

Yealink



HD 

SIP-T20P
SIP-T20

Enterprise IP Phone

User Guide

Version 73.40

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

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

WEEE Warning



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

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

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Yealink SIP-T20P/SIP-T20 firmware contains third-party software under the GNU General Public License (GPL). Yealink uses software under the specific terms of the GPL. Please refer to the GPL for the exact terms and conditions of the license.

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

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

About This Guide

Thank you for choosing the SIP-T20P/SIP-T20 IP phone, exquisitely designed to provide business telephony features, such as Call Hold, Call Transfer, Busy Lamp Field, Multicast Paging and Conference over an IP network. The difference between the SIP-T20P and SIP-T20 IP phone is that the former supports PoE, while the latter does not.

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/SIP-T20 IP phone.

Note

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

In This Guide

Topics provided in this guide include:

- Chapter 1 [Overview](#)
- Chapter 2 [Getting Started](#)
- Chapter 3 [Customizing Your Phone](#)
- Chapter 4 [Basic Call Features](#)
- Chapter 5 [Advanced Phone Features](#)

Summary of Changes

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

Changes for Release 73, Guide Version 73.40

The following section is new:

- [Optional Accessor](#) on page 10

Major updates have occurred to the following sections:

- [DSS Keys](#) on page 39
- [Appendix A - Time Zones](#) on page 115

Changes for Release 73, Guide Version 73.16

The following sections are new:

- [BLF List](#) on page 82

Major updates have occurred to the following sections:

- [LED Instructions](#) on page 4
- [Ring Tones](#) on page 26
- [Anonymous Call Rejection](#) on page 78
- [Multicast Paging](#) on page 90
- [Troubleshooting](#) on page 101
- [Appendix A - Time Zones](#) on page 115

Changes for Release 72, Guide Version 72.25

Major updates have occurred to the following sections:

- [Documentations](#) on page 7
- [Packaging Contents](#) on page 9
- [Phone Status](#) on page 13

Changes for Release 72, Guide Version 72.1

Documentations of the SIP-T20 IP phone have also been added.

Major updates have occurred to the following sections:

- [LED Instructions](#) on page 4
- [Anonymous Call](#) on page 77

Changes for Release 71, Guide Version 71.165

Major updates have occurred to the following sections:

- [Appendix A - Time Zones](#) on page 115

Changes for Release 71, Guide Version 71.140

Major updates have occurred to the following sections:

- [LED Instructions](#) on page 4
- [Ring Tones](#) on page 26

- [Anonymous Call](#) on page 77
- [Troubleshooting](#) on page 101

Changes for Release 71, Guide Version 71.125

Major updates have occurred to the following sections:

- [Appendix A - Time Zones](#) on page 115

Changes for Release 71, Guide Version 71.110

Major updates have occurred to the following sections:

- [Basic Network Settings](#) on page 15
- [Phone Lock](#) on page 23
- [Contact Management](#) on page 27
- [DSS Keys](#) on page 39

Changes for Release 70, Guide Version 70

Major updates have occurred to the following sections:

- [Phone Lock](#) on page 23
- [Volume](#) on page 25
- [Ring Tones](#) on page 26
- [Call Completion](#) on page 61
- [DSS Keys](#) on page 39
- [Do Not Disturb \(DND\)](#) on page 63
- [Call Forward](#) on page 66
- [Busy Lamp Field \(BLF\)](#) on page 81

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Overview

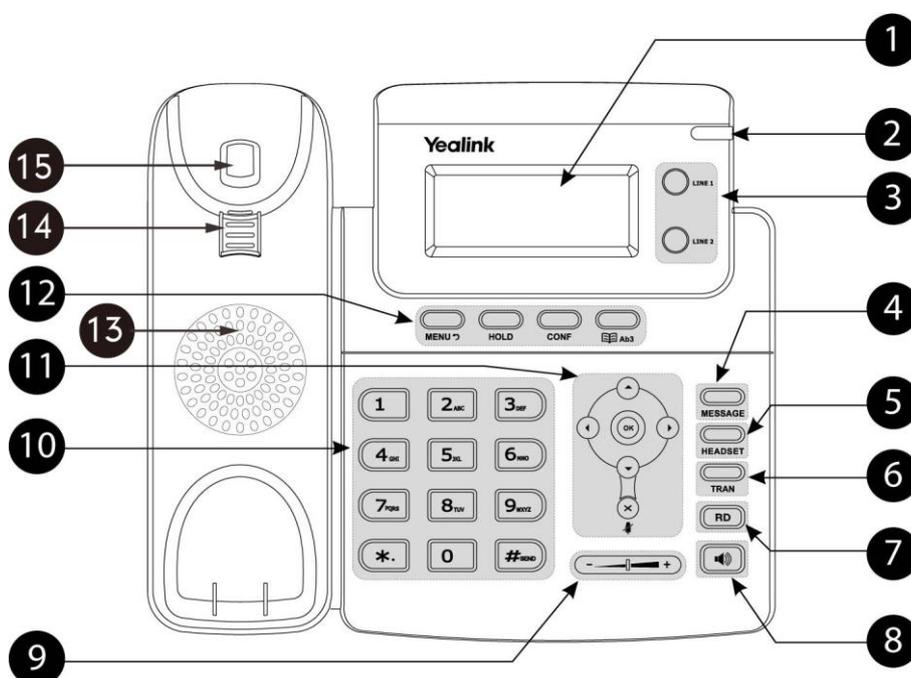
This chapter provides the overview of the SIP-T20P/SIP-T20 IP phone. Topics include:

- [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/SIP-T20 IP phone are the LCD screen and the keypad.



Hardware component instructions of the SIP-T20P/SIP-T20 IP phone are:

	Item	Description
①	LCD Screen	Shows information about calls, messages, time, date and other relevant data. <ul style="list-style-type: none"> • Call information — Caller ID, call duration • 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 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, speaker, and ringer.
⑩	Keypad	Provides the digits, letters and special characters in context-sensitive applications.
⑪		Scroll through the displayed information.
		Confirms actions or answers an incoming call.
		Cancels actions, rejects incoming calls, mutes or un-mutes calls.
⑫	MENU Key	Enters the main menu of phone or returns to the previous interface.
	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 screen or switches the input mode.
⑬	Speaker	Provides ringer and hands-free (speakerphone) audio output.
⑭	Hookswitch Tab	Secures the handset in the handset cradle when the IP phone is mounted vertically.
⑮	Hookswitch	Picking up the handset from the handset cradle, the hookswitch bounces and the phone connects to the line, laying the handset down on the handset cradle, the phone disconnects from the line.

Icon Instructions

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

Icon	Description
	Network is unavailable
	Hands-free speakerphone mode
	Handset mode
	Headset mode
123	Numeric input mode
abc	Multi-lingual lowercase letters input mode
ABC	Multi-lingual uppercase letters input mode
2aB	Alphanumeric input mode
	Voice Mail
AA	Auto Answer
DND	Do Not Disturb
	Call Forward/Forwarded Calls
	Call Mute
	Phone Lock
	Received Calls
	Placed Calls
	Missed Calls

LED Instructions

Power Indicator LED

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

Line key LED

LED Status	Description
Solid green	The line is in conversation. 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 a BLF key)

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

Note

The above introduces the default LED status. The statuses of the power indicator LED and BLF key are configurable via web user interface. For more information, refer to *Yealink SIP-T2xP IP Phones Administrator Guide*.

User Interfaces

Two ways to customize configurations of your SIP-T20P/SIP-T20 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 19.

Web User Interface

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

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

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

Options	Phone User Interface	Web User Interface
--Ring Tones	√	
--Contact Management	×	
--Local Directory	√	
--Blacklist	√	
--Call History Management	√	
--Logo Customization		
--DSS Keys	√	
--Account Registration	√	
--Dial Plan		
--Emergency Number	×	
--Live Dialpad	×	
--Hotline	√	
Basic Call Features		
--Auto Answer	√	
--Auto Redial	√	
--Call Completion	√	
-- Recall	√	
-- Do Not Disturb (DND)	√	
--Call Forward	√	
--Call Transfer	√	
--Call Waiting	√	
--Conference	×	
--Call Park	√	
--Call Pickup	√	
--Anonymous Call	√	
--Anonymous Call Rejection	√	
Advanced Phone Features		√
-- Busy Lamp Field (BLF)	√	
--BLF List	×	
--Call Recording	√	
--Hot Desking	√	
--Intercom	√	
--Multicast Paging	×	
--Music on Hold	×	
--Messages	√	
SIP Account		√
--User Options	×	
--Activation	√	
--Label	√	
--Display Name	√	
--Register Name	√	

Options	Phone User Interface	Web User Interface
--User Name	√	
--Password	√	
--SIP Server1/2	√	
--Server Option	×	
--Registrar Port	×	
--Outbound Status	√	
--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/SIP-T20 IP phone.

Name	Contents	Where found	Language
Quick Start Guide	Basic call features and phone customization	In the package	English
User Guide	Phone or web user interface settings Basic call features and advanced phone features	On the website	English

Note You can also download the latest documentations online:
http://www.yealink.com/SupportDownloadfiles_detail.aspx?Cateld=187&flag=142.

Getting Started

This chapter provides basic installation instructions and information for obtaining the best performance with the SIP-T20P/SIP-T20 IP phone. Topics include:

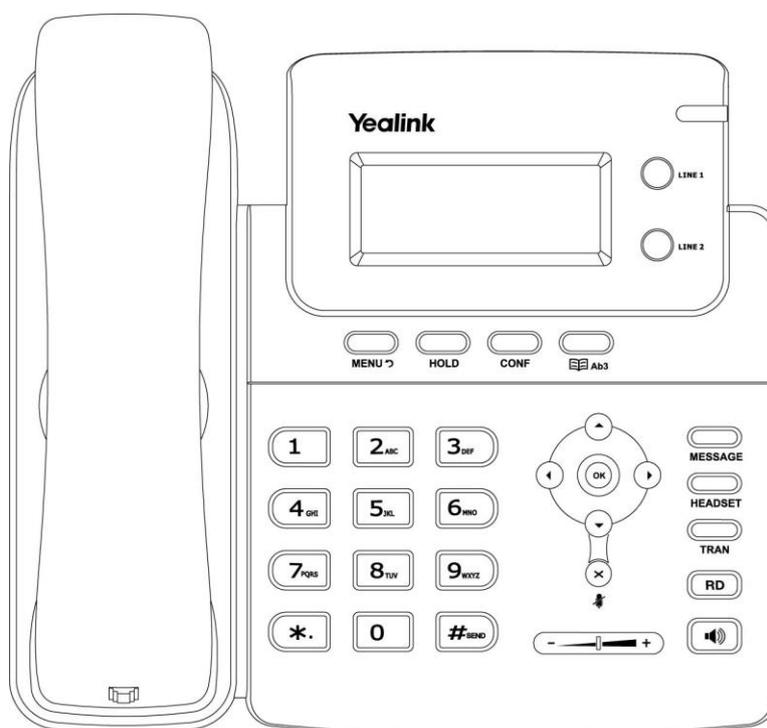
- [Packaging Contents](#)
- [Phone Installation](#)
- [Phone Initialization](#)
- [Phone Status](#)
- [Basic Network Settings](#)
- [Registration](#)
- [Idle Screen](#)

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

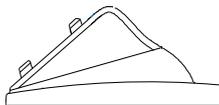
Packaging Contents

The following components are included in your SIP-T20P/SIP-T20 IP phone package:

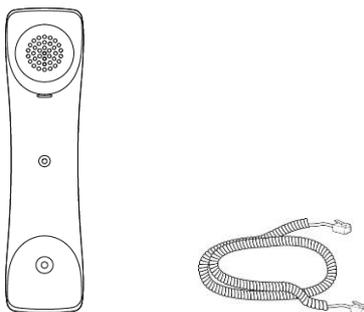
- **SIP-T20P/SIP-T20 IP phone**



- **Phone Stand**



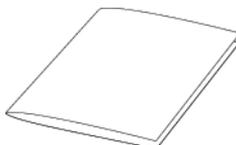
- **Handset & Handset Cord**



- **Ethernet Cable**



- **Quick Start Guide**

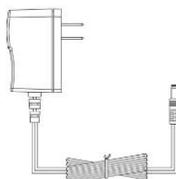


Check this list before installation. If you find anything missing, contact your system administrator.

Optional Accessories

The following items are optional accessories for your SIP-T20P/SIP-T20 IP phone. You need to purchase them separately if required.

- **Power Adapter**



- Headset



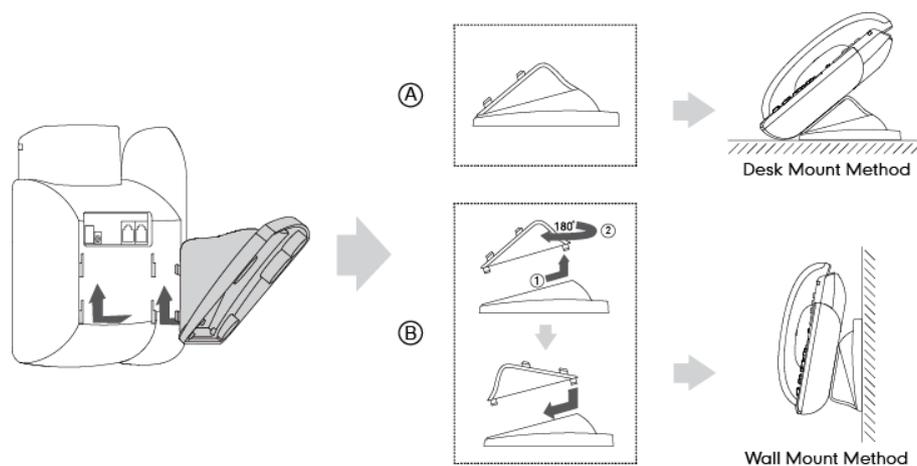
Phone Installation

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

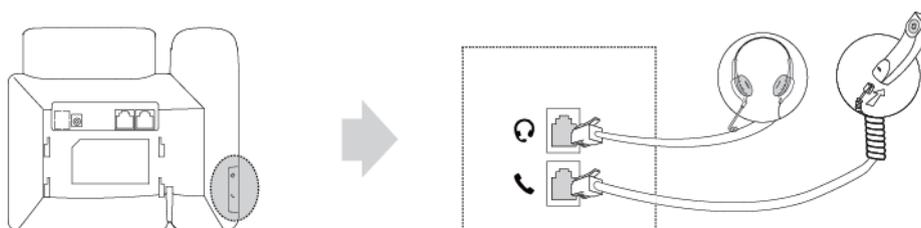
This section introduces how to install the phone:

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

1) Attach the Stand



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

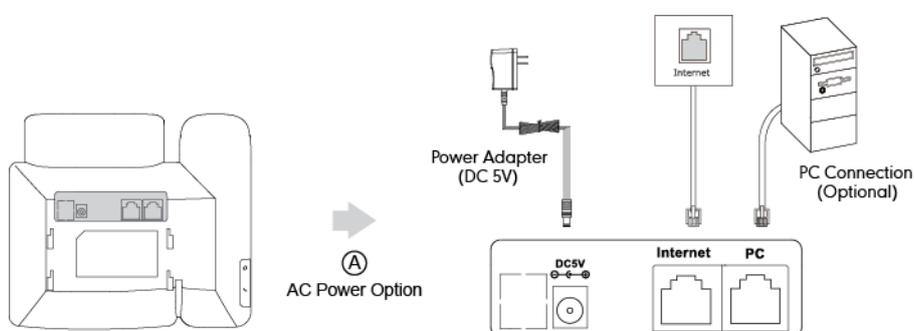
You have two options for power and network connections. Your system administrator will advise you which one to use.

- 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.

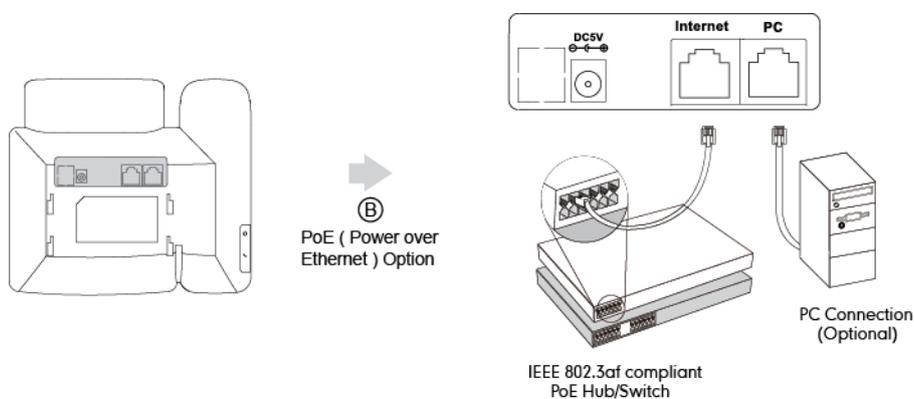


Power over Ethernet

With the included or a regular Ethernet cable, the SIP-T20P IP phone can be powered from a PoE-compliant switch or hub. PoE is not applicable to the SIP-T20 IP phone.

To connect the PoE for the SIP-T20P IP phone:

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



Note

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

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

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

Phone Initialization

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

Automatic Phone Initialization

The phone finishes the initialization by loading the saved configuration. The LCD screen displays "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 15.

Phone Status

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

Available information of phone status includes:

- Network status (e.g., IPv4 status, IP mode and MAC address).

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

Note

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

To view the phone status via phone user interface:

1. Press  or press  and then select **Status**.
2. Press  or  to scroll through the list and view the specific information.

1. IP Address: 10.3.6.219

To view the phone status via web user interface:

1. Open a web browser on your computer.
2. Enter the IP address in the browser's address bar, and then press **Enter**.
3. Enter the user name (admin) and password (admin) in the login page.



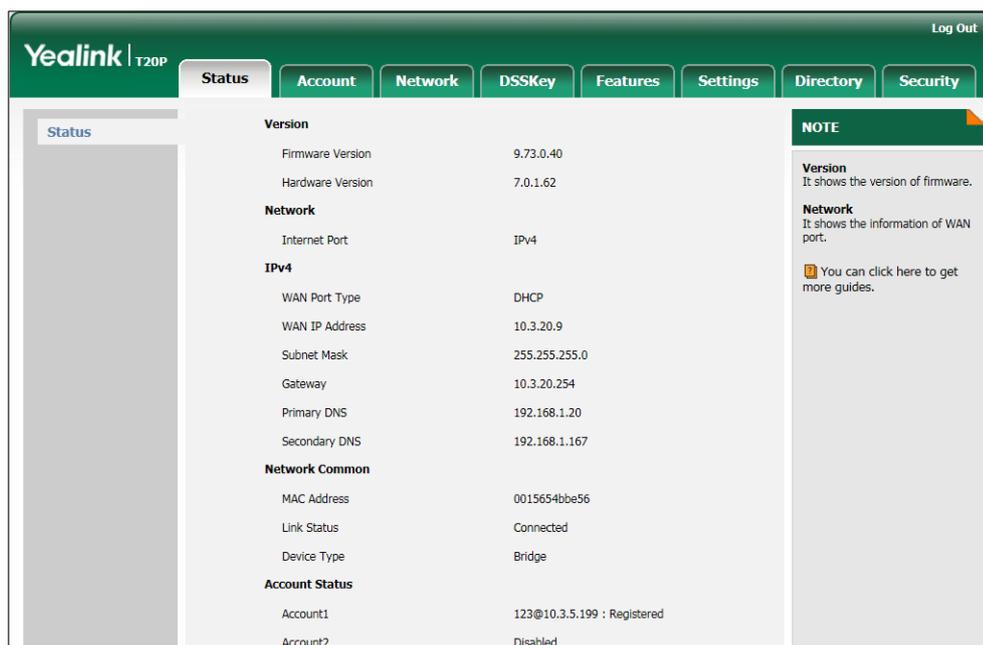
Login Enterprise IP phone SIP-T20P

Username

Password

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 Settings** (default password: admin) ->**Network->WAN Port**.
3. Press  or  to select **IPv4**, **IPv6** or **IPv4 & IPv6** from the **IP Mode** field.

1. IP Mode:
◀ IPv4 & IPv6 ▶

4. Press  to accept the change.

To configure a static IPv4 address via phone user interface:

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

Make sure that the IP address mode is configured as **IPv4** or **IPv4 & IPv6**.

3. Press  to select **IPv4** and press .

4. Press  to select **Static IP** and press .
5. Enter the desired values in the **IP Address, Subnet Mask, Default Gateway, IPv4 Pri.DNS** and **IPv4 Sec.DNS** fields respectively.

1. IP Address: 192.168.0.3
123

6. Press  to accept the change.

To configure a static IPv6 address via phone user interface:

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

Make sure that the IP address mode is configured as **IPv6** or **IPv4 & IPv6**

3. Press  to select **IPv6** and press .
4. Press  to select **Static IPv6** and press .
5. 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

6. Press  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 .
2. Select **Settings->Advanced Settings** (default password: admin) ->**Network->WAN Port**.
3. Press  to select **IPv4** and press .
4. Press  to select **PPPoE** and press .

1. PPPoE User:
2aB

5. Enter the user name and password in the corresponding fields.
6. Press  to accept the change.

Note

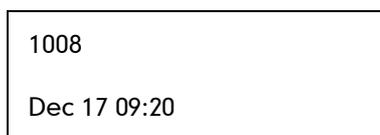
Wrong network settings may result in inaccessibility of your phone and may also have an impact on your network performance. For more information on these parameters, contact your system administrator.

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. The SIP-T20P/SIP-T20 IP phone supports up to 2 accounts. 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 45.

Idle Screen

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



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

Customizing Your Phone

You can customize your SIP-T20P/SIP-T20 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. If the language of your web browser is not supported by the phone, the web user interface will use English. You can change the language for the phone user interface and the web user interface respectively.

To change the language for the phone user interface:

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

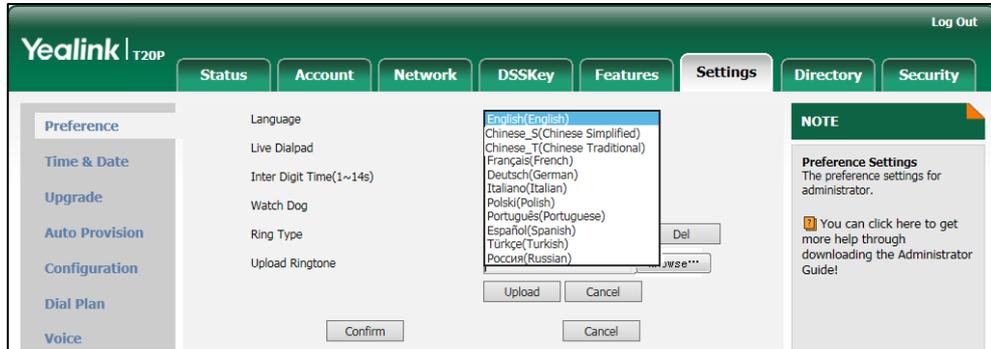
*1. English(English)
2. Français(French)

4. Press  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 Settings->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)”.



4. 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  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 Settings->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):

14-05-18

5. Press  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 Settings->Time & Date->Time & Date**.

1. Time Format:

◀ *24 Hour ▶

3. Press  or  to select the desired time format from the **Time Format** field.
4. Press  or  to select the desired date format from the **Date Format** field.
5. Press  to accept the change.

There are 3 available date formats. For example, for the date format “MM/DD/YY”, “MM” represents the first three letters of the month, “DD” represents a two-digit day, and “YY” represents a two-digit year which is not displayed on the LCD screen of the SIP-T20P/SIP-T20 IP phone.

The date formats available:

Date Format	Example (2014-11-15)
MM/DD/YY	Nov 15
DD/MM/YY	15 Nov
YY/MM/DD	15 Nov

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

Administrator Password

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

To change the administrator password via phone user interface:

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

1. Current PWD:

abc

3. Enter the old password in the **Current PWD** field.
4. Enter the new password in the **New PWD** field.
5. Re-enter the new password in the **Confirm PWD** field.
6. Press  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**.

Key as Send:

◀ * # ▶

3. Press  or  to select # or * from the **Key as Send** field, or select **Disabled** from the **Key as Send** field to disable this feature.
4. Press  to accept the change.

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

Phone Lock

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

Phone lock consists of the following:

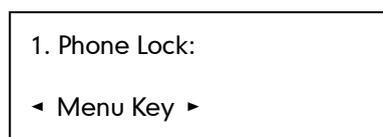
- Menu Key:** The MENU key and MESSAGE key are locked. You cannot access the menu of the phone until unlocked.
- Function Keys:** The function keys are locked. You cannot use the MESSAGE, CONF, HOLD, MENU, TRAN, RD, Directory, OK, X, line keys and navigation keys until unlocked.
- All Keys:** All keys 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 52.

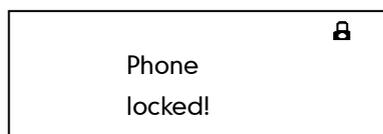
To activate the phone lock via phone user interface:

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



4. Press  to accept the change.
5. Long press  to lock the phone immediately when the phone is idle.

The LCD screen prompts "Phone locked!" and displays the icon .

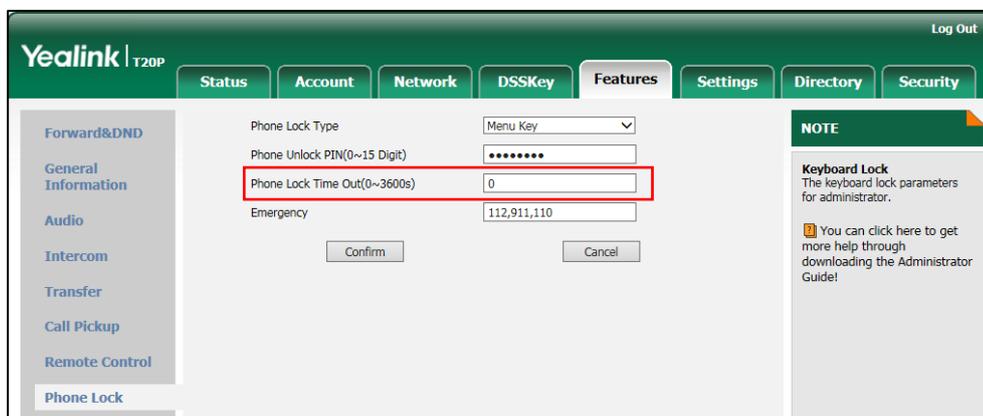


You can specify the interval (in seconds) to automatically lock the phone instead of long

pressing  .

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

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



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

Note

The default interval is 0 second, that is, you need to long press  to lock the phone. The interval of automatic phone lock is configurable via web user interface only.

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

To change the phone unlock PIN via phone user interface:

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

1. Current PIN:

123

4. Enter the new PIN in the **New PIN** field.
5. Re-enter the new password in the **Confirm PIN** field.
6. Press  to accept the change.

Note

The unlock PIN length must be within 15 digits.

To unlock the phone via phone user interface:

1. Press any locked key.
2. Enter the PIN in the **Phone Unlock PIN** field.
3. Press  to unlock the phone.

The  icon disappears from the LCD screen.

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

Note

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

To deactivate the phone lock via phone user interface:

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

1. Phone Lock:

◀ Disabled ▶

4. Press  to accept the change.

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

Audio Settings

Volume

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

To adjust the volume when the phone is idle:

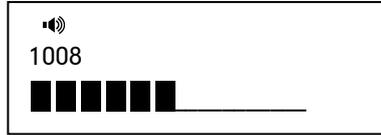
1. Press  to adjust the ringer volume.

100%



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

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

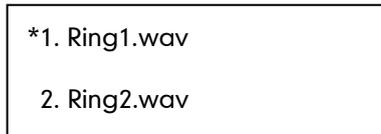


Ring Tones

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

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

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



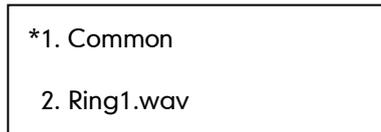
4. Press  to accept the change.

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

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

1. Press .
2. Press **Settings->Basic Settings ->Sound->Ring Tones.**
3. Press  or  to select the desired account and then press the **Enter** soft key.
4. Press  or  to select the desired ring tone.

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



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

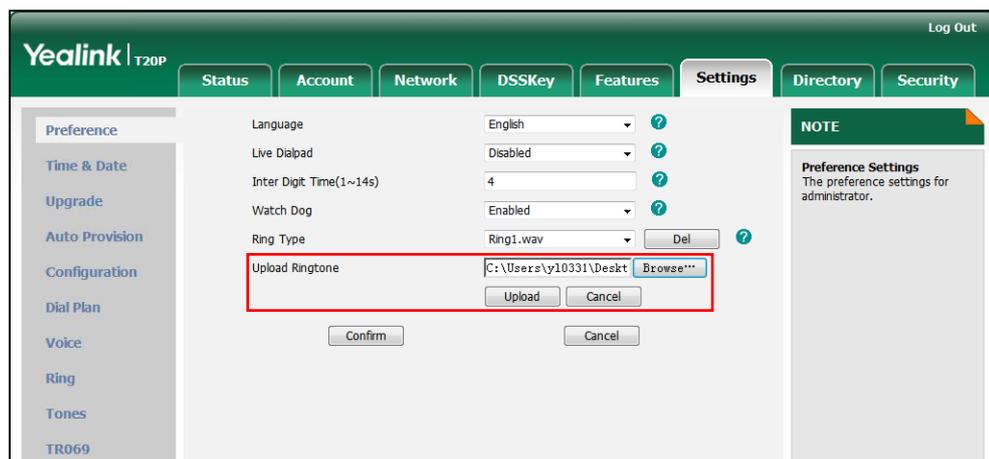
A ring tone for the account is configurable via web user interface at the path **Account->Basic->Ring Type**.

Note

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

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

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



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

Note

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

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

Contact Management

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

- [Local Directory](#)
- [Blacklist](#)

Local Directory

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

Adding Groups

To add a group to the local directory:

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

Name: Test

abc

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 specified for the contact. If no ring tone is specified for the contact, it will then play the ring tone specified for the account. For more information on ring tone for the account, refer to [Ring Tones](#) on page 26.
7. Press  to accept the change.

Editing Groups

To edit a group in the local directory:

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

Name: Test

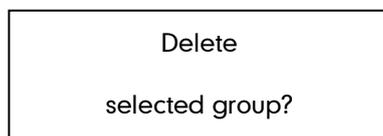
abc

6. Press  or  to select the contact information and then edit.
7. Press  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  or  to select **Delete** and then press .
- The LCD screen prompts “Delete selected group?”.



5. Press  to accept the change.

Adding Contacts

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

- Manually
- From call history

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 .
- If the groups have been added to the local directory, select the desired group, then press  or  to select **Enter** and then press .
4. Press  or  to scroll to **New Item**.
 5. Press  to add a new contact.
 6. Press .



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  to accept the change.

Note

If the contact already exists in the directory, the LCD screen prompts "Contact Name existed!".

Adding Contacts from Call History

To add a contact to the local directory from 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  to accept the change.
The entry is successfully saved to the local directory.

Editing Contacts

To edit a contact in the local directory:

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

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

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  to accept the change.

Deleting Contacts

To delete a contact from the local directory:

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

If the groups have been added to the local directory, select the desired group, then press  or  to select **Enter** and then press .
4. Press  or  to select the desired contact.
5. Press  or  to select **Delete** and then press .

The LCD screen prompts "Delete selected item?".
6. Press  to confirm the deletion.

Placing Calls to Contacts

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

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

If the groups have been added to the local directory, select the desired group, then press  or  to select **Enter** and then press .
4. Press  or  to select the desired contact.
5. Press  or  to select **Send**.
6. Do one of the following:
 - If only one number for the contact is stored in the local directory, press  to dial out the number.
 - If multiple numbers for the contact are stored in the local directory, press  to display a list of numbers.

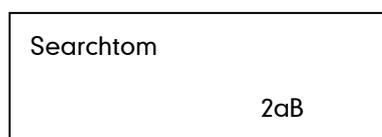
Press  or  to select the desired number.

Press  to dial out the number.

Searching for Contacts

To search for a contact in the local directory:

1. Press .
 2. Select **Local Directory**.
 3. Press  or  to select **Enter** and then press .
- If the groups have been added to the local directory, select the desired group, then press  or  to select **Enter** and then press .
4. Enter a few continuous characters of the contact name or continuous numbers of the contact number (office, mobile or other number) using the keypad.



5. Press .
- 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.

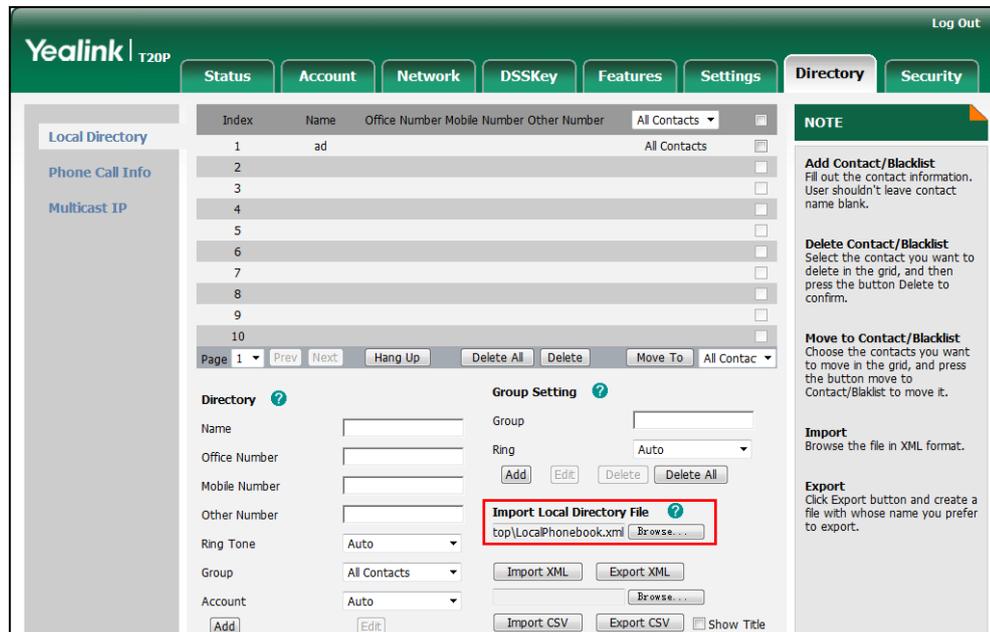
Importing/Exporting Contact Lists

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

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

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

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

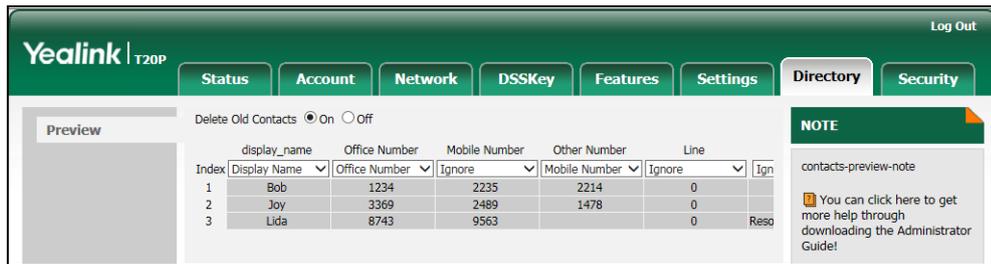


- Click **Import XML** to import the contact list.
The web user interface prompts "The original contact will be covered, Continue?".
- Click **OK** to complete importing the contact list.

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

- Click on **Directory->Local Directory**.
- Click **Browse** to locate a contact list file (the 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.
- 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.
- (Optional.) Select the contact information you want to import into the local directory from the pull down list of **Index**.

At least one item should be selected to be imported into the local directory.



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

To export contact list via web user interface:

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

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

Blacklist

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

To add a contact to the blacklist manually:

1. Press .
2. Select **Blacklist**.
3. Press  or  to scroll to **New Item**.
4. Press  to add a new contact.
5. Press  to switch the input mode.
6. Enter the name and the office, mobile or other numbers in the corresponding fields.

Name: Test

abc

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  to accept the change.

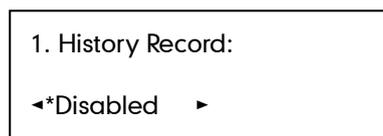
For operating instructions on adding, editing, deleting, placing calls to and/or searching for contacts in the blacklist, refer to the operating instructions of [Local Directory](#) on page 28.

Call History Management

The SIP-T20P/SIP-T20 IP phone maintains call history lists of Placed Calls, Received Calls, Missed Calls and Forwarded Calls. Call history lists support up to 100 entries. You can view call history, place a call, add a contact or delete an entry from the call history list. History record feature is enabled by default. If you don't want to save the call history, you can disable the feature.

To disable history record via phone user interface:

1. Press .
2. Select **Features->History Setting**.
3. Press  or  to select **Disabled** from the **History Record** field.



4. Press  to accept the change.

To view 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  or  to select the desired entry.
4. Press .

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  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  or  to select the desired entry.
4. Press  .
5. Enter the desired values in the corresponding fields.
6. Press  .

For more information, refer to [Contact Management](#) on page 27.

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  to delete the entry.

System Customizations

Logo Customization

You can customize text logo which would be displayed 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**.

- Enter the desired text in the **Text Logo** field.

The screenshot shows the Yealink T20P web interface. The 'Features' tab is selected. In the 'General Information' section, the 'Text Logo' field is highlighted with a red box and contains the text 'Yealink'. Other settings include 'Use Logo' (Enabled), 'Voice Mail Tone' (Enabled), 'DHCP Hostname' (SIP-T20P), 'Reboot In Talking' (Disabled), 'Hide Feature Access Codes' (Disabled), 'Display Method on Dialing' (User Name), and 'Auto Linekeys' (Disabled). A 'NOTE' section on the right provides information about 'Call Waiting', 'Key As Send', and 'Hotline Number'.

- 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:

- Press  on the phone.

The  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 , the phone will then place a call using the headset automatically. For more information on using the headset to place a call, refer to [Placing Calls](#) on page 55.

To deactivate the headset:

- Press  again on the phone.

The headset icon disappears when the headset mode is deactivated.

Headset Prior

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

To enable Headset Prior via web user interface:

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

The screenshot shows the 'General Information' settings page for a Yealink T20P phone. The 'Headset Prior' option is highlighted with a red box and is set to 'Enabled'. Other settings include Call Waiting (Enabled), Auto Redial (Disabled), Auto Redial Interval (10), Auto Redial Times (10), Key As Send (#), Reserve # in User Name (Enabled), Hotline Number, Hotline Delay (4), DTMF Replace Tran (Disabled), Time-Out for Dial-Now Rule (1), Voice Mail Tone (Enabled), Hide Feature Access Codes (Disabled), and Display Method on Dialing (Label). A 'NOTE' section on the right provides information about Call Waiting, Key As Send, and Hotline Number.

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.

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

Headset prior is configurable via web user interface only.

Dual Headset

You can use two headsets when dual headset feature is enabled. 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**.

The screenshot shows the Yealink T20P web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. The 'Dual-Headset' setting is highlighted with a red box and is set to 'Enabled'. Other settings include Call Waiting (Enabled), Auto Redial (Disabled), Auto Redial Interval (10), Auto Redial Times (10), Key As Send (#), Reserve # in User Name (Enabled), Auto-Answer Delay (1), Headset Prior (Enabled), DTMF Replace Tran (Disabled), Time-Out for Dial-Now Rule (1), Voice Mail Tone (Enabled), Hide Feature Access Codes (Disabled), and Display Method on Dialing (Label). A 'NOTE' section on the right provides details for Call Waiting, Key As Send, and Hotline Number.

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

Note

Dual headset is configurable via web user interface only.

DSS Keys

There are two types of DSS keys: Line Keys and Programmable Keys. The details will be introduced in the following. SIP-T20P/SIP-T20 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 recall 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 a line key via phone user interface:

1. Press  .

2. Select **Features->DSS Keys**.
3. Select the desired DSS key, and then press .
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  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
- PhoneLock

For the features not listed above, refer to [Basic Call Features](#) on page 55 and [Advanced Phone Features](#) on page 81. For more information, contact your system administrator.

Speed Dial

You can use this key feature to speed up dialing the numbers frequently 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 quickly connect voice mail. For more information, refer to [Voice Mail](#) on page 95.

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. Then follow the voice prompt to listen to 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, the phone will then enter the pre-dialing screen and display the prefix number that you specified in the **Value** field. You can enter the remaining digits and then dial out.

Local Group

You can use this key feature to quickly access a contact group in the local directory. For more information, refer to [Local Directory](#) on page 28.

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 [72](#).

Dependencies: *Type (Key Event)*

Key Type (Conference)

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

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

Note

If the **Value** field is left blank, the DSS key performs the same function as the **CONF** 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 [66](#).

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

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

Transfer

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

Dependencies: *Type (Key Event)*

Key Type (Tran)

Value (the number you want to transfer to)

Usage:

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

Note

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

If the **Value** field blank is left blank, the DSS key performs the same function as the **TRAN** key during a call. For more information, refer to [Call Transfer](#) on page [71](#).

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 screen. For more information, refer to [Do Not Disturb \(DND\)](#) on page 63.

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

Dependencies: *Type (Key Event)*

Key Type (Group Listening)

Usage:

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

Zero Touch

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

Dependencies: *Type (Key Event)*

Key Type (Zero Touch)

Usage:

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

The phone will reboot to update configurations.

Phone Lock

You can use this key feature to immediately lock your phone instead of long pressing . For more information, refer to [Phone Lock](#) on page 23.

Dependencies: *Type (Phone Lock)*

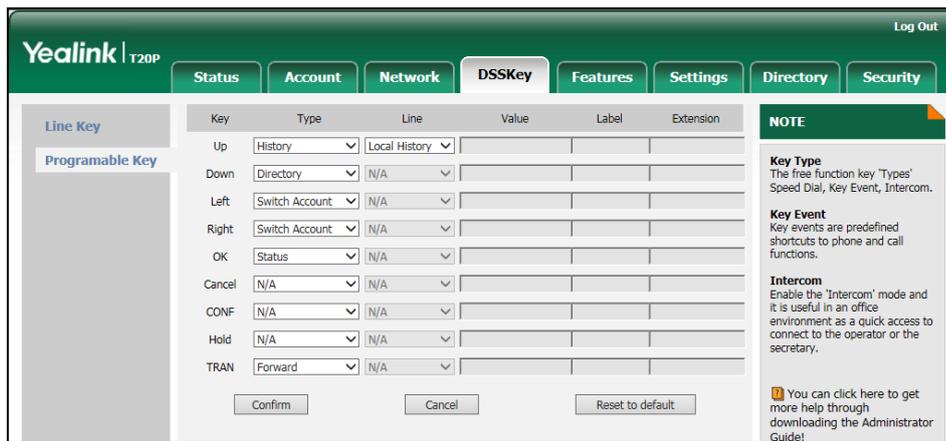
Usage: Press the DSS key to immediately lock your phone instead of long pressing .

Programmable Keys

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

To customize programmable keys via web user interface:

1. Click on **DSSKey-> Programmable Key**.
2. Customize specific features for these keys.



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

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

Then you can press the keys on the phone to perform the features you configured.

For example:

Switch Account

You can press this key to switch default account.

Dependencies: *Type (Switch Account)*

Usage: Press the programmable key to scroll down the account list to select the desired default account.

Note

Programmable keys are configurable via web user interface only.

Account Management

You can register two accounts at most on the SIP-T20P/SIP-T20 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 Settings->Accounts**.
3. Select the desired account.
4. Select **Enabled** from the **Activation** field.

1. Activation:

◀ Enabled ▶

5. 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  to accept the change.

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

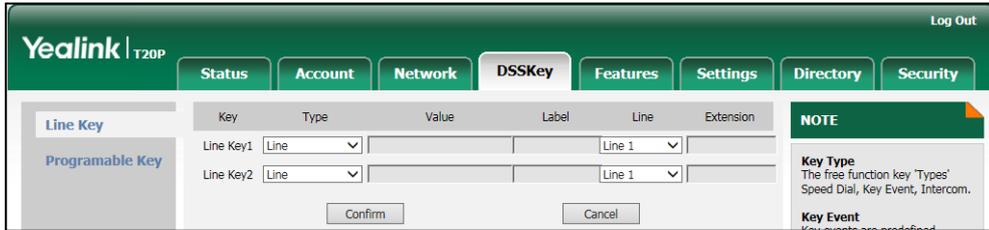
To disable an account via phone user interface:

1. Press .
2. Select **Settings->Advanced Settings ->Accounts**.
3. Select the desired account.
4. Select **Disabled** from the **Activation** field.
5. Press  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/SIP-T20 IP phone processes the inputs received from your phone keypad. The SIP-T20P/SIP-T20 IP phone supports the following dial plan features:

- [Replace Rule](#)
- [Dial-now](#)
- [Area Code](#)
- [Block Out](#)

The basic expression syntax you need to know:

.	The dot "." can be used as a placeholder or multiple placeholders for any character. Example: "12." would match "123", "1234", "12345", etc.
x	An "x" can be used as a placeholder for any character. Example: "12x" would match "121", "122", "123", etc.
[]	The square brackets "[]" can be used as a placeholder for a single character which matches any of a set of characters. Example: "91[5-7]1234" would match "9151234", "9161234", "9171234".
()	The parentheses "()" can be used to group together patterns, for instance, to logically combine two or more patterns. Example: "91([5-7])1(x)" would match "91511", "91618", "91715".

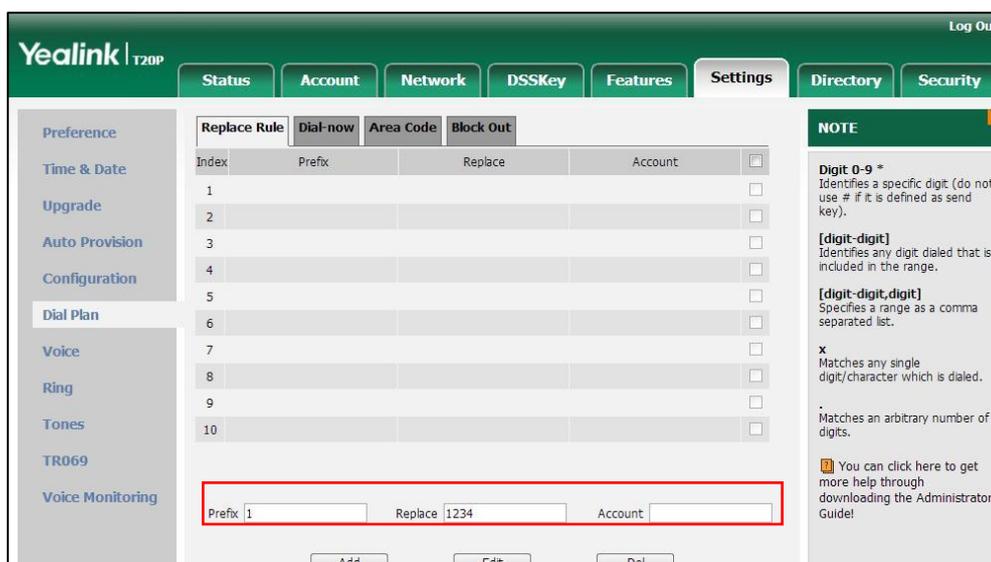
The "\$" should be followed by the sequence number of a parenthesis. The "\$" plus the sequence number means the whole character or characters placed in the parenthesis. The number directs to the right parenthesis when there are more than one. Example:
 A replace rule configuration, Prefix: "001(xxx)45(xx)", Replace: "9001\$145\$2".
 When you dial out "0012354599" on your phone, the IP phone will replace the number with "9001**235**45**99**". "\$1" means 3 digits in the first parenthesis, that is, "235". "\$2" means 2 digits in the second parenthesis, that is, "99".

Replace Rule

You can configure one or more replace rules (up to 100) to remove the specified string and replace it with another string. You can configure a pattern with wildcards (refer to the expression syntax in the table above), so that any string that matches the pattern will be replaced. This feature is convenient for you to dial out a long number. For example, a replace rule is configured as "Prefix: 1" and "Replace: 1234", when you try to dial out the number "1234", you just need to enter "1" on the phone and then press the Send soft key.

To add a replace rule via web user interface:

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Enter the string (e.g., 1) in the **Prefix** field.
3. Enter the string (e.g., 1234) in the **Replace** field.
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 for the **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”, any entered three-digit string beginning with 2 will then 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., 2xx) in the **Rule** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

For more information on the valid values for the **Account** field, refer to [Replace Rule](#) on page 47.

The screenshot shows the 'Replace Rule' configuration page in the Yealink T20P web interface. The page has a green header with 'Yealink | T20P' and 'Log Out'. Below the header are tabs for 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Settings' tab is active, and the 'Replace Rule' sub-tab is selected. A table with columns 'Index', 'Dial-now Rule', and 'Account' is shown, with rows 1 through 10. Below the table is a form with a 'Rule' field containing '2xx', an 'Account' field, and 'Add', 'Edit', and 'Del' buttons. A 'NOTE' box on the right contains information about digit ranges and symbols like 'x' and '.'.

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

When you enter the number “234” using the keypad, the phone will dial out “234” automatically without the pressing of any key.

Note

You can also edit or delete the dial-now rule, refer to [Replace Rule](#) on page 47 for more information.

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

Delay Time for Dial-now Rule

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

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

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

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

The screenshot shows the Yealink T20P web user interface. The 'Features' tab is selected, and the 'General Information' section is expanded. The 'Time-Out for Dial-Now Rule' field is highlighted with a red box and contains the value '1'. Other fields in the 'General Information' section include Call Waiting (Enabled), Call Waiting On Code, Call Waiting Off Code, Auto Redial (Disabled), Auto Redial Interval (1~300s) (10), Auto Redial Times (1~300) (10), Key As Send (#), Reserve # in User Name (Enabled), Hotline Number, Hotline Delay(0~10s) (4), Busy Tone Delay (Seconds) (0), Return Code When Refuse (486 (Busy Here)), Return Code When DND (480 (Temporarily Not A)), Call Completion (Disabled), and Feature Key Synchronization (Disabled). A 'NOTE' section on the right provides information about Call Waiting, Key As Send, and Hotline Number.

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

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

Area Code

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

To configure the area code via web user interface:

- Click on **Settings->Dial Plan->Area Code**.
- Enter the desired values in the **Code**, **Min Length (1-15)** and **Max Length (1-15)** fields.

3. Enter the desired line ID in the **Account** field or leave it blank.

For more information on the valid values for the **Account** field, refer to [Replace Rule](#) on page 47.

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.

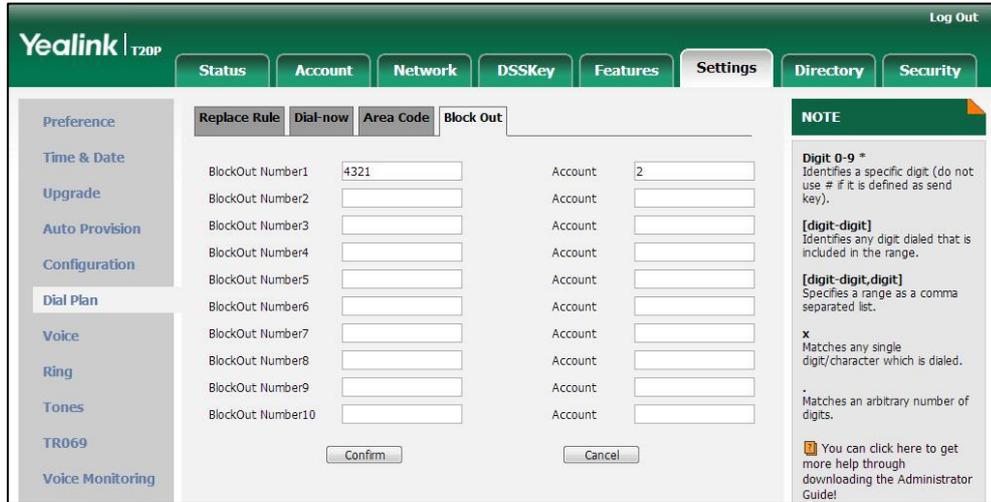
Block Out

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.
3. Enter the desired line ID in the **Account** field or leave it blank.

For more information on the valid values for the **Account** field, refer to [Replace Rule](#) on page 47.



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.

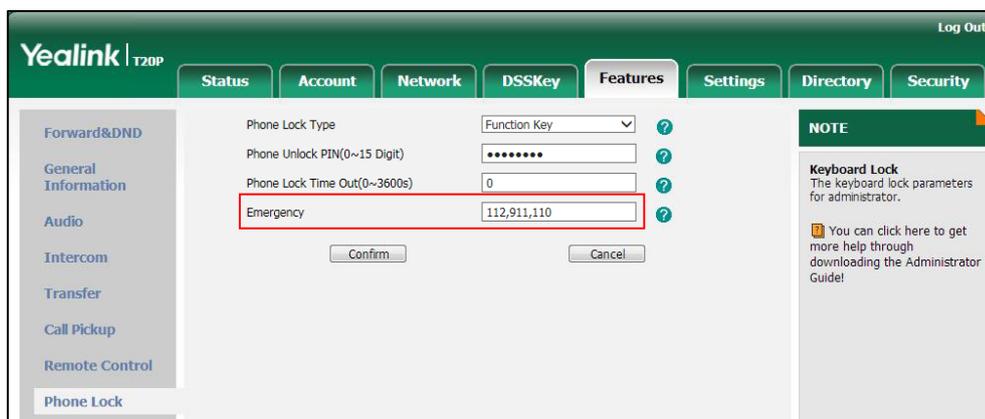
You can specify the emergency telephone numbers on the IP phone for contacting the emergency services in an emergency situation.

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.

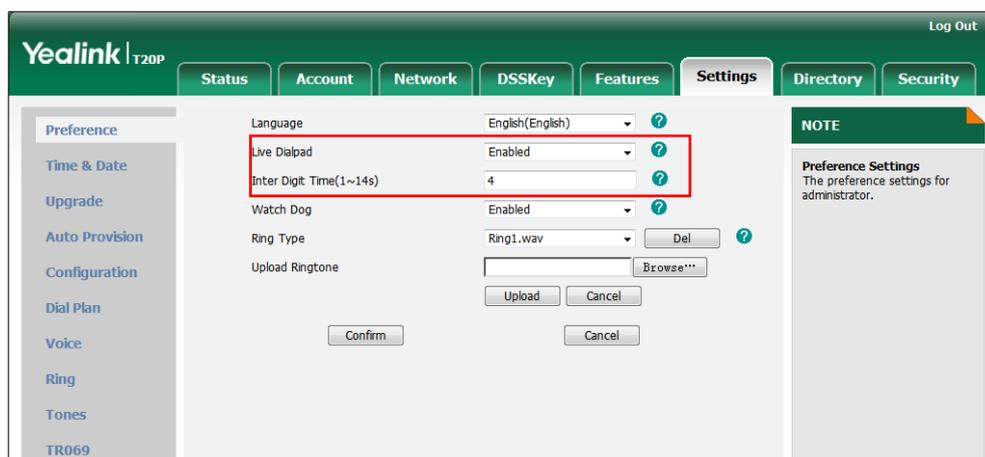
Live Dialpad

You can enable live dialpad on the SIP-T20P/SIP-T20 IP phone, which enables the IP phone to automatically dial out a phone number without the pressing of the send key. You can also configure a delay, where the phone will dial out the phone number automatically after the designated period of time.

To enable live dialpad via web user interface:

1. Click on **Settings->Preference**.
2. Select **Enabled** from the pull-down list of **Live Dialpad**.
3. Enter the desired delay time in the **Inter Digit Time (1~14s)** field.

The default delay time is 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 designated period of time.

To configure hotline number via phone user interface:

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

The valid values range from 0 to 10.

1. Number: 123

5. Press  to accept the change.

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

Basic Call Features

The SIP-T20P/SIP-T20 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/SIP-T20 IP phone.

Topics include:

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

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

Placing Calls

You can place a call in one of three ways using your SIP-T20P/SIP-T20 IP phone:

- Using the handset
- Using the speakerphone

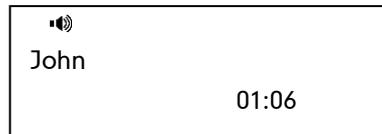
- Using the headset

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 27 and [Call History Management](#) on page 35.

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

The call duration of the active call is visible on the LCD screen. In the figure below, the call 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  or .

The # key is configured as a send key by default. You can also set the * key as the send key, or set neither. For more information, refer to [Key as Send](#) on page 22.

Note

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

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

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

Do one of the following:

- With the handset on-hook, press  or the line key to obtain a dial tone.
Enter the desired number using the keypad.
Press  or .
- With the handset on-hook, enter the desired number using the keypad.
Press ,  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  or .
- With the optional headset connected, press  to activate headset mode.
Enter the desired number using the keypad.
Press  or .

Note

To permanently activate the headset mode, refer to [Headset Prior](#) on page 38.

The SIP-T20P IP phone can handle multiple calls at a time. However, only one active call (the call that has audio associated with it) can be in progress at any given time. The SIP-T20P IP phone can handle a maximum of 6 calls at one time.

To place multiple calls:

You can have more than one call on your SIP-T20P/SIP-T20 IP phone.

1. Press the line key.
The active call is placed on hold.
2. Enter the desired number using the keypad.
3. Press ,  or .

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 one of three ways:

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

Note

You can reject incoming calls by pressing the **X** key. You can also activate Do Not Disturb mode to reject all incoming calls without ringing on your phone. For more information, refer to [Do Not Disturb \(DND\)](#) on page 63.

You can forward incoming calls to someone else by pressing the **TRAN** key. For more information, refer to [Call Forward](#) on page 66.

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 .
- 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  HEADSET.
- With the headset mode activated, press .
- 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 .

The incoming call is answered and the original call is placed on hold.
- Press  to access the new call.

Press .

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  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  twice.
A call to your last dialed number is attempted.

To redial a previously dialed number from your phone:

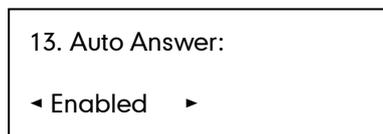
1. Press  when the phone is idle.
2. Press  or  to select the desired entry from the placed calls list, and then press  or .

Auto Answer

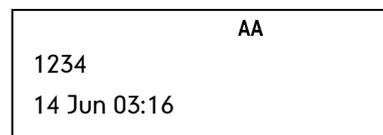
You can use auto answer to automatically answer an incoming call on a line. Auto answer is configurable on a per-line basis.

To configure auto answer via phone user interface:

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



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

To configure auto redial via phone user interface:

1. Press .
2. Select **Features->Auto Redial**.
3. Press  or  to select **Enabled** from the **Auto Redial** field.

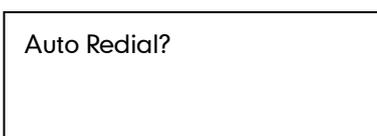


4. Enter the desired time in the **Interval** field.
The default time interval is 10 seconds.
5. Enter the desired number of redial attempts in the **Times** field.
The default times are 10.
6. Press  to accept the change.

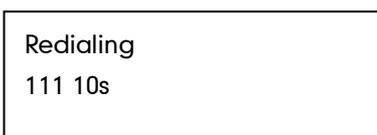
Auto redial is configurable via web user interface at the path **Features->General Information**.

To use auto redial:

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



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



2. Wait for the designated period of time or press  to redial the phone number.
The phone will retry as many times as configured until the called party is idle.

Call Completion

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

To configure call completion via phone user interface:

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



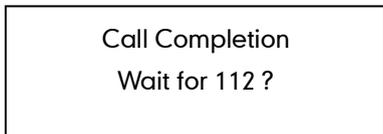
1. Call Completion:
◀ Enabled ▶

3. Press  or  to select **Enabled** from the **Call Completion** field.
4. Press  to accept the change.

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

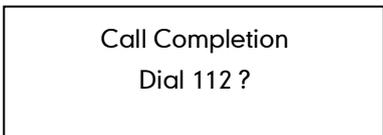
To use call completion:

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 . 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  to redial the number.

Note

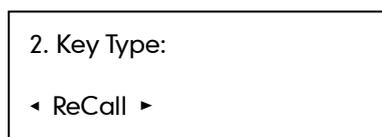
Call completion is not available on all servers. For more information, contact your system administrator.

Recall

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

To configure a recall key via 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 **ReCall** from the **Key Type** field.



6. Press  to accept the change.

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

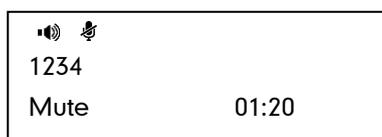
Call Mute

You can mute the microphone of the active audio device during an active call so that the other party cannot hear you. Call mute applies to all modes (Handset, Headset and Speakerphone).

To mute a call:

1. Press  during an active call.

The LCD screen indicates that the call is now muted.



To un-mute a call:

1. Press  again to un-mute a call.

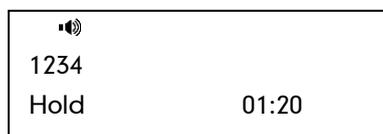
Call Hold/Resume

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 to the other party while waiting.

To place a call on hold:

1. Press  during a call.

The LCD screen indicates that 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.

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. The prompt message "**n New Missed Call(s)**" ("n" indicates the number of missed calls) will appear on the LCD screen, and callers can hear a busy signal or a message depending on your server. All calls you receive while DND is enabled are logged to your missed calls list.

Note

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

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

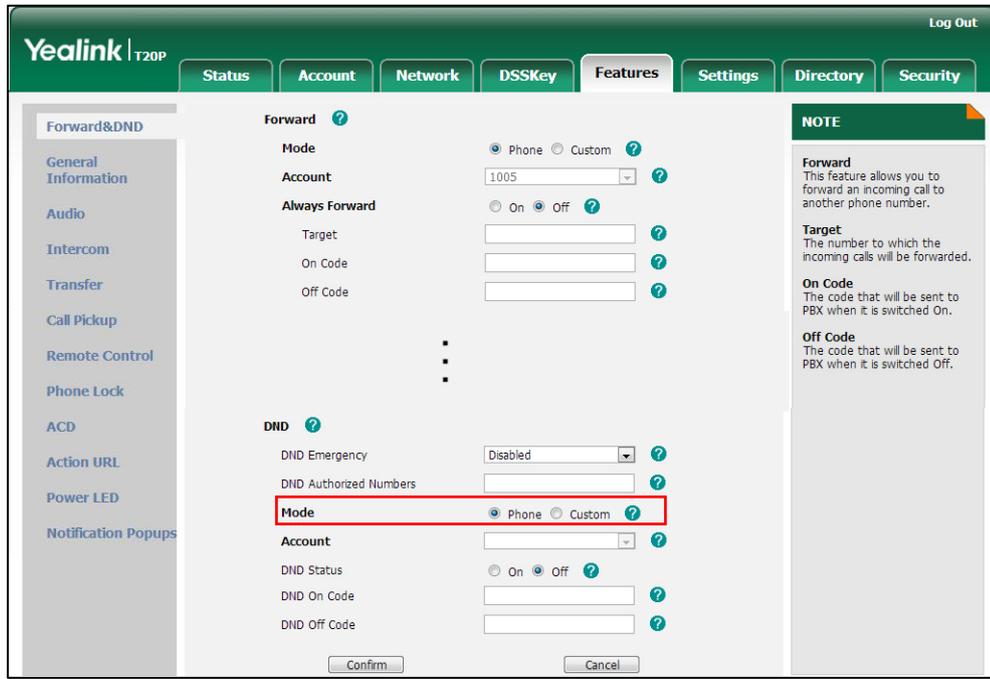
You can enable/disable DND for the phone system, or you can customize DND for each account. Two DND modes:

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

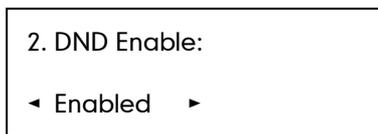


3. Click **Confirm** to accept the change.

Note DND mode is configurable via web user interface only.

To activate DND in phone mode:

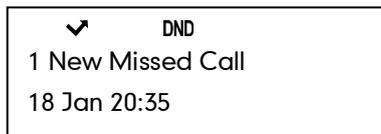
1. Press .
2. Select **Features->DND**.
3. Select **Enabled** from the **DND Enabled** field.



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

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

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

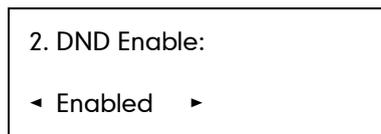


Note

When DND and busy forward are enabled in phone mode, calls will be sent to the configured destination number. For more information on busy forward, refer to [Call Forward](#) on page 66.

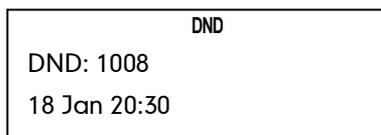
To activate DND in custom mode:

1. Press .
2. Select **Features->DND**.
3. Press  or  to select the desired account.
4. Press **Enter** soft key.
5. Select **Enabled** from the **DND Enable** field.

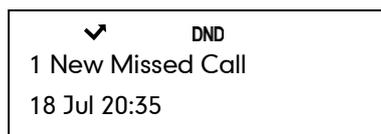


6. (Optional.) Enter the DND on code and off code respectively in the **On Code** and **Off Code** field.
7. Press  to accept the change.

If you activate DND for the default account, 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 New Missed Call**" ("n" indicates the number of missed calls) will prompt on the LCD screen.



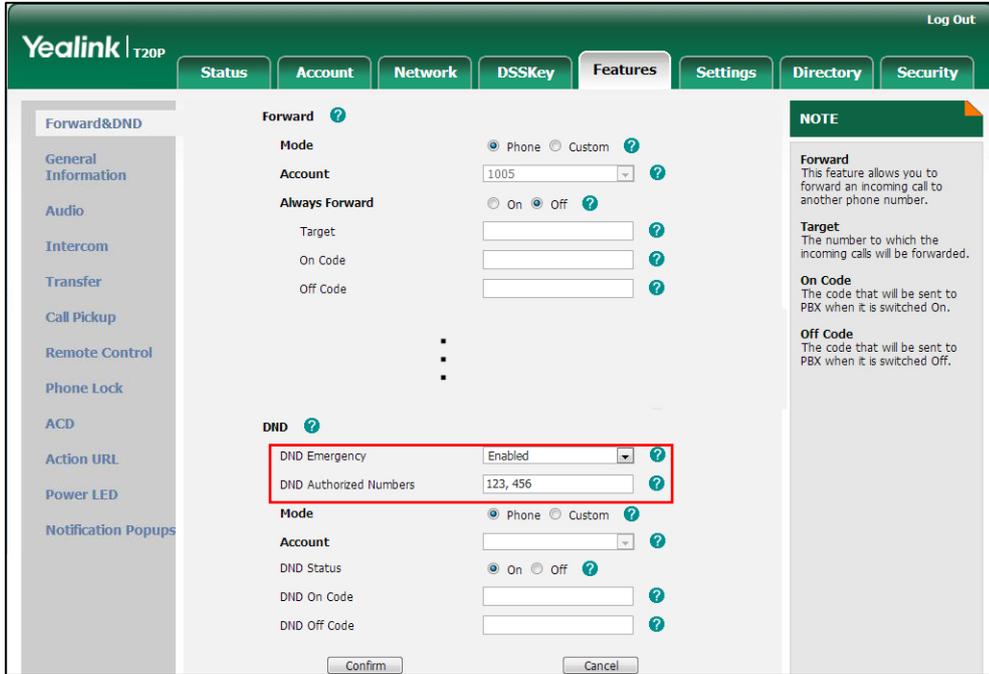
Note

When DND and busy forward are enabled for a specific account, calls from the specific account will be sent to the configured destination number. For more information on busy forward, refer to [Call Forward](#) on page 66.

To configure the DND authorized numbers via web user interface:

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

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



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

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

Note DND authorized number is configurable via web user interface only.
 When the phone receives a voice mail, the voice mail prompt window will pop up by default, if you want to disable the feature, contact your system administrator for more information.

Call Forward

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

Static Forwarding

Three types of static forwarding:

- **Always Forward:** Incoming calls are immediately forwarded.

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

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

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

To configure the call forward mode via web user interface:

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

The screenshot shows the Yealink T20P web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Features' section is active, showing 'Forward & DND' settings. The 'Forward' section has a 'Mode' field with 'Phone' selected (indicated by a red box). Below it are 'Account' (1005), 'Always Forward' (On/Off), 'Target', 'On Code', and 'Off Code' fields. The 'DND' section has 'DND Emergency' (Disabled), 'DND Authorized Numbers', 'Mode' (Phone/Custom), 'Account', 'DND Status' (On/Off), 'DND On Code', and 'DND Off Code' fields. A 'NOTE' panel on the right explains the Forward feature: 'Forward: This feature allows you to forward an incoming call to another phone number. Target: The number to which the incoming calls will be forwarded. On Code: The code that will be sent to PBX when it is switched On. Off Code: The code that will be sent to PBX when it is switched Off.' At the bottom are 'Confirm' and 'Cancel' buttons.

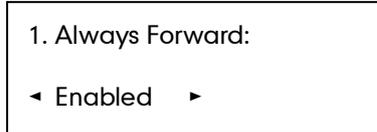
3. Click **Confirm** to accept the change.

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

To enable call forward in phone mode:

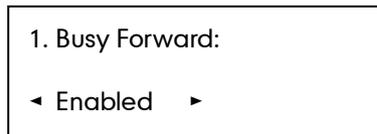
1. Press .
2. Select **Features->Call Forward**.
3. Press  or  to select the forwarding type, and then press .
4. Depending on your selection:
 - a.) If you select **Always Forward**:
 - 1) Press  or  to select **Enabled** from the **Always Forward** field.

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



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

- 1) Press ◀ or ▶ to select **Enabled** from the **Busy Forward** field.
- 2) Enter the destination number you want to forward incoming calls to when the phone is busy in the **Forward to** field.
- 3) (Optional.) Enter the busy forward on code and off code respectively in the **On Code** and **Off Code** field.

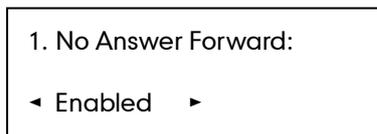


c.) If you select **No Answer Forward**:

- 1) Press ◀ or ▶ to select **Enabled** from the **No Answer Forward** field.
- 2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward to** field.
- 3) Press ◀ or ▶ 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.



5. Press  to accept the change.

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

To enable call forward in custom mode:

1. Press .
2. Select **Features->Call Forward**.

3. Press  or  to select the desired account, and then press .
4. Press  or  to select the forwarding type, and then press .
5. Depending on your selection:

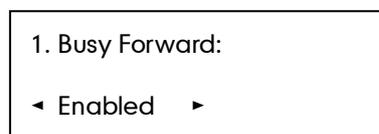
a.) If you select **Always Forward**:

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



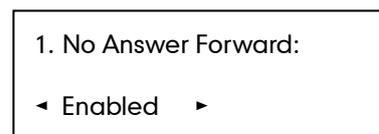
b.) If you selected **Busy Forward**:

- 1) Press  or  to select **Enabled** from the **Busy Forward** field.
- 2) Enter the destination number you want to forward all incoming calls to when the phone is busy in the **Forward to** field.
- 3) (Optional.) Enter the busy forward on code 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 **Enabled** from the **No Answer Forward** field.
- 2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward to** field.
- 3) Press  or  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  to accept the change.

If you activate call forward for the default account, 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 URI or IP address in the **Forward to** field. For more information on using the SIP URI or IP address, refer to [Placing Calls](#) on page 55.

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  or  to select the forwarding type, then press .
Press  or  to select **Disabled** to disable the call forward.
Press  to accept the change.

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

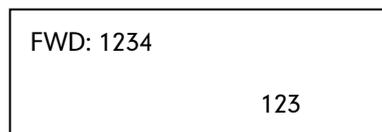
Do one of the following:

1. Press , select **Features->Call Forward** or press  when the phone is idle.
2. Press  or  to select the desired account, then press .
3. Press  or  to select the desired forwarding type, then press .
4. Press  or  to select **Disabled** to disable the call forward.
5. Press  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  or .

The LCD screen prompts a call forwarded message.

Forward to 1234

12 Sep 10:53

Note

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

Call Transfer

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

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

Tran: 1234

123

3. Press  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  or  to dial out.
4. Press  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  or  to dial out.
4. After the party answers the call, press  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  .

Call Waiting

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

To configure call waiting via phone user interface:

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

1. Call Waiting:

◀ *Enabled ▶

3. Press  or  to select **Enabled** from the **Call Waiting** field.
4. Press  or  to select **Enabled** from the **Play Tone** field.
5. (Optional.)Enter the call waiting on code and off code respectively in the **On Code** and **Off Code** field.
6. Press  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

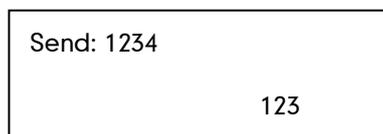
Network conference is not available on all servers. For more information, contact your system administrator.

Local Conference

The SIP-T20P/SIP-T20 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  to place a new call.
The active call is placed on hold.
3. Enter the number of second party and press  or .



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

To join two calls in a conference:

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

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

Network Conference

You can use network conference on the SIP-T20P/SIP-T20 IP phone to conduct a conference with multiple participants.

This feature allows you to perform the following:

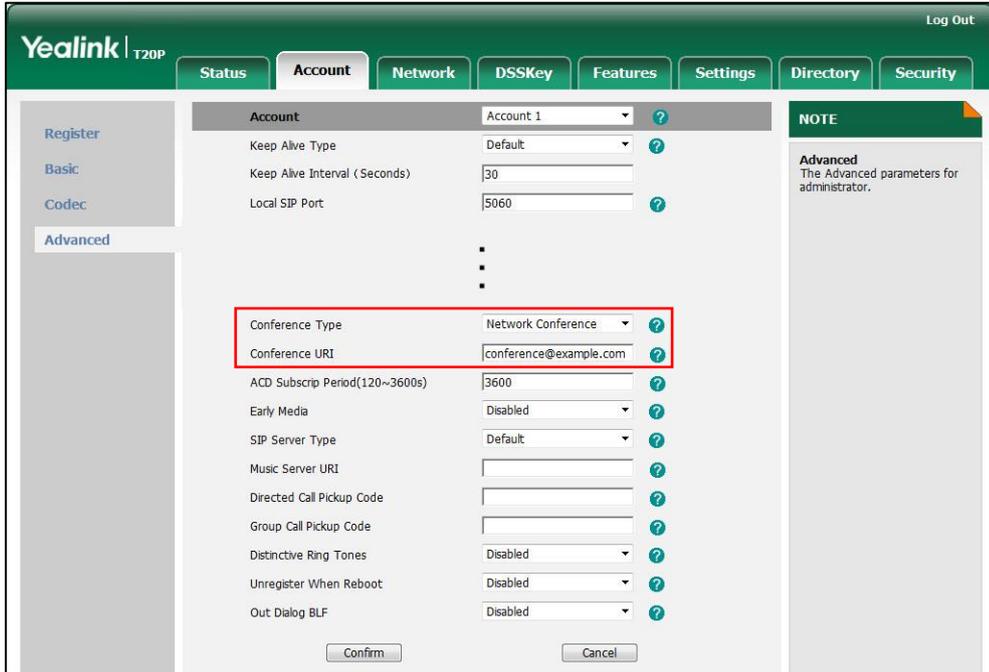
- Join two calls together into a conference call.
- Invite another party into an active conference call.

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  or .
4. When the second party answers the call, press  to add the second party to the conference.
5. Press  to place a new call.
The conference is placed on hold.
6. Enter the number of the new party and then press  or .
7. When the new party answers the call, press  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  or  to select the desired account from the **Account ID** field.
7. Enter the call park code in the **Value** field.

4. Value: *68
2aB

8. Press  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, and so presses the call park key on phone A.
3. User on phone A walks to an available conference room where the phone is designated as phone C. The user dials the call park retrieve code to retrieve the parked call.

The system establishes call between phone C and B.

Note

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

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

Call Pickup

You can use call pickup to answer someone else’s incoming call on your phone. The SIP-T20P/SIP-T20 IP phone supports directed call pickup and group call pickup. Directed call pickup is used for picking up a call that is ringing at a target phone number. Group call pickup is used for picking up a call that is ringing at any phone number in a certain group. The pickup group should be predefined, contact your system administrator for more information.

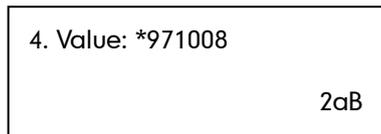
Note

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  or  to select **DPickup** from the **Key Type** field.
6. Press  or  to select the desired account from the **Account ID** field.
7. Enter the directed pickup code followed by the specific phone number you want to pick up in the **Value** field.



8. Press  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:

1. Press the **DPickup** key on your phone when the target 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 ◀ or ▶ to select the desired account from the **Account ID** field.
7. Enter the group pickup feature code in the **Value** field.

4. Value: *98	2aB
---------------	-----

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

Note

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

To configure anonymous call via phone user interface:

1. Press **MENU**.
2. Select **Features->Anonymous Call**.
3. Press ◀ or ▶ to select the desired line from the **Account ID** field.
4. Press ◀ or ▶ to select **Enabled** from the **Local Anony** field.

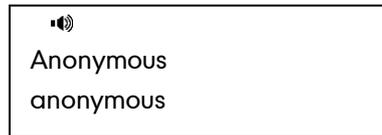
2. Local Anony:	◀ Enabled ▶
-----------------	-------------

5. (Optional.) Press  or  to select the desired value from the **Send Anony Code** field.
6. (Optional.) Enter the anonymous call on code and off code respectively in the **On Code** and **Off Code** field.
The phone will send the configured on code or off code depending on your selection when you enable or disable anonymous call feature on the phone.
7. Press  to accept the change.

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

To place an anonymous call:

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

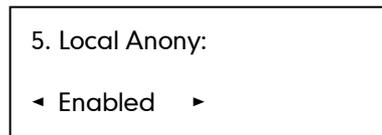


Anonymous Call Rejection

You can use anonymous call rejection to reject incoming calls from anonymous callers. Anonymous call rejection automatically rejects incoming calls from callers who deliberately block their identities and numbers from being displayed. Anonymous call rejection is configurable on a per-line basis. You can also configure the phone to send anonymous call rejection on/off code to the server to activate/deactivate anonymous call rejection on the server side.

To configure anonymous call rejection via phone user interface:

1. Press .
2. Select **Features->Anonymous Call**.
3. Press  or  to select the desired line from the **Account ID** field.
4. Press  or  to scroll to the **Local Anony** field.
5. Press  or  to select **Enabled** from the **Send Reject Code** field.



6. (Optional.) Enter the anonymous call rejection on code and off code respectively in the **On Code** and **Off Code** field.
The phone will send the configured on code or off code depending on your

selection when you enable or disable local anonymous rejection feature on the phone.

7. Press  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/SIP-T20 IP phone. Topics include:

- [Busy Lamp Field \(BLF\)](#)
- [BLF List](#)
- [Call Recording](#)
- [Hot Desking](#)
- [Intercom](#)
- [Multicast Paging](#)
- [Music on Hold](#)
- [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 to monitor 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 calls that are received on the monitored line. For more information, contact your system administrator.

To configure a BLF key via phone user interface:

1. Press .
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.

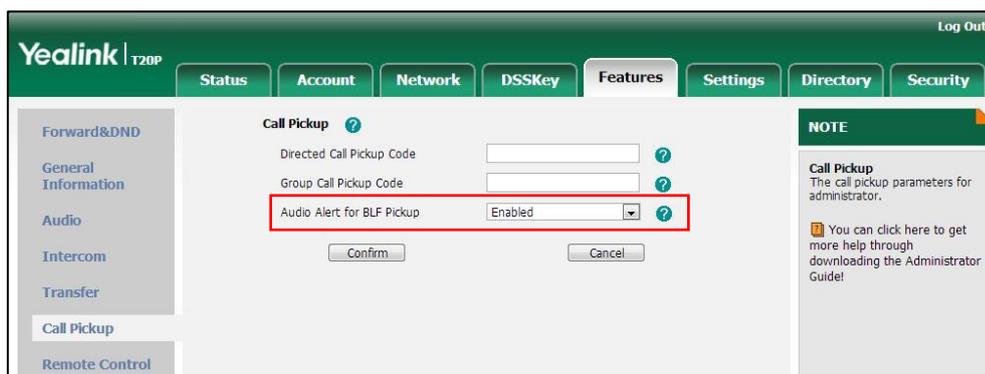
7. (Optional.) Enter the pickup code in the **Extension** field.
8. Press  to accept the change.

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

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

To enable an audio alert feature 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 following occurs on your 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 handles the active call differently depending on the transfer mode on DSS key. For more information on the transfer mode on DSS key, refer to transfer in the [DSS Keys](#) section on page 39.

BLF List

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

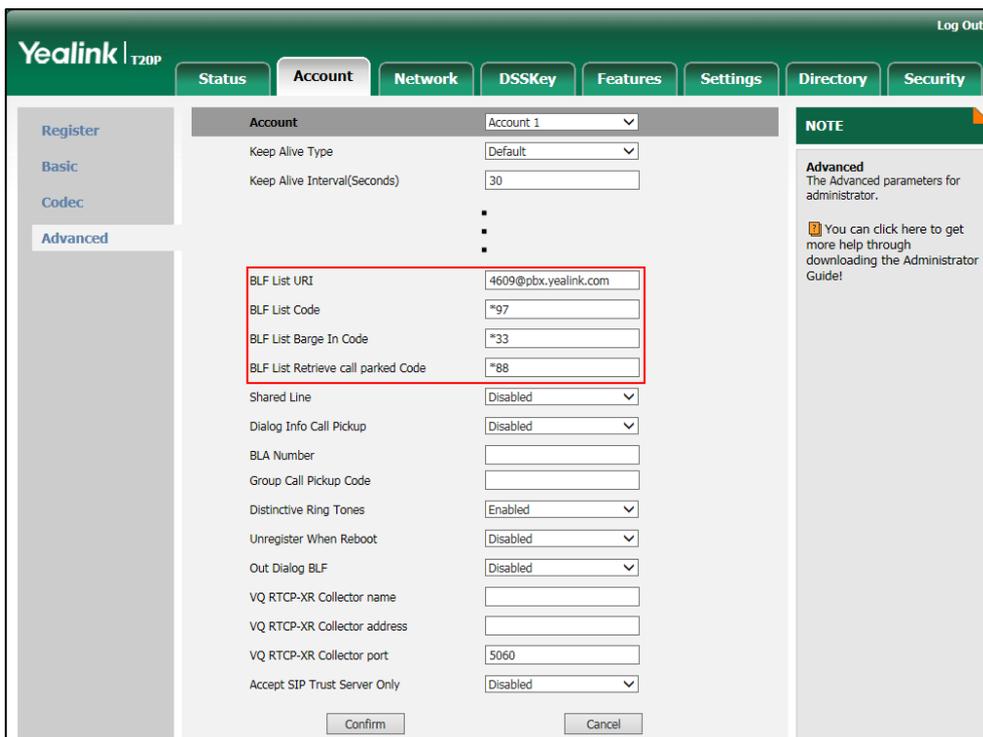
You can use the BLF List keys in the following ways:

- When the monitored user is idle, press the BLF list key to dial out the phone number.
- When there is already an active call on the IP phone, you can transfer the active call to the monitored user by pressing the BLF List key. The phone handles the active call differently depending on the transfer mode on DSS key. For more information on the transfer mode on DSS key, refer to [Transfer](#) on page 42.
- When the monitored user receives an incoming call, press the BLF list key to pick up the call directly. Before picking up an incoming call, ensure that the directed pickup code has been configured in advance. If the directed pickup code is not configured, the phone will place a call to the monitored user instead of picking up the incoming call of the monitored user when you press the BLF List key.
- When there is a conversation on the monitored user, press the BLF list key to barge in and set up a conference call. Before barging in an active call, ensure that the barge-in code has been configured in advance. If the BLF list barge in code is not configured, the phone will place a call to the monitored user instead of barging in an active call of the monitored user when you press the BLF List key.
- When a call is being parked against the monitored phone, press the BLF List key to retrieve the parked call from the monitored user. Before retrieving the parked call, ensure that the BLF List Retrieve call parked Code has been configured in advance. If the code is not configured, the phone will place a call to the monitored user instead of retrieving the parked call when you press the BLF List key.

To configure BLF List settings via web user interface:

1. Click on **Account->Advanced**.
2. Select the desired account from the pull-down list of **Account**.
3. Enter the BLF List URI in the **BLF List URI** field.
4. (Optional.) Enter the directed pickup code in the **BLF List code** field.
5. (Optional.) Enter the barge-in code in the **BLF List Barge In Code** field.

- (Optional.) Enter the call park retrieve code in the **BLF List Retrieve call parked Code** field.



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

Note

For more information on BLF List URI/BLF List Code/BLF List Barge In Code/BLF List Retrieve Call Parked Code, contact your system administrator.

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

You can receive a visual or/and an audio alert (if enabled) on your phone when the monitored user receives an incoming call. For more information, refer to [Busy Lamp Field \(BLF\)](#) section on page 81.

Note

The pickup code is used in the following order of preference: BLF List Code (Account->Advanced)>Directed Call Pickup Code (Account->Advanced)>Directed Call Pickup Code (Features->Call Pickup). If all of them are not configured, pressing the BLF List key will directly call the monitored user when he/she receives an incoming call. For more information about pickup Code, refer to [Call Pickup](#) on page 76.

Call Recording

You can record calls by pressing a record key on the phone. The SIP-T20P/SIP-T20 IP phone supports record and URL record.

Two ways of call recording:

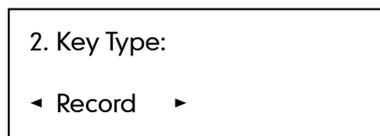
- **Record:** the phone sends SIP INFO message containing a specific header “Record: on/off” to trigger a recording.
- **URL Record:** the phone sends HTTP URL request to trigger a recording. Contact your system administrator for the predefined URL.

Note

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

To configure a record key via phone user interface:

1. Press .
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  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  to accept the change.

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

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

- During an active call
- When calls are on hold or muted
- 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:

1. 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.
2. 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 depending on the server. Contact your system administrator for more information.

Hot Desking

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

You can use hot desking on the SIP-T20P/SIP-T20 IP phone to log out of existing accounts and then log into a new account, that is, many users can share the phone resource at different times. To use this feature, you need to configure a hot desking key in advance.

Note

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

To configure a hot desking key via phone user interface:

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

- Press ◀ or ▶ to select **Hot Desking** from the **Key Type** field.

2. Key Type:

◀ Hot Desking ▶

- Press **OK** to accept the change.

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

To use hot desking:

- Press the hot desking key when the phone is idle.
The LCD screen prompts "Clear all account config?".

Warning

Clear all account config?

- Press **OK**, registration configurations of all accounts will be cleared immediately.
The login wizard will be displayed 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.
- Press **OK** to login.

Intercom

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

Note

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

Outgoing Intercom Calls

To configure an intercom key via phone user interface:

1. Press .
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  to accept the change.

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

To place an intercom call:

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

Incoming Intercom Calls

The SIP-T20P/SIP-T20 IP phone supports automatically to answer an incoming intercom call by default. The phone automatically plays a warning tone when it receives an incoming intercom call. In addition, you can enable the phone to mute the microphone when it automatically answers an incoming intercom call. You can also enable the phone to automatically answer an incoming intercom call while there is already an active call on the phone. The active call is then placed on hold.

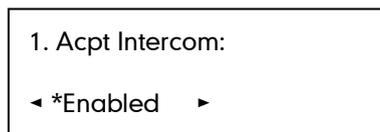
Intercom features you need to know:

Intercom Feature	Description
Accept Intercom	Enable or disable the IP phone to automatically answer an incoming intercom call.
Intercom Mute	Enable or disable the IP phone's microphone for intercom calls.
Intercom Tone	Enable or disable the IP phone to play a warning tone when it receives an incoming intercom call.
Intercom Barge	Enable or disable the IP phone to automatically

Intercom Feature	Description
	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  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 will automatically answer an incoming intercom call. If Accept Intercom is disabled, the phone will reject 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 phone's microphone for intercom calls automatically. If Intercom Mute is enabled, the microphone will be muted for intercom calls. If Intercom Mute is disabled, the microphone will work for intercom calls. Intercom Mute is disabled by default.

Intercom Tone

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

Intercom Barge

You can enable or disable the phone to automatically answer an incoming intercom call while there is already an active call on the phone. If Intercom Barge is enabled, the phone will automatically answer the intercom call and place the active call on hold. If

Intercom Barge is disabled, the phone will handle an incoming intercom call like a waiting call. Intercom Barge is disabled by default.

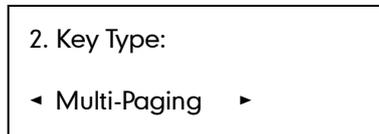
Multicast Paging

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

Sending RTP Stream

To configure a multicast paging key via phone user interface:

1. Press .
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 **Multi-Paging** from the **Key Type** field.



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

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

7. Press  to accept the change.

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

To configure a paging list key via phone user interface:

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

- Press  or  to select **Paging List** from the **Key Type** field.

2. Key Type:

◀ Paging List ▶

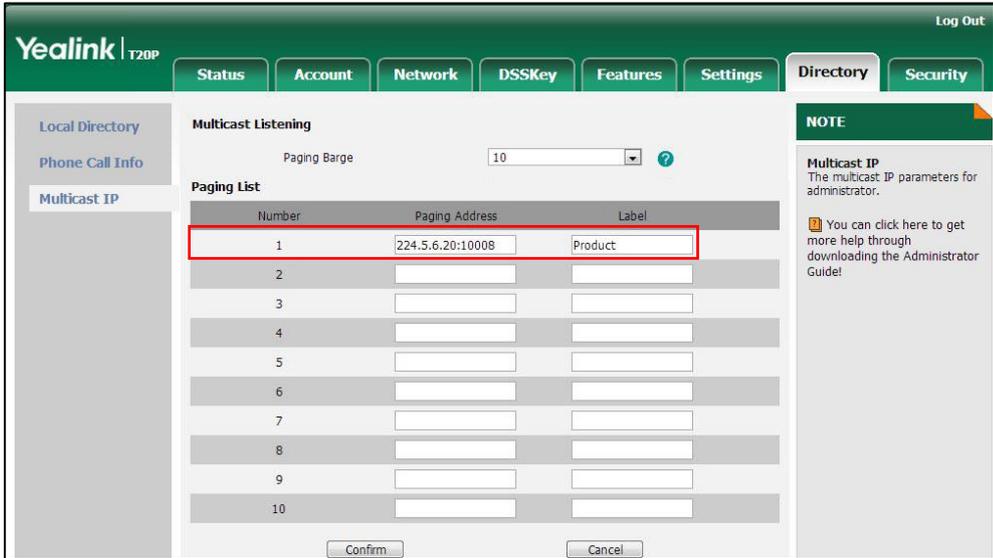
- Press  to accept the change.

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

To configure paging list via web user interface:

- Click on **Directory**.
- Click on **Multicast IP**.
- Enter the multicast IP address and port number (e.g., 224.5.6.20:10008) in the **Paging Address** field.

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.



The screenshot shows the Yealink T20P web interface. The 'Directory' tab is selected, and the 'Multicast IP' sub-tab is active. The 'Multicast Listening' section shows a 'Paging Barge' dropdown set to '10'. Below it is a 'Paging List' table with 10 rows. The first row is highlighted with a red box, showing '1' in the 'Number' column, '224.5.6.20:10008' in the 'Paging Address' column, and 'Product' in the 'Label' column. A 'NOTE' box on the right states: 'Multicast IP: The multicast IP parameters for administrator. You can click here to get more help through downloading the Administrator Guide!'. At the bottom are 'Confirm' and 'Cancel' buttons.

Number	Paging Address	Label
1	224.5.6.20:10008	Product
2		
3		
4		
5		
6		
7		
8		
9		
10		

- Enter the paging group name in the **Label** field.
- Click **Confirm** to accept the change.

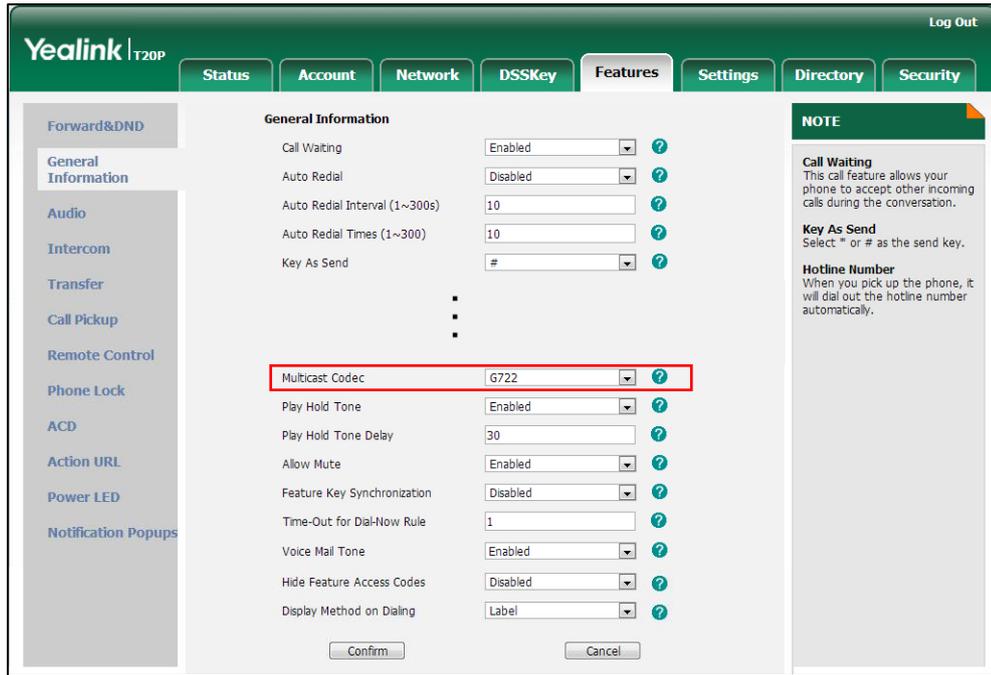
Note Paging list is configurable via web user interface only.

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:

- 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.

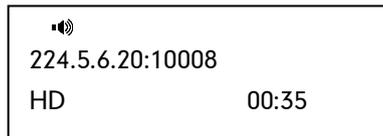
To send RTP stream via a multicast paging key:

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

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

The multicast paging key LED illuminates solid green.

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



2. Press  to place the current multicast RTP session on hold.

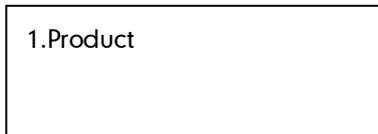
3. Press  to cancel the multicast RTP session.

Note

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

To send RTP stream via a paging key list:

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



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

Receiving RTP Stream

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

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

Paging Barge

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

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

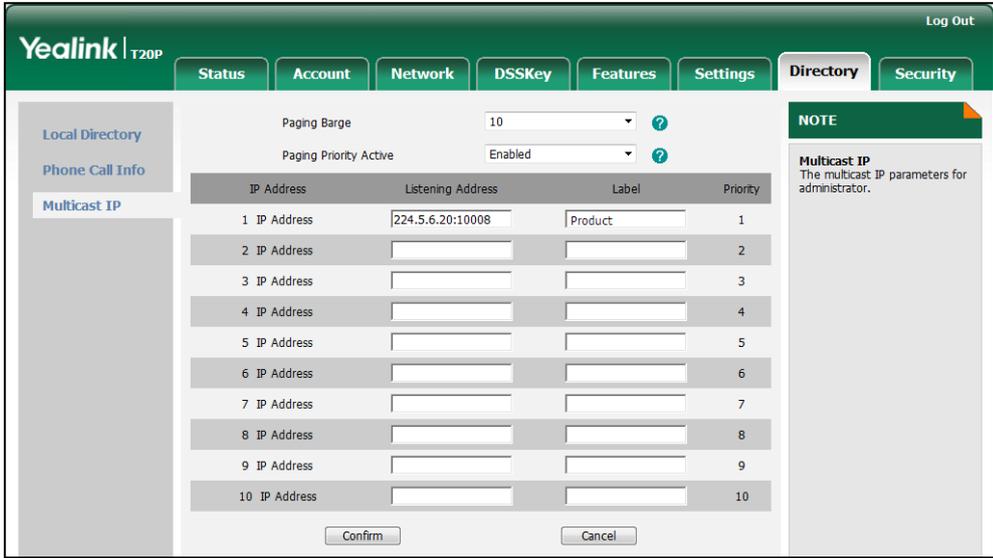
Paging Priority Active

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

To configure multicast listening addresses via web user interface:

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

3. Select the desired value from the pull-down list of **Paging Priority Active**.
Enter the multicast IP address(es) and port number(e.g., 224.5.6.20:10008) which the phone listens to for incoming RTP multicast in the **Listening Address** field.
The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.



Yealink T20P Log Out

Status Account Network DSSKey Features Settings **Directory** Security

Local Directory
Phone Call Info
Multicast IP

Paging Barge: 10 ?
Paging Priority Active: Enabled ?

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

Confirm Cancel

NOTE
Multicast IP
The multicast IP parameters for administrator.

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

Note

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

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

Multicast listening addresses is configurable via web user interface only.

Music on Hold

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

To configure music on hold server via web user interface:

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

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

The screenshot shows the Yealink T20P web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Music Server URI' field is highlighted with a red box and contains the text 'sip:moh@sip.com'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5060), 'RPort' (Disabled), 'SIP Session Timer T1 (0.5~10s)' (0.5), 'SIP Session Timer T2 (2~40s)' (4), 'SIP Session Timer T4 (2.5~60s)' (5), and 'Subscribe Period (Seconds)' (1800). A 'NOTE' box on the right states: 'Advanced: The Advanced parameters for administrator.'

- Click **Confirm** to accept the change.
When you placed a call on hold, the held party can hear the music.

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

Message

Voice Mail

You can leave voice mails for someone else using the SIP-T20P/SIP-T20 IP phone. You can also listen to voice mails that are stored in a voice mailbox. When receiving a new voice mail, the phone will play a warning tone and the MESSAGE key LED will illuminate. The LCD screen will prompt that the phone receives a new message and display a flashing icon.



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

Note

The voice mail feature is not available on all servers.

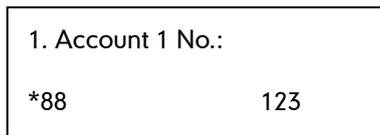
You can configure the phone not to display the pop-up prompt, contact your system administrator for more information.

To leave a voice mail:

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

To configure voice mail access codes via phone user interface:

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



3. Press  or  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  to accept the change.

Note

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

To listen to voice mails:

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

Note

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

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

To view voice mails via phone user interface:

1. Press .

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  to listen to voice mails.

Message Waiting Indicator (MWI)

The SIP-T20P/SIP-T20 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/SIP-T20 IP phone only handles the MWI messages sent from the server. But for other servers, the MWI service is solicited, so the SIP-T20P/SIP-T20 IP phone must enable subscription for MWI.

Note

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

The MWI Subscription parameters you need to know:

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

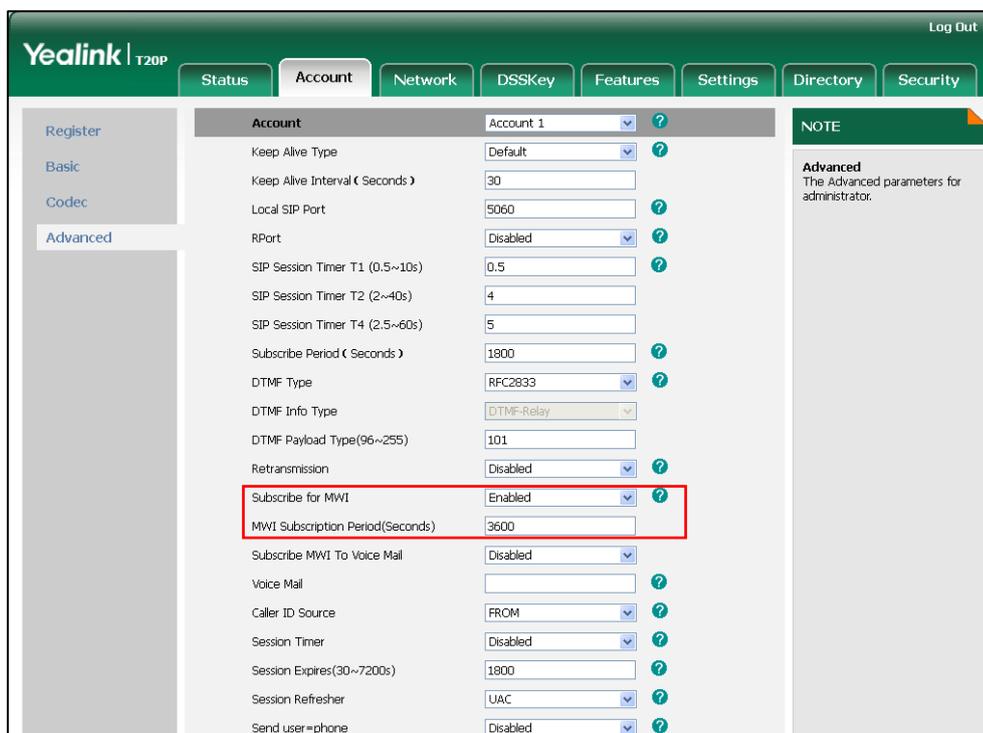
Note

The phone will send SUBSCRIBE messages for the MWI service to the account or the voice number MWI service depending on the server. Contact your system administrator for more information.

To 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.

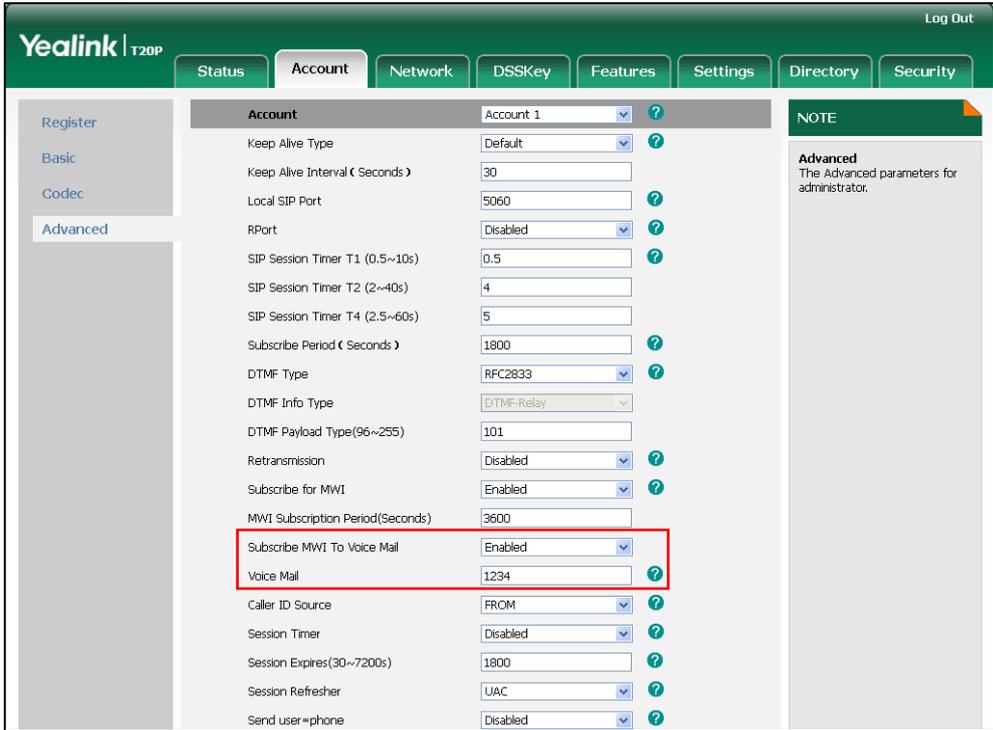


6. Click **Confirm** to accept the change.

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**.

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



The screenshot shows the Yealink T20P web interface. The 'Account' tab is selected, and the 'Advanced' sub-tab is active. The 'Voice Mail' field is highlighted with a red box, and the 'Subscribe MWI To Voice Mail' dropdown is set to 'Enabled'. The 'Voice Mail' field contains the number '1234'. A 'NOTE' box on the right states: 'Advanced: The Advanced parameters for administrator.'

Parameter	Value
Account	Account 1
Keep Alive Type	Default
Keep Alive Interval (Seconds)	30
Local SIP Port	5060
RPort	Disabled
SIP Session Timer T1 (0.5~10s)	0.5
SIP Session Timer T2 (2~40s)	4
SIP Session Timer T4 (2.5~60s)	5
Subscribe Period (Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~255)	101
Retransmission	Disabled
Subscribe for MWI	Enabled
MWI Subscription Period(Seconds)	3600
Subscribe MWI To Voice Mail	Enabled
Voice Mail	1234
Caller ID Source	FROM
Session Timer	Disabled
Session Expires(30~7200s)	1800
Session Refresher	UAC
Send user=phone	Disabled

- 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/SIP-T20 IP phone.

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

Why is the LCD screen blank?

- Ensure that the phone is properly plugged into a functional AC outlet.
- Ensure that the phone is plugged into a socket controlled by a switch that is on.
- If the phone is plugged into a power strip, try plugging it directly into a wall outlet instead.
- If your SIP-T20P IP phone is powered from PoE, ensure that you use a PoE-compliant switch or hub.

Why does the phone display "Network Unavailable"?

- Ensure that the Ethernet cable is plugged into the Internet port 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 to the SNTP server, you need to configure the time and date manually.

How do I find the basic information of the phone?

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

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

Three ways to obtain the MAC address of a phone:

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

Why can't I get a dial tone?

- Check for any loose connections and that the phone has been installed properly. For the installation instructions, refer to [Phone Installation](#) on page 11.
- Switch between the Handset, Headset (if present) 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 25.

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 63.
- Check that call forward is disabled on the phone. Refer to [Call Forward](#) on page 66.
- Check whether the caller number is stored in the blacklist directory. Refer to [Blacklist](#) on page 34.

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 37.

- Check that the headset volume is adjusted to an appropriate level. Refer to [Volume](#) on page 25.

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?

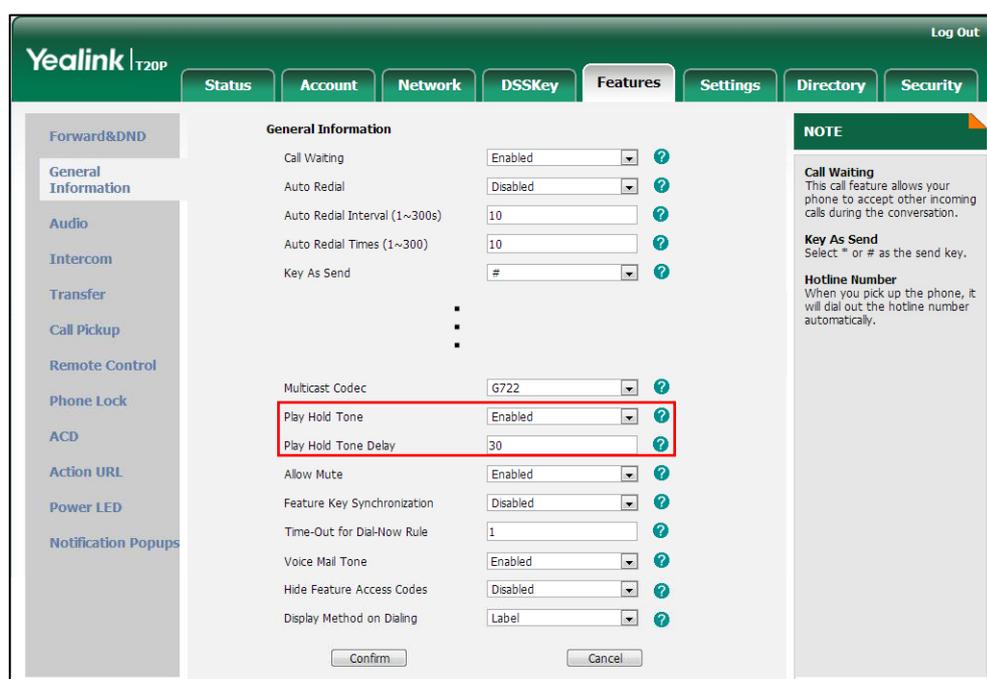
Both user name and register name are defined by the server. A user name is used to identify the account while a register name matched with a password is used for authentication if the server requires. Display name is the caller ID that will be displayed on the callee’s LCD screen. Server configurations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

The screenshot shows the 'Yealink T20P' web interface. The 'Security' tab is active, and the 'Password' sub-tab is selected. The form contains the following elements:

- User Type:** A dropdown menu with 'User' selected.
- Old Password:** An empty text input field.
- New Password:** A text input field with masked characters (dots).
- Confirm Password:** A text input field with masked characters (dots).
- Buttons:** 'Confirm' and 'Cancel' buttons at the bottom.
- NOTE:** A green box on the right with the text: 'User Type: Select your type. If you log in as user, you can only change your own password. If you login as an administrator, you can modify both the user's and admin's passwords. You can click here to get more help through downloading the Administrator Guide!'

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

Note

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

User password is configurable via the web user interface only.

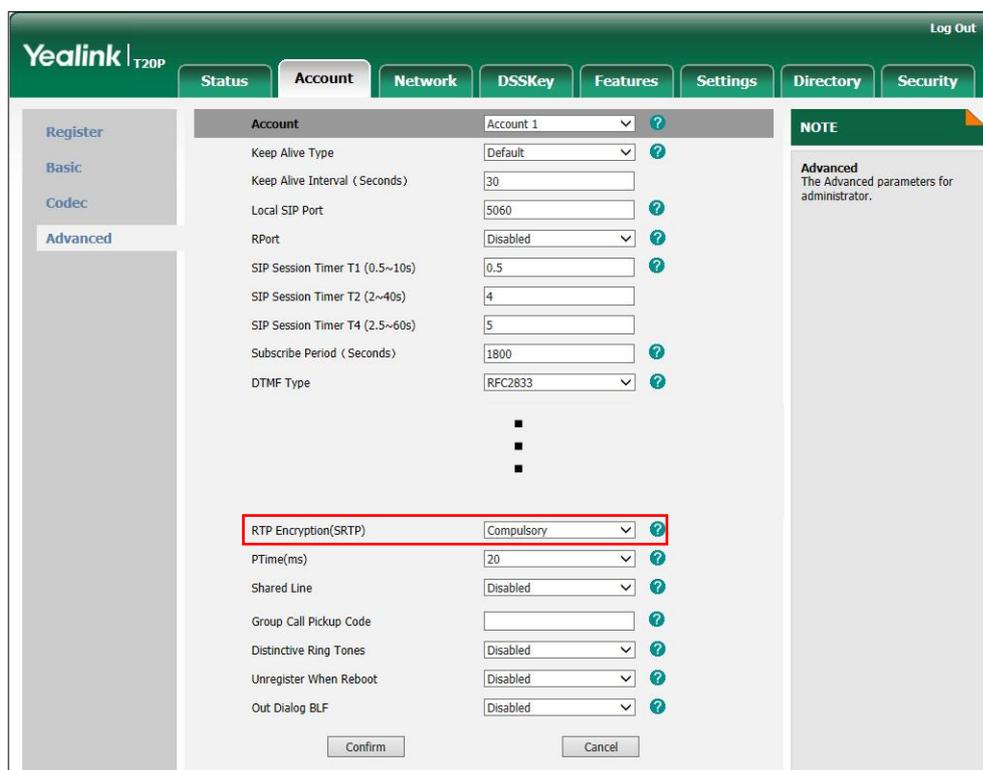
How to make a call using SRTP?

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

To enable SRTP via web user interface:

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

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



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

Note

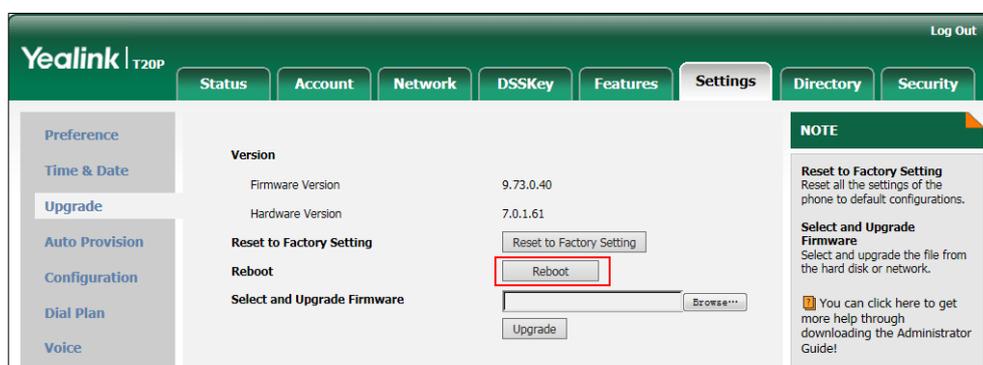
SRTP is not available on all servers. Contact your system administrator for more information.

SRTP is configurable via web user interface only.

How to reboot the phone?

To reboot the phone via web user interface:

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



Note

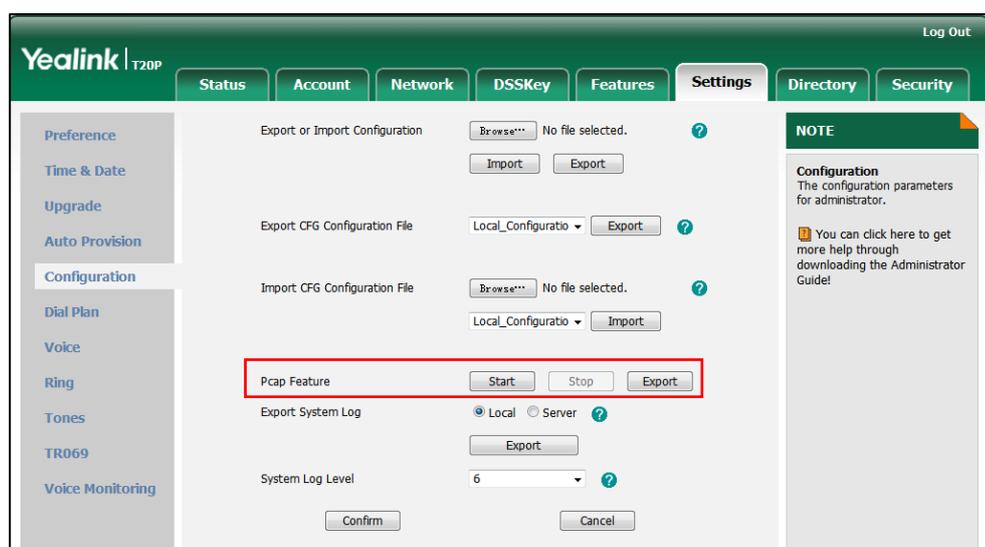
Any reboot of your phone may take a few minutes.

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.



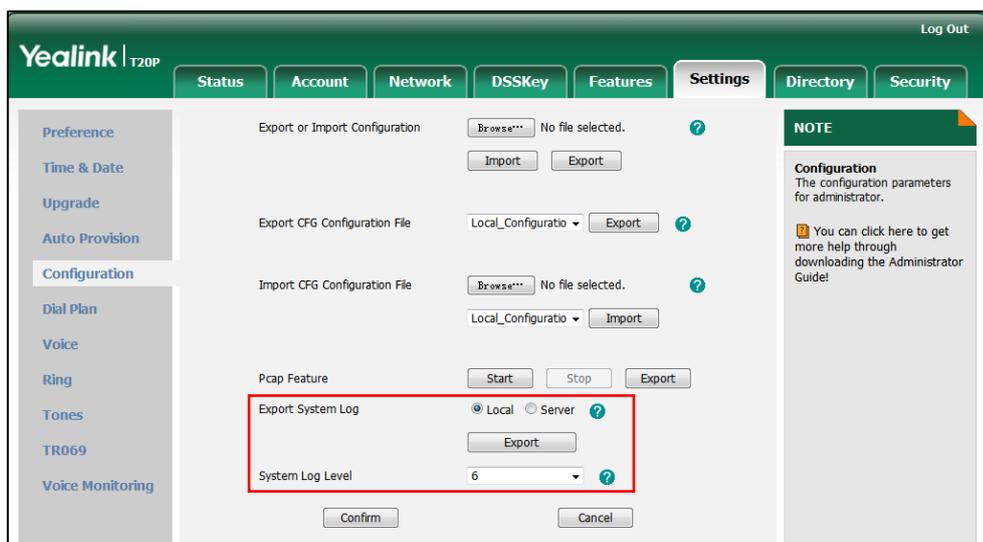
How to export system log?

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

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

1. Click on **Settings->Configuration**.
2. Select **6** from the pull-down list of **System Log Level**.
3. Click **Confirm** to accept the change.
4. Mark the **Local** radio box in the **Export System Log** field.

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



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

Note

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

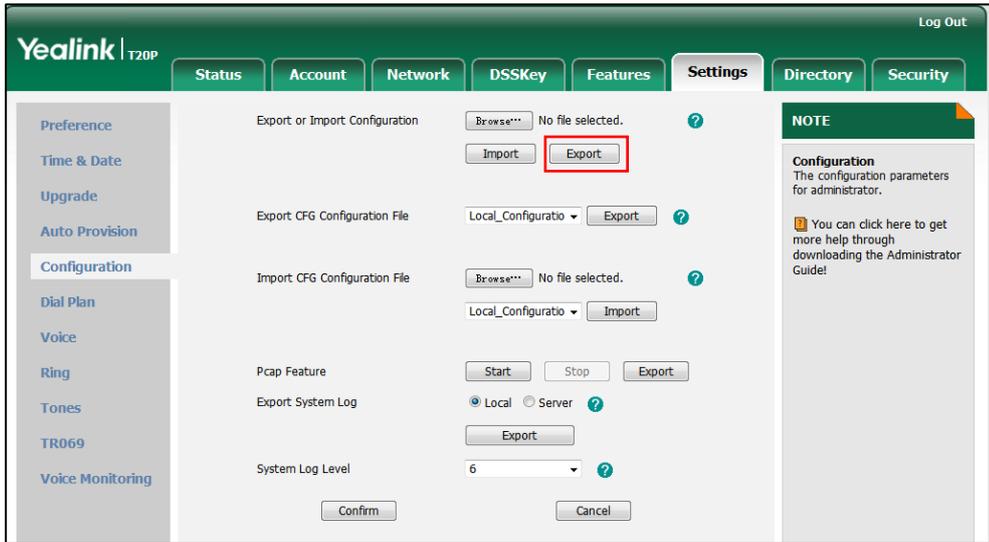
How to export/ import phone configurations?

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

To export the phone configurations via web user interface:

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

2. In the **Export or Import Configuration** block, click **Export** to open file download window, and then save the file to your local system.



To import the phone configurations via web user interface:

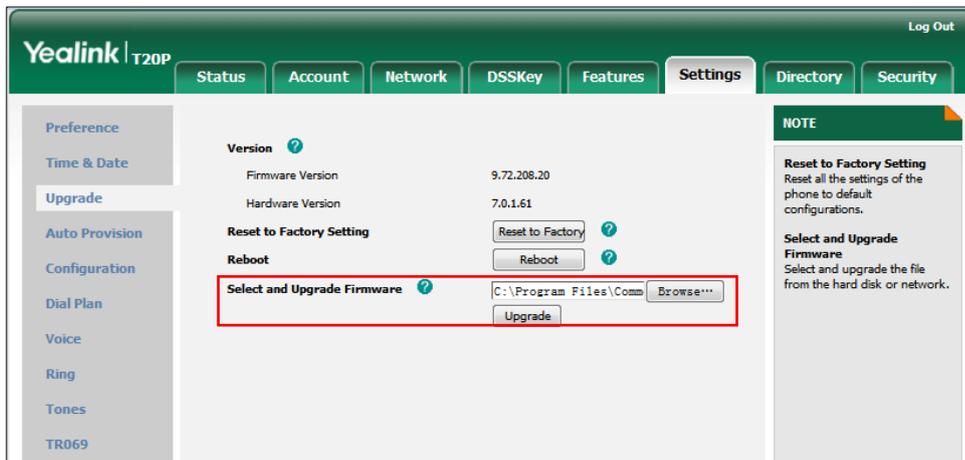
1. Click on **Settings->Configuration**.
2. In the **Export or Import Configuration** block, click **Brower** to locate a configuration file from your local system.
3. Click **Import** to import the configuration file.

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 required firmware from your local system.



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

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

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

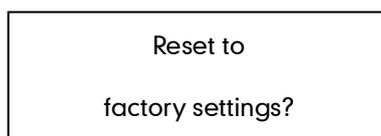
How to reset the phone?

Reset the phone to factory configurations after you have tried all troubleshooting suggestions but do not solve the problem. You need to note that all custom settings will be overwritten after resetting. We recommend asking your system administrator for advice before resetting the phone.

To reset the phone via phone user interface:

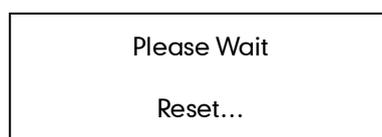
1. Press  .
2. Select **Settings->Advanced Settings** (default password: admin).
3. Press  or  to scroll to **Reset to Factory**, and then press  .

The LCD screen prompts "Reset to factory settings?".

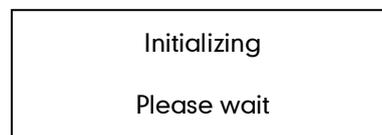


4. Press  to confirm.

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



The LCD screen prompts "Initializing Please wait".



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

Note

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

Regulatory Notices

Service Agreements

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

Limitations of Liability

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

Safety Instructions

Save these instructions. Read these safety instructions before use!

The following basic safety precautions should always be followed to reduce the risk of fire, electrical shock, and other personal injury.

General Requirements

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

Environmental Requirements

- Place the device at a well-ventilated place. Do not expose the device under direct sunlight.
- Keep the device dry and free of dust.

- Place the device on a stable and level platform.
- Please do not place heavy objects on the device in case of damage and deformation caused by the heavy load.
- Keep at least 10 cm between the device and the closest object for heat dissipation.
- Do not place the device on or near any inflammable or fire-vulnerable object, such as rubber-made materials.
- Keep the device away from any heat source or bare fire, such as a candle or an electric heater.
- Keep the device away from any household appliance with a strong magnetic field or electromagnetic field, such as a microwave oven or a refrigerator.

Operating Requirements

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



Cleaning Requirements

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

Appendix A - Time Zones

Time Zone	Time Zone Name
- 11:00	Samoa
- 10:00	United States-Hawaii-Aleutian
- 09:30	French Polynesia
- 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)
- 02:30	Newfoundland and Labrador
- 02:00	Brazil(no DST)
- 01:00	Portugal(Azores)
0	GMT
0	Greenland
0	Denmark-Faroe Islands(Torshavn)
0	Ireland(Dublin)
0	Portugal(Lisboa, Porto, Funchal)
0	Spain-Canary Islands(Las Palmas)
0	United Kingdom(London)
0	Morocco
+ 01:00	Albania(Tirane)
+ 01:00	Austria(Vienna)
+ 01:00	Belgium(Brussels)
+ 01:00	Caicos
+ 01:00	Chad
+ 01:00	Spain(Madrid)
+ 01:00	Croatia(Zagreb)
+ 01:00	Czech Republic(Prague)
+ 01:00	Denmark(Kopenhagen)

Time Zone	Time Zone Name
+01:00	France(Paris)
+01:00	Germany(Berlin)
+01:00	Hungary(Budapest)
+01:00	Italy(Rome)
+01:00	Luxembourg(Luxembourg)
+01:00	Macedonia(Skopje)
+01:00	Netherlands(Amsterdam)
+01:00	Namibia(Windhoek)
+02:00	Estonia(Tallinn)
+02:00	Finland(Helsinki)
+02:00	Gaza Strip(Gaza)
+02:00	Greece(Athens)
+02:00	Israel(Tel Aviv)
+02:00	Jordan(Amman)
+02:00	Latvia(Riga)
+02:00	Lebanon(Beirut)
+02:00	Moldova(Kishinev)
+02:00	Russia(Kaliningrad)
+02:00	Romania(Bucharest)
+02:00	Syria(Damascus)
+02:00	Turkey(Ankara)
+02:00	Ukraine(Kyiv, Odessa)
+03:00	East Africa Time
+03:00	Iraq(Baghdad)
+03:00	Russia(Moscow)
+03:30	Iran(Teheran)
+04:00	Armenia(Yerevan)
+04:00	Azerbaijan(Baku)
+04:00	Georgia(Tbilisi)
+04:00	Kazakhstan(Aktau)
+04:00	Russia(Samara)
+04:30	Afghanistan(Kabul)
+05:00	Kazakhstan(Aqtobe)
+05:00	Kyrgyzstan(Bishkek)
+05:00	Pakistan(Islamabad)
+05:00	Russia(Chelyabinsk)
+05:30	India(Calcutta)
+05:45	Nepal(Katmandu)
+06:00	Kazakhstan(Astana, Almaty)
+06:00	Russia(Novosibirsk, Omsk)
+06:30	Myanmar(Naypyitaw)
+07:00	Russia(Krasnoyarsk)
+07:00	Thailand(Bangkok)
+08:00	China(Beijing)
+08:00	Singapore(Singapore)
+08:00	Australia(Perth)
+08:00	Russia(Irkutsk, Ulan-Ude)
+08:45	Eucla
+09:00	Korea(Seoul)
+09:00	Japan(Tokyo)
+09:00	Russia(Yakutsk, Chita)
+09:30	Australia(Adelaide)
+09:30	Australia(Darwin)
+10:00	Australia(Sydney, Melbourne, Canberra)
+10:00	Australia(Brisbane)
+10:00	Australia(Hobart)

Time Zone	Time Zone Name
+ 10:00	Russia(Vladivostok)
+ 10:30	Australia(Lord Howe Islands)
+ 11:00	New Caledonia(Noumea)
+ 11:00	Russia(Srednekolymk Time)
+ 11:30	Norfolk Island
+ 12:00	New Zealand(Wellington, Auckland)
+ 12:00	Russia(Kamchatka Time)
+ 12:45	New Zealand(Chatham Islands)
+ 13:00	Tonga(Nukualofa)
+ 13:30	Chatham Islands
+ 14:00	Kiribati

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