

Yealink



HD 

SIP-T20P

Enterprise IP Phone

User Guide

Version 71.110

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

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

Class B Digital Device or Peripheral

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

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

WEEE Warning



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

GNU GPL INFORMATION

Yealink SIP-T20P firmware contains third-party software under the GNU General Public License (GPL).

Yealink uses software under the specific terms of the GPL. Please refer to the GPL for the exact terms and conditions of the license.

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

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

About This Guide

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

This guide provides everything you need to quickly use your new phone. Be sure to verify with your system administrator that your network is prepared for configuring your IP phone. As well, be sure to read the Packaging Contents and Regulatory Notices sections in this guide before you set up and use the SIP-T20P IP phone.

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

<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Note

The Shared Line, Busy Lamp Field List, Network Directory and Network Call Log features are designed for BroadWorks environment. Please refer to BroadWorks documents or contact your system administrator for more information.

In This Guide

The topics provided in this guide include:

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

Summary of Changes

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

Changes for Release 71, Guide Version 71.110

Major updates have occurred to the following sections:

- [Basic Network Settings](#) on page 14
- [Keypad Lock](#) on page 21

- [Contact Management](#) on page 26
- [DSS Keys](#) on page 37

Changes for Release 70, Guide Version 70

Major updates have occurred to the following:

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

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Overview

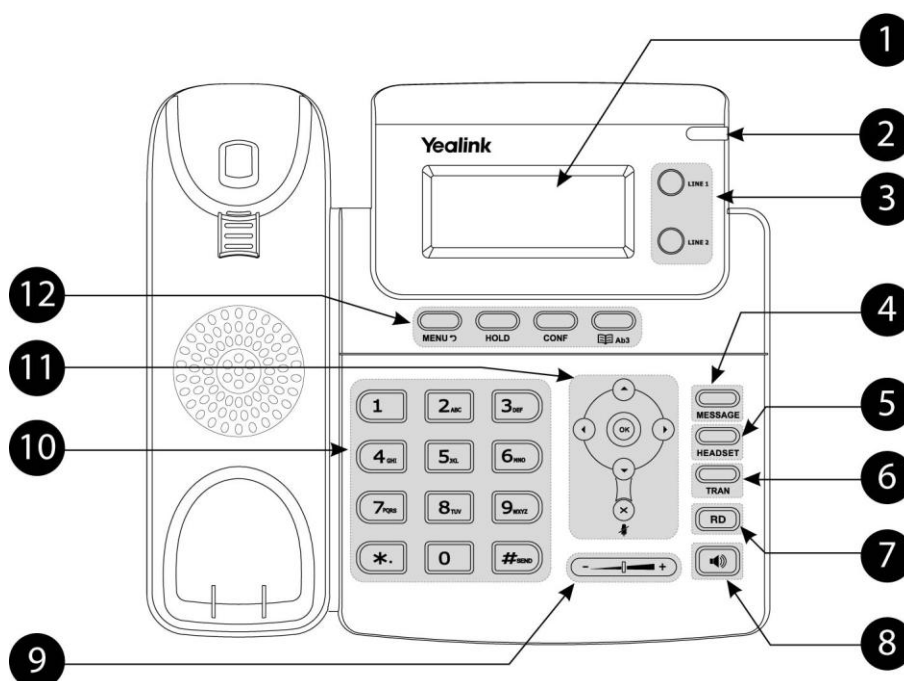
This chapter provides the overview of the SIP-T20P IP phone. The topics include:

- [Hardware Component Instructions](#)
- [Icon Instructions](#)
- [LED Instructions](#)
- [User Interfaces](#)
- [Documentations](#)




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

Hardware Component Instructions

The main hardware components of the SIP-T20P IP phone are the LCD screen and the keypad.














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

	Item	Description
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 a call.
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 interface or switches the input mode.

Icon Instructions

Icons appear on the phone 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
	Keypad Lock
	Received Calls
	Placed Calls
	Missed Calls

LED Instructions

Power Indicator LED

LED Status	Description
Solid green	The phone is powered on.
Fast Flashing green	The phone is ringing.
Slow flashing green	The phone receives a voice mail.
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 BLF key)

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

User Interfaces

There are two ways to customize specific configurations on your SIP-T20P IP phone:

- Using the User Interface on the IP phone.
- Using the User Interface in an Internet browser window from 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 instances, it is possible to use both the phone user interface and the web user interface to operate the phone and change settings. However, in some instances, it is only possible to use the phone user interface or the web user interface.

Phone User Interface

You can customize your phone by pressing the **MENU** key to access the phone user interface. The Advanced options are only accessible to an administrator, and the default administrator password is admin (case-sensitive). For more information on customizing your phone using the available options from the phone user interface, refer to [Customizing Your Phone](#) on page 17.

Web User Interface

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

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

Options	Phone User Interface	Web User Interface
Status	√	√
--IPv4		
--MAC		
--Firmware		
--Network		
--Phone		
--Accounts		
Basic Call Features	√	√
--Auto Answer		
--Auto Redial		
--Call Completion		
--Call Forward		
--Call Waiting		
--Call Park		
--Call Pickup		
--Call Return		
--Conference		
--Anonymous Call		
--Anonymous Call Rejection		
--DND		
Basic Phone Settings		
--Language	√	√

Options	Phone User Interface	Web User Interface
--Time & Date	√	√
--Administrator Password	√	√
--Ring Tones	√	√
--Keypad Lock	√	√
--Phone Volume	√	
--Logo Customization		√
--Call History Management	√	√
--Contact Management		
--Local Directory	√	√
--Blacklist	√	√
--Dial Plan		√
--DSS Keys	√	√
--Key as Send	√	√
--Hot Line	√	√
--Live Dialpad		√
--Emergency Call		√
Advanced Phone Features		√
--BLF	√	
--Call Recording	√	
--Hot Desking	√	
--Intercom	√	
--Multicast Paging		
--Music on Hold		
--ACD	√	
--Messages	√	
SIP Account		√
--User Options		
--Register Status	√	
--Activation	√	
--Label	√	
--Display Name	√	
--Register Name	√	
--User Name	√	
--Password	√	
--SIP Server1/2	√	
--Server Option		
--Registrar Port		
--Outbound Status	√	
--Outbound Proxy	√	
--NAT Traversal		
--STUN Status	√	

Options	Phone User Interface	Web User Interface
--STUN Server	√	

Note

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

Documentations

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

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

Note

You can also download the latest documentations from website:
<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Getting Started

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

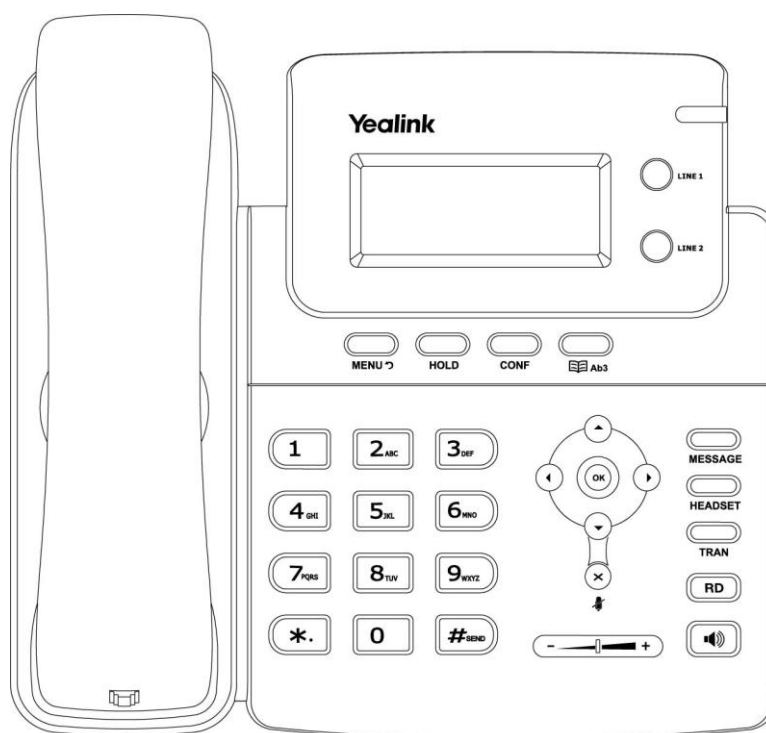
- [Packaging Contents](#)
- [Phone Installation](#)
- [Phone Initialization](#)
- [Phone Status](#)
- [Basic Network Settings](#)
- [Registration](#)
- [Idle Screen](#)

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

Packaging Contents

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

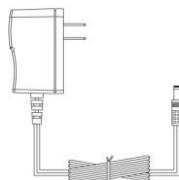
- **SIP-T20P IP phone**



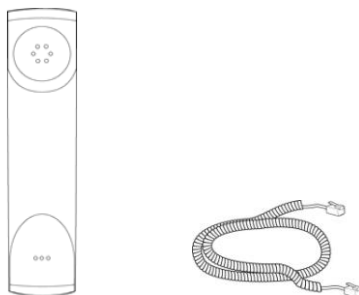
- **Phone Stand**



- **Power Adapter**



- **Handset & Handset Cord**



- **Ethernet Cable**



- **Quick Installation Guide & Quick Reference Guide**



- **CD-ROM**



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

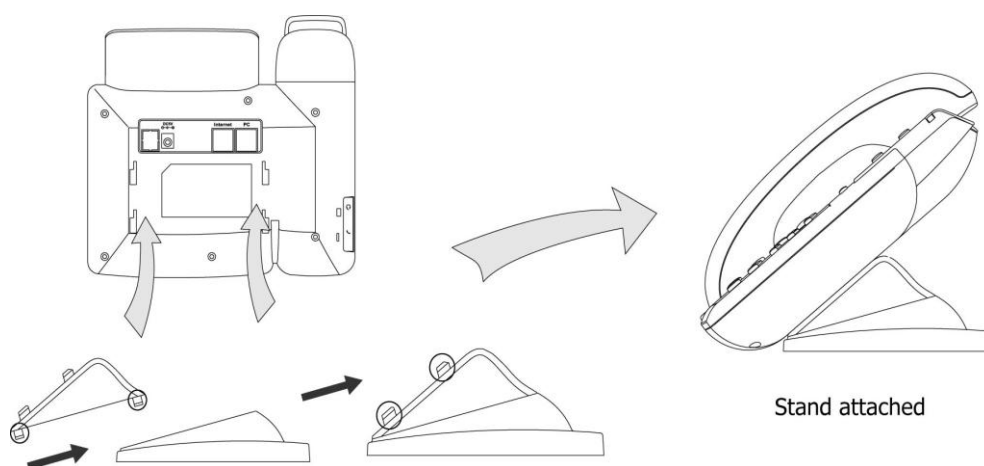
Phone Installation

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

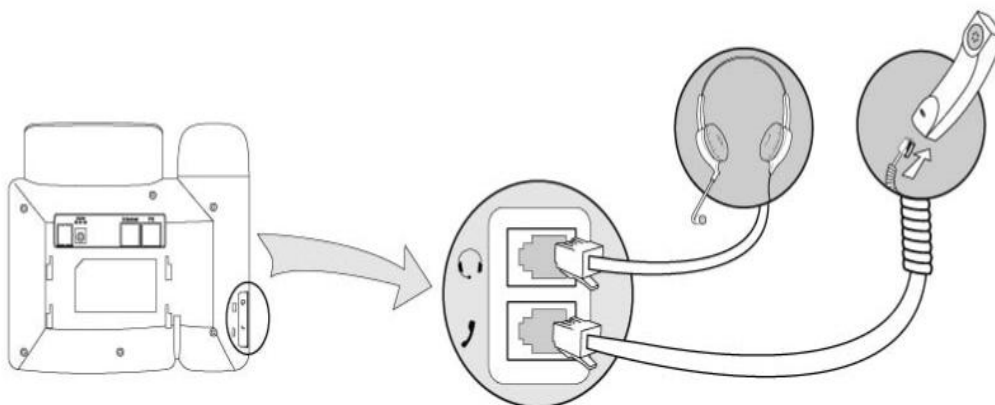
This section introduces how to install the phone with the components in the packaging contents:

- Attach the stand
- Connect the handset and optional headset
- Connect the network and power

1) Attach the Stand



2) Connect the Handset and optional Headset



Note

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

3) Connect the Network and Power

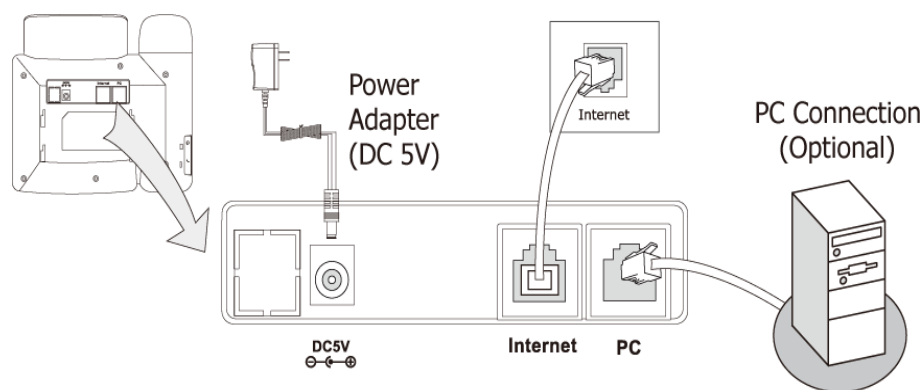
You have two options for power and network connections. Your system administrator will advise you which one to use.

- 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 supplied Ethernet cable between the Internet port on the phone and the Internet port in your network or switch/hub device port.

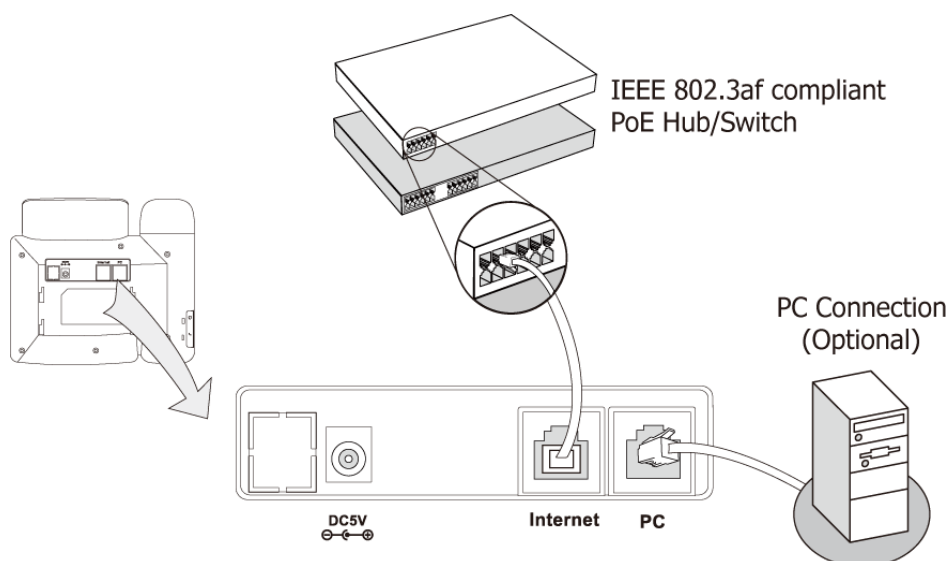


Power over Ethernet

Using a regular Ethernet cable, the SIP-T20P IP phone can be powered from a PoE (IEEE 802.3af) compliant switch or hub.

To connect the PoE:

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

**Note**

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

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

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

Phone Initialization

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

Automatic Phone Initialization

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

DHCP (Dynamic Host Configuration Protocol)

By default the phone attempts to contact a DHCP server in your network to obtain valid IPv4 and IPv6 network settings, e.g., IP address, subnet mask, default gateway address and DNS address.

Note

If your network does not use DHCP, proceed to [Basic Network Settings](#) on page 14.





Phone Status

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

Available information of phone status includes:

- Network status including IPv4 status, IPv6 status, IP mode, MAC address, LAN type, LAN IP and LAN subnet.
- Phone status including product name, hardware version, firmware version, product ID, MAC address and device certificate status (including factory, installed and not installed).
- Account status indicating the register status of SIP accounts.

To view the phone status via phone user interface:

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

1. IPv4:

192.168.0.10

To view the phone status via web user interface:

1. Open the web browser of your computer.
2. Enter the IP address in the browser's address bar, and then press **Enter**.
3. Enter the user name (admin) and password (admin) in the login page.
4. Click **Confirm** to log in.

The phone status displays 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. IP phones support to use the IPv4 address only, the IPv6 address only or both IPv4 and IPv6 addresses.

To configure the IP address mode via phone user interface:

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


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** (password: admin) ->**Network->WAN Port->IPv4->Static IP**.
3. Enter the desired values in the **IPv4**, **Subnet Mask**, **Default Gateway**, **Pri DNS** and **Sec DNS** fields respectively.

1. IPv4: 192.168.0.3

123


4. Press  to accept the change.

To configure a static IPv6 address via phone user interface:

1. Press .
2. Select **Settings->Advanced** (password: admin) ->**Network->WAN Port->IPv6->Static IPv6**.
3. Enter the desired values in the **IPv6 IP**, **IPv6 IP Prefix**, **IPv6 Default Gateway**, **IPv6 Pri.DNS** and **IPv6 Sec.DNS** fields respectively.

1. IPv6 IP: 2005:1:1:1::12

2aB

4. Press  to accept the change.

If you are using an xDSL modem, you can connect your phone to the Internet via PPPoE mode. You can set the WAN port to be 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** (password: admin) ->**Network->WAN Port->IPv4->PPPoE**.

1. PPPoE User:

2aB

3. Enter the user name and password in the corresponding fields.
4. Press  to accept the change.

Note

Using the wrong network parameters 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 to use. If your phone is not registered, you may have to register it. For more information on how to register your phone, refer to [Account Management](#) on page 43.

Idle Screen

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

1234

17 Jan 09:20

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

Customizing Your Phone

You can customize your SIP-T20P IP phone by configuring the language, time & date, ring tones and so on. You can handle incoming calls from different contacts in different ways.

This chapter provides basic operating instructions for customizing your phone. The topics include:

- [General Settings](#)
- [Audio Settings](#)
- [Contact Management](#)
- [Call History Management](#)
- [System Customizations](#)

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

General Settings

Language


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

Note

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


To change the language for the phone user interface:

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

- Press  or  to select the desired language.

*1. English

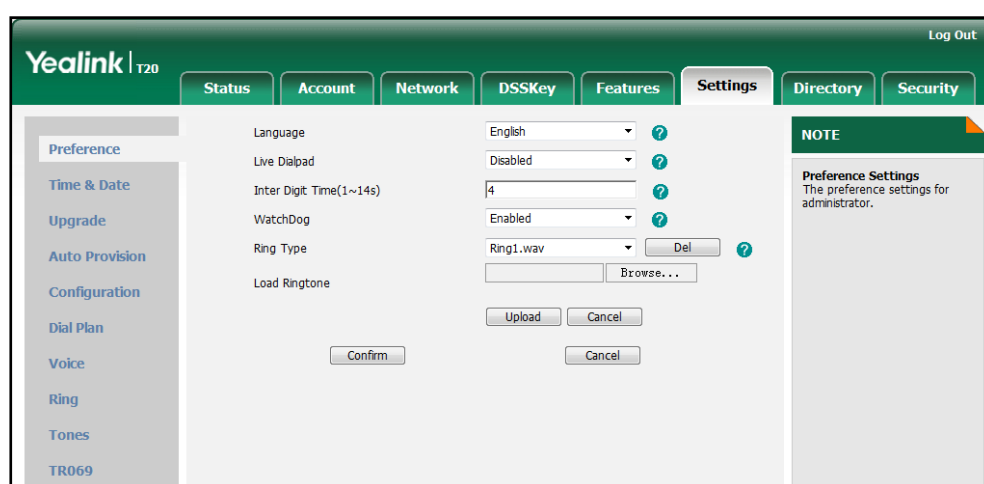
2. Deutsch

- Press  to accept the change.

Text appearing on the LCD screen will use the selected language.

To change the language for web user interface:

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






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

Text displaying on the web user interface will use the selected language.

Time & Date

The time and date display 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:

- Press .
- Select **Settings->Basic->Time & Date->SNTP**.
- 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)".


1. Time Zone:
◀ *+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 **Daylight Saving** field.
6. Press OK to accept the change.

Note

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

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

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

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

5. Press OK to accept the change.

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

To configure the time format via phone user interface:

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

1. Clock:
◀ *24 Hour ▶


3. Press ◀ or ▶ to select the desired time format from the **Clock** field.
4. Press OK to accept the change.

You can also configure the time and date via web user interface at the path **Settings->Time & Date**.

Administrator Password


The Advanced Settings option is only accessible to an 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** (password: admin) ->**Set Password**.

1. Old PWD:

abc

3. Enter the old password in the **Old PWD** field.
4. Enter the new password in the **New PWD** field.
5. Enter the new password again in the **Confirm PWD** field.
6. Press  to accept the change.

You can also configure the administrator password 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 the send key feature 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 **Disable** from the **Key as Send** field to disable this feature.
4. Press  to accept the change.

You can also configure a send key via web user interface at the path **Features->General Information**.

Keypad Lock

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




Keypad Lock consists of the following:

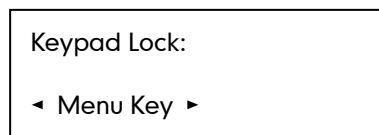
- Menu Key:** The **MENU** key and **MESSAGE** key are locked. You cannot access the menu of the phone until unlocked.
- Function Key:** The function keys are locked. You cannot use the **MESSAGE, CONF, HOLD, MENU, TRAN, RD, Directory, OK, X**, line keys and navigation keys until unlocked.
- All Keys:** 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, place an active call on hold by pressing the **HOLD** key, mute/un-mute the call during an active call and end the call by pressing the **X** key.



Note


The emergency number setting, if desired, must be made before lock activation. For more information, refer to [Emergency Number](#) on page 49.

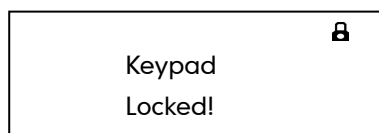
To activate the keypad lock via phone user interface:


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



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

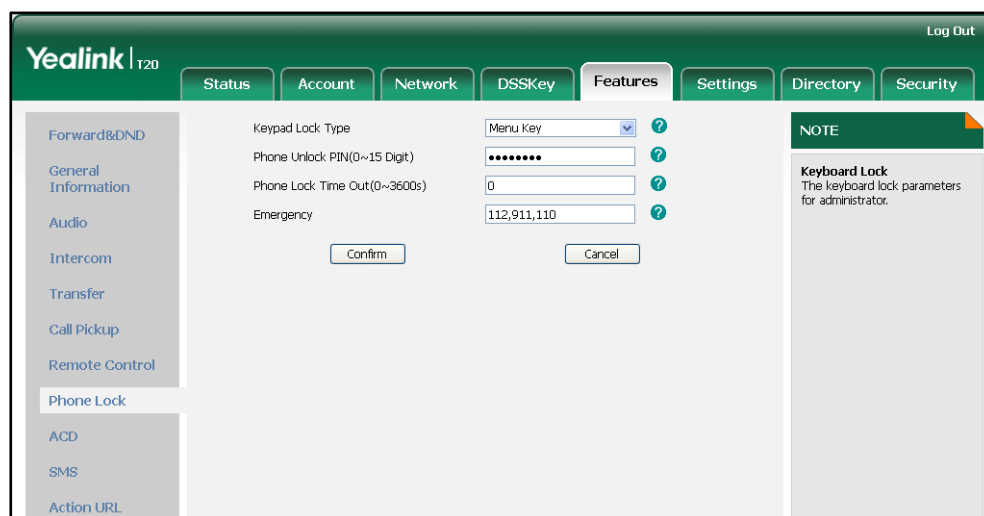
The LCD screen prompts "Keypad Locked!" and displays the icon .



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


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

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



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


Note

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

You can configure the interval of automatic keypad lock via web user interface only.


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

To change the keypad unlock PIN via phone user interface:

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

1. Current PIN:


123

4. Enter the new PIN in the **New PIN** field.
5. Enter the new PIN again in the **Confirm PIN** field.
6. Press  to accept the change.


Note

The unlock PIN length must be within 15 characters.

To unlock the keypad via phone user interface:

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




The  icon disappears from the LCD screen.

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

Note


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

To deactivate the keypad lock via phone user interface:

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

Keypad Lock:

◀ Disable ▶

4. Press  to accept the change.


You can also activate or deactivate the keypad lock 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 used audio devices (handset, speakerphone or headset) when the phone is during a call.

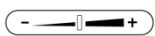
To adjust the volume when the phone is idle:

1. Press  to adjust the ringer volume.

1009



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




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




Ring Tones

Ring tones are used to indicate incoming calls. You can select different ring tones to distinguish the 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 ->Ring Tones**.
3. Press  or  to select the desired ring tone.

*1. Ring1.wav
2. Ring2.wav

4. Press  to accept the change.

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

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Basic**.
4. Select the desired ring tone from the pull-down list of **Ring Type**.

If **Common** is selected, this account will use the ring tone selected for the phone at the path of the web user interface **Settings->Preference**. Refer to the above instruction.

The screenshot shows the 'Account' configuration page for 'Account 1'. The 'Ring Type' is set to 'Common'. The interface includes a sidebar with options like Register, Basic, Codec, and Advanced. A 'NOTE' section on the right explains the 'Basic' and 'Proxy Require' parameters.

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

Note

You can select a ring tone for the account via web user interface only.

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

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

The screenshot shows the 'Preference' configuration page. The 'Ring Type' is set to 'Ring1.wav'. The interface includes a sidebar with options like Preference, Time & Date, Upgrade, Auto Provision, Configuration, Dial Plan, Voice, Ring, Tones, and TR069. A 'NOTE' section on the right explains the 'Preference Settings'.

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

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](#) in the [Contact Management](#) section). Otherwise, the phone will play the ring tone assigned to the account. If both the contact ring tone and the account ring tone are not assigned, then the phone will play the ring tone assigned to the phone.

All custom ring tone files must be within 100KB. You can upload custom ring tones for your phone via web user interface only.

Contact Management

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






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

Local Directory

The built-in phone directory stores the names and phone numbers of your contacts. You can store up to 1000 contacts and 7 groups (including the default All Contacts and Blacklist) in your phone's local directory. You can add new groups and add new contacts to different groups in the local directory. You can edit, delete or search for a contact in the local directory. You can also dial a contact 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 **AddGroup** 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.

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 **Del** and then press .

The LCD screen prompts "Delete Selected Group?".

Delete

Selected Group?

5. Press  to accept the change.





Adding Contacts




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





- Manually
- From call history

Adding Contacts Manually

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  to switch the input mode.

Name:

abc






7. Enter the name and the office, mobile or other numbers in the corresponding fields
8. Press  or  to select the desired account from the **Account** field.
If **Auto** is selected, the phone will use the first available account when placing calls to the contact from the local directory.
9. Press  or  to select the desired ring tone from the **Ring Tones** field.
10. Press  to accept the change.

Note

If the contact has existed in the directory, the LCD screen prompts "ContactName existed!".


Adding Contacts from Call History

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

1. Press .
2. Press  or  to select the desired entry.
3. Press .
4. Press  to switch the input mode, and enter the contact name in the **Name** field.

Name:










abc

5. Press  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 **Del** and then press .
- The LCD screen prompts "Delete selected Item?".
6. Press  to confirm the deleting.

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 **Dial**.
 6. Do one of the following:
 - If only one number of the contact is stored in the local directory, press  to dial out the number.
 - If multiple numbers of 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 phone number using the keypad.

Search: tom

2aB

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 file of contact list via web user interface:

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

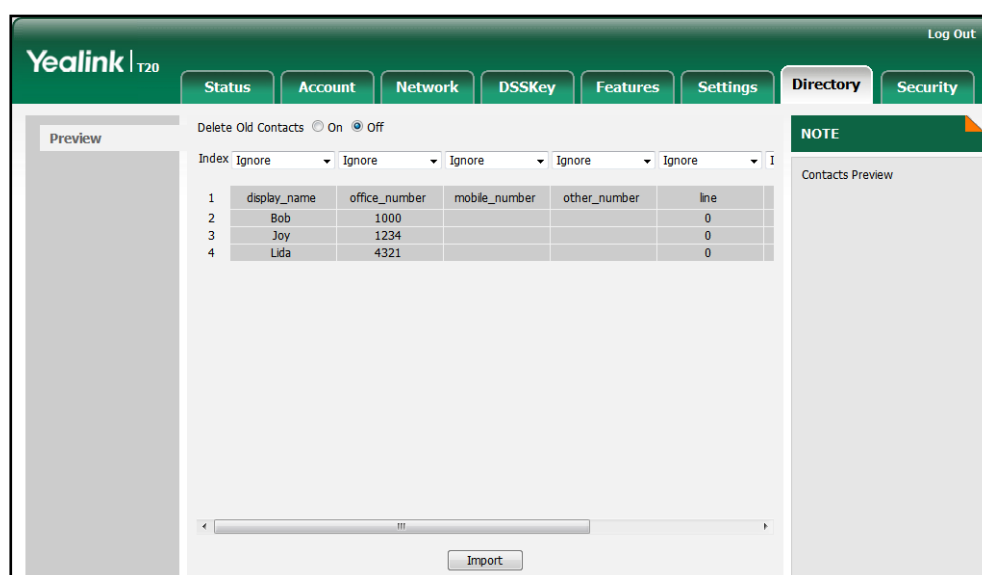
- Click **Browse** to locate a contact list file (file format must be *.xml) from your local system.

- 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 file of contact list via web user interface:

- Click on **Directory->Local Directory**.
- Click **Browse** to locate a contact list file (file format must be *.csv) from your local system.
- (Optional.) Check the **Show Title** checkbox.
It will prevent importing the title of the contact information which is located in the first line of the CSV file.
- 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**.



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

To export contact list via web user interface:

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

Note You can import/export contact lists via web user interface only.

Blacklist

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




Operating instructions of adding blacklists, editing blacklists, deleting blacklists, placing call to blacklists and searching for a contact in the blacklist, refer to the operating instructions of [Local Directory](#) on page 26.

Call History Management

The SIP-T20P IP phone maintains call history lists of Placed Calls, Received Calls, Missed Calls and Forwarded Calls. The call history lists support to store 100 entries in all. You can check the call history, dial a call, add a contact or delete an entry from the call


history list. You should enable the history record feature in advance.

To enable the history record feature via phone user interface:







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

1. History Record:







◀*Enable ▶

4. Press  to accept the change.








To check the call history:

1. Press .
- The LCD screen displays the last call history.
2. Press  or  to switch between **Placed Calls**, **Received Calls**, **Missed Calls** and **Forwarded Calls**.
3. Press  or  to select the desired entry.
4. Press .
- The detailed information of the entry appears on the LCD screen.

To dial 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 26.

To delete an entry from the call history list:

1. Press  .
The LCD screen displays the call list.
2. Press  or  to switch between **Placed Calls**, **Received Calls**, **Missed Calls** and **Forwarded Calls**.
3. Press  or  to select the desired entry.
4. Press  to delete the entry.

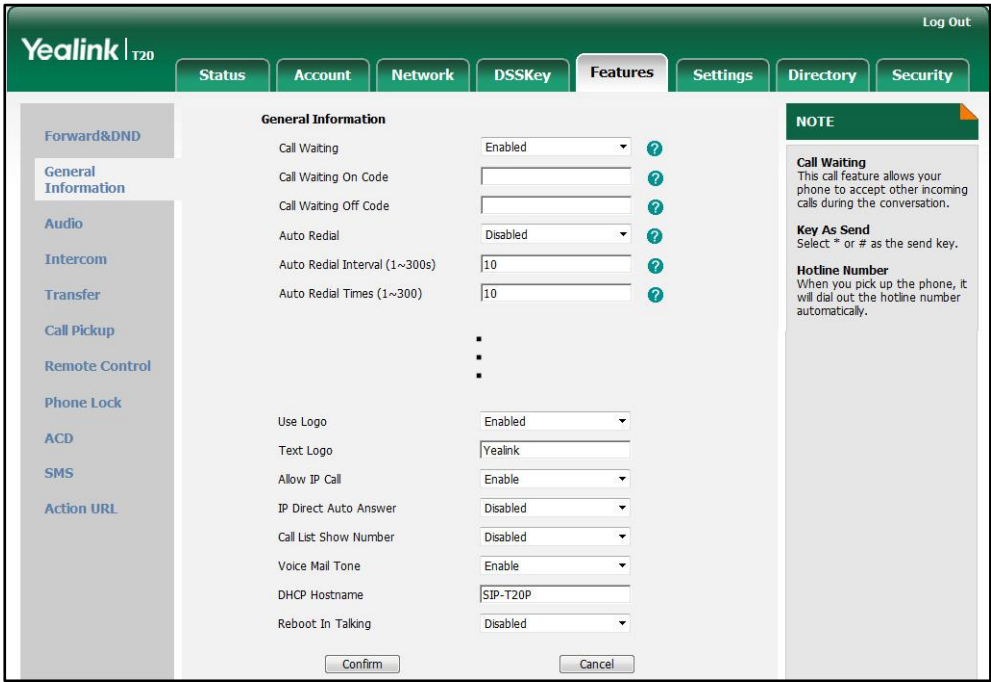
System Customizations

Logo Customization

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

To customize a text logo via web user interface:

1. Click on **Features->General Information**.
2. Select **Enabled** from the pull-down list of **Use Logo**.
3. Enter the desired text in the **Text Logo** field.



The screenshot shows the Yealink T20 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Features' tab is selected, and the 'General Information' sub-tab is active. On the left sidebar, 'General Information' is highlighted. The main content area shows various settings: 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Auto Redial Times (1~300)' (10), 'Use Logo' (Enabled), 'Text Logo' (Yealink), 'Allow IP Call' (Enable), 'IP Direct Auto Answer' (Disabled), 'Call List Show Number' (Disabled), 'Voice Mail Tone' (Enable), 'DHCP Hostname' (SIP-T20P), and 'Reboot In Talking' (Disabled). There are 'Confirm' and 'Cancel' buttons at the bottom. A 'NOTE' section on the right provides details about 'Call Waiting', 'Key As Send', and 'Hotline Number'.

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

Note

The maximum length of text logo is 15 characters.

You can configure logo customization via web user interface only.


Headset Use


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

Headset Mode Activation/Deactivation

To activate the headset:

1. Press  on the phone.

The  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 then press , the phone will use the headset to place a call automatically. For more information on using the headset to place a call, refer to [Placing Calls](#) on page 53.

To deactivate the headset:

1. Press  again on the phone.

The headset icon disappears when the headset mode is deactivated.

Headset Prior

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

To enable the 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 Yealink T20 web interface. The 'Features' tab is selected, and the 'General Information' section is active. The 'Headset Prior' dropdown menu is set to 'Enabled'. Other settings include 'Call Waiting' (Enabled), 'Call Waiting On Code' (empty), 'Call Waiting Off Code' (empty), 'Auto Redial' (Disabled), and 'Auto Redial Interval (1~300s)' (10). There are also settings for 'DTMF Replace Tran' (Disabled), 'Tran Send DTMF' (empty), 'Send Pound Key' (Disabled), 'IP Direct Auto Answer' (Disabled), 'Call List Show Number' (Disabled), 'Voice Mail Tone' (Enable), 'DHCP Hostname' (SIP-T20P), and 'Reboot In Talking' (Disabled). 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 the headset prior feature is enabled, the headset mode will not be deactivated until you press the **HEADSET** key again.

You can configure headset prior via web user interface only.

Dual Headset

You can use two headsets when enabling the dual headset feature. To use this feature, you must physically connect headsets to the headset jack and handset jack respectively. Once the phone joins in a call, people with the headset connected to the headset jack has a full-duplex conversation, another people with the headset connected to the handset jack is only allowed to listen to.

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 T20 web interface with the 'Features' tab selected. The 'General Information' section contains various settings. The 'Dual-Headset' setting is currently set to 'Enabled'. Other settings include 'Call Waiting' (Enabled), 'Call Waiting On Code', 'Call Waiting Off Code', 'Auto Redial' (Disabled), 'Auto Redial Interval (1~300s)' (10), 'Headset Prior' (Disabled), 'DTMF Replace Tran' (Disabled), 'Tran Send DTMF', 'IP Direct Auto Answer' (Disabled), 'Call List Show Number' (Disabled), 'Voice Mail Tone' (Enable), 'DHCP Hostname' (SIP-T20P), and 'Reboot In Talking' (Disabled). A 'NOTE' section on the right provides additional information: 'Call Waiting' allows accepting other incoming calls; 'Key As Send' allows selecting * or # as the send key; and 'Hotline Number' allows dialing the hotline number automatically when the phone is picked up.

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

Note

You can configure dual headset 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 IP phone supports 2 line keys.

Line Keys

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

To configure the line key via phone user interface:

1. Press .
2. Select **Features->DSS Keys**.
3. Select the desired DSS key, and then press .
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 line from the **Account ID** field.

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

You can also configure the line key 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.

The line key features are explained in the following subchapters in detail:

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

For more information, contact your system administrator.

Speed Dial

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

Dependencies: *Type (Speed Dial)*

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

Value (the number you want to dial out)

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

Voice Mail

You can use this key feature to connect voice mail quickly. For more information, refer to [Voice Mail](#) on page 92.

Dependencies: *Type (Key Event)*

Key Type (Voice Mail)

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

Value (the voice mail access code)

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

DTMF

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

Dependencies: *Type (Key Event)*

Key Type (DTMF)

Value (DTMF sequence)

Note

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

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

Prefix

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

Dependencies: *Type (Key Event)*

Key Type (Prefix)

Value (the prefix number)

Usage: Press the DSS key when the phone is idle, the phone enters into the dialing interface and displays the prefix number which you specified in the **Value** field, enter other digits to dial out.

Local Group

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

Dependencies: *Type (Key Event)*

Key Type (Local Group)

Local Group (the contact group you want to access)

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

Conference

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

Dependencies: *Type (Key Event)*

Key Type (Conf)

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

Usage: Press the DSS key during an active call to set up a conference with the number

specified in the **Value** field.

Note

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

Forward

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

Dependencies: *Type (Key Event)*

Key Type (Forward)

Value (the number you want to forward to)

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

Note

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

Transfer

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

Dependencies: *Type (Key Event)*

Key Type (Transfer)

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

You can configure the transfer mode on DSS key via web user interface at the path **Features->Transfer**. For more information on how to configure the transfer mode on DSS key, refer to [Busy Lamp Field \(BLF\)](#) on page 79.

When leaving the **Value** field blank, the DSS key performs the same function as the **TRAN** key during a call. For more information, refer to [Call Transfer](#) on page 68.

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 the DND feature. You can also use this key feature to access the custom DND interface. For more information, refer to [Do Not Disturb \(DND\)](#) on page 61.

Dependencies: *Type (Key Event)*

Key Type (DND)

Usage:

When the phone is in phone mode:

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

When the phone is in custom mode:

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

Note

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

Group Listening

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

Dependencies: *Type (Key Event)*

Key Type (Group Listening)

Usage:

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




Zero Touch

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

Dependencies: *Type (Key Event)*


Key Type (Zero Touch)

Usage:


1. Press the DSS key to access the zero touch screen.
2. Press  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.

Keypad Lock

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

Dependencies: *Type (Keypad Lock)*

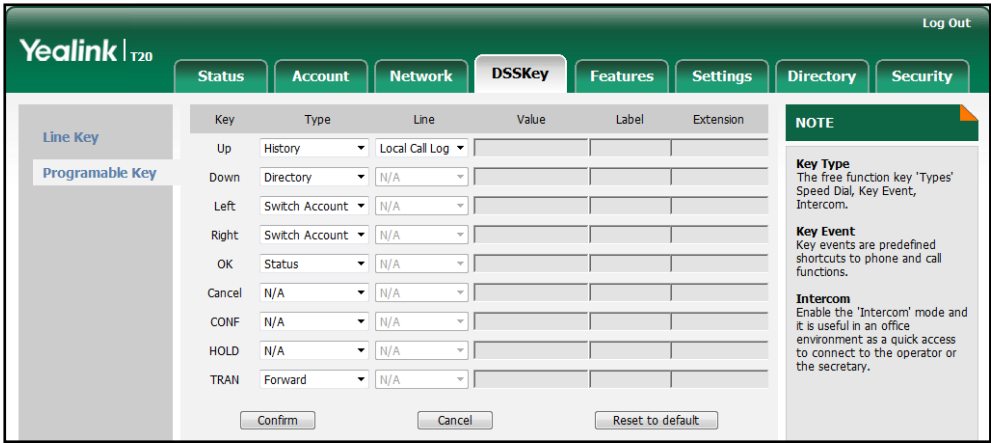
Usage: Press the DSS key to immediately lock the keypad of your phone instead of long pressing .

Programmable Keys

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

To customize the programmable keys via web user interface:

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



Key	Type	Line	Value	Label	Extension
Up	History	Local Call Log			
Down	Directory	N/A			
Left	Switch Account	N/A			
Right	Switch Account	N/A			
OK	Status	N/A			
Cancel	N/A	N/A			
CONF	N/A	N/A			
HOLD	N/A	N/A			
TRAN	Forward	N/A			

NOTE

Key Type
The free function key 'Types' Speed Dial, Key Event, Intercom.

Key Event
Key events are predefined shortcuts to phone and call functions.

Intercom
Enable the 'Intercom' mode and it is useful in an office environment as a quick access to connect to the operator or the secretary.

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

Note


You can configure the programmable keys via web user interface only.

Account Management

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


Account Registration

To register an account via phone user interface:



1. Press .
2. Select **Settings->Advanced->Accounts**.
3. Select the desired line.
4. Select **Enable** from the **Active Line** field.

1. Active Line:

◀ Enable ▶

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

To disable an account via phone user interface:

1. Press .
2. Select **Settings->Advanced ->Accounts**.
3. Select the desired line.
4. Select **Disable** from the **Active Line** field.
5. Press  to accept the change.

You can also register or disable an account via web user interface at the path **Account->Register**.

Multiple Line Keys per Account

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

Incoming calls to this line will be distributed evenly among the available line keys. Similarly, outgoing calls will be distributed.

Your phone can be configured to have a combination of lines with a line key and lines

with two line keys.

Dial Plan

Dial plan is a string of characters that governs the way your SIP-T20P IP phone processes the inputs received from your phone keypad. The SIP-T20P IP phone supports the following dial plan features:

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

The basic expression syntax you need to know:

.	The dot "." can be used as a placeholder or multiple placeholders for any character. Example: "12." would match "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: "9([5-7]) (.)", Replace: "5\$2". When you enter "96123" to dial out on your phone, the number will be replaced as "5123" and then dialed out. "\$2" means the characters in the second parenthesis, that is, "123".

Replace Rule

You can configure one or more replace rules to remove the specified string and replace it with another string. You can configure a pattern with wildcards (refer to the expression syntax in the table above), so that any string that matches the pattern will be replaced. This feature is convenient for you to dial out a long number. For example, a replace rule is configured as "Prefix: 1" and "Replace: 1234567", when you try to dial out the number "1234567", you just need to enter "1" on the phone and then press the OK key to dial

out.

To add a replace rule via web user interface:

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Enter the string (e.g., 1) in the **Prefix** field.
3. Enter the string (e.g., 1234) in the **Replace** field.
4. Enter the desired line ID in the **Account** field or leave it blank.

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

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

Note

The valid values of **Account** field can be one or more digits between 1 and 2. Each digit must be separated by a “,”. 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).

- Click **Del** to delete the replace rule(s).

Note

You can configure the replace rule via web user interface only.

Dial-now

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

To add dial-now rule via web user interface:

- Click on **Settings->Dial Plan->Dial now**.
- Enter the desired value (e.g., 1234) in the **Rule** field.
- Enter the desired line ID in the **Account** field or leave it blank.

For more information on the valid values of **Account** field, refer to [Replace Rule](#) on page 44.

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

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

Note

You can also edit or delete the dial-now rule, refer to [Replace Rule](#) on page 44 for more information.

You can configure the dial-now rule via web user interface only.

Delay Time for Dial-now Rule

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

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

1. Click on **Features->General Information**.
2. Enter the desired time within 1-14 (in seconds) in the **Time-Out For Dial-Now Rule** field.

The screenshot shows the Yealink T20 web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. The 'Time-Out For Dial-Now Rule' field is set to 1. A 'NOTE' section on the right contains the following information:

- Call Waiting**: This call feature allows your phone to accept other incoming calls during the conversation.
- Key As Send**: Select * or # as the send key.
- Hotline Number**: When you pick up the phone, it will dial out the hotline number automatically.

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

Note You can configure the delay time for dial-now rule via web user interface only.

Area Code

Area codes are also known as Numbering Plan Areas (NPAs). They usually indicate geographical areas in a country. This feature is necessary only when dialing the number outside the code area. For example, area code is configured as "code: 0592, Min Length: 4, Maxi Length: 11", then when you dial out the number "56789", which has the digits between 4 to 11, the phone will add the area code and dial out the number "059256789".

To configure the area code via web user interface:

1. Click on **Settings->Dial Plan->Area Code**.
2. Enter the desired values in the **Code**, **Min Length (1-15)** and **Max Length (1-15)** fields.
3. Enter the desired line ID in the **Account** field or leave it blank.

For more information on the valid values of **Account** field, refer to [Replace Rule](#) on page 44.

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

Note

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

You can configure the area code via web user interface only.

Block Out

You can block some 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".

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 of **Account** field, refer to [Replace Rule](#) on page 44.

Yealink | T20 Log Out

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

Replace Rule **Dial-now** **Area Code** **Block Out**

BlockOut Number1	4321	Account	1
BlockOut Number2		Account	
BlockOut Number3		Account	
BlockOut Number4		Account	
BlockOut Number5		Account	
BlockOut Number6		Account	
BlockOut Number7		Account	
BlockOut Number8		Account	
BlockOut Number9		Account	
BlockOut Number10		Account	

NOTE

Digit 0-9 *
Identifies a specific digit (do not use # if it is defined as send key).

[digit-digit]
Identifies any digit dialed that is included in the range.

[digit-digit,digit]
Specifies a range as a comma separated list.

x
Matches any single digit/character which is dialed.

.
Matches an arbitrary number of digits.

Confirm **Cancel**

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

Note

You can configure a block out number 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 required. 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:

- Click on **Features->Phone Lock**.
- Enter the emergency services number in the **Emergency** field.

For multiple numbers, enter a “,” between each emergency number. The default emergency numbers are 112, 911 and 110.

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

Note

You can configure emergency number via web user interface only.

Live Dialpad

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

To enable the live dialpad feature 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.

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

Note


The default delay time of live dialpad is 4s.

You can configure live dialpad via web user interface only.

Hot Line

You can dial a hotline number immediately once you lift the handset, press the **Speakerphone** key or press a line key. You can also configure a period of delay time to dial out the hotline number. The phone will dial out the hotline number automatically after the specified delay time.


To configure the hot line number via phone user interface:

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

The valid values range from 0 to 10.

1. Hot Number:

123

5. Press  to accept the change.

You can also configure hot line via web user interface at the path **Features->General Information**.

Basic Call Features

The SIP-T20P IP phone is designed to be easily used like a regular phone on a public switched telephone network (PSTN). You can place calls, answer calls, transfer a call to someone else, or conduct a conference call.

This chapter provides basic operating instructions for the SIP-T20P IP phone. The topics include:

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

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

Placing Calls

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

- Using the handset
- Using the speakerphone

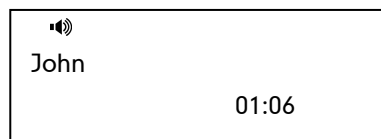
- Using the headset

You can also dial the number first, and then choose the way you want to speak to the other party.



You can search the call history, the local contact directory and dial from the search results. For more information, refer to [Contact Management](#) on page 26 and [Call History Management](#) on page 32.

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

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



To place a call using the handset:

1. Pick up the handset.
2. Enter the desired number using the keypad.
3. Press  or .

By default, the **#** key is configured as send. You can set the ***** key as send key or set neither of them as send keys. For more information, refer to [Key as Send](#) on page 20.







Note

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

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







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

Do one of the following:

- With the handset on-hook, press  or the line key to obtain a dial tone.
Enter the desired number using the keypad.
Press  or .
- 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 enable your headset, refer to [Headset Prior](#) on page 35.

To place multiple calls:

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

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

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

Note

You can ignore incoming calls by pressing the **X** key. You can also activate Do Not Disturb mode to ignore all incoming calls without ring on your phone. For more information, refer to [Do Not Disturb \(DND\)](#) on page 61.

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

Answering When Not in Another Call



In all cases, the active call will appear on the LCD screen showing call duration and destination.

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  .
- 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 the auto answer feature to automatically answer an incoming call on a line. Auto answer is configurable on a per-line basis.

To configure auto answer via phone user interface:

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

13. Auto Answer:

◀ Enable ▶

5. Press  to accept the change.
The **AA** icon appears on the LCD screen.

AA

1234
14 Jun 03:16

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




Note

The auto answer feature is only applicable when there is no other call in progress on the phone.

Auto Redial


You can enable the auto redial feature to redial the phone number automatically when the called party is busy. You can also configure the times of the auto redial 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 **Enable** from the **Auto Redial** field.

1. Auto Redial:

◀ Enable ▶


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

You can also configure auto redial 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:

Auto Redial?

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

Redialing
111 10s

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

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

Call Completion




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

To configure call completion via phone user interface:

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

1. Call Completion:

◀ Enable ▶


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

You can also configure call completion 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

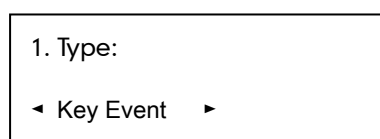
The call completion feature is not available on all servers. For more information, contact your system administrator.


Call Return

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

To configure a call return key via phone user interface:

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




6. Press  to accept the change.

You can also configure a call return key via web user interface at the path **DSSKey->Line Key**.

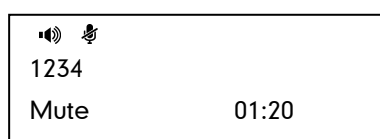
Call Mute

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

To mute a call:

1. Press  during an active call.

The phone LCD screen indicates that the call is on mute.



To un-mute a call:

1. Press  again to un-mute a call.

Call Hold/Resume

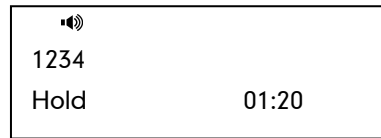
You can place an active call on hold. At any time, at most one active call can be in progress on your phone, other calls can be received and made while placing the original call on hold. When placing a call on hold, your IP PBX might play a melody to

the other party while waiting.

To place a call on hold:

1. Press  during a call.

The phone LCD screen shows 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, press  or  to switch between the calls, and then press  to retrieve the desired call.

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

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

Do Not Disturb (DND)

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

You can enable/disable the DND feature for the phone system, or you can customize the DND feature for each account. The following describes the DND modes:

- **Phone** (default): DND in phone mode means that the DND feature is effective for the phone system.
- **Custom**: DND in custom mode means that you can configure the DND feature for each account or all accounts.

You can also configure the phone to receive incoming calls from authorized numbers when the DND feature is enabled.

To configure the DND mode via web user interface:

1. Click on **Features->Forward & DND**.




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

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

Note


You can configure the DND mode via web user interface only.

To activate DND in phone mode:

- Press .
- Select **Features->DND Code**.
- Press  or  to select **Phone Mode** from the **User Name** field.
- Select **Enable** from the **DND** field.

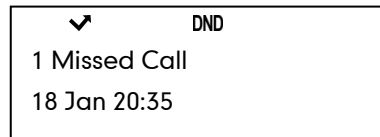
2. DND:

◀ Enable ▶




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

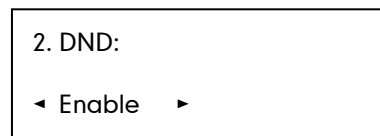
The **DND** icon on the idle screen indicates that the DND feature is enabled.


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



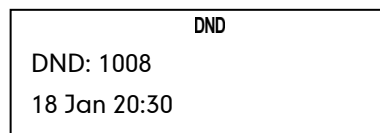
To activate DND in custom mode:

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

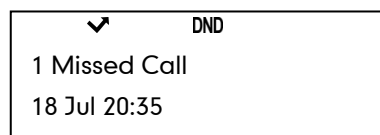


5. (Optional.) Enter the DND on code or off code respectively in the **DND On Code** or **DND Off Code** field.
6. Press  to accept the change.

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



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



Note

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

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

To configure the DND authorized numbers via web user interface:

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 "," between each number.
4. Click **Confirm** to accept the change.
When the DND feature is enabled on the phone, the phone can still receive incoming calls from the numbers specified in the **DND Authorized Numbers** field.

Note

You can configure the DND authorized numbers via web user interface only.

Call Forward

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

Static Forwarding

You can configure three types of call forward:

- **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 the call forward feature for the phone system, or you can customize the call forward feature for each account or all accounts. The following describes the call forward modes:

- **Phone** (default): Call forward in phone mode means that the call forward feature is effective for the phone system.
- **Custom:** Call forward in custom mode means that you can configure the call forward feature for each account.

To configure the call forward mode via web user interface:

1. Click on **Features->Forward & DND**.







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

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

Note

You can configure the call forward mode via web user interface only.



To enable call forward in phone mode:

- Press .
- Select **Call Control->Call Forwarding**.
- Press  or  to select the forwarding type, and then press .
- Depending on your selection:
 - If you select **Always**:
 - Press  or  to select **Enable** from the **Always** field.
 - Enter the destination number you want to forward all incoming calls to in the **Forward To** field.
 - (Optional.) Enter the always forward on code or off code respectively in the **On Code** or **Off Code** field.

1. Always:

◀ Enable ▶

- If you select **Busy**:

- Press  or  to select **Enable** from the **Busy** field.
- 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 or off code respectively in the **On Code** or **Off Code** field.

1. Busy:

◀ Enable ▶

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


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


The default ring time is 12 seconds.

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




1. No Answer:

◀ Enable ▶

5. Press  to accept the change.

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

To enable call forward in custom mode:

1. Press .
2. Select **Call Control->Call Forwarding**.
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**:

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

1. Always:

◀ Enable ▶

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

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

1. Busy:


◀ Enable ▶


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


- 1) Press ◀ or ▶ to select **Enable** from the **No Answer** field.
- 2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward To** field.
- 3) Press ◀ or ▶ to select the ring time to wait before forwarding in the **After Ring Time** field.
The default ring time is 12 seconds.
- 4) (Optional.) Enter the no answer forward on code or off code respectively in the **On Code** or **Off Code** field.

1. No Answer:

◀ Enable ▶

6. Press  to accept the change.

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



FWD: 1003
24 Jul 10:39

You can also configure the call forward feature via web user interface at the path **Features->Forward & DND**.









Note

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

The call forward feature 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 **Call Control->Call Forwarding**.
Press  or  to select the forwarding type, then press .
Press  or  to select **Disable** to disable the call forward.
Press  to accept the change.


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

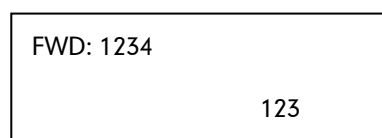
Do one of the following:

1. Press , select **Call Control->Call Forwarding** 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 **Disable** 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.

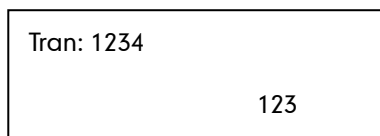
Call Transfer


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

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





To perform a blind transfer:

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







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






To configure call waiting via phone user interface:

1. Press .

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

1. Call Waiting:

◀ *Enable ▶

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

You can also configure the call waiting feature via web user interface at the path **Features->General Information**.

Conference

You can create a conference with other parties using the phone's local conference feature. 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. The network conference feature allows you to add specific conference parties.




Note

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

Local Conference

The SIP-T20P IP phone supports up to 3 parties (including yourself) in a conference call. This is the default method of conference called Local Conference.

To set up a local conference call:


1. Place a call to the first party.
2. When the first party answers the call, press  to place a new call.
The active call is placed on hold.
3. Enter the number of second party and press  or .

Dial: 1234




123



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 make sure the call is active (for example, select the call on account 1).
3. Press  to join the two calls in the conference on account 1.

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

Network Conference

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

This feature allows you to perform the following:

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

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

To configure the network conference feature 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**.

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









The screenshot shows the Yealink T20 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Account' dropdown is set to 'Account 1'. The 'Conference URI' field is filled with 'conference@example.com'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5060), 'Conference Type' (Network Conference), 'ACD Subscrip Period(120~3600s)' (3600), 'Early Media' (Disabled), 'SIP Server Type' (Default), 'Music Server URI', 'Directed Call Pickup Code', 'Group Call Pickup Code', 'Distinctive Ring Tones' (Disabled), 'Unregister When Reboot' (Disabled), and 'Out Dialog BLF' (Disabled). A 'NOTE' box on the right says 'Advanced: The Advanced parameters for administrator.' At the bottom are 'Confirm' and 'Cancel' buttons.

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

Note

You can configure network conference via web user interface only.

To set up a network conference call:

- Place a call to the first party.
- Press  to create a new call.
The active call is placed on hold.
- Enter the second party's number and press  or .
- When the second party answers the call, press  to add the second party to the conference.
- Press  to create a new call.
The conference is placed on hold.
- Enter the number of the new party and then press  or .
- When the new party answers the call, press  to add the new party to the conference.
- Repeat steps 5 to 7 until you have added all intended parties.

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








Call Park

You can use call park feature to place a call on hold, and then retrieve the call from another phone in the system (for example, a phone in another office or conference room). You can park 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 indication on the LCD screen.

Note


The call park feature 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 specific line from the **Account ID** field.
7. Enter the call park feature code in the **Value** field.

1. Type:

◀ Key Event ▶

8. Press  to accept the change.

You can also configure a call park key via web user interface at the path **DSSKey->Line Key**.

To use the call park feature:

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

The system establishes call between phone C and B.

Note

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

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

Call Pickup








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

Note

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


Directed Call Pickup

To configure a directed pickup key via phone user interface:

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

1. Type:

◀ Key Event ▶

8. Press  to accept the change.

You can also configure a directed pickup key via web user interface at the path **DSSKey->Line Key**.

To pick up a call directly:








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

incoming call.

The incoming call is answered on your phone.


Group Call Pickup

To configure a group pickup key via phone user interface:

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

1. Type:

◀ Key Event ▶

8. Press  to accept the change.

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

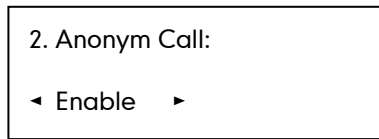
Note

The anonymous call feature 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 .
2. Select **Features->Anonym Call**.

3. Press ◀ or ▶ to select the desired line from the **Line ID** field.
4. Press ◀ or ▶ to select **Enable** from the **Anonym Call** field.

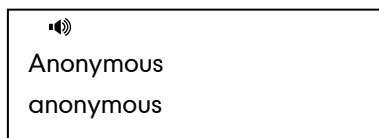


5. (Optional.) Enter the anonymous call on code in the **Call On Code** field.
6. (Optional.) Enter the anonymous call off code in the **Call Off Code** field.
7. Press OK to accept the change.

You can also configure anonymous call 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.

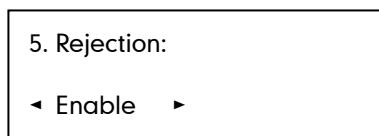


Anonymous Call Rejection


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

To configure anonymous call rejection via phone user interface:

1. Press  .
2. Select **Features->Anonym Call**.
3. Press ◀ or ▶ to select the desired line from the **Line ID** field.
4. Press ▲ or ▼ to scroll to the **Rejection** field.
5. Press ◀ or ▶ to select **Enable** from the **Rejection** field.



6. (Optional.) Enter the anonymous call rejection on code in the **Reject On Code** field.

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

You can also configure anonymous call rejection via web user interface at the path **Account->Basic**.

Advanced Phone Features

This chapter provides operating instructions for the advanced features of the SIP-T20P IP phone. The topics include:

- [Busy Lamp Field \(BLF\)](#)
- [Call Recording](#)
- [Hot Desking](#)
- [Intercom](#)
- [Multicast Paging](#)
- [Music on Hold](#)
- [Automatic Call Distribution \(ACD\)](#)
- [Voice Mail](#)
- [Message Waiting Indicator \(MWI\)](#)






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

Busy Lamp Field (BLF)

You can use the BLF feature to monitor a specific user for status changes on the phone. For example, you can configure a BLF key on the phone for monitoring the status of a friend's phone (busy or idle). When the friend makes a call, the BLF key LED illuminates flashing green on your phone to indicate that the friend's phone is in use and busy. For more 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 phone is idle. You can receive an audio alert when the monitored phone number receives an incoming call. You can also pick up a call directly by pressing the BLF key when the monitored phone number receives an incoming call. 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 line 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.

1. Type:

◀ BLF ▶

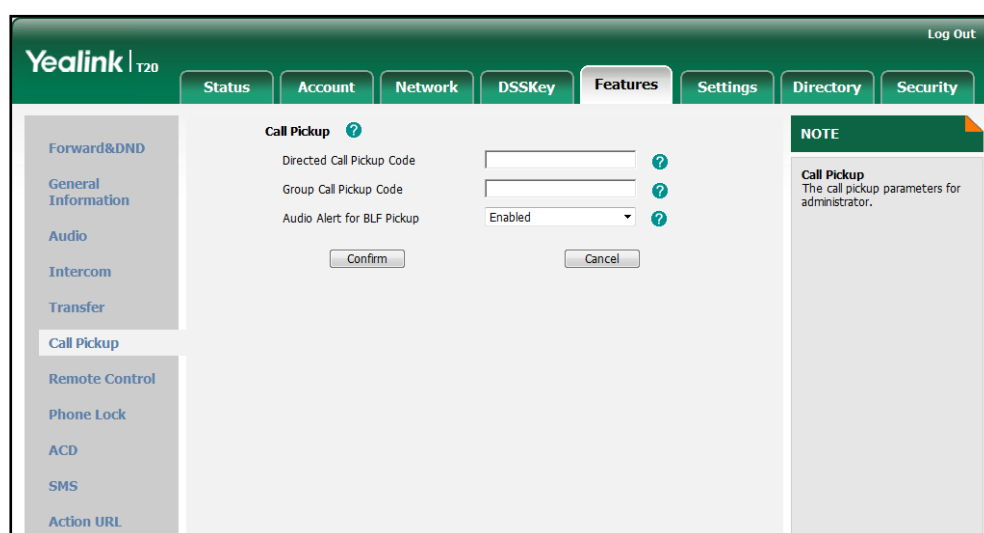
8. Press  to accept the change.

You can also configure the BLF key via web user interface at the path **DSSKey->Line Key**.

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

To enable the visual and 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 You can enable the audio alert feature via web user interface only.

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

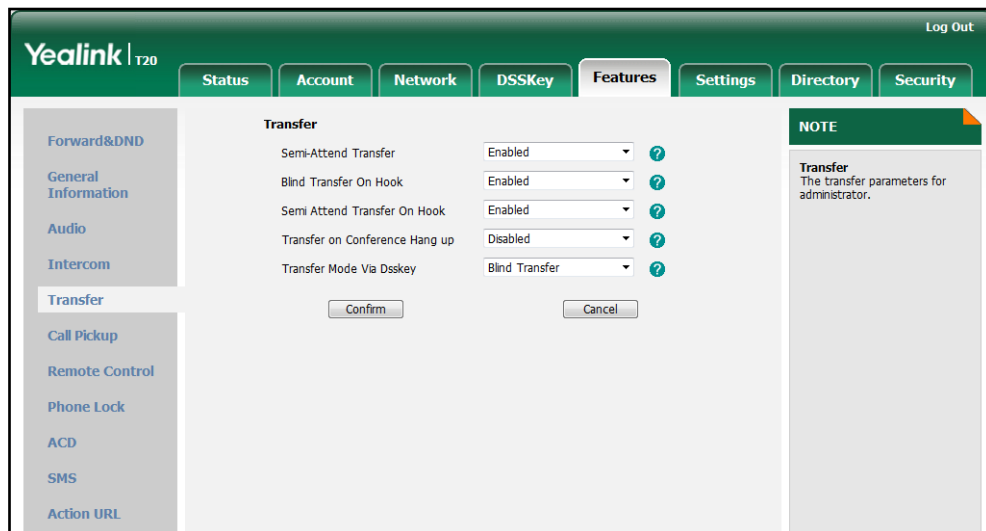
- The phone plays a warning tone (if enabled).
- The BLF key LED flashes.

When there is an active call on the IP phone, you can transfer the active call to the monitored phone number directly by pressing the BLF key. The phone transfers the

active call differently depending on the transfer mode on DSS key. For more information on performing call transfer, refer to [Call Transfer](#) on page 68.

To configure the transfer mode on DSS key via web user interface:

1. Click on **Features->Transfer**.
2. Select the desired transfer mode from the pull-down list of **Transfer Mode Via Dsskey**. Depending on your selection:
 - If you select **Blind Transfer**, press the BLF key to complete the blind transfer to the monitored phone number.
 - If you select **Attended Transfer**, press the BLF key to dial out the monitored phone number and then you can complete the attended or semi-attended transfer.
 - If you select **New Call**, press the BLF key to place a new call to the monitored user.



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

Note You can configure the transfer mode on DSS key via web user interface only.

Call Recording

You can record calls by pressing a record key on the SIP-T20P IP phone. There are 2 different ways of configuring call recording and they even work differently:






- **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


The record feature 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.

2. Key Type:

◀ Record ▶


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.

1. Type:

◀ URL Record ▶

6. Press  to accept the change.

You can also configure the record key or URL record key via web user interface at the path **DSSKey->Line Key**.

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

- During an active call
- When calls are on hold or mute
- During a blind or attended transfer
- During a conference call

- When the phone prompts you to answer an incoming call

The record and URL record key is not available:

- When there are no connected calls on your phone
- When 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 flashes green.
2. Press the record key or URL record key again to stop recording.
The record or URL record key LED turns off.

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

Note

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

Hot Desking




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

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

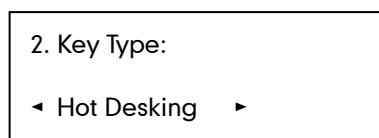
Note

The hot desking feature 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.

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



2. Key Type:

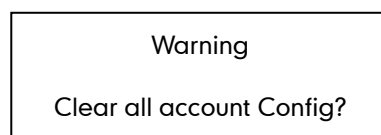
◀ Hot Desking ▶

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

You can also configure a hot desking key via web user interface at the path **DSSKey**.

To use hot desking:

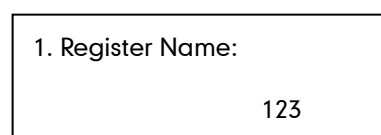
1. Press the hot desking key when the phone is idle.
The LCD screen prompts “Clear all account config?”.



Warning

Clear all account Config?

2. Press **OK** , registration configurations of all accounts will be cleared immediately.
The login wizard will be shown as below:



1. Register Name:

123

3. Enter the login information in each field.

Intercom




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

Note

The intercom feature 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 line from the **Account ID** field.
6. Enter the remote extension number in the **Value** field.


1. Type:

◀ Intercom ▶

7. Press  to accept the change.

You can also configure a DSS key as intercom 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

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

Intercom features you need to know:

Intercom features	Description
Intercom Allow	Allows you to enable or disable the IP phone to automatically answer an incoming intercom call.
Intercom Mute	Allows you to enable or disable the microphone on the IP phone for Intercom calls.
Intercom Tone	Allows you to enable or disable the IP phone to play a warning tone when the IP phone receives an


	incoming intercom call.
Intercom Barge	Allows you to enable or disable the IP phone to automatically answer an incoming intercom call while there is already an active call on the phone.

To configure intercom features via phone user interface:

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

1. Intercom Allow:

◀ *Enable ▶

4. Press  to accept the change.

You can also configure these specific parameters via web user interface at the path **Features->Intercom**.

Intercom Allow

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

Note

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

Intercom Mute

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

Intercom Tone

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

Intercom Barge






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

Multicast Paging

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

1. Type:

◀ Key Event ▶

7. Press  to accept the change.

You can also configure a multicast paging key via web user interface at the path **DSSKey->Line Key**.

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

To configure a default codec for multicast paging:

1. Click on **Features->General Information**.
2. Select the desired codec from the pull-down list of **Multicast Codec**.

The default codec is G722.

The screenshot shows the Yealink T20 web interface. The 'Features' tab is selected, and the 'General Information' sub-tab is active. On the left, a sidebar lists various settings categories: Forward&DND, General Information, Audio, Intercom, Transfer, Call Pickup, Remote Control, Phone Lock, ACD, SMS, and Action URL. The main content area displays settings for General Information. The 'Multicast Codec' is set to 'G722'. Other settings 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), Play Hold Tone (Enabled), Play Hold Tone Delay (30), Allow Mute (Enabled), Call List Show Number (Disabled), Voice Mail Tone (Enable), DHCP Hostname (SIP-T20P), and Reboot In Talking (Disabled). A 'NOTE' box on the right contains information about Call Waiting, Key As Send, and Hotline Number. At the bottom, there are 'Confirm' and 'Cancel' buttons.

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

Note

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

You can configure a default codec for multicast paging via web user interface only.

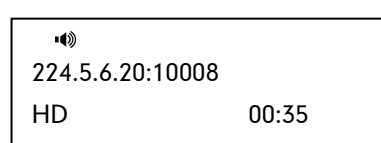
To send RTP stream:



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

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

The multicast paging key LED illuminates solid green.

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



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

Receiving RTP Stream

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

You can also change the behavior of how the phone handles incoming multicast paging calls by configuring specific parameters via web user interface. The specific parameters are: Paging Barge and Paging Priority Active.

Paging Barge

You can use the paging barge feature to define 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, then it will be ignored automatically. If Disabled is selected from the pull-down list of Paging Barge, the voice call in progress shall take precedence over all incoming multicast paging calls. The valid values in the Paging Barge field are:

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

Paging Priority Active

You can enable or disable this feature to decide 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 the incoming multicast paging call with a lower priority, otherwise, the phone will play the incoming multicast RTP 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 for incoming RTP multicast in the **Listening Address** field.

Yealink | T20

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		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.

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

Note

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

Both the multicast paging sender and the receiver will play a warning tone when pressing the multicast paging key.

You can configure multicast listening addresses 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 callers who have been placed on hold. To use this feature, you should specify a SIP URI pointing to a Music on Hold Server account, when placing a call on hold, the phone will invite this SIP URI to the Music on Hold Server account. The Music on Hold account automatically answers to the SIP INVITE messages and immediately plays audio from some source located anywhere (LAN, Internet) to the held party. Contact your system administrator for the SIP URI.

To configure music on hold server via web user interface:

- Click on **Account**.
- 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 T20 web interface with the 'Account' tab selected. The 'Music Server URI' field is set to '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

All involved parties cannot use encrypted RTP.

You can configure the music on hold server via web user interface only.

Automatic Call Distribution (ACD)

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

Note

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


To configure an ACD key via phone user interface:

- Press .
- Select **Features->DSS Keys**.
- Select the desired DSS key.

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

1. Type:

◀ ACD ▶

5. Press  to accept the change.

You can also configure an ACD key via web user interface at the path **DSSKey->Line Key**.

To log in the ACD system:

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

The phone LCD screen prompts you the following information:

User ID: the identity used to log in the queue.

Password: the password used to log in the queue.


1. User ID:

123

2. Press  to log in.

Note

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

After configuring an ACD key, you can press the ACD key to log in the ACD system. After logging in, you are ready to receive calls from the ACD system. You can press the ACD key to show your current phone status. You can press the OK key to set your phone status to **Available /Unavailable**. The system server monitors your phone status. When you set the phone status to available, the ACD key LED illuminates solid green, and then the server begins distributing calls to your phone. When you set the phone to unavailable, the ACD key LED illuminates flashing green, and then the server temporarily stops distributing calls to your phone. To log out the ACD system, press .

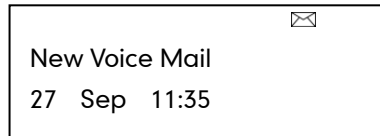
Note

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

Voice Mail

You can leave voice mails for someone else on the SIP-T20P IP phone. You can also listen to the voice mails stored in a centralized location. When receiving a new voice mail, the

phone will play a warning tone and the **MESSAGE** key LED will illuminate and the phone LCD screen will appear a flashing icon.



Note

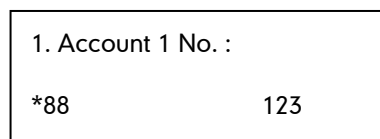
The voice mail feature is not available on all servers. Contact your system administrator for more information.





To leave a voice mail:

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

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

To listen to the voice mails:

1. When the phone user interface prompts receiving new voice mails, press  or  to dial out the voice mail access code.
2. Follow the voice prompt to listen to the 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 turn off.

To view the voice mail via phone user interface:

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

The phone LCD screen displays the amount of voice mails that includes new or old voice mails.

1. 1234

2 new 1 old Mail

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

Message Waiting Indicator (MWI)

The SIP-T20P IP phone supports MWI feature 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, the **MESSAGE** key LED illuminates solid green and an indicator message (including a voice mail icon) appears on the LCD screen. This is cleared only when you retrieve all voice mails or delete them.

For some particular servers, the MWI service is unsolicited. So the SIP-T20P IP phone just needs to handle the MWI messages sent from the server. But for some servers, the MWI service is solicited. In this case, the SIP-T20P IP phone must enable MWI Subscription for MWI messages.

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 re-sends a MWI subscription before expiring.
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 the subscribe for MWI feature and the voice mail number in advance.

Note

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

To enable MWI subscription via web user interface:

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

The screenshot shows the Yealink T20 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Account' dropdown menu shows 'Account 1'. The 'Subscribe for MWI' dropdown is set to 'Enabled', and the 'MWI Subscription Period (Seconds)' field is set to 3600. Other settings like 'Keep Alive Type', 'Local SIP Port', 'RPort', and 'DTMF Type' are also visible. A 'NOTE' section 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	Disabled
Voice Mail	
Caller ID Source	FROM
Session Timer	Disabled
Session Expires(30~7200s)	1800
Session Refresher	UAC
Send user=phone	Disabled

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

To enable the Subscribe MWI to Voice Mail feature 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 T20 web interface with the 'Account' tab selected. The 'Voice Mail' field is set to 1234. The 'Subscribe MWI To Voice Mail' field is set to Enabled. The 'Voice Mail' field is highlighted with a red box.

Field	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 WM feature.

Note

You can configure the MWI subscription via web user interface only.

Troubleshooting

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

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

Why is the phone LCD screen blank?

- Ensure that the phone is properly plugged into a functional AC outlet.
- Ensure that the phone isn't plugged into a plug controlled by a switch that is off.
- If the phone is plugged into a power strip, try plugging it directly into a wall outlet instead.
- If the phone is powered from PoE, ensure you use a PoE compliant switch or hub.

Why does the phone display "Network Unavailable"?

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

Why does the phone display "No Service"?

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

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

Check if you have configured the phone to obtain the time and date from the SNTP server automatically. If the phone fails to connect the SNTP server, 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 14.

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

There are three ways to obtain the MAC address of a phone:

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

Why can't I get a dial tone?

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

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

Why doesn't the phone ring?

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

Why can't I receive calls?

- Check that the SIP registration with your system administrator.
- Check that the DND (Do Not Disturb) mode is turned off on the phone. Refer to [Do Not Disturb \(DND\)](#) on page 61.
- Check that the call forward feature is disabled on the phone. Refer to [Call Forward](#) on page 64.
- Check whether the caller number is stored in the blacklist directory. Refer to [Blacklist](#) on page 32.

Why is my handset not working?

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

Why is my headset not working?

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

[Activation/Deactivation](#) on page 35.

- Check that the headset volume is adjusted to an appropriate level. Refer to [Volume](#) on page 23.

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

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

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

Both user name and register name are defined by the server. A user name is used to identify the account while a register name matched with a password is used for authentication if the server requires. Display name is the caller ID that will be displayed on the callee's phone 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 is on hold, the phone will play a hold tone every 30 seconds. The call hold tone feature is enabled by default. You can disable it or change the interval to play a hold tone via web user interface only.

To configure the call hold tone and call 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.

The screenshot shows the Yealink T20 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features' (selected), 'Settings', 'Directory', and 'Security'. The left sidebar lists various features: Forward&DND, General Information (selected), Audio, Intercom, Transfer, Call Pickup, Remote Control, Phone Lock, ACD, SMS, and Action URL. The main content area is titled 'General Information' and contains a list of settings:

Setting	Value	Help
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	?
Multicast Codec	G722	?
Play Hold Tone	Enabled	?
Play Hold Tone Delay	30	?
Allow Mute	Enabled	?
Call List Show Number	Disabled	?
Voice Mail Tone	Enable	?
DHCP Hostname	SIP-T20P	?
Reboot In Talking	Disabled	?

At the bottom of the settings list are 'Confirm' and 'Cancel' buttons. On the right side, there is a 'NOTE' section with the following text:

Call Waiting
This call feature allows your phone to accept other incoming calls during the conversation.

Key As Send
Select * or # as the send key.

Hotline Number
When you pick up the phone, it will dial out the hotline number automatically.

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

How to change the user password?

To change the user password via web user interface:

1. Click on **Security->Password**.
2. Select **user** from the pull-down list of **User Type**.
3. Enter the new user password in the **New Password** and **Confirm Password** fields.

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

Note

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

You can change the user password via web user interface only.

How to make a call using SRTP?

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

To enable SRTP via web user interface:

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

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

The screenshot shows the Yealink T20 web interface with the 'Account' tab selected. The 'RTP Encryption(SRTP)' dropdown is set to 'Forbin SRTP'. Other visible settings include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5060), 'RPort' (Disabled), and 'SIP Session Timer T1 (0.5~10s)' (0.5). There are also fields for 'Directed Call Pickup Code', 'Group Call Pickup Code', 'Distinctive Ring Tones', 'Unregister When Reboot', and 'Out Dialog BLF', all currently set to 'Disabled'. The interface includes a 'Confirm' button and a 'Cancel' button at the bottom.

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

Note

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

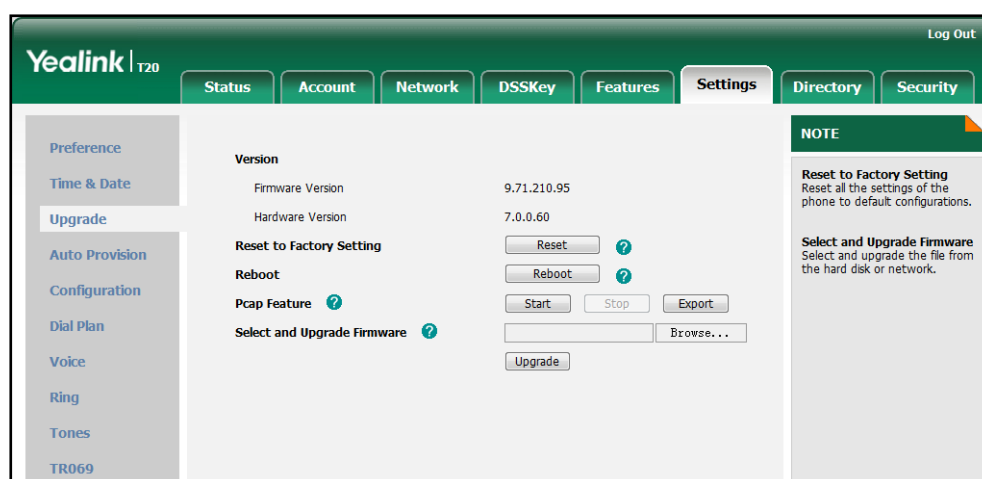
You can configure the SRTP via web user interface only.

How to reboot the phone?

To reboot the phone via web user interface:

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

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



Note

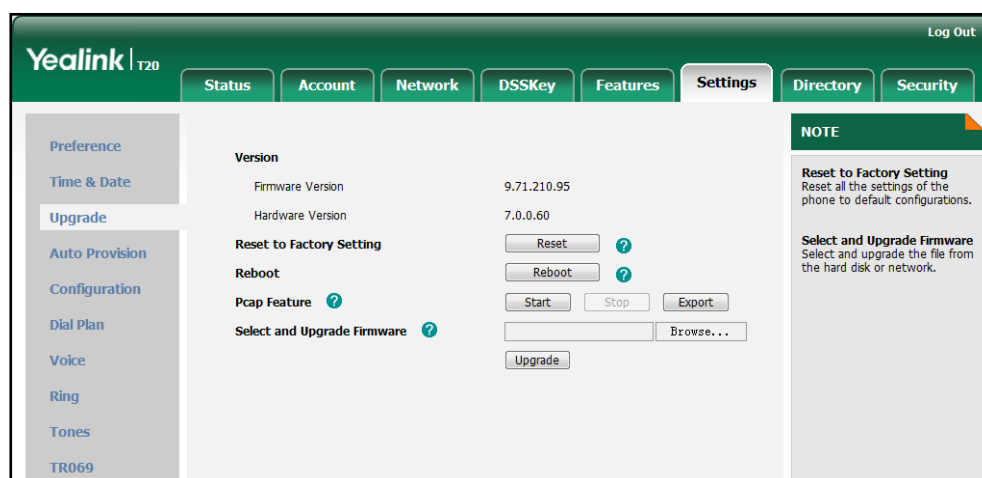
You can also long press the **X** key to reboot the phone directly. Any reboot of the phone may take a few minutes.

How to export PCAP trace?

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

To export a PCAP trace via web user interface:

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



How to export system log?

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

To export a system log via web user interface:

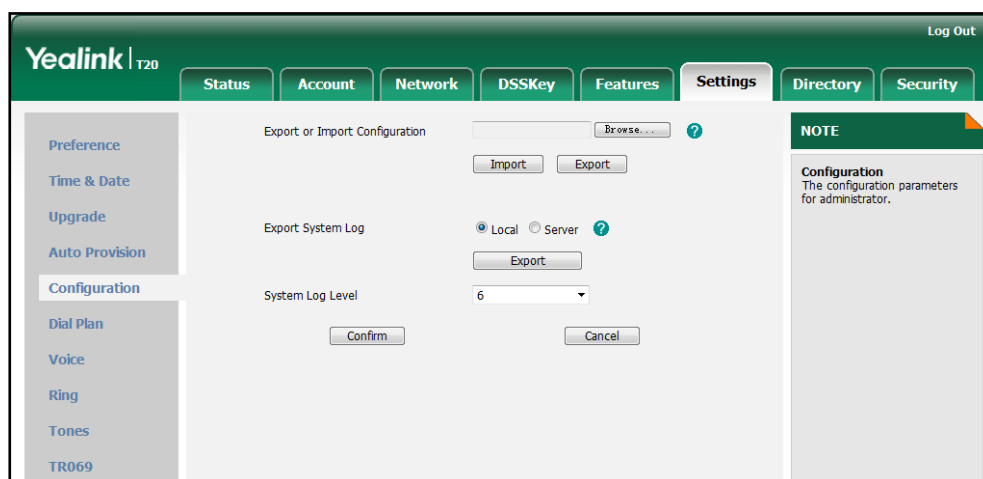
1. Click on **Settings->Configuration**.
2. Select **6** from the pull-down list of **System Log Level**.
3. Click **Confirm** to accept the change.

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

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

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

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



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

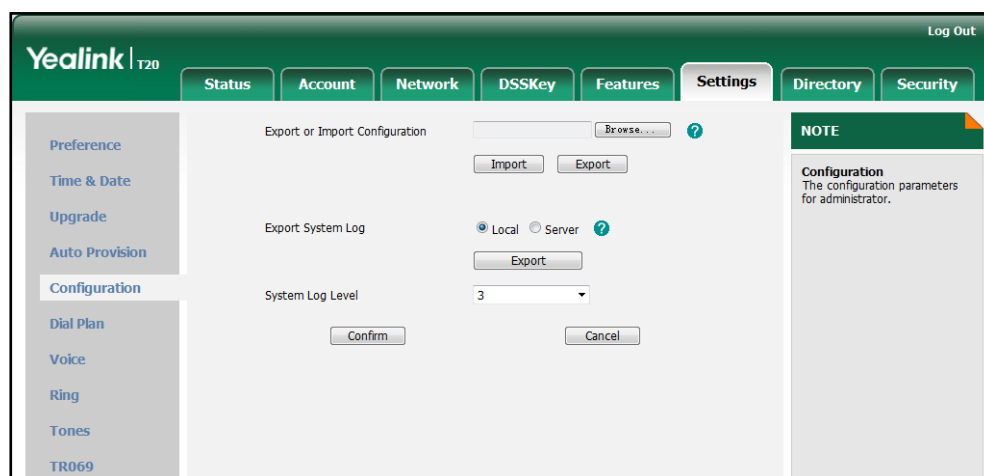
How to export/ import phone configurations?

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

To export the phone configurations via web user interface:

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

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



To import the phone configurations via web user interface:

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

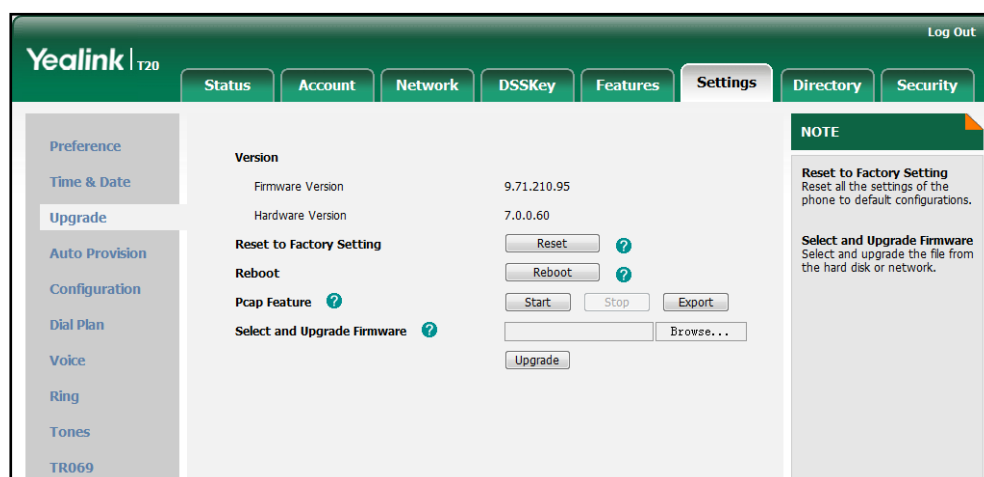
Note

The file format of configuration file must be “.bin”.

How to upgrade firmware?

To upgrade firmware via web user interface:

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



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





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

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

How to reset the phone?

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

To reset the phone via phone user interface:

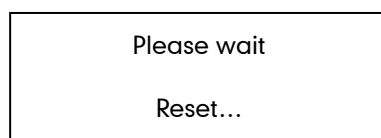
1. Press  .
2. Select **Settings->Advanced** (password: admin).
3. Press  or  to scroll to **Reset Factory**, and then press  .

The LCD screen prompts "Reset To Factory?".

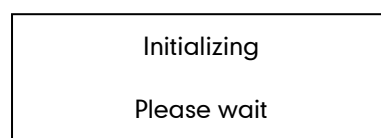


4. Press  to confirm.

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



The LCD screen prompts "Initializing Please wait".



The phone will be reset to factory successfully after startup.

Note

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

Regulatory Notices

Service Agreements

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

Limitations of Liability

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

Safety Instructions

Save these instructions. Read these safety instructions before use!

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

General Requirements

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

Environmental Requirements

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

Operating Requirements

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



Cleaning Requirements

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

Appendix A - Time Zones

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

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

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