



Enterprise IP Phone User Guide SIP-T19 E2 & T19P E2

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CE Mark Warning

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

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

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

Class B Digital Device or Peripheral

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

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

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GNU GPL INFORMATION

Yealink SIP-T19(P) E2 IP phone firmware contains third-party software under the GNU General Public License (GPL). Yealink uses software under the specific terms of the GPL. Please refer to the GPL for the exact terms and conditions of the license.

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

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

About This Guide

Thank you for choosing the SIP-T19(P) E2 IP phone, exquisitely designed to provide business telephony features, such as Call Hold, Call Transfer, Multicast Paging and Conference over an IP network. The difference between the SIP-T19 E2 and SIP-T19P E2 IP phones is that only SIP-T19P E2 supports PoE.

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-T19(P) E2 IP phone.

Note

Network Directory and Network Call Log features are hidden for IP phones in neutral firmware, 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

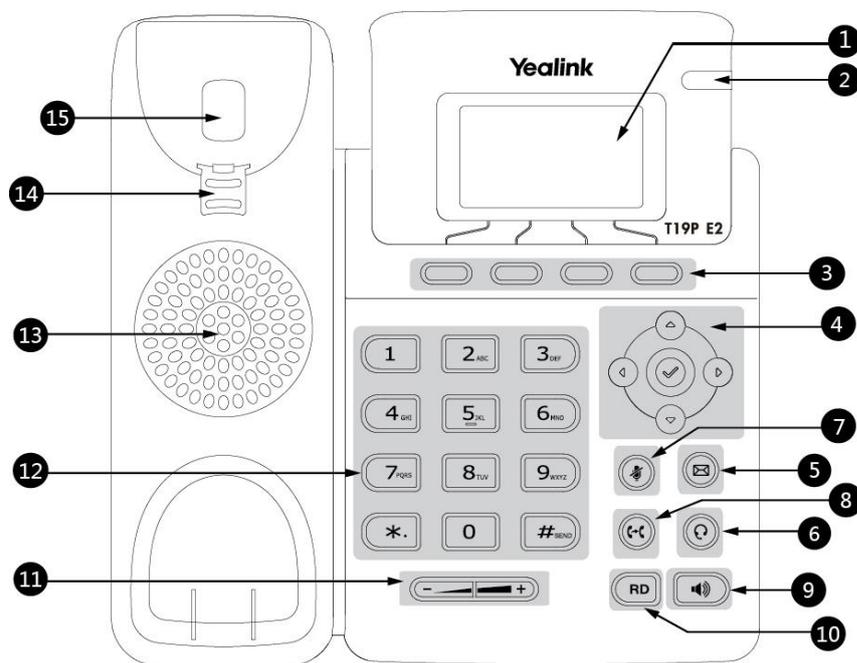
This chapter provides the overview of the SIP-T19(P) E2 IP phone. Topics include:

- [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-T19(P) E2 IP phone are the LCD screen and the keypad.



Hardware component instructions of the SIP-T19(P) E2 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	Soft Key	Label automatically to identify their context-sensitive features.
④		Scroll through the displayed information.
		Confirms actions or answers incoming calls.
⑤	Message Key	Accesses voice mails.
⑥	Headset Key	Toggles the headset mode.
⑦	Mute Key	Mutes or un-mutes an active call.
⑧	Transfer Key	Transfers a call to another party.
⑨	Speakerphone Key	Toggles the hands-free speakerphone mode.
⑩	RD Key	Redials a previously dialed number.
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. For more information on how to adjust the hookswitch tab, refer to <i>Yealink Wall Mount Quick Installation Guide for Yealink IP Phones</i> .
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
	The private line registers successfully
	Register failed
	Registering
	The shared/bridged line registers successfully
	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

Icon	Description
	Ringer volume is 0
	Phone Lock
	Call Mute
	Received Calls
	Placed Calls
	Missed Calls
	The contact icon

LED Instructions

Power Indicator LED

LED Status	Description
Solid green	The phone is initializing.
Fast flashing green (300ms)	The phone is ringing.
Slow flashing green (1s)	The phone receives a voice mail or text message.
Off	The phone is powered off. The phone is idle. The phone is busy. The call is placed on hold or is held. The call is muted.

Note The above introduces the default LED status. The statuses of the power indicator LED are configurable via web user interface. For more information, refer to [Yealink_SIP-T2_Series_T19\(P\)_E2_T4_Series_IP_Phones_Administrator_Guide](#).

User Interfaces

Two ways to customize configurations of your SIP-T19(P) E2 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 19.

Web User Interface

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

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

Options	Phone User Interface	Web User Interface
Status		
--IPv4		
--MAC		
--Firmware	√	√
--Network		
--Phone		
--Accounts		
Basic Phone Settings		
--Contrast	√	
--Language	√	
--Time & Date	√	
--Administrator Password	√	
--Key as Send	√	
--Phone Lock	√	
--Ring Tones	√	

Options	Phone User Interface	Web User Interface
--Contact Management		
--Directory	x	
--Local Directory	√	
--Blacklist	√	
--Remote Phone Book	x	
--Call History Management	√	
--Logo Customization	x	
--Programable Keys	x	
--Account Registration	√	
--Dial Plan	x	
--Emergency Number	x	
--Live Dialpad	x	
--Hotline	√	
Basic Call Features		
--Recent Call In Dialing	x	
--Auto Answer	√	
--Auto Redial	√	
--Call Completion	√	
--ReCall	x	
--Do Not Disturb (DND)	√	√
--Call Forward	√	
--Call Transfer	√	
--Call Waiting	√	
--Conference	x	
--Call Pickup	√	
--Anonymous Call	√	
--Anonymous Call Rejection	√	
Advanced Phone Features		
--Hot Desking	√	
--Intercom	√	
--Multicast Paging	x	
--Shared Call Appearance (SCA)	x	√
--Bridged Line Appearance (BLA)	x	
--Music on Hold	x	
--Messages	√	
SIP Account		
--User Options		
--Active Line	√	√
--Label	√	
--Display Name	√	
--Register Name	√	

Options	Phone User Interface	Web User Interface
--User Name	√	
--Password	√	
--Server Option		
--SIP Server 1/2	√	
--Register Port	x	
--Outbound Status	√	
--Outbound Proxy1/2	√	
--Proxy Fallback Interval	√	
--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 SIP-T19(P) E2 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-T19(P) E2 IP phone. Topics include:

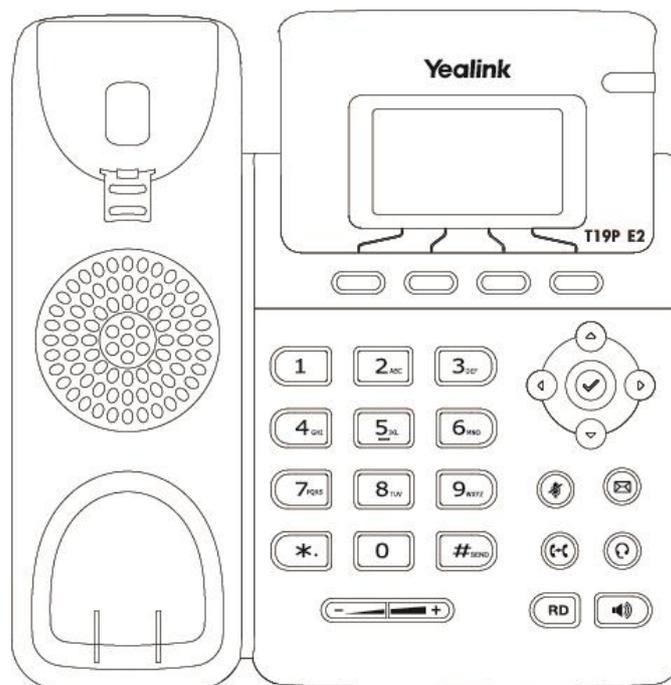
- [Packaging Contents](#)
- [Phone Installation](#)
- [Phone Initialization](#)
- [Phone Status](#)
- [Basic Network Settings](#)
- [Registration](#)
- [Idle Screen](#)

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-T19(P) E2 IP phone package:

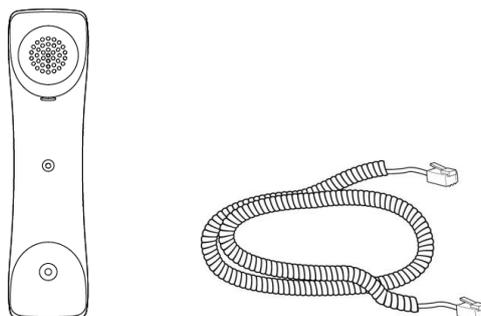
- **SIP-T19(P) E2 IP Phone**



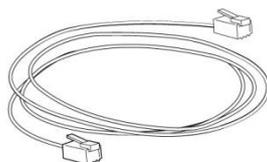
- **Phone Stand**



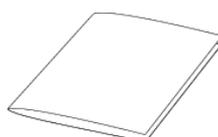
- **Handset & Handset Cord**



- **Ethernet Cable**



- **Quick Start Guide**

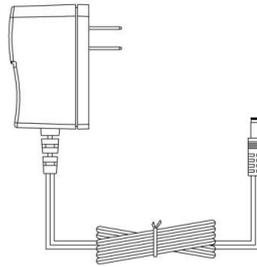


Check the list before installation. If you find anything missing, contact your system administrator.

Optional Accessories

The following items are optional accessories for your SIP-T19(P) E2 IP phone. You need to purchase them separately if required.

- **Power Adapter**



- **Headset**



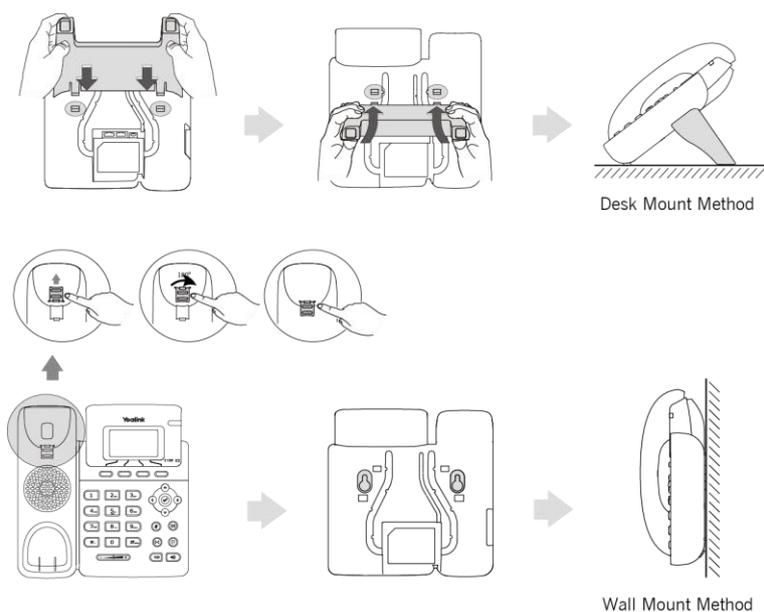
Phone Installation

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

This section introduces how to install the phone:

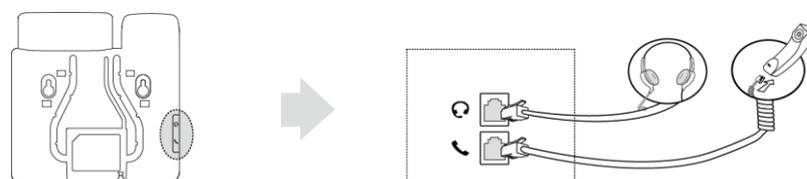
- 1) Attach the stand
- 2) Connect the handset and optional headset
- 3) Connect the network and power

1) Attach the stand



Note The hookswitch tab has a lip which allows the handset to stay on-hook when the IP phone is mounted vertically.

2) Connect the handset and optional headset



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)

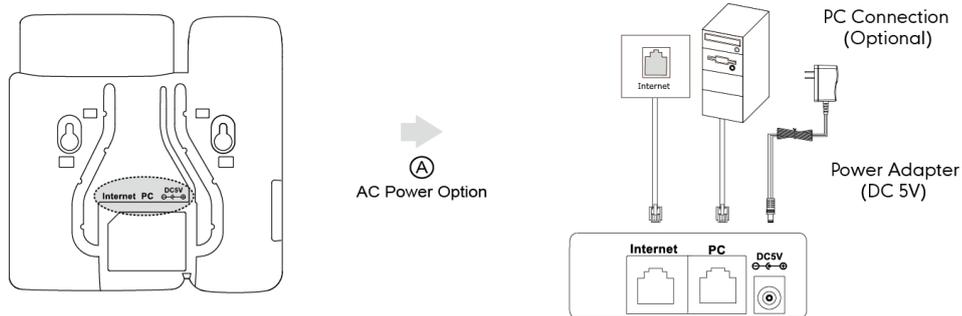
Note PoE is not applicable to the SIP-T19 E2 IP phone.

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.

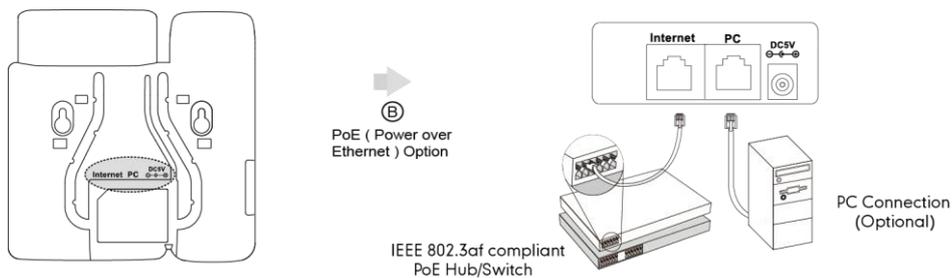


Power over Ethernet

With the included or a regular Ethernet cable, the SIP-T19P E2 IP phone can be powered from a PoE-compliant switch or hub.

To connect the PoE for the SIP-T19P E2 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 from 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 IPv4 network settings (e.g., IP address, subnet mask, default gateway address and DNS address) by default.

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

Phone Status

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

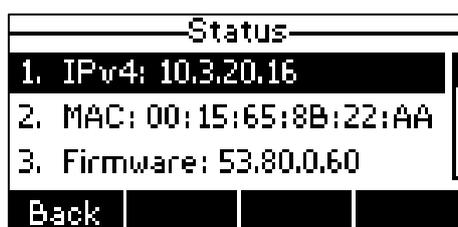
Available information of phone status includes:

- Network status (e.g., IPv4 Status, IPv6 Status, IP Mode and MAC address).
- Phone status (e.g., Product Name, Hardware, Firmware, Product ID, MAC address and Device Cert).
- Account status (e.g., register status of SIP accounts).

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

To view the phone status via phone user interface:

1. Press , 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 of your computer.
2. Enter the IP address in the browser's address bar, and then press **Enter**.

- Enter the user name (admin) and password (admin) in the login page.

- Click **Confirm** to login.

The phone status is displayed on the first page of the web user interface.

Version	
Firmware Version	53.80.0.60
Hardware Version	53.0.0.128.0.0.0
Network	
Internet Port	IPv4
IPv4	
WAN Port Type	DHCP
WAN IP Address	10.3.20.16
Subnet Mask	255.255.255.0
Gateway	10.3.20.254
Primary DNS	192.168.1.22
Secondary DNS	192.168.1.20
Network Common	
MAC Address	0015658822AA
Link Status	Connected
Device Type	Bridge
Account Status	
Account	106@10.2.1.199 : Registered

NOTE

Version
It shows the version of firmware and hardware.

Network
It shows the network settings of Internet (WAN) port.

Account
It shows the registration status of SIP accounts.

You can click here to get more guides.

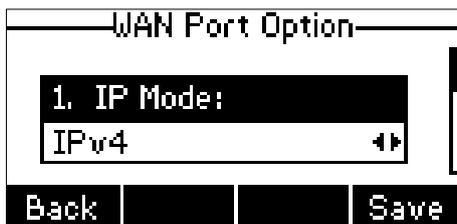
Basic Network Settings

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:

- 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 the **IPv4** and then press the **Enter** soft key.
- Press  to select **Static IPv4 Client** and then press the **Enter** soft key.
- Enter the desired value 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.**

- Press  to select **IPv6** and then press the **Enter** soft key.
- Press  to select **Static IPv6 Client** and then press the **Enter** soft key.
- Enter the desired value 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, 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 then press the **Enter** soft key.
3. Press  to select **PPPoE IPv4 Client** and then 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

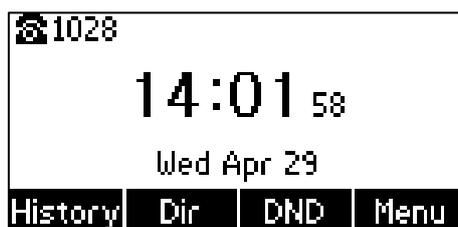
Wrong network settings may result in inaccessibility of your phone and may also have an impact on your network performance. For more information on these parameters, contact your system administrator.

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. The SIP-T19(P) E2 IP phone supports only one account. If your phone is not registered, you may have to register it. For more information on how to register your phone, refer to [Account Registration](#) on page 53.

Idle Screen

If the phone has successfully started up, the idle LCD 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-T19(P) E2 IP phone by personally configuring certain settings, for example, contrast, language and time & date. 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

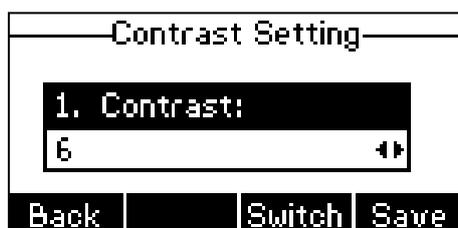
Contrast

You can configure the LCD screen contrast of SIP-T19(P) E2 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**.

Language

The default language of the phone user interface is English. If the language of your web browser is not supported by the phone, the web user interface will use English by default. You can change the language for the phone user interface and the web user interface respectively.

To change the language for the phone user interface:

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

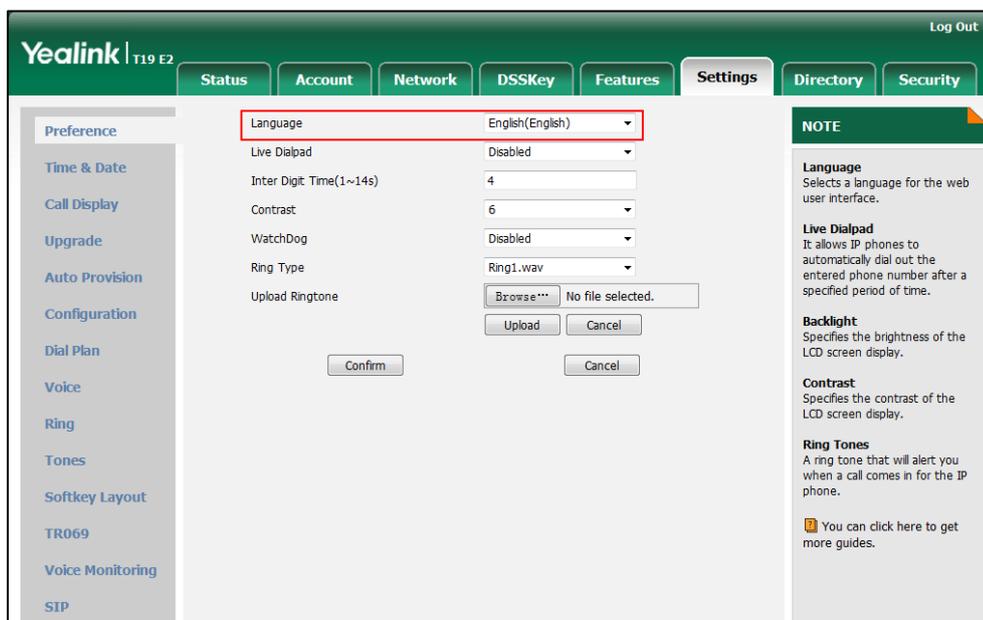


3. Press the **Save** soft key to accept the change.

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

To change the language for the web user interface:

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



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

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

Time & Date

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

To configure the SNTP settings via phone user interface:

1. Press **Menu->Settings->Basic Settings->Time & Date->SNTP Settings**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the time zone that applies to your area from the **Time Zone** field.

The default time zone is "+8".



3. Enter the domain name or IP address of SNTP server in the **NTP Server1** and **NTP Server2** fields respectively.
4. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Daylight Saving** field.
5. Press **◀** or **▶**, or the **Switch** soft key to select the desired time zone name from the **Location** field.

This field appears only if **Daylight Saving** field is selected **Automatic**.

The default time zone name is China(Beijing).

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

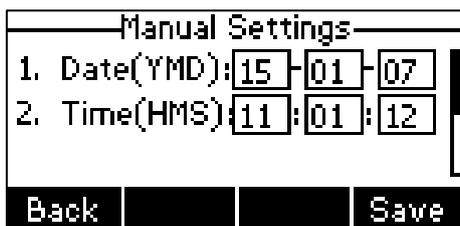
Note

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(YMD)** field.

3. Enter the specific time in the **Time(HMS)** field.

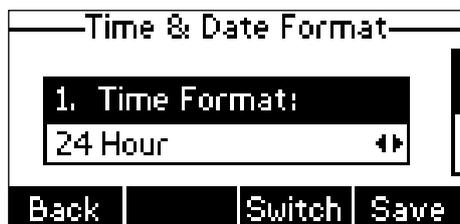


4. Press the **Save** soft key to accept the change.

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 (12 Hour or 24 Hour) from the **Time Format** 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 (2015-04-29)
WWW MMM DD	Wed Apr 29
DD-MMM-YY	29-Apr-15
YYYY-MM-DD	2015-04-29
DD/MM/YYYY	29/04/2015
MM/DD/YY	04/29/15
DD MMM YYYY	29 Apr 2015
WWW DD MMM	Wed 29 Apr

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

Administrator Password

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

To change the administrator password via phone user interface:

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



3. Enter the new password in the **New PWD** field.
4. 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.

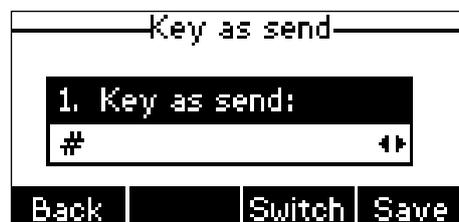
Administrator password is configurable via web user interface at the path **Security->Password**.

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 \leftarrow or \rightarrow , or the **Switch** soft key to select # or * from the **Key as send** field, or select **Disabled** 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**.

Phone Lock

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

Phone lock consists of the following:

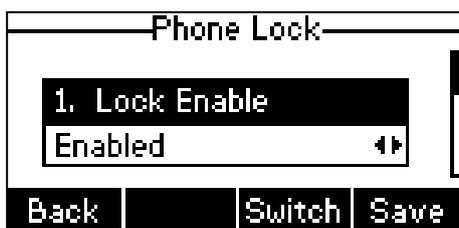
- Menu Key:** The Menu soft key is locked. You cannot access the menu of the phone until unlocked.
- Function Keys:** The function keys are locked. You cannot use the Message, RD, Mute, Transfer, √, navigation keys and soft keys until unlocked.
- All Keys:** All keys are locked except the Volume key, digit keys and speakerphone key. You are only allowed to dial emergency numbers, reject incoming calls by pressing the **Reject** soft key, answer incoming calls by lifting the handset, pressing the Speakerphone key, the HEADSET key, the √ key, or the **Answer** soft key, and end the call by hanging up the handset, pressing the Speakerphone key or the **EndCall** soft key.

Note

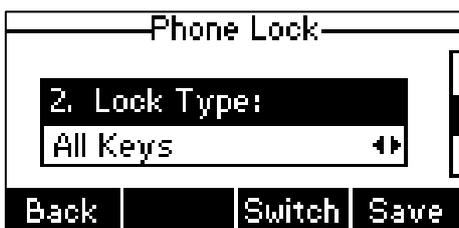
The emergency number setting, if desired, must be set before lock activation. For more information, refer to [Emergency Number](#) on page 60.

To activate the phone lock via phone user interface:

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



3. Press ◀ or ▶, or the **Switch** soft key to select the desired type from the **Lock Type** field.



4. (Optional.) Enter the desired interval of automatic phone lock in the **Lock Time Out**

field.

The default timeout is 0. It means the phone will not be automatically locked. You need to long press  to lock it immediately when the phone is idle.

If it is set to other values except 0 (e.g., 5), the phone will be locked when the phone is inactive in idle screen for the designated time (in seconds).

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

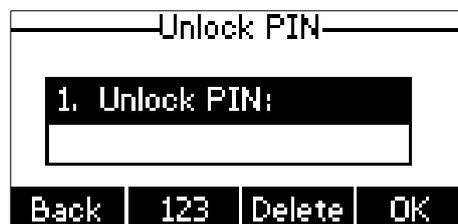
When the phone is locked, the LCD screen prompts "Phone locked." and displays the icon .



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

To unlock the phone via phone user interface:

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



2. Enter the PIN in the **Unlock PIN** field.
3. Press the **OK** soft key to unlock the phone.

The  icon disappears from the LCD screen.

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

Note

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

To change the phone unlock PIN via phone user interface:

1. Press **Menu->Settings->Basic Settings->Change PIN**.

2. Enter the desired value in the **Current PIN**, **New PIN** and **Confirm PIN** fields respectively.



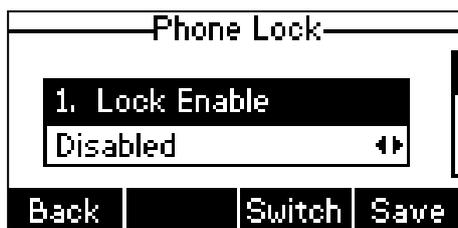
3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

Note

The unlock PIN length must be within 15 digits.

To deactivate the phone lock via phone user interface:

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



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

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

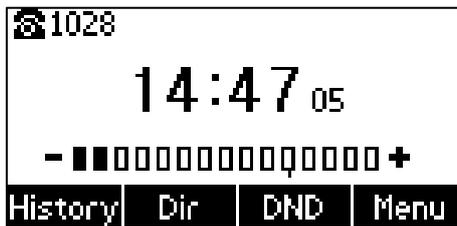
Audio Settings

Volume

You can press the Volume key to adjust the ringer volume when the phone is idle. You can also press the Volume key to adjust the receiver volume of currently engaged audio devices (handset, speakerphone or headset) when the phone is in use.

To adjust the volume when the phone is idle:

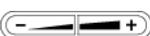
1. Press  to adjust the ringer volume.



Note

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

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

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



Ring Tones

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

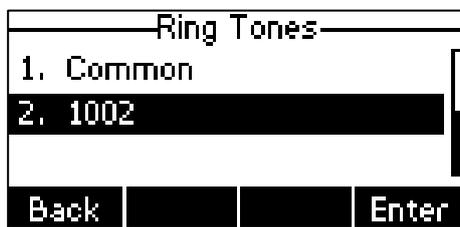


3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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 and then press the **Enter** soft key.



3. Press  or  to select the desired ring tone.
If **Common** is selected, this account will use the ring tone selected for the phone.

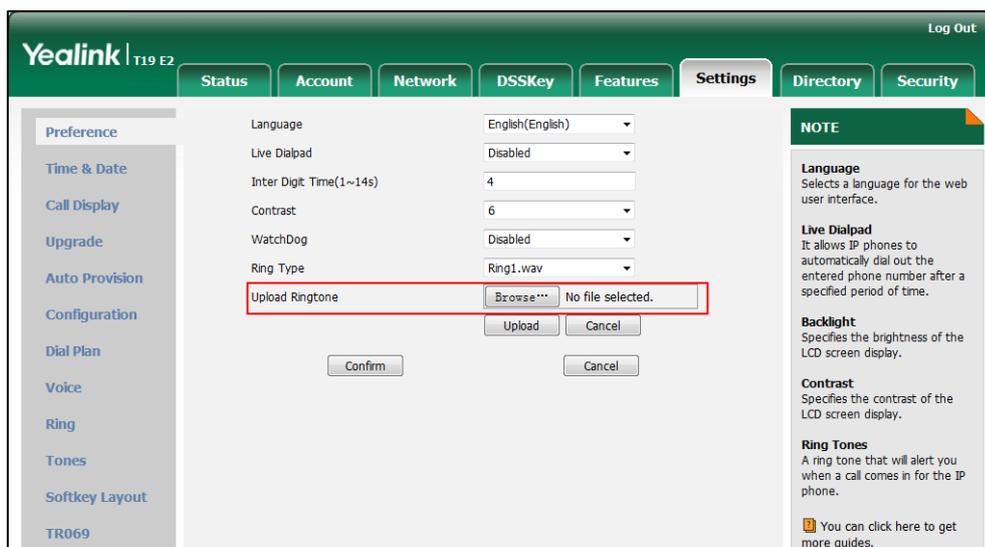


4. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

A ring tone for the account is configurable via web user interface at the path **Account->Basic->Ring Type**.

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

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



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

Note

The priority of ring tone for an incoming call on the phone is as follows:
Contact ring tone (refer to [Adding Contacts](#)) > Group ring tone (refer to [Adding Groups](#)) > Account ring tone > Phone ring tone.

Both single custom ring tone file and total custom ring tone files must be within 100KB.
Uploading custom ring tones for your phone is configurable via web user interface only.

Contact Management

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

- [Directory](#)
- [Local Directory](#)
- [Blacklist](#)
- [Remote Phone Book](#)

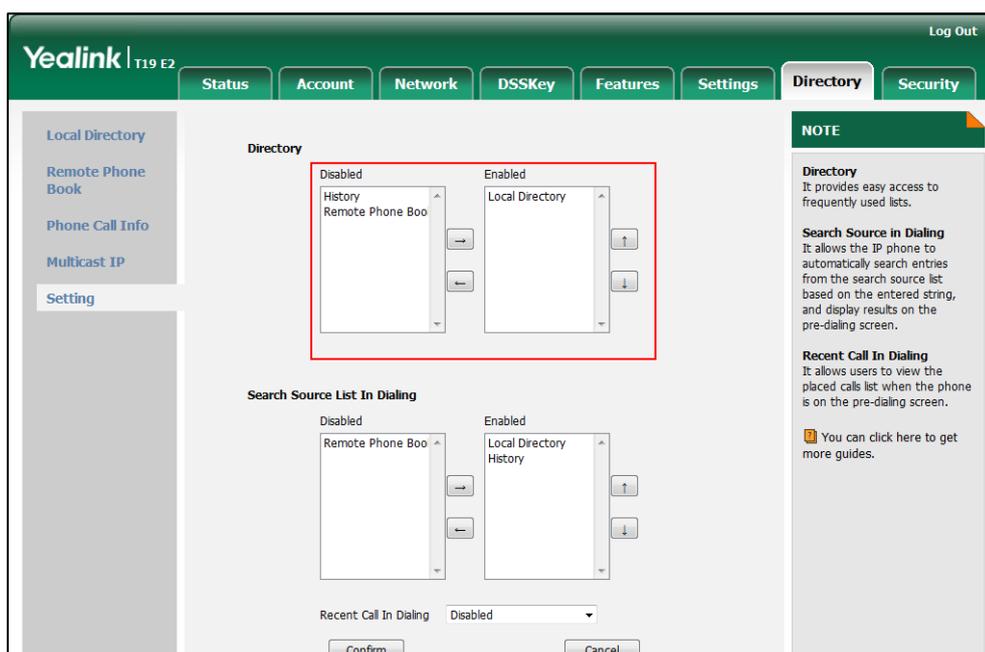
Directory

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

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 enabled lists, select the desired list and then click  or  .

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



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

Note Directory is configurable via web user interface only.

To check the directory via phone user interface:

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



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

Note If the remote phone book is not configured in advance, you cannot see remote phone book list on the phone user interface. For more information on remote phone book, refer to [Remote Phone Book](#) on page 42.

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 **AddGr** soft key.
3. Enter the desired group name in the **Name** field.
4. Press **◀** or **▶**, or **Switch** soft key to select the desired group ring tone from the **Ring** field.

If **Auto** is selected, this group will use the ring tone according to the priority: Contact ring tone (refer to [Adding Contacts](#)) >Account ring tone (refer to [Ring Tones](#)) >Phone ring tone (refer to [Ring Tones](#)). If a specific ring tone is selected, this group will use the ring tone according to the priority: Contact ring tone (refer to [Adding Contacts](#)) >Group ring tone.



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

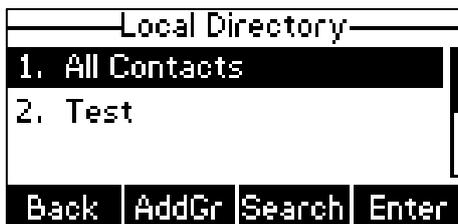
You can also edit or delete any newly added contact groups.

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.

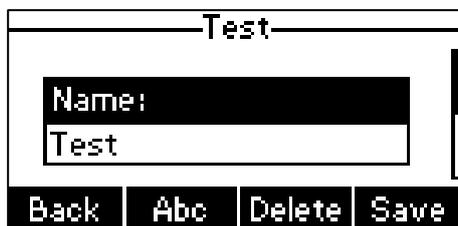


If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired group.
3. Press the **Option** soft key, and then select **Detail**.



4. Press  or  to scroll through 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.
3. Press the **Option** soft key, and then select **Delete**.

The LCD screen prompts the following warning:



4. Press the **OK** soft key to confirm the deletion or the **Cancel** soft key to cancel.
You can also delete all groups by pressing the **Option** soft key and then select **Delete All**.

Adding Contacts

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

- Manually
- From call history
- From 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. Select the desired contact group and then press the **Enter** soft key.
3. Press the **Add** soft key.

- Enter the name and the office, mobile or other numbers in the corresponding fields.



- Press or , 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 according to the priority:
 Group ring tone (refer to [Adding Groups](#)) >Account ring tone (refer to [Ring Tones](#))>Phone ring tone (refer to [Ring Tones](#)).

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

Note

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

Adding Contacts from Call History

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

- Press the **History** soft key.
- Press or to highlight the desired entry.
- Press the **Option** soft key, and then select **Add to Contacts**.



- Press the **OK** soft key, and then edit the contact name.
- Press the **Save** soft key to accept the change.
 The entry is successfully saved to the local directory.

Adding Contacts from Remote Phone Book

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

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

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

If the contact has already existed in the local directory, the LCD screen will prompt "Contact name existed, overwrite?". Press the **OK** soft key to overwrite the original contact in 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. Select the desired contact group and then press the **Enter** soft key.
3. Press \uparrow or \downarrow to highlight the desired contact.
4. Press the **Option** soft key, and then select **Detail**.
5. Press \uparrow or \downarrow to highlight the contact information and then edit.



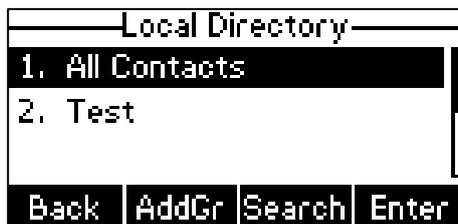
6. Press the **Save** soft key to accept change or the **Back** soft key to cancel.

Deleting Contacts

To delete a contact from the local directory:

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

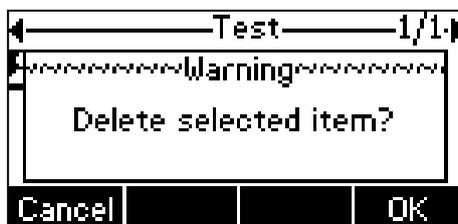
The IP phone enters the local directory directly as there is only Local Directory enabled in the directory by default.



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

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

The LCD screen prompts the following warning:



5. Press the **OK** soft key to confirm the deletion or the **Cancel** soft key to cancel.
- You can also delete all contacts by pressing the **Option** soft key, and then select **Delete All**.

Placing Calls to Contacts

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

1. Press the **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 contact group and then press the **Enter** soft key.
3. Press \uparrow or \downarrow to highlight the desired contact.

4. Do one of the following:
 - If only one number of the contact is stored in the local directory, press the **Send** soft key to dial out the number.
 - If multiple numbers of the contact are stored in the local directory, press the **Send** soft key to display a list of numbers.
Press  or  to highlight 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 the **Search** soft key.
3. 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.

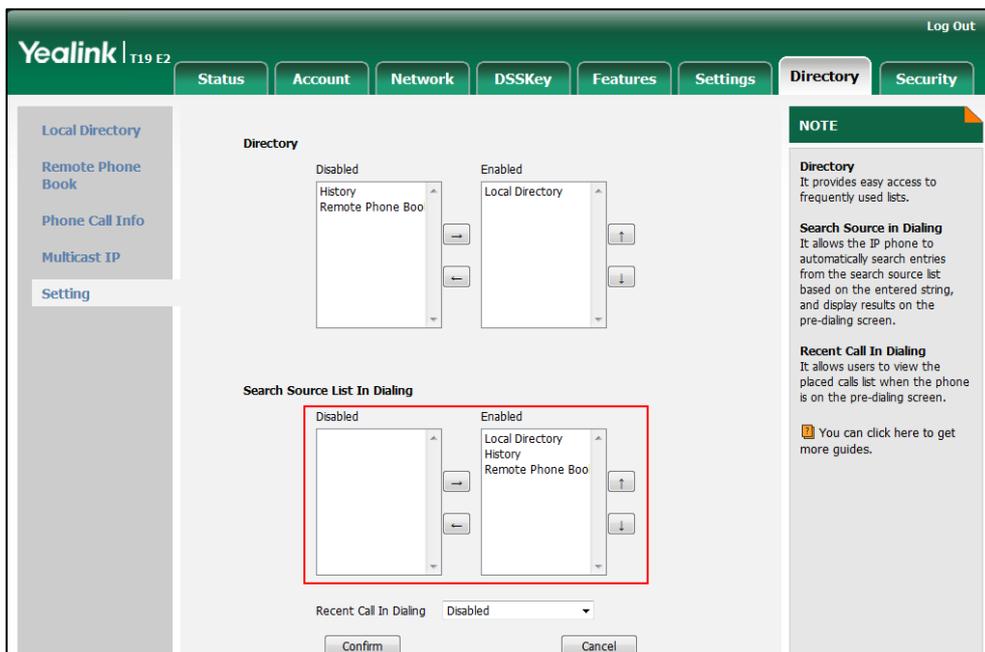
Search Source List in Dialing

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

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.
2. Enter a few continuous characters of the entry’s name or continuous numbers of the entry’s phone number (office, mobile or other number) using the keypad.

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

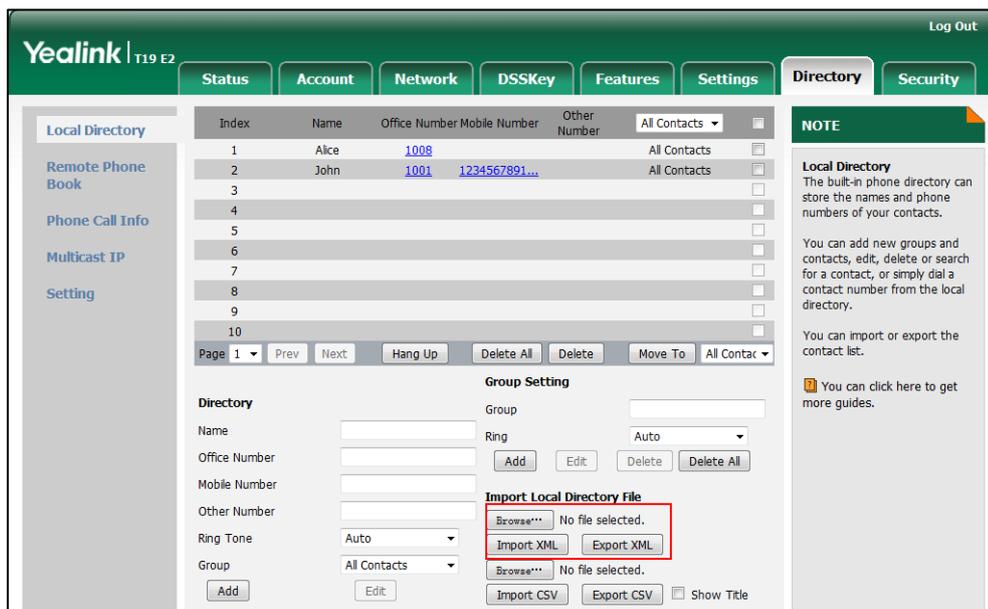


Importing/Exporting Contact Lists

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

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

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

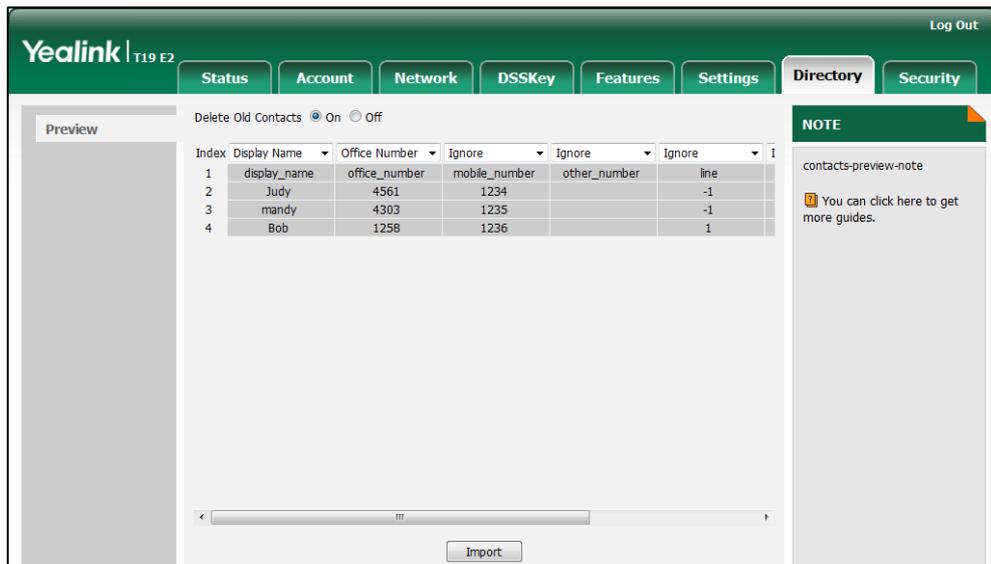


3. Click **Import XML** to import the contact list.
The web user interface prompts “The original contact will be covered, continue?”.
4. Click **OK** to complete importing the contact list.

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

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

3. (Optional.) Check the **Show Title** checkbox.
It will prevent importing the title of the contact information which is located in the first line of the CSV file.
4. Click **Import CSV** to import the contact list.
5. (Optional.) Mark the **On** radio box in the **Delete Old Contacts** field.
It will delete all existing contacts while importing the contact list.
6. Select the contact information you want to import into the local directory from the pull-down list of **Index**.
At least one item should be selected to be imported into the local directory.



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

To export a contact list via web user interface:

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

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

Blacklist

The built-in phone directory can store 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 directory manually:

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



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

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

1. Press 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 contact group and then press the **Enter** soft key.
3. Press \uparrow or \downarrow to highlight the desired contact.
4. Press the **Option** soft key and then select **Add to Blacklist**.

The LCD screen prompts the following warning:



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

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

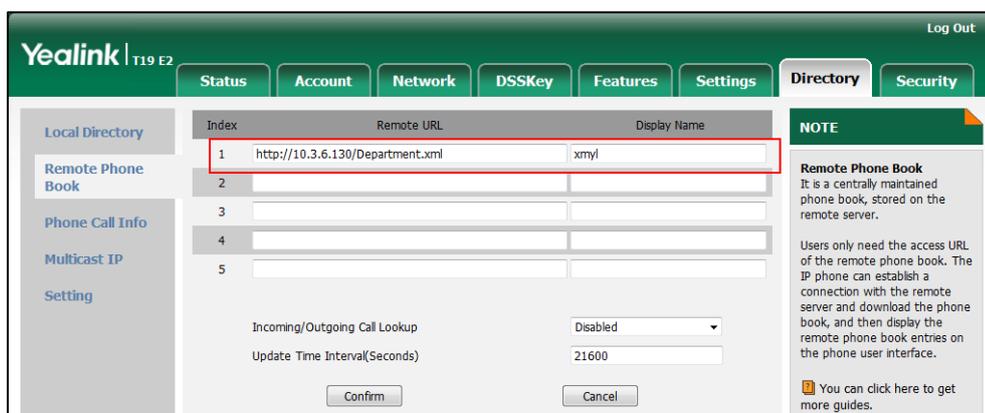
Remote Phone Book

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. 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.

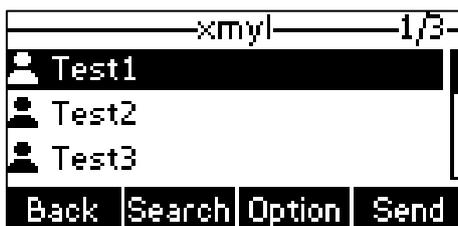
Note

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.

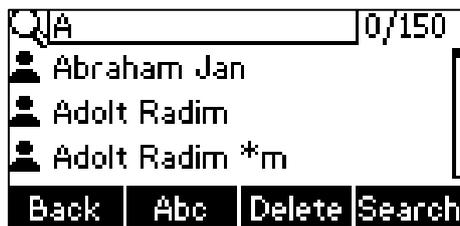
The phone then connects to the remote phone book and proceeds to load it. The contacts in the remote phone book are displayed on the LCD screen.



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

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. Press the **Abc** soft key to change the input mode. And then enter a few continuous characters of the contact name or continuous numbers of the contact number using the keypad.



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

To place a call from the remote phone book:

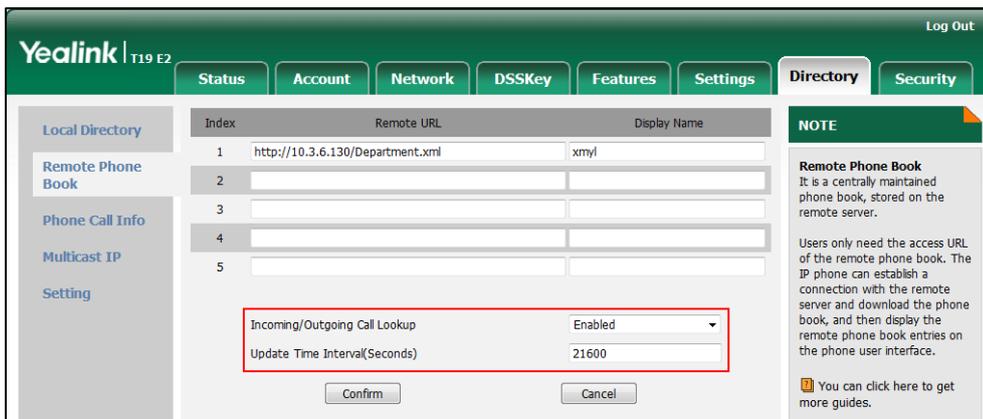
1. Press **Menu->Directory->Remote Phone Book**.
If Remote Phone Book is added to the directory, press **Dir->Remote Phone Book** to enter the remote phone book.
2. Select the desired remote group, and then press the **Enter** soft key to load the remote phone book.
3. Select the desired contact in the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

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

To configure incoming/outgoing call lookup and update time interval via web user interface:

1. Click on **Directory->Remote Phone Book**.
2. Select **Enabled** from the pull-down list of **Incoming/Outgoing Call Lookup**.

- Enter the desired refresh period in the **Update Time Interval(Seconds)** field.
The default value is 21600 seconds.



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

Call History Management

The SIP-T19(P) E2 IP phone maintains call history lists of Placed calls, Received calls, Missed calls and Forwarded calls. Each call history list supports up to 100 entries. 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:

- Press **Menu->Features->History Setting**.
- Press **◀** or **▶**, or the **Switch** soft key to select **Disabled** from the **History Record** field.



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

To view the call history:

- Press the **History** soft key.
The LCD screen displays all call records.
- Press **◀** or **▶** to switch between **All Calls, Placed Calls, Received Calls, Missed Calls** and **Forwarded Calls**.

3. Press  or  to select the desired entry.
4. Press the **Option** soft key, and then select **Detail**.

The detailed information of the entry appears on the LCD screen.

To place a call from the call history list:

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

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

1. Press the **History** soft key.
2. Press  or  to switch between **All Calls, Placed Calls, Received Calls, Missed Calls** and **Forwarded Calls**.
3. Press  or  to select the desired entry.
4. Press the **Option** soft key, and then select **Add to Contacts** (or **Add to Blacklist**).
5. Enter the desired values in the corresponding fields.
6. Press the **Save** soft key.

For more information, refer to [Contact Management](#) on page 29.

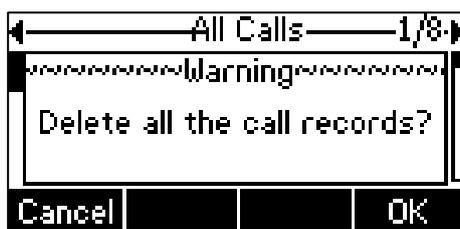
To delete an entry from the call history list:

1. Press the **History** soft key.
2. Press  or  to switch between **All Calls, Placed Calls, Received Calls, Missed 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, Placed Calls, Received Calls, Missed Calls** and **Forwarded Calls**.
3. Press the **Option** soft key, and then select **Delete All**.
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.

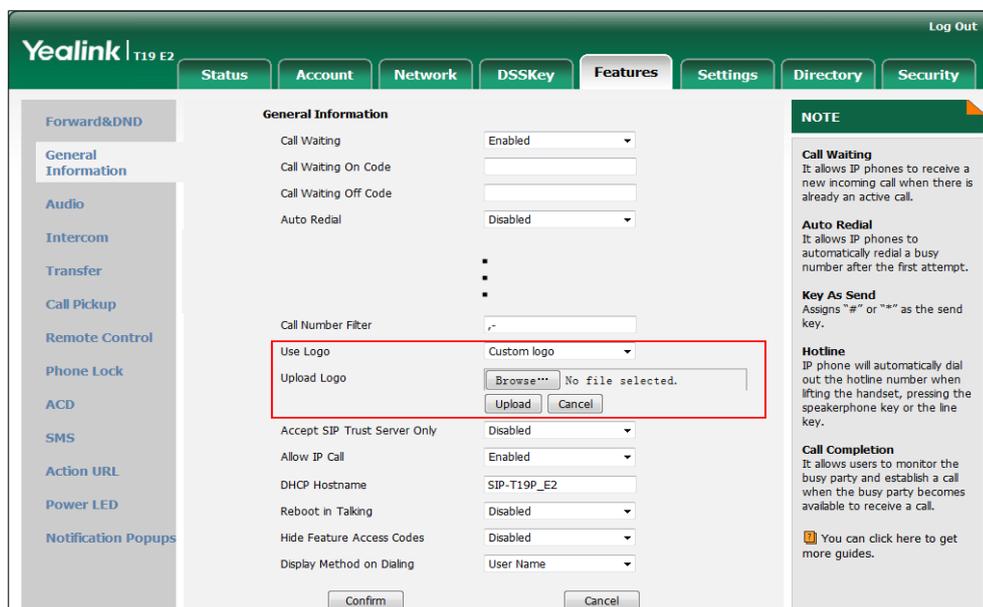
System Customizations

Logo Customization

You can upload your custom logo which will be shown 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 locate the logo file from your local system.



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

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

Note

Delete item will appear after you upload a custom logo, you can click **Delete** to delete the custom logo.

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

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

Headset Mode Activation/Deactivation

To activate the headset mode:

1. Press  on the phone.

The headset icon on the idle screen indicates that the headset mode is activated.

Press the **Answer** soft key to answer an incoming call. The call will be connected to your headset automatically.

Enter the desired number and then press the **Send** soft key, the phone will 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 63.

To deactivate the headset mode:

1. Press  again on the phone.

The headset icon disappears from the idle screen indicates the headset mode is deactivated.

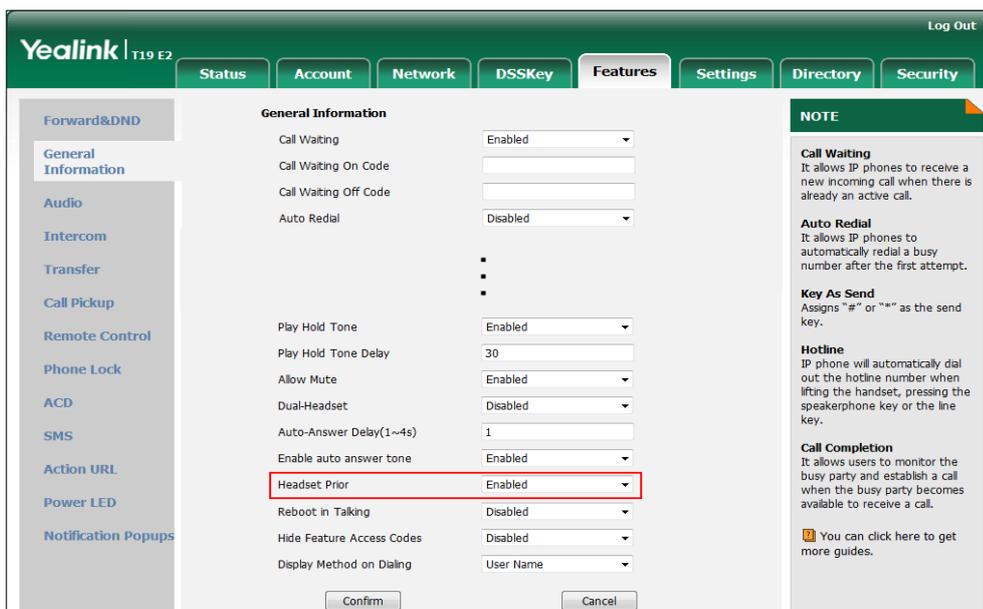
Headset Prior

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

To enable headset prior via web user interface:

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

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



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

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

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

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

If headset prior is disabled, the headset mode can be deactivated by pressing the speakerphone key or the **Headset** key.

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**.

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

The screenshot shows the Yealink T19 E2 web interface. The 'Features' tab is selected, and the 'General Information' section is active. The 'Dual-Headset' option is highlighted with a red box and set to 'Enabled'. Other settings include Call Waiting (Enabled), Call Waiting On Code, Call Waiting Off Code, Auto Redial (Disabled), Play Hold Tone (Enabled), Play Hold Tone Delay (30), Allow Mute (Enabled), Auto-Answer Delay (1), Enable auto answer tone (Enabled), Headset Prior (Disabled), Reboot in Talking (Disabled), Hide Feature Access Codes (Disabled), and Display Method on Dialing (User Name). A 'NOTE' section on the right provides details for various features:

- Call Waiting:** It allows IP phones to receive a new incoming call when there is already an active call.
- Auto Redial:** It allows IP phones to automatically redial a busy number after the first attempt.
- Key As Send:** Assigns "*" or "#" as the send key.
- Hotline:** IP phone will automatically dial out the hotline number when lifting the handset, pressing the speakerphone key or the line key.
- Call Completion:** It allows users to monitor the busy party and establish a call when the busy party becomes available to receive a call.

At the bottom of the settings page, there are 'Confirm' and 'Cancel' buttons.

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

Note Dual headset is configurable via web user interface only.

Programmable Keys

You can customize the soft keys, navigation keys and function keys on the keypad. The SIP-T19(P) E2 IP phone supports 11 programmable keys.

To customize the programmable keys via web user interface:

- Click on **DSSKey->Programmable Key**.

2. Customize specific features for these keys.

Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local History			
SoftKey 2	Directory	N/A			
SoftKey 3	DND	N/A			
SoftKey 4	Menu	N/A			
Up	History	Local History			
Down	Directory	N/A			
Left	N/A	N/A			
Right	N/A	N/A			
OK	Status	N/A			
Mute	N/A	N/A			
Tran	Forward	N/A			

3. (Optional.) Enter a string that will appear on the LCD screen in the **Label** field.

Label is configurable only when customizing SoftKey 1-4.

4. 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.

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

- Speed Dial
- Direct Pickup
- Group Pickup
- Prefix
- Local Directory
- Local Group
- XML Directory
- XML Group
- XML Brower
- SMS
- New SMS
- Zero Touch
- Phone Lock
- Directory

For the features not listed above, refer to [Basic Call Features](#) on page 63 and [Advanced Phone Features](#) on page 89. 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)

Usage: Press the programable key to dial out the number specified in the **Value** field.

Direct Pickup

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

Dependencies: *Type (Direct Pickup)*

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

Usage: Press the programable 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 (Group Pickup)*

Value (the group pickup feature code)

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

Prefix

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

Dependencies: *Type (Prefix)*

Value (the prefix number)

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

Local Directory

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

Dependencies: *Type (Local Directory)*

Usage: Press the programable key to access the local directory quickly.

Local Group

You can use this key feature to access the group in the local directory quickly. For more

information, refer to [Local Directory](#) on page 30.

Dependencies: *Type (Local Group)*

Line (the contact group you want to access)

Usage: Press the programable key to access the contact group specified in the **Line** field.

XML Directory

You can use this key feature to access the corporate directory quickly. For more information, refer to [Remote Phone Book](#) on page 42.

Dependencies: *Type (XML Directory)*

Usage: Press the programable key to access the corporate directory quickly.

XML Group

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

Dependencies: *Type (XML Group)*

Line (the remote group you want to access if the remote phone book is configured)

Usage: Press the programable key to access the remote group specified in the **Line** field.

XML Browser

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

Dependencies: *Type (XML Browser)*

Value (the access URL for xml browser)

Usage: Press the programable key to access the XML browser specified in the **Value** field.

SMS

You can use this key feature to quick access text message. For more information, refer to [Short Message Service \(SMS\)](#) on page 109.

Dependencies: *Type (SMS)*

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

New SMS

You can use this key feature to quick access the new text message. For more information, refer to [Short Message Service \(SMS\)](#) on page 109.

Dependencies: *Type (New SMS)*

Usage: Press the programable key when the phone is idle to access the New Message screen. You can enter the text message and then send it.

Zero Touch

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

Dependencies: *Type (Zero Touch)*

Usage:

1. Press the programable 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.

Phone Lock

You can use this key feature to immediately lock your phone instead of long pressing . For more information, refer to [Phone Lock](#) on page 24.

Dependencies: *Type (Phone Lock)*

Usage: When the phone lock feature is enabled, press the programable key to immediately lock 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 (Directory)*

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

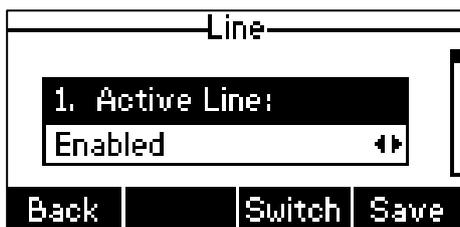
Account Registration

You can only register one account on the SIP-T19(P) E2 IP phone.

To register an account via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (default password: admin)
->**Accounts**.
2. Press the **Enter** soft key.

3. Press  or , or the **Switch** soft key to select **Enabled** from the **Active Line** field.



4. Enter the desired value in **Label**, **Display Name**, **Register Name**, **User Name**, **Password** and **SIP Server1/2** fields respectively. Contact your system administrator for more information.
5. If you use the outbound proxy servers, do the following:
 - 1) Press  or , or the **Switch** soft key to select **Enabled** from the **Outbound Status** field.
 - 2) Enter the desired value in the **Outbound Proxy1/2** and **Proxy Fallback Interval** fields respectively. Contact your system administrator for more information.
6. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

To disable an account via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (default password: admin) **->Accounts**.
2. Press  or , or the **Switch** soft key to select **Disabled** from the **Active Line** field.
3. 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**.

Dial Plan

Dial plan is a string of characters that governs the way your SIP-T19(P) E2 IP phone processes the inputs received from your phone keypad. The SIP-T19(P) E2 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.
-	Numeric ranges are allowed within the brackets: Digit "-" Digit. Example: "[5-7]" would match the number "5", "6" or "7".
[]	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: "([1-9])([2-7])3" would match "923", "153", "773", etc.
\$	The "\$" should be followed by the sequence number of a parenthesis. The "\$" plus the sequence number means the whole character or characters placed in the parenthesis. The number directs to the right parenthesis when there are more than one. Example: A replace rule configuration, Prefix: "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 2354599 ". "\$1" means 3 digits in the first parenthesis, that is, "235". "\$2" means 2 digits in the second parenthesis, that is, "99".

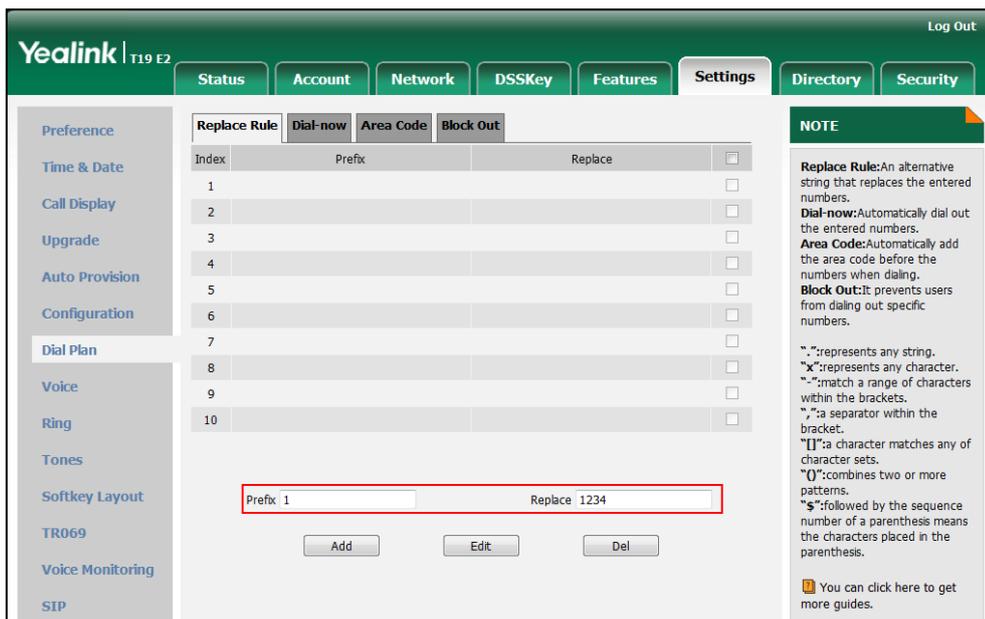
Replace Rule

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: 1" and "Replace: 1234". When trying to dial out the number "1234", you just need to enter "1" on the phone and then press the **Send** soft key.

To add a replace rule via web user interface:

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Enter the string (e.g., 1) in the **Prefix** field.

- Enter the string (e.g., 1234) in the **Replace** field.



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

When you enter the number “1” using the keypad and then press the **Send** soft key, the phone will dial out “1234” instead.

To edit a replace rule via web user interface:

- Click on **Settings->Dial Plan->Replace Rule**.
- Select the desired replace rule by checking the check box.
- Edit the values in the **Prefix** and **Replace** fields.
- Click **Edit** to accept the change.

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

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

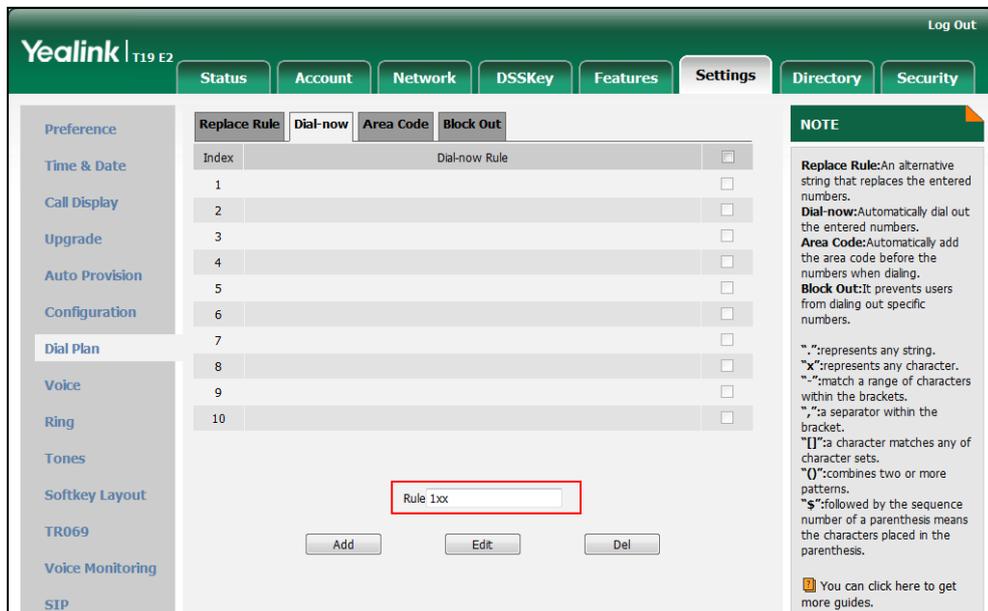
Note Replace rule is configurable via web user interface only.

Dial-now

You can configure one or more dial-now rules (up to 100) on your phone. When the dialed 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., 1xx) in the **Rule** field.



3. 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 pressing any key.

Note

You can also edit or delete the dial-now rule, refer to [Replace Rule](#) on page 55 for more information.

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

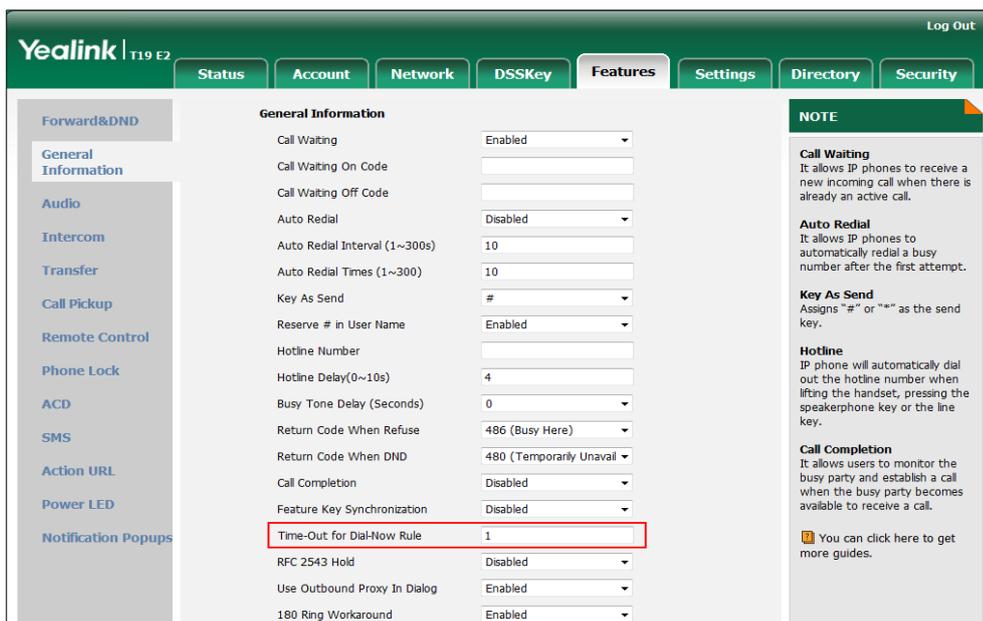
Delay Time for Dial-now Rule

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**.

- Enter the time between 1 and 14 (seconds) in the **Time-Out for Dial-Now Rule** field.



- 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:

- Click on **Settings->Dial Plan->Area Code**.

- Enter the desired values in the **Code**, **Min Length (1-15)** and **Max Length (1-15)** fields.

The screenshot shows the Yealink T19 E2 web interface. The 'Settings' menu is open, and the 'Block Out' tab is selected. The 'Code' field contains '0592', 'Min Length (1-15)' is '1', and 'Max Length (1-15)' is '15'. There are 'Confirm' and 'Cancel' buttons below the input fields. On the right, a 'NOTE' section provides definitions for 'Replace Rule', 'Dial-now', and 'Area Code', along with a legend for special characters like '.', 'x', '-', '[]', '()', and '\$'.

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

Note

The default value of minimum and maximum length is 1 and 15 respectively.
Area code is configurable via web user interface only.

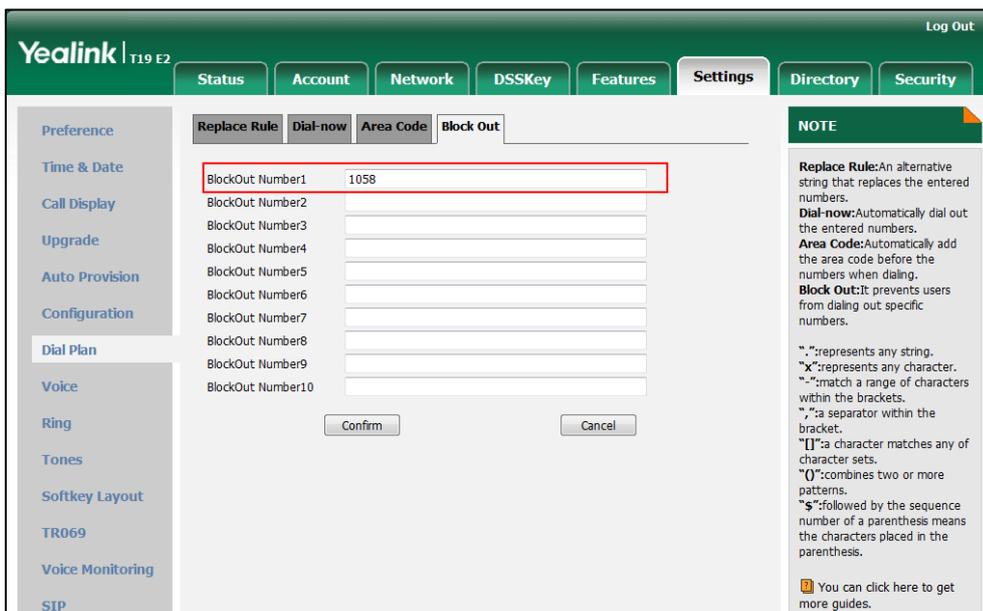
Block Out

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

To add a block out number via web user interface:

- Click on **Settings->Dial Plan->Block Out**.

2. Enter the desired value in the **BlockOut Number** field.



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

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

Emergency Number

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

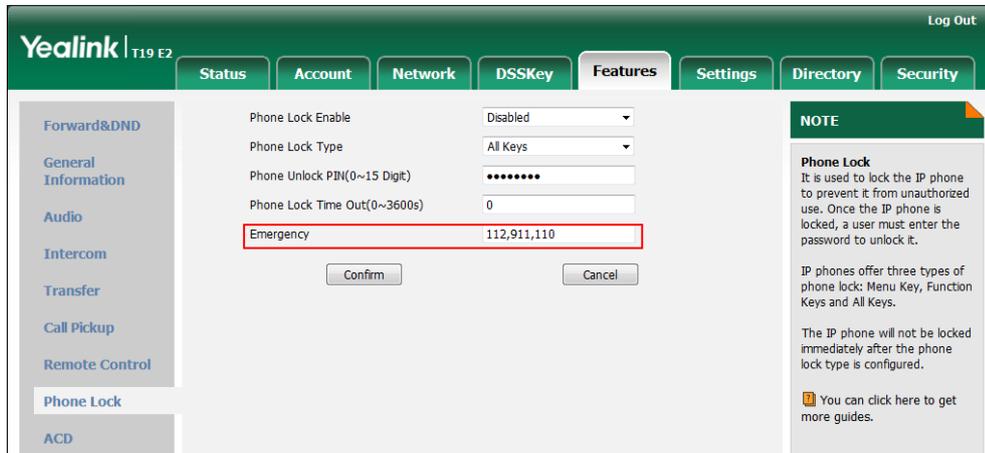
You can specify the emergency telephone numbers on the IP phone for contacting the emergency services in an emergency situation. You can dial these numbers when the phone is locked. For more information on phone lock, refer to Phone Lock on page 24.

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.



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

Note Emergency number is configurable via web user interface only.

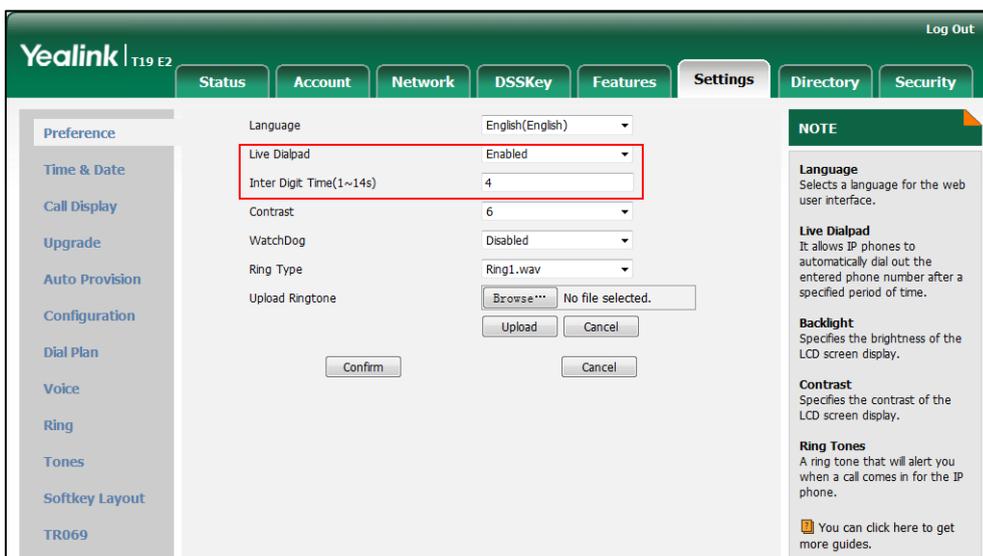
Live Dialpad

You can enable live dialpad feature on the SIP-T19(P) E2 IP phone, which enables the IP phone to automatically dial out a phone number without pressing the send key. You can also configure a delay, and then the phone will dial out the phone 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 4 seconds.



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

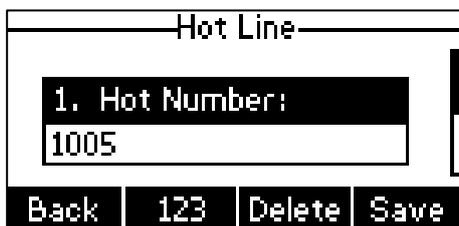
Note Live dialpad is configurable via web user interface only.

Hotline

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

To configure the hot line 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 the **Hotline Delay** field.
The valid values range from 0 to 10 (seconds) and the default value is 4.
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-T19(P) E2 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-T19(P) E2 IP phone. Topics include:

- [Placing Calls](#)
- [Answering Calls](#)
- [Ending Calls](#)
- [Redialing Numbers](#)
- [Recent Call In Dialing](#)
- [Auto Answer](#)
- [Auto Redial](#)
- [Call Completion](#)
- [ReCall](#)
- [Call Mute](#)
- [Call Hold/Resume](#)
- [Do Not Disturb \(DND\)](#)
- [Call Forward](#)
- [Call Transfer](#)
- [Call Waiting](#)
- [Conference](#)
- [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-T19(P) E2 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, called party's name and phone number are visible on the LCD screen. In the figure below, the call to "Tom" (the phone number: 1002) has lasted 34 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 23.

Note

You can also dial using the SIP URI or IP address. To obtain the IP address of a phone, press the  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  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:

1. With the optional headset connected, press  to activate the headset mode.
2. Enter the desired number using the keypad.
3. Press , , or the **Send** soft key.

Note

To permanently use the headset mode, refer to [Headset Prior](#) on page 47.

The SIP-T19(P) E2 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 time, other calls are placed on hold. The SIP-T19(P) E2 IP phone can handle a maximum of 2 calls at one time.

To place multiple calls:

You can have more than one call on your SIP-T19(P) E2 IP phone. To place a new call during an active call:

1. Press the **Hold** soft key to place the original call on hold.
2. Press the **NewCall** soft key.
3. Enter the desired number using the keypad.
4. 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 reject incoming calls by pressing the **Reject** soft key. You can also activate Do Not Disturb mode to ignore all incoming calls without ring on your phone. For more information, refer to [Do Not Disturb \(DND\)](#) on page 73.

You can forward incoming calls to someone else by pressing the **FWD** soft key. For more information, refer to [Call Forward](#) on page 74.

Answering When Not in Another Call

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

To answer a call using the handset:

1. Pick up the handset.

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

Do one of the following:

- Press  .
- With the handset on-hook and the headset mode deactivated, press the **Answer** soft key.

To answer a call using the headset:

Do one of the following:

- Press  .
- With the headset mode activated, press the **Answer** soft key.

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.

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:

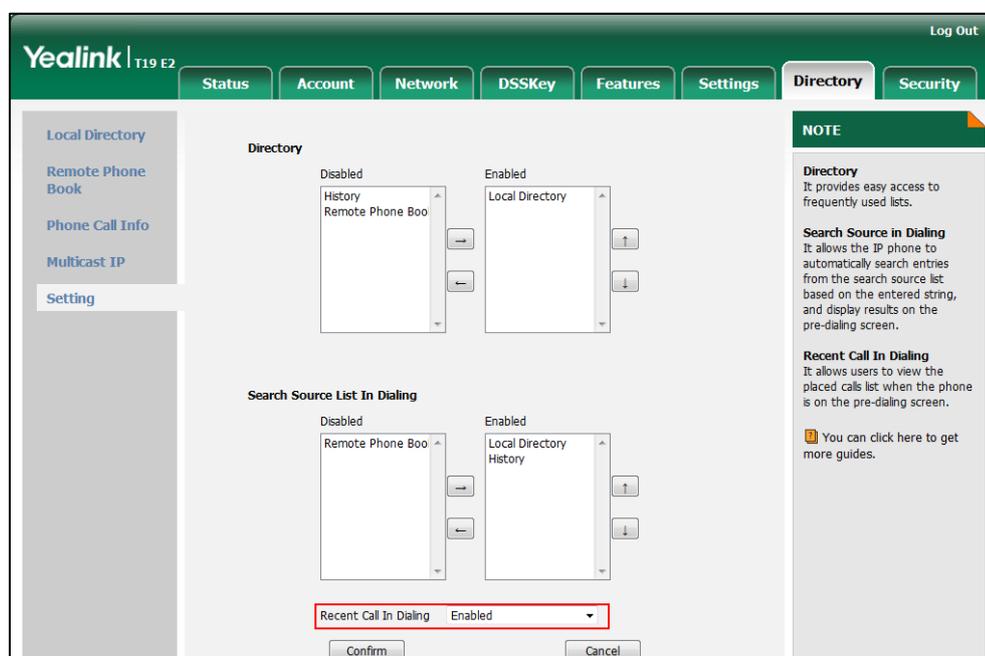
1. Press **RD** when the phone is idle.
2. Press **▲** or **▼** to select the desired entry from the placed calls list, and then press **RD** or the **Send** soft key.

Recent Call In Dialing

To view the placed calls list when the phone is on the pre-dialing screen, 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.

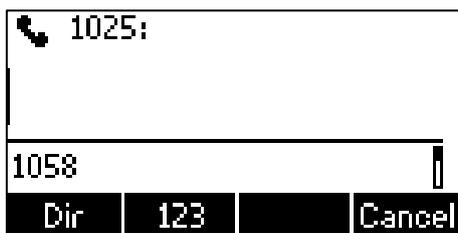
Note

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

To view placed calls list when the phone is on the pre-dialing screen:

1. Pick up the handset or press the speakerphone.

The LCD screen displays the placed calls list.

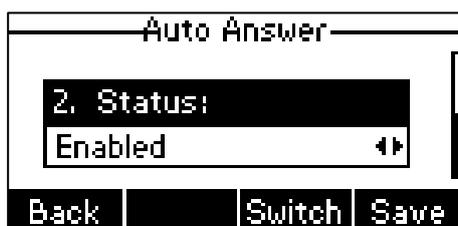


Auto Answer

You can use auto answer feature to automatically answer an incoming call.

To configure auto answer via phone user interface:

1. Press **Menu->Features->Auto Answer**.
2. Press to select the **Status**, and the press or , or the **Switch** soft key to select **Enabled** from the **Status** field.



3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.
The 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 **◀** or **▶**, or the **Switch** soft key to select **Enabled** from the **Auto Redial** field.



3. Enter the desired time (in seconds) in the **Redial Interval** field.
The default time interval is 10 seconds.
4. Enter the desired 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 LCD screen prompts the following:



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.

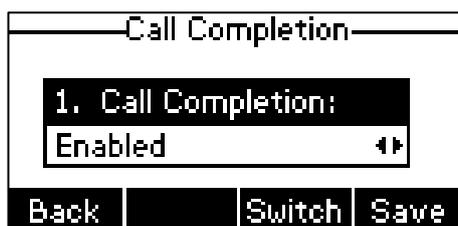
The phone will retry as many times as configured until the called party is idle.

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 **Enabled** from the **Call Completion** field.

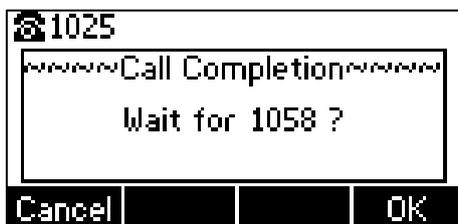


3. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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 will appear 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.

ReCall

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

To configure a recall key via web user interface:

1. Click on **DSSKey->Programable Key**.
2. Select the desired programable key.
3. Select **ReCall** from the pull-down list of **Type**.

Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local History			
SoftKey 2	Directory	N/A			
SoftKey 3	Direct Pickup	N/A			
SoftKey 3	Group Pickup	N/A			
SoftKey 4	Prefix	N/A			
Up	Local Group	Local History			
	XML Group				
Down	XML Browser				
	History	N/A			
Left	Menu	N/A			
	Forward	N/A			
Right	DND	N/A			
	ReCall	N/A			
OK	SMS	N/A			
	New SMS	N/A			
Mute	XML Directory	N/A			
	Status	N/A			
Tran	Local Directory	N/A			
	Hot Desking	N/A			
	Zero Touch	N/A			
	Phone Lock				
	Directory				

NOTE: Programmable Keys Customizes the soft keys, navigation keys and function keys. You can click here to get more guides.

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

Note

A recall key is configurable via web user interface only.

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.

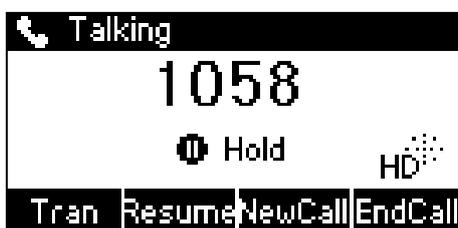
Call Hold/Resume

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.



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:

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

If more than one call is placed on hold, a numbered prompt appears on the LCD screen, for example "1/2", indicating that this is the first call out of two calls.

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

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

To activate DND via phone user interface:

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

The  icon on the status bar indicates that DND is enabled.

Incoming calls will be rejected automatically and "**1 New Missed Call(s)**" will appear on the LCD screen.



Note

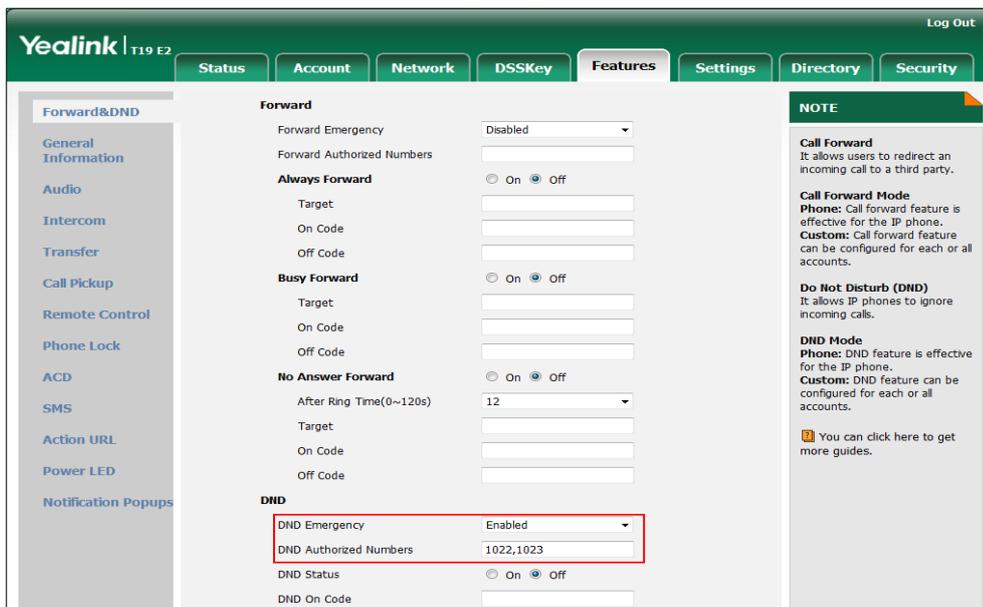
When DND and busy forward are enabled at the same time, calls will be sent to the configured destination number. For more information on busy forward, refer to [Call Forward](#) on page 74.

DND is configurable via web user interface at the path **Features->Forward & DND**.

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, a 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 the dynamic forwarding.

Static Forwarding

Three types of static forwarding:

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

To enable call forward via phone user interface:

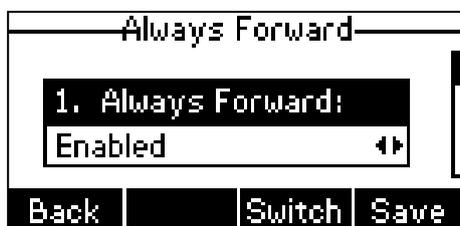
1. Press **Menu->Features->Call Forward**.
2. Press **▲** or **▼** to select the desired forwarding type, and then press the **Enter**

soft key.

3. Depending on your selection:

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

- 1) Press ◀ or ▶, or the **Switch** soft key to select **Enabled** from the **Always Forward** field.



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

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

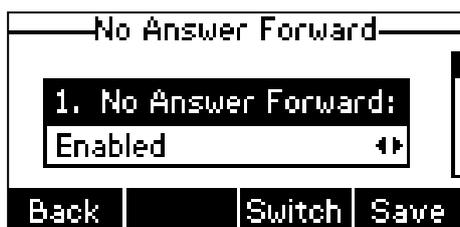
- 1) Press ◀ or ▶, or the **Switch** soft key to select **Enabled** from the **Busy Forward** field.



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

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

- 1) Press ◀ or ▶, or the **Switch** soft key to select **Enabled** 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 or off code respectively in the **On Code** or **Off Code** field.

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

The  icon on the status bar indicates that the call forward is enabled.

Call forward is configurable via web user interface at the path **Features->Forward & DND**.

Note

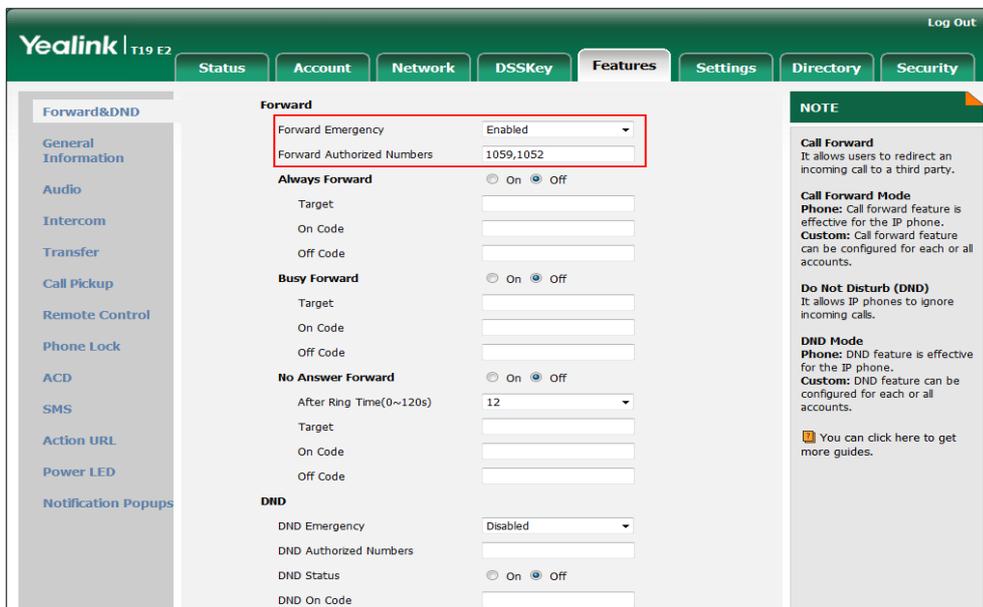
You can also enter the SIP URI or IP address in the **Forward to** field. For more information on using the SIP URI or IP address, refer to [Placing Calls](#) on page 63.

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 configure the forward authorized numbers via web user interface:

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

For multiple numbers, enter a comma between every two numbers.



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

When call forward is enabled on the phone, the phone cannot forward incoming calls from the numbers specified in the **Forward Authorized Numbers** field.

Note

Forward authorized number is configurable via web user interface only.

To disable call forward via phone user interface:

Do one of the following:

- Press  when the phone is idle.
- Press **Menu->Features->Call Forward**.

Press  or  to select the desired forwarding type and then press the **Enter** soft key.

Press  or , or the **Switch** soft key to select **Disabled** to disable the call forward.

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

Dynamic Forwarding**To forward an incoming call to another party:**

1. When the phone is ringing, press the **FWD** soft key.
2. Enter the number you want to forward the incoming call to.



Forward to:
1058
1058
Send 123 Delete Cancel

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

The LCD screen prompts a call forward message.

Note

When the phone forwards a call, a 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 the target phone is ringing.

- **Attended Transfer:** Transfer a call with prior consulting.

To perform a blind transfer:

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

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

To perform a semi-attended transfer:

1. Press  or the **Tran** soft key during a call.
2. Do one of the following:
 - 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 then press  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 should be configured in advance. Refer to [Directory](#) on page 29 and [Remote Phone Book](#) on page 42 for more information).
3. Press  or  to dial out.
4. Press  or the **Tran** soft key to complete the transfer when receiving 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 then press  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 should be configured in advance. Refer to [Directory](#) on page 29 and [Remote Phone Book](#) on page 42 for more information).
3. Press  or  to dial out.

- 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.

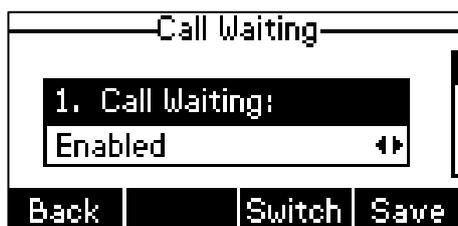
You can cancel the transfer before the call is connected by pressing the **Cancel** soft key.

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:

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



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

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

Conference

You can create a conference with other two parties using the phone's local conference. You can create a conference between an active call and a call on hold by pressing the **Conf** soft key. The SIP-T19(P) E2 IP phone also supports network conference.

Note

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

Local Conference

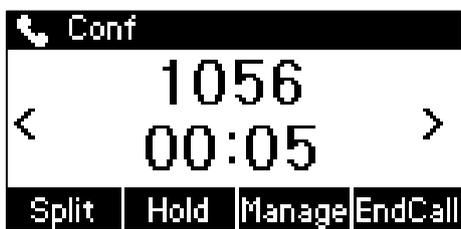
The SIP-T19(P) E2 IP phone supports up to 3 parties (including yourself) in a conference call. This is the default method of conference called Local Conference.

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 **NewCall** soft key to place a new call.

The active call is placed on hold.

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



You can press  or  to see all parties in the conference.

During the conference call, you can do the following actions:

- 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 screen.
- Press  to mute the conference call.
- Press the **EndCall** soft key to drop the conference call.

Network Conference

You can use network conference feature on the SIP-T19(P) E2 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->Advanced**.
2. Select **Network Conference** from the pull-down list of **Conference Type**.
3. Enter the conference URI (e.g., conference@example.com) in the **Conference URI** field.

The screenshot shows the Yealink T19 E2 web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. In the configuration area, the 'Conference Type' dropdown menu is set to 'Network Conference' and the 'Conference URI' text field contains 'conference@example.com'. These two fields are highlighted with a red rectangular box. Other settings visible include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'DTMF Type' (RFC2833), 'SIP Send MAC' (Disabled), 'SIP Send Line' (Disabled), and 'SIP Registration Retry Timer(0~1800s)' (30). A 'NOTE' section on the right provides details about DTMF, Session Timer, Busy Lamp Field/BLF List, and Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA). At the bottom of the configuration area, there are 'Confirm' and 'Cancel' buttons.

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

To set up a network conference call:

1. Place a call to the first party.
2. Press the **NewCall** soft key to place a new call.
The active call is placed on hold.
3. Enter the number of the second party and press  ,  , or the **Send** soft key.
4. When the second party answers the call, press the **Conf** soft key to add the second party to the conference.
5. Press the **NewCall** soft key to place a new call.
6. The conference is placed on hold.
7. Enter the number of the new party and then press  ,  , or the **Send** soft key.
8. 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 on specific servers may be different from introduced above. Contact your system administrator for more information.

Call Pickup

You can use call pickup to answer someone else’s incoming call on your phone. The SIP-T19(P) E2 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.

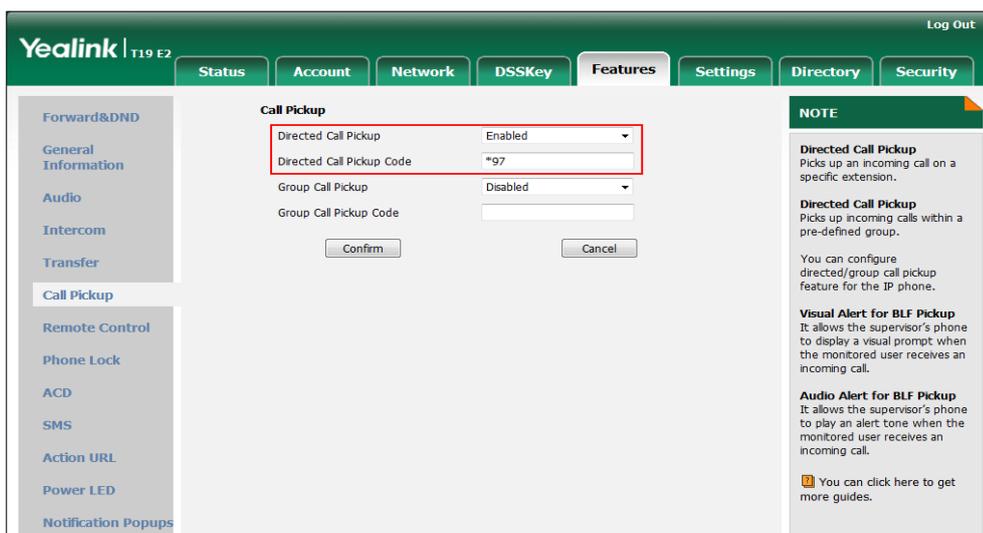
You can pick up an incoming call by using the **DPickup/GPickup** soft key. To use call pickup, you need to configure the call pickup code beforehand on a global or per-line basis via web user interface.

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

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



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

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

To pick up a call directly:

1. Pick up the handset or press the speakerphone and then press the **More** soft key. The **DPickup** soft key appears on the LCD screen.



2. Press the **DPickup** soft key on your phone when the target phone receives an incoming call.
3. Enter the phone number which is receiving an incoming call.
4. Press the **DPickup** soft key again. The call is answered on your phone.

Group Call Pickup

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

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

The screenshot shows the 'Call Pickup' configuration page in the Yealink T19 E2 web interface. The 'Group Call Pickup' dropdown menu is set to 'Enabled', and the 'Group Call Pickup Code' text field contains '*98'. A red rectangular box highlights these two elements. Below the fields are 'Confirm' and 'Cancel' buttons. On the right side, a 'NOTE' section provides details about 'Directed Call Pickup' and 'Visual Alert for BLF Pickup'.

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

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

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

The screenshot shows the 'Advanced' configuration page under the 'Account' section in the Yealink T19 E2 web interface. The 'Group Call Pickup Code' text field is highlighted with a red rectangular box and contains '*98'. Other fields include 'Keep Alive Type', 'Keep Alive Interval(Seconds)', 'RPort', 'Subscribe Period(Seconds)', 'DTMF Type', 'SIP Server Type', 'Music Server URI', 'Directed Call Pickup Code', 'Distinctive Ring Tones', 'Unregister When Reboot', 'VQ RTPC-XR Collector name', 'VQ RTPC-XR Collector address', and 'VQ RTPC-XR Collector port'. 'Confirm' and 'Cancel' buttons are at the bottom. A 'NOTE' section on the right provides information about 'DTMF', 'Session Timer', 'Busy Lamp Field/BLF List', and 'Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA)'.

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

To pick up a call in the group:

1. Pick up the handset or press the speakerphone.

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.

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

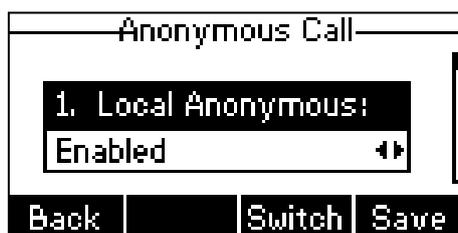
You can use anonymous call to block your identify and phone number from appearing to the called party when you call someone. For example, you want to call to consult some services, but don't want to be harassed. Anonymous call is configurable on a per-line basis. You can also configure the phone to send anonymous call on/off code to the server to activate/deactivate anonymous call on the server side.

Note

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

To configure anonymous call via phone user interface:

1. Press **Menu->Features->Anonymous Call**.
2. Press **◀** or **▶**, or the **Switch** soft key to select **Enabled** from the **Local Anonymous** field.



- (Optional.) Press **◀** or **▶** to select the desired value from the **Send Anony Code** field.

The phone will send the configured on code or off code depending on your selection when you enable or disable anonymous call feature on the phone.

- (Optional.) Enter the anonymous call on code in the **On Code** field.
- (Optional.) Enter the anonymous call off code in the **Off Code** field.
- 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.

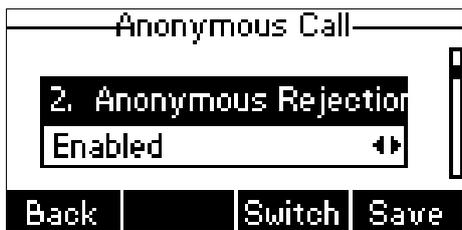


Anonymous Call Rejection

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. You can also configure the phone to send anonymous call rejection on/off code to the server to activate/deactivate anonymous call rejection on the server side.

To configure anonymous call rejection via phone user interface:

- Press **Menu->Features->Anonymous Call**.
- Press **▲** or **▼** to scroll to the **Anonymous Rejection** field.
- Press **◀** or **▶** , or the **Switch** soft key to select **Enabled** from the **Anonymous Rejection** field.



- (Optional.) Press **◀** or **▶** , or the **Switch** soft key to select the desired value from the **Send Rejection Code** field.

The phone will send the configured reject on code or reject off code depending on your selection when you enable or disable anonymous call rejection feature on the phone.

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

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

Advanced Phone Features

This chapter provides operating instructions for the advanced features of the SIP-T19(P) E2 IP phone. Topics include:

- [Hot Desking](#)
- [Intercom](#)
- [Multicast Paging](#)
- [Music on Hold](#)
- [Shared Call Appearance \(SCA\)](#)
- [Bridged Line Appearance \(BLA\)](#)
- [Messages](#)

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

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.

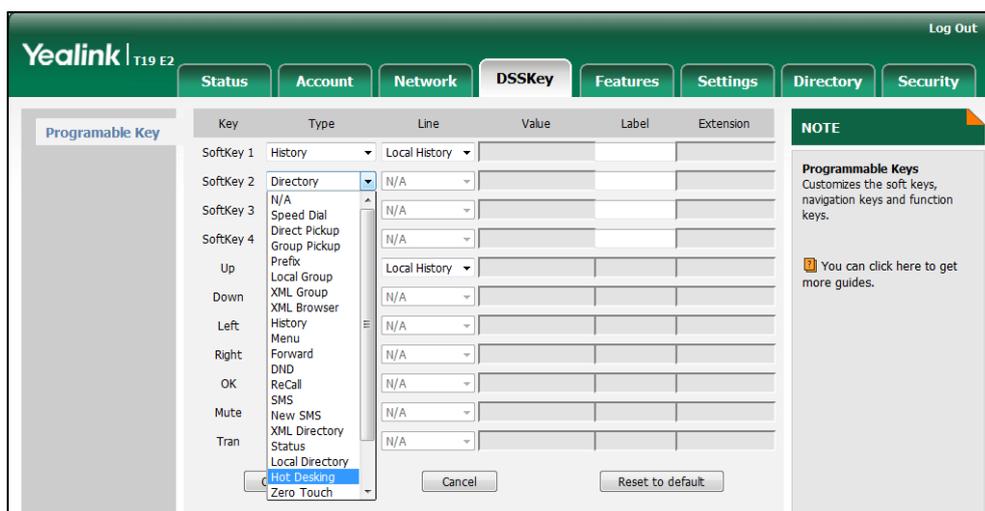
You can use hot desking on the SIP-T19(P) E2 IP phone to log out of the existing accounts and then log into a new account. As a result, many users can share the phone resource at different times. To use this feature, you need to configure a hot desking key in advance.

Note Hot desking is not available on all servers. Contact your system administrator for more information.

To configure a hot desking key via web user interface:

1. Click on **DSSKey->Programable Key**.
2. Select the desired programable key.

3. Select **Hot Desking** from the pull-down list of **Type**.



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

Note A hot desking key is configurable via web user interface only.

To use hot desking:

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

The LCD screen prompts the following warning:



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

The login wizard will be displayed as below:



3. Enter the login information in each field.
4. Press the **Save** soft key to login or the **Cancel** soft key to cancel.

Intercom

Intercom is a useful feature in an office environment to quickly connect with the operator or the secretary. The SIP-T19(P) E2 IP phone supports automatically to answer an incoming intercom call by default. The phone automatically plays a warning tone when it receives an incoming intercom call. In addition, you can enable the phone to mute the microphone when it automatically answers an incoming intercom call. You can also enable the phone to automatically answer an incoming intercom call while there is already an active call on the phone. The active call is then placed on hold.

Intercom features you need to know:

Intercom Feature	Description
Accept Intercom	Enable or disable the IP phone to 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.

Accept Intercom

You can enable or disable the phone to answer an incoming intercom call. If Accept Intercom is enabled, the phone will automatically answer an incoming intercom call. If Accept Intercom is disabled, the phone will reject incoming intercom calls and send a busy message to the caller. Accept Intercom is enabled by default.

Note

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

Intercom Mute

You can mute or un-mute the 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

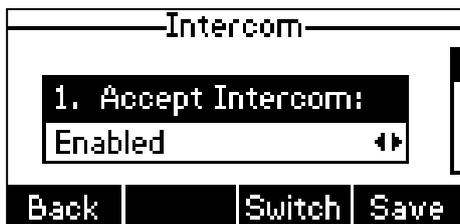
You can enable or disable the phone to play a warning tone when receiving an intercom call. If Intercom Tone is enabled, the phone will play a warning tone before answering the intercom call. If Intercom Tone is disabled, the phone will automatically answer the intercom call without warning. Intercom 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.

To configure intercom features via phone user interface:

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



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

These specific parameters are configurable via web user interface at the path **Features->Intercom**.

Multicast Paging

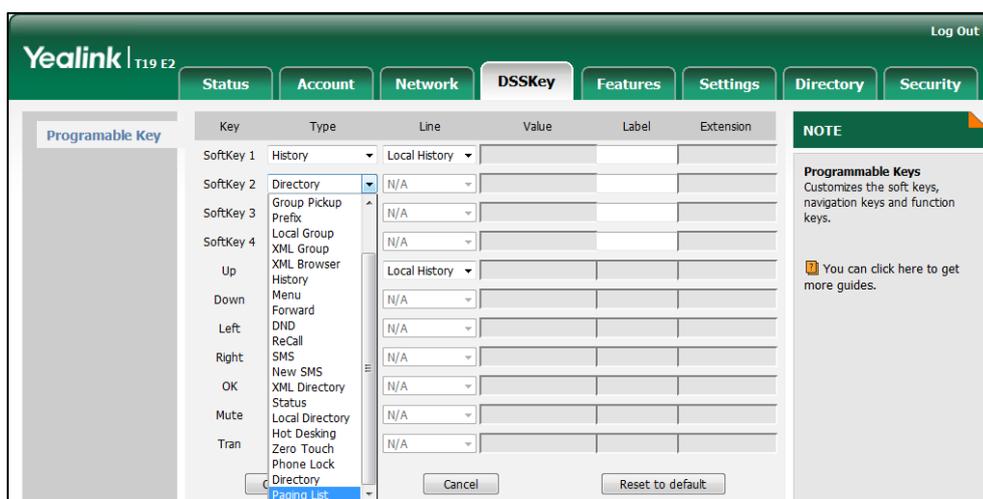
You can use multicast paging to quickly and easily broadcast time sensitive announcements to users who are listening to a specific multicast group. You can configure a multicast paging key or the paging list key on the phone, which allows you to send a Real Time Transport Protocol (RTP) stream to the 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 paging list key via web user interface:

1. Click on **DSSKey->Programable Key**.
2. Select the desired programable key.

3. Select **Paging List** from the pull-down list of **Type**.



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

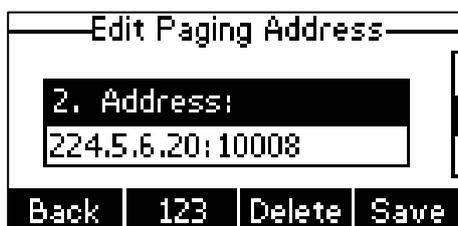
To configure paging list via phone user interface:

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 and then press the **Edit** soft key.
4. 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.



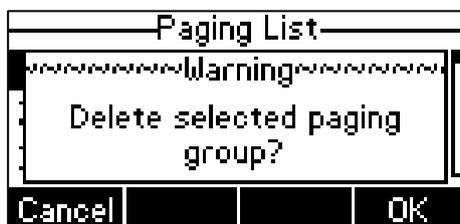
5. Enter the group name in the **Label** field.
6. Press the **Save** soft key to accept the change.
7. Repeat the step 2-6, you can add more paging groups.

Paging list is configurable via web user interface at the path: **Directory->Multicast IP**.

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 and then press **Delete** soft key.

The LCD screen prompts “Delete selected paging group?”.



4. Press the **OK** soft key to accept the change or the **Cancel** soft key to cancel.

If you want to delete all paging groups, you can press the **Del All** soft key.

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 the active call, 1 with the highest priority, 10 with 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 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.
5. Enter the label in the **Label** field.

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

IP Address	Listening Address	Label	Priority
1 IP Address	224.5.6.20:10008	Sale	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

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

Note

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

Multicast listening addresses are configurable via web user interface only.

Using Multicast Paging

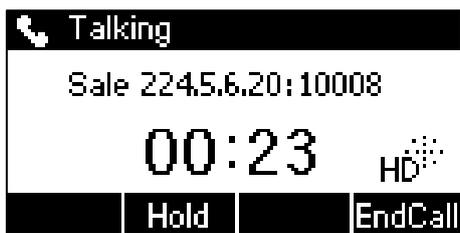
To send RTP stream via a paging list key when the receiver's phone is idle:

1. Press the paging list key when the phone is idle.
2. Press  or  to select the desired paging group.
3. Press  or the **Paging** soft key to send RTP.

The phone sends RTP to a preconfigured multicast address (IP: Port).

Both the sender's and receiver's phones play a warning tone and the receiver automatically answers the multicast RTP session in the speakerphone mode.

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



4. To place the current multicast RTP session on hold, press the **Hold** soft key.
The sender's phone places the multicast RTP session on hold and receiver's phone releases the session.
5. To resume the held multicast RTP session, press the **Resume** soft key.
The multicast RTP session is established again.
6. To end the multicast RTP session, press the **EndCall** soft key.

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.

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->Advanced**.

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

The screenshot shows the Yealink T19 E2 web interface. The 'Account' tab is selected. The 'Music Server URI' field is highlighted with a red box and contains the value 'sip:moh@sp.com'. The interface includes a navigation menu on the left with options like Register, Basic, Codec, and Advanced. The main content area has various configuration fields for Keep Alive Type, RPort, DTMF Type, etc. A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

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

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

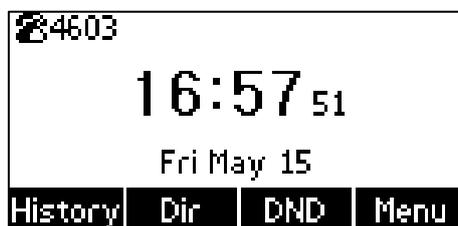
Note

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

Shared Call Appearance (SCA)

You can use SCA feature to share an extension which can be registered on two or more IP phones at the same time. The shared line is indicated by a different line icon.

In the following figure, the registered line is shared:

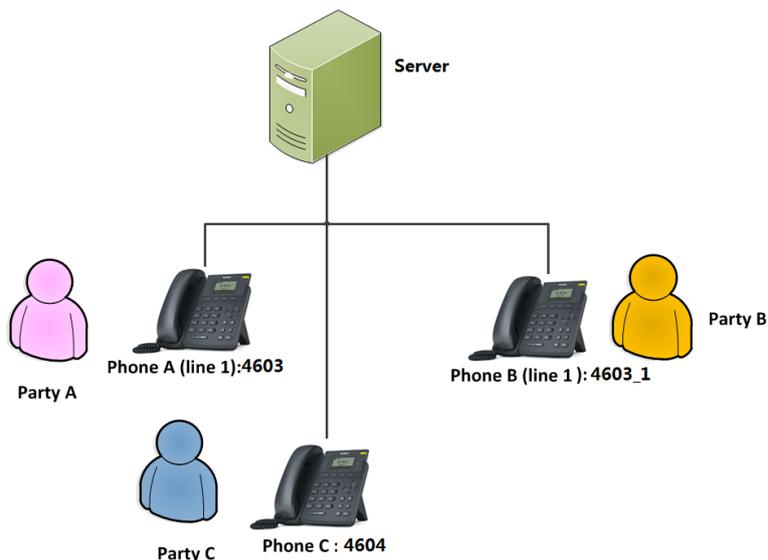


If two phones share a line, an incoming call to this extension will cause both phones to ring simultaneously. The incoming call can be answered on either phone but not both.

This feature is very useful in the boss and secretary scenario. For example, the secretary can share the boss's extension on her phone. When there is an incoming call to the extension of the boss, both the phones of the boss and the secretary will ring simultaneously. Either the boss or the secretary can answer the call.

Configuring SCA Feature on the IP Phone

You can configure a primary account on the IP phone and other alternate accounts on the other IP phones. For example, party A, party B share the account 4603, phone A registers the primary account 4603, phone B registers the alternate account 4603_1, phone C registers the account 4604.



To configure the shared line settings on phone A via web user interface:

1. Register the account 4603.

The screenshot shows the Yealink T19 E2 web user interface. The 'Account' tab is selected. The 'Register Status' section is expanded, showing the following configuration:

Register Status	Registered
Line Active	Enabled
Label	4603
Display Name	4603
Register Name	4603
User Name	4603
Password	*****
SIP Server 1	
Server Host	pbx.yealink.com Port: 5060
Transport	UDP
Server Expires	3600
Server Retry Counts	3
SIP Server 2	
Server Host	Port: 5060
Transport	UDP
Server Expires	3600
Server Retry Counts	3
Outbound Proxy Server	
Enable Outbound Proxy Server	Enabled
Outbound Proxy Server 1	10.1.8.11 Port: 5060
Outbound Proxy Server 2	Port: 5060
Proxy Falback Interval	3600
NAT	Disabled

The 'NOTE' section on the right contains the following information:

- Account Registration:** Registers account(s) for the IP phone.
- Server Redundancy:** It is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.
- NAT Traversal:** A general term for techniques that establish and maintain IP connections traversing NAT gateways. STUN is one of the NAT traversal techniques.

At the bottom of the page, there are 'Confirm' and 'Cancel' buttons.

- Click on **Advanced**, and then select **Shared Call Appearance** from the pull-down list of **Shared Line**.

The screenshot shows the Yealink T19 E2 web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Shared Line' dropdown menu is set to 'Shared Call Appearance', which is highlighted with a red box. Other settings visible include: Keep Alive Type (Default), Keep Alive Interval (30), RPort (Disabled), Subscribe Period (1800), DTMF Type (RFC2833), PTime (20), Call Pull Feature Access Code, Group Call Pickup Code, Distinctive Ring Tones (Enabled), Unregister When Reboot (Disabled), VQ RTPC-XR Collector name, VQ RTPC-XR Collector address, and VQ RTPC-XR Collector port (5060). A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, and Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA).

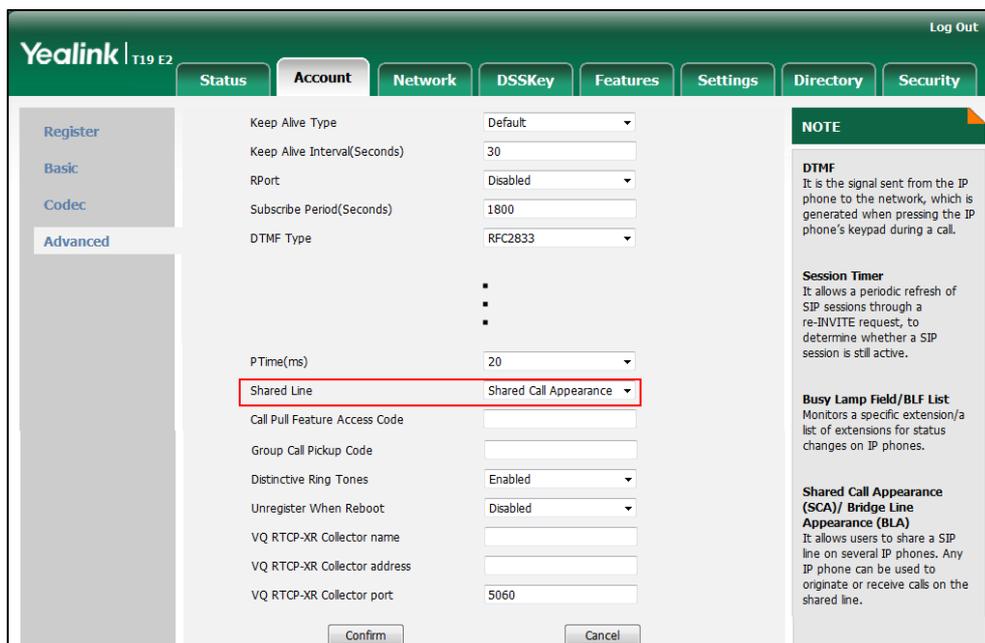
- Click **Confirm** to accept the change.

To configure the shared line settings on phone B via web user interface:

- Register the alternate account 4603_1.
(Enter the primary account 4603 in the **Register Name** field.)

The screenshot shows the Yealink T19 E2 web interface. The 'Account' tab is active, and the 'Register Status' is 'Registered'. The 'SIP Server 1' and 'SIP Server 2' sections are highlighted with red boxes. The 'SIP Server 1' section includes fields for: Line Active (Enabled), Label (4603_1), Display Name (4603_1), Register Name (4603), User Name (4603_1), Password (*****), Server Host (pbx.yealink.com), Port (5060), Transport (UDP), Server Expires (3600), and Server Retry Counts (3). The 'SIP Server 2' section includes fields for: Server Host, Port (5060), Transport (UDP), Server Expires (3600), and Server Retry Counts (3). The 'Enable Outbound Proxy Server' section is also highlighted with a red box, showing fields for: Enable Outbound Proxy Server (Enabled), Outbound Proxy Server 1 (10.1.8.11), Port (5060), Outbound Proxy Server 2, Port (5060), Proxy Fallback Interval (3600), and NAT (Disabled). A 'NOTE' section on the right provides information about Account Registration, Server Redundancy, and NAT Traversal.

- Click on **Advanced**, and then select **Shared Call Appearance** from the pull-down list of **Shared Line**.



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

Configuring call pull feature

Call pull feature allows users to retrieve an existing call from another shared location that is in active or public hold status.

To configure the call pull feature access code via web user interface:

- Click on **Account->Advanced**.

- Enter the call pull feature access code (e.g., *11) in the **Call Pull Feature Access Code** field.

The screenshot shows the Yealink T19 E2 web interface. The 'Account' tab is selected. The 'Call Pull Feature Access Code' field is highlighted with a red box and contains the value '*11'. Other settings include Keep Alive Type (Default), Keep Alive Interval (30), RPort (Disabled), Subscribe Period (1800), DTMF Type (RFC2833), PTime (20), Shared Line (Shared Call Appearance), Group Call Pickup Code, Distinctive Ring Tones (Enabled), Unregister When Reboot (Disabled), VQ RTPC-XR Collector name, VQ RTPC-XR Collector address, and VQ RTPC-XR Collector port (5060). A 'NOTE' section on the right explains DTMF, Session Timer, Busy Lamp Field/BLF List, and Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA).

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

The phone will dial out “*11” automatically when you press the **CallPull** soft key.

Using SCA Feature on the IP Phone

This section provides you with detailed information on using the SIP-T19(P) E2 IP phone in a SCA scenario.

You can do the following using SIP-T19(P) E2 IP phone in a SCA scenario:

- Placing calls
- Answering calls
- Place a call on hold
- Retrieving a held call
- Call Pull

Placing Calls

You can have one call or multiple calls on the shared line.

To place a call on the shared line:

- Enter the desired number using the keypad when the phone is idle.
- Press , , or the **Send** soft key.

To place multiple calls on the shared line:

You can have more than one call on the shared line. To place a new call when there is an active call on phone A, do one of the following on phone A:

1. Press the **Hold** soft key. The original call is placed on hold.
2. Press the **NewCall** soft key to enter the pre-dialing screen.
3. Enter the desired number using the keypad.
4. Press  ,  , or the **Send** soft key.
Phone A will dial the entered number.

Answering Calls

You can have one call or multiple calls on the shared line. Incoming calls will be distributed evenly among the available shared line.

To answer a call on the shared line:

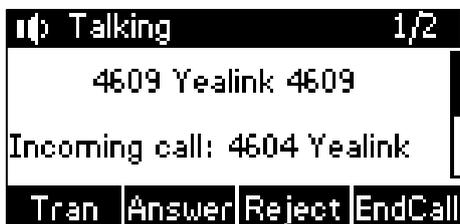
When an incoming call arrives on the shared line, the phone A and phone B will ring simultaneously. You can answer the incoming call on either phone A or phone B but not both.

Do one of the following on phone A or phone B:

- Press  ,  or the **Answer** soft key on phone A.
Phone B stops ringing.
- Press  ,  or the **Answer** soft key on phone B.
Phone A stops ringing.

To answer multiple calls on the shared line:

An incoming call arrives on the shared line when there is an active call on phone A. You can answer the incoming call on either phone A or phone B. The LCD screen of phone A displays the information of the incoming call (e.g., "Incoming call: 4604 Yealink").



Note Make sure call waiting feature is enabled on phone A. For more information, refer to [Call Waiting](#) on page 79.

Do one of the following on phone A:

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

You can also answer the call on phone B:

1. Press  or the **Answer** soft key. Phone A stop ringing.

Placing a Call on Hold

To place a call on hold:

1. Press the **Hold** soft key on phone A when party A and party C are talking.
The shared line call is placed on hold.

Retrieving a Held Call

If there is a held call between phone A and phone C, you can retrieve a held call on phone A.

To retrieve the held call on phone A:

1. Press the **Resume** soft key on phone A.
The conversation between phone A and phone C is retrieved.

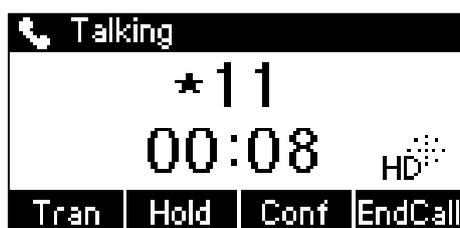
Call Pull

Call pull feature allows users to retrieve an existing call from another shared phone that is in active or hold status. For example, when there is a call between phone A and phone C, you can use call pull feature on phone B to retrieve this call from phone A. Then the call is established between phone B and phone C.

To retrieve a call from another shared phone:

If there is an active call between phone A and phone C, do the following:

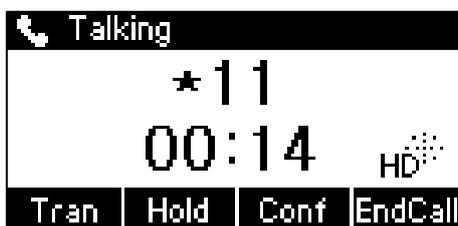
1. Enter the call pull feature access code (e.g., *11), and then press the **Send** soft key on the phone B.



The active call has been retrieved from the phone A successfully.

If there is a held call between phone A and phone C, do the following:

1. Enter the call pull feature access code (e.g., *11), and then press the **Send** soft key on the phone B.



The held call has been retrieved from the phone A successfully.

Bridged Line Appearance (BLA)

BLA allows users to share a SIP line on two or more IP phones. Users can monitor the specific extension (BLA number) for status changes on each IP phone. To use this feature, a BLA group should be pre-configured on the server and one of them is specified as a BLA number. BLA depends on support from a SIP server.

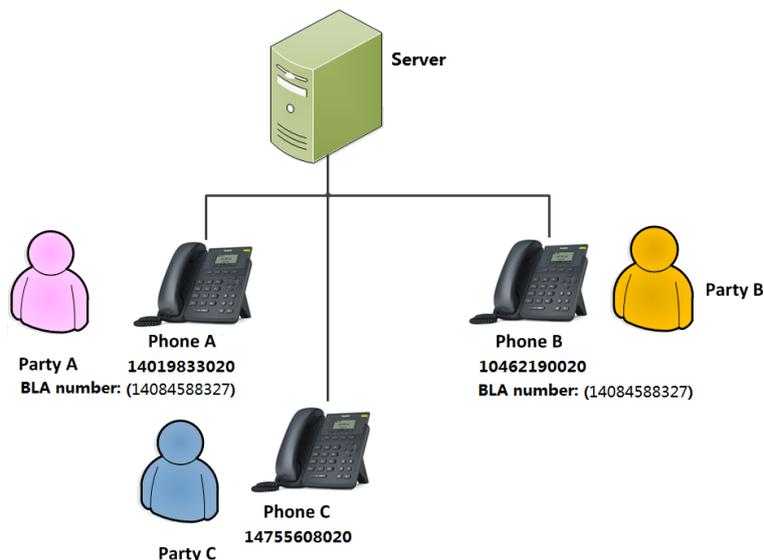
In the following figure, the registered account is shared:



Any IP phone can be used to originate or receive calls on the bridged line. An incoming call to the BLA number can be presented to multiple phones in the group simultaneously. The incoming call can be answered on any IP phone of the group but not all.

Configuring BLA Feature on the IP Phone

You can share a BLA number on two or more phones. For example, phone A registers the account 14019833020 and assigns BLA number, phone B registers the account 10462190020 and assigns BLA number, phone C registers the account 14755608020. Phone A and phone B share the BLA number 14084588327.



To register account and configure BLA feature on phone A via web user interface:

1. Register the account 14019833020.

The screenshot shows the Yealink T19 E2 web user interface. The 'Account' tab is selected, and the 'Register' section is active. The 'SIP Server 1' section is highlighted with a red box, showing fields for Server Host, Transport, Server Expires, and Server Retry Counts. The 'Enable Outbound Proxy Server' section is also highlighted with a red box, showing fields for Outbound Proxy Server 1 and Outbound Proxy Server 2. A 'NOTE' section on the right provides information about Account Registration, Server Redundancy, and NAT Traversal.

Field	Value
Line Active	Enabled
Label	14019833020
Display Name	14019833020
Register Name	14019833020
User Name	14084588327
Password	*****
SIP Server 1	
Server Host	sp.ringcentral.com Port: 5060
Transport	UDP
Server Expires	3600
Server Retry Counts	3
SIP Server 2	
Server Host	Port: 5060
Transport	UDP
Server Expires	3600
Server Retry Counts	3
Enable Outbound Proxy Server	
Outbound Proxy Server 1	sp114.ringcentral.com Port: 5099
Outbound Proxy Server 2	Port: 5060
Proxy Fallback Interval	3600
NAT	Disabled

NOTE

Account Registration
Registers account(s) for the IP phone.

Server Redundancy
It is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server falls, or the connection between the IP phone and the server fails.

NAT Traversal
A general term for techniques that establish and maintain IP connections traversing NAT gateways. STUN is one of the NAT traversal techniques.

You can configure NAT traversal for this account.

You can click here to get more guides.

2. Click on **Advanced**, and then select **Draft BLA** from the pull-down list of **Shared Line**.
3. Enter the desired number in the **BLA Number** field.

The screenshot shows the Yealink T19 E2 web interface. The 'Advanced' tab is selected in the sidebar. The 'Account' section is active, showing various settings. The 'Shared Line' dropdown menu is set to 'Draft BLA', and the 'BLA Number' text field contains the number '14084588327'. A red rectangular box highlights these two fields. Other settings include 'Keep Alive Type' (Default), 'Keep Alive Interval(Seconds)' (30), 'RPort' (Disabled), 'Subscribe Period(Seconds)' (1800), 'DTMF Type' (RFC2833), 'PTime(ms)' (20), 'BLA Subscription Period' (300), 'SIP Send MAC' (Disabled), 'SIP Send Line' (Disabled), 'Unregister When Reboot' (Disabled), and 'VQ RTPC-XR Collector port' (5060). The 'Confirm' and 'Cancel' buttons are at the bottom. A 'NOTE' section on the right provides details about DTMF, Session Timer, Busy Lamp Field/BLF List, and Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA).

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

To register account and configure BLA feature on phone B via web user interface:

1. Register the account 10462190020.

Yealink T19 E2

Log Out

Status Account Network DSSKey Features Settings Directory Security

Register

Register Status Registered

Line Active Enabled

Label 10462190020

Display Name 10462190020

Register Name 10462190020

User Name 14084588327

Password *****

SIP Server 1

Server Host sp.ringcentral.com Port: 5060

Transport UDP

Server Expires 3600

Server Retry Counts 3

SIP Server 2

Server Host Port: 5060

Transport UDP

Server Expires 3600

Server Retry Counts 3

Enable Outbound Proxy Server Enabled

Outbound Proxy Server 1 sp214.ringcentral.com Port: 5099

Outbound Proxy Server 2 Port: 5060

Proxy Fallback Interval 3600

NAT Disabled

Confirm Cancel

NOTE

Account Registration
Registers account(s) for the IP phone.

Server Redundancy
It is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.

NAT Traversal
A general term for techniques that establish and maintain IP connections traversing NAT gateways. STUN is one of the NAT traversal techniques.

You can configure NAT traversal for this account.

You can click here to get more guides.

2. Click on **Advanced**, and then select **Draft BLA** from the pull-down list of **Shared Line**.
3. Enter the desired number in the **BLA Number** field.

Yealink T19 E2

Log Out

Status Account Network DSSKey Features Settings Directory Security

Register

Basic

Codec

Advanced

Keep Alive Type Default

Keep Alive Interval(Seconds) 30

RPort Disabled

Subscribe Period(Seconds) 1800

DTMF Type RFC2833

PTime(ms) 20

Shared Line Draft BLA

BLA Number 14084588327

BLA Subscription Period 300

SIP Send MAC Disabled

SIP Send Line Disabled

Unregister When Reboot Disabled

VQ RTPC-XR Collector name

VQ RTPC-XR Collector address

VQ RTPC-XR Collector port 5060

Confirm Cancel

NOTE

DTMF
It is the signal sent from the IP phone to the network, which is generated when pressing the IP phone's keypad during a call.

Session Timer
It allows a periodic refresh of SIP sessions through a re-INVITE request, to determine whether a SIP session is still active.

Busy Lamp Field/BLF List
Monitors a specific extension/a list of extensions for status changes on IP phones.

Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA)
It allows users to share a SIP line on several IP phones. Any IP phone can be used to originate or receive calls on the shared line.

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

Using BLA Feature on the IP Phone

This section provides you with detailed information on using the SIP-T19(P) E2 IP phone in a BLA scenario.

You can do the following using SIP-T19(P) E2 IP phone in a BLA scenario:

- Placing calls
- Answering calls
- Place a call on hold
- Retrieving a held call

Placing Calls

You can have one call or multiple calls on the shared line.

To place a call on the shared line:

1. Enter the desired number using the keypad when the phone is idle.
2. Press  ,  , or the **Send** soft key.

To place multiple calls on the shared line:

You can have more than one call on the shared line. To place a new call when there is an active call on phone A, do one of the following on phone A:

1. Press the **Hold** soft key. The original call is placed on hold.
2. Press the **NewCall** soft key to enter the pre-dialing screen.
3. Enter the desired number using the keypad.
4. Press  ,  , or the **Send** soft key.

Phone A will dial the entered number.

Answering Calls

When the phone C dials the BLA number "14084588327", an incoming call will arrive on the bridged line. The phone A and phone B ring simultaneously. You can answer the incoming call on either phone A or phone B but not both.

Do one of the following on phone A or phone B:

- Press  ,  or the **Answer** soft key on phone A.
Phone B stops ringing.
- Press  ,  or the **Answer** soft key on phone B
Phone A stops ringing.

Placing a Call on Hold

To place a call on hold:

2. Press the **Hold** soft key on phone A when party A and party C are talking.
The bridged line call is placed on hold.

Retrieving a Held Call

If there is a held call between phone A and phone C, you can retrieve a held call on phone A.

To retrieve the held call on phone A:

2. Press the **Resume** soft key on phone A.
The conversation between phone A and phone C is retrieved.

Messages

Short Message Service (SMS)

You can send and receive text messages using the SIP-T19(P) E2 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 power indicator LED will slow flash green. The LCD screen will prompt receiving new text messages with the number of waiting messages (e.g., 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. Enter the number you want to send the message to in the **To** field.
5. 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.

- Compose the new text message. You can press the **abc** soft key to change the input mode.



- Press the **Send** soft key after completing the content.
- Check the **From** and **To** fields, and then press the **Send** soft key.

To delete a text message:

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



- Select **Delete** to delete the desired message, then press the **OK** soft key. The LCD screen prompts "Delete message?".



- 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.

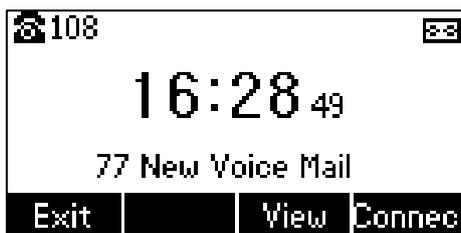
Note

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-T19(P) E2 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 power indicator LED will slow flash green. The LCD screen will prompt that the phone receives a new message with the

number of waiting voice mails (e.g., 77 New Voice Mail) and display an icon.



If the voice mail pop-up message box disappears, it won't pop up again unless the user receives a new voice mail or the user re-registers the account that has unread voice mail(s).

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 the navigation keys to highlight the account which you want to set.
3. Press the **123** soft key to select the proper input mode and then enter the voice mail access code (e.g., *97).



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

Note

Voice mail access 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 and the power indicator LED slow flashes green, you can press  or the **Connect** soft key to dial out the voice mail access code.

- Follow the voice prompt to listen to your voice mails.

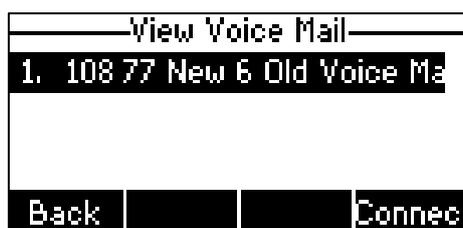
Note

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

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

To view the voice mail via phone user interface:

- Press **Menu->Message->Voice Mail->View Voice Mail**.
- The LCD screen displays the amount of new and old voice mails.



- Press the **Connect** soft key to listen to voice mails.

Message Waiting Indicator (MWI)

The SIP-T19(P) E2 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, an indicator message (including a voice mail icon) on the LCD screen, and the power indicator LED slow flashes 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-T19(P) E2 IP phone only handles the MWI messages sent from the server. But for other servers, the MWI service is solicited, so the SIP-T19(P) E2 IP phone must enable subscription for MWI.

Note

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

The MWI subscription parameters you need to know:

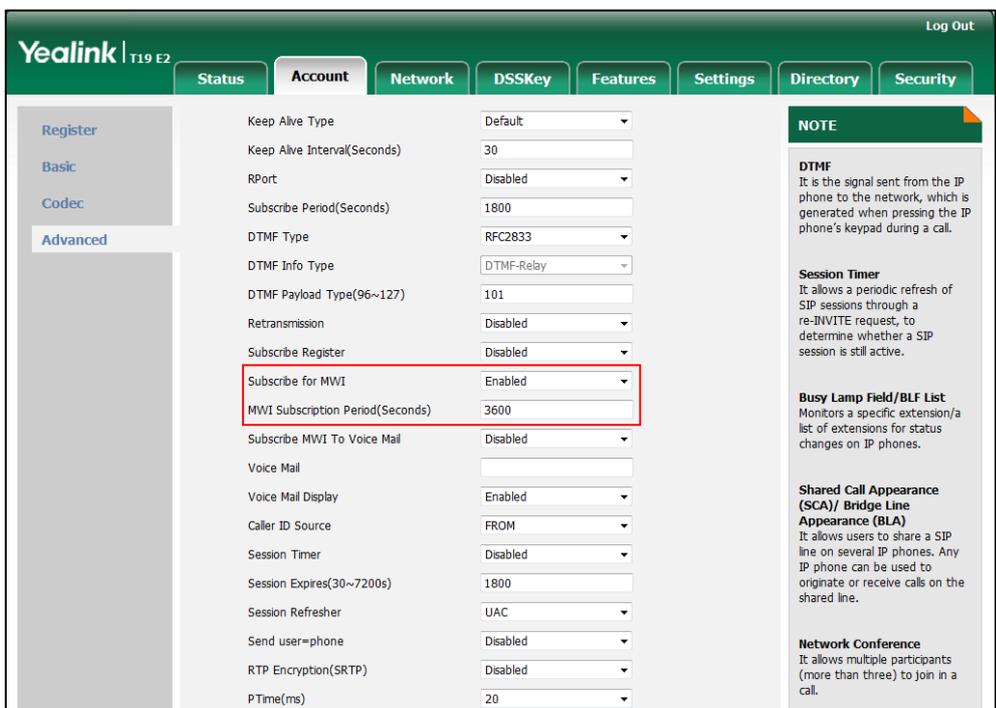
Options	Description
Subscribe for MWI	Enable or disable a subscription for MWI service.
MWI Subscription Period	Period of MWI subscription. The IP phone sends a refresh SUBSCRIBE request before initial SUBSCRIBE expiration.
Subscribe MWI to Voice Mail	Enable or disable a subscription to the voice mail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

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

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->Advanced**.
2. Select **Enabled** from the pull-down list of **Subscribe for MWI** field.
3. Enter the period time in the **MWI Subscription Period (Seconds)** field.



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

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

To enable subscribe MWI to voice mail via web user interface:

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

- Enter the desired voice mail number in the **Voice Mail** field.

The screenshot shows the Yealink T19 E2 web interface with the 'Account' tab selected. The 'Advanced' sub-tab is active, displaying various settings. The following settings are highlighted with red boxes:

- Subscribe for MWI**: Enabled
- Subscribe MWI To Voice Mail**: Enabled
- Voice Mail**: *88

Other visible settings include: Keep Alive Type (Default), Keep Alive Interval (30), RPort (Disabled), Subscribe Period (1800), DTMF Type (RFC2833), DTMF Info Type (DTMF-Relay), DTMF Payload Type (101), Retransmission (Disabled), Subscribe Register (Disabled), MWI Subscription Period (3600), Voice Mail Display (Enabled), Caller ID Source (FROM), Session Timer (Disabled), Session Expires (1800), Session Refresher (UAC), Send user=phone (Disabled), RTP Encryption (SRTP) (Disabled), and PTime (20).

A 'NOTE' section on the right provides information about DTMF, Session Timer, Busy Lamp Field/BLF List, Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA), and Network Conference.

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

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

Note

MWI subscription is configurable via web user interface only.

Troubleshooting

This chapter provides general troubleshooting information to help you solve the problems you might encounter when using your SIP-T19(P) E2 IP phone.

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

Why is the LCD screen blank?

- Ensure that the phone is properly plugged into a functional AC outlet.
- Ensure that the phone is plugged into a socket controlled by a switch that is on.
- If the phone is plugged into a power strip, try to plug it directly into a wall outlet instead.
- If your SIP-T19P E2 IP phone is powered from PoE, ensure you use a PoE-compliant switch or hub.

Why does the phone display “Network unavailable”?

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

Why does the phone display “No Service”?

The LCD screen displays “No Service” 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 can I find the basic information of the IP phone?

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

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

Three ways to obtain the MAC address of a phone:

- You can ask your supplier for 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) and Hands-Free Speakerphone to check whether the dial tone is present for one of the audio modes.

If the dial tone exists on another audio mode, connect a different handset or headset to isolate the problem.

Why doesn't the phone ring?

Check the ringer volume on the phone. To adjust the ringer volume setting, press the **Volume** key when the phone is on-hook and idle. For more information, refer to [Volume](#) on page 26.

Why can't I receive calls?

- Check the SIP registration with your system administrator.
- Check that DND (Do Not Disturb) mode is deactivated on your phone. Refer to [Do Not Disturb \(DND\)](#) on page 73.
- Check that call forward is disabled on the phone. Refer to [Call Forward](#) on page 74.
- 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 configuration may override the local configuration.

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. Call hold tone is enabled by default. Call hold tone and the interval of playing a hold tone are configurable via web user interface only.

To configure call hold tone and call hold tone delay via web user interface:

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

The screenshot shows the Yealink T19 E2 web user interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. The configuration table is as follows:

Field	Value
Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Play Local DTMF Tone	Enabled
DTMF Repetition	3
Play Hold Tone	Enabled
Play Hold Tone Delay	30
Allow Mute	Enabled
Voice Mail Tone	Enabled
DHCP Hostname	SIP-T19P_E2
Reboot in Talking	Disabled
Hide Feature Access Codes	Disabled
Display Method on Dialing	User Name

The 'NOTE' section on the right contains the following information:

- Call Waiting:** It allows IP phones to receive a new incoming call when there is already an active call.
- Auto Redial:** It allows IP phones to automatically redial a busy number after the first attempt.
- Key As Send:** Assigns "*" or "***" as the send key.
- Hotline:** IP phone will automatically dial out the hotline number when lifting the handset, pressing the speakerphone key or the line key.
- Call Completion:** It allows users to monitor the busy party and establish a call when the busy party becomes available to receive a call.

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

Why can't I send an 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.

The screenshot shows the 'Security' tab in the Yealink T19 E2 web interface. The 'Password' sub-tab is active. The form contains the following elements:

- User Type:** A dropdown menu with 'user' selected.
- Old Password:** A text input field with masked characters (dots).
- New Password:** A text input field with masked characters (dots).
- Confirm Password:** A text input field with masked characters (dots).
- Buttons:** 'Confirm' and 'Cancel' buttons at the bottom.
- NOTE sidebar:** A green-bordered box on the right with the text: 'User Password/ Administrator Password. When logging into the web user interface, you need to enter the user name and password. You can change the user/ administrator password for security reasons. You can click here to get more guides.'

4. Click **Confirm** to accept the change.
You can also contact your system administrator for help.

Note

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

User password is configurable via web user interface only.

How to make a call using SRTP?

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

To enable SRTP on a per-line basis via web user interface:

1. Click on **Account->Advanced**.

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

The screenshot shows the Yealink T19 E2 web interface. The 'Account' tab is selected. The 'RTP Encryption(SRTP)' option is highlighted with a red box and set to 'Optional'. Other settings include Keep Alive Type (Default), Keep Alive Interval (30), RPort (Disabled), Subscribe Period (1800), DTMF Type (RFC2833), DTMF Info Type (DTMF-Relay), DTMF Payload Type (101), Retransmission (Disabled), Subscribe Register (Disabled), Subscribe for MWI (Disabled), MWI Subscription Period (3600), Subscribe MWI To Voice Mail (Disabled), Voice Mail, Voice Mail Display (Enabled), Caller ID Source (FROM), Session Timer (Disabled), Session Expires (1800), Session Refresher (UAC), Send user=phone (Disabled), and PTime (20).

- 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 IP phone.

The screenshot shows the Yealink T19 E2 web interface. The 'Settings' tab is selected. The 'Reboot' button is highlighted with a red box. Other settings include Version (Firmware Version: 53.80.0.50, Hardware Version: 53.0.0.128.0.0.0), Reset to Factory Setting (Reset to Factory Setting button), Select and Upgrade Firmware (Browse... button, No file selected, Upgrade button), and a NOTE section with links for Reset to Factory Setting, Reboot, and Upgrading Firmware.

Note

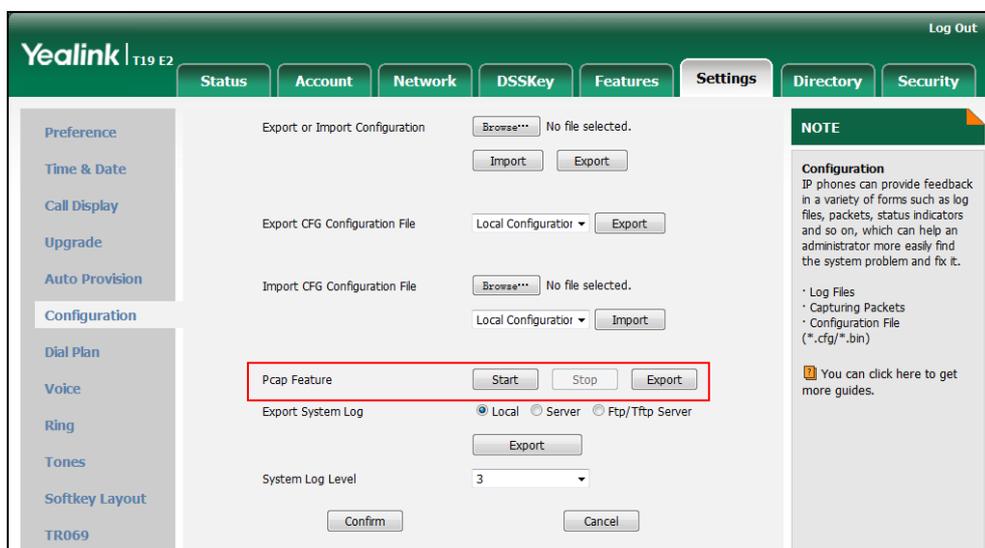
Any reboot of the IP phone may take a few minutes.

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 capturing signal traffic.
3. Recreate the error to be documented in the trace.
4. Click **Stop** to stop the capture.
5. Click **Export** to open file download window, and then save the file to your local system.



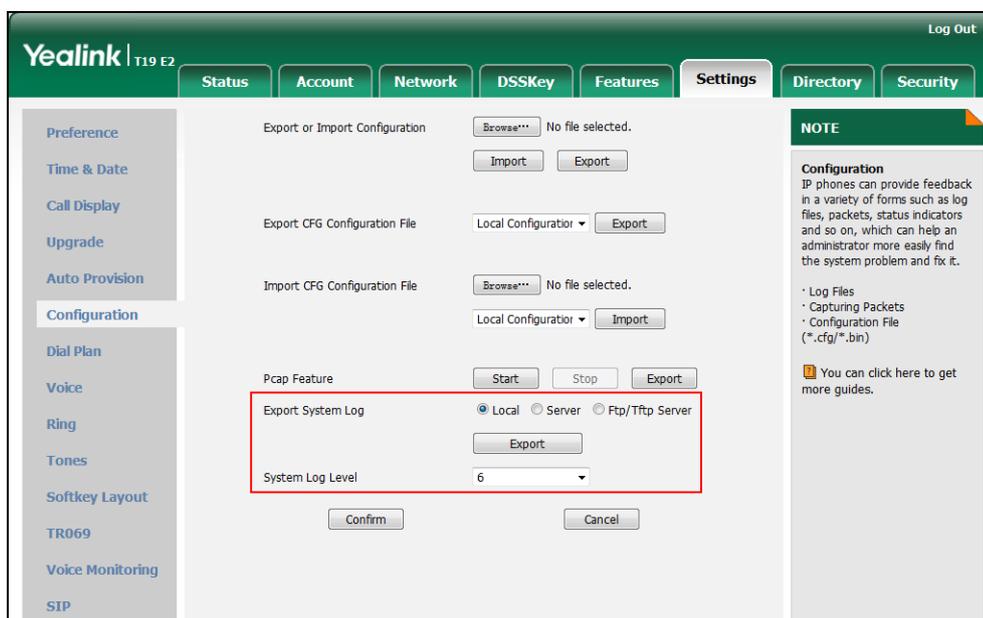
How to export system log?

We may need you to provide 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**.
The default system log level is 3.
3. Click **Confirm** to accept the change.

4. Mark the **Local** radio box in the **Export System Log** field.
5. Click **Export** to open file download window, and then save the file to your local system.



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

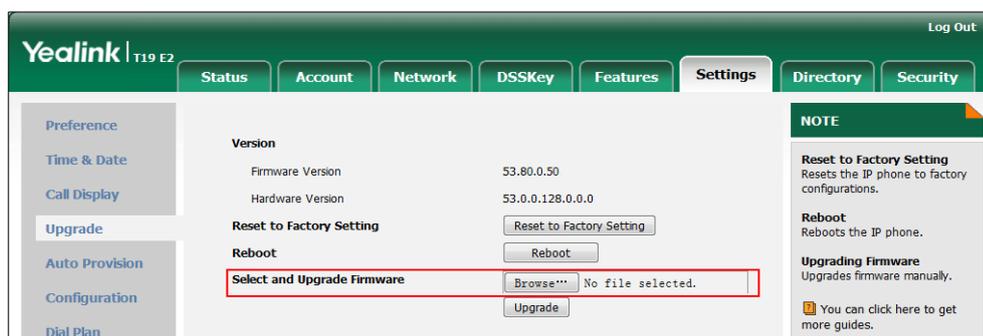
Note

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

How to 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 customized settings will be overwritten after reset.

To reset the phone via phone user interface:

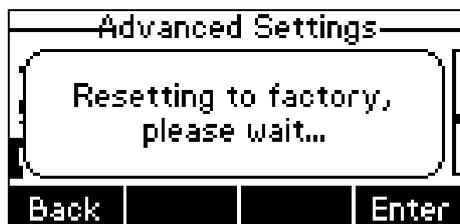
1. Press **Menu->Settings->Advanced Settings** (default password: admin) -> **Reset to Factory**.
2. Press the **Enter** soft key.

The LCD screen prompts the following warning:

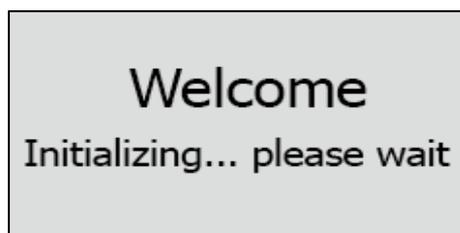


3. Press the **OK** soft key.

The LCD screen prompts "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 risk of fire, electrical shock, and other personal injury.

General Requirements

- Before you install and use the device, read the safety instructions carefully and observe the situation during operation.
- During the process of storage, transportation, and operation, please always keep the device dry and clean.
- During the process of storage, transportation, and operation, please avoid collision and crash of the device.
- Please attempt not to dismantle the device by yourself. In case of any discrepancy, please contact the appointed maintenance center for repair.
- Without prior written consent, no organization or individual is permitted to make any change to the structure or the safety design of the device. Yealink is under no circumstance liable to consequences or legal issues caused by such changes.
- Please refer to the relevant laws and statutes while using the device. Legal rights of others should be respected as well.

Environmental Requirements

- Place the device at a well-ventilated place. Do not expose the device under direct sunlight.
- Keep the device dry and free of dusts.
- Place the device on a stable and level platform.

- Please place no heavy objects on the device in case of damage and deformation caused by the heavy load.
- Keep at least 10 cm between the device and the closest object for heat dissipation.
- Do not place the device on or near any inflammable or fire-vulnerable object, such as rubber-made materials.
- Keep the device away from any heat source or bare fire, such as a candle or an electric heater.
- Keep the device away from any household appliance with strong magnetic field or electromagnetic field, such as a microwave oven or a refrigerator.

Operating Requirements

- Do not let a child operate the device without guidance.
- Do not let a child play with the device or any accessory in case of accidental swallowing.
- Please use the accessories provided or authorized by the manufacturer only.
- The power supply of the device shall meet the requirements of the input voltage of the device. Please use the provided surge protection power socket only.
- Before plugging or unplugging any cable, make sure that your hands are completely dry.
- Do not spill liquid of any kind on the product or use the equipment near water, for example, near a bathtub, washbowl, kitchen sink, wet basement or near a swimming pool.
- Do not tread on, pull, or over-bend any cable in case of malfunction of the device.
- During a thunderstorm, stop using the device and disconnect it from the power supply. Unplug the power plug and the Asymmetric Digital Subscriber Line (ADSL) twisted pair (the radio frequency cable) to avoid lightning strike.
- If the device is left unused for a rather long time, disconnect it from the power supply and unplug the power plug.
- When there is smoke emitted from the device, or some abnormal noise or smell, disconnect the device from the power supply, and unplug the power plug immediately. Contact the specified maintenance center for repair.
- Do not insert any object into equipment slots that is not part of the product or auxiliary product.
- Before connecting a cable, connect the grounding cable of the device first. Do not disconnect the grounding cable until you disconnect all other cables.

Cleaning Requirements

- Before cleaning the device, stop using it and disconnect it from the power supply.
- Use a piece of soft, dry and anti-static cloth to clean the device.
- Keep the power plug clean and dry. Using a dirty or wet power plug may lead to electric shock or other perils.

Appendix A - Time Zones

Time Zone	Time Zone Name
-11	Samoa
-10	US-Hawaii-Aleutian, US-Alaska-Aleutian
-9:30	French Polynesia
-9	US-Alaska Time
-8	Canada(Vancouver,Whitehorse), Mexico(Tijuana,Mexicali), US-Pacific Time
-7	Canada(Edmonton,Calgary), Mexico(Mazatlan,Chihuahua), US-MST no DST, US-Mountain Time
-6	Canada-Manitoba(Winnipeg), Chile(Easter Islands), Mexico(Mexico City,Acapulco), US-Central Time
-5	Bahamas(Nassau), Canada(Montreal,Ottawa,Quebec), Cuba(Havana), US-Eastern Time
-4:30	Venezuela(Caracas)
-4	Canada(Halifax,Saint John), Chile(Santiago), Paraguay(Asuncion), UK(Falkland Islands), UK-Bermuda(Bermuda), Trinidad&Tobago
-3:30	Canada-New Foundland(St.Johns)
-3	Argentina(Buenos Aires), Brazil(DST), Brazil(no DST), Denmark-Greenland(Nuuk)
-2:30	Newfoundland and Labrador
-2	Brazil(no DST)
-1	Portugal(Azores)
0	Denmark-Faroe Islands(Torshavn), GMT, Greenland, Ireland(Dublin), Morocco, Portugal(Lisboa,Porto,Funchal), Spain-Canary Islands(Las Palmas),UK(London)
+1	Albania(Tirane), Austria(Vienna), Belgium(Brussels), Caicos, Chad, Croatia(Zagreb), Czech Republic(Prague), Denmark(Kopenhagen), France(Paris), Germany(Berlin), Hungary(Budapest), Italy(Rome), Luxembourg(Luxembourg), Macedonia(Skopje), Namibia(Windhoek), Netherlands(Amsterdam), Spain(Madrid)
+2	Estonia(Tallinn), Finland(Helsinki), Gaza Strip(Gaza), Greece(Athens), Israel(Tel Aviv), Jordan(Amman), Latvia(Riga), Lebanon(Beirut), Moldova(Kishinev), Romania(Bucharest), Russia(Kaliningrad), Syria(Damascus), Turkey(Ankara), Ukraine(Kyiv, Odessa)
+3	East Africa Time, Iraq(Baghdad), Russia(Moscow)
+3:30	Iran(Teheran)
+4	Armenia(Yerevan), Azerbaijan(Baku), Georgia(Tbilisi), Kazakhstan(Aktau), Russia(Samara)
+4:30	Afghanistan(Kabul)
+5	Kazakhstan(Aqtobe), Kyrgyzstan(Bishkek), Pakistan(Islamabad), Russia(Chelyabinsk)
+5:30	India(Calcutta)
+5:45	Nepal(Katmandu)
+6	Kazakhstan(Astana, Almaty), Russia(Novosibirsk,Omsk)
+6:30	Myanmar(Naypyitaw)
+7	Russia(Krasnoyarsk), Thailand(Bangkok)
+8	Australia(Perth), China(Beijing), Russia(Irkutsk, Ulan-Ude), Singapore(Singapore)
+8:45	Eucla
+9	Japan(Tokyo), Korea(Seoul), Russia(Yakutsk,Chita)

Time Zone	Time Zone Name
+9:30	Australia(Adelaide), Australia(Darwin)
+10	Australia(Brisbane), Australia(Hobart), Australia(Sydney,Melbourne,Canberra), Russia(Vladivostok)
+10:30	Australia(Lord Howe Islands)
+11	New Caledonia(Noumea), Russia(Srednekolymsk Time)
+11:30	Norfolk Island
+12	New Zealand(Wellington,Auckland), Russia(Kamchatka Time)
+12:45	New Zealand(Chatham Islands)
+13	Tonga(Nukualofa)
+13:30	Chatham Islands
+14	Kiribati

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