



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

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CE

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

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

http://www.yealink.com/GPLOpenSource.aspx?BaseInfoCateId=293&NewsCateId=293&CateId=293&CateId=293&NewsCateId=

About This Guide

Thank you for choosing the SIP-T19 E2/SIP-T19P E2 (hereinafter referred to as 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

Summary of Changes

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

Changes for Release 80, Guide Version 80.130

The following section is new:

Call Park on page 86

Major updates have occurred to the following sections:

- Volume on page 28
- Ring Tones on page 29

• Programable Keys on page 52

Changes for Release 80, Guide Version 80.95

The following section is new:

• Entering Data and Editing Fields on page 18

Major updates have occurred to the following sections:

- Icon Instructions on page 3
- Optional Accessories on page 10
- Phone Installation on page 11
- Troubleshooting on page 125
- Appendix A Time Zones on page 139

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Overview

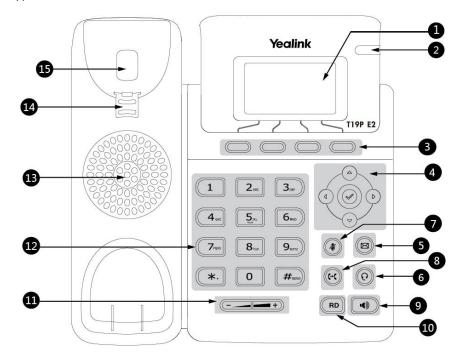
This chapter provides the overview of the SIP-T19(P) E2 IP phone. Topics include:

- 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.



	ltem	Description
		Shows information about calls, messages, soft keys,
		time, date and other relevant data:
		 Call information—caller ID, call duration
(1)	LCD Screen	 Icons (for example, DND)
		Missed call text or second incoming caller information
		 Prompt text (for example, "Saving config file!")
		Time and date
		Indicates phone power status and phone status:
2	Power Indicator LED	Receives an incoming call—Fast flashing
		Receives a voice mail or text message—Slow flashing
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.
\bigcirc	(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 hands-free (speakerphone) audio output.
		Secures the handset in the handset cradle when the IP
(14)	Hookswitch Tab	phone is mounted vertically. For more information on
)		how to adjust the hookswitch tab, refer to Yealink Wall
		Mount Quick Installation Guide for Yealink IP Phones.
		Picking up the handset from the handset cradle, the
(15)	Hookswitch	hookswitch bounces and the phone connects to the line,
		laying the handset down on the handset cradle, the
		phone disconnects from the line.

Hardware component instructions of the SIP-T19(P) E2 IP phone are:

Icon Instructions

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

lcon	Description	
	Network is unavailable	
6	The private line registers successfully	
	Register failed	
\bigcirc	Registering	
2	The shared/bridged line registers successfully	
•••)	Hands-free speakerphone mode	
۲.	Handset mode	
N	Headset mode	
00	Voice Mail	
	Text Message	
AA	Auto Answer	
DND	Do Not Disturb	
0	Call Hold	
□¢ ×	Ringer volume is 0	
B	Phone Lock	
Ą	Call Mute	
	Received Calls	

lcon	Description	
~	Placed Calls	
~	Missed Calls	
Ċ	Call Forward/Forwarded Calls	
	The contact icon	

LED Instructions

Power Indicator LED

LED Status	Description
Solid yellow	The phone is initializing.
Fast flashing yellow (300ms)	The phone is ringing.
Slow flashing yellow (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_CP860_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 21.

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 *(w)* 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	\checkmark	\checkmark
Network		
Phone		
Accounts		
Basic Phone Settings		
Contrast	\checkmark	
Language	\checkmark	
Time & Date	\checkmark	
Administrator Password	\checkmark	
Key as Send	\checkmark	1
Phone Lock	\checkmark	v
Ring Tones	\checkmark	
Contact Management		
Directory	х	
Local Directory	\checkmark	
Blacklist	\checkmark	

Options	Phone User Interface	Web User Interface
Remote Phone Book	x	
Call History Management	\checkmark	
Logo Customization	x	
Programable Keys	x	
Account Registration	\checkmark	
Dial Plan	x	
Emergency Number	x	
Live Dialpad	x	
Hotline	\checkmark	
Basic Call Features		
Recent Call In Dialing	x	
Auto Answer	\checkmark	
Auto Redial	\checkmark	
Call Completion	\checkmark	
ReCall	x	
Do Not Disturb (DND)	\checkmark	1
Call Forward	\checkmark	V
Call Transfer	\checkmark	
Call Waiting	\checkmark	
Conference	x	
Call Pickup	\checkmark	
Anonymous Call	\checkmark	
Anonymous Call Rejection	\checkmark	
Advanced Phone Features		
Hot Desking	\checkmark	
Intercom	\checkmark	
Multicast Paging	x	1
Shared Call Appearance (SCA)	x	V
Bridged Line Appearance (BLA)	x	
Music on Hold	x	
Messages	\checkmark	
SIP Account		
User Options		
Active Line	\checkmark	
Label	\checkmark	
Display Name	\checkmark	\checkmark
Register Name	\checkmark	
User Name	\checkmark	
Password	\checkmark	
Server Option		

Options	Phone User Interface	Web User Interface
SIP Server 1/2	\checkmark	
Register Port	х	
Outbound Status	\checkmark	
Outbound Proxy1/2	\checkmark	
Proxy Fallback Interval	\checkmark	
NAT Status	\checkmark	

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	In the package	English
	customizations	On the website	English/Chinese
User Guide	Phone/Web user interface settings Basic call features and advanced phone features	On the website	English/Chinese

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:

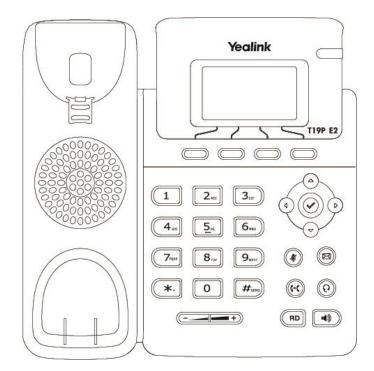
- Packaging Contents
- Phone Installation
- Phone Initialization
- Phone Status
- Basic Network Settings
- Registration
- Idle Screen
- Entering Data and Editing Fields

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:

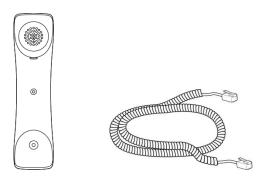
SIP-T19(P) E2 IP Phone



• Phone Stand



• Handset & Handset Cord



Ethernet Cable



Quick Start Guide

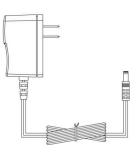


Check the list before installation. If you find anything missing, contact your system administrator.

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



Note We recommend that you use the accessories provided or approved by Yealink. The use of unapproved third-party accessories may result in reduced performance.

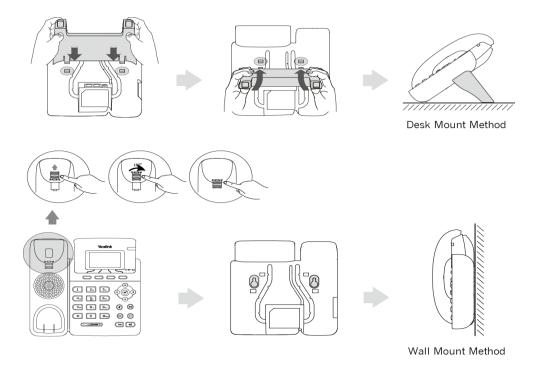
Phone Installation

If your phone is already installed, proceed to Phone Initialization on page 14.

This section introduces how to install the phone:

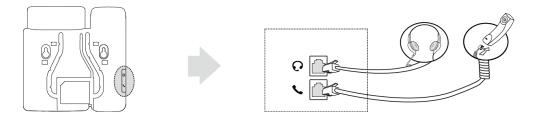
- 1) Attach the stand
- 2) Connect the handset and optional headset
- 3) Connect the network and power

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.

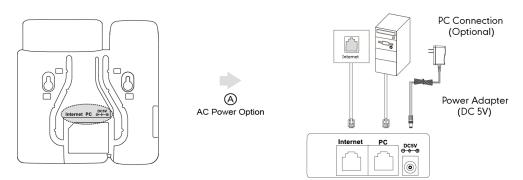
- AC power
- Power over Ethernet (PoE)

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.



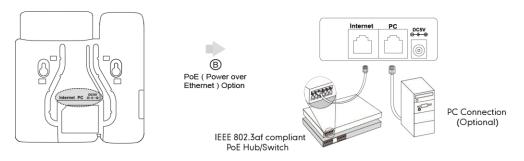
Note The IP phone should be used with Yealink original power adapter (5V/600mA) only. The use of the third-party power adapter may cause the damage to the phone.

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

Note

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

To view the phone status via phone user interface:

- 1. Press (✓), or press Menu->Status.
- 2. Press () or (\neg) to scroll through the list and view the specific information.

Status				
1.	IPv	4: 10.3.2	0.16	
2. MAC: 00:15:65:8B:22:AA				
3. Firmware: 53.80.0.90				
В	ack			

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 the Enter key.

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

Login	Enterprise IP phone SIP-T19P_E2		
Username Password			
Cor	nfirm Cancel		

4. Click **Confirm** to login.

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

Yealink			Log Out
	Status Account	Network DSSKey Features	Settings Directory Security
Status	Version		NOTE
	Firmware Version	53.80.0.90	Version
	Hardware Version	53.0.0.128.0.0.0	It shows the version of firmware and hardware.
	Network		Network
	Internet Port	IPv4	It shows the network settings of Internet (WAN) port.
	IPv4		Account
	WAN Port Type	DHCP	It shows the registration status of SIP accounts.
	WAN IP Address	10.3.20.16	You can click here to get
	Subnet Mask	255.255.255.0	more guides.
	Gateway	10.3.20.254	
	Primary DNS	192.168.1.22	
	Secondary DNS	192.168.1.20	
	Network Common		
	MAC Address	0015658B22AA	
	Link Status	Connected	
	Device Type	Bridge	
	Account Status		
	Account	106@10.2.1.199 : Registered	

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 mode via phone user interface:

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

	JAN Por	t Option	
1. IF	• Mode:		
IPv4	ŀ		••
Back			Save

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

You can configure a static IPv4 address for the IP phone. Before configuring it, make sure that the IP mode is configured as **IPv4** or **IPv4 & IPv6**.

To configure a static IPv4 address via phone user interface:

- Press Menu->Settings->Advanced Settings (default password: admin)
 ->Network->WAN Port.
- 2. Press (\neg) to select the **IPv4** and then press the **Enter** soft key.
- 3. Press (\neg) to select Static IPv4 Client and then press the Enter soft key.
- 4. Enter the desired value in the IPv4, Subnet Mask, Default Gateway, IPv4 Pri.DNS and IPv4 Sec.DNS field respectively.

S [.]	tatic IF	^p v4 Clien	t
1. IF	ν 4 :		
192.1	68.1.10		
Back	123	Delete	Save

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

You can configure a static IPv6 address for the IP phone. Before configuring it, make sure that the IP mode is configured as **IPv6** or **IPv4 & IPv6**.

To configure a static IPv6 address via phone user interface:

- Press Menu->Settings->Advanced Settings (default password: admin)
 ->Network->WAN Port.
- 2. Press (\neg) to select **IPv6** and then press the **Enter** soft key.
- 3. Press (\neg) 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 field respectively.

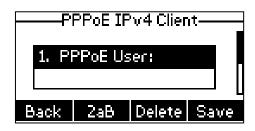


5. Press the Save soft key to accept the change or the Back soft key to cancel.

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 (\neg) to select **IPv4** and then press the **Enter** soft key.
- **3.** Press (\neg) 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 The 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 57.

Idle Screen

2 1028			
-	14:0	01 58	
	Wed A	ipr 29	
History	Dir	DND	Menu

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.

Entering Data and Editing Fields

You can enter data and edit fields using the phone keypad.

Keypad on the phone provides different characters (or numbers) when using the **2aB**, **abc**, **Abc**, **ABC** or **123** input mode. You can change the following input modes to enter data and edit fields on your phone. When your phone keypad matches the input mode, you can press the keypad repeatedly to view the character (or numbers) options and stop to select. When the character (or numbers) you want to enter displays in the field, wait one second, and enter the next character (or numbers).

The following table lists the input modes and character (or numbers) options for the keypad:

Input Mode Keypad	2aB	abc	Abc (initials in capitals)	ABC	123
	1				1
2	2abcABC	abc2äæå àáâãç	abc2äæå àáâãç	ABC2ÄÆÅ ÀÁÂÃÇ	2
3	3defDEF	def3èéêë ð	def3èéêë ð	DEF3ÈÉÊË Đ	3
(4 _{ost})	4ghiGHI	ghi4ìíîï	ghi4ìíîï	GHI4ÌÍĨĨ	4
5	5jklJKL	jkl5£	jkl5£	JKL5£	5
6 _{mc}	6mnoMN O	mno6öøò óôõñ	mno6öøò óôõñ	MNO6ÖØ ÒÓÔÕÑ	6

Input Mode Keypad	2aB	abc	Abc (initials in capitals)	ABC	123
77005	7pqrsPQR S	pqrs7BS	pqrs7BS	PQRS7S	7
8 _{tw}	8tuvTUV	tu∨8ùúûü	tu∨8ùúûü	TUV8ÙÚÛ Ü	8
9.erz	9wxyzWX YZ	wxyz9ýÞ	wxyz9ýÞ	WXYZ9ÝÞ	9
0	0	space	space	space	0
*.	*.,'?!\-()@/: _;+&%=< > £\$¥€[]{} ~^i¿\$#"	*.,'?!\-()@/: _;+&%=< > £ \$¥€[]{} ~^i¿\$#"	*.,'?!\-()@/: _;+&%=< > £ \$¥€[]{} ~^i&\$#"	*.,'?!\-()@/: _;+&%=< > £\$¥€[]{} ~^i&\$#"	.*:/@[]
# 100	#	#	#	#	#

To enter or edit data:

Do one of the following:

If you want to	Then you can
Enter only digits (1), uppercase (A) characters, lowercase (a) characters, or alphanumeric (2aB) characters.	Press a keypad key one or more times (depending what input mode you're in) to enter the characters that is displayed on the keypad key. You can press the abc soft key one or more times to switch among uppercase (ABC soft key), numeric (123 soft key), alphanumeric (2aB soft key), uppercase and lowercase (Abc soft key) and lowercase (abc soft key) input modes. For example, if the input mode is ABC : - To enter "A", press 2 once. - To enter "B", press 2 twice quickly. - To enter "C", press 2 three times
	quickly.

If you want to	Then you can
	 To enter "2ÄÆÅÀÁÂÂÇ", press 2 more than three times quickly.
	Note: When you are in the uppercase (ABC soft key), uppercase and lowercase (Abc soft key) or lowercase (abc soft key) input mode, 1 is not available. Press the keypad key #==> or (*.), or
	 press o . For o Key: If it is in the uppercase (ABC soft key), uppercase and lowercase (Abc soft key) or lowercase (abc soft key) input mode, it will provide the space character. If it is in the numeric (123 soft key) or alphanumeric (2aB soft key) input mode,
Enter special characters.	 it will only provide the digit 0. For # key: It only provides the pound character #. For ★ key: If it is in the uppercase (ABC soft key), lowercase (abc soft key), uppercase and lowercase (Abc soft key) or alphanumeric (2aB soft key) input mode, it will provide the following special characters: *.,?!\-()@/:_;+&%=<>f\$¥€[]{}~^i¿§#"]. If it is in the numeric (123 soft key) input
	mode, it will provide the following special characters: .*:/@[] .
Delete text you entered.	Press (1) or (b) to position the cursor to the right of the text you want to delete, and then press the Delete soft key to delete one character at a time.

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. The intensity of contrast ranges from 1 to 10 and the highest intensity is 10.

To configure the contrast via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Display->Contrast.
- Press (d) or (b), or the Switch soft key to increase or decrease the intensity of contrast.

The default contrast level is "6".

(Contrast	: Setting	
-			
1. C	ontrast		
6			41
Back		Switch	Save

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 (\diamond) or (\bigtriangledown) 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.

Yealink								Log Out
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
PreferenceTime & DateCall DisplayUpgradeAuto ProvisionConfigurationDial PlanVoiceRingTonesSoftkey LayoutTR069Voice MonitoringSIP	Live Inte Con Wat Ring	juage Dialpad r Digit Time(1~14s trast .chDog .Type .ad Ringtone .Confi		English (English) Disabled 4 6 Disabled Ring1.wav Browse ^{***} N Upload	v		user interface. Live Dialpad It allows IP ph automaticaly c entered phono- specified perio Backlight Specifies the t LCD screen dis Contrast Specifies the c LCD screen dis Ring Tones A ring tone th when a call co phone.	ones to fial out the e number after a d of time. orightness of the play. contrast of the

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

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

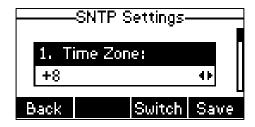
Time & Date

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 (a) or (b), 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".



- Enter the domain name or IP address of SNTP server in the NTP Server1 and NTP Server2 field respectively.
- Press (1) or (b), or the Switch soft key to select the desired value from the Daylight Saving field.
- Press (1) or (b), 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.

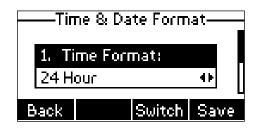
Manual 9 1. Date(YMD):	Gettings 15 -01 -07 ∎
2. Time(HMS)	
Back	L Save

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.
- Press (1) or (b), or the Switch soft key to select the desired time format (12 Hour or 24 Hour) from the Time Format field.



- Press (1) or (b), 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	

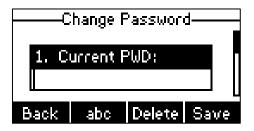
Time and date are configurable via web user interface at the path **Settings**->**Time & Date**.

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:

- 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 "#" key or "*" key to perform as a send key while dialing.

To configure key as send via phone user interface:

- 1. Press Menu->Features->Key as send.
- Press (a) or (b), or the Switch soft key to select # or * from the Key as send field, or select Disabled to disable this feature.

Key as send-			
1. Key as send:			
#			41
Back		Switch	Save

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, # key, * key 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 63.

To activate the phone lock via phone user interface:

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

Phone Lock			
1. Lock Enable			
Enabled		••	
Back	Swi	tch Save	

3. Press (1) or (b), or the Switch soft key to select the desired type from the Lock Type field.

Phone Lock			
2. Lock Type:			
All Keys		<u>-</u> ₽ [
Back		Switch	Save

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 $\#_{ee}$ 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.

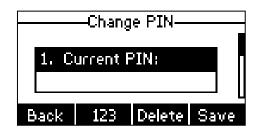
The **b** icon disappears from the LCD screen.

You can long press $\boxed{\#}$ 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 Change 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 field respectively.

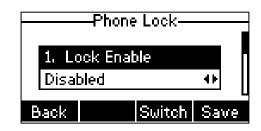


3. Press the Save soft key to accept the change or the Back soft key to cancel.

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 (\triangleleft) or (\triangleright), 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

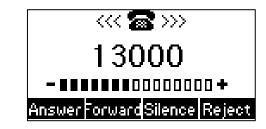
You can press the Volume key to adjust the ringer volume when the phone is idle or ringing. You can also press the Volume key to adjust the receiver volume of currently engaged audio devices (handset, speakerphone or headset) when the phone is in use.

To adjust the ringer volume:

When the phone is idle:

[2 1028			
		14:	47 05	
		100000	000000	10 +
	History	Dir	DND	Menu

When the phone is ringing:



Note If the ringer volume is adjusted to minimum, the \mathbb{Q}^{\times} icon will appear on the LCD screen.

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



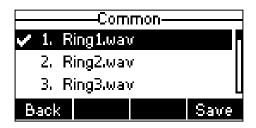
Ring Tones

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. (Optional.) Press (-___+) to adjust the ringer volume.
- 4. 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 (\diamond) or (\neg) to select the desired account and then press the **Enter** soft key.

Ring Tones-	
1. Common	Γ
2. 1002	
Back	Enter

3. Press (a) or (a) to select the desired ring tone.

If **Common** is selected, this account will use the ring tone selected for the phone.

\checkmark	1.		ommon			
	2. Ring1.wav					
	3. Ring2.wav					
B	ack	1			Save	

- 5. 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.

	Status	Account	Network	DSSKey	Features	Settings	Directory Security
	Lan	guage		English(English)			NOTE
Preference							NOTE
Time & Date	Live	Dialpad		Disabled	-		Language
Time & Date	Inte	er Digit Time(1~14s	;)	4			Selects a language for the w
Call Display	Con	trast		6	-		user interface.
							Live Dialpad
Upgrade	Wat	tchDog		Disabled	•		It allows IP phones to
Auto Provision	Ring	туре		Ring1.wav	•		automatically dial out the entered phone number after
AULO PROVISION	Unk	Upload Ringtone		Browser No file selected.		specified period of time.	
Configuration	opic	ad hangcone					Backlight
5				Upload	Cancel		Specifies the brightness of th
Dial Plan							LCD screen display.
		Confi	rm		Cancel		Contrast
Voice							Specifies the contrast of the
Ring							LCD screen display.
							Ring Tones
Tones							A ring tone that will alert you
							when a call comes in for the phone.
Softkey Layout							

3. Click Upload to upload the file.

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

Note

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.

In the Directory block, select the desired list from the Disabled column and then click .

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

- 3. Repeat the step 2 to add more lists to the **Enabled** column.
- 4. To remove a list from the **Enabled** column, select the desired list and then click 🦳 .
- 5. To adjust the display order of enabled lists, select the desired list and then click to or 1.

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

		Log Out
Yealink T19 E2		
	Status Account Network DSSKey Features Settings	Directory Security
Local Directory Remote Phone Book Phone Call Info Multicast IP Setting	Directory Disabled Enabled Remote Phone Book Local Directory History ↑ ↑ ↓	NOTE Directory It provides easy access to frequently used lists. Search Source in Dialing It allows the IP phone to automatically search entries from the search source list based on the sear
	Search Source List In Dialing	calls list when the phone is on the pre-dialing screen.
	Disabled Enabled	You can click here to get more guides.
	Remote Phone Book	
	Confirm	

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.

	——————————————————————————————————————						
1.	1. Local Directory						
2.	Hist	ory					
В.	ack			Enter			

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 44.

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 (refer to Directory on page 31), 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.
- Press (1) or (b), 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.

Add Group							
Name:							
Test							
Back	Abc	Delete	Add				

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 (refer to Directory on page 31), 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.

Local Di	rectory	
Detail		
Delete		
Delete All		
Cancel		OK

4. Press (a) or (a) to scroll through the group information and then edit.

Test							
Name:							
Test							
Back	Abc	Delete	Save				

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 (refer to Directory on page 31), 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:

Local Directory						
พ๛๛๛๛๛๛๛๛๚						
Delete selected group?						
Cancel OK						

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 (refer to Directory on page 31), press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group and then press the Enter soft key.

If the contact is not in any group, select **All Contacts** and then press the **Enter** soft key.

- 3. Press the Add soft key.
- 4. Enter the name and the office, mobile or other numbers in the corresponding fields.

Add Contact						
Name	:					
Bob						
Back	abc	Delete	Add			

5. Press (1) or (b), 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 (1) or (b), or the Switch soft key to select the desired group from the Group field.
- 7. Press the Add soft key to accept the change or the Back soft key to cancel.

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

Adding Contacts from Call History

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

- 1. Press the History soft key.
- **2.** Press (a) or (r) to highlight the desired entry.

3. Press the Option soft key, and then select Add to Contacts.



- 4. Press the OK soft key, and then edit the contact name.
- 5. Press the **Save** soft key to accept the change.

The entry is successfully saved to the local directory.

Adding Contacts from Remote Phone Book

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

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

If Remote Phone Book is added to the directory (refer to Directory on page 31), press **Dir->Remote Phone Book** to enter the remote phone book.

- 2. Select the desired remote group and then press the Enter soft key.
- **3.** Press (\diamond) or (\neg) to highlight the desired entry.
- 4. Press the Option soft key, and then select Add to Contacts.
- 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 44.

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.

Local Directory						
1. All Contacts						
2. Tes	2. Test					
Back	AddGr	Search	Enter			

If Local Directory is removed from the directory (refer to Directory on page 31),

press Menu->Directory->Local Directory to enter the local directory.

- Select the desired contact group and then press the Enter soft key.
 If the contact is not in any group, select All Contacts and then press the Enter soft key.
- **3.** Press (\triangle) or (∇) to highlight the desired contact.
- 4. Press the **Option** soft key, and then select **Detail**.
- 5. Press (\triangle) or (\neg) to highlight the contact information and then edit.

	B	ob	
Nam	e:		
Bob			
Back	abc	Delete	⊔ Save

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.



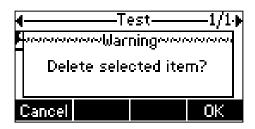
If Local Directory is removed from the directory (refer to Directory on page 31), press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group and then press the Enter soft key.

If the contact is not in any group, select **All Contacts** and then press the **Enter** soft key.

- **3.** Press (\bigcirc) or (\neg) 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.

	Local Directory—					
1. All (Contacts					
2. Test						
Baak	AddGr Search E	U atac				
Dauk	Huudi Search E					

If Local Directory is removed from the directory (refer to Directory on page 31), press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group and then press the Enter soft key.

If the contact is not in any group, select **All Contacts** and then press the **Enter** soft key.

- **3.** Press (\triangle) or (∇) to highlight the desired contact.
- 4. Do one of the following:
 - If only one number of the contact is stored in the local directory, press the **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 (refer to Directory on page 31), 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.

্ষা			0/1
ad			
Back	abc	Delete	

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

								Log Out
Yealink 119 E2								
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
Local Directory	Direc	tory					NOTE	
Remote Phone Book Phone Call Info		Disabled History Remote P	hone Boo	Enabled Local Directory	*		Directory It provides eas frequently use	d lists.
Multicast IP			→ +				Search Source It allows the IF automatically s from the search	P phone to earch entries th source list
Setting			Ŧ		Ŧ		based on the e and display res pre-dialing scre	ults on the een.
	Sear	ch Source List In	Dialing				Recent Call In It allows users placed calls list is on the pre-d	to view the when the phone
		Disabled	1	Enabled Local Directory History Remote Phone B			You can cli more guides.	ick here to get
			-		*			
		Recent Cal	ll In Dialing Disab	led	•			
		Confir	m		Cancel			

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 key.
- 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 \bigcirc or (\bigtriangledown) to scroll to the desired entry, and then place a call to the entry.

\$ 1029	5:		
1			
Test1(2	3000)		
Send	123	Delete	Cancel

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.

 Click Browse to locate a contact list file (the file format must be *.xml) from your local system.

Yealink 119 E2								Log Out
	Status	Account	Networ	k DSSKey	/ Feature	s Sett	ings	Directory Security
Local Directory	Index	Name	Office Numb	er Mobile Number	Other Al Number Al	Contacts 👻		NOTE
Remote Phone Book	1 2 3 4	Alice John	<u>1008</u> <u>1001</u>	<u>1234567891</u>		All Contacts All Contacts		Local Directory The built-in phone directory can store the names and phone
Phone Call Info Multicast IP	4 5 6 7							numbers of your contacts. You can add new groups and contacts, edit, delete or search for a contact, or simply dial a
Setting	8 9 10 Page 1 -	Prev Next	Hang Up	Delete All	Delete Mo	we To All C	Contac 🗸	contact number from the local directory. You can import or export the contact list.
	Directory		Thing op	Group Sett				You can click here to get more guides.
	Name Office Number Mobile Number			Ring Add	Edit Dele		All	
	Other Number Ring Tone	Auto)	Import Loc Browse*** Import XM	A Directory File			
	Group Add		ontacts Edit	Browse*** Import CS	No file selected.		Title	

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.
- 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.

	tatus	Acco	unt Netw	ork DSSK	ey Feature	s Settings	Directory Secur
Del	ete Old Co	ontacts 🔍 O	in 🔘 Off				NOTE
Ind	ex Display	/ Name 🛛 👻	Office Number	- Ignore ·	• Ignore •	Ignore 👻	I
1	disp	lay_name	office_number	mobile_number	other_number	line	contacts-preview-note
2		Judy	4561	1234		-1	You can click here to
3		nandy	4303	1235		-1	more guides.
4		Bob	1258	1236		1	June geneen

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.

	-Add B	lacklist—	
Nam	91		
Bob			
Back	Abc	Delete	Add

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 (refer to Directory on page 31), press **Menu->Directory->Local Directory** to enter the local directory.

2. Select the desired contact group and then press the Enter soft key.

If the contact is not in any group, select **All Contacts** and then press the **Enter** soft key.

- **3.** Press (\triangle) or (\neg) 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 accept the change.

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 33.

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.

Configuring an Access URL

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.

Yealink 119 E2	Status Account Nets	work DSSKey Features	Settings Directo	Log Out	
Local Directory	Index Remote	URL Disp	lay Name NOTE		
Remote Phone	1 http://10.3.6.130/Department.	xml xmyl	Remot	e Phone Book	
Book	2		It is a c	entrally maintained book, stored on the	
Phone Call Info	3			server.	
	4			nly need the access URL	
Multicast IP	5			remote phone book. The ne can establish a	
Setting				tion with the remote and download the phone	
	Incoming/Outgoing Call Lookup	Disabled	🚽 book, a	nd then display the	
	Update Time Interval(Seconds)	21600		remote phone book entries on the phone user interface.	
	Confirm	Cancel	You more g	can click here to get uides.	

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

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

Accessing the Remote Phone Book

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 (refer to Directory on page 31), press **Dir**->**Remote Phone Book** to enter the remote phone book.

Press (a) or (v) 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.

	——×п	nyl——	1/3-
🚨 Test	:1		
💄 Test	:2		
🚨 Test	3		
Back	Search	Option	Send

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

Placing Calls to Contacts

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 (refer to Directory on page 31), 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.

Searching for Contacts

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 (refer to Directory on page 31), 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.
- 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.

Incoming/Outgoing Call Lookup

You can enable the phone to present the caller/callee identity stored in the remote phone book when receiving/placing 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.

	Status	Account Network DS	6SKey Features Settings	Directory Security
Local Directory	Index	Remote URL	Display Name	NOTE
	1 http:	//10.3.6.130/Department.xml	xmyl	
Remote Phone Book	2			Remote Phone Book It is a centrally maintained
Phone Call Info	3			phone book, stored on the remote server.
Phone Call Into	4			Users only need the access U
Multicast IP	5			of the remote phone book. T
Setting				IP phone can establish a connection with the remote server and download the pho
	Incoming/Outgoing Call Lookup		Enabled 👻	book, and then display the remote phone book entries o
	Updat	e Time Interval(Seconds)	21600	the phone user interface.

4. 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:

- 1. Press Menu->Features->History Setting.
- Press (1) or (b), or the Switch soft key to select Disabled from the History Record field.



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

To view the call history:

1. Press the History soft key.

The LCD screen displays all call records.

Press (a) or (b) to switch among All Calls, Placed Calls, Received Calls, Missed Calls and Forwarded Calls.

- **3.** Press \bigcirc or \bigtriangledown 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 (d) or (b) to switch among All Calls, Placed Calls, Received Calls, Missed Calls and Forwarded Calls.
- **3.** Press (\bigtriangleup) or (\bigtriangledown) 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.
- Press (d) or (b) to switch among All Calls, Placed Calls, Received Calls, Missed Calls and Forwarded Calls.
- **3.** Press (a) or $(\neg$ 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 31.

To delete an entry from the call history list:

- 1. Press the History soft key.
- Press (1) or (b) to switch among All Calls, Placed Calls, Received Calls, Missed Calls and Forwarded Calls.
- 3. Press \bigcirc or \bigtriangledown 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.
- Press (1) or (b) to switch among 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.

All Calls—1/8• •••••••••Warning••••••• Delete all the call records? Cancel OK

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.

Yealink 119 E2								Log Out
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
Forward&DND	Ge	eneral Informati	on				NOTE	
General Information		Call Waiting Call Waiting On Co	ode	Enabled	•			ones to receive a
Audio		Call Waiting Off Co Auto Redial	ode	Disabled			already an acti	call when there is ive call.
Intercom		Auto Rediai		Disabled	Ţ		Auto Redial It allows IP ph automatically r	
Transfer				:			number after	the first attempt.
Call Pickup		Call Number Filter		-			Key As Send Assigns "#" or key.	"*" as the send
Remote Control	-	Use Logo		,- Custom logo			Hotline	
Phone Lock		Upload Logo		Browse No	file selected.		out the hotlin	automatically dial e number when dset, pressing the
ACD					icel			key or the line
SMS		Accept SIP Trust Allow IP Call	Server Only	Disabled	-		Call Complet	ion to monitor the
Action URL	DHCP Hostnan			SIP-T19P_E2			busy party and	to monitor the d establish a call y party becomes
Power LED		Reboot in Talking		Disabled	•		available to rea	
Notification Popups		Hide Feature Acce		Disabled	•		You can cl more guides.	lick here to get
		Display Method or		User Name	Cancel			

- 4. Click Upload to upload the file.
- 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 67.

To deactivate the headset mode:

1. Press (0) 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.

	Status	Account	Network	DSSKey	Features	Settings	Directory Securit	
Forward&DND	6	eneral Informati	ion				NOTE	
General		Call Waiting		Enabled	•		Call Waiting	
Information		Call Waiting On C	ode				It allows IP phones to receiv	
Audio		Call Waiting Off C	ode				new incoming call when the already an active call.	
Audio		Auto Redial		Disabled	•		Auto Redial	
Intercom							It allows IP phones to automatically redial a busy	
Transfer				•			number after the first attem	
o-II pi-lass				:			Key As Send	
Call Pickup							Assigns "#" or "*" as the ser key.	
Remote Control		Play Hold Tone		Enabled	•			
Phone Lock		Play Hold Tone D	elay	30			Hotline IP phone will automatically d	
FIIORC LOCK		Allow Mute		Enabled	•		out the hotline number whe lifting the handset, pressing	
ACD		Dual-Headset		Disabled	•		speakerphone key or the line key.	
SMS		Auto-Answer Del	ay(1~4s)	1				
Action URL		Enable auto answ	rer tone	Enabled	•		Call Completion It allows users to monitor th	
ACUOITOKL		Headset Prior		Enabled	•		busy party and establish a ca when the busy party becom	
Power LED		Reboot in Talking		Disabled	•		available to receive a call.	
Notification Popups		Hide Feature Acc	ess Codes	Disabled	•		🛽 You can click here to ge	
		Display Method or	Dipling	User Name	•		more guides.	

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 (0) 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.

ealink 119 E2	tatus Account Network	DSSKey Features	Settings	Directory Security	
Forward&DND	General Information			NOTE	
General	Call Waiting	Enabled -		Coll Mailting	
Information	Call Waiting On Code			Call Waiting It allows IP phones to receive	
Audio	Call Waiting Off Code			new incoming call when ther already an active call.	
Audio	Auto Redial	Disabled -		Auto Redial	
Intercom				It allows IP phones to automatically redial a busy	
Transfer		:		number after the first attemp	
Call Pickup		•		Key As Send Assigns "#" or "*" as the sen	
Remote Control	Play Hold Tone	Enabled 🔹		key.	
Kemote control	Play Hold Tone Delay	30		Hotline	
Phone Lock	Allow Mute	Enabled 🗸		IP phone will automatically di out the hotline number when	
ACD	Dual-Headset	Enabled 👻		lifting the handset, pressing t speakerphone key or the line	
SMS	Auto-Answer Delay(1~4s)	1		key.	
	Enable auto answer tone	Enabled 👻		Call Completion It allows users to monitor the	
Action URL	Headset Prior	Disabled 👻		busy party and establish a cal when the busy party become	
Power LED	Reboot in Talking	Disabled 🗸		available to receive a call.	
Notification Popups	Hide Feature Access Codes	Disabled 👻		You can click here to get	
	Display Method on Dialing	User Name		more guides.	

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

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

Note Dual headset is configurable via web user interface only.

Programable Keys

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

To customize the programable keys via web user interface:

1. Click on DSSKey->Programable Key.

2. Customize specific features for these keys.

	Status	Account	Network	DSSKey	Features	Settings	Directory Security
rogramable Key	Кеу	Туре	Line	Value	Label	Extension	NOTE
5 1	SoftKey 1	History	- Local History -				
	SoftKey 2	Directory	• N/A •				Programmable Keys Customizes the soft keys,
	SoftKey 3	DND	• N/A •				navigation keys and function keys.
	SoftKey 4	Menu	• N/A •				
	Up	History	- Local History -				You can click here to get
	Down	Directory	• N/A •				more guides.
	Left	N/A	• N/A •				
	Right	N/A	• N/A •				
	ок	Status	• N/A •				
	Mute	N/A	• N/A •				
	Tran	Forward	• N/A •				

- (Optional.) Enter the 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 Programable keys are configurable via web user interface only.

Common used programable 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 Browser
- SMS
- New SMS
- Zero Touch
- Phone Lock
- Directory

For the features not listed above, refer to Basic Call Features on page 67 and Advanced Phone Features on page 97. For more information, contact your system administrator.

Speed Dial

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

Dependencies: Type (Speed Dial)

Value (the number you want to dial out)
Label (key label displayed on the LCD screen)

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

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) Label (key label displayed on the LCD screen)

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. You can only configure the SoftKey (1-4) as the prefix key.

Dependencies: Type (Prefix)

Value (the prefix number)

Label (key label displayed on the LCD screen)

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 33.

Dependencies: Type (Local Directory)

Label (key label displayed on the LCD screen)

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 33.

Dependencies: Type (Local Group)

Line (the contact group you want to access) Label (key label displayed on the LCD screen)

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 44.

Dependencies: Type (XML Directory)

Label (key label displayed on the LCD screen)

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 <u>Remote Phone Book</u> on page 44.

Dependencies: Type (XML Group)

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

Label (key label displayed on the LCD screen)

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 an 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)

Label (key label displayed on the LCD screen)

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 117.

Dependencies: Type (SMS)

Label (key label displayed on the LCD screen)

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 117.

Dependencies: Type (New SMS)

Label (key label displayed on the LCD screen)

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)

Label (key label displayed on the LCD screen)

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 26.

Dependencies: Type (Phone Lock)

Label (key label displayed on the LCD screen)

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 31.

Dependencies: Type (Directory)

Label (key label displayed on the LCD screen)

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.
- Press (1) or (b), or the Switch soft key to select Enabled from the Active Line field.

	——Lii	ne	
1. Ac	stive Lir	ne:	
Enab	led		41
Back		Switch	Save

- Enter the desired value in Label, Display Name, Register Name, User Name, Password and SIP Server1/2 field respectively. Contact your system administrator for more information.
- 5. If you use the outbound proxy servers, do the following:
 - Press (1) or (b), 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** field 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.
- Press (d) or (b), 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 "12 3 ", "12 34 ", "12 345 ", "12 abc ", etc.
	An "x" can be used as a placeholder for any character. Example:
X	"12x" would match "12 1 ", "12 2 ", "12 3 ", "12 a ", 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
0	character which matches any of a set of characters. Example:
	"91[5-7]1234" would match "91 5 1234", "91 6 1234", "91 7 1234".
	The parentheses "()" can be used to group together patterns, for
()	instance, to logically combine two or more patterns. Example:
	"([1-9])([2-7])3" would match " 92 3", " 15 3", " 77 3", etc.
	The "\$" should be followed by the sequence number of a parenthesis.
	The "\$" plus the sequence number means the whole character or
	characters placed in the parenthesis. The number directs to the right
	parenthesis when there are more than one. Example:
\$	A replace rule configuration, Prefix: "001(xxx)45(xx)", Replace:
	"9001\$145\$2". When you dial out "0012354599" on your phone, the IP
	phone will replace the number with "9001 235 45 99 ". "\$1" means 3 digits
	in the first parenthesis, that is, "235". "\$2" means 2 digits in the second
	parenthesis, that is, "99".

Replace Rule

You can configure one or more replace rules (up to 100) to remove the specified string and replace it with another string. You can configure a pattern with wildcards (refer to the expression syntax in the table above), so that any string that matches the pattern will be replaced. This feature is convenient for you to dial out a long number. For example, a replace rule is configured as "Prefix: 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.
- 3. Enter the string (e.g., 1234) in the **Replace** field.

Yealink 119 E2							Log Out
	Status	Account	Network	DSSKey	Features	Settings	Directory Security
Preference	Replace Rule	e Dial-now Area	a Code Block	Out			NOTE
Time & Date	Index	Prefix			Replace		Replace Rule: An alternative
	1						string that replaces the entered numbers.
Call Display	2						Dial-now: Automatically dial out
Upgrade	3						the entered numbers. Area Code:Automatically add
Auto Provision	4						the area code before the numbers when dialing.
AULO PTOVISION	5						Block Out: It prevents users
Configuration	6						from dialing out specific numbers.
Dial Plan	7						".":represents any string.
	8						"x":represents any character.
Voice	9						"-":match a range of characters within the brackets.
Ring	10						",":a separator within the bracket.
Tones							"[]":a character matches any of character sets. "()":combines two or more
Softkey Layout	Prefit	(1		Replace	1234		patterns. "\$":followed by the sequence
TR069		Add	E	dit	Del	,	number of a parenthesis means the characters placed in the parenthesis.
Voice Monitoring SIP							You can click here to get more guides.

4. 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:

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

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

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

Note

Replace rule is configurable via web user interface only.

Dial-now

You can configure one or more dial-now rules (up to 100) on your phone. When the dialed 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.

Me erlinde							Log Out
Yealink 119 E2	Status	Account	Network	DSSKey	Features	Settings	Directory Security
Preference	Replace Rule	e Dial-now A	rea Code Block	Out			NOTE
Time & Date	Index		Dial-nov	w Rule			Replace Rule: An alternative
Call Display	1 2						string that replaces the entered numbers. Dial-now:Automatically dial out
Upgrade	3						the entered numbers. Area Code:Automatically add
Auto Provision	4 5						the area code before the numbers when dialing. Block Out: It prevents users
Configuration	6						from dialing out specific numbers.
Dial Plan	7						".":represents any string.
Voice	8						"x":represents any character. "-":match a range of characters within the brackets.
Ring	10						",":a separator within the bracket.
Tones							"[]":a character matches any of character sets. "O":combines two or more
Softkey Layout			Rule 1xx				patterns. "\$":followed by the sequence
TR069		Add		Edit	Del		number of a parenthesis means the characters placed in the parenthesis.
Voice Monitoring							You can click here to get more guides.

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 58 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.

	Status	Account	Network	DSSKey	Features	Settings	Directory	Security	
Forward&DND	Ger	neral Informatio	n				NOTE		
General		Call Waiting		Enabled	¥		Call Waiting		
Information		Call Waiting On Co	de				It allows IP ph	ones to receive call when there	
Audio		Call Waiting Off Co	de				already an acti		
		Auto Redial		Disabled	¥		Auto Redial It allows IP ph	ones to	
Intercom		Auto Redial Interva	10			automatically redial a busy number after the first attempt			
Transfer	Auto Redial Times (1~300)			10 Key As Sen					
Call Pickup	Key As Send			#	¥			s "#" or "*" as the send	
Remote Control	1	Reserve # in User	Name	Enabled	¥		Hotline		
Phone Look	1	Hotline Number					IP phone will automatically dia out the hotline number wher lifting the handset, pressing t speakerphone key or the line key.		
Phone Lock	1	Hotline Delay(0~10)s)	4					
ACD		Busy Tone Delay (Seconds)	0	•		Call Completi	on.	
SMS		Return Code Wher	n Refuse	486 (Busy Here)	¥		It allows users	to monitor the	
Action URL	Return Code When DND		480 (Temporarily Unava 🔻			busy party and establish a call when the busy party becomes available to receive a call.			
Power LED	Call Completion Disabled			You can click here to	ick hore to get				
Power LED	1	Feature Key Synchronization			Disabled			more guides.	
Notification Popups		Time-Out for Dial-N	low Rule	1					
		RFC 2543 Hold		Disabled	T				
		Use Outbound Pro	xy In Dialog	Enabled	•				
		180 Ring Workarou	ind	Enabled	•				
		Logon Wizard		Disabled					

2. Enter the time between 0 and 14 (seconds) in the Time-Out for Dial-Now Rule field.

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

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

Area Code

Area codes are also known as Numbering Plan Areas (NPAs). They usually indicate geographical areas in a country. This feature is necessary when dialing a phone number outside the code area. For example, an area code is configured as "Code: 0592, Min Length: 1, Max Length: 15". When you dial out the number "56789" (the length of the number is between 1 and 15), the phone will add the area code and dial out the number "059256789". You can only configure one area code rule on your phone.

To configure the area code via web user interface:

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

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

		Log Out
Yealink 119 E2	Status Account Network DSSKey Features Sett	ings Directory Security
Preference	Replace Rule Dial-now Area Code Block Out	NOTE
Time & Date	Code 0592	Replace Rule: An alternative string that replaces the entered
Call Display	Min Length (1-15) 1	numbers. Dial-now:Automatically dial out
Upgrade	Max Length (1-15) 15	the entered numbers. Area Code:Automatically add
Auto Provision	Confirm	the area code before the numbers when dialing.
Configuration		Block Out: It prevents users from dialing out specific numbers.
Dial Plan		".":represents any string.
Voice		"x":represents any character. "-":match a range of characters within the brackets.
Ring		",":a separator within the bracket.
Tones		"[]":a character matches any of character sets. "Q":combines two or more
Softkey Layout		patterns. "\$":followed by the sequence
TR069		number of a parenthesis means the characters placed in the parenthesis.
Voice Monitoring		You can click here to get
SIP		more guides.

3. 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:

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

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

	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
Preference	Replace Rule	Dial-now A	rea Code Block	Out			NOTE	
Time & Date	BlockOut Num	ber1 105	8				Replace Rules	An alternative
Call Display	BlockOut Num	ber2					numbers.	
	BlockOut Num	ber3					the entered n	omatically dial or umbers.
Upgrade	BlockOut Num	ber4					Area Code:Au the area code	tomatically add
Auto Provision	BlockOut Num	ber5					numbers when	
	BlockOut Num	ber6					Block Out: It p from dialing ou	
Configuration	BlockOut Num	ber7					numbers.	L Specific
Dial Plan	BlockOut Num	ber8					".":represents	any string
DRITRIT	BlockOut Num	ber9					"x":represents	any character.
Voice	BlockOut Num	ber10					"-":match a ra within the brad	nge of characte rkets.
Ring		Confi			Cancel		",":a separator	
lang		Com			Cancer		bracket. "[]":a characte	er matches any
Tones							character sets.	
Softkey Layout							"()":combines patterns.	cwo or more
Softkey Layout								y the sequence arenthesis mear

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 26.

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, 110.

			Log Out
Yealink 119 E2	Status Account Network	DSSKey Features Settings	Directory Security
Forward&DND	Phone Lock Enable Phone Lock Type	Disabled	NOTE
General Information	Phone Unlock PIN(0~15 Digit)	•••••	Phone Lock It is used to lock the IP phone to prevent it from unauthorized
Audio	Phone Lock Time Out(0~3600s)	0	use. Once the IP phone is locked, a user must enter the
Intercom	Emergency	112,911,110	password to unlock it.
Transfer	Confirm	Cancel	IP phones offer three types of phone lock: Menu Key, Function Keys and All Keys.
Call Pickup			The IP phone will not be locked immediately after the phone
Remote Control			lock type is configured.
Phone Lock			You can click here to get more guides.
ACD			

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.

alink 119 E2	Status	Account	Network	DSSKey	Features	Settings	Directory Security		
Preference	Lan	iguage		English(English)	•		NOTE		
Time & Date		e Dialpad er Digit Time(1~14:	s)	Enabled 4	•		Language Selects a language for the web		
Call Display	Cor	ntrast		6	•		user interface.		
Upgrade	Wa	tchDog		Disabled	•		Live Dialpad It allows IP phones to		
Auto Provision	Rin	д Туре		Ring1.wav	•		automatically dial out the entered phone number after		
Configuration	Upload Ringtone		Browse M	No file selected.		specified period of time. Backlight Specifies the brightness of the			
Dial Plan		Conf	ìrm	ſ	Cancel		LCD screen display.		
Voice							Contrast Specifies the contrast of the		
Ring							LCD screen display.		
Tones							Ring Tones A ring tone that will alert you when a call comes in for the		
Softkey Layout							phone.		

The default delay time is 4 seconds.

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

Live dialpad is configurable via web user interface only.

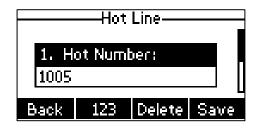
Hotline

Note

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 Park
- Call Pickup
- Anonymous Call
- Anonymous Call Rejection

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

Placing Calls

You can place a call in one of three ways using your SIP-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 31 and Call History Management on page 47.

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 and far-site's information (name or 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 25.

Note You can also dial using the SIP URI or IP address. To obtain the IP address of a phone, press the A 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 (\checkmark), #, or the **Send** soft key.

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

Press \mathbf{w} , $\mathbf{\#}$, $\mathbf{\#}$, $\mathbf{\#}$, or the **Send** soft key.

To place a call using the headset:

- 1. With the optional headset connected, press (0) to activate the headset mode.
- 2. Enter the desired number using the keypad.
- **3.** Press (\checkmark) , $\#_{\text{seed}}$, or the **Send** soft key.

Note To permanently use the headset mode, refer to Headset Prior on page 50.

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 (\checkmark) , (#), or the **Send** soft key.

You can press (a) 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 77.

You can forward incoming calls to someone else by pressing the **FWD** soft key. For more information, refer to Call Forward on page 78.

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 (\neg) to access the new call.

Press (\checkmark) 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 (RD) 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 (a) or (v) 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 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.

Yealink 119 E2							Log Out
	Status	Account	Network	DSSKey	Features	Settings	Directory Security
Local Directory Remote Phone	Direc	tory Disabled		Enabled			NOTE
Book Phone Call Info Multicast IP Setting		History Remote P	hone Bool	Local Directory			It provides easy access to frequently used lists. Search Source in Dialing It allows the IP phone to automatically search entries from the search source list based on the entered string, and display results on the
	Searc	h Source List In Disabled	Tialing	Enabled	Ŧ		pre-dialing screen. Recent Call In Dialing It allows users to view the placed calls list when the phone is on the pre-dialing screen.
		Remote P	hone Boo	Local Directory History			You can click here to get more guides.
		Recent Cal		led	▼ Cancel		

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

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

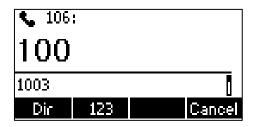
To view placed calls list when the phone is on the dialing screen:

1. Pick up the handset or press the Speakerphone key.

The LCD screen displays the placed calls list.

% 106	:	
1058		
Dir	123	Cancel

You can also enter a few continuous characters of the contact name or continuous numbers of the contact number (office, mobile or other number) to search from 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 v to select the Status, and the press (1) or (b), or the Switch soft key to select Enabled from the Status field.

	-Auto A	Answer—	Г
2. St	tatus:		
Enab	led		41
Back		Switch	Save

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

2 1025			88
	10:	39 59	
	Thu A	pr 30	
History	Dir	DND	Menu

The AA icon appears on the LCD screen.

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

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

Auto Redial

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

To configure auto redial via phone user interface:

- 1. Press Menu->Features->Auto Redial.
- Press (1) or (1), 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 value is "10".

- Enter the desired number of redial attempts in the Redial Times field. The default value is "10".
- 5. Press the Save soft key to accept the change or the Back soft key to cancel.

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

To use auto redial:

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



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

ιφ Ca	all Finish	
<u>ر ا</u>		
	Redialing 1	058 7s
<u>k</u>		/
OK		Cancel

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.
- Press (a) or (b), or the Switch soft key to select Enabled from the Call Completion field.

	Call Cor	npletion	
1. C	all Comp	oletion:	
Enab	led		41
Back		Switch	Save

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:

2 1025	
wwwwCall Con	npletion~~~~
Wait for	1058 ?
Cancel	ОК

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:

21025			
๛๛๛๛	Call Con	npletion	๛๛๛๛
	Dialing	1058 ?	
Cancel			OK

1. 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**.

	Status	Account	Network	DSSKey	Features	Settings	Directory Security
Programable Key	Кеу	Туре	Line	Value	Label	Extension	NOTE
	SoftKey 1	History 🗸	Local History 🗸				Description of the second
	SoftKey 2	Directory 🗸	N/A 🗸				Programmable Keys Customizes the soft keys,
	SoftKey 3	DND 🗸	N/A 🗸				navigation keys and function keys.
	SoftKey 4	ReCall 🗸	N/A 🗸]
	Up	History 🗸	Local History 🗸				You can click here to get more guides.
	Down	Directory 🗸	N/A 🗸				
	Left	N/A 🗸	N/A 🗸				
	Right	N/A 🗸	N/A 🗸				
	ОК	Status 🗸	N/A 🗸				
	Mute	N/A 🗸	N/A 🗸				
	Tran	Forward V	N/A 🗸				

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 (a) or (v) 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 DND icon on the status bar indicates that DND is enabled.

Incoming calls will be rejected automatically and "**n New Missed Call(s)**" ("n" indicates the number of missed calls. e.g., 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 79.

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.

				Log Out
Yealink T19 E2	Status Account Network	DSSKey Features	Settings	Directory Security
Forward&DND	Forward			NOTE
General Information	Forward Emergency Forward Authorized Numbers	Disabled 👻		Call Forward It allows users to redirect an
Audio	Always Forward	🗢 On 🖲 Off		incoming call to a third party. Call Forward Mode Phone: Call forward feature is
Intercom	On Code			effective for the IP phone. Custom: Call forward feature
Transfer	Off Code			can be configured for each or all accounts.
Call Pickup	Busy Forward	© On ◉ Off		Do Not Disturb (DND) It allows IP phones to ignore
Remote Control	On Code			incoming calls.
Phone Lock	Off Code			Phone: DND feature is effective for the IP phone.
ACD	No Answer Forward After Ring Time(0~120s)	© On		Custom: DND feature can be configured for each or all accounts.
SMS Action URI	Target			You can click here to get
Power LED	On Code			more guides.
Notification Popups	Off Code			
Nouncation Popups	DND Emergency	Enabled 👻		
	DND Authorized Numbers	1022,1023		
	DND Status	◎ On		
	DND On Code			

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

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

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

Call Forward

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

Static Forwarding

Three types of static forwarding:

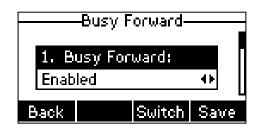
- Always Forward: Incoming calls are immediately forwarded.
- Busy Forward: Incoming calls are immediately forwarded if the phone is busy.
- No Answer Forward: Incoming calls are forwarded if not answered after a period of time.

To enable call forward via phone user interface:

- 1. Press Menu->Features->Call Forward.
- Press (a) or (v) 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 (d) or (b), or the Switch soft key to select Enabled from the Always Forward field.

	Always Forward					
	1. Always Forward:					
Enab	led	<u>+</u> [
Back	Switch	Save				

- 2) Enter the destination number you want to forward all incoming calls to in the **Forward to** field.
- (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:
 - Press (1) or (b), 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.
- (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:
 - Press (1) or (b), or the Switch soft key to select Enabled from the No Answer Forward field.

No Answer Forward					
1. No Answer Forward:					
Enabled 🔸					
Back		Switch	Save		

- 2) Enter the destination number you want to forward unanswered incoming calls to in the **Forward to** field.
- Press (1) or (1), 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 67.

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

To 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.

ealink 119 E2								Log O
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
Forward&DND		Forward					NOTE	
General Information		Forward Emerger Forward Authoriz		Enabled 1059,1052	-		Call Forward It allows users	to redirect an
Audio		Always Forward	I	◎ On ම Off			Call Forward Phone: Call fo	Mode rward feature is
Intercom Transfer		On Code Off Code					effective for the IP phone Custom: Call forward feat can be configured for each accounts.	
Call Pickup		Busy Forward		🗇 On 🖲 Off			Do Not Distu	
Remote Control		Target On Code					incoming calls.	ones to ignore
Phone Lock		Off Code					DND Mode Phone: DND 1 for the IP pho	eature is effectiv
ACD		No Answer Forv		⊙ On				feature can be
SMS		After Ring Tin Target	ne(U~12Us)	12	•		accounts.	
Action URL		On Code					You can cl more guides.	ick here to get
Power LED		Off Code						
Notification Popups		DND						
		DND Emergency		Disabled	-			
		DND Authorized I	Numbers	⊙ On . Off				
		DND Status		Un On				

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 (\mathbf{r}, \mathbf{t}) when the phone is idle.
- Press Menu->Features->Call Forward.

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

Press (d) or (b), 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:						
105	1058						
1058							
Send	123	Delete	Cancel				

3. Press (), #m, 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:
 - 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 31 for more information).
 - Press the **Dir** soft key, and then select **History**. Select the desired list and then press a or v to select the entry (Directory should be configured in advance. Refer to Directory on page 31 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 31 and Remote Phone Book on page 44 for more information).

- **3.** Press (\checkmark) 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 31 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 31 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 31 and Remote Phone Book on page 44 for more information).
- **3.** Press (\checkmark) or # to dial out.
- 4. After the party answers the call, press (c) or the **Tran** soft key to complete the transfer.

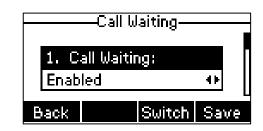
If you are using a handset, the transfer can be completed by hanging up the handset. You can cancel the transfer before the call is connected by pressing the **Cancel** soft key.

Call Waiting

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

To configure call waiting via phone user interface:

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



- **3.** Press (4) or (\triangleright), 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.
- 5. Press the Save soft key to accept the change or the Back soft key to cancel.

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

Conference

You can create a conference with 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 SIPT19(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.

Enter the number of the second party and then 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 (4) or (b) 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 \bigcirc or \bigcirc to select the desired party:
 - Press the **FarMute** soft key to mute the party. 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, all other participants can hear each other, but they cannot hear you.
- 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.

Enter the conference URI (e.g., conference@example.com) in the Conference URI field.

Ve elinte			Log Out
Yealink T19 E2	Status Account Network	DSSKey Features Settings	Directory Security
Register Basic	Keep Alive Type Keep Alive Interval(Seconds) RPort	Default • 30 Disabled •	NOTE DTMF It is the signal sent from the IP
Codec	Subscribe Period(Seconds) DTMF Type	1800 RFC2833 -	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
	SIP Send MAC SIP Send Line	Disabled	session is still active.
	SIP Registration Retry Timer(0~1800s) Conference Type	30 Network Conference	Busy Lamp Field/BLF List Monitors a specific extension/a list of extensions for status
	Conference URI	conference@example.com	changes on IP phones.
	VQ RTCP-XR Collector name VQ RTCP-XR Collector address VQ RTCP-XR Collector port Confirm	5060	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.

To set up a network conference call:

- 1. Place a call to the first party.
- 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 then 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 (\checkmark) , $(\#_{\infty})$, 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.
- 9. 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 Park

You can use call park feature to place a call on hold, and then retrieve the call from another phone in the system (for example, a phone in another office or conference room). You can park an active call by pressing the **Park** soft key. If the call is parked successfully, there is a voice prompt confirming that the call was parked. You can retrieve the parked call by pressing the **Retrieve** soft key. If the parked call is not retrieved within a period of time defined by the system, the phone performing call park will receive a call back.

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

The IP phone supports call park feature under the following modes:

- FAC mode (dial the call park code to park the call to local extension or a desired extension)
- Transfer mode (park the call to directional parking lot directly)

You may need to configure the call park code/call park number or park retrieve code/park retrieval number before using call park feature.

Note The call park code/call park number and park retrieve code/park retrieval number are predefined on the system server. Contact your system administrator for more information.

FAC Mode

To configure call park feature in FAC mode via web user interface:

- 1. Click on Features->Call Pickup.
- 2. Select FAC from the pull-down list of Call Park Mode.
- 3. Select **Enabled** from the pull-down list of **Call Park**.

If **Enabled** is selected, the **Park** soft key will display on the LCD screen during a call, and the **Retrieve** soft key will display on the dialing screen.

- (Optional.) Enter the call park code in the Call Park Code field. It is configured for the Park soft key.
- 5. (Optional.) Enter the park retrieve code in the Park Retrieve Code field.

It is configured for the Retrieve soft key.

6 1 - I - I			Log Out
ealink 119 E2	Status Account Network	DSSKey Features	Settings Directory Security
Forward&DND	Call Pickup		NOTE
General	Directed Call Pickup	Disabled 👻	Disasted Call Dislam
Information	Directed Call Pickup Code		Directed Call Pickup Picks up an incoming call on a
Audio	Group Call Pickup	Disabled 👻	specific extension.
Audio	Group Call Pickup Code		Directed Call Pickup Picks up incoming calls within a
Intercom	Call Park		pre-defined group.
Transfer	Call Park Mode	FAC 💌	You can configure
Call Pickup	Call Park	Enabled 🔹	directed/group call pickup feature for the IP phone.
Сан Ріскир	Call Park Code	*68	Visual Alert for BLF Pickup
Remote Control	Park Retrieve Code	*88	It allows the supervisor's phone to display a visual prompt when
Phone Lock			the monitored user receives an incoming call.
ACD	Confirm	Cancel	
ACD			Audio Alert for BLF Pickup

6. Click Confirm to accept the change.

Note If the Park or Retrieve soft key doesn't appear on the LCD screen, please select Disabled from the pull-down list of Custom Softkey via web user interface at path Settings->Softkey Layout.

To park a call in FAC mode:

- 1. During a call, press the **Park** soft key (You may need to press the **More** soft key to see the **Park** soft key).
 - If the call park code is not configured, you need to enter the call park code.

3 D	elete (Cancel
	3 D	3 Delete (

Press \checkmark , $\#_{\text{sevo}}$ or the **Park** soft key.

 If the call park code is configured, the phone will dial the configured call park code shown as below:

🔩 Talk	king						
*68							
00:03							
Tran	Hold	NewCall	More				

- 2. Do one of the following:
 - a) If you want to park the call against the local extension.

1) Press **#** SEND .

If the call is parked successfully, you will hear a voice prompt confirming that the call is parked.

- b) If you want to park the call against desired extension.
 - 1) Enter an extension where you want to park the call.
 - 2) Press (✓) or (#₅€№).

If the call is parked successfully, you will hear a voice prompt confirming that the call is parked. The call is parked against the extension you entered.

To retrieve a parked call in FAC mode:

- 1. Do one of the following:
 - If the park retrieve code is not configured, dial the park retrieve code (e.g., *88).
 - If the park retrieve code is configured, press the **Retrieve** soft key on the dialing screen.

The phone will dial the configured park retrieve code and the Retrieve screen appears as below:



- 2. Follow the voice prompt, do one of the following:
 - Press #second on the phone where the call is parked.
 - Enter the desired extension follow by # (e.g., 4603#) on any phone.

Transfer Mode

To configure call park feature in transfer mode via web user interface:

- 1. Click on Features->Call Pickup.
- 2. Select Transfer from the pull-down list of Call Park Mode.
- 3. Select Enabled from the pull-down list of Call Park.

If **Enabled** is selected, the **Park** soft key will display on the LCD screen during a call, and the **Retrieve** soft key will display on the dialing screen.

4. (Optional.) Enter the call park number in the Call Park Code field.It is configured for the Park soft key.

5. (Optional.) Enter the park retrieval number in the Park Retrieve Code field.

It is configured for the **Retrieve** soft key.

ealink 119 E2			
	Status Account Network	DSSKey Features	Settings Directory Security
Forward&DND	Call Pickup		NOTE
General Information	Directed Call Pickup Directed Call Pickup Code	Disabled -	Directed Call Pickup Picks up an incoming call on a
Audio	Group Call Pickup Group Call Pickup Code	Disabled 👻	specific extension. Directed Call Pickup Picks up incoming calls within a
Intercom	Call Park		pre-defined group.
Transfer	Call Park Mode	Transfer 👻	You can configure directed/group call pickup
Call Pickup	Call Park	Enabled 👻	feature for the IP phone.
Remote Control	Call Park Code	*01	Visual Alert for BLF Pickup It allows the supervisor's phone
Phone Lock	Park Retrieve Code	*11	to display a visual prompt when the monitored user receives an incoming call.
	Confirm	Cancel	inconning cail.

- 6. Click Confirm to accept the change.
- Note If the Park or Retrieve soft key doesn't appear on the LCD screen, please select Disabled from the pull-down list of Custom Softkey via web user interface at path Settings->Softkey Layout.

To park a call in Transfer mode:

- During a call, press the Park soft key (You may need to press the More soft key to see the Park soft key).
 - If the call park number is not configured, you need to enter the call park number (e.g., *01).

📞 Park to:							
*01							
Park	123	Delete	Cancel				

Press \checkmark , $\#_{mo}$ or the **Park** soft key. The call will be transferred to the directional parking lot where the call park number directed to.

 If the call park number is configured, the call will be directly transferred to the directional parking lot where the call park number directed to.

To retrieve a parked call in Transfer mode:

- 1. Do one of the following:
 - If the park retrieval number is not configured, you need to enter the park retrieval number (e.g., *11).

€ 101;★11 	1		
Send	123	Delete	Cancel

Press (\checkmark) , $[\#_{sevo}]$ or the **Send** soft key.

 If the park retrieval number is configured, press the **Retrieve** soft key on the dialing screen.

The phone will retrieve the parked call from the directional parking lot where the park retrieval number directed to.

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:

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

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

ealink 119 E2	Status Account Netwo	rk DSSKey Feat	tures Settings	Log 0 Directory Security
Forward&DND	Call Pickup			ΝΟΤΕ
General Information	Directed Call Pickup Directed Call Pickup Code	Enabled *97		Directed Call Pickup Picks up an incoming call on a
Audio	Group Call Pickup Group Call Pickup Code	Disabled	•	specific extension. Directed Call Pickup Picks up incoming calls within a
Intercom	Call Park			pre-defined group.
Transfer	Call Park Mode	Transfer	•	You can configure directed/group call pickup
Call Pickup	Call Park	Disabled	•	feature for the IP phone.
Remote Control	Call Park Code Park Retrieve Code			Visual Alert for BLF Pickup It allows the supervisor's phone to display a visual prompt when
Phone Lock	Confirm	Cancel		the monitored user receives an incoming call.

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

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

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

Yealink 119 E2						Log Out
	Status Account	Network	DSSKey	Features	Settings	Directory Security
Register	Keep Alive Type		Default	•		NOTE
Basic	Keep Alive Interval(S	econds)	30			DTMF
	RPort		Disabled	•		It is the signal sent from the IP phone to the network, which is
Codec	Subscribe Period(Sec	onds)	1800			generated when pressing the IP
Advanced	DTMF Type		RFC2833	•		phone's keypad during a call.
			:			Session Timer It allows a periodic refresh of SIP sessions through a re-IINVITE request, to determine whether a SIP session is still active.
	SIP Server Type		Default	•		session is still active.
	Music Server URI					Busy Lamp Field/BLF List
	Directed Call Pickup (Code	*97			Monitors a specific extension/a list of extensions for status
	Group Call Pickup Co	le				changes on IP phones.
	Distinctive Ring Tone	S	Enabled	•		alternation in the second second
	Unregister When Rel	poot	Disabled	•		Shared Call Appearance (SCA)/ Bridge Line
	VQ RTCP-XR Collecto	r name				Appearance (BLA) It allows users to share a SIP
	VQ RTCP-XR Collecto	r address				line on several IP phones. Any IP phone can be used to
	VQ RTCP-XR Collecto	r port	5060			originate or receive calls on the shared line.
	Con	ıfirm		Cancel		

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

To pick up a call directly:

 Pick up the handset or press the Speakerphone key (You may need to press the More soft key to see the DPickup soft key). The **DPickup** soft key appears on the LCD screen.

\$ 1025:	
1058	
Cancel DPickup	More

- 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.

Yealink 119 E2	Status Account Network	DSSKey Features	Log Out Settings Directory Security
Forward&DND General	Call Pickup Directed Call Pickup	Disabled 🗸	NOTE
Information	Directed Call Pickup Code Group Call Pickup	Enabled -	Picks up an incoming call on a specific extension.
Intercom	Group Call Pickup Code Call Park Call Park	*98	Picks up incoming calls within a pre-defined group. You can configure
Call Pickup	Call Park Call Park Call Park Code	Disabled -	directed/group call pickup feature for the IP phone. Visual Alert for BLF Pickup
Remote Control Phone Lock	Park Retrieve Code	Cancel	It allows the supervisor's phone to display a visual prompt when the monitored user receives an incoming call.

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

	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
Register	Kee	p Alive Type		Default	•		NOTE	
Basic	Kee	p Alive Interval(See	conds)	30			DTME	
Basic	RPo	nt		Disabled	•		It is the signal	
Codec	Sub	scribe Period(Seco	nds)	1800			generated wh	network, which en pressing the
Advanced	DTM	ИГ Туре		RFC2833	•		phone's keypa	d during a call.
	Mus	Server Type sic Server URI ected Call Pickup Co	de	Default	•		re-INVITE requ determine who session is still a Busy Lamp Fi Monitors a spe	ether a SIP ctive.
		up Call Pickup Code		*98			list of extensio changes on IP	ns for status
	Dist	inctive Ring Tones		Enabled	•			
	Unr	egister When Rebo	ot	Disabled	•		Shared Call A (SCA)/ Bridg	e Line
	VQ	RTCP-XR Collector	name				Appearance (It allows users	
	VQ	RTCP-XR Collector	address				line on several IP phone can l	IP phones. Any
		RTCP-XR Collector		5060			originate or re	

3. Click Confirm to accept the change.

To pick up a call in the group:

1. Pick up the handset or press the Speakerphone key.

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

% 102!	5:	
1058		
Dic	123	GPickup Cancel

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.
- Press (1) or (1), or the Switch soft key to select Enabled from the Local Anonymous field.

	Anonymous	Call
1. Lo	ocal Anonym	10US:
Enab	led	•
Back	Swi	tch Save

(Optional.) Press (1) or (b) 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.

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

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

To place an anonymous call:

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

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



Anonymous Call Rejection

You can use anonymous call rejection to reject incoming calls from anonymous callers. Anonymous call rejection automatically rejects incoming calls from callers who deliberately block their identities and numbers from 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:

- 1. Press Menu->Features->Anonymous Call.
- **2.** Press (\triangle) or (\neg) to scroll to the **Anonymous Rejection** field.
- 3. Press (a) or (b), or the Switch soft key to select Enabled from the Anonymous Rejection field.

	Anonymous	s Call
	nonymous	Rejectior
Enab	led	<u>•</u>
Back	St	uitch Save

4. (Optional.) Press (1) or (b) , 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.

	Status	Account	Network	DSSKey	Features	Settings	Directory Securi
rogramable Key	Key	Туре	Line	Value	Label	Extension	NOTE
,	SoftKey 1	History 🗸	Local History 🗸				
	SoftKey 2	Directory V	N/A 🗸				Programmable Keys Customizes the soft keys,
	SoftKey 3	DND	N/A 🗸				navigation keys and functio keys.
	SoftKey 4	Hot Desking 🗸	N/A 🗸				
	Up	Hot Desking 🗸	N/A 🗸				You can click here to o more guides.
	Down	Directory V	N/A 🗸				
	Left	N/A 🗸	N/A 🗸				
	Right	N/A 🗸	N/A 🗸				
	ОК	Status 🗸	N/A 🗸				
	Mute	N/A 🗸	N/A 🗸				
	Tran	Forward	N/A 🗸				

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

4. 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:

& 1025					
ง๛๛๛๛๛๛Warning๛๛๛๛๛					
Clear all account config?					
Cancel			OK		

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:

Set Hot Desking	
1. User Name:	
Cancel 2aB Delete Sa	ve

- 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					
Intercom ione	tone when it receives an incoming intercom call.					
	Enable or disable the IP phone to automatically					
Intercom Barge	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

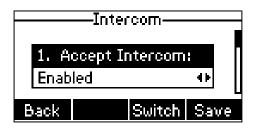
You can enable or disable the phone to play a warning tone when receiving an intercom call. If Intercom Tone is enabled, the phone will play a warning tone before answering the intercom call. If Intercom Tone is disabled, the phone will automatically answer the intercom call without warning. Intercom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

	Status	Account	Network	DSSKey	Features	Settings	Directory Security
Programable Key	Кеу	Туре	Line	Value	Label	Extension	NOTE
	SoftKey 1	History 🗸	Local History 🗸				
	SoftKey 2	Directory V	N/A 🗸				Programmable Keys Customizes the soft keys,
	SoftKey 3	DND V	N/A 🗸				navigation keys and function keys.
	SoftKey 4	Paging List 🗸	N/A 🗸				
	Up	History 🗸	Local History 🗸				You can click here to get more guides.
	Down	Directory V	N/A 🗸				
	Left	N/A 🗸	N/A 🗸				
	Right	N/A 🗸	N/A 🗸				
	ОК	Status 🗸	N/A 🗸				
	Mute	N/A V	N/A 🗸				
	Tran	Forward V	N/A 🗸				

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 (\bigcirc) or (\bigtriangledown) to select a desired paging group.

The default tag is Empty if it is not configured before.

	—Pagin	g List—	
1. (Emp	oty)		
2. (Emp			
3. (Emp	oty)		Ц
Back		Option	Paging

- 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.

Edi	t Pagir	ng Addre:	ss						
2. Address:									
224.5	.6.20:1	.0008							
Back	123	Delete	Save						

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

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

To delete a paging group via phone user interface:

- 1. Press the paging list key when the phone is idle.
- **2.** Press (\triangle) or (\bigtriangledown) to select a desired group.
- 3. Press the **Option** soft key and then press the **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.

	Status	Account	Network	DSSKey	Features	Settings	Directory Security	
Local Directory	Multicast List	tening					NOTE	
Remote Phone		Paging Barge		10			Multicast Paging	
Book		Paging Priority A	ctive Enabled		nabled		Multicast paging allows IP phones to send/receive Real-	
Phone Call Info	IP Address		Listening Ad	ldress	Label	Priorit,	time Transport Protocol (RTP) streams to/from the pre- configured multicast address(e	
Multicast IP	1 IP	1 IP Address		20:10008 Sale		1	without involving SIP signaling Up to 10 listening multicast	
etting	2 IP	Address				2	addresses can be specified on the IP phone.	
botting	3 IP	Address				3	You can click here to get	
	4 IP	Address				4	more guides.	
	5 IP	Address				5		
	6 IP	Address				6		
	7 IP	Address				7		
	8 IP	Address				8		
	9 IP	Address				9		
	10.10	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 (\triangle) or (∇) to select the desired paging group.
- **3.** Press (\checkmark) 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.
- 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.

alink 119 E	Status	Account	Network	DSSKey	Features	Settings	Directory	Security	
Register	Keep	Alive Type		Default	•		NOTE		
	Keep	Keep Alive Interval(Seconds)		30					
Basic	RPort	:		Disabled	•		DTMF It is the signal sent from the IF		
Codec	Subs	Subscribe Period(Seconds)		1800		phone to the network, which generated when pressing the			
Advanced	DTM	- Туре		RFC2833	•		phone's keypa	d during a call.	
				• Disabled			Session Timer It allows a perior SIP sessions th re-INVITE required determine whe	odic refresh of rough a lest, to other a SIP	
	Early	Early Media			*		session is still active.		
	SIP S	Server Type		Default	•		Busy Lamp Field/BLF List		
	Music	Music Server URI			sip:moh@sip.com			cific extension/a	
	Direc	ted Call Pickup Co	ode				changes on IP		
	Grou	p Call Pickup Code	e						
	Distin	ctive Ring Tones		Enabled	•		Shared Call A (SCA)/ Bridge		
	Unre	gister When Rebo	oot	Disabled	•		Appearance (It allows users		
	VQ R	TCP-XR Collector	name				line on several		
	VQ R	VQ RTCP-XR Collector address					IP phone can be used to originate or receive calls on th shared line.		
	VO R	TCP-XR Collector	port	5060			snared iné.		

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

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

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

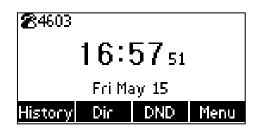
Note

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.

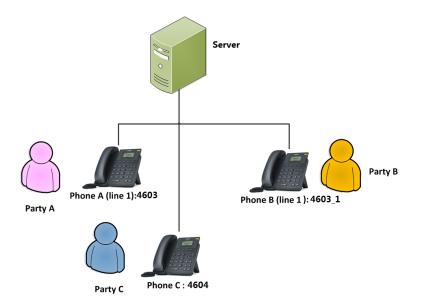
In the following figure, the registered line is shared:



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:

- Log Out Yealink 119 E2 Status Account Network DSSKey Features Settings Directory Security Register Status Registered NOTE Register Line Active Enabled Account Registration Registers account(s) for the IP phone. Basic Label 4603 Codec 4603 Display Name Server Redundancy It is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offine for maintenance, the server fails, or the connection between the IP phone and the server fails. 4603 Register Name Advanced User Name 4603 Password SIP Server 1 Server Host pbx.vealink.com Port 5060 NAT Traversal A general term for techniques that establish and maintain IP connections traversing NAT gateways. STUN is one of the NAT traversal techniques. Transport UDF Server Expires 3600 Server Retry Counts 3 SIP Server 2 Server Host Port 5060 You can configure NAT traversal for this account. Transport UDP You can click here to get more guides. Server Expires 3600 Server Retry Counts 3 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 . Confirm Cancel
- 1. Register the account 4603.

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

ealink 119 E2		_					Log Oı	
C CHI II (119 E 2	Status Account	Network	DSSKey	Features	Settings	Directory	Security	
Register	Keep Alive Type		Default	•		NOTE	l	
Basic	Keep Alive Interval(S	Seconds)	30			DTME		
Dasic	RPort		Disabled	•		It is the signal s	sent from the IP	
Codec	Subscribe Period(Se	conds)	1800			generated whe	network, which i en pressing the I	
Advanced	DTMF Type		RFC2833	•		phone's keypad	d during a call.	
	PTime(ms)		20	•		Session Time It allows a per SIP sessions t re-INVITE req determine wh session is still :		
	Shared Line Call Pull Feature Acc	acc Codo	Shared Call Appe	Shared Call Appearance 🔻			eld/BLF List cific extension/a	
	Group Call Pickup Co					list of extension changes on IP		
	Distinctive Ring Ton	es	Enabled	•		Shared Call A		
	Unregister When Re	boot	Disabled	•		(SCA)/ Bridge	Line	
	VQ RTCP-XR Collect	or name				Appearance (It allows users	to share a SIP	
	VQ RTCP-XR Collect	or address				line on several IP phone can b	IP phones. Any oe used to	
	VQ RTCP-XR Collect	or port	5060			originate or rec shared line.	eive calls on the	
	Co	nfirm	Γ	Cancel				

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

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

1. Register the alternate account 4603_1.

(Enter the primary account 4603 in the Register Name field.)

ealink Inse				Log (
	Status Account	Network DSSKey	Features Settings	Directory Security		
Register	Register Status	Registered		NOTE		
Basic	Line Active	Enabled	•	Account Registration		
Dasic	Label	4603_1		Registers account(s) for the 1		
Codec	Display Name	4603_1		phone.		
Advanced	Register Name	4603		Server Redundancy It is often required in VoIP		
	User Name	4603_1		deployments to ensure continuity of phone service, f		
	Password	•••••		events where the server need to be taken offline for		
	SIP Server 1			maintenance, the server fails,		
	Server Host	pbx.yealink.com	Port 5060	the connection between the phone and the server fails.		
	Transport	UDP	*	NAT Traversal		
	Server Expires	3600		A general term for technique that establish and maintain IP		
	Server Retry Counts	3		connections traversing NAT gateways. STUN is one of the		
	SIP Server 2			NAT traversal techniques.		
	Server Host		Port 5060	You can configure NAT trave		
	Transport	UDP		for this account.		
	Server Expires	3600		You can click here to get		
	Server Retry Counts	3		more guides.		
	Enable Outbound Proxy Se	erver Enabled	-			
	Outbound Proxy Server 1	10.1.8.11	Port 5060			
	Outbound Proxy Server 2		Port 5060			
	Proxy Falback Interval	3600				
	NAT	Disabled	•			
	Confirm		ancel			

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

ealink 119 E2								Log Oı	
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security	
Register	Keep	p Alive Type		Default	•		NOTE		
	Keep	p Alive Interval(Sec	conds)	30					
Basic	RPort Subscribe Period(Seconds)			Disabled	•		DTMF It is the signal sent from the IP		
Codec				1800				phone to the network, which is generated when pressing the II	
Advanced	DTM	IF Type		RFC2833	•		phone's keypa		
	PTin	PTime(ms)		20	¥		Session Timer It allows a periodic n SIP sessions througi re-INVITE request, determine whether session is still active.		
	Shar	red Line		Shared Call Appearance 🔻			Busy Lamp Field/BLF List		
	Call F	Pull Feature Access	Code				Monitors a spe list of extensio	cific extension/a	
	Grou	up Call Pickup Code					changes on IP		
	Disti	inctive Ring Tones		Enabled	•		Channel C. II.		
	Unre	egister When Rebo	ot	Disabled	•		Shared Call A (SCA)/ Bridg	e Line	
	VQ I	RTCP-XR Collector	name					to share a SIP	
	VQ I	RTCP-XR Collector	address				line on several IP phone can	IP phones. Any be used to	
	VQ I	RTCP-XR Collector	port	5060			originate or re shared line.	ceive calls on the	
		Confi	rm	Γ	Cancel				

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

Configuring call pull feature

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

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

1. Click on Account->Advanced.

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

	Status	Account	Network	DSSKey	Features	Settings	Directory Security		
Register	Kee	p Alive Type		Default 👻			NOTE		
	Kee	Keep Alive Interval(Seconds)		30					
Basic	RPo	rt		Disabled	•		DTMF It is the signal sent from the IF		
Codec	Subscribe Period(Seconds)			1800			phone to the network, which is generated when pressing the IF phone's keypad during a call.		
Advanced	DTMF Type			RFC2833 -					
		PTime(ms) Shared Line					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		
		Pull Feature Access	s Code	Shared Call Appearance V					
	Gro	up Call Pickup Code	e				list of extensions for status changes on IP phones.		
	Dist	inctive Ring Tones		Enabled	•		Shared Call Appearance		
	Unr	egister When Rebo	pot	Disabled	•		(SCA)/ Bridge Line		
	VQ	RTCP-XR Collector	name				Appearance (BLA) It allows users to share a SIP		
	VQ	VQ RTCP-XR Collector address VQ RTCP-XR Collector port					line on several IP phones. Any IP phone can be used to		
	VO					originate or receive calls on th shared line.			

3. 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:

- 1. Enter the desired number using the keypad when the phone is idle.
- **2.** Press (\checkmark) , $(\#_{\text{seed}})$, 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 dialing screen.
- 3. Enter the desired number using the keypad.
- 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.

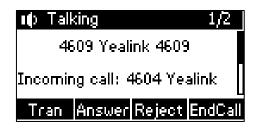
Do one of the following on phone A or phone B:

- Press), or the Answer soft key on phone A.
 Phone B stops ringing.
- Press \bigcirc , (\checkmark) 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 83.
```

Do one of the following on phone A:

- Press the Answer soft key. Phone B stops ringing.
- Press (\neg) to access the new call.

Press (\checkmark) or the **Answer** soft key. Phone B stops 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 (\checkmark) or the **Answer** soft key. Phone A stops ringing.

Placing a Call on Hold

To place a call on hold:

 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.

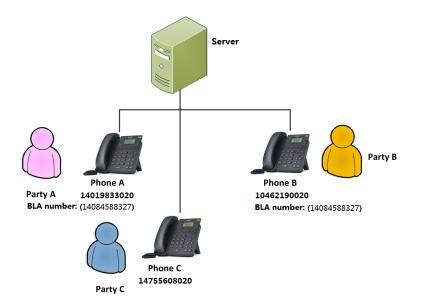


In the following figure, the registered account is shared:

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

ealink Ing	Status	Account	etwork DSSKey	Featu	ires	Settings	Directory	Security	
Register	Re	gister Status	Registered				NOTE		
Basic	Lin	e Active	Enabled	•			Account Reg		
DdSK	Lat	oel	14019833020				Registers acco	unt(s) for the I	
Codec	Dis	play Name	14019833020				phone.		
Advanced	Re	gister Name	14019833020				Server Redundancy It is often required in VoIP		
	Usi	er Name	14084588327	14084588327			deployments to ensure continuity of phone service, f		
	Pas	sword					events where the server nee to be taken offine for		
	SI	P Server 1					maintenance,	the server fails, n between the	
	Ser	ver Host	sip.ringcentral.co	m	Port 506	0	phone and the		
	Tra	insport	UDP	•			NAT Traversal A general term for technique		
	Ser	ver Expires	3600	3600			that establish and		
	Sei	ver Retry Counts	3	3			connections traversing NAT gateways. STUN is one of the		
	SI	P Server 2		Port 5060			NAT traversal techniques. You can configure NAT trave		
	Ser	ver Host							
	Tra	insport	UDP	•			for this account.		
	Sei	ver Expires	3600					lick here to get	
	Sei	ver Retry Counts	3	3			more guides.		
	En	able Outbound Proxy Serv	er Enabled	•	1				
	Ou	tbound Proxy Server 1	sip114.ringcentr	al.com	Port 509	9			
	Ou	Outbound Proxy Server 2				0			
	Pro	xy Falback Interval	3600						
	NA	т	Disabled						

1. Register the account 14019833020.

2. Click on Advanced, and then select Draft BLA from the pull-down list of Shared Line.

Yealink 119 E2								Log Out	
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security	
Register	Kee	p Alive Type		Default	•		NOTE		
Basic	Kee	p Alive Interval(Seo	conds)	30 Disabled			DTMF It is the signal sent from the IP		
Codec	Subscribe Period(Seconds) DTMF Type			1800			phone to the network, which generated when pressing the phone's keypad during a call.		
Advanced	DTI	ИЕ Туре		RFC2833	•			-	
				:			Session Timer It allows a periodic refresh of SIP sessions through a re-INVITE request, to determine whether a SIP		
		me(ms)		20	•		Busy Lamp Field/BLF List		
		red Line Number		Draft BLA 14084588327	-				
	BLA	Subscription Perio	ł	300			list of extensio changes on IP		
		Send MAC Send Line		Disabled	~		Shared Call A	opearance	
		register When Rebo	ot	Disabled	•		(SCA)/ Bridg Appearance	(BLA)	
	VQ RTCP-XR Collector name		name				It allows users to share a SIP line on several IP phones. Any IP phone can be used to		
	VQ RTCP-XR Collector address VQ RTCP-XR Collector port			5060		originate or receive calls on the shared line.			
		Conf	-		Cancel				

3. Enter the desired number in the **BLA Number** field.

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

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

1. Register the account 10462190020.

ealink 119 E2 /				_	-	_	_	Log Out
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
Register	Reg	jister Status		Registered			NOTE	
Basic	Line	e Active		Enabled	•		1	
Basic	Lab	el		10462190020				unt(s) for the IP
Codec	Disp	olay Name		10462190020			phone.	
Advanced	Reg	jister Name		10462190020			Server Redu It is often req	
	Use	er Name		14084588327			deployments t	
	Pas	sword		•••••				the server needs
	SIF	Server 1					maintenance,	the server fails, or
	Ser	ver Host		sip.ringcentral.co	m Po	rt 5060	phone and th	n between the IP e server fails.
	Tra	nsport		UDP	•		NAT Traversa	
	Ser	ver Expires		3600			that establish	n for techniques and maintain IP
	Ser	ver Retry Counts		3			connections t gateways. ST	aversing NAT UN is one of the
	SIF	Server 2					NAT traversal	techniques.
	Ser	ver Host			Po	rt 5060	You can confi	gure NAT traversal
	Tra	nsport		UDP	•		for this account	
	Ser	ver Expires		3600			💾 You can d	ick here to get
	Ser	ver Retry Counts		3			more guides.	
	Ena	ble Outbound Prox	/ Server	Enabled	•			
	Out	tbound Proxy Serve	r 1	sip214.ringcentra	l.com Po	rt 5099		
	Out	tbound Proxy Serve	r 2		Po	rt 5060		
	Pro	xy Fallback Interval		3600				
	NA	т		Disabled	-			
		Confi	m		Cancel			

- 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.

ealink 119 E2								Log O
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
Register	Kee	p Alive Type		Default	•		NOTE	
Basic	Kee RPo	p Alive Interval(Seo rt	onds)	30 Disabled	•		DTMF It is the signal	sent from the I
Codec	Sub	scribe Period(Secor	nds)	1800			phone to the generated wh	network, which en pressing the
Advanced	DTM	IF Type		RFC2833	-		phone's keypa	d during a call.
				:			Session Time It allows a per SIP sessions th re-INVITE req determine wh	iodic refresh of nrough a uest, to
	PTir	me(ms)		20	•		session is still a	
		red Line Number		Draft BLA 14084588327	-		Busy Lamp F Monitors a spe	ield/BLF List
	BLA	Subscription Period	i	300			list of extension changes on IP	
	SIP	Send MAC		Disabled	-			
	SIP	Send Line		Disabled	•		Shared Call / (SCA)/ Bridg	e Line
	Unr	egister When Rebo	ot	Disabled	•		Appearance It allows users	(BLA) to share a SIP
	VQ	RTCP-XR Collector	name				line on several IP phone can	IP phones. Any be used to
	VQ	RTCP-XR Collector	address				originate or re shared line.	ceive calls on th
	VQ	RTCP-XR Collector	port	5060				
		Confi	rm		Cancel			

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 (\mathscr{A}) , $(\texttt{\#}_{sew})$, 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 dialing screen.
- 3. Enter the desired number using the keypad.
- 4. Press (\checkmark) , $\#_{sev}$, 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:

 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:

1. 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. The power indicator LED will slow flash yellow, and the LCD screen will prompt "n New Text Message(s)" ("n" indicates the number of unread text messages. e.g., 1 New Text Message(s)) and a flashing icon $\boxed{\sim}$.

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.

	——Int	00×	
B	056		
Today	y 16:18		
Back	Reply	Delete	View

2. Select the desired message and then 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.

Hi	-New M	lessage—	
Back	abc	Delete	Send

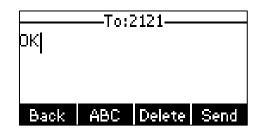
- 3. Press the Send soft key after completing the content.
- 4. Enter the number you want to send the message to in the To field.
- 5. Press the Send soft key to send the message or the Back soft key to cancel.

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 then press the Reply soft key.

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



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

To delete a text message:

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

Inbox					
Delete					
Delete All					
Cancel	Oł	<			

3. Select **Delete** to delete the desired message, then press the **OK** soft key.

The LCD screen prompts "Delete message?".



4. Press the OK soft key to delete this message or the Cancel soft key to cancel.

You can also delete all text messages by pressing the **Delete** soft key and then select **Delete All**. For more information, refer to the above steps.

Note

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

Voice Mail

You can leave voice mails for someone else using the SIP-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. The power indicator LED will slow flash yellow, and the LCD screen will prompt "n New Voice Mail(s)" ("n" indicates the number of

unread voice messages. e.g., 4 New Voice Mail(s)) and a flashing icon oo.

2 104			60
	21:	1225	
4 N	lew Voic	e Mail(s)
Exit		View	Connect

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:

 When the LCD screen prompts that the phone receives a new voice mail and the power indicator LED slow flashes yellow, you can press or the Connect soft key to dial out the voice mail access code.

- 2. Follow the voice prompt to listen to your voice mails.
- Note Before listening to voice mails, make sure the voice mail access code has been configured.

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

To view the voice mail via phone user interface:

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

	View Voice Mail						
1.	104	4 New i	2 Old	Voice	Mail		
B.	ack			Co	nnec:		

3. 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 yellow. 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.

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.

The MWI subscription parameters you need to know:

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.

Yealink 119 E2			Log Out
	Status Account Network	DSSKey Features Settings	Directory Security
Register	Keep Alive Type	Default 🔹	NOTE
Basic	Keep Alive Interval(Seconds)	30	DTMF
Codec	RPort Subscribe Period(Seconds)	Disabled -	It is the signal sent from the IP phone to the network, which is generated when pressing the IP
Advanced	DTMF Type	RFC2833 -	phone's keypad during a call.
	DTMF Info Type	DTMF-Relay 👻	Session Timer
	DTMF Payload Type(96~127)	101	It allows a periodic refresh of SIP sessions through a re-INVITE request, to
	Retransmission Subscribe Register	Disabled	determine whether a SIP session is still active.
	Subscribe for MWI	Enabled -	
	MWI Subscription Period(Seconds)	3600	Busy Lamp Field/BLF List Monitors a specific extension/a
	Subscribe MWI To Voice Mail	Disabled 👻	list of extensions for status changes on IP phones.
	Voice Mail	T-abled	Shared Call Appearance
	Voice Mail Display Caller ID Source	Enabled	(SCA)/ Bridge Line Appearance (BLA)
	Session Timer	Disabled	It allows users to share a SIP line on several IP phones. Any
	Session Expires(30~7200s)	1800	IP phone can be used to originate or receive calls on the shared line.
	Session Refresher	UAC 🗸	Silaicu mic.
	Send user=phone	Disabled	Network Conference It allows multiple participants
	RTP Encryption(SRTP) PTime(ms)	Disabled	(more than three) to join in a call.

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.

4. Enter the desired voice mail number in the Voice Mail field.

alink	E2 Status	Account	Network	DSSKey	Features	Settings	Directory	ty	
Decistor	Keep	Alive Type		Default	•		NOTE		
Register	Keer	Alive Interval(Se	conds)	30					
Basic	RPor	t		Disabled	•		DTMF It is the signal sent from th	e Ti	
Codec	Subs	cribe Period(Seco	nds)	1800			phone to the network, whi generated when pressing t	ich	
Advanced	DTM	F Туре		RFC2833	•		phone's keypad during a ca		
avancea	DTM	F Info Type		DTMF-Relay	~				
	DTM	F Payload Type(9	6~127)	101			Session Timer It allows a periodic refresh of	of	
	Retr	ansmission		Disabled	•		SIP sessions through a re-INVITE request, to determine whether a SIP session is still active.		
	Subs	cribe Register		Disabled	•				
	Subs	cribe for MWI		Enabled	•				
	MWI	Subscription Peri	od(Seconds)	3600			Busy Lamp Field/BLF List Monitors a specific extension/a		
	Subs	cribe MWI To Voi	ce Mail	Enabled	•		list of extensions for status changes on IP phones.		
	Voic	e Mail		*88			changes on a phones		
	Voic	e Mail Display		Enabled	•		Shared Call Appearance		
	Calle	r ID Source		FROM	•		(SCA)/ Bridge Line Appearance (BLA)		
	Sess	ion Timer		Disabled	•		It allows users to share a SI line on several IP phones. A		
	Sess	ion Expires(30~72	200s)	1800			IP phone can be used to originate or receive calls o		
	Sess	ion Refresher		UAC	•		shared line.		
	Send	d user=phone		Disabled	•		Network Conference	nference	
	RTP	Encryption(SRTP)		Disabled	•		It allows multiple participant (more than three) to join in		
	PTim	ne(ms)		20	•		call.		

5. 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.

General Issues

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

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 required by the server. Display name is the caller ID that will be displayed on the called party's LCD screen. Server configuration may override the local configuration.

Display Issues

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 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, contact your system administrator for more information. You can also configure the time and date manually. For more information, refer to Time & Date on page 23.

Password Issues

Why can't I access web user interface using the default administrator user name and password (admin/admin)?

- It is case-sensitive, ensure the case is correct.
- Ensure the password has not been changed.
- If the password has been changed, but the computer remembers the old password, try to clear the browser cache and try again or select another browser to login.

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.

Yealink 119 E2	Status	Account	etwork DSSKe	/ Features	Settings	Log Out Directory Security
Password		User Type	user	-		NOTE
Trusted Certificates		Old Password New Password	•••••			User Password/ Administrator Password
Server Certificates		Confirm Password	•••••			When logging into the web user interface, you need to enter the user name and password.
		Confirm		Cancel		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.

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

User password is configurable via web user interface only.

Call Issues

Note

Why can't I receive calls?

- Check the SIP registration with your system administrator.
- Check that the DND (Do Not Disturb) mode is deactivated on your phone. Refer to Do Not Disturb (DND) on page 77.
- Check that call forward is disabled on the phone. Refer to Call Forward on page 79.
- Check whether the caller number is stored in the blacklist directory. Refer to Blacklist on page 43.

Headset & Handset Issues

Why does my handset not work?

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 does my headset not work?

 Check that the headset cord is properly connected to the headset jack on the phone. Refer to Phone Installation on page 11.

- Check that the headset mode is activated. Refer to Headset Use on page 50.
- Check that the headset volume is adjusted to an appropriate level. Refer to Volume on page 28.

Audio Issues

Why can't I get a dial tone?

- Check for any loose connections and that the phone has been installed properly. For the installation instructions, refer to Phone Installation on page 11.
- Switch between the Handset, Headset (if present) or Hands-Free Speakerphone to check whether the dial tone is present for one of the audio modes.

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

Why doesn't the phone ring?

Check the ringer volume on the phone. To adjust the ringer volume setting, press the Volume key when the phone is on-hook and idle. For more information, refer to Volume on page 28.

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.

- Log Out Yealink | T19 E2 Network DSSKey Features Settings Directory Security Account Status General Information NOTE Forward&DND Call Waiting Enabled • General Information Call Waiting It allows IP phones to receive a new incoming call when there is already an active call. Call Waiting On Code Call Waiting Off Code Audio Auto Redial Auto Redial It allows IP phones to automatically redial a busy number after the first attempt. Disabled Intercom Transfer Key As Send Assigns "#" or "*" as the send Call Pickup Play Local DTMF Tone Enabled kev **Remote Control** Hotline IP phone will automatically dial out the hotline number when lifting the handset, pressing the speakerphone key or the line DTMF Repetition 3 Phone Lock Play Hold Tone Enabled ÷ ACD Play Hold Tone Delay 30 spea key. Allow Mute Enabled SMS Call Completion It allows users to monitor the busy party and establish a call when the busy party becomes available to receive a call. Voice Mail Tone Enabled . Action URL SIP-T19P_E2 DHCP Hostname Power LED Reboot in Talking Disabled • You can click here to get more guides. Notification Popups Hide Feature Access Codes Disabled User Name Display Method on Dialing Confirm Cancel
- 3. Enter the desired time in the Play Hold Tone Delay field.

4. Click Confirm to accept the change.

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.

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

ealink 119 E2 /							Log Ou	
	Status	Account	Network	DSSKey	Features	Settings	Directory Security	
Register	Kee	ep Alive Type		Default	•		NOTE	
Basic	Keep Alive Interval(Seconds) RPort			30 Disabled			DTMF It is the signal sent from the IP	
Codec	Sub	oscribe Period(Seco	nds)	1800			phone to the network, which is generated when pressing the I	
Advanced		MF Type		RFC2833	-		phone's keypad during a call.	
	DTMF Info Type DTMF Payload Type(96~127)		DTMF-Relay	v		Session Timer It allows a periodic refresh of SIP sessions through a		
	Retransmission Subscribe Register			Disabled	•		re-INVITE request, to determine whether a SIP	
				Disabled	•		session is still active.	
	Subscribe for MWI MWI Subscription Period(Seconds)		Disabled 3600	•		Busy Lamp Field/BLF List Monitors a specific extension/a list of extensions for status		
	Subscribe MWI To Voice Mail Voice Mail			Disabled	-		changes on IP phones.	
		ce Mail Display		Enabled	•		Shared Call Appearance (SCA)/ Bridge Line	
	Caller ID Source Session Timer		FROM	•		Appearance (BLA) It allows users to share a SIP		
			Disabled	•		line on several IP phones. Any IP phone can be used to		
		sion Expires(30~72	00s)	1800			originate or receive calls on the shared line.	
		sion Refresher		UAC	•			
	Send user=phone RTP Encryption(SRTP)			Disabled	•		Network Conference It allows multiple participants	
		me(ms)		20			(more than three) to join in a call.	

3. 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.

Log Issues

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.

Yealink			Log Out
TECHINK T19 E2	Status Account Network	DSSKey Features Settings	Directory Security
Preference	Export or Import Configuration	Browse No file selected.	NOTE
Time & Date		Import Export	Configuration IP phones can provide feedback
Call Display	Export CFG Configuration File	Local Configuratior Export	in a variety of forms such as log files, packets, status indicators
Upgrade	Export of a configuration file		and so on, which can help an administrator more easily find the system problem and fix it.
Auto Provision	Import CFG Configuration File	Browse No file selected.	· Log Files
Configuration		Local Configuratior - Import	 Capturing Packets Configuration File (*,cfg/*,bin)
Dial Plan			
Voice	Pcap Feature	Start Stop Export	You can click here to get more guides.
Ring	Export System Log	Local Server Ftp/Tftp Server	
Tones	System Log Level	Export •	
Softkey Layout	Export All Diagnostic Files	Start Stop Export	
TR069			
Voice Monitoring	Confirm	Cancel	

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.
- 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. Reproduce the issue.

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

Yealink 119 E2								Log Out
	Status	Account	Network	DSSKey	Features	Settings	Directory	Security
Preference	Б	xport or Import Cor	nfiguration	Browse No fil	e selected.		NOTE	
Time & Date				Import	Export		Configuration	
Call Display	5	xport CFG Configura	tion File	Local Configuratior Export			IP phones can provide feedback in a variety of forms such as log files, packets, status indicators	
Upgrade	D	xpore ero comigura	Cion File	Ebcar configuration	 Export 		administrator n	ich can help an nore easily find oblem and fix it.
Auto Provision	In	nport CFG Configura	ation File	Browse No fil	e selected.		· Log Files	
Configuration				Local Configuration	▼ Import		 Capturing Pace Configuration (*.cfg/*.bin) 	
Dial Plan								
Voice		cap Feature			Stop Export		You can cli more guides.	ick here to get
Ring	Ð	xport System Log			r C Ftp/Tftp Serve	er		
Tones				Export				
	Sy	ystem Log Level		6	•			
Softkey Layout	Ð	xport All Diagnostic	Files	Start	Stop Export			
TR069								
Voice Monitoring		Confi	rm		Cancel			

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

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

How to export all diagnostic files?

We may need you to provide three types of diagnostic files (including PCAP trace, system log and BIN configuration file) to help analyze your problem. You can export these files at a time.

To export all diagnostic files via web user interface:

- 1. Click on Settings->Configuration.
- Click Start to begin capturing signal traffic.
 The system log level will be automatically set to 6.
- 3. Reproduce the issue.
- 4. Click **Stop** to stop the capture.

The system log level will be reset to 3.

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

Yealink 119 E2		Log			
	Status Account Network	DSSKey Features Settings	Directory Security		
Preference	Export or Import Configuration	Browse No file selected.	NOTE		
Time & Date		Import Export	Configuration IP phones can provide feedback		
Call Display	Export CFG Configuration File	Local Configuration - Export	in a variety of forms such as log files, packets, status indicators		
Upgrade			and so on, which can help an administrator more easily find the system problem and fix it.		
Auto Provision	Import CFG Configuration File	Browse No file selected.	Log Files		
Configuration		Local Configuratior	 Capturing Packets Configuration File (*.cfg/*.bin) 		
Dial Plan			You can click here to get		
Voice	Pcap Feature	Start Stop Export	more guides.		
Ring	Export System Log	Local Server Ftp/Tftp Server			
Tones		Export			
Softkey Layout	System Log Level	3 🔹			
	Export All Diagnostic Files	Start Stop Export			
TR069					
Voice Monitoring	Confirm	Cancel			

Reboot & Upgrade & Reset Issues

How to reboot the phone?

To reboot the phone via web user interface:

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

Mandal			Log Out
Yealink 119 E2	Status Account Netwo	ork DSSKey Features Set	tings Directory Security
Preference	Version		NOTE
Time & Date	Firmware Version	53.80.0.90	Reset to Factory Setting Resets the IP phone to factory
Call Display	Hardware Version	53.0.0.128.0.0.0	configurations.
Upgrade	Reset to Factory Setting	Reset to Factory Setting	Reboot Reboots the IP phone.
Auto Provision	Reboot	Reboot	Upgrading Firmware
Configuration	Select and Upgrade Firmware	Browse No file selected.	Upgrades firmware manually.
Dial Plan		Upgrade	You can click here to get more guides.

Note

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

Note If the issue cannot be reproduced, just directly click **Export** to export all diagnostic files.

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.

Veglink			Log Out
Yealink T19 E2	Status Account Network	DSSKey Features Settings	Directory Security
Preference	Version		NOTE
Time & Date	Firmware Version	53.80.0.90	Reset to Factory Setting Resets the IP phone to factory
Call Display	Hardware Version	53.0.0.128.0.0.0	configurations.
Upgrade	Reset to Factory Setting	Reset to Factory Setting	Reboots the IP phone.
Auto Provision	Reboot	Reboot	Upgrading Firmware Upgrades firmware manually.
Configuration	Select and Upgrade Firmware	Browse No file selected.	You can click here to get
Dial Plan			more guides.

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:

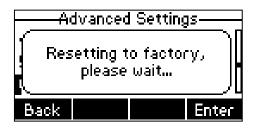
- Press Menu->Settings->Advanced Settings (default password: admin)->Reset to Factory.
- 2. Press the Enter soft key.

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".

Welcome
Initializing please wait

The phone will be reset to factory settings sucessfully 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 AMAGES, AND LIMITS ITS LIABILITY TO REPAIR, REPLACEMENT, OR REFUND OF THE PURCHASE RICE 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.

A 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.

A 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.

A 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	United States-Hawaii-Aleutian, United States-Alaska-Aleutian		
-9:30	French Polynesia		
-9	United States-Alaska Time		
	Canada(Vancouver,Whitehorse), Mexico(Tijuana,Mexicali),		
-8	United States-Pacific Time		
_	Canada(Edmonton,Calgary), Mexico(Mazatlan,Chihuahua),		
-7	United States-MST no DST, United States-Mountain Time		
	Canada-Manitoba(Winnipeg), Chile(Easter Islands),		
-6	Mexico (Mexico City, Acapulco), United States-Central Time		
	Bahamas(Nassau), Canada(Montreal,Ottawa,Quebec),		
-5	Cuba(Havana), United States-Eastern Time		
-4:30	Venezuela(Caracas)		
1.00	Canada(Halifax,Saint John), Chile(Santiago),		
-4	Paraguay(Asuncion), United Kingdom-Bermuda(Bermuda),		
-7	United Kingdom(Falkland Islands), Trinidad&Tobago		
-3:30	Canada-New Foundland (St. Johns)		
	Argentina(Buenos Aires), Brazil(DST), Brazil(no DST),		
-3	Denmark-Greenland(Nuuk)		
-2:30	Newfoundland and Labrador		
-2.30	Brazil(no DST)		
-			
-1	Portugal(Azores)		
0	Denmark-Faroe Islands(Torshavn), GMT, Greenland,		
0	Ireland(Dublin), Morocco, Portugal(Lisboa,Porto,Funchal),		
	Spain-Canary Islands(Las Palmas), United Kingdom(London)		
	Albania(Tirane), Austria(Vienna), Belgium(Brussels), Caicos, Chad, Croatia(Zagreb), Czech Republic(Prague),		
	Denmark(Kopenhagen), France(Paris), Germany(Berlin),		
+1			
	Hungary(Budapest), Italy(Rome), Luxembourg(Luxembourg), Macedonia(Skopje), Namibia(Windhoek),		
	Netherlands(Amsterdam), Spain(Madrid)		
	Estonia(Tallinn), Finland(Helsinki), Gaza Strip(Gaza),		
	Greece(Athens), Israel(Tel Aviv), Jordan(Amman), Latvia(Riga),		
+2	Lebanon(Beirut), Moldova(Kishinev), Romania(Bucharest),		
72	Russia(Kaliningrad), Syria(Damascus), Turkey(Ankara),		
	Ukraine(Kyiv, Odessa)		
+3	East Africa Time, Iraq(Baghdad), Russia(Moscow)		
+3:30	Iran(Teheran) Armenia(Yerevan), Azerbaijan(Baku), Georgia(Tbilisi),		
+4	Kazakhstan(Aktau), Russia(Samara)		
+4:30	Afghanistan(Kabul)		
±4.50	Kazakhstan(Aqtobe), Kyrgyzstan(Bishkek),		
+5			
L E-70	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			
+9	Japan(Tokyo), Korea(Seoul), Russia(Yakutsk,Chita)		

Time Zone	Time Zone Name
+9:30	Australia(Adelaide), Australia(Darwin)
+10	Australia(Brisbane), Australia(Hobart),
+10	Australia(Sydney, Melboume, 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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