Yealink HAAGCOM







SIP-T20P

Enterprise IP Phone User Guide

Copyright

Copyright © 2013 YEALINK NETWORK TECHNOLOGY

Copyright © 2013 Yealink Network Technology CO., LTD. All rights reserved. No parts of this publication may be reproduced or transmitted in any form or by any means, electronic or mechanical, photocopying, recording, or otherwise, for any purpose, without the express written permission of Yealink Network Technology CO., LTD. Under the law, reproducing includes translating into another language or format.

When this publication is made available on media, Yealink Network Technology CO., LTD. gives its consent to downloading and printing copies of the content provided in this file only for private use but not for redistribution. No parts of this publication may be subject to alteration, modification or commercial use. Yealink Network Technology CO., LTD. will not be liable for any damages arising from use of an illegally modified or altered publication.

Warranty

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS GUIDE ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS GUIDE ARE BELIEVED TO BE ACCURATE AND PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF PRODUCTS.

YEALINK NETWORK TECHNOLOGY CO., LTD. MAKES NO WARRANTY OF ANY KIND WITH REGARD TO THIS GUIDE, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. Yealink Network Technology CO., LTD. shall not be liable for errors contained herein nor for incidental or consequential damages in connection with the furnishing, performance, or use of this guide.

Declaration of Conformity



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

CE Mark Warning

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

Part 15 FCC Rules

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

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

Class B Digital Device or Peripheral

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

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

WEEE Warning



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

Customer Feedback

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

GNU GPL INFORMATION

Yealink SIP-T20P firmware contains third-party software under the GNU General Public License (GPL). Yealink uses software under the specific terms of the GPL. Please refer to the GPL for the exact terms and conditions of the license.

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

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

About This Guide

Thank you for choosing the SIP-T20P IP phone, exquisitely designed to provide business telephony features, such as Call Hold, Call Transfer, Busy Lamp Field, Multicast Paging and Conference over an IP network.

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-T20P IP phone.

If this is your first time using the SIP-T20P IP phone, we recommend that you first refer to the *Quick Installation Guide* and *Quick Reference Guide*, available online:

http://www.yealink.com/DocumentDownload.aspx? Cateld = 142& flag = 142.

Note

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

In This Guide

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 71, Guide Version 71.140

Major updates have occurred to the following:

- LED Instructions on page 4
- Ring Tones on page 24

- Anonymous Call on page 75
- Troubleshooting on page 97

Changes for Release 71, Guide Version 71.125

Major updates have occurred to the following:

• Appendix A - Time Zones on page 111

Changes for Release 71, Guide Version 71.110

Major updates have occurred to the following sections:

- Basic Network Settings on page 14
- Keypad Lock on page 21
- Contact Management on page 26
- DSS Keys on page 37

Changes for Release 70, Guide Version 70

Major updates have occurred to the following:

- Keypad Lock on page 21
- Volume on page 23
- Ring Tones on page 24
- Call Completion on page 59
- DSS Keys on page 37
- Do Not Disturb (DND) on page 61
- Call Forward on page 64
- Busy Lamp Field (BLF) on page 79

Table of Contents

About This Guide	V
In This Guide	V
Summary of Changes	
Changes for Release 71, Guide Version 71.140	
Changes for Release 71, Guide Version 71.125	
Changes for Release 71, Guide Version 71.110	
Changes for Release 70, Guide Version 70	
Table of Contents	vii
Overview	1
Hardware Component Instructions	1
Icon Instructions	3
LED Instructions	4
User Interfaces	4
Phone User Interface	4
Web User Interface	5
Documentations	7
Getting Started	9
Packaging Contents	9
Phone Installation	11
Phone Initialization	13
Phone Status	13
Basic Network Settings	14
Registration	16
Idle Screen	16
Customizing Your Phone	17
General Settings	17
Language	17
Time & Date	18
Administrator Password	20
Key as Send	20
Keypad Lock	21
Audio Settings	23

Volume	23
Ring Tones	24
Contact Management	26
Local Directory	26
Blacklist	32
Call History Management	33
System Customizations	
Logo Customization	34
Headset Use	
DSS Keys	
Account Management	43
Dial Plan	44
Emergency Number	50
Live Dialpad	51
Hotline	52
D : C !! E .	
Basic Call Features	53
Placing Calls	53
Answering Calls	55
Ending Calls	56
Redialing Numbers	57
Auto Answer	57
Auto Redial	58
Call Completion	59
Call Return	60
Call Mute	60
Call Hold/Resume	60
Do Not Disturb (DND)	61
Call Forward	64
Call Transfer	69
Call Waiting	70
Conference	70
Local Conference	71
Network Conference	71
Call Park	73
Call Pickup	
Anonymous Call	
Anonymous Call Rejection	76
Advanced Phone Features	79
Busy Lamp Field (BLF)	79
Call Recording	
Hot Deskina	

Intercom	84
Outgoing Intercom Calls	84
Incoming Intercom Calls	84
Multicast Paging	86
Sending RTP Stream	86
Receiving RTP Stream	88
Music on Hold	89
Automatic Call Distribution (ACD)	90
Message	92
Voice Mail	92
Message Waiting Indicator (MWI)	93
Troubleshooting	97
•	
Regulatory Notices	107
•	
Regulatory Notices	107
Regulatory Notices	107 107
Regulatory Notices Service Agreements Limitations of Liability	

Overview

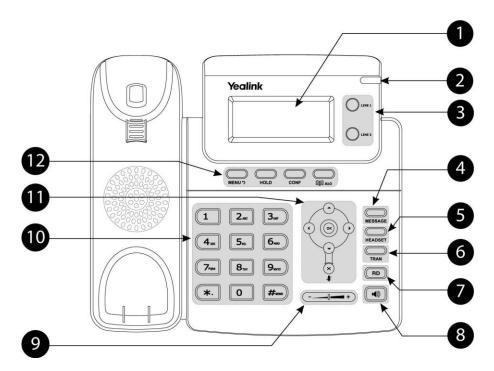
This chapter provides the overview of the SIP-T20P IP phone. Topics include:

- 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-T20P IP phone are the LCD screen and the keypad.



Hardware component instructions of the SIP-T20P IP phone are:

	Item	Description	
		Shows information about calls, messages, 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, "New Voice Mail")	
		Time and date	
2	Power Indicator LED	Indicates phone power status and phone status.	
3	Line Keys	Use these keys to activate up to two accounts and	
	Ellie Reyo	assign various features.	
4	MESSAGE Key	Indicates and accesses voice mails.	
(5)	HEADSET Key	Toggles the headset mode.	
6	TRAN Key	Transfers a call to another party.	
7	RD Key	Redials a previously dialed number.	
8	Speakerphone Key	Toggles the hands-free speakerphone mode.	
(9)	Volume Key Adjusts the volume of the handset, headset, spe		
	and ringer.		
Reypad Provides the digits, letters and spe		Provides the digits, letters and special characters in	
	context-sensitive applications.		
	$ \bigcirc \bigcirc$	Scroll through the displayed information.	
(1)	ОК	Confirms actions or answers an incoming call.	
	×	Cancels actions, rejects incoming calls, mutes or	
		un-mutes a call.	
	MENU Key	Enters the main menu of phone or returns to the previous	
	,	interface.	
(12)	HOLD Key	Places a call on hold or resumes a held call.	
	CONF Key	Conducts a conference call with multiple other parties.	
	Directory Key	Enters the directory interface or switches the input	
	Directory itey	mode.	

Icon Instructions

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

Icon	Description
	Network is unavailable
■●》	Hands-free speakerphone mode
<i>C</i> .,,	Handset mode
6	Headset mode
123	Numeric input mode
abc	Multi-lingual lowercase letters input mode
ABC	Multi-lingual uppercase letters input mode
2aB	Alphanumeric input mode
\bowtie	Voice Mail
AA	Auto Answer
DND	Do Not Disturb
	Call Forward/Forwarded Calls
\$	Call Mute
8	Keypad Lock
↓	Received Calls
1	Placed Calls
✓	Missed Calls

LED Instructions

Power Indicator LED

LED Status	Description	
Solid green	The phone is initializing.	
Fast Flashing green	The phone is ringing.	
Off	The phone is powered off.	
	The phone is idle when powered on	

Line key LED

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

Line key LED (configured as BLF key)

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

User Interfaces

Two ways to customize configurations of your SIP-T20P 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, each phone has a 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 key to access the phone user

interface. The Advanced option is only accessible to the administrator, and the default administrator password is "admin" (case-sensitive). For more information on customizing your phone with the available options from the phone user interface, refer to Customizing Your Phone on page 17.

Web User Interface

In addition to the phone user interface, you can also customize your phone via web user interface. In order to access the web user interface, you need to know the IP address of your new phone. To obtain the IP address, press the OK key on the phone. 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		
IP Address		
MAC		
Firmware	\checkmark	$\sqrt{}$
Network		
Phone		
Accounts		
Basic Phone Settings		
Language	\checkmark	
Time & Date	√	
Administrator Password	√	
Key as Send	√	
Keypad Lock	√	
Ring Tones	√	
Contact Management		
Local Directory	√	. /
Blacklist	√) v
Call History Management	√	
Logo Customization		
DSS Keys	√	
Account Registration	√	
Dial Plan		
Emergency Number		
Live Dialpad		
Hotline	√	

Options	Phone User Interface	Web User Interface
Basic Call Features		
Auto Answer	√	
Auto Redial	V	
Call Completion	V	
Call Return	V	
Do Not Disturb (DND)	√	
Call Forward	√	
Call Transfer	√	,
Call Waiting	√ ·	√
Conference	,	
Call Park	V	
Call Pickup	√	
Anonymous Call	√ √	
Anonymous Call Rejection	V √	
Advanced Phone Features	V	
	./	
Busy Lamp Field (BLF)	√ √	
Call Recording		
Hot DeskingIntercom	√ √	-/
Multicast Paging	V	V
Music on Hold		
Automatic Call Distribution (ACD)	√	
Messages	V √	
SIP Account	V	
User Options		
Activation	√	
Label	√	
Display Name	√	
Register Name	√	
User Name	√	
Password	√	,
SIP Server1/2	√	√
Server Option	·	
Registrar Port		
Outbound Status	V	
Outbound Proxy	· √	
NAT Traversal		
STUN Status	√	
STUN Server	√	

Note

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

Documentations

The following table shows documentations available for the SIP-T20P IP phone.

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

Note

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

Getting Started

This chapter provides basic installation instructions and information for obtaining the best performance with the SIP-T20P IP phone. Topics include:

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

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

Packaging Contents

The following components are included in your SIP-T20P IP phone package:

• SIP-T20P IP phone



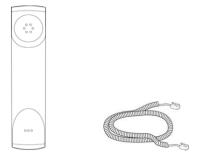
• Phone Stand



• Power Adapter



• Handset & Handset Cord



• Ethernet Cable



• Quick Installation Guide & Quick Reference Guide



CD-ROM



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

administrator.

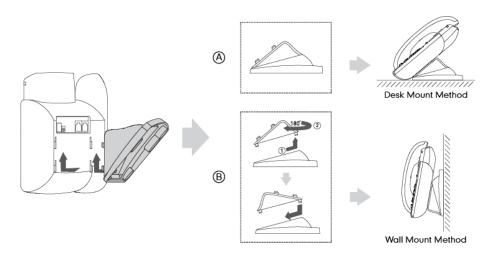
Phone Installation

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

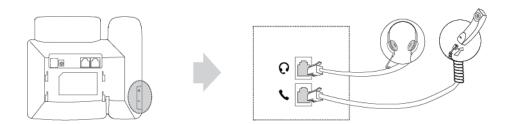
This section introduces how to install the phone:

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

1) Attach the Stand



2) Connect the Handset and optional Headset



Note

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

3) Connect the Network and Power

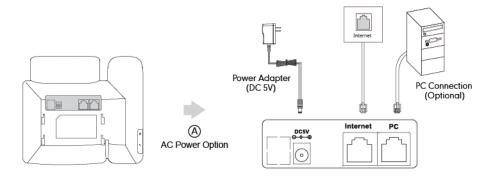
You have two options for power and network connections. Your system administrator will advise you which one to use.

- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power:

- 1. Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.
- 2. Connect the included or a standard Ethernet cable between the Internet port on the phone and the one on the wall or switch/hub device port.

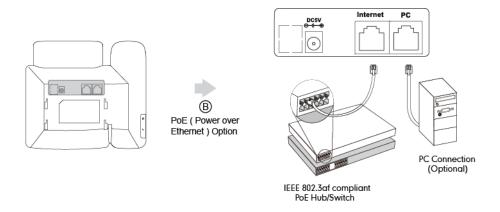


Power over Ethernet

With the included or a regular Ethernet cable, the SIP-T20P IP phone can be powered from a PoE-compliant switch or hub.

To connect the PoE:

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



Note

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

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

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

Phone Initialization

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

Automatic Phone Initialization

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

DHCP (Dynamic Host Configuration Protocol)

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

Note

If your network does not use DHCP, proceed to Basic Network Settings on page 14.

Phone Status

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

Available information of phone status includes:

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

To view the phone status via phone user interface:

- 1. Press or press and then select **Status**.
- **2.** Press (\bullet) or (ullet) to scroll through the list and view the specific information.

1. IP Address: 192.168.0.10

To view the phone status via web user interface:

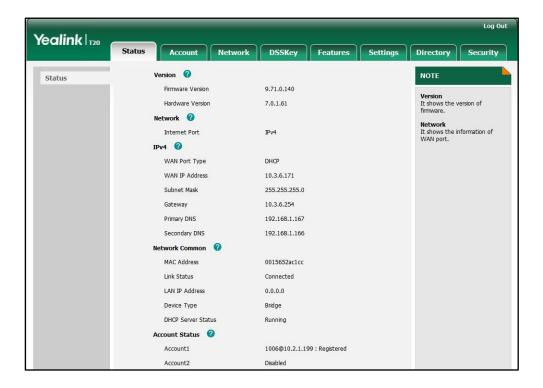
- 1. Open the web browser on your computer.
- 2. Enter the IP address in the browser's address bar, and then press **Enter**.

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



4. Click Confirm to login.

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



Basic Network Settings

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

To configure the IP address mode via phone user interface:

- 1. Press .
- 2. Select Settings->Advanced (password: admin) -> Network-> WAN Port.

1. IP Mode:
IPv4 & IPv6 ►
4. Press ok to accept the change.
To configure a static IPv4 address via phone user interface:
1. Press MENU2.
2. Select Settings->Advanced (password: admin) ->Network->WAN
Port->IPv4->Static IP.
3. Enter the desired values in the IP Address, Subnet Mask, Default Gateway, Pri.DNS
and Sec.DNS fields respectively.
1. IP Address: 192.168.0.3
123
4. Press ox to accept the change.
To configure a static IPv6 address via phone user interface:
1. Press .
2. Select Settings->Advanced (password: admin) ->Network->WAN
Port->IPv6->Static IPv6.
3. Enter the desired values in the IPv6, IPv6 Prefix, IPv6 Gateway, IPv6 Pri.DNS and
IPv6 Sec.DNS fields respectively.
1. IPv6: 2005:1:1:1::12
2aB
4. Press (ok) to accept the change.
If you are using an xDSL modem for IPv4 network connection, you can connect your
phone to the Internet via PPPoE mode. Set the WAN port as a PPPoE port. The PPPoE port
will perform a PPP negotiation to obtain the IP address. Contact your system
administrator for the PPPoE user name and password.
To configure PPPoE via phone user interface:

1. Press .

3. Press • or • to select IPv4, IPv6 or IPv4 & IPv6 from the IP Mode field.

 Select Settings->Advanced (password: admin) ->Network->WAN Port->IPv4->PPPoE.

> 1. PPPoE User: 2aB

- 3. Enter the user name and password in the corresponding fields.
- **4.** Press (oK) to accept the change.

Note

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

Registration

Generally, your phone will be deployed with multiple other phones. In this case, your system administrator will configure the phone parameters beforehand, so that after you start up your phone, the phone will be registered and ready for use. If your phone is not registered, you may have to register it. For more information on how to register your phone, refer to Account Management on page 43.

Idle Screen

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

1234 17 Jan 09:20

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

Customizing Your Phone

You can customize your SIP-T20P IP phone by personally configuring certain settings, for example, language, time & date and ring tones. 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

Language

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

Note

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

To change the language for the phone user interface:

- 1. Press .
- 2. Select Settings->Basic->Language.

3. Press ♠ or ♠ to select the desired language.

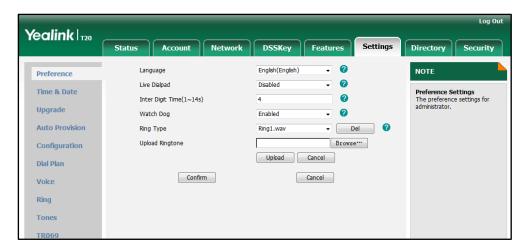


4. Press (ok) to accept the change.

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

To change the language for web user interface:

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



3. Click Confirm to accept the change.

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

Time & Date

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

To configure the SNTP settings via phone user interface:

- 1. Press .
- 2. Select Settings->Basic->Time & Date->SNTP.
- 3. Press or to select the time zone that applies to your area from the **Time Zone** field.

The default time zone is "+8 China (Beijing)".

- Enter the domain names or IP addresses in the NTP Server 1 and NTP Server 2 fields respectively.
- 5. Press () or () to select the desired value from the **DST** field.
- **6.** Press (ox) to accept the change.

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 .
- 2. Select Settings->Basic->Time & Date->Manual.
- 3. Enter the specific date in the Date field.
- 4. Enter the specific time in the **Time** field.

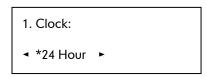
1. Date (Y-M-D): 13-05-18

5. Press (ok) to accept the change.

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

To configure the time format via phone user interface:

- 1. Press .
- 2. Select Settings->Basic->Time & Date->Time & Date.



- 3. Press lacksquare or lacksquare to select the desired time format from the **Clock** field.
- **4.** Press $\binom{OK}{OK}$ to accept the change.

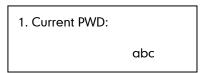
Time and date is configurable via web user interface at the path Settings->Time & Date.

Administrator Password

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

To change the administrator password via phone user interface:

- 1. Press MENUD
- 2. Select Settings->Advanced (password: admin) ->Set Password.



- 3. Enter the old password in the Current PWD field.
- 4. Enter the new password in the New PWD field.
- 5. Enter the new password again in the Confirm PWD field.
- **6.** Press (ox) to accept the change.

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

Key as Send

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

To configure key as send via phone user interface:

- 1. Press .
- 2. Select Features->Key as Send.



- 3. Press or to select # or * from the Key as Send field, or select Disable from the Key as Send field to disable this feature.
- **4.** Press (oK) to accept the change.

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

Keypad Lock

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

Keypad lock consists of the following:

Menu Key: The MENU key and MESSAGE key are locked. You cannot access

the menu of the phone until unlocked.

Function Keys: The function keys are locked. You cannot use the MESSAGE, CONF,

HOLD, MENU, TRAN, RD, Directory, OK, X, line keys and navigation

keys until unlocked.

All Keys: All keys are locked except the Volume key. You are only allowed to

dial emergency numbers, reject incoming calls by pressing the X key, answer incoming calls by lifting the handset, pressing the Speakerphone key, the HEADSET key or the OK key, mute an active call by pressing the X key, place an active call on hold by pressing the HOLD key, resume the held call by pressing the HOLD key, and end the call by hanging up the handset or

pressing the Speakerphone key.

Note

The emergency number setting, if desired, must be made before lock activation. For more information, refer to Emergency Number on page 50.

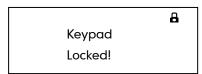
To activate the keypad lock via phone user interface:

- 1. Press .
- 2. Select Settings->Advanced (password: admin) -> Keypad Lock.
- 3. Press () or () to select desired type from the **Keypad Lock** field.



- **4.** Press (ok) to accept the change.
- 5. Long press $\boxed{\textbf{\#}_{\text{moo}}}$ to lock the keypad immediately when the phone is idle.

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

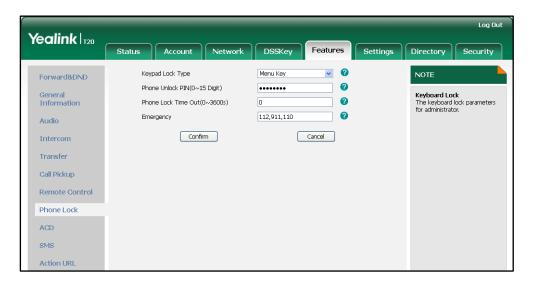


You can specify the interval (in seconds) to automatically lock the keypad instead of

long pressing #

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

- 1. Click on Features->Phone Lock.
- 2. Enter the desired interval in the Phone Lock Time Out (0~3600s) field.



3. Click Confirm to accept the change.

Note

The default interval is 0 second, that is, you need to long press #second to lock the keypad.

The interval of automatic keypad lock is configurable via web user interface only.

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

To change the keypad unlock PIN via phone user interface:

- 1. Press .
- 2. Select Settings->Basic->Phone Unlock.
- 3. Enter the old PIN in the Current PIN field.



- Enter the new PIN in the New PIN field.
- 5. Enter the new PIN again in the Confirm PIN field.
- **6.** Press (ok) to accept the change.

Note

The unlock PIN length must be within 15 characters.

To unlock the keypad via phone user interface:

- 1. Press any locked key.
- 2. Enter the PIN in the PIN field.
- 3. Press (ok) to unlock the keypad.

The \blacksquare icon disappears from the LCD screen.

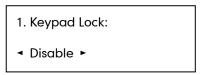
You can long press # or wait for a period of time (if configured) to lock the keypad again.

Note

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

To deactivate the keypad lock via phone user interface:

- 1. Press .
- 2. Select Settings->Advanced (password: admin) ->Keypad Lock.
- 3. Press () or () to select **Disable** from the **Keypad Lock** field.



4. Press (ok) to accept the change.

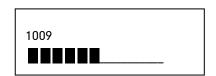
Keypad lock is configurable via web user interface at the path Features->Phone Lock.

Audio Settings

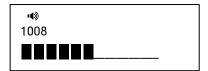
Volume

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

To adjust the volume when the phone is idle:



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

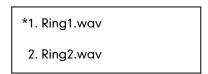


Ring Tones

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

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

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



4. Press (ok) to accept the change.

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

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

4. Select the desired ring tone from the pull-down list of Ring Type.

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



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

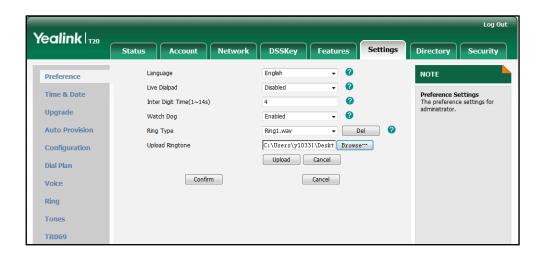
Note

A ring tone for the account is configurable via web user interface only.

The ring tone for an incoming call on the phone may be different. For example, when the phone receives an incoming call from a contact stored in the local directory, it will play the ring tone assigned to the contact in the contact directory (refer to Adding Contacts). If no ring tone is assigned to the contact, the phone will play the ring tone assigned to the associated group (refer to Adding Groups). Otherwise, the phone will play the ring tone assigned to the account. If no ring tone is assigned to the account, the phone will play the ring tone assigned for the phone.

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

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



3. Click Upload to upload the file.

Note

All custom ring tone files must be within 100KB.

Custom ring tones for your phone can be uploaded via web user interface only.

Contact Management

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

- Local Directory
- Blacklist

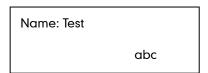
Local Directory

The built-in phone directory stores the names and phone numbers of your contacts. You can store up to 1000 contacts and 5 groups in your phone's local directory; add new contacts and groups; edit, delete or search for a contact; or simply dial a contact number from the local directory.

Adding Groups

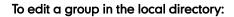
To add a group to the local directory:

- 1. Press .
- 2. Select Local Directory.
- 3. Press \bullet or \bullet to select **AddGrp** and then press \bullet .
- 4. Press to switch the input mode.
- 5. Enter the group name in the Name field.

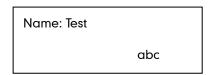


- 6. Press or to select the desired group ring tone from the **Ring Tones** field. If **Auto** is selected, this group will use the ring tone assigned to the account. For more information on ring tone for a account, refer to Ring Tones on page 24.
- 7. Press (ok) to accept the change.

Editing Groups



- 1. Press .
- 2. Select Local Directory.
- **5.** Press or to select the desired group (e.g., Test).
- **4.** Press lacksquare or lacksquare to select **Edit** and then press lacksquare .
- 5. Press to switch the input mode.



- **6.** Press \bigcirc or \bigcirc to select the contact information and then edit.
- 7. Press (ok) to accept the change.

Deleting Groups

To delete a group from the local directory:

- 1. Press .
- 2. Select Local Directory.
- **3.** Press ♠ or ♥ to select the desired group (e.g., Test).
- **4.** Press \bullet or \bullet to select **Del** and then press \bullet .

The LCD screen prompts "Delete selected group?".



5. Press (ox) to accept the change.

Adding Contacts

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

- Manually
- From call history

Adding Contacts Manually

Adding Contacts Manually
To add a contact to the local directory manually:
1. Press .
2. Select Local Directory.
3. Press or to select Enter and then press ok.
If the groups have been added to the local directory, select the desired group, then
press or to select Enter and then press ox.
4. Press or to scroll to New Item.
5. Press ox to add a new contact.
6. Press to switch the input mode.
Name:
Nume.
abc
7. Enter the name and the office, mobile or other numbers in the corresponding fields
8. Press or to select the desired account from the Account field.
If Auto is selected, the phone will use the first available account when placing calls
to the contact from the local directory.
9. Press or to select the desired ring tone from the Ring Tones field.
If Auto is selected, this contact will use the ring tone assigned to the group.
10. Press ox to accept the change.
If the contact has existed in the directory, the LCD screen prompts "Contact name existed!".
Adding Contacts from Call History
To add a contact to the local directory from the call history:
1. Press (*).
2. Press • or • to select the desired entry.
3. Press .
4. Press to switch the input mode, and enter the contact name in the Name field.
Name:

abc

5. Press \bigcirc to accept the change.

Note

The entry is successfully saved to the local directory.

Editing Contacts

То	edit a contact in the local directory:
1.	Press Abs .
2.	Select Local Directory.
3.	Press or to select Enter and then press ox.
	If the groups have been added to the local directory, select the desired group, then
	press or to select Enter and then press or .
4.	Select the desired contact and press () or () to select Edit .
	Name: john
	abc
5.	Press (or (to select the contact information and then edit.
6.	Press (ox) to accept the change.
Deleting	Contacts
J 222323	
То	delete a contact from the local directory:
1.	Press And .
2.	
3.	Press () or () to select Enter and then press (ok) .
	If the groups have been added to the local directory, select the desired group, then
	press or to select Enter and then press ox.
4.	Press • or • to select the desired contact.
5.	Press or to select Del and then press ox.
	The LCD screen prompts "Delete selected Item?".
6.	Press (ox) to confirm the deletion.

Placing Calls to Contacts

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

- 1. Press Ab3.
- 2. Select Local Directory.
- 3. Press lacktriangledown or lacktriangledown to select **Enter** and then press lacktriangledown .

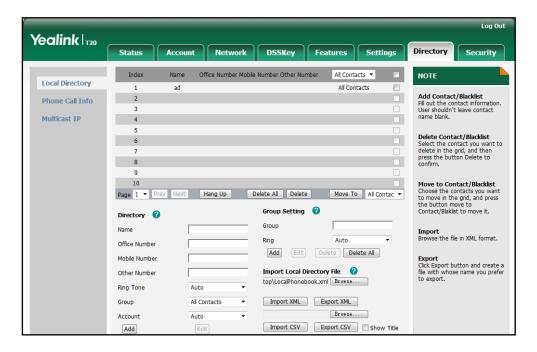
		If the groups have been added to the local directory, select the desired group, then
		press (•) or (•) to select Enter and then press (ox).
4	١.	Press () or () to select the desired contact.
5	5.	Press or to select Send .
6	5 .	Do one of the following:
		- If only one number of the contact is stored in the local directory, press ${}^{\!}$ or dial out the number.
		- If multiple numbers of the contact are stored in the local directory, press (ox) to display a list of numbers.
		Press () or () to select the desired number.
		Press ok to dial out the number.
Searchi	ing	g for Contacts
T	o s	earch for a contact in the local directory:
1	١.	Press .
2	2.	Select Local Directory.
3	5.	Press \bullet or \bullet to select Enter and then press \bullet .
		If the groups have been added to the local directory, select the desired group, then
		press or to select Enter and then press or .
4	l .	Enter a few continuous characters of the contact name or continuous numbers of the contact phone number using the keypad.
		Search: tom
		2aB
5	5.	Press OK .
		The contacts whose name or phone number matches the characters you entered
		will appear on the LCD screen. You can dial from the query result.
Import	ing	g/Exporting Contact Lists
Υ	⁄ου	can manage your phone's local directory via phone or web user interface. But you

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

can only import or export the contact list via web user interface.

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

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

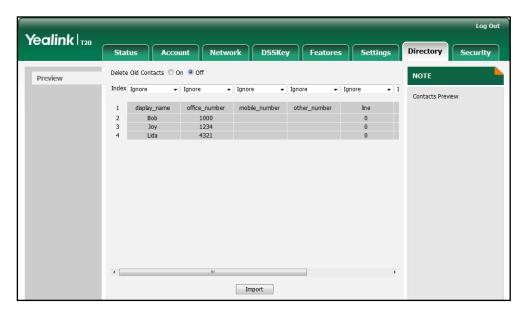


- 3. Click Import XML to import the contact list.
 The web user interface prompts "The original contact will be covered, Continue?".
- 4. Click **OK** to complete importing the contact list.

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

- 1. Click on **Directory**->**Local Directory**.
- 2. Click **Browse** to locate a contact list file (file format must be *.csv) from your local system.
- (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.
- (Optional.) Mark the On radio box in the Delete Old Contacts field.It will delete all existing contacts while importing the contact list.

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



7. Click Import to complete importing the contact list.

To export contact list via web user interface:

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

Note

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

Blacklist

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

To add a contact to the blacklist manually:

- 1. Press .
- 2. Select Blacklist.
- 3. Press () or () to scroll to New Item.
- **4.** Press (ox) to add a new contact.
- 5. Press \bigcirc to switch the input mode.

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



7. Press or to select the desired account from the Account field.
If Auto is selected, the phone will use the first available account when placing calls to the contact from the blacklist.

8. Press (ok) to accept the change.

For operating instructions on adding, editing, deleting, placing calls to and/or searching for contacts in the blacklist, refer to the operating instructions of Local Directory on page 26.

Call History Management

The SIP-T20P IP phone maintains call history lists of Placed Calls, Received Calls, Missed Calls and Forwarded Calls. Call history lists support 100 entries in all. You can view the call history, place a call, add a contact or delete an entry from the call history list. You should enable history record in advance.

To enable history record via phone user interface:

- 1. Press .
- 2. Select Features->History Setting.
- 3. Press (\cdot) or (\cdot) to select **Enable** from the **History Record** field.



4. Press (ok) to accept the change.

To view the call history:

1. Press • .

The LCD screen displays the last call history.

- 2. Press or to switch between placed calls, received calls, missed calls and forwarded calls.
- 3. Press (\bullet) or (\bullet) to select the desired entry.
- 4. Press Ab3.

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

To place a call from the call history list:

1. Press (•).

The LCD screen displays the call list.

- 2. Press or to switch between placed calls, received calls, missed calls and forwarded calls.
- 3. Press () or () to select the desired entry.
- 4. Press (ok) to dial out.

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

1. Press (•).

The LCD screen displays the call list.

- 2. Press or to switch between placed calls, received calls, missed calls and forwarded calls.
- 3. Press (\bullet) or (ullet) to select the desired entry.
- 4. Press
- 5. Enter the desired values in the corresponding fields.
- **6.** Press (oĸ).

For more information, refer to Contact Management on page 26.

To delete an entry from the call history list:

1. Press (•).

The LCD screen displays the call list.

- 2. Press or to switch between placed calls, received calls, missed calls and forwarded calls.
- 3. Press or to select the desired entry.
- 4. Press (x) to delete the entry.

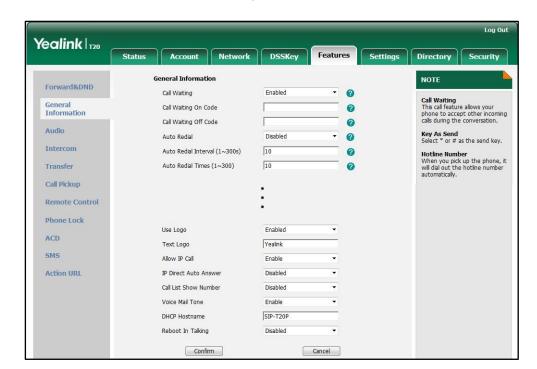
System Customizations

Logo Customization

You can customize text logo which would be shown on the idle screen.

To customize a text logo via web user interface:

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



3. Enter the desired text in the Text Logo field.

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

Note

The maximum length of text logo is 15 characters.

A text logo is configurable via web user interface only.

Headset Use

Physically connect your headset and activate the headset mode for use. For more information on physically connecting a headset, refer to Phone Installation on page 11.

Headset Mode Activation/Deactivation

To activate the headset:

1. Press on the phone.

The \bigcirc icon on the LCD screen indicates that the headset mode is activated. Press the line key to answer a call. The call will connect to your headset automatically. Enter the desired number and press \bigcirc , then the phone will use the headset to place a call automatically. For more information on using the headset to place a call, refer to Placing Calls on page 53.

To deactivate the headset:

1. Press again on the phone.

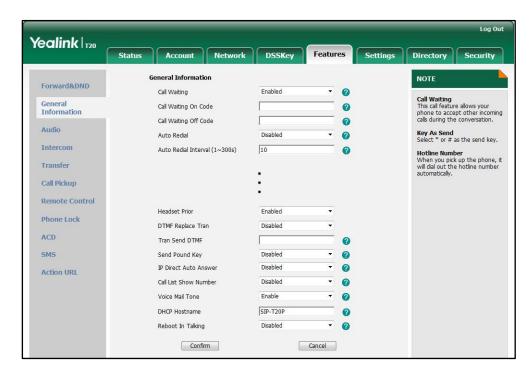
The headset icon disappears when the headset mode is deactivated.

Headset Prior

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

To enable Headset Prior via web user interface:

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



3. Click Confirm to accept the change.

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

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

Note

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

Headset prior is configurable via web user interface only.

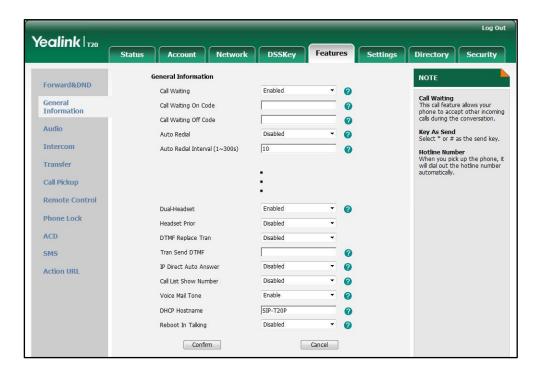
Dual Headset

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

listen.

To enable Dual Headset via web user interface:

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



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

Note

Dual headset is configurable via web user interface only.

DSS Keys

Two types of DSS keys: Line Keys and Programmable Keys. The details will be introduced in the following. SIP-T20P IP phone supports 2 line keys.

Line Keys

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

To configure the line key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.

- **3.** Select the desired DSS key, and then press (ooO
- 4. Select the desired type from the **Type** field.
- 5. (Optional.) Select the desired key event from the **Key Type** field.
- 6. (Optional.) Select the desired account from the Account ID field.
- 7. (Optional.) Enter the corresponding value in the **Extension** field.
- **8.** Press (or) to accept the change.

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

Note

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

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

- Speed Dial
- Voice Mail
- DTMF
- Prefix
- Local Group
- Conference
- Forward
- Transfer
- Hold
- DND
- Group Listening
- Zero Touch
- Keypad Lock

For more information, contact your system administrator.

Speed Dial

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

Dependencies: Type (Speed Dial)

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

Value (the number you want to dial out)

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

Voice Mail

You can use this key feature to connect voice mail quickly. For more information, refer to Message

Voice Mail on page 92.

Dependencies: Type (Key Event)

Key Type (Voice Mail)

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

Value (the voice mail access code)

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

DTMF

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

Dependencies: Type (Key Event)

Key Type (DTMF)

Value (DTMF sequence)

Note

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

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

Prefix

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

Dependencies: Type (Key Event)

Key Type (Prefix)

Value (the prefix number)

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

Local Group

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

Dependencies: Type (Key Event)

Key Type (Local Group)

Local Group (the contact group you want to access)

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

Conference

You can use this key feature to set up a conference call. For more information, refer to Conference on Page 70.

Dependencies: Type (Key Event)

Key Type (Conf)

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

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

Note

When you leave the **Value** field blank, the DSS key performs the same function as the **CONF** key during a call.

Forward

You can use this key feature to forward an incoming call to someone else. For more information, refer to Call Forward on page 64.

Dependencies: Type (Key Event)

Key Type (Forward)

Value (the number you want to forward to)

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

Note

When you leave the **Value** field blank, the DSS key performs the same function as the **TRAN** key when receiving an incoming call.

Transfer

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

Dependencies: Type (Key Event)

Key Type (Tran)

Value (the number you want to transfer to)

Usage:

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

 When the transfer mode on DSS key is New Call, press the DSS key to place a new call to the number specified in the Value field.

Note

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

When you leave the **Value** field blank, the DSS key performs the same function as the **TRAN** key during a call. For more information, refer to Call Transfer on page 69.

Hold

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

Dependencies: Type (Key Event)

Key Type (Hold)

Usage:

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

DND

You can use this key feature to activate or deactivate DND. You can also use this key feature to access the custom DND interface. For more information, refer to Do Not Disturb (DND) on page 61.

Dependencies: Type (Key Event)

Key Type (DND)

Usage:

When the phone is in phone mode:

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

When the phone is in custom mode:

- 1. Press the DSS key to access the custom DND interface.
- 2. Activate or deactivate DND for an account.

Note

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

Group Listening

You can use this key feature to activate the Speakerphone and Handset/Headset mode at the same time. It is suitable for the group conversation which has more than one person at one side. You are able to speak and listen through the handset/headset, meanwhile the others nearby can only listen through the speaker.

Dependencies: Type (Key Event)

Key Type (Group Listening)

Usage:

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

Zero Touch

You can use this key feature to configure the network parameters quickly.

Dependencies: Type (Key Event)

Key Type (Zero Touch)

Usage:

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

The phone will reboot to update configurations.

Keypad Lock

You can use this key feature to immediately lock the keypad of your phone instead of long pressing #see. For more information, refer to Keypad Lock on page 21.

Dependencies: Type (Keypad Lock)

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

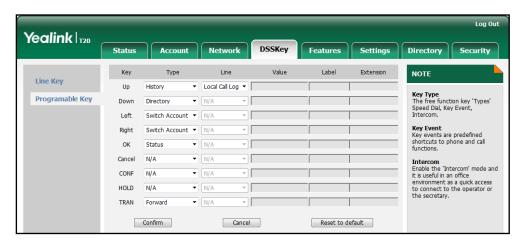
Programmable Keys

You can customize the navigation keys, HOLD, CONF, TRAN keys on the keypad.

To customize the programmable keys via web user interface:

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

2. Customize specific features for these keys.



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

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

Note

Programmable keys are configurable via web user interface only.

Account Management

You can register two accounts at most on the SIP-T20P IP phone. You can also configure each line key to associate with an account or configure two line keys to associate with an account.

Account Registration

To register an account via phone user interface:

- 1. Press .
- 2. Select Settings->Advanced->Accounts.
- 3. Select the desired account.
- 4. Select **Enable** from the **Activation** field.



- Enter the desired values in the Label, Display Name, Register Name, User Name, Password and SIP Server1/2 fields respectively. Contact your system administrator for more information.
- **6.** Press (ok) to accept the change.

You can repeat steps 3 to 5 to register more accounts.

To disable an account via phone user interface:

- 1. Press .
- 2. Select Settings->Advanced ->Accounts.
- 3. Select the desired account.
- 4. Select **Disable** from the **Activation** field.
- **5.** Press (ok) to accept the change.

Account registration is configurable via web user interface at the path **Account->Register**.

Multiple Line Keys per Account

You can configure two line keys to associate with an account. This enhances call visualization and simplifies call handling.

Incoming calls to this account will be distributed evenly among the available line keys. Outgoing calls will be distributed similarly.

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

Dial Plan

Dial plan is a string of characters that governs the way your SIP-T20P IP phone processes the inputs received from your phone keypad. The SIP-T20P 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 ", etc.
.,	An "x" can be used as a placeholder for any character. Example:
Х	"12x" would match "12 1 ", "12 2 ", "12 3 ", etc.
	The square brackets "[]"can be used as a placeholder for a single character
	which matches any of a set of characters. Example:
	"91[5-7]1234" would match "91 5 1234", "91 6 1234", "91 7 1234".
()	The parentheses "() "can be used to group together patterns, for instance, to
()	logically combine two or more patterns. Example:

"91([5-7])1(x)" would match "91**511**", "91**618**", "91**715**".

The "\$" should be followed by the sequence number of a parenthesis. The "\$" plus the sequence number means the whole character or characters placed in the parenthesis. The number directs to the right parenthesis when there are more than one. Example:

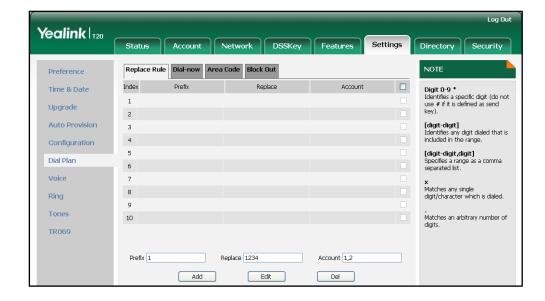
A replace rule configuration: Prefix: "9([5-7]) (.)", Replace: "5\$2". When you enter "96123" to dial out on your phone, the number will be replaced as "5123" and then dialed out. "\$2" means the characters in the second parenthesis, that is, "123".

Replace Rule

You can configure one or more replace rules (up to 100) to remove the specified string and replace it with another string. You can configure a pattern with wildcards (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: 1234567", when you try to dial out the number "1234567", you just need to enter "1" on the phone and then press the OK key to dial out.

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.
- 4. Enter the desired line ID in the Account field or leave it blank.



5. Click Add to add the replace rule.

When you enter the number "1" using the keypad and then press the OK key, the phone will dial out "1234" instead.

Note

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

If you leave the **Account** field blank or enter 0, the replace rule will apply to all accounts.

To edit a replace rule via web user interface:

- 1. Click on Settings->Dial Plan->Replace Rule.
- 2. Select the desired replace rule by checking the check box.
- 3. Edit the values in the **Prefix** and **Replace** fields.
- 4. Enter the desired line ID in the **Account** field or leave it blank.
- 5. Click Edit to accept the change.

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

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

Note

Replace rule is configurable via web user interface only.

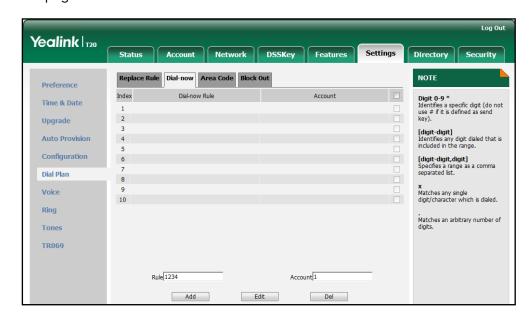
Dial-now

You can configure one or more dial-now rules (up to 100) on your phone. When the dialed out number matches the dial-now string, the number will be dialed out automatically. For example, a dial-now rule is configured as "2xx", then any entered three-digit string beginning with 2 will be dialed out automatically on the phone.

To add dial-now rule via web user interface:

- 1. Click on Settings->Dial Plan->Dial now.
- 2. Enter the desired value (e.g., 1234) in the Rule field.

Enter the desired line ID in the Account field or leave it blank.
 For more information on the valid values of Account field, refer to Replace Rule on page 45.



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

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

Note

You can also edit or delete the dial-now rule, refer to Replace Rule on page 45 for more information.

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

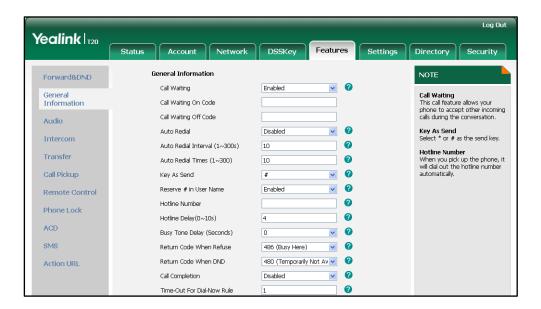
Delay Time for Dial-now Rule

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

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

1. Click on Features->General Information.

2. Enter the desired time within 1-14 (in seconds) in the **Time-Out For Dial-Now Rule** field.



3. Click Confirm to accept the change.

Note

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

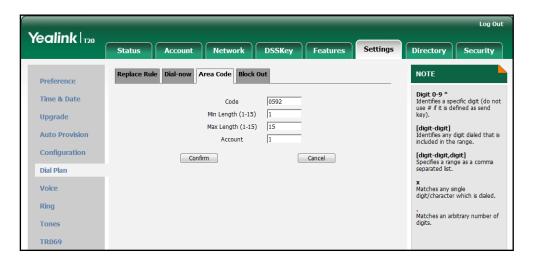
Area Code

Area codes are also known as Numbering Plan Areas (NPAs). They usually indicate geographical areas in a country. This feature is necessary only when dialing the number outside the code area. For example, area code is configured as "code: 0592, Min Length: 4, Maxi Length: 11", then when you dial out the number "56789", which has the digits between 4 to 11, 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.
- Enter the desired values in the Code, Min Length (1-15) and Max Length (1-15) fields.

Enter the desired line ID in the Account field or leave it blank.
 For more information on the valid values of Account field, refer to Replace Rule on page 45.



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

Note

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

Area code is configurable via web user interface only.

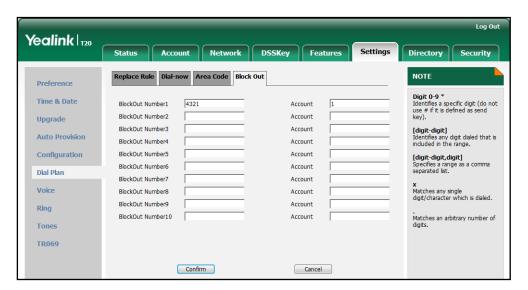
Block Out

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

To add a block out number via web user interface:

- 1. Click on Settings->Dial Plan->Block Out.
- 2. Enter the desired value in the BlockOut Number field.

Enter the desired line ID in the Account field or leave it blank.
 For more information on the valid values of Account field, refer to Replace Rule on page 45.



4. Click Confirm to add the block out number.

Note

Block out number is configurable via web user interface only.

Emergency Number

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

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

Note

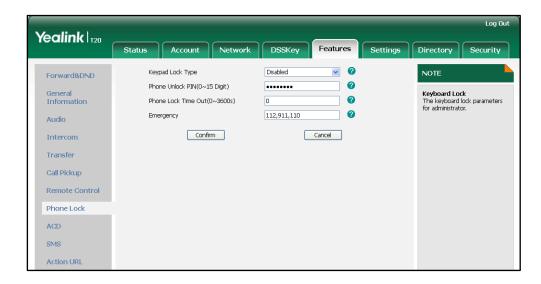
Contact your local phone service provider for available emergency numbers in your area.

To specify emergency numbers via web user interface:

1. Click on Features->Phone Lock.

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

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



3. Click Confirm to accept the change.

Note

Emergency number is configurable via web user interface only.

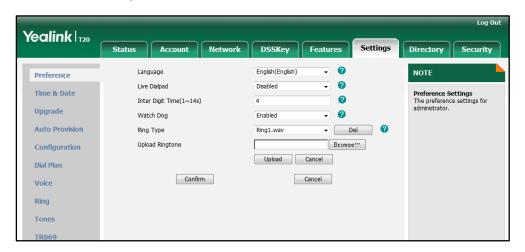
Live Dialpad

You can enable live dialpad on the SIP-T20P IP phone, which enables the IP phone to automatically dial out phone numbers without pressing any other key. You can also configure a delay, where the phone will dial out the phone number automatically after the specified period of time.

To enable live dialpad via web user interface:

- 1. Click on **Settings**->**Preference**.
- 2. Select **Enabled** from the pull-down list of **Live Dialpad**.

Enter the desired delay time in the Inter Digit Time (1~14s) field.
 The default delay time is 4s.



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

Note

Live dialpad is configurable via web user interface only.

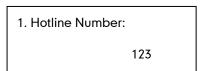
Hotline

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

To configure hotline number via phone user interface:

- 1. Press .
- 2. Select Features->Hotline.
- 3. Enter the desired numbers in the Hotline Number field.
- 4. Enter the delay time (in seconds) in the Hotline Delay field.

The valid values range from 0 to 10.



5. Press (ox) to accept the change.

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

Basic Call Features

The SIP-T20P 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-T20P IP phone. Topics include:

- Placing Calls
- Answering Calls
- Ending Calls
- Redialing Numbers
- Auto Answer
- Auto Redial
- Call Completion
- Call Return
- Call Mute
- Call Hold/Resume
- Do Not Disturb (DND)
- Call Forward
- Call Transfer
- Call Waiting
- Conference
- Call Park
- Call Pickup
- Anonymous Call
- Anonymous Call Rejection

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

Placing Calls

You can place a call in three ways using your SIP-T20P IP phone:

- Using the handset
- Using the speakerphone

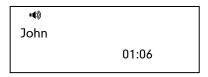
• Using the headset

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 or local directory. For more information, refer to Contact Management on page 26 and Call History Management on page 33.

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

The call duration of active calls is visible on the LCD screen. In the figure below, the call to John has lasted 1 minute and 6 seconds.



To place a call using the handset:

- 1. Pick up the handset.
- 2. Enter the desired number using the keypad.
- 3. Press (ok) or #sexo).

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 20.

Note

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

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

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

Do one of the following:

- With the handset on-hook, press or the line key to obtain a dial tone.

Enter the desired number using the keypad.

Press or a.

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

Press (\bullet) , (\circ) or (#) .

To place a call using the headset:

Do one of the following:

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

Press the line key to obtain a dial tone.

Enter the desired number using the keypad.

Press OK or #seno .

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

Enter the desired number using the keypad.

Press \bigcirc or $\boxed{\text{#}_{\text{\tiny SEND}}}$.

Note

To permanently activate the headset mode, refer to Headset Prior on page 36.

To place multiple calls:

You can have more than one call on your SIP-T20P IP phone.

1. Press the line key.

The active call is placed on hold.

- 2. Enter the desired number using the keypad.
- 3. Press (ok), (mal) or (#seno).

You can press • or • to switch between the calls, and then press again to resume the desired call.

Answering Calls

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

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

Note

You can ignore incoming calls by pressing the \mathbf{X} 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 61.

You can forward incoming calls to someone else by pressing the **TRAN** key. For more information, refer to Call Forward on page 64.

Answering When Not in Another Call

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

To answer a call using the handset:

1. Pick up the handset.

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

Do one of the following:

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

To answer a call using the headset:

Do one of the following:

- Press ____.
- With the headset mode activated, press (ok).
- With the headset mode activated, press the line key (the line LED flashes green).

Answering When in Another Call

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

- Press (ok).

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

- Press (ullet) to access the new call.

Press (ok).

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, hangs up the handset or press ______.
- If you are using the headset, press ______.
- If you are using the speakerphone, press (4) or .

Note

To end a call placed on hold, you should press the **HOLD** key to resume the call firstly before ending it.

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:

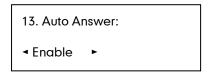
- 1. Press (RD) when the phone is idle.

Auto Answer

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

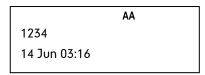
To configure auto answer via phone user interface:

- 1. Press .
- 2. Select Settings->Advanced (password: admin) ->Accounts.
- 3. Select the desired account and then press (ok).
- 4. Press () or () to select **Enable** from the **Auto Answer** field.



5. Press (ok) to accept the change.

The AA icon appears on the LCD screen.



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

Note

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

Auto Redial

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

To configure auto redial via phone user inte	erface:
--	---------

- 2. Select Features->Auto Redial.
- 3. Press () or () to select **Enable** from the **Auto Redial** field.



4. Enter the desired time in the Interval field.

The default time interval is 10s.

5. Enter the desired times in the Times field.

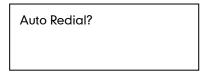
The default times are 10.

6. Press (ok) to accept the change.

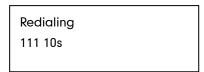
Auto redial is configurable via web user interface at the path **Features**->**General Information**.

To use auto redial:

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



1. Press ok to activate auto redial. The following prompt will appear on the LCD screen of the phone:



2. Wait for a period of time or press (ox) to redial the phone number.

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

Call Completion

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

To configure call completion via phone user interface:

- 1. Press .
- 2. Select Features->Call Completion.



- 3. Press (\cdot) or (\cdot) to select **Enable** from the **Call Completion** field.
- **4.** Press (o) to accept the change.

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:

Call Completion
Wait for 112?

1. Press ox . The phone returns to the idle screen and call completion is activated. When the called party becomes idle, the following prompt appears on the LCD screen of the phone:

Call Completion Dial 112?

2. Press (ok) to redial the number.

Note

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

Call Return

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

To configure a call return key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.
- 3. Select a desired DSS key.
- 4. Press () or () to select **Key Event** from the **Type** field.
- 5. Press () or () to select **Call Return** from the **Key Type** field.



6. Press (ox) to accept the change.

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

Call Mute

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

To mute a call:

1. Press (\times) during an active call.

The LCD screen indicates that the call is on mute.



To un-mute a call:

1. Press (\times) again to un-mute a 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 on hold to the other party while waiting.

To place a call on hold:

1. Press during a call.

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



Note

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

To resume a held call:

1. Press again.

Multiple Calls on Hold:

If multiple calls are placed on hold, do one of the following:

- Press or to switch between the calls, and then press to retrieve the desired call.
- Press the corresponding line key to retrieve the desired call.

If more than two calls are placed on hold, a numbered prompt appears on the LCD screen, for example "1/3", indicating that this is the first call out of three calls.

Do Not Disturb (DND)

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

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

- Phone (default): DND is effective for the phone system.
- Custom: DND can be configured for each account.

You can receive incoming calls from authorized numbers when DND is enabled.

To configure the DND mode via web user interface:

1. Click on Features->Forward & DND.

Log Out Yealink 120 Status Security Forward NOTE Forward&DND Mode Phone Custom **Call Waiting**This call feature allows your phone to accept other incoming calls during the conversation. General Information Account Audio Key As Send Select * or # as the send key. Target Intercom On Code Hotline Number
When you pick up the phone, it
will dial out the hotline number Off Code Transfer Call Pickup Remote Control Phone Lock DND 🕜 ACD DND Emergency Disabled SMS DND Authorized Numbers Mode Phone Custom Action URL Account On Off **DND** Status DND On Code DND Off Code

Cancel

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

3. Click Confirm to accept the change.

Note

DND mode is configurable via web user interface only.

Confirm

To activate DND in phone mode:

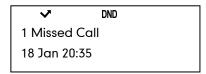
- 1. Press .
- 2. Select Features->DND Code.
- 3. Press () or () to select **Phone Mode** from the **DND Account** field.
- 4. Select **Enable** from the **DND** field.



- (Optional.) Enter the DND on code and off code respectively in the DND On Code and DND Off Code field.
- **6.** Press (or to accept the change.

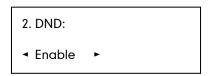
The **DND** icon on the idle screen indicates that DND is enabled.

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



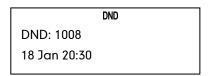
To activate DND in custom mode:

- 1. Press .
- 2. Select Features->DND Code.
- 3. Press () or () to select the desired account from the **DND Account** field.
- 4. Select **Enable** from the **DND** field.

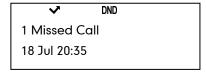


- (Optional.) Enter the DND on code and off code respectively in the DND On Code and DND Off Code field.
- **6.** Press (ox) to accept the change.

The **DND** icon appears on the idle screen and the prompt "**DND**: "appears in front of the associated account.



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



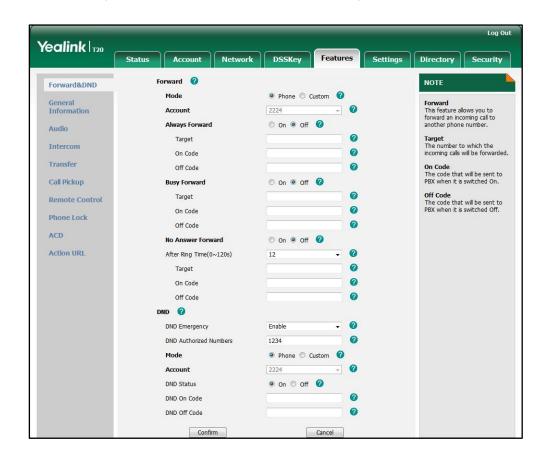
Note

The prompt message displays only if Missed Call Log for the line is enabled. Missed Call Log is configurable via web user interface at the path **Account->Basic**.

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

To configure the DND authorized numbers via web user interface:

- Click on Features->Forward & DND.
- 2. Select **Enabled** from the pull-down list of **DND Emergency**.
- Enter the numbers in the DND Authorized Numbers field.
 For multiple numbers, enter a comma between every two numbers.



4. Click Confirm to accept the change.

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

Note

DND authorized number is configurable via web user interface only.

Call Forward

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

Static Forwarding

Three types of static forwarding:

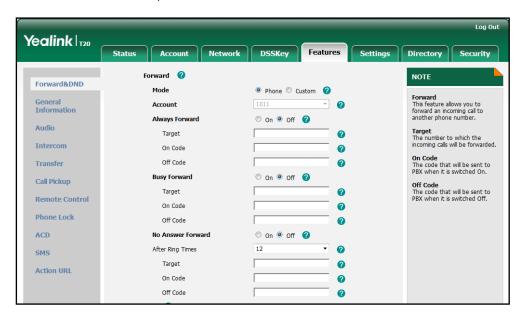
- Always Forward: Incoming calls are immediately forwarded.
- Busy Forward: Incoming calls are immediately forwarded if the phone is busy.
- No Answer Forward: Incoming calls are forwarded if not answered after a period
 of time.

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

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

To configure the call forward mode via web user interface:

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



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

To enable call forward in phone mode:

1. Press .

Note

- 2. Select Features->Call Forward.
- **3.** Press (-) or (-) to select the forwarding type, and then press (-).

Call forward mode is configurable via web user interface only.

a.) If you select Always Forward :	
1) Press () or () to select Enable from the Always fiel	eld.
Enter the destination number you want to forward all i	incoming calls to in
the Forward To field.	
3) (Optional.) Enter the always forward on code and off	f code respectively in
the On Code and Off Code field.	
1. Always:	
▼ Enable ►	
b.) If you select Busy Forward :	
 Enter the destination number you want to forward incommend the phone is busy in the Forward To field. 	coming cans to when
3) (Optional.) Enter the busy forward on code and off co	ode respectively in the
On Code and Off Code field.	
1. Busy:	
Inable →	
Endoic	
c.) If you select No Answer Forward :	
1) Press or to select Enable from the No Answer	er field.
2) Enter the destination number you want to forward all u	unanswered incoming
calls to in the Forward To field.	
3) Press () or () to select the ring time to wait before	e forwarding from the
After Ring Time field. The default ring time is 12 seconds.	
-	d off and a vaccatively
 (Optional.) Enter the no answer forward on code and in the On Code and Off Code field. 	a oir code respectively
die Chi Codo and Chi Codo nota.	
1. No Answer:	
The Image The I	

66

5. Press (OK) to accept the change.

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

1.	Press
2.	Select Features->Call Forward.
3.	Press $lacksquare$ or $lacksquare$ to select the desired account, and then press $lacksquare$.
4.	Press $lacksquare$ or $lacksquare$ to select the forwarding type, and then press $lacksquare$.
5.	Depending on your selection:
	a.) If you select Always Forward :
	1) Press () or () to select Enable from the Always field.

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



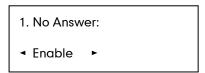
- b.) If you selected Busy Forward:
 - 1) Press () or () to select **Enable** from the **Busy** field.
 - 2) Enter the destination number you want to forward all incoming calls to when the phone is busy in the **Forward To** field.
 - 3) (Optional.) Enter the busy forward on code and busy off code respectively in the **On Code** and **Off Code** field.



- c.) If you select No Answer Forward:
 - 1) Press () or () to select **Enable** from the **No Answer** field.
 - 2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward To** field.
 - 3) Press or to select the ring time to wait before forwarding from the After Ring Time field.

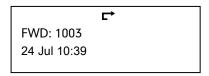
The default ring time is 12 seconds.

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



6. Press (oK) to accept the change.

The icon appears on the LCD screen and the prompt "FWD:" appears in front of the associate account.



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

Note

You can also enter the SIP URL or IP address in the **Forward To** field. For more information on using the SIP URL or IP address, refer to Placing Calls on page 53.

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

To disable call forward in phone mode:

Do one of the following:

- Press when the phone is idle.
- Press ____, select Features->Call Forward.
 - Press lacksquare or lacksquare to select the forwarding type, then press lacksquare .
 - Press \bullet or \bullet to select **Disable** to disable the call forward.
 - Press (oK) to accept the change.

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

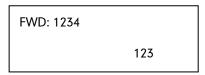
Do one of the following:

- 1. Press , select Features->Call Forward or press when the phone is idle.
- 2. Press (a) or (b) to select the desired account, then press (ok).
- 3. Press \bullet or \bullet to select the desired forwarding type, then press \bullet .
- **4.** Press (•) or (•) to select **Disable** to disable the call forward.
- **5.** Press (oK) to accept the change.

Dynamic Forwarding

To forward an incoming call to another party:

- 1. When the phone is ringing, press _____.
- 2. Enter the number you want to forward the incoming call to.



3. Press (ok) or $(\#_{BEND})$.

The LCD screen prompts a call forwarded message.

FWD To 1234 12 Sep 10:53

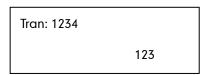
Call Transfer

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

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

To perform a blind transfer:

- 1. Press during a call.
- 2. Enter the number you want to transfer the call to.



3. Press $\underset{\text{TRAN}}{\bigcirc}$ again to complete the transfer.

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

To perform a semi-attended transfer:

- 1. Press during a call.
- 2. Enter the number you want to transfer the call to.
- 3. Press (oK) or (#ssso) to dial out.

4. Press \bigcirc to complete the transfer when receiving ringback.

To perform an attended transfer:

- 1. Press during a call.
- 2. Enter the number you want to transfer the call to.
- 3. Press (ok) or (#) to dial out.
- 4. After the party answers the call, press \bigcirc 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 \bigcirc

Call Waiting

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

To configure call waiting via phone user interface:

- 1. Press .
- 2. Select Features->Call Waiting.

1. Call Waiting:		
◆ *Enable	•	

- 3. Press or to select **Enable** from the **Call Waiting** field.
- **4.** Press (\cdot) or (\cdot) to select **Enable** from the **Play Tone** field.
- (Optional.)Enter the call waiting on code and off code respectively in the CW On Code and CW Off Code field.
- **6.** Press (ok) to accept the change.

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

Conference

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

Note

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

Local Conference

The SIP-T20P 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 on to place a new call. The active call is placed on hold.
- 3. Enter the number of second party and press (ok) or (#) or (#)

Dial: 1234

- **4.** When the second party answers the call, you can consult with him or her before adding to the conference.
- 5. Press \bigcirc again to join all parties in the conference.

To join two calls in a conference:

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

You can press \bigcirc to place the conference on hold. You can press \bigcirc to split the conference call into two individual calls. To drop the conference call, press \bigcirc .

Network Conference

You can use network conference on the SIP-T20P IP phone to conduct a conference with multiple participants.

This feature allows you to perform the following:

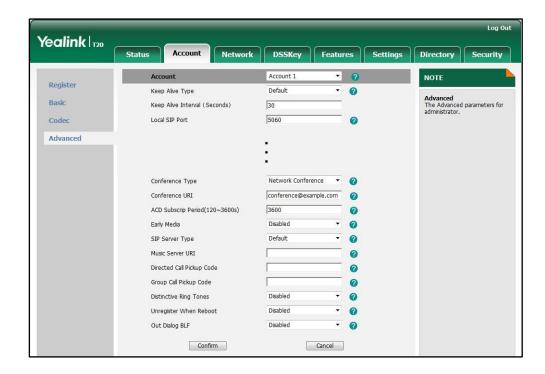
- Join two calls together into a conference call.
- Invite another party into an active conference call.

To use this feature, contact your system administrator for the network conference URI in advance.

To configure network conference via web user interface:

1. Click on Account.

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



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

Note

Network conference is configurable via web user interface only.

To set up a network conference call:

- 1. Place a call to the first party.
- 2. Press to place a new call.

The active call is placed on hold.

- **3.** Enter the second party's number and press (N) or $(\#_{SDD})$.
- **4.** When the second party answers the call, press to add the second party to the conference.
- 5. Press \bigcirc to place a new call.

The conference is placed on hold.

- 6. Enter the number of the new party and then press (ok) or (#===).
- 7. When the new party answers the call, press only to add the new party to the conference.
- 8. Repeat steps 5 to 7 until you have added all intended parties.

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

Call Park

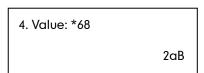
You can use call park to place a call on hold, and then retrieve the call from another phone in the system (for example, a phone in another office or conference room). You can park the active call by pressing the call park key on the phone. If the call is parked successfully, the response is either a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

Note

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

To configure a call park key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.
- **4.** Press (•) or (•) to select **Key Event** from the **Type** field.
- 5. Press () or () to select Call Park from the Key Type field.
- **6.** Press (\cdot) or (\cdot) to select the desired account from the **Account ID** field.
- 7. Enter the call park code in the Value field.



8. Press $(\mathbf{o}\mathbf{k})$ to accept the change.

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

To use call park:

- 1. User on phone A places a call to phone B.
- 2. User on phone A wants to take the call in a conference room for privacy, then presses the call park key on phone A.

User on phone A walks to an available conference room where the phone is designated as phone C. The user dials the call park retrieve code to retrieve the parked call.

The system establishes call between phone C and B.

Note

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

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

Call Pickup

You can use call pickup to answer someone else's incoming call on the phone. The SIPT20P IP phone supports directed call pickup and group call pickup. Directed call pickup is used for picking up a call that is ringing at a specific phone number. Group call pickup is used for picking up a call that is ringing at any phone number in the group. The pickup group should be predefined, contact your system administrator for more information.

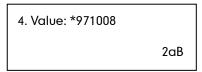
Note

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

Directed Call Pickup

To configure a directed pickup key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.
- 4. Press () or () to select **Key Event** from the **Type** field.
- 5. Press (\cdot) or (\cdot) to select **Directed Pickup** from the **Key Type** field.
- 6. Press () or () to select the desired account from the **Account ID** field.
- Enter the directed pickup code followed by the specific phone number you want to pick up in the Value field.



8. Press $(\mathbf{o}\mathbf{k})$ to accept the change.

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

To pick up a call directly:

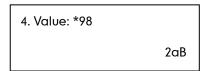
 Press the pickup key on your phone when the specific phone number receives an incoming call.

The incoming call is answered on your phone.

Group Call Pickup

To configure a group pickup key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.
- **4.** Press () or () to select **Key Event** from the **Type** field.
- 5. Press () or () to select **Group Pickup** from the **Key Type** field.
- **6.** Press (\cdot) or (\cdot) to select the desired account from the **Account ID** field.
- 7. Enter the group pickup feature code in the Value field.



8. Press (ok) to accept the change.

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

To pick up a call in the group:

1. Press the group pickup key on your phone when a phone number in the group receives an incoming call.

The incoming call is answered on your phone.

Anonymous Call

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



- 1. Press .
- 2. Select Features->Anonym Call.
- 3. Press () or () to select the desired line from the **Line ID** field.
- **4.** Press () or () to select **Enable** from the **Send Anon** field.

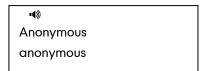
2. Send An	on:
∢ Enable	•

- 5. (Optional.) Press () or () to select **Enable** from the **Anon Code** field.
- 6. (Optional.) Enter the anonymous call on code in the Call On Code field.
- 7. (Optional.) Enter the anonymous call off code in the Call Off Code field.
- **8.** Press $\binom{OK}{OK}$ to accept the change.

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

To place an anonymous call:

Using the specific line on the phone to place a call to phone B.
 The LCD screen of phone B prompts an incoming call from anonymity.



Anonymous Call Rejection

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

To configure anonymous call rejection via phone user interface:

- 1. Press .
- 2. Select Features->Anonym Call.
- 3. Press () or () to select the desired line from the **Line ID** field.
- **4.** Press () or () to scroll to the **Rejection** field.

5. Anon Rejection:
4 Engillo

5. Press or to select **Enable** from the **Anon Rejection** field.

- 6. (Optional.) Enter the anonymous call rejection on code in the **Reject On Code** field.
- 7. (Optional.) Enter the anonymous call rejection off code in the Reject Off Code field.
- **8.** Press (ok) to accept the change.

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-T20P IP phone. Topics include:

- Busy Lamp Field (BLF)
- Call Recording
- Hot Desking
- Intercom
- Multicast Paging
- Music on Hold
- Automatic Call Distribution (ACD)
- Message

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

Busy Lamp Field (BLF)

You can use BLF to monitor a specific line for status changes on the phone. For example, you can configure a BLF key (line key) on the phone for monitoring the status of a friend's line (busy or idle). The BLF key LED flashes green when the friend's line is in use. For more information on BLF key LED indications, refer to LED Instructions on page 4.

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

To configure a BLF key via phone user interface:

- 1. Press enun.
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.
- **4.** Press () or () to select **BLF** from the **Type** field.



5. Press () or () to select the desired account from the **Account ID** field.

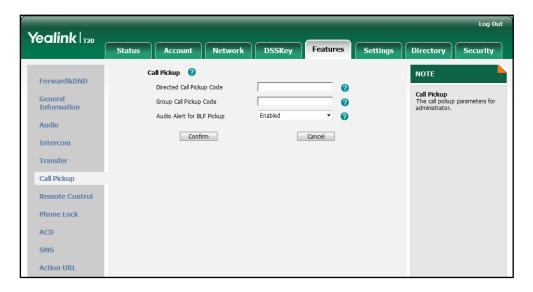
- 6. Enter the phone number you want to monitor in the Value field.
- 7. (Optional.) Enter the pickup code in the Extension field.
- **8.** Press (ok) to accept the change.

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

You can enable audio alert for BLF pickup on the phone. This allows the monitoring phone to play a warning tone when the monitored line receives an incoming call.

To enable audio alert via web user interface:

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



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

Note

Audio alert is configurable via web user interface only.

When the monitored line receives an incoming call, the followings occur on the phone:

- The phone plays a warning tone (if enabled).
- The BLF key LED flashes.

When there is an active call on the IP phone, you can transfer the active call to the monitored phone number directly by pressing the BLF key. The phone transfers the active call differently depending on the transfer mode on DSS key. For more information on the transfer mode on DSS key, refer to transfer in the DSS Keys section on page 37.

Call Recording

You can record calls by pressing a record key on the phone. The SIP-T20P IP phone supports record and URL record.

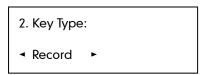
Two ways of call recording:

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

Note

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

- 1. Press MENUS
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.
- **4.** Press () or () to select **Key Event** from the **Type** field.
- 5. Press () or () to select **Record** from the **Key Type** field.



6. Press (ok) to accept the change.

To configure a URL record key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.
- **4.** Press () or () to select **URL Record** from the **Type** field.



- 5. Enter the URL (e.g., http://10.1.2.224/phonerecording.cgi) in the Value field.
- **6.** Press (ok) to accept the change.

Record key or URL record key is configurable via web user interface at the path **DSSKey->Line Key**.

The record and URL record keys control recording, and are available:

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

The record and URL record key is not available when:

- There are no connected calls on your phone
- You place a new call

To record a call:

- Press the record key or URL record key during a call.
 If the recording starts successfully, the record or URL record key LED will flash green.
- Press the record key or URL record key again to stop recording.The record or URL record key LED goes out.

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

Note

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

Hot Desking

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

You can use hot desking on the SIP-T20P IP phone to log out of the existing accounts and then log into a new account, that is, many users can share the phone resource in different time. To use this feature, you need to configure a hot desking key in advance.

Note

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

1.	Press MENU 2.	
2.	Select Features->DSS Keys.	
3.	Select the desired DSS key.	
4.	Press or to select Key Event from the Type field.	
5.	Press or to select Hot Desking from the Key Type field.	
	2. Key Type:	
	2. Key Type: ◆ Hot Desking ▶	
6.	Press (ok) to accept the change.	
Ho	t desking key is configurable via web user interface at the path DSSKey->Line Key	
То	use hot desking:	
1.	Press the hot desking key when the phone is idle.	
	The LCD screen prompts "Clear all account?".	
	Warning	
	Clear all account?	
2.	Press $(\mathbf{o}\mathbf{k})$, registration configurations of all accounts will be cleared immediately	
	The login wizard will be shown as below:	
	1. Register Name:	
	123	

 Enter the desired values in the Register Name ,User Name,Password,SIP Server1/2 and Outbound Proxy Server fields respectively.Contact your system administrator

for more information.

4. Press (ok) to login.

To configure a hot desking key via phone user interface:

Intercom

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

Note

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

Outgoing Intercom Calls

To configure an intercom key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.
- **4.** Press () or () to select **Intercom** from the **Type** field.



- 5. Select the desired account from the Account ID field.
- 6. Enter the remote extension number in the Value field.
- 7. Press $(\mathbf{o}\mathbf{k})$ to accept the change.

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

To place an intercom call:

- Press the intercom key when the phone is idle.
 The phone is automatically connected to the extension specified in the Value field.
- 2. Press the intercom key again or \bigcirc to end the intercom call.

Incoming Intercom Calls

The SIP-T20P IP phone supports automatically to answer an incoming intercom call by default. The phone automatically plays a warning tone when it receives an incoming intercom call. In addition, you can enable the phone to mute the microphone when it automatically answers an incoming intercom call. You can also enable the phone to automatically answer an incoming intercom call while there is already an active call on the phone, the active call is placed on hold.

Intercom features you need to know:

Intercom Feature	Description
A coopt Intercore	Enable or disable the IP phone to automatically
Accept Intercom	answer an incoming intercom call.
Intercent Mute	Enable or disable the microphone on the IP phone for
Intercom Mute	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.

To configure intercom features via phone user interface:

- 1. Press .
- 2. Select Features->Intercom.
- 3. Make the desired changes.



4. Press (0K) to accept the change.

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

Accept Intercom

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

Note

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

Intercom Mute

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

Intercom Tone

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

Intercom Barge

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

Multicast Paging

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

Sending RTP Stream

To configure a multicast paging key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.
- 4. Press () or () to select **Key Event** from the **Type** field.
- 5. Press (\cdot) or (\cdot) to select **Multi-Paging** from the **Key Type** field.

2. Key Type:	
■ Multi-Paging	•

6. Enter the multicast IP address and port number (e.g., 224.5.6.20:10008) in the **Value** field

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

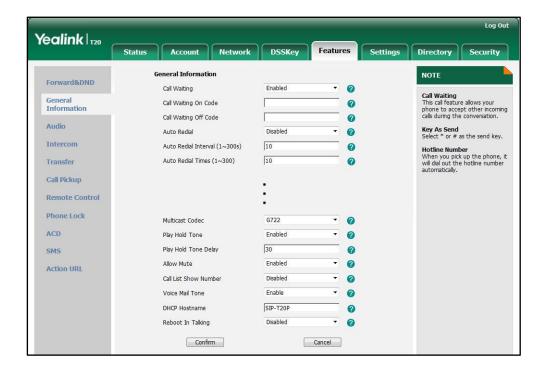
7. Press (ok) to accept the change.

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

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

To configure a default codec for multicast paging:

- 1. Click on Features->General Information.
- Select the desired codec from the pull-down list of Multicast Codec.
 The default codec is G722.



3. Click Confirm to accept the change.

Note

If G722 codec is used for multicast paging, the LCD screen prompts "**HD**" to indicate that it is providing high definition voice.

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

To send RTP stream:

1. Press the multicast paging key when the phone is idle.

The phone sends RTP to a preconfigured multicast address (IP: Port). Any phone in the local network then listens to the RTP on the preconfigured multicast address (IP: Port). For both sending and receiving of the multicast RTP, there is no SIP signaling involved.

The multicast paging key LED illuminates solid green.

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

224.5.6.20:10008 HD 00:35

- 2. Press \bigcirc to place the current multicast RTP session on hold.
- 3. Press to cancel the multicast RTP session.

Note

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

Receiving RTP Stream

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

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

Paging Barge

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

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

Paging Priority Active

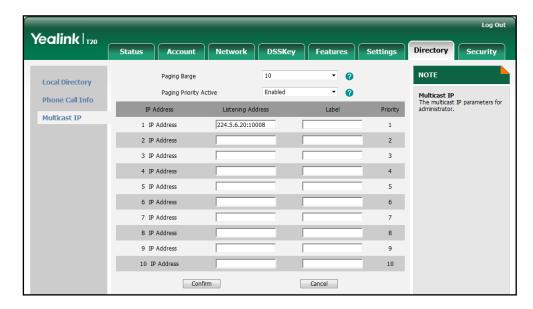
Paging priority active parameter decides how the phone handles the incoming multicast paging calls when there is already a multicast paging call on the phone. If enabled, the phone will ignore incoming multicast paging calls with lower priorities, otherwise, the phone will answer incoming multicast paging calls automatically and place the previous multicast paging call on hold. If disabled, the phone will automatically ignore all incoming multicast paging calls.

To configure multicast listening addresses via web user interface:

- 1. Click on Directory->Multicast IP.
- 2. Select the desired value from the pull-down list of Paging Barge.

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

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.



- Enter the label in the Label field.
 Label will appear on the LCD screen when receiving the RTP multicast.
- 6. Click Confirm to accept the change.

Note

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

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

Multicast listening addresses is configurable via web user interface only.

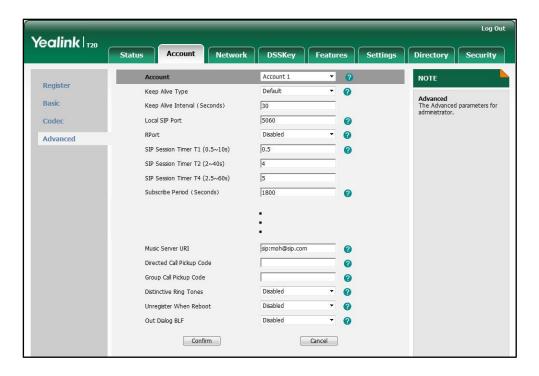
Music on Hold

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

To configure music on hold server via web user interface:

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

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



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

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

Note

All involved parties cannot use encrypted RTP.

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

Automatic Call Distribution (ACD)

ACD is often used in offices for customer service, such as call center. The ACD system handles the large volumes of incoming calls from callers who have no need to talk to a specific person but who require assistance from any of multiple persons at the earliest opportunity. ACD on the SIP-T20P IP phone allows the ACD system to distribute large volumes of incoming calls to the registered ACD users. To use this feature, you should configure an ACD key in advance.

Note

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

To configure an ACD key via phone user interface:

- 1. Press .
- 2. Select Features->DSS Keys.
- 3. Select the desired DSS key.

4. Press () or () to select **ACD** from the **Type** field.

1. Type: **→** ACD **→**

5. Press (o) to accept the change.

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

To log into the ACD system:

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

The LCD screen prompts you the following information:

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

Password: the password used to log into the queue.

1. User ID:

2. Press (ok) to login.

Note

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

After configuring an ACD key, you can press the ACD key to log into the ACD system. After logging in, you are ready to receive calls from the ACD system. You can press the ACD key to show your current ACD user status. You can press the OK key to set your ACD user status to **Available** /**Unavailable**. ACD user status synchronizes on both IP phone and ACD system. When you set the ACD user status to be available, the ACD key LED illuminates solid green, and then the server begins distributing calls to your phone. When you set the ACD user status to be unavailable, the ACD key LED flashes green, and then the server temporarily stops distributing calls to your phone. To log out of the ACD system, press (\times) .

Note

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

Message

Voice Mail

You can leave voice mails for someone else on the SIP-T20P IP phone. You can also listen to the voice mails stored in a centralized location. When receiving a new voice mail, the phone will play a warning tone and the MESSAGE key LED will illuminate and the LCD screen will display a flashing icon.



Note

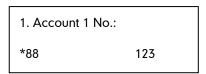
The voice mail feature is not available on all servers. Contact your system administrator for more information.

To leave a voice mail:

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

To configure voice mail access codes via phone user interface:

- 1. Press .
- 2. Select Messages->Set Voice Mail.



- **3.** Press lacksquare or lacksquare to select the account you want to set.
- 4. Press to select the proper input mode and then enter the voice mail access code (e.g., *88).
- **5.** Press (ox) to accept the change.

Note

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

To listen to the voice mails:

1. When the phone user interface prompts receiving new voice mails, press

- or $(\mathbf{o}\mathbf{k})$ to dial out the voice mail access code.
- 2. Follow the voice prompt to listen to voice mails.

Note

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

When all new voice mails are retrieved, the MESSAGE key LED will go out.

To view the voice mail via phone user interface:

- 1. Press .
- 2. Select Messages->View Voice Mail.

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



3. Select an account and then press (or) to listen to the voice mails.

Message Waiting Indicator (MWI)

The SIP-T20P 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 prompt message (including a voice mail icon) on the LCD screen, and the MESSAGE key LED illuminates solid green. This will be cleared when you retrieve all voice mails or delete them.

The MWI service is unsolicited for some servers, so the SIP-T20P IP phone only handles the MWI messages sent from the server. But for other servers, the MWI service is solicited, so the SIP-T20P IP phone must enable subscription for MWI.

Note

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

The MWI Subscription parameters you need to know:

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

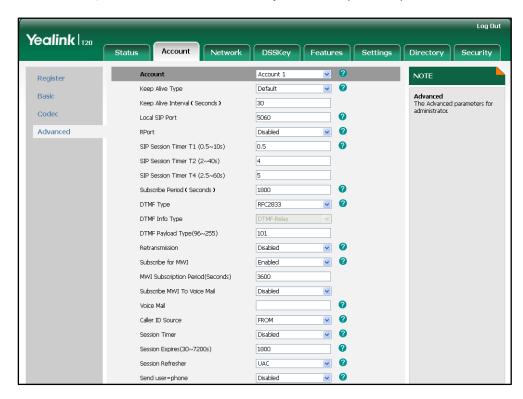
Option	Description
	subscribe for MWI and the voice mail number in advance.

Note

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

To enable MWI subscription via web user interface:

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



6. Click Confirm to accept the change.

To enable Subscribe MWI to Voice Mail via web user interface:

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

Yealink 120 Status Network DSSKey Features Settings Account 1 NOTE Register Keep Alive Type 0 **Advanced** The Advanced parameters for administrator. Keep Alive Interval (Seconds) 30 Codec 0 Local SIP Port 5060 0 Advanced RPort Disabled SIP Session Timer T1 (0.5~10s) 0.5 SIP Session Timer T2 (2~40s) SIP Session Timer T4 (2.5~60s) <u>v</u> 0 RFC2833 DTMF Info Type 101 DTMF Payload Type(96~255) 0 Disabled Retransmission Subscribe for MWI Enabled MWI Subscription Period(Seconds) 3600 Subscribe MWI To Voice Mail Caller ID Source FROM 0 Disabled Session Timer 0 Session Expires(30~7200s) 1800 v 0 Session Refresher UAC Send user=phone Disabled 0

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

6. Click Confirm to accept the change.

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

Note

MWI subscription is configurable via web user interface only.

Troubleshooting

This chapter provides general troubleshooting information to help to solve the problems you might encounter when using your SIP-T20P IP phone.

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

Why is the LCD screen blank?

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

Why does the phone display "Network Unavailable"?

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

Why does the phone display "No Service"?

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

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

Check if you have configured the phone to obtain the time and date from the SNTP server automatically. If the phone fails to connect the SNTP server, you need to configure the time and date manually.

How do I find the basic information of the phone?

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

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

Three ways to obtain the MAC address of a phone:

- You can ask your supplier for shipping information sheet which includes MAC addresses according to the corresponding PO (Purchase Order).
- You can find the MAC address in the label of carton box.
- You can also find the MAC address from the phone's bar code on the back of the phone.

Why can't I get a dial tone?

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

Why doesn't the phone ring?

Check that the ringer volume on the phone. To adjust the ringer volume setting, press the **Volume** key when the phone is idle. For more information, refer to Volume on page 23.

Why can't I receive calls?

- Check that the SIP registration with your system administrator.
- Check that the DND (Do Not Disturb) mode is deactivated on the phone. Refer to Do Not Disturb (DND) on page 61.
- Check that call forward is disabled on the phone. Refer to Call Forward on page
- Check whether the caller number is stored in the blacklist directory. Refer to Blacklist on page 32.

Why is my handset not working?

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

Why is my headset not working?

- Check that the headset cord is fully connected to the headset jack on the phone.
 Refer to Phone Installation on page 11.
- Check that the headset mode is activated. Refer to Headset Mode

Activation/Deactivation on page 35.

 Check that the headset volume is adjusted to an appropriate level. Refer to Volume on page 23.

What will happen if I connect both PoE cable and power adapter? Which has the priority?

The phones manufactured before February 2010 use the power adapter preferentially, otherwise the phones use PoE preferentially.

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

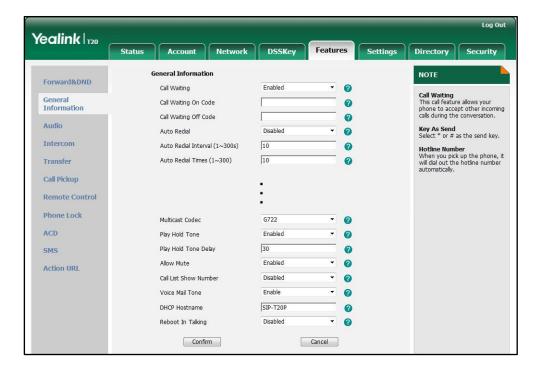
Both user name and register name are defined by the server. A user name is used to identify the account while a register name matched with a password is used for authentication if the server requires. Display name is the caller ID that will be displayed on the callee's LCD screen. Server configurations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

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

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

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

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



4. Click Confirm to accept the change.

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** and **Confirm Password** fields.



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

Note

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

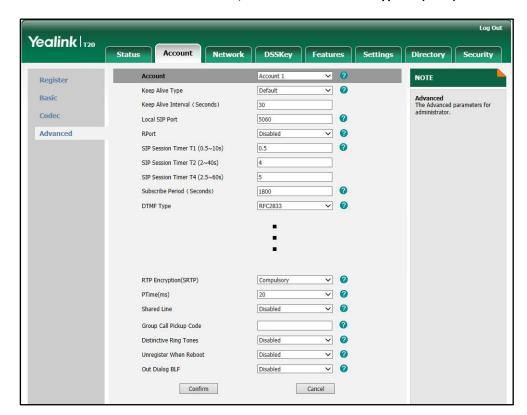
User password is configurable via web user interface only.

How to make a call using SRTP?

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

To enable SRTP via web user interface:

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



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

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

Note

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

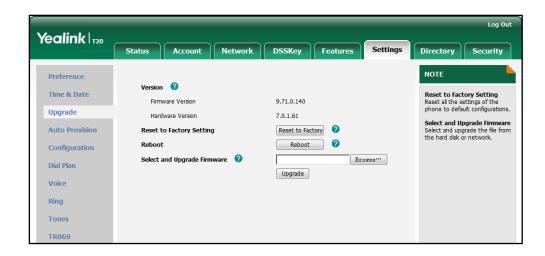
SRTP is configurable via web user interface only.

How to reboot the phone?

To reboot the phone via web user interface:

1. Click on Settings->Upgrade.

2. Click **Reboot** to reboot the phone.



Note

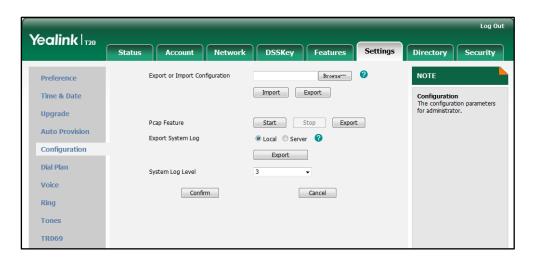
Any reboot of your phone may take a few minutes.

How to export PCAP trace?

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

To export a PCAP trace via web user interface:

- 1. Click on **Settings-> Configuration**.
- 2. Click Start to begin recording signal traffic.
- 3. Recreate the error to be documented in the trace.
- 4. Click Stop to end recording.
- **5.** Click **Export** to open file download window, and then save the file to your local system.



How to export system log?

We may need you to provide a system log to help analyze your problem.

To export a system log via web user interface:

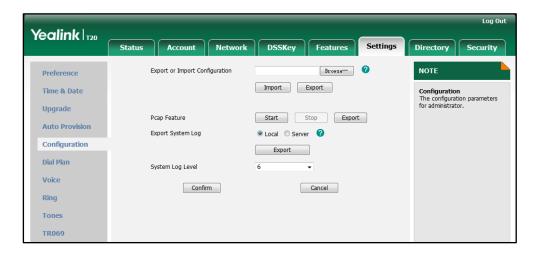
- 1. Click on **Settings->Configuration**.
- 2. Select 6 from the pull-down list of System Log Level.
- 3. Click Confirm to accept the change.

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

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

After rebooting the phone, the system log level is set as 6, the debug level.

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



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

Note

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

How to export/import phone configurations?

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

To export the phone configurations via web user interface:

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

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



To import the phone configurations via web user interface:

- 1. Click on **Settings->Configuration**.
- 2. Click **Brower** to locate a configuration file from your local system.
- 3. Click **Import** to import the configuration file.

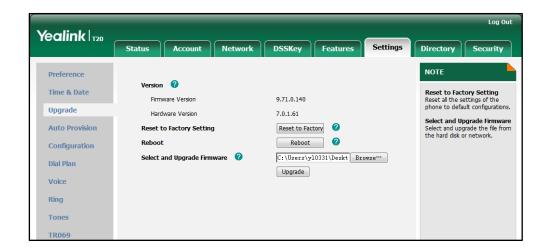
Note

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

How to upgrade firmware?

To upgrade firmware via web user interface:

- 1. Click on **Settings**->**Upgrade**.
- 2. Click **Browse** to locate the firmware from your local system.



3. Click **Upgrade** to upgrade the firmware.

The web user interface prompts "Firmware of the SIP Phone will be updated. It will take 5 minutes to complete. Please don't power off!".

4. Click **OK** to confirm upgrading.

How to reset the phone?

Reset the phone when other troubleshooting suggestions do not correct the problem. You need to note that all customized settings will be overwritten after resetting. So we recommend asking your system administrator for advice before resetting the phone.

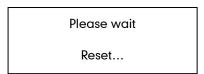
To reset the phone via phone user interface:

- 1. Press .
- 2. Select Settings->Advanced (password: admin).
- 3. Press or to scroll to Reset Factory, and then press ok .
 The LCD screen prompts "Reset To Factory?".

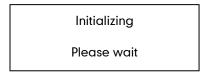


4. Press (ok) to confirm.

The LCD screen prompts "Please wait Reset...".



The LCD screen prompts "Initializing Please wait".



The phone will be reset to factory sucessfully after startup.

Note

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

Regulatory Notices

Service Agreements

Contact your Yealink Authorized Reseller for information 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.

⚠ General Requirements

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

/!\ Environmental Requirements

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

Operating Requirements

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

Cleaning Requirements

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

Appendix A - Time Zones

Time Zone	Time Zone Name	
-11:00	Samoa	
-10:00	United States-Hawaii-Aleutian	
-10:00	United States-Alaska-Aleutian	
-09:00	United States-Alaska Time	
-08:00	Canada(Vancouver, Whitehorse)	
-08:00	Mexico(Tijuana, Mexicali)	
-08:00	United States-Pacific Time	
-07:00	Canada(Edmonton, Calgary)	
-07:00	Mexico(Mazatlan, Chihuahua)	
-07:00	United States-Mountain Time	
-07:00	United States-MST no DST	
-06:00	Canada-Manitoba(Winnipeg)	
-06:00	Chile(Easter Islands)	
-06:00	Mexico(Mexico City, Acapulco)	
-06:00	United States-Central Time	
-05:00	Bahamas(Nassau)	
-05:00	Canada(Montreal, Ottawa, Quebec)	
-05:00	Cuba(Havana)	
-05:00	United States-Eastern Time	
-04:30	Venezuela(Caracas)	
-04:00	Canada(Halifax, Saint John)	
-04:00	Chile(Santiago)	
-04:00	Paraguay(Asuncion)	
-04:00	United Kingdom-Bermuda(Bermuda)	
-04:00	United Kingdom(Falkland Islands)	
-04:00	Trinidad&Tobago	
-03:30	Canada-New Foundland(St.Johns)	
-03:00	Denmark-Greenland(Nuuk)	
-03:00	Argentina(Buenos Aires)	
-03:00	Brazil(no DST)	
-03:00	Brazil(DST)	
-02:00	Brazil(no DST)	
-01:00	Portugal(Azores)	
0	GMT	
0	Greenland	
0	Denmark-Faroe Islands(Torshavn)	
0	Ireland(Dublin)	
0	Portugal(Lisboa, Porto, Funchal)	
0	Spain-Canary Islands(Las Palmas)	
0	United Kingdom(London)	
0	Morocco	
+01:00	Albania(Tirane)	
+01:00	Austria(Vienna)	
+01:00	Belgium(Brussels)	
+01:00	Caicos	
+01:00	Chad	
+01:00	Croatia(Zagreb)	
+01:00	Czech Republic(Prague)	
+01:00	Denmark(Kopenhagen)	
+01:00	France(Paris)	
+01:00	Germany(Berlin)	
T01.00	Oermany(Denin)	

Time Zone	Time Zone Name	
+01:00	Hungary(Budapest)	
+01:00	Italy(Rome)	
+01:00	Luxembourg(Luxembourg)	
+01:00	Macedonia(Skopje)	
+01:00	Netherlands(Amsterdam)	
+01:00	Namibia(Windhoek)	
+02:00	Estonia(Tallinn)	
+02:00	Finland(Helsinki)	
+02:00	Gaza Strip(Gaza)	
+02:00	Greece(Athens)	
+02:00	Israel(Tel Aviv)	
+02:00	Jordan(Amman)	
+02:00	Latvia(Riga)	
+02:00	Lebanon(Beirut)	
+02:00	Moldova(Kishinev)	
+02:00	Russia(Kaliningrad)	
+02:00	Romania(Bucharest)	
+02:00	Syria(Damascus)	
+02:00	Turkey(Ankara)	
+02:00	Ukraine(Kyiv, Odessa)	
+02:00	Syria(Damascus)	
+03:00	East Africa Time	
+03:00	Iraq(Baghdad)	
+03:00	Russia(Moscow)	
+03:30	Iran(Teheran)	
+04:00	Armenia(Yerevan)	
+04:00	Azerbaijan(Baku)	
+04:00	Georgia(Tbilisi)	
+04:00	Kazakhstan(Aktau)	
+04:00	Russia(Samara)	
+04:30	Afghanistan	
+05:00	Kazakhstan(Aqtobe)	
+05:00	Kyrgyzstan(Bishkek)	
+05:00	Pakistan(Islamabad)	
+05:00	Russia(Chelyabinsk)	
+05:30	India(Calcutta)	
+06:00	Kazakhstan(Astana, Almaty)	
+06:00	Russia(Novosibirsk, Omsk)	
+07:00	Russia(Krasnoyarsk)	
+07:00	Thailand(Bangkok)	
+08:00	China(Beijing)	
+08:00	Singapore(Singapore)	
+08:00	Australia(Perth)	
+09:00	Korea(Seoul)	
+09:00	Japan(Tokyo)	
+09:30	Australia(Adelaide)	
+09:30	Australia(Darwin)	
+10:00	Australia(Sydney, Melbourne, Canberra)	
+10:00	Australia(Brisbane)	
+10:00	Australia(Hobart)	
+10:00	Russia(Vladivostok)	
+10:30	Australia(Lord Howe Islands)	
+11:00	New Caledonia(Noumea)	
+12:00	New Zealand(Wellington, Auckland)	
+12:45	New Zealand(Chatham Islands)	
+13:00	Tonga(Nukualofa)	
5.00	1 - 1 - 3 - 1 - 1 - 1 - 1 - 1 - 1 - 1 -	

Index

A	
About This Guide v	Documentations 7
Account Management 43	DSS Keys 37
Account Registration 43	Dual Headset 36
Adding Contacts 27	Dynamic Forwarding 63
Adding Groups 26	E
Administrator password 20	Editing Contacts 29
Always Forward 64	Editing Groups 27
Anonymous Call 75	Emergency Number 50
Anonymous Call Rejection 76	Ending Calls 56
Answering Calls 55	-
Area Code 48	G
Attaching Stand 11	General Settings 17
Attended Transfer 69	Getting Started 9
Audio Settings 23	Н
Automatic Call Distribution (ACD) 90	п
B	Hardware Components Instruction 1 Headset Mode Activation/Deactivation
Basic Network Settings 14 Blacklist 32	35
Blacklist 32 Blind Transfer 69	Headset Prior 36
Block Out 49	Hotline 52
Busy Forward 64	Hot Desking 82
Busy Lamp Field (BLF) 79	1
С	Icon Instructions 3
Call Completion 59	Idle Screen 16
Call Forward 64	Importing/Exporting Contact lists 30 Incoming Intercom Calls 84
Call History Management 33	Incoming Intercom Calls 84 Intercom 84
Call Hold/Resume 60	intercom 04
Call Mute 60	K
Call Park 73	Key as Send 20
Call Pickup 74	Keypad Lock 21
Call Return 60	•
Call Recording 80	L
Call Weiting 70	Language 17
Call Waiting 70	LED Instructions 4
CE Mark Warning ii Conference 70	Limitation of Liability 107
Contact Management 26	Line Keys 37
Copyright i	Live Dialpad 51
Customizing Your Phone 17	Local Conference 71
-	Local Directory 26
D	Logo Customization 34
Declaration of Conformity ii	M
Deleting Contacts 29	Message Waiting Indicator 93
Deleting Groups 27	Message Waiting Indicator 93 Multicast Paging 86
Dial-now 46	Music on Hold 89
Dial Plan 44	
Do Not Disturb (DND) 61	N

Network Conference 71 Network Connection 11 No answer Forward 64
0
Outgoing Intercom Call 84 Overview 1
P
Packaging Contents 9 Phone Installation 11 Phone Initialization 13 Phone Status 13 Phone User Interface 4 Placing Calls 53 Placing Calls to Contacts 29 Programmable Keys 42
R
Receiving RTP Stream 88 Redialing Numbers 57 Replace Rule 45 Ring Tones 24
S
Safety Instructions 107 Searching for Contacts 30 Semi-attended Transfer 69 Sending RTP Stream 86 Service Agreement 107
Т
Time and Date 18 Time Zone 111 Troubleshooting 93
V
Voice Mail 92 Volume 23
W
Warranty ii Web User Interface 5 WEEE Warning ii