Auto Answer

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**->**Settings**->**Advanced Settings** (default password: admin)
->**Accounts**.
2. Select the desired account and then press the **Enter** soft key.
3. Press **▲** or **▼** to select the **Auto Answer** field.
4. Press **◀** or **▶**, or the **Switch** soft key to select **Enable** from the **Auto Answer** field.



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

To configure auto redial via phone user interface:

1. Press **Menu->Features->Auto Redial**.



2. Press \leftarrow or \rightarrow , or the **Switch** soft key to select **Enable** from the **Auto Redial** field.
3. Enter the desired time in the **Redial Interval** field.
The default time interval is 10 seconds.
4. Enter the desired number of redial attempts in the **Redial Times** field.
The default times are 10.
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

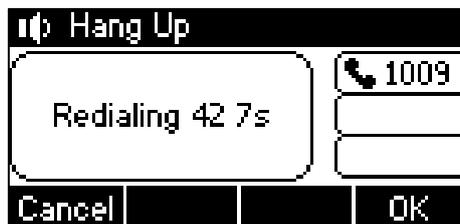
Auto redial is configurable via web user interface at the path **Features->General Information**.

To use auto redial:

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



1. Press the **OK** soft key to activate auto redial. The following prompt will appear on the LCD screen of the phone:



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

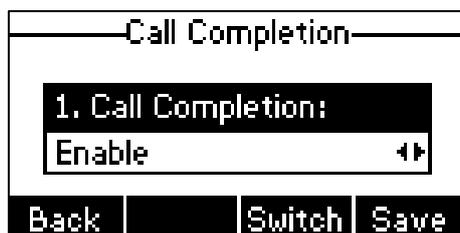
The phone will retry as many times as configured until the called party is idle.

Call Completion

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

To configure call completion via phone user interface:

1. Press **Menu->Features->Call Completion**.

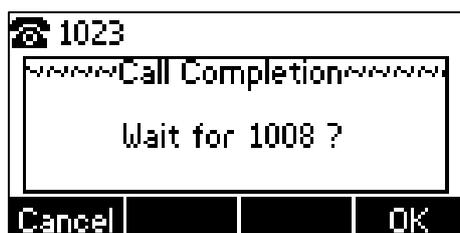


2. Press **◀** or **▶**, or the **Switch** soft key to select **Enable** from the **Call Completion** field.
3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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

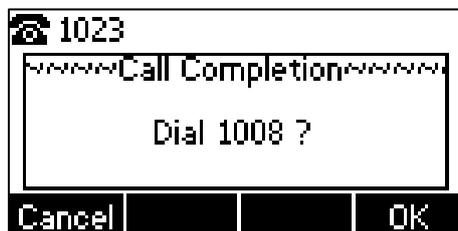
To use call completion:

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

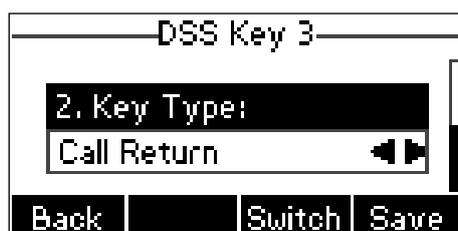
Call completion is not available on all servers. For more information, contact your system administrator.

Call 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 the 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 so that the other party cannot hear you. Call mute applies to all modes (Handset, Headset and Speakerphone).

To mute a call:

1. Press  during an active call.

The LCD screen indicates that the call is now muted.

**To un-mute a call:**

1. Press  again to un-mute the call.

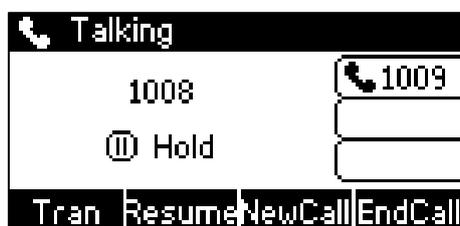
Call Hold/Resume

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 to the other party while waiting.

To place a call on hold:

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

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

**Note**

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

To resume a held call:

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

Multiple Calls on Hold:

If multiple calls are placed on hold, 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 desired call.

A numbered prompt appears on the LCD screen, for example "2/3", indicating that this is the second call out of three calls.

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

Do Not Disturb (DND)

You can use DND to reject incoming calls automatically on the phone. The prompt message "**n New Missed Call(s)**" ("n" indicates the number of missed calls) will appear on the LCD screen, and callers will receive a busy message. All calls you receive while DND is enabled are logged to your missed calls list.

Note

The prompt message will display 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.

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

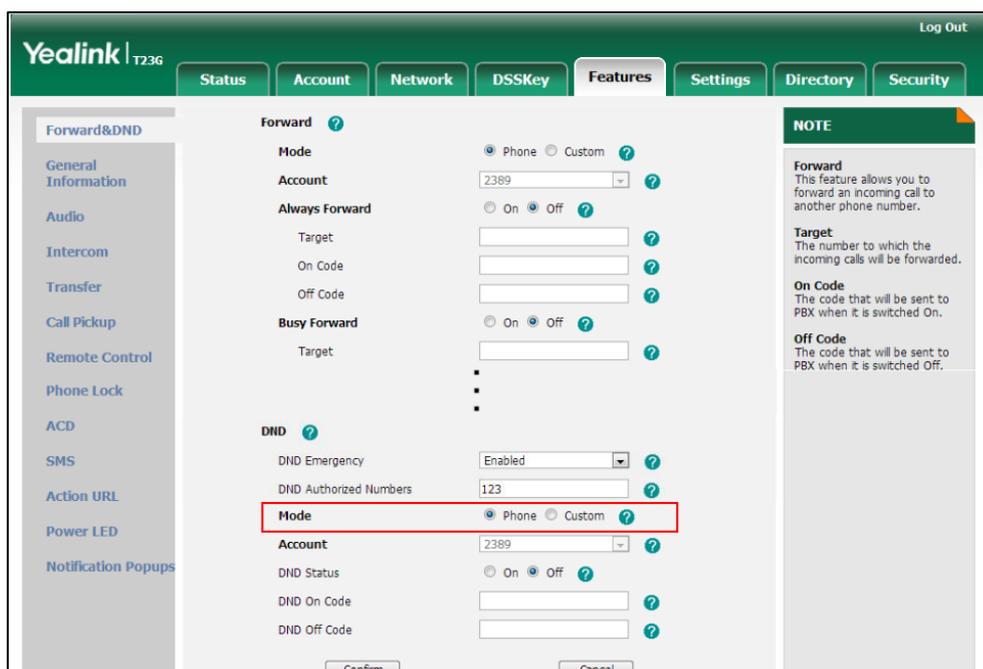
- **Phone** (default): DND is effective for the phone system.
- **Custom**: DND can be configured for each or all accounts.

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**.

- In the **DND** block, mark the desired radio box in the **Mode** field.



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

Note DND mode is configurable via web user interface only.

To activate DND in phone mode:

- 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 New Missed Call(s)" ("n" indicates the number of missed calls) will appear on the LCD screen.



Note When DND and busy forward are enabled in phone mode, calls will be sent to the configured destination number. For more information on busy forward, refer to [Call Forward](#) on page 81.

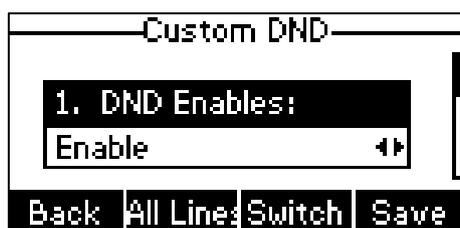
To activate DND in custom mode for a specific account:

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

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



2. Press  or  to select the desired account, then press the **Enter** soft key.
3. Press  or  soft key to select **Enable** in **DND Enables** field.



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

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

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



Note

When DND and busy forward are enabled for a specific account, calls from the specific account will be sent to the configured destination number. For more information on busy forward, refer to [Call Forward](#) on page 81.

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 the accounts 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 **⊖**.

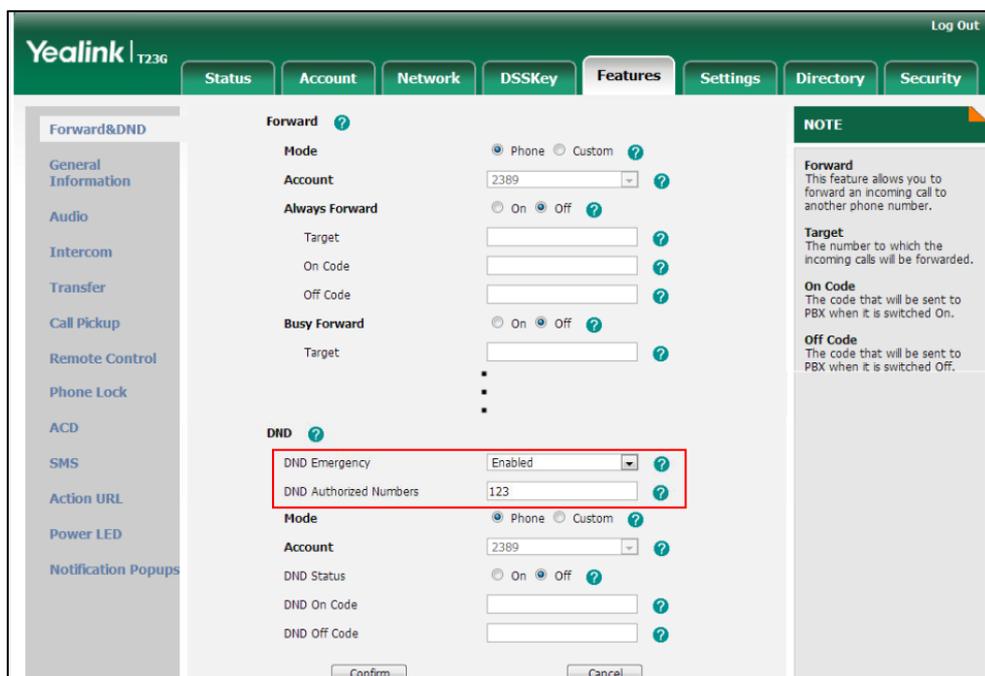
Incoming calls will be rejected automatically and "n New Missed Call(s)" ("n" indicates the number of missed calls) will prompt on the LCD screen.



To configure the DND authorized numbers via web user interface:

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

For multiple numbers, enter a comma between 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.
 When the phone receives a voice mail, the voice mail prompt window will pop up by default, if you want to disable the feature, contact your system administrator for more information.

Call Forward

You can configure your phone to forward incoming calls to another party through static forwarding. You can also forward calls while your phone is ringing; refer to 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 system, 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.

The screenshot shows the Yealink T236 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Features' tab is active, and the 'Forward&DND' sub-tab is selected. The main content area is divided into 'Forward' and 'DND' sections. In the 'Forward' section, the 'Mode' field has two radio buttons: 'Phone' (selected) and 'Custom'. Below this are fields for 'Account' (2389), 'Always Forward' (On/Off), 'Target', 'On Code', and 'Off Code'. The 'DND' section includes 'DND Emergency' (Enabled), 'DND Authorized Numbers' (123), and another 'Mode' field (Phone/Custom) with 'Account' (2389), 'DND Status' (On/Off), 'DND On Code', and 'DND Off Code'. A 'NOTE' panel on the right explains the 'Forward' feature and defines 'Target', 'On Code', and 'Off Code'. 'Confirm' and 'Cancel' buttons are at the bottom.

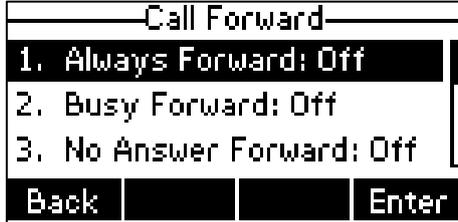
3. Click **Confirm** to accept the change.

Note

Call forward mode is configurable via web user interface only.

To enable call forward in phone mode:

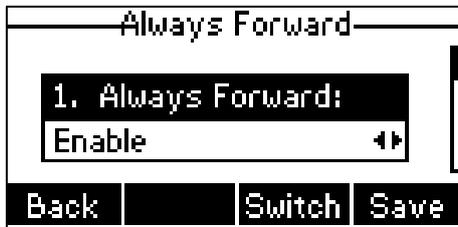
1. Press **Menu->Features->Call Forward**.
2. Press  or  to select the desired forwarding type, and then press the **Enter** soft key.



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.

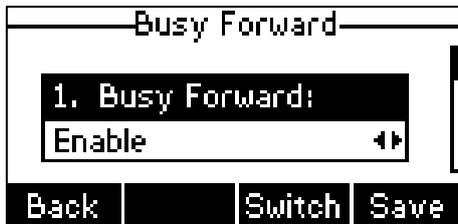


- 2) Enter the destination number you want to forward incoming calls to in the **Forward to** field.
- 3) (Optional.) Enter the always forward on code and off code respectively in the **On Code** and **Off Code** field.

Always forward on code/ off code is used to activate/deactivate the server-side always forward feature when the user enables/ disables always forward feature on the phone.

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

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



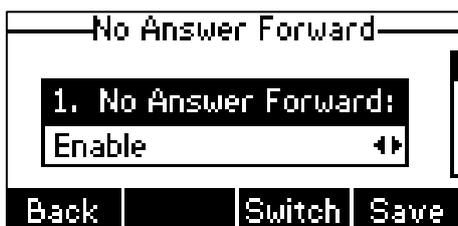
- 2) Enter the destination number you want to forward incoming calls to when the phone is busy in the **Forward to** field.
- 3) (Optional.) Enter the busy forward on code and off code respectively in the

On Code and Off Code field.

Busy forward on code/ off code is used to activate/deactivate the server-side always forward feature when the user enables/ disables busy forward feature on the phone.

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 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 and off code respectively in the **On Code** and **Off Code** field.

No answer forward on code/ off code is used to activate/deactivate the server-side always forward feature when the user enables/ disables no answer forward feature on the phone.

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

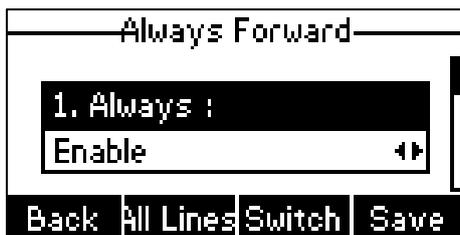
The  icon on the LCD screen indicates that call forward is enabled.

**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. The options are:
4. Depending on your selection:
 - a.) If you select **Always Forward**, you can enable it for a specific account.

- 1) Press  or  , or the **Switch** soft key to select **Enable** from the **Always** field.
- 2) Enter the destination number you want to forward incoming calls to in the **Forward to** field.
- 3) (Optional.) Enter the always forward on code and off code respectively in the **On Code** and **Off Code** field.

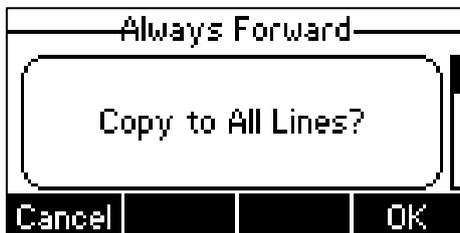
Always forward on code/ off code is used to activate/deactivate the server-side always forward feature when the user enables/ disables always forward feature for this account.



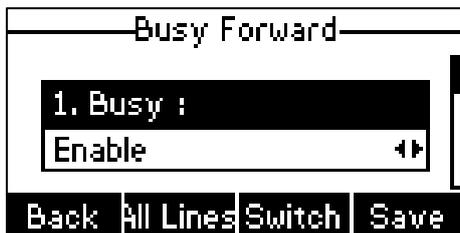
You can also enable always forward for all accounts. After always forward was enabled for a specific account, do the following:

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

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



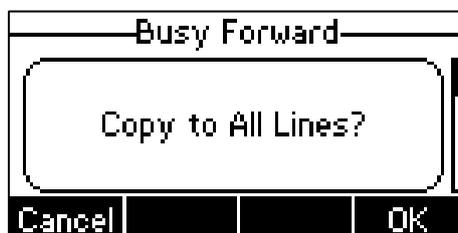
- 3) Press the **OK** soft key to accept the change or the **Cancel** soft key to cancel.
- b.) If you select **Busy Forward**, you can enable it for a specific account.
- 1) Press  or  , or the **Switch** soft key to select **Enable** from the **Busy** field.
 - 2) Enter the destination number you want to forward incoming calls to when the phone is busy in the **Forward to** field.
 - 3) (Optional.) Enter the busy forward on code and off code respectively in the **On Code** and **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 select the **Busy** field.
- 2) Press the **All Lines** soft key.

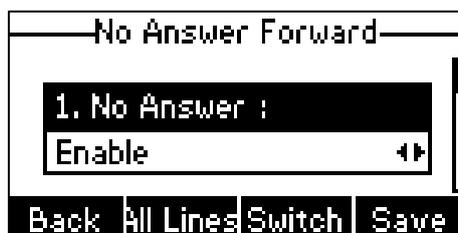
The LCD screen prompts "Copy to All Lines?".



- 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** field.
 - 2) Enter the destination number you want to forward 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 Times** field.

The default ring time is 12 seconds.

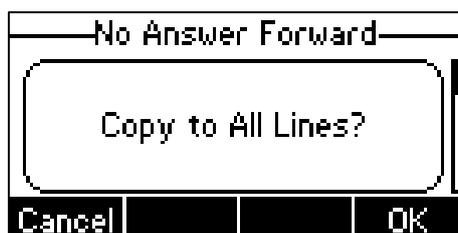
- 4) (Optional.) Enter the no answer forward on code and off code respectively in the **On Code** and **Off Code** field.



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 select the **No Answer** 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 associate 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 67.

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:

Do one of the following:

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

Dynamic Forwarding

To forward an incoming call to another party:

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

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



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

The incoming call is forwarded. The LCD screen prompts a call forward message.

Note

When the phone forwards a call, the forward call prompt window will pop up by default, if you want to disable the feature, contact your system administrator for more information.

Call Transfer

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

- Blind Transfer:** Transfer a call directly to another party without consulting.
- Semi-Attended Transfer:** Transfer a call when receiving ringback.
- Attended Transfer:** Transfer a call with prior consulting.

To perform a blind transfer:

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

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

To perform a semi-attended transfer:

- Press  or the **Tran** soft key during a call.
- Do one of the following:
 - Enter the number you want to transfer the call to.
 - Press the **Dir** 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 **Dir** soft key, and then select **History**. Select the desired list and use  or  to select the entry (Directory should be configured in advance. Refer to [Directory](#) on page 29 for more information.).
 - Press the **Dir** 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 42 for more information.).

3. Press the **Send** soft key,  or  to dial out.
4. Press  or the **Tran** soft key to complete the transfer when receiving the ringback.

To perform an attended transfer:

1. Press  or 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 **Dir** 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 **Dir** soft key, and then select **History**. Select the desired list and use  or  to select the entry (Directory should be configured in advance. Refer to [Directory](#) on page 29 for more information.).
 - Press the **Dir** 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 42 for more information.).
3. Press the **Send** soft key,  or  to dial out.
4. After the party answers the call, press  or the **Tran** soft key to complete the transfer.

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

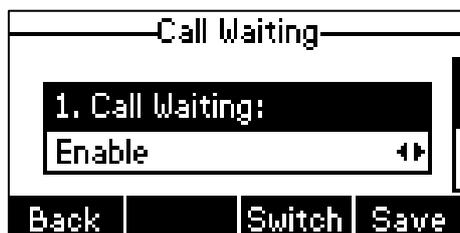
You can cancel the transfer before the call is connected by pressing the **Cancel** soft key.

Call Waiting

You can enable or disable call waiting on the phone. If call waiting is enabled, you can receive another call while there is already 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. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.
5. (Optional.)Enter the call waiting on code and off code respectively in the **On Code** and **Off Code** field.

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

Conference

You can create a conference with other two parties using the phone's local conference. You can create a conference between an active call and a call on hold (on the same or another line) by pressing the **Conf** soft key.

Note

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

Local Conference

The SIP-T23(P/G) 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 , , or the **Send** soft key.



4. When the second party answers the call, you can consult with him or her before adding to the conference.
5. 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 ensure that 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 the selected account.

During the conference call, you can do the followings:

- 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 on hold.
- 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. The muted party can hear everyone, but no one can hear the muted party.
 - 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 **EndCall** soft key to drop the conference call.

Network Conference

You can use network conference on the SIP-T23(P/G) 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.

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

The screenshot shows the Yealink T236 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' section is active, showing a list of parameters for 'Account 1'. The 'Advanced' tab is selected on the left sidebar. In the main configuration area, the 'Conference Type' dropdown is set to 'Network Conference' and the 'Conference URI' text field contains 'conference@example.com'. A red rectangular box highlights these two fields. Other parameters like 'Keep Alive Type', 'RPort', and 'DTMF Type' are also visible. A 'NOTE' section on the right provides additional information for administrators.

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

To set up a network conference call:

1. Place a call to the first party.
2. Press the **Conf** soft key to 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.

- Repeat steps 5 to 7 until you have added all intended parties.

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

Call Park

You can use call park 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:

- Press **Menu->Features->DSS Keys**.
- Press \uparrow or \downarrow to select the desired DSS key, then press the **Enter** soft key
- Press \leftarrow or \rightarrow , or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press \leftarrow or \rightarrow , or the **Switch** soft key to select **Call Park** from the **Key Type** field.



- Press \leftarrow or \rightarrow , or the **Switch** soft key to select the desired account from the **Account ID** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Enter the call park code in the **Value** field.
- 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:

- User on phone A places a call to phone B.
- User on phone A wants to take the call in a conference room for privacy, and so presses the call park key on phone A.
- 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 will receive a call back.

Call Pickup

You can use call pickup to answer someone else's incoming call on your phone. The SIP-T23(P/G) 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 target phone number. Group call pickup is used for picking up a call that is ringing at any phone number in a certain 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

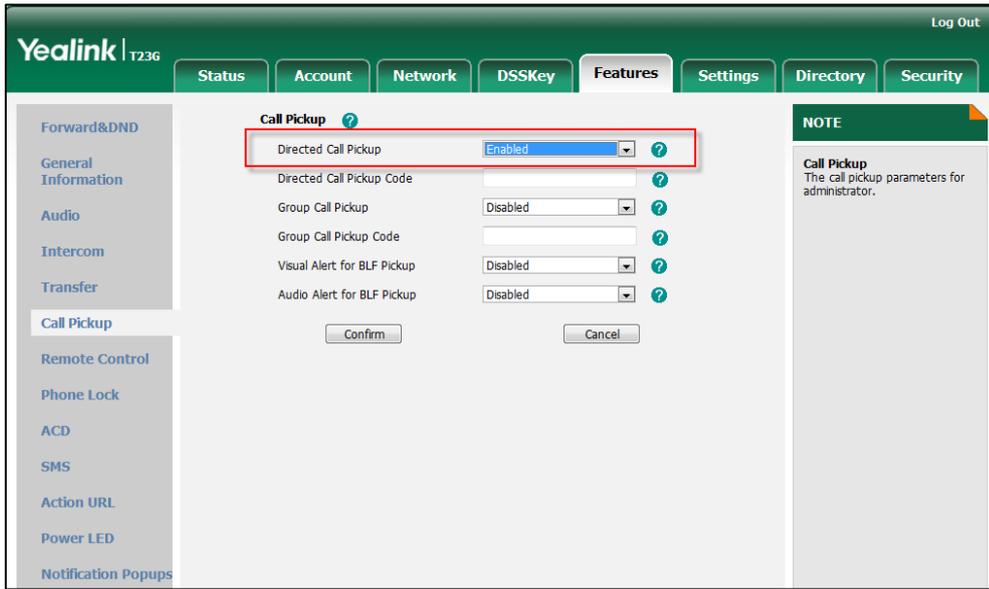
If there are many incoming calls at the same time, pressing the **GPickup** soft key on the phone will pick up the call that rings first.

Directed Call Pickup

To enable the directed call pickup via web user interface:

1. Click on **Features->Call Pickup**.

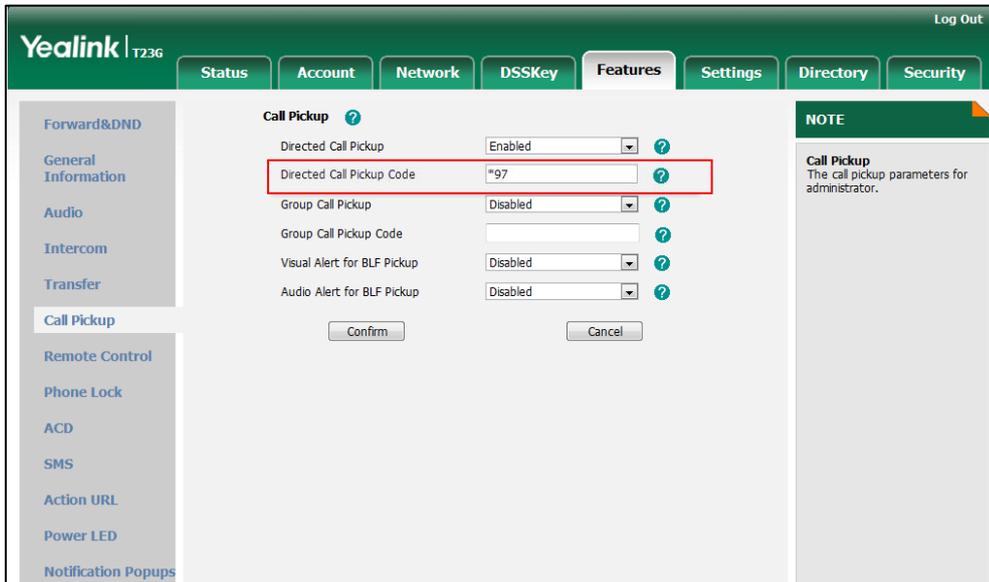
2. Select **Enabled** from the pull-down list of **Directed Call Pickup**.



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

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

1. Click on **Features->Call Pickup**.
2. Enter the directed call pickup code in the **Directed Call Pickup Code** field.



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

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

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

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

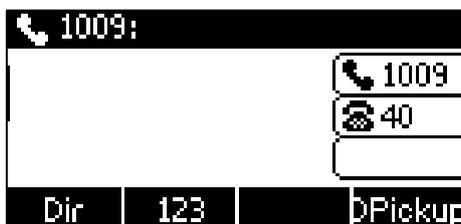
The screenshot shows the Yealink T236 web interface with the 'Account' tab selected. The 'Account' configuration page is displayed for 'Account 1'. The 'Directed Call Pickup Code' field is highlighted with a red box and contains the value '*97'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'DTMF Type' (RFC2833), 'DTMF Info Type' (DTMF-Relay), 'DTMF Payload Type(96~127)' (101), 'Conference Type' (Network Conference), 'Conference URI' (conference@example.com), 'ACD Subscribe Period(120~3600s)' (3600), 'Group Call Pickup Code', 'Distinctive Ring Tones' (Enabled), 'Unregister When Reboot' (Disabled), and 'Out Dialog BLF' (Disabled). A 'NOTE' section on the right indicates that advanced parameters are for administrators and provides a link to the Administrator Guide.

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

To pick up a call directly:

- Pick up the handset and press the **more** soft key..

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



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

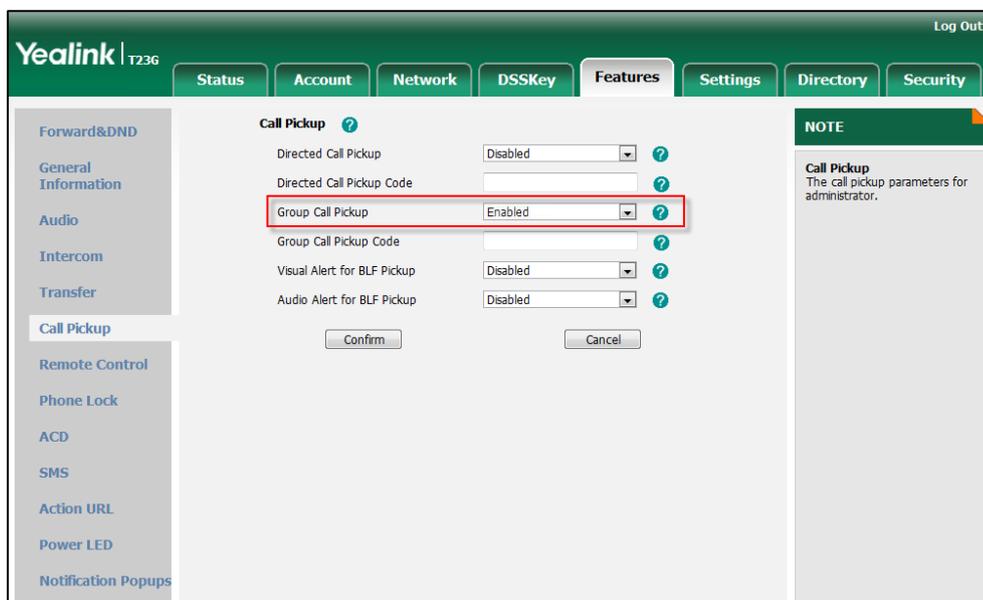
The call is answered on your phone.

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

Group Call Pickup

To enable group call pickup via web user interface:

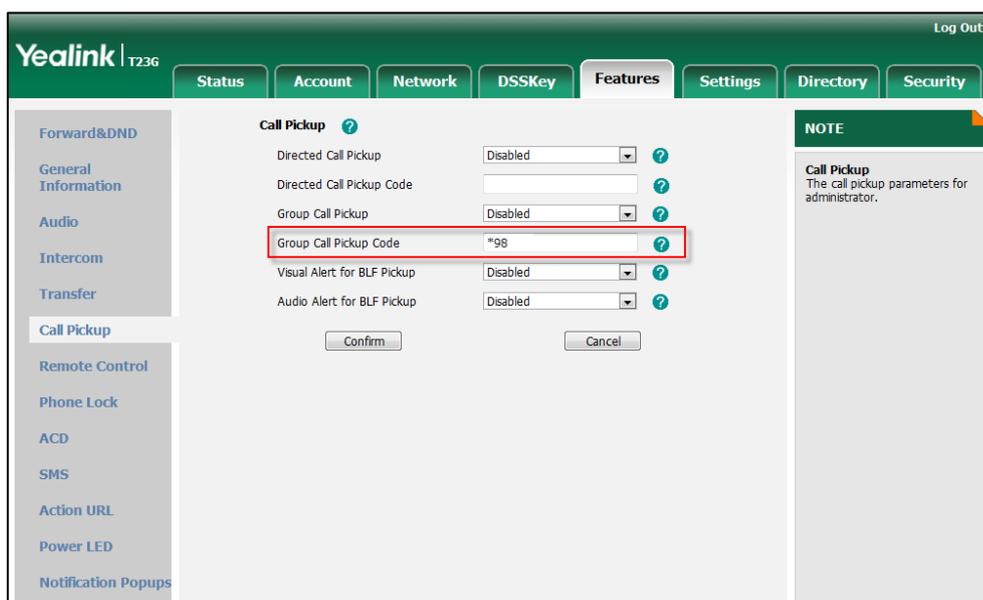
1. Click on **Features**->**Call Pickup**.
2. Select **Enabled** from the pull-down list of **Group Call Pickup**.



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

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

1. Click on **Features**->**Call Pickup**.
2. Enter the group call pickup code in the **Group Call Pickup Code** field.



3. 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.

The screenshot shows the Yealink T23G web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Group Call Pickup Code' field is highlighted with a red box and contains the value '*98'. Other fields include 'Keep Alive Type', 'Keep Alive Interval(Seconds)', 'RPort', 'Subscribe Period(Seconds)', 'DTMF Type', 'DTMF Info Type', 'DTMF Payload Type(96~127)', 'Conference Type', 'Conference URI', 'ACD Subscribe Period(120~3600s)', 'Directed Call Pickup Code', 'Distinctive Ring Tones', 'Unregister When Reboot', and 'Out Dialog BLF'. A 'NOTE' section on the right indicates that advanced parameters are for administrators and provides a link to the Administrator Guide.

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 a group pickup key via phone user interface or web user interface. Once configured, 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

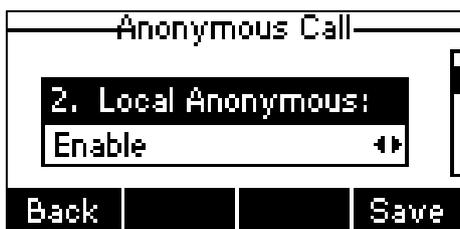
You can use anonymous call to block your identify and phone number from appearing to the called party when you call someone. For example, you want to call to consult some of the services, but don't want to be harassed. Anonymous call is configurable on a per-line basis. You can also configure the phone 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 enable 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  to select **Enable** from the **Local Anonymous** field.



4. (Optional.) Press  or  to select the desired value from the **Send Anony Code** field.
5. (Optional.) Enter the anonymous call on code and off code respectively in the **Call On Code** and **Call Off Code** field.

Anonymous call on code/ off code is used to activate/deactivate the server-side anonymous call feature when the user enables/ disables anonymous call feature on the phone

6. 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:

- 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.

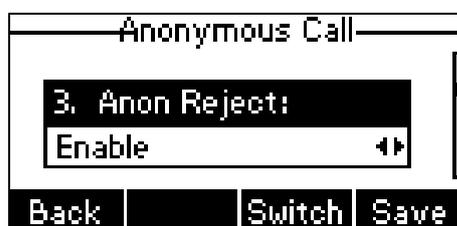


Anonymous Call Rejection

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 being displayed. Anonymous call rejection is configurable on a per-line basis.

To enable anonymous call rejection via phone user interface:

- Press **Menu->Features->Anonymous Call**.
- Press \uparrow or \downarrow , or the **Switch** soft key to select the desired line from the **Line ID** field.
- Press \uparrow or \downarrow to scroll to the **Anon Reject** field.



- Press \leftarrow or \rightarrow , or the **Switch** soft key to select **Enable** from the **Send Rejection Code** field.
- (Optional.) Enter the anonymous call rejection on code and off code respectively in the **Reject On Code** and **Reject Off Code** field.

Anonymous call rejection on code/ off code is used to activate/deactivate the server-side anonymous call rejection feature when the user enables/ disables anonymous call rejection feature on the phone

- 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-T23(P/G) IP phone. Topics include:

- [Busy Lamp Field \(BLF\)](#)
- [BLF List](#)
- [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)

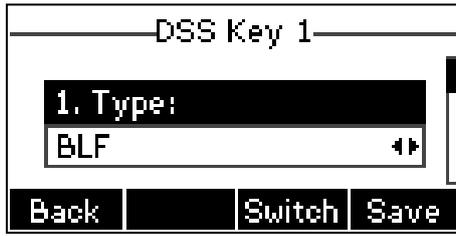
You can use BLF to monitor a specific line for status changes on the phone. For example, you can configure a BLF key (line key) on the phone to monitor the status of a friend's line (busy or idle). The BLF key LED flashes red when the friend's line receives an incoming call. For more information on BLF 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 or/and an audio alert (if enabled), and also pick up calls that are received on the monitored line. 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.

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



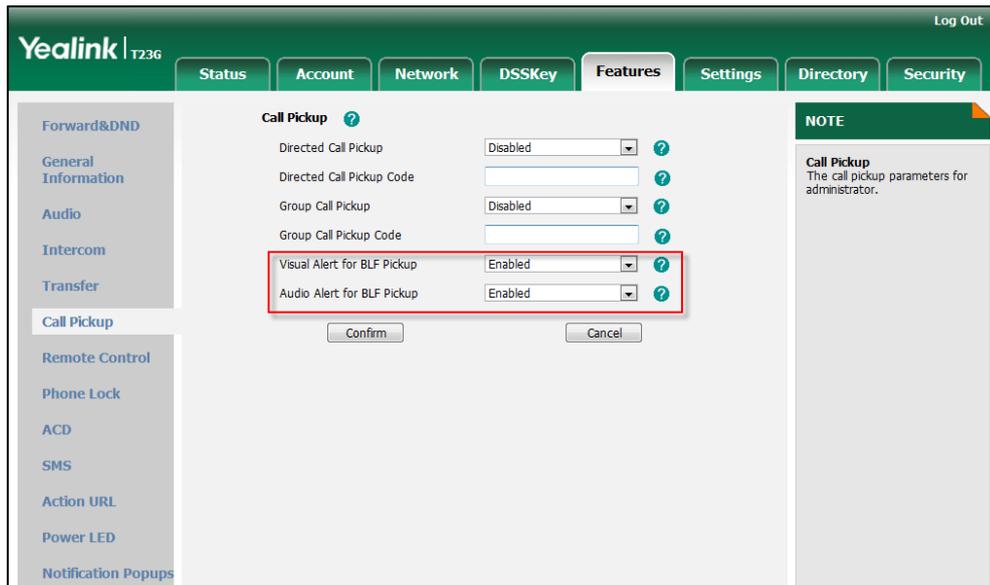
- Press  or  , or the **Switch** soft key to select the desired account from the **Account ID** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
- Enter the phone number or extension you want to monitor in the **Value** field.
- (Optional.) Enter the pickup code in the **Extension** field.
- 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 feature 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 feature 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 features via web user interface:

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



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

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

When the monitored line receives an incoming call, the following occurs on your 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 1002 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 **Pickup**, **Send**, **NewCall** 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 the [DSS Keys](#) section on page 49.

BLF List

You can use the BLF List feature to monitor a list of users defined by your system administrator. For example, your system administrator enables BLF List, and creates a BLF List URI (e.g., BLFList@example.com) including a list of user1, user2 on the server. The BLF List keys on the IP phone can present the status of user1 and user2. The key LEDs illuminate either flashing or solid depending on the status of those users. For more BLF List key LED indications, refer to [LED Instructions](#) on page 4.

You can use the BLF List keys in the following ways:

- When the monitored user is idle, press the BLF list key to dial out the phone number.
- When there is already an active call on the IP phone, you can transfer the active call to the monitored user by pressing the BLF List key. The phone handles 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 49.

- When the monitored user receives an incoming call, press the BLF list key to pick up the call directly. Before picking up an incoming call, ensure that the directed pickup code has been configured in advance.
- When there is a conversation on the monitored user, press the BLF list key to barge in and set up a conference call. Before barging in an active call, ensure that the barge-in code has been configured in advance.
- When there is an active call parked to the monitored phone, press the corresponding BLF List key to retrieve the call. Before retrieving the parked call, ensure that the call park retrieve code has been configured in advance.

To configure BLF List settings via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of Account.
3. Enter the BLF List URI in the BLF List URI field.
4. (Optional.) Enter the directed pickup code in the **BLF List code** field.
5. (Optional.) Enter the barge-in code in the **BLF List Bargeln Code** field.
6. (Optional.) Enter the call park retrieve code in the **BLF List Retrieve call parked Code** field.
7. Click **Confirm** to accept the change.

Note

For more information about the BLF List pickup/BLF List barge in feature and BLF List URI, contact your system administrator.

According to the response message from the server, the IP phone will automatically configure the BLF List keys beginning from the first unused DSS key. Once any DSS key is seized, the IP phone will skip to configure the next DSS key.

You can receive a visual or/and an audio alert (if enabled) on your phone when the monitored user receives an incoming call. For more information, refer to [Busy Lamp Field \(BLF\)](#) on page 101.

Note

The pickup code is used in the following order of preference: (Account->Advanced)>Directed Call Pickup Code (Account->Advanced)>Directed Call Pickup Code (Features->Call Pickup). If all of them are not configured, pressing the BLF List key will directly call the monitored user when he/she receives an incoming call. For more information about pickup Code, refer to [Call Pickup](#) on page 93.

Call Recording

You can record calls by pressing a record key on the phone. The SIP-T23(P/G) IP phone supports record and URL record.

Two ways of call recording:

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

Note

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 \leftarrow or \rightarrow , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press \leftarrow or \rightarrow , or the **Switch** soft key to select **Record** from the **Key Type** field.



5. (Optional.) Enter the 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 \leftarrow or \rightarrow , or the **Switch** soft key to select **URL Record** from the **Type** field.



4. (Optional.) Enter the 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 function, 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 a record or a 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 depending on the server. 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 that actual personal offices would be often vacant, consuming valuable space and resources.

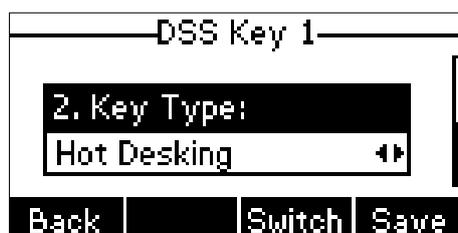
You can use hot desking on the SIP-T23(P/G) IP phone to log out of existing accounts and then log into a new account, that is, many users can share the phone resource at different times. To use this feature, you need to configure a hot desking key in advance.

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.

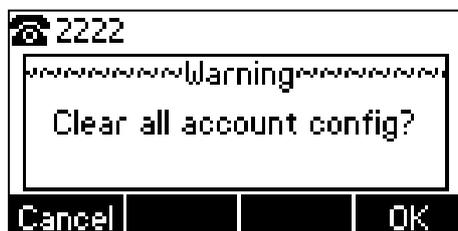


5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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 "Clear all account config?".



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

The login wizard will be displayed as below:

3. Enter the desired values in the **User Name** and **Password** fields respectively. Contact your system administrator for more information.
4. Press the **Save** soft key to login or **Cancel** 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 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 account 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 **EndCall** soft key to end the intercom call.

Incoming Intercom Calls

The SIP-T23(P/G) 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. Then 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 IP phone's microphone 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 (Press  or  to switch between different intercom features).



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 Intercom Allow is enabled, the phone will automatically answer an incoming intercom call. If Intercom Allow is disabled, the phone will reject incoming intercom calls and send a busy message to the caller. Intercom Allow is enabled by default.

Note

Your administrator can set a 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 phone's microphone for intercom calls automatically. If Intercom Mute is enabled, the microphone will be muted for intercom calls. If Intercom Mute is disabled, the microphone will work 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 will play a warning tone before answering the intercom call. If Intercom Tone is disabled, the phone will automatically answer 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 will automatically answer the intercom call and place the active call on hold. If Intercom Barge is disabled, the phone will handle an incoming intercom call like a waiting call. Intercom Barge is disabled by default.

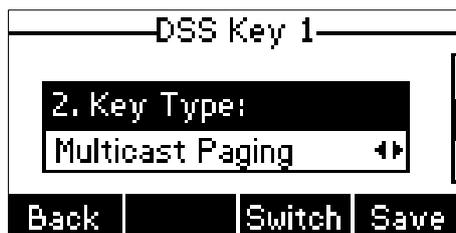
Multicast Paging

You can use multicast paging to quickly and easily forward out time sensitive announcements to people within the multicast group. You can configure a multicast paging key or a paging list 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 (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**.

To configure a paging list 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 **Paging List** from the **Key Type** field.



5. (Optional.) Enter the 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.

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

To configure paging list via phone user interface:

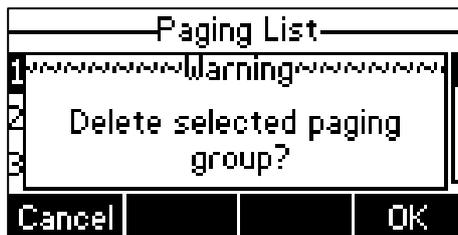
1. Press the paging list key when the phone is idle.
2. Press  or  to select a desired paging group.
The default tag is (Empty) if it is not configured before.
3. Press the **Option** soft key.
4. Press the **Edit** soft key.



5. Enter the multicast IP address and port number (e.g., 224.5.6.20:10008) in the **Address** field.
The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
6. Enter the group name in the **Label** field.
7. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.
8. Repeat the step 1-7, you can add more paging groups.

To delete paging group via phone user interface:

1. Press the Paging list key when the phone is idle.
2. Press  or  to select a desired group.
3. Press the **Option** soft key.
4. Press the **Delete** soft key.
The LCD screen prompts "Delete selected paging group?".



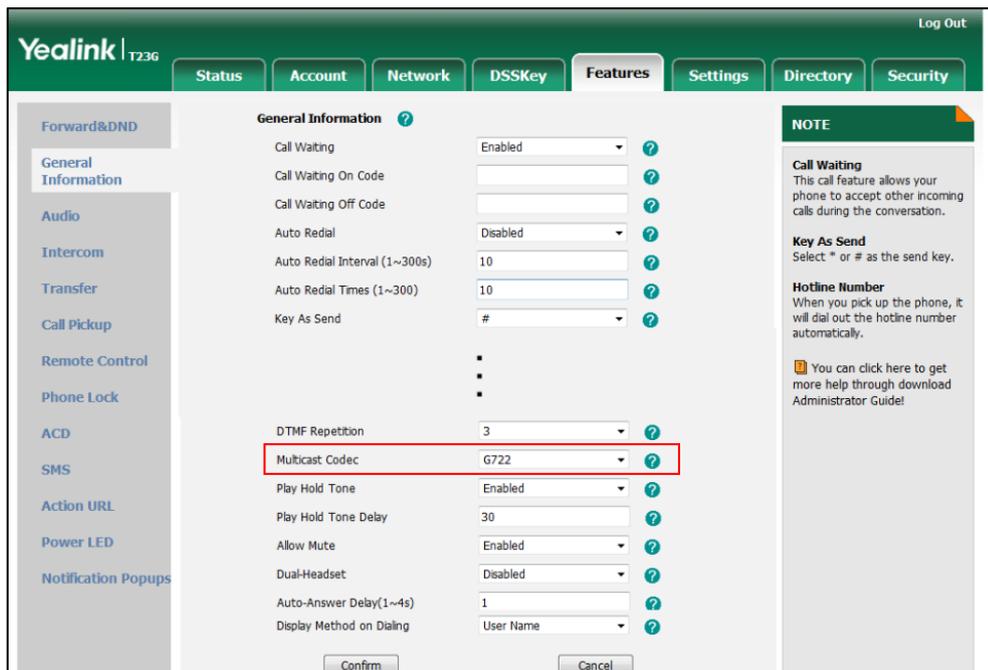
5. Press the **OK** soft key to accept the change or the **Cancel** soft key to cancel.
If you want to delete all paging group, you can press the **Del All** soft key.

You can also configure the phone to use a default codec for sending multicast RTP stream via web 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 prompts "HD" to indicate that it is providing high definition voice.

Default codec for multicast paging is configurable via web user interface only.

To send RTP stream via a multicast paging key:

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 **EndCall** soft key to cancel the multicast RTP session.

Note

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

To send RTP stream via a paging list key:

1. Press the paging list key when the phone is idle.
2. Press  or  to select the desired group.



3. Press  or the **Paging** soft key to send RTP.
4. Press the **Hold** soft key to place the current multicast RTP session on hold.
5. Press the **EndCall** soft key to cancel the multicast RTP session.

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

The 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 will take precedence over all incoming multicast paging calls. Valid values in the Paging Barge field:

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

Paging Priority Active

The paging priority active parameter decides how the phone handles the incoming multicast paging calls when there is already a multicast paging call on the phone. If enabled, the phone will ignore 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 to for incoming RTP multicast in the **Listening Address** field.

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

Multicast Listening

Paging Barge: 10

Paging Priority Active: Enabled

	IP Address	Listening Address	Label	Priority
1	IP Address	224.5.6.20:10008		1
2	IP Address			2
3	IP Address			3
4	IP Address			4
5	IP Address			5
6	IP Address			6
7	IP Address			7
8	IP Address			8
9	IP Address			9
10	IP Address			10

NOTE
Multicast IP
 The multicast IP parameters for administrator.
 You can click here to get more help through download Administrator Guide!

5. Enter the label in the **Label** field.
Label will appear on the LCD screen when receiving the RTP multicast.
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 receiver's phones play a warning tone when establishing a multicast paging call.

Multicast listening addresses is configurable via web user interface only.

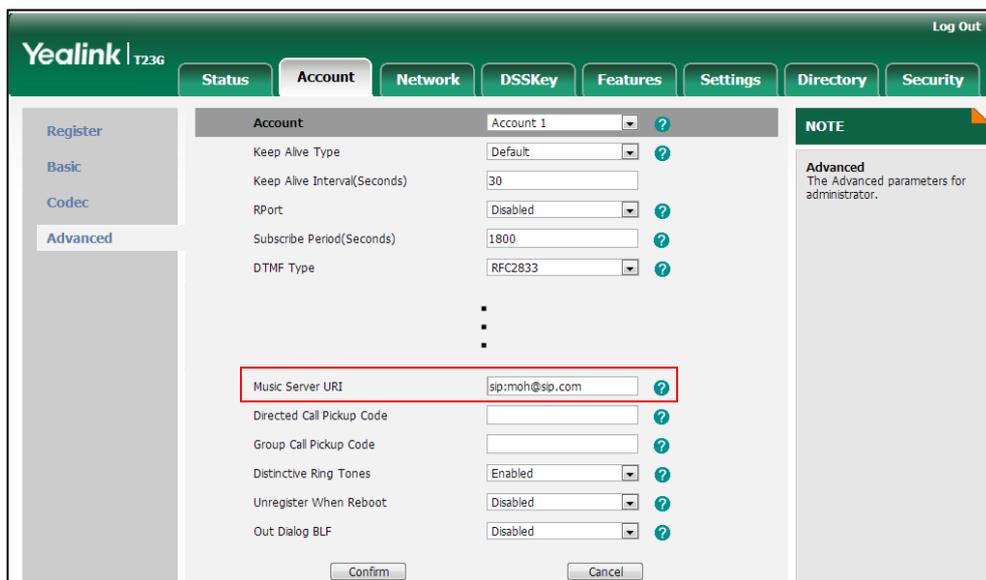
Music on Hold

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 answers 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

For this feature to function, all involved parties cannot use encrypted RTP (SRTP). Music on hold server is configurable via the web user interface only.

Automatic Call Distribution (ACD)

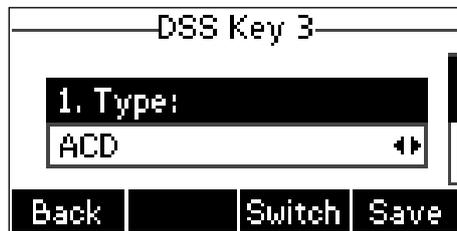
ACD is often used in offices for customer service, such as call center. The ACD system handles large volumes of incoming calls from callers who have no need to talk to a specific person but who require assistance from any of multiple persons at the earliest opportunity. ACD on the SIP-T23(P/G) IP phone 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 identify used to log into the queue.

Password: the password used to log into the queue.



2. Press the **Login** soft key to login.

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 **Avail/Unavail** 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 phone 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

Short Message Service (SMS)

You can send and receive text messages using the SIP-T23(P/G) IP phone. New text messages can be indicated both acoustically and visually. When receiving a new text message, the phone will play a warning tone, and the LCD screen will prompt receiving new text messages with the number of waiting messages (example: 1 New Text Message(s)) and a flashing icon.

Note

When the phone receives a text message, the text message prompt window will pop up by default, if you want to disable the feature, contact your system administrator for more information.



You can store text messages in your phone's Inbox, Sentbox, Outbox or Draftbox. Each of the boxes can store up to 100 text messages. If the number of the text messages in one box is more than 100, the phone will directly delete the oldest text message in the box.

Note

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

To read a text message:

1. Press **Menu->Message->Text Message->Inbox**.



2. Select the desired message and press the **View** soft key.

Note

If the phone prompts receiving new text messages, you can also press the **View** soft key to read the new messages directly.

To send a text message:

1. Press **Menu->Message->Text Message->New Message**.
2. Compose the new text message. You can press the **abc** soft key to change the input mode.



3. Press the **Send** soft key after completing the content.
4. (Optional.) Press **◀** or **▶**, or the **Switch** soft key to select the desired account from the **From** field.
5. Enter the number you want to send the message to in the **To** field.
6. Press the **Send** soft key to send the message or the **Back** soft key to cancel.

Sending a text message is configurable via web user interface at the path **Features>SMS**.

To reply a text message:

1. Press **Menu->Message->Text Message->Inbox**.
2. Select the desired message and press the **Reply** soft key.



3. Compose the new text message. You can press the **abc** soft key to change the input mode.
4. Press the **Send** soft key after completing the content.

To delete a text message:

1. Press **Menu->Message->Text Message->Inbox (Sentbox, Outbox or Draftbox)**.
2. Select the desired message and then press the **Delete** soft key.



3. Select **Delete** to delete the desired message.
The LCD screen prompts "Delete Message?".

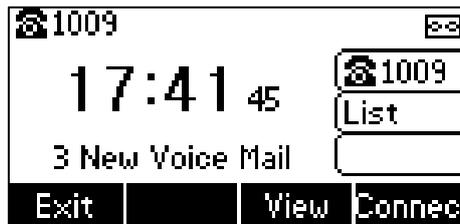


4. Press the **OK** soft key to delete this message or the **Cancel** soft key to cancel.
You can also delete all text messages by pressing the **Delete** soft key and then select **Delete All**. For more information, refer to the above steps.

Note You can also delete a specific message after retrieving by pressing the **Delete** soft key.

Voice Mail

You can leave voice mails for someone else using the SIP-T23(P/G) IP phone. You can also listen to voice mails that are stored in a voice mailbox. When receiving a new voice mail, the phone will play a warning tone and the MESSAGE key LED will illuminate. The LCD screen will prompt that the phone receives a new message and display a flashing icon.



Note

Voice mail is not available on all servers.

You can configure the phone not to display the pop-up prompt, 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->Message->Voice Mail->Set Voice Mail**.



2. Press \uparrow or \downarrow to select the account which you want to set, press the **123** soft key to select the proper input mode and then enter the voice mail access code.
3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Note

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

To listen to voice mails:

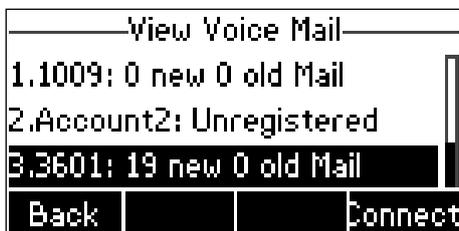
1. When the LCD screen prompts that the phone receives a new voice mail, press  or the **Connect** soft key to dial out the voice mail access code.
2. Follow the voice prompt to listen to your voice mails.

Note

Before listening to the voice mails, make sure the voice mail access code has been configured.
When all new voice mails are retrieved, the MESSAGE key LED will go out.

To view voice mails 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.

Note

When the phone receives a voice mail, the voice mail prompt window will pop up by default, if you want to disable the feature, contact your system administrator for more information.

Message Waiting Indicator (MWI)

The SIP-T23(P/G) IP phone supports MWI 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, a prompt message (including a voice mail icon) on the LCD screen, and the MESSAGE key LED illuminates solid green. This will be cleared when you retrieve all voice mails or delete them.

The MWI service is unsolicited for some servers, so the SIP-T23(P/G) IP phone only handles the MWI messages sent from the server. But for other servers, the MWI service is solicited, so the SIP-T23(P/G) 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:

Option	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 configure subscribe for MWI and the voice mail number in advance.

Note

The phone will send SUBSCRIBE messages for the MWI service to the account or the voice number MWI service depending 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** field.
5. Enter the period time in the **MWI Subscription Period (Seconds)** field.

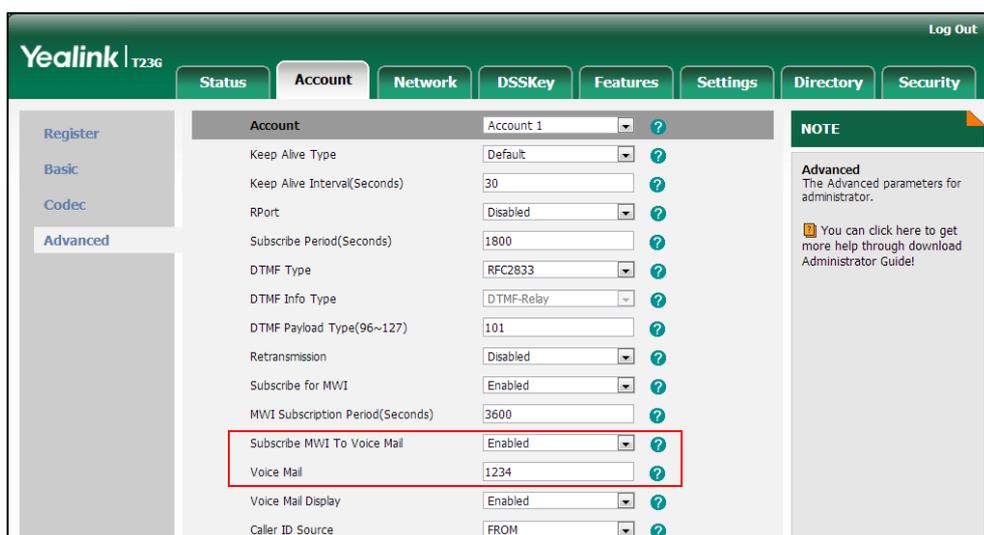
The screenshot shows the Yealink T236 web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Subscribe for MWI' field is highlighted with a red box and set to 'Enabled'. The 'MWI Subscription Period(Seconds)' field is set to 3600. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'DTMF Type' (RFC2833), 'DTMF Info Type' (DTMF-Relay), 'DTMF Payload Type(96~127)' (101), 'Retransmission' (Disabled), and 'Subscribe MWI To Voice Mail' (Disabled). A 'NOTE' box on the right states: 'Advanced: The Advanced parameters for administrator. You can click here to get more help through download Administrator Guide!'.

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

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 to solve the problems you might encounter when using your SIP-T23(P/G) 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 plugging it directly into a wall outlet instead.
- If your SIP-T23P/T23G IP phone is powered from PoE, ensure that 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 of the phone and the Ethernet cable is not loose.
- Ensure that the switch or hub in your network is operational.
- Contact your system administrator for more information.

Why does the phone display “No Service”?

The LCD screen prompts “No Service” message when no SIP account registers successfully.

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 fails to connect to the SNTP server, you need to configure the time and date manually.

How do I find the basic information of the phone?

Press the **OK** key when the phone is idle to check the basic information of the phone, such as the IP address and firmware version. For more the basic information, refer to [Phone Status](#) on page 13.

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

Three ways you can obtain the MAC address of a phone:

- You can ask your supplier for the shipping information sheet which includes MAC addresses according to the corresponding PO (Purchase Order).
- You can find the MAC address on the label of the 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 that the ringer volume on the phone. To adjust the ringer volume setting, press the **Volume** key when the phone is idle. For more information, refer to [Volume](#) on page 26.

Why can't I receive calls?

- Check that the SIP registration with your system administrator.
- Check that the DND (Do Not Disturb) mode is deactivated on the 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 81.
- Check whether the caller number is stored in the blacklist directory. Refer to [Blacklist](#) on page 40.

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 Mode](#)

[Activation/Deactivation](#) on page 47.

- 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?

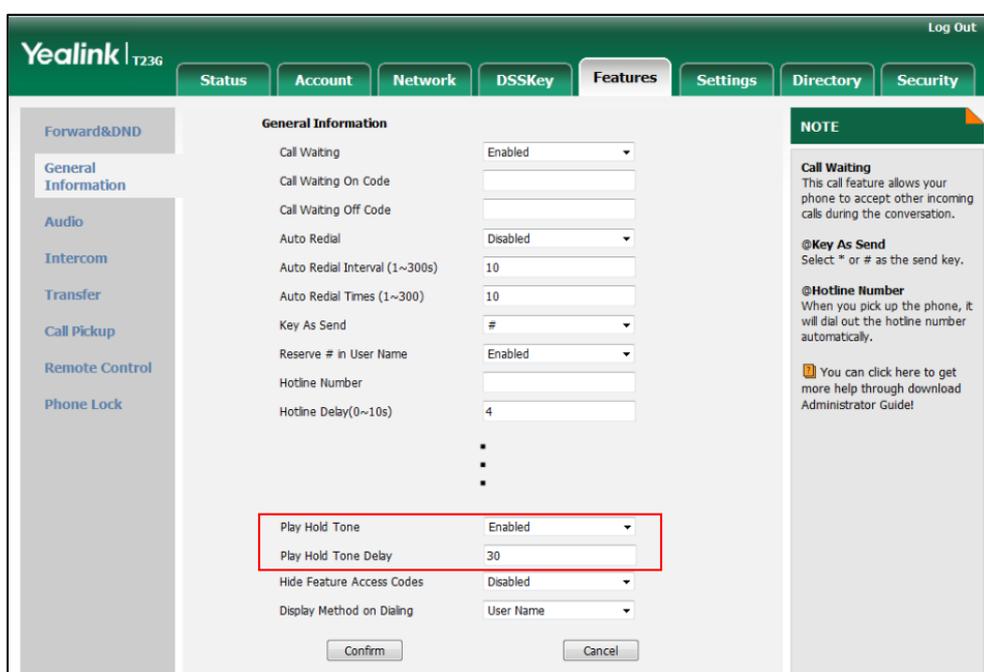
Both user name and register name are defined by the server. A user name is used to identify the account while a register name matched with a password is used for authentication if the server requires. Display name is the caller ID that will be displayed on the callee’s LCD screen. Server configurations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

Why does the phone play a tone when there is a call on hold? How to disable it?

When there is a call 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 play hold tone and play hold tone delay via web user interface:

1. Click on **Features->General Information**.
2. Select the **Enabled** or **Disabled** from the pull-down list of **Play Hold Tone**.
3. Enter the desired time (in seconds) in the **Play Hold Tone Delay** field.



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

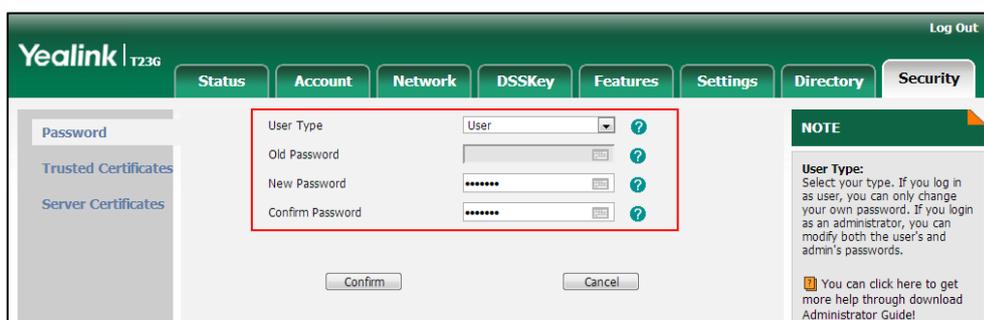
Why can't I send a SMS to any other phone?

SMS depends on support from a SIP server. Contact your system administrator for more information.

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.

Note

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

User password is configurable via the web user interface only.

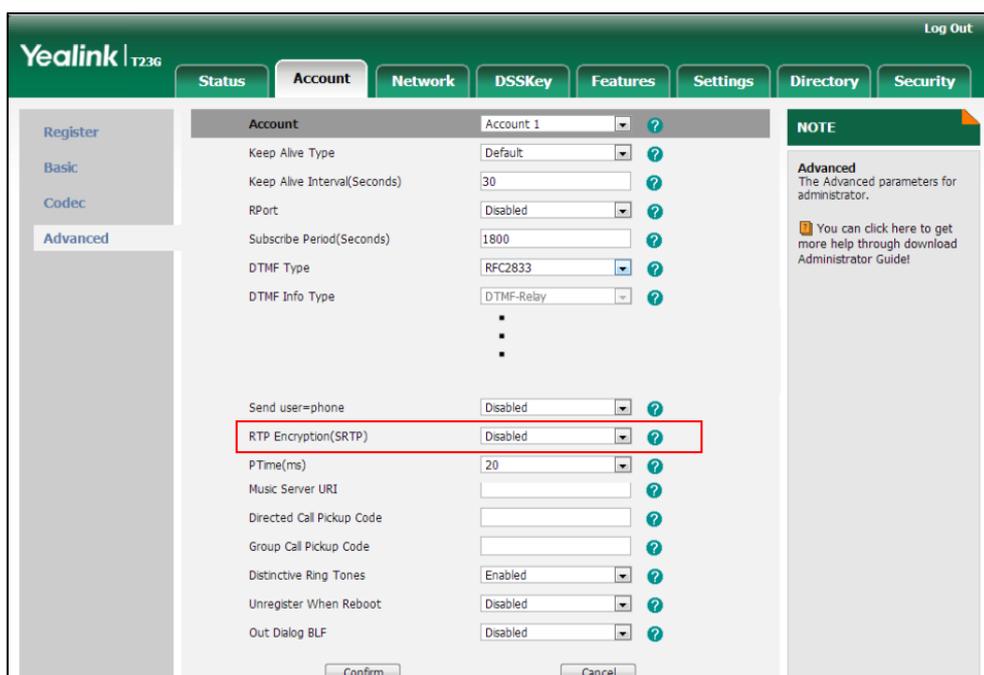
How to make a call using SRTP?

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

- Select the desired value from the pull-down list of **RTP Encryption (SRTP)**.



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

Note

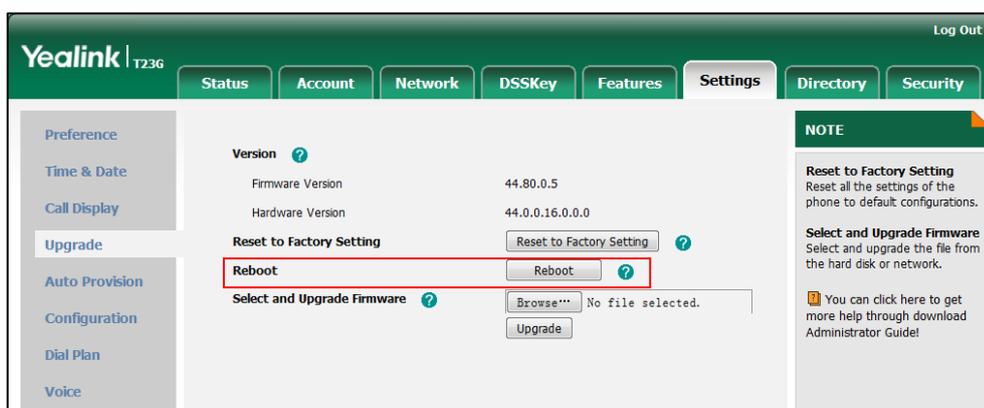
SRTP is not available on all servers. Contact your system administrator for more information.

SRTP is configurable via web user interface only.

How to reboot the phone?

To reboot the phone via web user interface:

- Click on **Settings->Upgrade**.
- Click **Reboot** to reboot the phone.



Note

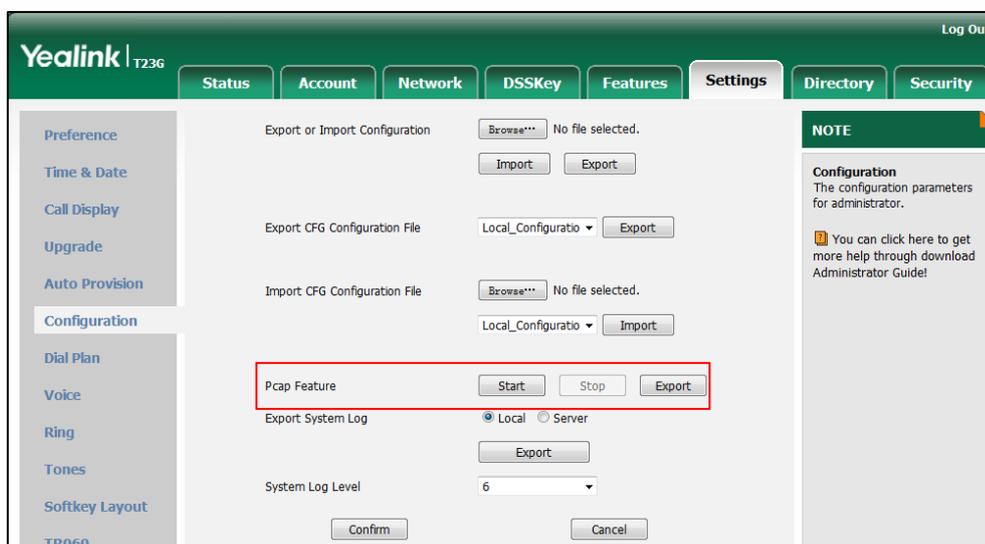
Any reboot of the phone may take a few minutes.

How to export PCAP trace?

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

To export a PCAP trace via web user interface:

1. Click on **Settings->Configuration**.
2. Click **Start** to begin recording signal traffic.
3. Recreate the error to be documented in the trace.
4. Click **Stop** to end recording.
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 a system log to help analyze your problem.

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

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

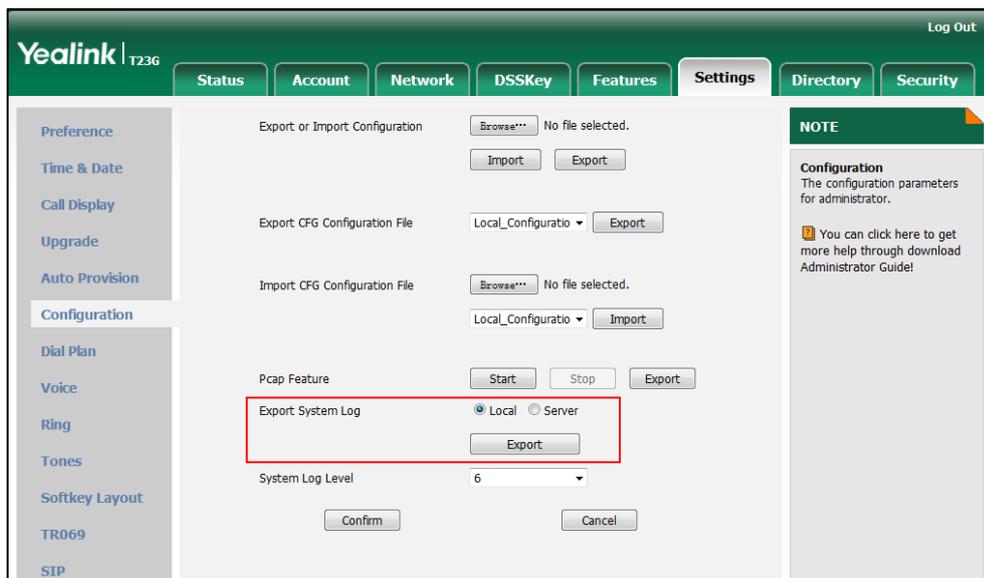
The web user interface prompts "Do you want to restart your machine?". The configuration will take effect after reboot.

4. Click OK to reboot the phone.

After the IP phone finishes a reboot, the system log level is set as 6, the debug level.

5. Mark the **Local** radio box in the **Export System Log** field.

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



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

Note

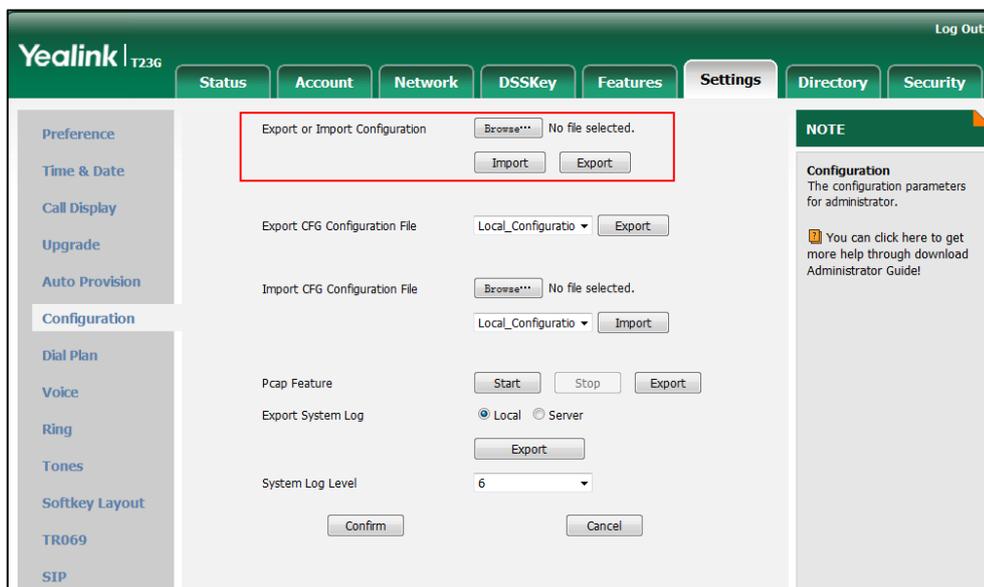
It is recommended you reset the system log level to 3 after you have the syslog file provided.

How to export/import phone configurations?

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

To export the phone configurations via web user interface:

1. Click on **Settings->Configuration**.
2. In the **Export or Import Configuration** field, 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. In the **Export or Import Configuration** field, click **Browse** to locate a configuration file from your local system.
3. Click **Import** to import the configuration file.

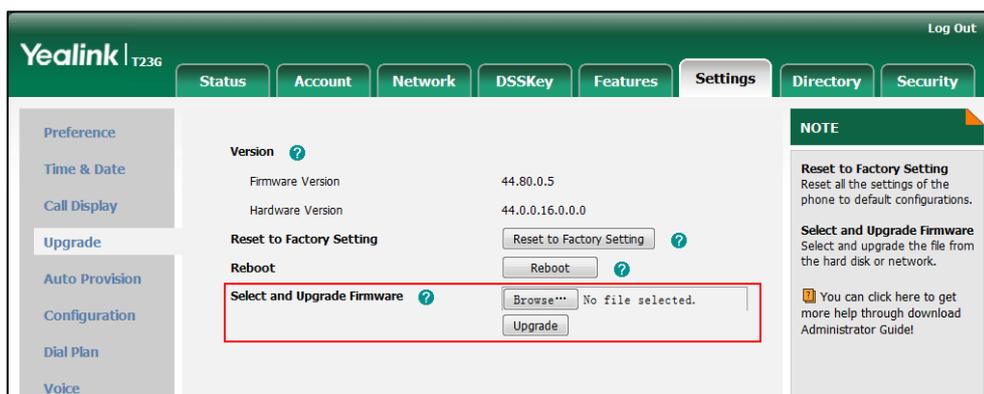
Note The file format of configuration file must be *.bin.

How to upgrade firmware?

To upgrade firmware via web user interface:

1. Click on **Settings->Upgrade**.

2. Click **Browse** to locate the required 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 all troubleshooting suggestions but do not solve the problem. You need to note that all custom settings will be overwritten after resetting. We recommend asking your system administrator for advice before resetting the phone.

To reset the phone via phone user interface:

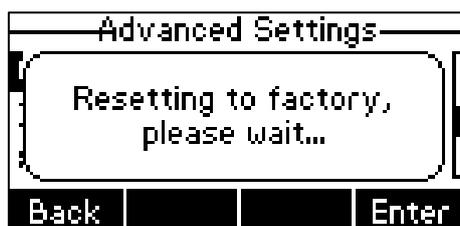
1. Press **Menu->Settings->Advanced Settings** (default password: admin).
2. Press **▲** or **▼** to scroll to **Reset to Factory**, and then press the **Enter** soft key.

The LCD screen prompts "Reset To Factory Setting?".

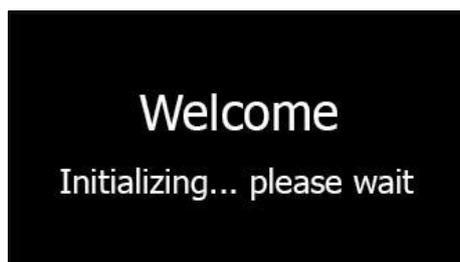


3. Press the **OK** soft key.

The LCD screen prompts “Resetting to factory, please wait...”.



The LCD screen prompts “Welcome Initializing... please wait”.



The phone will be reset to factory settings successfully after startup.

Note

Reset of your phone may take a few minutes. Do not power off until the phone has started up successfully.

Regulatory Notices

Service Agreements

Contact your Yealink Authorized Reseller for information about service agreements applicable to your product.

Limitations of Liability

TO THE FULL EXTENT ALLOWED BY LAW, YEALINK EXCLUDES FOR ITSELF AND ITS SUPPLIERS ANY LIABILITY, WHETHER BASED IN CONTRACT OR TORT (INCLUDING NEGLIGENCE), FOR INCIDENTAL, CONSEQUENTIAL, INDIRECT, SPECIAL, OR PUNITIVE DAMAGES OF ANY KIND, OR FOR LOSS OF REVENUE OR PROFITS, LOSS OF BUSINESS, LOSS OF INFORMATION OR DATA, OR OTHER FINANCIAL LOSS ARISING OUT OF OR IN CONNECTION WITH THE SALE, INSTALLATION, MAINTENANCE, USE, PERFORMANCE, FAILURE, OR INTERRUPTION OF ITS PRODUCTS, EVEN IF YEALINK OR ITS AUTHORIZED RESELLER HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES, AND LIMITS ITS LIABILITY TO REPAIR, REPLACEMENT, OR REFUND OF THE PURCHASE PRICE PAID, AT YEALINK'S OPTION. THIS DISCLAIMER OF LIABILITY FOR DAMAGES WILL NOT BE AFFECTED IF ANY REMEDY PROVIDED HEREIN SHALL FAIL OF ITS ESSENTIAL PURPOSE.

Safety Instructions

Save these instructions. Read these safety instructions before use!

The following basic safety precautions should always be followed to reduce the 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 do not attempt 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 circumstances 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 also 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 dust.
- Place the device on a stable and level platform.

- Please do not place heavy objects on the device in case of damage and deformation caused by the heavy load.
- Keep at least 10 cm between the device and the closest object for heat dissipation.
- Do not place the device on or near any inflammable or fire-vulnerable object, such as rubber-made materials.
- Keep the device away from any heat source or bare fire, such as a candle or an electric heater.
- Keep the device away from any household appliance with a 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 only use the accessories provided or authorized by the manufacturer.
- The power supply of the device must meet the requirements of the input voltage of the device. Please use the provided surge protection power socket only.
- Before plugging or unplugging any cable, make sure that your hands are completely dry.
- Do not spill liquid of any kind on the product or use the equipment near water, for example, near a bathtub, washbowl, kitchen sink, wet basement or near a swimming pool.
- Do not tread on, pull, or over-bend any cable in case of malfunction of the device.
- During a thunderstorm, stop using the device and disconnect it from the power supply. Unplug the power plug and the Asymmetric Digital Subscriber Line (ADSL) twisted pair (the radio frequency cable) to avoid lightning strike.
- If the device is left unused for a rather long time, disconnect it from the power supply and unplug the power plug.
- When there is smoke emitted from the device, or some abnormal noise or smell, disconnect the device from the power supply, and unplug the power plug immediately. Contact the specified maintenance center for repair.
- Do not insert any object into equipment slots that is not part of the product or auxiliary product.
- Before connecting a cable, connect the grounding cable of the device first. Do not disconnect the grounding cable until you have disconnected 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
- 09:30	French Polynesia
- 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)
- 02:30	Newfoundland and Labrador
- 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)

Time Zone	Time Zone Name
+01:00	Denmark(Kopenhagen)
+01:00	France(Paris)
+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(Kabul)
+05:00	Kazakhstan(Aqtobe)
+05:00	Kyrgyzstan(Bishkek)
+05:00	Pakistan(Islamabad)
+05:00	Russia(Chelyabinsk)
+05:30	India(Calcutta)
+05:45	Nepal(Katmandu)
+06:00	Kazakhstan(Astana, Almaty)
+06:00	Russia(Novosibirsk, Omsk)
+06:30	Myanmar(Naypyitaw)
+07:00	Russia(Krasnoyarsk)
+07:00	Thailand(Bangkok)
+08:00	China(Beijing)
+08:00	Singapore(Singapore)
+08:00	Australia(Perth)
+08:45	Eucla
+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)

Time Zone	Time Zone Name
+ 11:00	New Caledonia(Noumea)
+ 11:30	Norfolk Island
+ 12:00	New Zealand(Wellington, Auckland)
+ 12:45	New Zealand(Chatham Islands)
+ 13:00	Tonga(Nukualofa)
+ 13:30	Chatham Islands
+ 14:00	Kiribati

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