



HD IP Conference Phone CP860 User Guide

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CE

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Statements of compliance can be obtained by contacting support@yealink.com.

CE Mark Warning

This device is marked with the CE mark in compliance with EC Directives 2014/35/EU and 2014/30/EU.

Part 15 FCC Rules

Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies 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.

Industry Canada (IC)

This Class [B] digital apparatus complies with Canadian ICES-003 Rules.

Class B Digital Device or Peripheral

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

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

WEEE Warning



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

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GNU GPL INFORMATION

Yealink CP860 IP conference phone firmware contains third-party software under the GNU General Public License (GPL). Yealink uses software under the specific terms of the GPL. Please refer to the GPL for the exact terms and conditions of the license.

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

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

About This Guide

Thank you for choosing the CP860 IP conference phone, a Yealink's first IP conference phone which is exquisitely designed for small and medium-sized conference room, meeting for 10 people below. Users can also benefit from optional expansion microphones for wider reception. This conference phone provides business telephony features, such as Call Hold, Call Transfer, Call and Conference Recording, Multicast Paging and five-way Conference over an IP network.

This guide provides everything you need to quickly use your new phone and expansion microphone. First, verify with your system administrator that the IP network is ready for phone configuration. Also be sure to read the Packaging Contents and Regulatory Notices sections in this guide before you set up and use the CP860 IP conference phone.

In This Guide

Topics provided in this guide include:

- Chapter 1 Overview
- Chapter 2 Getting Started
- Chapter 3 Customizing Your Phone
- Chapter 4 Basic Call Features
- Chapter 5 Advanced Phone Features

Summary of Changes

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

Changes for Release 81, Guide Version 81.10

The following sections are new:

- Connecting the Optional Expansion PSTN Box CPN10 on page 15
- Entering Data and Editing Fields on page 20
- Power Saving on page 23
- Key Tone on page 35
- Favorite Directory on page 46
- Yealink Cloud Account on page 59
- Call Park on page 90

- Idle Recording on page 102
- Using Your Phone with PSTN Account on page 131
- Appendix B Menu Structure on page 157
- Appendix C Unavailable Features for PSTN on page 158

Major updates have occurred to the following sections:

- Icon Instructions on page 3
- Setup Wizard on page 16
- Phone Status on page 17
- Language on page 27
- Time & Date on page 28
- Phone Lock on page 30
- Volume on page 32
- Local Directory on page 37
- Programable Keys on page 53
- Dial Plan on page 62
- Emergency Number on page 67
- Local Conference on page 87
- Call and Conference Recording on page 99
- Incoming Intercom Calls on page 105
- Using SCA Feature on the IP Phone on page 116
- Short Message Service (SMS) on page 123
- Voice Mail on page 125

Changes for Release 80, Guide Version 80.10

Major updates have occurred to the following sections:

- Icon Instructions on page 3
- Setup Wizard on page 16

Changes for Release 80, Guide Version 80.5

The following sections are new:

- Logo Customization on page 52
- Call Completion on page 77
- Shared Call Appearance (SCA) on page 113
- Bridged Line Appearance (BLA) on page 118
- Short Message Service (SMS) on page 123

Major updates have occurred to the following sections:

- Backlight on page 26
- Time & Date on page 28
- Phone Lock on page 30
- Ring Tones on page 33
- Directory on page 36
- Remote Phone Book on page 48
- Programable Keys on page 53
- Account Registration on page 58
- Auto Answer on page 75
- Do Not Disturb (DND) on page 81
- Call Forward on page 82
- Intercom on page 104
- Multicast Paging on page 106
- Appendix on page 155

Changes for Release 72, Guide Version 72.8

Major updates have occurred to the following section:

• Connecting the Optional Expansion Microphone CPE80 on page 12

Changes for Release 72, Guide Version 72.3

Major updates have occurred to the following sections:

- Local Directory on page 37
- Programable Keys on page 52
- Dial Plan on page 62

- Conference on page 87
- Intercom on page 104
- Troubleshooting on page 139

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Overview

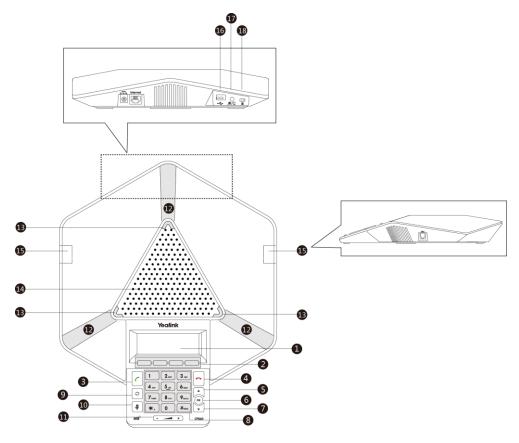
This chapter provides the overview of the CP860 IP conference phone. Topics include:

- Hardware Component Instructions
- Icon Instructions
- LED Instructions
- User Interfaces
- Documentations

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

Hardware Component Instructions

The following figure shows the primary hardware component of CP860 IP conference phone:

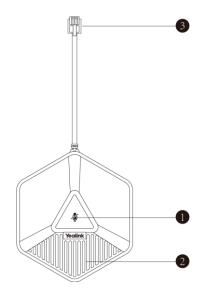


	Item	Description
		Shows information about calls, messages, soft keys, time, date
		and other relevant data:
		Call information-caller ID, call duration
(1)	LCD Screen	• Icons (for example, DND)
		Missed call text or second incoming caller information
		Prompt text (for example, "Saving config file!")
		Time and date
2	Soft Keys	Label automatically to identity their context-sensitive
	-	features.
3	Off-hook Key	Enters the dialing screen, places a call or answers a call.
4	On-hook Key	Ends a call or returns to the idle screen.
(5)		 Scrolls through the displayed information upwards.
		Views call history list when the phone is idle.
6	OK	Confirms actions or answers incoming calls.
(7)	M	Scrolls through the displayed information downwards.
		Views the directory list when the phone is idle.
8	Keypad	Provides the digits, letters, and special characters in
	, , , , , , , , , , , , , , , , , , ,	context-sensitive applications.
9	Redial Key	Redials a previously dialed number.
10	Mute Key	Toggles mute feature.
11	Volume Key	Adjusts the volume of the speaker, ringer or media.
(12)	Three Internal	Provide 10-feet (3-meters) and 360-degree coverage to
	Microphones	transmit sound to other phones.
(13)	LED Indicators	Indicate phone and call statuses.
14	Speaker	Provides hands-free (speakerphone) audio output.
(15)	Two MIC Darte	Allow you to connect two expansion microphones to your
10	Two MIC Ports	phone (optional).
		• Allows you to connect a USB flash drive (optional) to your
		phone so you can record calls/conferences and play back
		recorded files.
(16)	USB Port	• Allows you to connect expansion PSTN box(es) (optional) to
		experience calls in PSTN. Up to two cascaded expansion PSTN
		boxes can be connected, so you can experience the local
		three-way conference conveniently in excellent speech quality
		with PSTN.

Hardware component instructions of the CP860 IP conference phone are:

	Item	Description
17	PC/Mobile Port	Allows you to connect an optional PC or Mobile Device to your phone so you can connect the PC or mobile device audio to your phone.
18	Security Slot	Allows you to connect a universal security cable to your phone so you can lock down your phone. The phone will not be removed after locked.

The following figure shows the primary hardware component of CPE80 expansion microphone (optional):



Hardware component instructions of the CPE80 expansion microphone are:

	Item	Description
\bigcirc	Mute Indicator LED	Toggles and indicates mute feature.
2	Microphone	Transmits sound to other phones.
3	MIC Connector	Allows you to connect to the MIC port on the phone.

Icon Instructions

Certain features and icons do not apply to PSTN. Therefore, not all icons listed in the table below will display when using PSTN only. Please refer to the relevant sections for more information.

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

Icon	Description
	Network is unavailable
6	Registered successfully (SIP/PSTN account)

Icon	Description
0	Registered successfully (Yealink Cloud account)
2	Register failed (SIP account)
	Registering (SIP account)
•••>	Hands-free (speakerphone) mode
00	Voice Mail
\bowtie	Text Message
AA	Auto Answer
DND	Do Not Disturb (DND)
0	Call Hold
A	Call Mute
⊡Q́×	Ringer volume is 0
	Phone Lock
¢	Call Forward
₽	Forwarded Calls
~	Missed Calls
	Received Calls
`	Placed Calls
	Call and Conference Recording
<u>00</u>	Idle Recording
œ .	USB flash drive/PSTN box is detecting
4	USB flash drive is detected
PSTN	PSTN box is detected

Icon	Description
HD	High Definition Voice

LED Instructions

LED indicators on the CP860 IP conference phone and Mute indicator LED on the CPE80 expansion microphone (optional):

LED Status	Description
Calid and	The phone is initializing (This LED status is only
Solid red	applicable to CP860 IP conference phone). The call is muted.
Flashing red	The phone is ringing.
Solid green	The phone places a call. There is an active call on the phone.
Flashing green	The call is placed on hold or is held.
Off	The phone is powered off. The phone is idle.

User Interfaces

Two ways to customize configurations of your CP860 IP conference phone:

- The user interface on the IP phone.
- The user interface in a web browser on your PC.

The hardware components keypad and LCD screen constitute the phone user interface, which allows the user to execute all call operation tasks and basic configuration changes directly on the phone. In addition, you can use the web user interface to access all configuration settings. In many cases, it is possible to use either the phone user interface and/or the web user interface interchangeably. However, in some cases, it is only possible to use one or the other interface to operate the phone and change settings.

Phone User Interface

You can customize your phone by pressing the **Menu** soft key to access the phone user interface. The Advanced Settings option is only accessible to the administrator, and the default administrator password is "admin" (case-sensitive). For more information on customizing your phone with the available options from the phone user interface, refer to Customizing Your Phone on page 23.

Note For a better understanding of the menu structure, please refer to Appendix B – Menu Structure on page 157.

Web User Interface

In addition to the phone user interface, you can also customize your phone via web user interface. In order to access the web user interface, you need to know the IP address of your new phone. To obtain the IP address, press the OK key on the phone when the phone is idle. Enter the IP address (e.g., http://192.168.0.10 or 192.168.0.10 for IPv4; http://[2005:1:1:1:215:65ff:fe64:6e0a] or [2005:1:1:1:215:65ff:fe64:6e0a] for IPv6) in the address bar of a web browser on your PC. The default administrator user name and password are both "admin" (case-sensitive).

Note

The access to the Advanced settings of the Account or Network via web user interface may be blocked by the web browser (e.g., Chrome, Firebox) if you have installed "Adblock Plus plugin".

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 Address		
Firmware	\checkmark	\checkmark
Network		
Phone		
Accounts		
Basic Phone Settings		
Power Saving	х	
Contrast	\checkmark	
Backlight	\checkmark	\checkmark
Language	\checkmark	
Time & Date	\checkmark	
Administrator Password	\checkmark	

Options	Phone User Interface	Web User Interface
Key As Send	√	
Phone Lock	\checkmark	
Audio Settings		
Ring Tones	\checkmark	
Key Tone	\checkmark	
Contact Management		
Directory	x	
Local Directory	\checkmark	
Blacklist	\checkmark	
Favorite Directory	\checkmark	
Remote Phone Book	x	
Call History Management	\checkmark	
Search Source List in Dialing	x	
Logo Customization	х	
Programable Keys	x	
Account Registration	\checkmark	
Dial Plan	x	
Emergency Number	x	
Live Dialpad	x	
Hotline	\checkmark	
Basic Call Features		
Recent Call In Dialing	x	
Auto Answer	\checkmark	
Auto Redial	\checkmark	
Call Completion	\checkmark	
ReCall	x	
Do Not Disturb (DND)	\checkmark	\checkmark
Call Forward	\checkmark	
Call Waiting	\checkmark	
Conference	х	
Call Pickup	\checkmark	
Anonymous Call	\checkmark	
Anonymous Call Rejection	\checkmark	
Advanced Phone Features		
Call and Conference Recording	\checkmark	х
Idle Recording	\checkmark	х
Intercom	\checkmark	
Multicast Paging	x	
Music on Hold	x	\checkmark
SCA (Share Call Appearance)	x	
BLA (Bridge Line Appearance)	х	

Options	Phone User Interface	Web User Interface
Messages	\checkmark	
SIP Account		
User Options		
Туре	√	
Active Line	\checkmark	
Label	~	
Display Name	~	
Register Name	√	
User Name	\checkmark	
Password	~	v
Server Option		
SIP Server1/2	√	
Register Port	х	
Outbound Status	√	
Outbound Proxy1/2	√	
Fallback Interval	√	
NAT Status	√	
Yealink Cloud Account		
PIN Code		
PIN Code	\checkmark	x
Account		^
User Name	\checkmark	
Password	\checkmark	

Note

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

Documentations

The following table shows documentations available for the CP860 IP conference phone.

Name	Contents	Where found	Language
Quick Start Guide	Basic call features and	In the package	English
	phone customizations	On the website	Chinese/English
User Guide	Phone/Web user interface settings Basic call features and advanced phone features	On the website	Chinese/English

Note

You can also download the latest documentations online:

http://support.yealink.com/documentFront/forwardToDocumentDetailPage?documentId=34.

Getting Started

This chapter provides the following basic installation instructions and information for obtaining the best performance with the CP860 IP conference phone. Topics include:

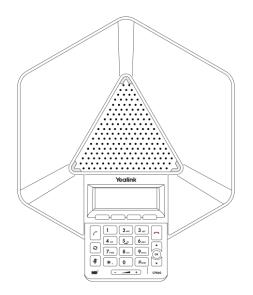
- Packaging Contents
- Phone Installation
- Setup Wizard
- Phone Status
- Basic Network Settings
- Registration
- Idle Screen

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

Packaging Contents

The following items are included in your CP860 IP conference phone package:

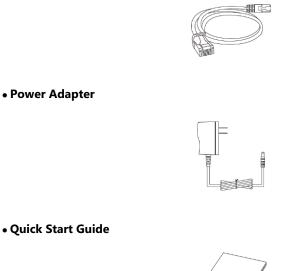
• CP860 IP Conference Phone



• 3.5mm Jack Cable



• Ethernet Cable





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

Optional Accessories

The following items are optional accessories for your CP860 IP conference phone. You need to purchase them separately if required.

• Expansion Microphone CPE80



• Expansion PSTN Box CPN10









PSTN Box CPN10

3m PSTN Cable

Double-side Tape

Quick Start Guide

Phone Installation

If your phone has already installed, proceed to Setup Wizard on page 16.

This section introduces how to install the phone:

- 1) Connecting the Network and Power
- 2) Connecting the Optional Expansion Microphone CPE80
- 3) Connecting the Optional USB Flash Drive
- 4) Connecting the Optional Expansion PSTN Box CPN10
- 5) Connecting the Optional PC or Mobile Device

Connecting the Network and Power

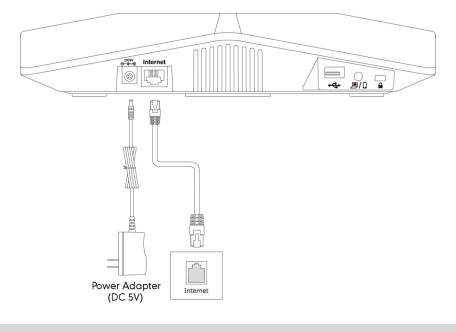
You have two options for power and network connections. Your system administrator will advise you which one to use.

- AC power (Optional)
- Power over Ethernet (PoE)

AC Power (Optional)

To connect the AC power:

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



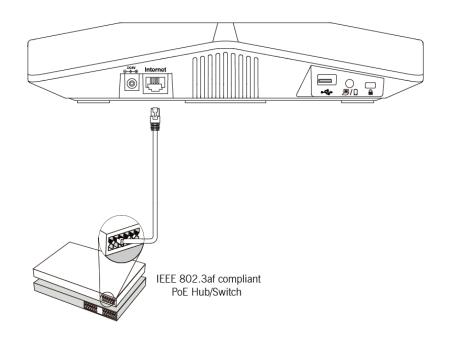
Note The IP phone should be used with Yealink original power adapter (5V/2A) only.

Power over Ethernet

With the included or a regular Ethernet cable, the CP860 IP conference phone can be powered from a PoE-compliant switch or hub.

To connect the PoE:

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



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

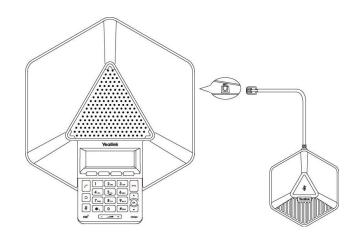
Important! Do not unplug or remove power to the phone while it is updating firmware and configurations.

Connecting the Optional Expansion Microphone CPE80

You can connect optional expansion microphone to enhance the room coverage of the conference phone.

To connect the expansion microphone:

1. Connect the free end of the optional expansion microphone cable to one of the MIC ports on the phone.

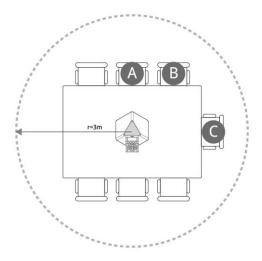




Up to two expansion microphones can be connected to an IP conference phone.

Positioning the CP860 and the Expansion Microphone CPE80

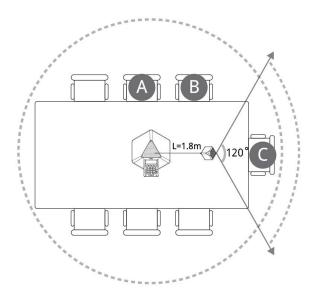
The microphone reception of the CP860 conference phone is ideal when placed within 3m of any speaker. The better reception is within 2m.



(CP860)

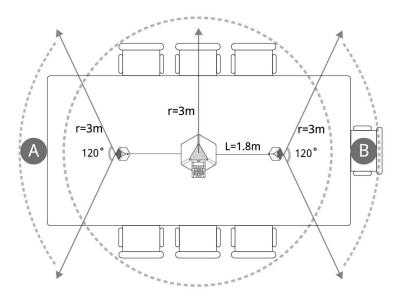
CPE80 is a directional microphone. Its coverage range is 120 degree. Ensure the microphone faces the speaker.

When the size of the conference room is more than 16 (4*4) square meters or the spacing between a speaker and CP860 IP conference phone is more than 2m, one expansion microphone should be connected to the CP860 IP conference phone.





When the size of the conference room is more than 36 (6*6) square meters, two expansion microphones should be connected to the CP860 IP conference phone.



(CP860+two expansion microphones)

Following the following guidelines to ensure optimum performance with the CP860 IP conference phone and the expansion microphones:

• Do not move or handle the CP860 IP conference phone or the expansion microphones

while on a call. Fix the microphone before a call and the better spacing between microphones and participants should be 0.5m to 2m.

- Minimize background noise from air conditioning units, fans or other equipment in conference room. Keep the microphone far away from the noise source.
- Do not speak at the same time; otherwise the microphones capture the voice of all the speakers, causing people in the far site cannot hear clearly.
- Do not move around in the conference room while on a call.

Connecting the Optional USB Flash Drive

You can connect a USB flash drive to record calls/conferences and play back recorded files.

To connect a USB flash drive:

- 1. Insert a USB flash drive into the USB port on the phone.

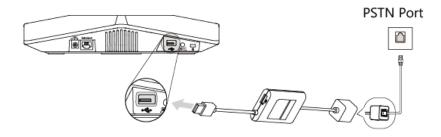
Connecting the Optional Expansion PSTN Box CPN10

You can connect optional expansion PSTN box CPN10 to make calls using the Public Switched Telephone Network (PSTN).

USB

To connect the expansion PSTN box:

1. Insert the USB plug on the expansion PSTN box into the USB port on the phone.





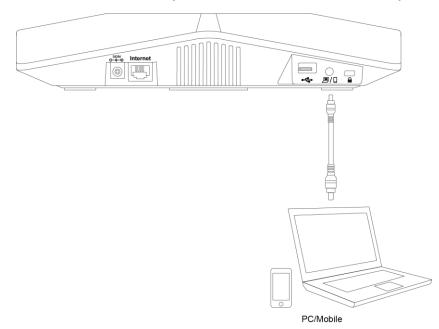
If you need to record calls when using the PSTN box, you can also connect the USB flash drive to the USB port on the PSTN box. Up to two expansion PSTN boxes can be connected to an IP conference phone. For more information, refer to *Yealink PSTN Box CPN10 Quick Start Guide*.

Connecting the Optional PC or Mobile Device

You can connect a PC or mobile device to listen to the PC or mobile audio using your conference phone.

To connect a PC or mobile device:

1. Connect one end of the 3.5mm jack cable to the PC/mobile port on the phone, and connect the other end to the headset jack on the mobile device or the AUX/MIC jack on the PC.



Setup Wizard

When the IP phone is first powered on or the phone settings are reset to factory defaults, the setup wizard will appear on the LCD screen after startup. You can disable this feature. For more information, contact your system administrator.

Configure the setup wizard:

1. Configure the language for the phone user interface.

The default language is English.

For more information, refer to Language on page 27.

- 2. Press the **Next** soft key to continue, or press the **Skip** soft key to skip the setup wizard and go to the idle screen.
- Configure time and date.
 For more information, refer to Time & Date on page 28.
- **4.** Press the **Next** soft key to continue.
- Configure basic network settings for the WAN port.
 For more information, refer to Basic Network Settings on page 18.

- 6. Press the Next soft key to continue.
- 7. Configure the SIP account information.

For more information, refer to Account Registration on page 58.

8. Press the Finish soft key to complete the setup wizard.

After you complete these steps, the LCD screen prompts the following, and then the phone goes to the idle screen.

Complete the Wizard!

Initializing...Please Wait

Phone Status

You can view phone status via phone user interface or web user interface. Available information of phone status includes:

- Network status (IPv4 status or IPv6 status, IP mode and MAC address).
 - IPv4 uses a 32-bit address.
 - IPv6 is an updated version of the current Internet Protocol to meet the increased demands for unique IP addresses, using a 128-bit address.
- Phone status (product name, hardware version, firmware version, product ID, MAC address and device certificate status).
- Account (SIP, Yealink Cloud or PSTN) status

To view the phone status via phone user interface:

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

Status					
1. IPv4:	10.2.20.189				
2. MAC Address:	00:15:65:65:45:D6				
3. Firmware:	37.81.0.10				
Back	آهي هي ا				

To view the phone status via web user interface:

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

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

Login	Enterprise IP Phone CP860
Username Password	admin
	ogin Cancel

4. Click Login to login.

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

Yealink CP860						Log Out English(English) 🔻
	Status	Network	Dsskey	Features	Settings	Directory Security
Status	Version					NOTE
	Firmware Ver	sion	37.81.0.10			
	Hardware Ve	sion	37.0.0.0.0).0		Version It shows the version of firmware and hardware.
	Device Certifica	te				Network
	Device Certifi	ate	Factory Insta	alled		It shows the network settings of Internet (WAN) port.
	Network					Account
	Internet Port		IPv4			It shows the registration status of SIP accounts.
	IPv4					You can click here to get
	WAN Port Ty	pe	DHCP			more guides.
	WAN IP Addr	ess	10.2.20.189			
	Subnet Mask		255.255.255	5.0		
	Gateway		10.2.20.254			
	Primary DNS		192.168.1.2	0		
	Secondary DI	IS	192.168.1.2	2		
	Network Comn	ion				
	MAC Address		0015656545	5D6		
	WAN Port St	atus	100Mbps Ful	II Duplex		
	Uptime		0 days 00:04	4		
	Account Statu	5				
	Account		Disabled			
	PSTN1		(PSTN1) : R	egistered		

Basic Network Settings

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

To configure the IP mode via phone user interface:

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

			AN Po	rt—	
1. I	P Mode	:			
IΡv	4				••
Bac	k 🛛	•		•	Save

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

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

To configure a static IPv4 address via phone user interface:

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

Sta	tic IPv	4 Client—	
1. IP Address			
192.168.1.20			
Back 📘 12	3 I	Delete 📘	Save

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

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

To configure a static IPv6 address via phone user interface:

- Press Menu->Settings->Advanced Settings (default password: admin)
 ->Network->WAN Port.
- 2. Press v to select **IPv6** and then press the **Enter** soft key.
- 3. Press 🖵 to select Static IPv6 Client and then press the Enter soft key.
- Enter the desired value in the IP Address, IPv6 IP Prefix, Default Gateway, Pri.DNS and Sec.DNS field respectively.

Static IPv6 Client-	_
1. IP Address	
2005:1:1:1:12	
Back ZaB Delete Savo	e

- 5. Press the Save soft key to accept the change or the Back soft key to cancel.
- **Note** Wrong network settings may result in inaccessibility of your phone and may also have an impact on your network performance. For more information on these parameters, contact your system administrator.

Registration

Generally, your phone will be deployed with multiple other phones. In this case, your system administrator will configure the phone parameters beforehand, so that after you start up your phone, the phone will be registered and ready for use. If your phone is not registered, you may have to register it. For more information on how to register your phone, refer to Account Registration on page 58.

Idle Screen

If the phone has successfully started up, the idle LCD screen is displayed as below.



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

Entering Data and Editing Fields

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

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

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

Input Mode Keypad	2aB	abc	Abc (initials in capitals)	ABC	123
1	1				1
2 ABC	2abcABC	abc2äæåàá âãç	abc2äæåàá âãç	ABC2ÄÆÅ ÀÁÂÃÇ	2
3 DEF	3defDEF	def3èéêëð	def3èéêëð	DEF3ÈÉÊËÐ	3
4 он	4ghiGHI	ghi4ìíîï	ghi4ìíîï	GHI4ÌÍÎÏ	4

Input Mode Keypad	2aB	abc	Abc (initials in capitals)	ABC	123
5 JAL	5jklJKL	jkl5£	jkl5£	JKL5£	5
быно	6mnoMNO	mno6öøòó ôõñ	mno6öøòó ôõñ	MNO6ÖØ ÒÓÔÕÑ	6
7 PORS	7pqrsPQRS	pqrs7ßS	pqrs7ßS	PQRS7S	7
8 TUV	8tuvTUV	tuv8ùúûü	tuv8ùúûü	TUV8ÙÚÛ Ü	8
9 _{wxvz}	9wxyzWXY Z	wxyz9ýÞ	wxyz9ýÞ	WXYZ9ÝÞ	9
0	0	space	space	space	0
*.	*.@,'?!\-() /:_;+&%=< > £\$¥¤[]{}~ ^j¿§#"	*.@,'?!\-() /:_;+&%=< > £\$¥¤[]{}~ ^j¿§#"	*.,'?!\-()@/: _;+&%=<> £ \$¥¤[]{}~ ^¡¿§#"	*.@,'?!\-() /:_;+&%=< > £\$¥¤[]{}~ ^j¿§#"	.*:/@[]
#send	#	#	#	#	#

To enter or edit data:

Do one of the following:

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

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

Customizing Your Phone

You can customize your CP860 IP conference phone by personally configuring certain settings, for example, backlight, time & date and ring tones. You can add contacts to the phone's local directory manually or from call history. You can also personalize different ring tones for different callers.

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

- General Settings
- Audio Settings
- Contact Management
- Call History Management
- Search Source List in Dialing
- System Customizations

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

General Settings

Power Saving

The power saving feature is used to turn off the backlight to conserve energy. The IP phone enters power-saving mode after it has been idle for a certain period of time.

The IP phone will exit power-saving mode if one of the following phone events occurs:

- Press any key.
- There is an incoming call.
- A new prompt (e.g., missed call, new voice mail or forwarded call).

You can configure the following power-saving settings:

- Office Hour
- Idle Timeout (minutes)

The office hour and idle timeout (minutes) settings work only if the power saving feature is enabled.

Note Power saving is configurable via web user interface only.

Enabling the Power Saving

To enable the power saving feature via web user interface:

- 1. Click on Settings->Power Saving.
- 2. Select Enabled from the pull-down list of Power Saving.

Yealink CP860		Log Out English(English) ▼		
	Status Account Network	Dsskey Features	Settings	Directory Security
Preference	Power Saving	Enabled v		NOTE
Time & Date	Office Hour	07 19		Settings Powersaving
Call Display	Tuesday	07 19		You can click here to get more guides.
Upgrade	Wednesday	07 19		
Auto Provision	Thursday	07 - 19		
Configuration	Friday	07 19		
Dial Plan	Saturday	07 07		
	Sunday	07 07		
Voice	Idle TimeOut (minutes)			
Ring	Office Hour Idle TimeOut	960		
Tones	Off Hour Idle TimeOut	10		
Softkey Layout	User Input Extension Idle TimeOut	10		
TR069	Confirm	Cancel		
Voice Monitoring				
SIP				
Power Saving				

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

Configuring the Office Hour

Office Hour specifies the starting time and ending time in the office each day.

To configure the office hour via web user interface:

1. Click on **Settings**->**Power Saving**.

2. Enter the starting time and ending time respectively in the desired day field.

alink cp860	Status	Account	Network	Dsskey	Features	Settings	Directory	Security
Preference	Po	ower Saving		Enabled	Ŧ		NOTE	
Time & Date	O	ffice Hour					Settings Pow	versaving
		Monday		07 - 19				-
Call Display		Tuesday		07 19			You can cli more guides.	ck here to get
Upgrade		Wednesday		07 19				
Auto Provision		Thursday		07 19				
Configuration		Friday		07 - 19				
-		Saturday		07 07				
Dial Plan		Sunday		07 07				
Voice	Id	lle TimeOut (minu	tes)					
Ring		Office Hour Idle Tir	meOut	960				
Tones		Off Hour Idle Time	Out	10				
		User Input Extensi	on Idle TimeOut	10				
Softkey Layout		00			Creat			
TR069		Confi	m		Cancel			
Voice Monitoring								
SIP								

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

Configuring the Idle Timeout

Idle Timeout specifies the period of time before the IP phone enters power-saving mode. The following three types of idle timeout you can configure:

- Office Hours Idle TimeOut: Configures the idle timeout for office hours.
- Off Hours Idle TimeOut: Configures the idle timeout for non-office hours.
- **User Input Extension Idle TimeOut**: Configures idle timeout that applies after you use the IP phone (for example, press a key on the phone).

By default, the Office Hours Idle Timeout is much longer than the Off Hours Idle TimeOut. If you use the IP phone, the idle timeout that applies (User Input Extension Idle TimeOut or Office Hours/Off Hours Idle TimeOut) is the timeout with the highest value. If the phone has an incoming call or new message, the User Input Extension Idle TimeOut is ignored.

To configure the idle timeout via web user interface:

- 1. Click on Settings->Power Saving.
- Enter the desired value in the Office Hours Idle TimeOut field.
 The default value is 960, you can set to 1-960.
- Enter the desired value in the Off Hours Idle TimeOut field.
 The default value is 10, you can set to 1-10.
- 4. Enter the desired value in the User Input Extension Idle TimeOut field.

alink CP860	Status	Account	Network	Dsskey	Features	Settings	Directory Security
Preference	P	ower Saving		Enabled	Ŧ		NOTE
Time & Date	0	ffice Hour		07 19			Settings Powersaving
Call Display		Tuesday		07 - 19			You can click here to ge more guides.
Upgrade		Wednesday		07 19			more guideon
Auto Provision		Thursday		07 19			
Configuration		Friday		07 - 19			
Dial Plan		Saturday		07 07			
Voice	Ic	Sunday lle TimeOut (minutes))	07 07			
Ring		Office Hour Idle TimeC	Dut	960			
Tones		Off Hour Idle TimeOut		10			
Softkey Layout		User Input Extension I	dle TimeOut	10			
TR069		Confirm			Cancel		
Voice Monitoring							
SIP							

The default value is 10, you can set to 1-30.

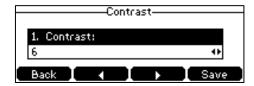
5. Click **Confirm** to accept the change.

Contrast

You can configure the contrast of the LCD screen to a comfortable level. The intensity of contrast ranges from 1 to 10 and the highest intensity is 10.

To configure the contrast via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Display->Contrast.
- Press the d or ► soft key to decrease or increase the intensity of contrast. The default contrast level is "6".



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

Contrast is configurable via web user interface at the path Settings->Preference.

Backlight

You can configure the backlight to adjust the brightness of the LCD screen. Backlight status on the LCD screen can be configured from the following options:

• Always On: Backlight is on permanently.

- Always Off: Backlight is off permanently.
- **15s**, **30s**, **1min**, **2min**, **5min**, **10min** or **30min**: Backlight is turned off when the phone is inactive after the designated time (in seconds).

To configure the backlight via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Display->Backlight.
- 2. Press the **d** or **b** soft key to select the desired value from the **Backlight Time** field.

Backlight-	
1. Backlight Time:	
30s	•
Back	I Save

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

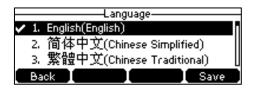
Backlight is configurable via web user interface at the path Settings->Preference.

Language

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

To change the language for the phone user interface:

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



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

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

To change the language for the web user interface:

1. Select the desired language from the pull-down list at the top-right corner of web user interface.

Yealink cp860							Log Out English(English) ▼
	Status	Account	Network	Dsskey	Features	Settings	Directory Security
Preference	Live	Dialpad		Disabled	T		NOTE
Time & Date	Inter Digit Time(1~14s)			4 30s			Live Dialpad It allows IP phones to
Call Display	Backlight Time(seconds) Contrast		6	Ŧ		automatically dial out the entered phone number after a specified period of time.	
Upgrade	Watch Dog		Disabled	¥		Backlight	
Auto Provision	Ring	Туре		Ring1.wav	T		Specifies the brightness of the LCD screen display.

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

Time & Date

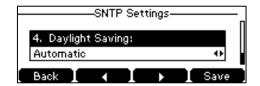
The time and date are displayed on the LCD screen when the phone is idle. You can configure the phone to obtain the time and date from the Simple Network Time Protocol (SNTP) server automatically, or configure the date and time manually. If the phone cannot obtain the time and date from the SNTP server,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

To configure the SNTP settings via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Time & Date->SNTP Settings.
- Press the
 I or
 Soft key to select the time zone that applies to your area from the Time Zone field.

The default time zone is "+8".

- Enter the domain name or IP address in the NTP Server1 and NTP Server2 field respectively.
- 4. Press the ◀ or ▶ soft key to select the desired value from the Daylight Saving field.



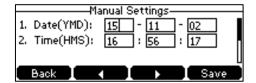
- 5. Press the ◀ or ▶ soft key to select the desired time zone name from the Location field. This field appears only if Daylight Saving field is selected to Automatic. The default time zone name is "China(Beijing)".
- 6. Press the Save soft key to accept the change or the Back soft key to cancel.

Note

Please refer to Appendix A - Time Zones for the list of available time zones on the IP phone.

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

- 1. Press Menu->Settings->Basic Settings->Time & Date->Manual Settings.
- 2. Enter the specific time and date.



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

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

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

- 1. Press Menu->Settings->Basic Settings->Time & Date->Time & Date Format.
- Press the d or ► soft key to select the desired time format (12 Hour or 24 Hour) from the Time Format field.

Time & Date Format	
1. Time Format:	
24 Hour	4
Back I 🕢 I 🕨] Save

- 3. Press the ◀ or ▶ soft key to select the desired date format from the Date Format field.
- 4. Press the Save soft key to accept the change or the Back soft key to cancel.

There are 7 available date formats. For example, for the date format "WWW DD MMM", "WWW" represents the abbreviation of the weekday, "DD" represents the two-digit day, and "MMM" represents the first three letters of the month.

The date formats available:

Date Format	Example (2016-09-02)
WWW MMM DD	Fri, Sep 02
DD-MMM-YY	02-Sep-16
YYYY-MM-DD	2016-09-02
DD/MM/YYYY	02/09/2016
MM/DD/YY	09/02/16
DD MMM YYYY	02 Sep, 2016
WWW DD MMM	Fri, 02 Sep

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

You can also customize the date format. Contact your system administrator for more information.

Administrator Password

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

To change the administrator password via phone user interface:

- Press Menu->Settings->Advanced Settings (default password: admin) ->Change Password.
- 2. Enter the current password in the Old PWD field.

Note

- 3. Enter the new password in the New PWD field.
- 4. Re-enter the new password in the **Confirm PWD** field.

		-Chang	e Pa	issword-			- п
3. C	onfirn s	n PWD:					
					_		
Back	: 1	abc	1	Delete	1	Save	

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

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

Key As Send

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

To configure key as send via phone user interface:

- 1. Press Menu->Features->Key As Send.
- Press the d or ► soft key to select # or * from the Key As Send field, or select Disabled to disable this feature.

Key As Send	
1. Key As Send:	
#	41
Back	Save

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

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

Phone Lock

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

Phone lock consists of the following:

Menu Key:	The Menu soft key is locked. You cannot access the menu of the
	phone until unlocked.
Function Keys:	The function keys are locked. You cannot use the mute key, redial
	key, OK key, up and down navigation keys and soft keys until
	unlocked.
All Keys:	All keys are locked, except the Volume key, on-hook key, off-hook
	key, * key, # key and digit keys. You are only allowed to dial
	emergency numbers, reject by pressing the Reject soft key or
	on-hook key, answer incoming calls by pressing the Answer soft
	key or off-hook key, and end a call by pressing the EndCall soft key

or on-hook key.

Note The emergency number setting, if desired, must be made before lock activation. For more information, refer to Emergency Number on page 67.

To activate the phone lock via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Phone Lock.
- Enter the desired PIN (default PIN: 123) in the Unlock PIN field, and then press the OK soft key.
- 3. Press the ◀ or ▶ soft key to select **Enabled** from the **Lock Enable** field.
- 4. Press the ◀ or ▶ soft key to select the desired type from the Lock Type field.
- 5. Enter the desired interval of automatic phone lock in the Lock Time Out field.

The default timeout is 0. It means the phone will not be automatically locked. You need to long press $\#_{mo}$ to lock it immediately when the phone is idle.

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

Phone Lock	
3. Lock Time Out: 0	
Back 123 Delete Save	

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

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



To change the phone unlock PIN via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Change PIN.
- 2. Enter the desired value in the Old PIN, New PIN and Confirm PIN field respectively.

		—Cha	nge	PIN		
3. 0	Confirm	n PIN:				

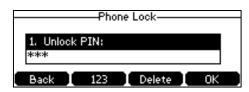
(Bac	k I	123		Delete	Save	

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

Note The unlock PIN length must be within 15 digits.

To unlock the phone via phone user interface:

- 1. Press any locked key, the LCD screen prompts "Unlock PIN".
- 2. Enter the PIN in the **Unlock PIN** field.



3. Press the **OK** soft key to unlock the keypad.

The icon of disappears from the LCD screen.

You can long press $[\#_{mo}]$ or wait for a period of time (if configured) to lock the keypad again.

Note You can also unlock the keypad by entering administrator password. When you enter the administrator password to unlock the keypad, the phone will turn to the Change PIN screen.

To deactivate the phone lock via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Phone Lock.
- 2. Enter the desired PIN (default PIN: 123) in the **Unlock PIN** field, and then press the **OK** soft key.
- 3. Press the **d** or **b** soft key to select **Disabled** from the **Lock Enable** field.

Phone Lock-	
Thone Look	
1. Look Enable:	
Disabled	••
Back] Save

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

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

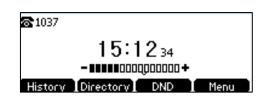
Audio Settings

Volume

You can press the volume key to adjust the ringer volume when the phone is idle or ringing. You can also press the volume key to adjust the speaker volume when the phone is during a call.

To adjust the ringer volume when the phone is idle:

1. Press - - + to adjust the ringer volume.



To adjust the ringer volume when the phone is ringing:

1. Press - - + to adjust the ringer volume.

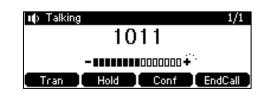


You can also press (- - + to adjust the ringer volume when selecting a ring tone. For more information, refer to Ring Tones on page 33.

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

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

1. Press - - + to adjust the speaker volume.



Note You can also press - - + to adjust the volume when playing back the recording calls. For more information, refer to Recorded Files Playback on page 102.

Ring Tones

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

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

1. Press Menu->Settings->Basic Settings->Sound->Ring Tones->Common.

2. Press \frown or \bigtriangledown to select the desired ring tone.



- 3. (Optional.) Press - + to adjust the ringer volume.
- 4. Press the Save soft key to accept the change or the Back soft key to cancel.

A ring tone for the phone is configurable via web user interface at the path

Settings->Preference->Ring Type.

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

- 1. Press Menu->Settings->Basic Settings->Sound->Ring Tones.
- 2. Press \frown or \bigtriangledown to select the desired account and then press the **Enter** soft key.

		-Ring Tones-	
1.	Common	-	Π
2.	1037		
	Back 📘		Enter

3. Press \frown or $\overleftarrow{}$ to select the desired ring tone.

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

🖌 1. Common		
2. Ring1.wav		
3. Ring2.wav		U
Back I	I Save	Ó

- 4. (Optional.) Press - + to adjust the ringer volume.
- 5. Press the Save soft key to accept the change or the Back soft key to cancel.

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

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

1. Click on Settings->Preference.

2. In the **Upload Ringtone** field, click **Browse** to locate a ring tone file (the file format must be *.wav) from your local system.

Yealink CP860	Status Account Network	Dsskey Features Sett	Log Out Englah(Englah) • Directory Security
Preference	Live Dialpad	Disabled -	NOTE
Time & Date	Inter Digit Time(1~14s) Backlight Time(seconds)	4 30s 👻	Live Dialpad It allows IP phones to
Call Display	Contrast	6 🗸	automatically dial out the entered phone number after a
Upgrade	Watch Dog	Disabled 👻	specified period of time.
Auto Provision	Ring Type	Ring1.wav 👻	Backlight Specifies the brightness of the
Configuration	Upload Ringtone	Browse No file selected. Upload Cancel	LCD screen display. Contrast Specifies the contrast of the
Dial Plan	Confirm	Cancel	LCD screen display.

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

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

Both single custom ring tone file and total custom ring tone files must be within 100KB.

Uploading custom ring tones for your phone is configurable via web user interface only.

Key Tone

Note

If you enable key tone, the phone will produce a sound when you press the keypad.

To configure key tone via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Sound->Key Tone.
- 2. Press the ◀ or ▶ soft key to select the desired value from the **Key Tone** field.

Key Tone—	
1. Key Tone	
Enabled	••
Back 🖌 🖌 🕨	Save

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

Key tone is configurable via web user interface at the path Features->Audio.

Contact Management

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

- Directory
- Local Directory
- Blacklist

- Favorite Directory
- Remote Phone Book

Directory

Directory provides an easy access to frequently used lists. The lists may contain Local Directory, Favorite Directory, History, Remote Phone Book and LDAP. You can configure the list(s) to be accessed for the **Directory** soft key.

Note LDAP list is hidden by default. To configure LDAP list, you need to enable LDAP feature in advance. For more information, contact your system administrator.

To configure the list(s) to access for the Directory soft key via web user interface:

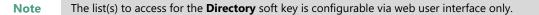
- 1. Click on Directory->Setting.
- In the **Directory** block, select the desired list from the **Disabled** column and then click →.

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

- 3. Repeat the step 2 to add more lists to the **Enabled** column.
- 4. To remove a list from the **Enabled** column, select the desired list and click _____.
- To adjust the display order of enabled lists, select the desired list and click or .
 The LCD screen will display the list(s) in the adjusted order.

Yealink	Status Account Network Dsskey Features Settings	Log Out English(English) • Directory Security
Local Directory	Directory	NOTE
Remote Phone Book Phone Call Info LDAP Multicast IP Setting	Disabled Enabled History Remote Phone Bool	Directory It provides easy access to frequently used lists. Search Source in Dialing It allows the IP phone to atommake address the source of the based on the entered string, and display results on the pre- dialing screen. Recent Call In Dialing It allows users to view the placed calls list when the phone is on the pre-dialing screen.

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



To view the directory list(s) via phone user interface:

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

The LCD screen displays the enabled list(s) in the directory.

	Directory	
1.	Local Directory	
2.	Favorite Directory	
	Back I I I	Enter

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

Note If the remote phone book or LDAP is not configured in advance, you cannot see remote phone book or LDAP lists on the phone user interface. For more information on remote phone book, refer to Remote Phone Book on page 48. For more information on the LDAP, contact your system administrator.

Local Directory

The built-in phone directory can store the names and phone numbers of your contacts. You can store up to 1000 contacts and 47 groups in your phone's local directory. You can add new groups and contacts, edit, delete or search for a contact, or simply dial a contact number from the local directory.

Note

Local directory can be backed up to the provisioning server. For more information, contact your system administrator.

Adding Groups

To add a group to the local directory:

1. Press Directory->Local Directory.



If the Local Directory is removed from the directory (refer to Directory on page 36), press **Menu->Directory->Local Directory** to enter the local directory.

- 2. Press the AddGr soft key.
- 3. Enter the desired group name in the Name field.
- 4. Press the **d** or **b** soft key to select the desired ring tone from the **Ring** field.

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

	- Ad	ld Gro	up—		
Ring:					
Auto				41	
Back	•		Þ	Add	

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

Editing Groups

To edit a group in the local directory:

1. Press Directory->Local Directory.

_	Local Directory	
1.	All Contacts	
2.	Test	
	Back 🚺 AddGr 📘 Search 📘	Enter

If Local Directory is removed from the directory (refer to Directory on page 36), press **Menu->Directory->Local Directory** to enter the local directory.

- **2.** Press \frown or \checkmark to highlight the desired contact group.
- 3. Press the Option soft key, and then select Detail.

Local Director	ry		
Detail			
Delete			
Delete All			U
Cancel I		ОK	

4. Press i or i to highlight the group information and then edit.

		Test——			_
Name:					
Test					
Back	Abc	[Delet	e I	Save	

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

Deleting Groups

To delete a group from the local directory:

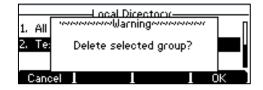
1. Press Directory->Local Directory.



If Local Directory is removed from the directory (refer to Directory on page 36), press Menu->Directory->Local Directory to enter the local directory.

- **2.** Press \frown or \bigtriangledown to highlight the desired contact group.
- 3. Press the **Option** soft key, and then select **Delete**.

The LCD screen prompts the following warning:



4. Press the OK soft key to confirm the deletion or the Cancel soft key to cancel.

You can also delete all groups by pressing the Option soft key, and then select Delete All.

Adding Contacts

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

- Manually
- From call history
- From a remote phone book

Adding Contacts Manually

To add a contact to the local directory manually:

1. Press Directory->Local Directory.

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

If Local Directory is removed from the directory (refer to Directory on page 36), press Menu->Directory->Local Directory to enter the local directory. 2. Select the desired contact group and then press the Enter soft key.

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

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

	Add	Contact—	
Name:			
Jim			
Back	Abc	Delete	Add

5. Press the \triangleleft or \triangleright soft key to select the desired ring tone from the **Ring** field.

If **Auto** is selected, this contact will use the ring tone according to the priority: Group ring tone (refer to Adding Groups) >Account ring tone (refer to Ring Tones)>Phone ring tone (refer to Ring Tones).

- 6. Press the \blacktriangleleft or \blacktriangleright soft key to select the desired group from the **Group** field.
- 7. Press the Add soft key to accept the change or the Back soft key to cancel.

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

Adding Contacts from Call History

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

- 1. Press the History soft key.
- 2. Select the desired call history list and then press the Enter soft key.
- **3.** Press \frown or \bigtriangledown to highlight the desired entry.
- 4. Press the Option soft key, and then select Add to Contact.

K	—All Calls—	—1/44 >
Detail		
Add to Contact		
Add to Blacklist		
Cancel		ОК

- 5. Enter the contact name.
- 6. Press the Save soft key to save the entry to the local directory.

If the contact already exists in the local directory, the LCD screen will prompt "Overwrite the original contact?". Press the **OK** soft key to overwrite the original contact in the local directory or the **Cancel** soft key to cancel.

Adding Contacts from Remote Phone Book

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

1. Press Menu->Directory->Remote Phonebook.

If Remote Phone Book is added to the directory (refer to Directory on page 36), press

Directory->Remote Phone Book to enter the remote phone book.

- 2. Select the desired remote group and then press the Enter soft key.
- **3.** Press \frown or \smile to highlight the desired entry.
- 4. Press the **Option** soft key, and then select **Add to Contact**.
- 5. Press the **Save** soft key to save the contact to the local directory.

If the contact already exists in the local directory, the LCD screen will prompt "Overwrite the original contact?". Press the **OK** soft key to overwrite the original contact in the local directory or the **Cancel** soft key to cancel.

For more information on remote phone book operating, refer to Remote Phone Book on page 48.

Editing Contacts

To edit a contact in the local directory:

1. Press Directory->Local Directory.

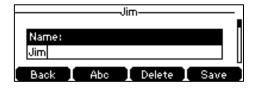
	Local Directory	
1.	All Contacts	
2.	Test	
		U
	Back 👖 AddGr 📘 Search 📘 Enter	

If Local Directory is removed from the directory (refer to Directory on page 36), press Menu->Directory->Local Directory to enter the local directory.

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

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

- **3.** Press \frown or \bigtriangledown to highlight the desired contact.
- 4. Press the Option soft key, and then select Detail.



- 5. Press or v to highlight the contact information and then edit.
- 6. Press the Save soft key to accept the change or the Back soft key to cancel.

Deleting Contacts

To delete a contact from the local directory:

1. Press Directory->Local Directory.



If Local Directory is removed from the directory (refer to Directory on page 36), press Menu->Directory->Local Directory to enter the local directory.

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

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

- **3.** Press \frown or \bigtriangledown to highlight the desired contact.
- 4. Press the **Option** soft key, and then select **Delete**.

The LCD screen prompts the following warning:



5. Press the OK soft key to confirm the deletion or the Cancel soft key to cancel.

You can also delete all contacts by pressing the **Option** soft key, and then select **Delete All**.

Placing Calls to Contacts

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

1. Press Directory->Local Directory.



If Local Directory is removed from the directory (refer to Directory on page 36), press **Menu->Directory->Local Directory** to enter the local directory.

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

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

- **3.** Press \frown or \bigtriangledown to highlight the desired contact.
- 4. Do one of the following:
 - If only one number for the contact is stored in the local directory, press the Send soft

key to dial out the number.

- If multiple numbers for the contact are stored in the local directory, press the Send soft key to display a list of numbers.
 - Press \frown or \bigtriangledown to highlight the desired number.

Press the **Send** soft key to dial out the number.

Searching for Contacts

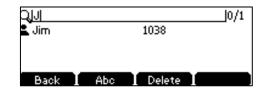
To search for a contact in the local directory:

1. Press Directory->Local Directory.

Local Directory	
1. All Contacts	
2. Test	
	U
. Back ${f I}$ AddGr ${f I}$ Search ${f I}$ Ent	er

If Local Directory is removed from the directory (refer to Directory on page 36), press Menu->Directory->Local Directory to enter the local directory.

- 2. Press the Search soft key.
- **3.** Enter a few continuous characters of the contact name or continuous numbers of the contact number (office, mobile or other number) using the keypad.



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

Importing/Exporting Contact Lists

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

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

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

3. Click Import XML to import the contact list.

						Log Out
alink cp860						English(English) -
	Status	Account	Network	Dsskey	eatures Settings	Directory Security
ocal Directory	Index	Name	Office Number	Mobile Other Number Number	All Contacts 👻 🔲	NOTE
	1	1	1		All Contacts	
emote Phone ook	2	2	2		All Contacts	Local Directory The built-in phone directory can
JOK	3					store the names and phone
ione Call Info	4					numbers of your contacts.
	5					You can add new groups and
AP	7					contacts, edit, delete or search for a contact, or simply dial a
ulticast IP	8					contact number from the local
incluse in	9					directory.
ting	10					You can import or export the
	Page 1 🔻 P	Next	Hang Up	Delete All Delete	Move To Al Contac -	contact list.
	Page I 🔹 P	INEXL	Hang Op	Сору То	Move to All contac +	You can click here to get
						more guides.
	Contacts			Group Setting		
	Name			Group		
	Office Number			Ring	Auto 👻	
	Mobile Number			Add Edit	Delete Delete All	
	Other Number			Import Local Directo		
	Ring Tone	Auto	-	Browse No file se	elected.	
	Group	All Co	ontacts -	Import XML Ex	port XML	
	Account	Auto	-	Browse No file se	elected.	
	Add		dit	Import CSV Ex	port CSV 🗌 Show Title	

The web user interface prompts "The original contact will be covered, Continue?".

4. Click **OK** to complete importing the contact list.

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

- 1. Click on Directory->Local Directory.
- Click Browse to locate a contact list file (the file format must be *.csv) from your local system.
- 3. (Optional.) Check the Show Title checkbox.

It will prevent importing the title of the contact information which is located in the first line of the CSV file.

- 4. Click Import CSV to import the contact list.
- 5. (Optional.) Mark the **On** radio box in the **Delete Old Contacts** field.

It will delete all existing contacts while importing the contact list.

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

Yealink CP860							Eng	Log Out Jish(English) 🔻
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security
Preview	Del Old contact displa Index (display_1 Bob 2 Joy 3 Linda	y_name offi	-	obile_number bile_number •	other_number	ine Ine	NOTE contacts-previe 10 You can cli more guides.	ew-note ck here to get
	4		In	nport	_	Þ		

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

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

To export a contact list via web user interface:

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

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

Blacklist

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

To add a contact to the blacklist directory manually:

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

	-Add	Blacklist—			-
Name:					
Kelly	A1		_		
🛛 Back 🛛	Abc	📘 Delete	21	Add	

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

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

1. Press Directory->Local Directory.

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

If Local Directory is removed from the directory (refer to Directory on page 36), press Menu->Directory->Local Directory to enter the local directory.

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

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

- **3.** Press \frown or \bigtriangledown to highlight the desired contact.
- 4. Press the Option soft key, and then select Add to Blacklist.

The LCD screen prompts following warning:



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

For operating instructions on editing, deleting, placing calls to and/or searching for contacts in the blacklist, refer to the operating instructions of Editing Contacts on page 41, Deleting Contacts on page 42, Placing Calls to Contacts on page 42 and/or Searching for Contacts on page 43.

Favorite Directory

Favorites are the contacts in your local directory that you call most often. You can add up to 1000 contacts as favorites from the local directory. You can view the favorites in the favorite directory. You can also remove contacts or search for a contact, or simply dial a contact number from the favorite directory.

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

1. Press Directory->Local Directory.

If Local Directory is removed from the directory (refer to Directory on page 36), press Menu->Directory->Local Directory to enter the local directory.

- Select the desired contact group and then press the Enter soft key.
 If the contact is not in any group, select All Contacts and then press the Enter soft key.
- **3.** Press \frown or \bigtriangledown to highlight the desired contact.
- 4. Press the Option soft key, and then select Copy to Favorite.

The LCD screen prompts following warning:



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

The contact added to the favorite directory from the local directory also exists in the local directory.

You can delete favorites to make room for new favorites.

To remove a contact from the favorite directory:

1. Press Directory->Favorite Directory.

If Favorite Directory is removed from the directory (refer to Directory on page 36), press **Menu->Directory->Favorite Directory** to enter the favorite directory.

- **2.** Press \frown or \bigtriangledown to highlight the desired contact.
- 3. Press the Remove soft key.

The LCD screen prompts following warning:



4. Press the OK soft key to confirm the deletion.

If the contact added to the favorite directory is deleted in the local directory, it will be automatically deleted from the favorite directory.

For operating instructions on placing a call to a contact from the favorite directory, refer to the operating instructions of Placing Calls to Contacts on page 42.

To search for a contact in the favorite directory:

1. Press Directory->Favorite Directory.

If Favorite Directory is removed from the directory (refer to Directory on page 36), press **Menu->Directory->Favorite Directory** to enter the favorite directory.

2. Enter a few continuous numbers of the contact number (office, mobile or other number) using the keypad.

QUI		_0/1
Jim	1038	
Back	Abc I Delete I	

The contacts whose phone number matches the characters entered will appear on the LCD screen. You can dial from the result list.

Remote Phone Book

You can add new contacts to the local directory, search for a contact, or simply dial a contact number from the remote phone book.

You can configure your new phone to access up to 5 remote phone books. The phone supports up to 5000 remote phone book entries. For the access URL of the remote phone book, contact your system administrator.

For operating instructions on placing calls to and/or searching for contacts in the remote phone book, refer to the operating instructions of Placing Calls to Contacts on page 42 and/or Searching for Contacts on page 43.

Configuring an Access URL and Update Time Interval

To configure an access URL and update time interval for a remote phone book via web user interface:

- 1. Click on Directory->Remote Phone Book.
- 2. Enter the access URL in the Remote URL field.
- 3. Enter the name in the **Display Name** field.
- Enter the desired refresh period in the Update Time Interval(Seconds) field. The default value is 21600.

Yealink	Status Account Network D	sskey Features Settings	Log Out English(English) v Directory Security
Local Directory	Index Remote URL	Display Name	NOTE
Remote Phone	1 http://10.3.6.184/Department.xml	Product	Remote Phone Book
Book	2		It is a centrally maintained phone book, stored on the
Phone Call Info	3		remote server.
	4		Users only need the access URL of the remote phone book. The
LDAP	5		IP phone can establish a connection with the remote
Multicast IP			server and download the phone book, and then display the
Setting	Incoming/Outgoing Call Lookup	Disabled v	remote phone book entries on the phone user interface.
	Update Time Interval(Seconds)	21600	You can click here to get
	Confirm	Cancel	more guides.

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

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

Accessing the Remote Phone Book

To access your remote phone book via phone user interface:

1. Press Menu->Directory->Remote Phonebook.

If Remote Phone Book is added to the directory (refer to Directory on page 36), press **Directory**->**Remote Phone Book** to enter the remote phone book.

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

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

	Pro	duct———	14/16-
🚨 Test1		32000	Π
🚨 Test2		303	
🛓 Test3		6650	H
Back	Search	Option	Send

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

Incoming/Outgoing Call Lookup

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

To configure incoming/outgoing call lookup via web user interface:

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

Yealink cp860	Status Account Network	Dsskey Features Settings	Log Out English(English) • Directory Security
Local Directory	Index Remote URL	Display Name	NOTE
Remote Phone	1 http://10.3.6.184/Department.xml	Product	Remote Phone Book
Book	2		It is a centrally maintained phone book, stored on the
Phone Call Info	3		remote server.
	4		Users only need the access URL of the remote phone book. The
LDAP	5		IP phone can establish a connection with the remote
Multicast IP			server and download the phone book, and then display the
Setting	Incoming/Outgoing Call Lookup	Enabled V	remote phone book entries on the phone user interface.
	Update Time Interval(Seconds)	21600	You can click here to get
	Confirm	Cancel	more guides.

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

Call History Management

The CP860 IP conference phone maintains call history lists of Missed Calls, Placed Calls, Received Calls and Forwarded Calls. Call history lists support 400 entries in all. You can view call history, place a call, add a contact or delete an entry from the call history list.

History record feature enables the phone to save the call history. If you don't want to save the call history, you can disable the feature. History record feature is enabled by default.

Viewing History Records

To view call history:

1. Press the **History** soft key.

The LCD screen displays all call history lists.

- 2. Select the desired call history list and then press the Enter soft key.
- **3.** Press \frown or \bigtriangledown to select the desired entry.
- 4. Press the Option soft key, and then select Detail.

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

Placing a Call from History Records

To place a call from the call history list:

- 1. Press the **History** soft key.
- 2. Select the desired call history list and then press the Enter soft key.
- **3.** Press \frown or \bigtriangledown to select the desired entry.
- 4. Press the **Send** soft key.

Adding a Contact to the Local Directory/Blacklist

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

- 1. Press the History soft key.
- 2. Select the desired call history list and then press the Enter soft key.
- **3.** Press \frown or \bigtriangledown to select the desired entry.
- 4. Press the Option soft key, and then select Add to Contact (or Add to Blacklist).

<	-All Calls	—1/44 >
Detail		0
Add to Contact		
Add to Blacklist		
Cancel		ОК

- 5. Enter the desired values in the corresponding fields.
- 6. Press the Save soft key to accept the change.

For more information on local directory and/or blacklist, refer to Local Directory on page 37 and/or Blacklist on page 45.

Deleting History Records

To delete an entry from the call history list:

- 1. Press the **History** soft key.
- 2. Select the desired call history list and then press the Enter soft key.
- **3.** Press \frown or \bigtriangledown to select the desired entry.
- 4. Press the **Delete** soft key.

To delete all entries from the call history list:

- 1. Press the **History** soft key.
- 2. Select the desired call history list and then press the Enter soft key.
- 3. Press the Option soft key, and then select Delete All.
- 4. Press the OK soft key.

The LCD screen prompts "Delete all the call records?".

< <u> </u>	All Calls	1/44>
N 10	wwwwwwWarningwwwwww	Í
> 10	Delete all the call records?	
> 10		
Cano	el 📘 🗼 📜 🕻	ĸ

5. Press the OK soft key to confirm the deletion or the Cancel soft key to cancel.

Disabling History Record

To disable history record feature via phone user interface:

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

History Setting	
1. History Record:	
Disabled	41
Back I ┥ I 🕨 I	Save

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

Search Source List in Dialing

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

Note LDAP list is hidden by default. To configure LDAP list, you need to enable LDAP feature in advance. For more information, contact your system administrator.

To configure search source list in dialing via web user interface:

- 1. Click on Directory->Setting.
- 2. In the Search Source List In Dialing block, select the desired list from the Disabled column and click →.

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

- 3. Repeat the step 2 to add more lists to the **Enabled** column.
- 4. To remove a list from the **Enabled** column, select the desired list and click [--]

To adjust the display order of the enabled lists, select the desired list and click f or .
 The LCD screen will display search results in the adjusted order.

Yealink CP860			Log Out Englsh(Englsh)					
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security
Local Directory Remote Phone Book Phone Call Info LDAP Multicast IP Setting	Direc	tory Disabled History Remote P Ch Source List In Disabled	hone Bool	Enabled Local Directory Favorite Director Enabled Local Directory History		Settings	NOTE Directory It provides ear frequently use Search Sourr Lailows the I sourmatically so from the search based on the and display calls lat tailows users placed calls lat is on the pre-	sy access to ed lists. Exe in Dialing P phone to search entries th source list exetered string, allist on the pre- it to view the when the phone allaing screen. lick here to get
		Confi	rm		Cancel			

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

Search source list in dialing is configurable via web user interface only.

To search for an entry in the enabled search source lists:

1. Press 🦿

Note

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

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

ф 1037:				1/1	
1					
Jim(1038)					
Send I	123	Delete	1	Cancel	Í

System Customizations

Logo Customization

You can upload your custom logo which will be shown on the idle screen.

To upload a custom logo via web user interface:

- 1. Click on Features->General Information.
- 2. Select Custom logo from the pull-down list of Use Logo.
- 3. Click Browse to locate the logo file from your local system.

Yealink CP860							Eng	Log Out lish(English) 🔻
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security
Forward&DND		General Information	DN				NOTE	
General Information		Call Waiting Call Waiting On Co	ode	Enabled	•			ones to receive a
Audio		Call Waiting Off Co Auto Redial	ode	Disabled			already an activ	call when there is ve call.
Intercom		Auto Redial Interv	ral (1~300s)	10			It allows IP pho automatically re	
Transfer Call Pickup			:				Key As Send	"*" as the send
Remote Control							key. Hotline	
Phone Lock		Use Logo Upload Logo		Custom logo	✓ file selected.		IP phone will au out the hotline	
ACD				Upload Can			speakerphone key.	iset, pressing the key or the line
SMS		Accept SIP Trust	Server Only	Disabled	~		Call Completio	
Action URL		Allow IP Call		Enabled	•		busy party and	
Notification Popups		Reboot in Talking Hide Feature Acce	ess Codes	Disabled Disabled	- -		available to rec	
		Display Method on	Dialing	User Name	•		You can cli more guides.	ck here to get
		Confir	m		Cancel			

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

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

The logo file format must be *.dob, contact your system administrator for more information.

Logo customization is configurable via web user interface only.

Programable Keys

Note

The CP860 IP conference phone supports 8 programable keys. You can assign predefined functionalities to programable keys. Programable keys allow you to quickly access features such as SMS. The key type N/A indicates that this programable key provides no functionality until configuration. Details will be introduced in the following. Some features listed in this chapter are not available for PSTN accounts. For more information, refer to Appendix C - Unavailable Features for PSTN on page 158.

To customize programable keys via web user interface:

1. Click on Dsskey->Programable Key.

2. Customize specific features for these keys.

Yealink CP860	Status	Account	Network	Dsskey	Features	Settings	Log Out English(English) • Directory Security
Programable Key	Key	Туре	Line	Value	Label	Extension	NOTE
	SoftKey 1	History	Local History 🔻				
	SoftKey 2	Directory •	N/A •				Programmable Keys Customizes the soft keys,
	SoftKey 3	DND	N/A 🔹				navigation keys and function keys.
	SoftKey 4	Menu 🔻	N/A 🔹				
	Up	History	Local History 🔻				You can click here to get more guides.
	Down	Local Favorite	N/A 🔻				
	ОК	Status 🔻	N/A 🔻				
	MUTE	N/A T	N/A •		, ,		
		Confirm	Cancel		Reset To D	efault	

- (Optional.) Enter the string that will appear on the LCD screen in the Label field.
 Label is configurable only when customizing Softkeys (1-4).
- 4. Click **Confirm** to accept the change.

You can click **Reset To Default** to reset custom settings to defaults.

Note You also can assign predefined functionalities to soft keys via phone user interface.

Programable key features are explained in the following subchapters in detail:

- Prefix
- Local Group
- XML Group
- LDAP
- Forward
- DND
- Phone Lock
- Directory
- Local Favorite
- Speed Dial
- Direct Pickup
- Group Pickup
- XML Browser
- SMS
- New SMS
- Zero Touch

For the features not listed above, refer to Basic Call Features on page 71 and Advanced Phone Features on page 98. For more information, contact your system administrator.

Prefix

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

Dependencies: Type (Prefix)

Label (key label displayed on the LCD screen)

Value (the prefix number)

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

Local Group

You can use this key feature to quickly access a contact group in the local directory. For more information, refer to Local Directory on page 37.

Dependencies: Type (Local Group)

Line (the group you want to access)

Label (key label displayed on the LCD screen)

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

XML Group

You can use this key feature to quickly access a remote group in your remote phone book. You should configure remote phone book in advance. For more information, refer to Remote Phone Book on page 48.

Dependencies: Type (XML Group)

Line (the remote group you want to access if the remote phone book is configured) *Label* (key label displayed on the LCD screen)

Usage: Press the XML Group key to access the remote group specified in the Line field.

LDAP

You can use this key feature to quickly access a LDAP search screen.

Dependencies: Type (LDAP)

Label (key label displayed on the LCD screen)

Usage:

- 1. Press the LDAP key to access the LDAP search screen.
- **2.** Enter a few continuous characters of the contact name or continuous numbers of the contact number using the keypad.

The contacts whose name or phone number matches the characters entered will appear on the LCD screen.

Forward

You can use this key feature to forward an incoming call to someone else. For more information, refer to Call Forward on page 82.

Dependencies: Type (Forward)

Label (key label displayed on the LCD screen)

Usage: Press the Forward key to enter the forward configuration screen.

DND

You can use this key feature to activate or deactivate DND. For more information, refer to Do Not Disturb (DND) on page 81.

Dependencies: Type (DND)

Label (key label displayed on the LCD screen)

Usage:

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

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

Phone Lock

You can use this key feature to immediately lock the keypad of your phone instead of long pressing $[\#_{mo}]$. For more information, refer to Phone Lock on page 30.

Dependencies: Type (Phone Lock)

Label (key label displayed on the LCD screen)

Usage: Press the **Phone Lock** key to immediately lock the keypad of your phone instead of long pressing $[\#_{mn}]$.

Directory

You can use this key feature to easily access frequently used lists. For more information, refer to Directory on page 36.

Dependencies: Type (Directory)

Label (key label displayed on the LCD screen)

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

Note The Directory key performs the same function as the Directory soft key when the phone is idle.

Local Favorite

You can use this key feature to easily access favorite directory. For more information, refer to Favorite Directory on page 46.

Dependencies: Type (Local Favorite)

Label (key label displayed on the LCD screen)

Usage: Press the Local Favorite key to immediately access favorite directory.

Speed Dial

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

Dependencies: Type (Speed Dial)

Label (key label displayed on the LCD screen)

Value (the number you want to dial out)

Usage: Press the Speed Dial key to dial out the number specified in the Value field.

Direct Pickup

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

Dependencies: Type (Direct Pickup)

Label (key label displayed on the LCD screen)

Value (the directed call pickup code followed by the target phone number)

Usage: Press the **Direct Pickup** key on your phone when the target phone number receives an incoming call. The call is then answered on your phone.

Group Pickup

You can use this key feature to answer incoming calls in a group that is associated with their own group.

Dependencies: Type (Group Pickup)

Label (key label displayed on the LCD screen)

Value (the group pickup feature code)

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

XML Browser

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

Dependencies: Type (XML Browser)

Label (key label displayed on the LCD screen) *Value* (the access URL for xml browser)

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

SMS

You can use this key feature to quick access text message. For more information, refer to Short Message Service (SMS) on page 123.

Dependencies: Type (SMS)

Label (key label displayed on the LCD screen)

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

New SMS

You can use this key feature to quick access the new text message. For more information, refer to Short Message Service (SMS) on page 123.

Dependencies: Type (New SMS)

Label (key label displayed on the LCD screen)

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

Zero Touch

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

Dependencies: Type (Zero Touch)

Label (key label displayed on the LCD screen)

Usage:

- 1. Press the Zero Touch key to access the zero touch screen.
- 2. Press the OK soft key in a few seconds.
- 3. Configure the network parameters in the corresponding fields.
- 4. Press the Next soft key.
- 5. Configure the auto provision parameters in the corresponding fields.
- 6. Press the OK soft key.

The phone will reboot to update configurations.

Account Registration

You can only register one SIP account (common account) or one Yealink Cloud account on the CP860 IP conference phone. Besides you can register one Yealink Cloud account on up to five Cloud endpoints.

If you have connected an expansion PSTN box CPN10 to the phone, a PSTN account is automatically registered on the phone. For more information, refer to Configuring the PSTN Account on page 131.

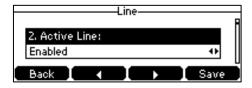
SIP Account

You can register only one SIP account on the phone.

To register a SIP account via phone user interface:

- 1. Press Menu->Settings->Advanced Settings (default password: admin) ->Accounts.
- 2. Press the Enter soft key.
- 3. Press the ◀ or ▶ soft key to select Common Account from the Type field.

4. Press the \blacktriangleleft or \blacktriangleright soft key to select **Enabled** from the **Active Line** field.



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

To disable a SIP account via phone user interface:

- 1. Press Menu->Settings->Advanced Settings (default password: admin) ->Accounts.
- 2. Press the Enter soft key.
- 3. Press the \blacktriangleleft or \blacktriangleright soft key to select **Common Account** from the **Type** field.
- 4. Press the ◀ or ▶ soft key to select **Disabled** from the **Account Status** field.
- 5. Press the Save soft key to accept the change or the Back soft key to cancel.

Registering a SIP account is configurable via web user interface at the path **Account->Register**.

Yealink Cloud Account

You can register only one Cloud account on the phone.

Configuring Cloud Feature

To register a Yealink Cloud account, you need to enable Cloud feature in advance.

To enable Cloud feature via web user interface:

1. Click on Features->General Information.

ealink CP860							Eng	lish(English)	
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security	
Forward&DND	(General Informati	ion				NOTE		
General		Call Waiting		Enabled	•		Call Waiting		
Information		Call Waiting On Co	ode				It allows IP ph		
Audio		Call Waiting Off C	ode				already an acti	call when there /e call.	
		Auto Redial		Disabled	•		Auto Redial		
Intercom		Auto Redial Interval (1~300s)		10		It allows IP phones to automatically redial a busy			
Transfer		Auto Redial Times (1~300)				number after the first attemp			
Call Pickup		Key As Send		#	#		Key As Send Assigns "#" or "*" as the send		
Remote Control							key.	as the sent	
Phone Lock							Hotline IP phone will a out the hotline	number when	
ACD							lifting the hand speakerphone key.		
SMS									
Action URL		Hide Feature Acc	ess Codes	Disabled	•			to monitor the	
ACTON ONE	_	Display Method on Dialing		User Name	•			establish a call party become	
Notification Popups		Cloud Enable		Enabled	•		available to rec	eive a call.	
							You can cl	ck here to get	
		Confi	rm		Cancel		more guides.	Ŭ	

2. Select Enabled from the pull-down list of Cloud Enable.

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

Note Cloud feature is configurable via web user interface only.

Yealink Cloud Account Registration

Two methods of registering a Yealink Cloud account are:

- PIN code: This method uses the user's PIN code to register the Yealink Cloud account. The PIN code consists of 9 numbers.
- Account: This method uses the user's username (Cloud number) and password to register the Yealink Cloud account.

You can obtain the Yealink Cloud account information from your Cloud enterprise administrator.

PIN Code

To register a Yealink Cloud account by using the PIN code:

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

3. Press the ◀ or ▶ soft key to select **Yealink Cloud** from the **Type** field.

Line	
1. Type:	
Yealink Cloud	•
Back 🖌 🖌 🕨	Login

- 4. Press the ◀ or ▶ soft key to select **PIN Code** from the **Login Type** field.
- 5. Enter the PIN code (9-digit) in the **PIN Code** field.

		-Line			
					П
3. PIN Cod	e:				
221795506					
2217 55566					
Back I	123		Delete	Login	

6. Press the Login soft key.

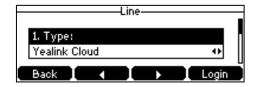
Note If you fail to register a Yealink Cloud account by using PIN code, you can re-enter the PIN code according to the prompt or contact your Cloud enterprise administrator.

The PIN code can be used once only. You can contact your Cloud enterprise administrator to get a new PIN code.

Account

To register a Yealink Cloud account by using an account:

- 1. Press Menu->Settings->Advanced Settings (default password: admin) ->Accounts.
- 2. Press the Enter soft key.
- 3. Press the \blacktriangleleft or \blacktriangleright soft key to select Yealink Cloud from the Type field.



- 4. Press the **d** or **b** soft key to select **Account** from the **Login Type** field.
- 5. Enter the username (Cloud number) in the User Name field.
- 6. Enter the password in the **Password** field.

	Line		
4. Password:			
Back 2	aB	Delete	Login

(Optional.) Press the ◀ or ▶ soft key to select the desired value from the Save Password field.

- 8. Press the Login soft key.
- **Note** If you fail to register a Yealink Cloud account by using an account, you can re-enter the account information according to the prompt or contact your Cloud enterprise administrator.

Yealink Cloud Account Exit

To exit the Yealink Cloud account:

- 1. Press Menu->Settings->Advanced Settings (default password: admin) ->Accounts.
- 2. Select the Yealink Cloud account and then press the Enter soft key.
- 3. Press the \blacktriangleleft or \blacktriangleright soft key to select **Yealink Cloud** from the **Type** field.
- 4. Press the Logout soft key.

The LCD screen prompts the following warning:



5. Press the OK soft key.

Dial Plan

Dial plan is a string of characters that governs the way your CP860 IP conference phone processes the inputs received from your phone keypad. The CP860 IP conference phone supports the following dial plan features:

- Replace Rule
- Dial Now
- Area Code
- Block Out

The basic expression syntaxes you need to know:

•	The dot "." can be used as a placeholder or multiple placeholders for any character. Example: "12." would match "12 3 ", "12 34 ", "12 345 ", "12 abc ", etc.
x	An "x" can be used as a placeholder for any character. Example: "12x" would match "12 1 ", "12 2 ", "12 3 ", "12 a ", etc.
-	Numeric ranges are allowed within the brackets: Digit "-" Digit. Example: "[5-7]" would match the number" 5 ", " 6 "or " 7 ".
[]	The square brackets "[]" can be used as a placeholder for a single character

	which matches any of a set of characters. Example: "91[5-7]1234" would match "91 5 1234", "91 6 1234", "91 7 1234".
0	The parentheses "()" can be used to group together patterns, for instance, to logically combine two or more patterns. Example: "([1-9])([2-7])3" would match " 92 3", " 15 3", " 77 3",etc.
\$	The "\$" should be followed by the sequence number of a parenthesis. The "\$" plus the sequence number means the whole character or characters placed in the parenthesis. The number directs to the right parenthesis when there are more than one. Example: A replace rule configuration: Prefix: "9([5-7])(.)", Replace: "5\$2". When you enter "96123" to dial out on your phone, the number will be replaced as "5 123 " and then dialed out. "\$2" means the characters in the second parenthesis, that is, "123".

Note

The IP phone supports a new dial plan mechanism – digit map. Digit maps are defined by a single string or a list of strings. If a number you dial matches any string of a digit map, the call is automatically placed.

Note that if digit map feature is enabled, the old dial plan rules (described in this chapter) will be ignored. For more information, contact your system administrator.

Replace Rule

You can configure one or more replace rules (up to 100) to remove the specified string and replace it with another string. You can configure a pattern with wildcards (expression syntax refer to 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 **Send** soft key.

To add a replace rule via web user interface:

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

3. Enter the string (e.g., 1234567) in the Replace field.

Veelink					Log Out English(English) 🔻
Yealink CP860	Status	Account	Dsskey Features	Settings	Directory Security
Preference	Replace Rul	e Dial Now Area Code Bloc	k Out		NOTE
Time & Date	Index	Prefix	Replace		Replace Rule: An alternative
Call Display	1				string that replaces the entered numbers.
	2				Dial-now: Automatically dial out the entered numbers.
Upgrade	3				Area Code:Automatically add the area code before the
Auto Provision	4				numbers when dialing. Block Out: It prevents users
	5				from dialing out specific numbers.
Configuration	6				".":represents any string.
Dial Plan	7				"x":represents any string. "x":represents any character. "-":match a range of characters
Voice	8				within the brackets.
VOICE	9				",":a separator within the bracket.
Ring	10				"[]":a character matches any of character sets.
Tones					"()":combines two or more patterns. "\$":followed by the sequence number of a parenthesis means
Softkey Layout	Pre	fix 1	Replace 1234567		the characters placed in the parenthesis.
TR069 Voice Monitoring		Add	Edit Del		You can click here to get more guides.

4. Click Add to add the replace rule.

When you enter the number "1" using the keypad and then press the **Send** soft key, the phone will dial out "1234567" instead.

To edit a replace rule via web user interface:

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

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

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

Note Replace rule is configurable via web user interface only.

Dial Now

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

To add a dial now rule via web user interface:

1. Click on Settings->Dial Plan->Dial Now.

2. Enter the desired value (e.g., 2xx) in the Rule field.

alink cp860	Status	Account	Network	Dsskey	Features	Settings	Directory Security
Preference	Replace Rul	e Dial Now A	rea Code Block	Out			NOTE
Time & Date	Index		Dial No	w Rule			Replace Rule: An alternative
	1						string that replaces the enter numbers.
Call Display	2						Dial-now:Automatically dial o the entered numbers.
Upgrade	3						Area Code:Automatically add
	4						the area code before the numbers when dialing.
Auto Provision	5						Block Out: It prevents users from dialing out specific
Configuration	6						numbers.
Dial Plan	7						".":represents any string. "x":represents any character.
	8						"-":match a range of character within the brackets.
Voice	9						",":a separator within the bracket.
Ring	10						"[]":a character matches any
	10						character sets. "()":combines two or more
Tones							patterns. "\$":followed by the sequence
Softkey Layout							number of a parenthesis mean the characters placed in the
			Rule 2xx				parenthesis.

3. Click Add to add the dial now rule.

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

Note You can also edit or delete the dial now rule, refer to Replace Rule on page 63 for more information.

Dial now rule is configurable via web user interface only.

Time Out for Dial Now Rule

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

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

- 1. Click on Features->General Information.
- 2. Enter the time between 0 and 14 (seconds) in the Time Out for Dial Now Rule field.

ealink cp860							Eng	Log O Jish(English)
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security
Forward&DND	(General Informati	on				NOTE	
General Information		Call Waiting Call Waiting On Co	ode	Enabled	•			ones to receive call when there
Audio		Call Waiting Off Co	ode				already an activ	
Intercom		Auto Redial Auto Redial Inten	ral (1~300s)	Disabled	¥		Auto Redial It allows IP pho automatically re	
Transfer		Auto Redial Times	(1~300)	10			Key As Send	ne nisc accempt
Call Pickup		Key As Send		#	Ŧ		Assigns "#" or key.	"*" as the send
Remote Control		Reserve # in User	Name	Enabled	¥		Hotline	utomatically dial
Phone Lock		Hotline Number					out the hotline lifting the hand	e number when lset, pressing th
ACD		Hotline Delay(0~1		4			speakerphone key.	key or the line
SMS		Busy Tone Delay		0	T		Call Completi It allows users	on to monitor the
		Return Code Whe		486 (Busy Here 480 (Temporari			busy party and establish a call when the busy party become	
Action URL		Call Completion		Disabled	v Ullava .		available to rec	eive a call.
Notification Popups		Feature Key Sync	hronization	Disabled	•		You can click here to more guides.	
		Time Out for Dial	Now Rule	1				
	L	RFC 2543 Hold		Disabled	•			

The default value is "1".

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

Note

Time out for dial now rule is configurable via web user interface only.

Area Code

Area codes are also known as Numbering Plan Areas (NPAs). They usually indicate geographical areas in a country. This feature is necessary when dialing a phone number outside the code area. For example, an area code is configured as "Code: 0592, Min Length: 4, Max Length: 11". When you dial out the number "56789" (the length of the number is between 4 and 11), the phone will add the area code and dial out the number "059256789". You can only configure one area code rule on your phone.

To configure the area code and lengths 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.

Yealink CP860	Status Account Network Dsskey Features Settings	Log Out English(English) • Directory Security
Preference	Replace Rule Dial Now Area Code Block Out	NOTE
Time & Date	Code 0592	Replace Rule: An alternative string that replaces the entered
Call Display	Min Length (1-15) 1	numbers. Dial-now:Automatically dial out
Upgrade	Max Length (1-15) 15	the entered numbers. Area Code:Automatically add the area code before the
Auto Provision	Confirm Cancel	numbers when dialing. Block Out:It prevents users
Configuration		from dialing out specific numbers.
Dial Plan		".":represents any string. "x":represents any character. "-":match a range of characters

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

Note The default value of minimum and maximum length is 1 and 15 respectively.

Area code is configurable via web user interface only.

Block Out

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

To add a block out number via web user interface:

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

Yealink CP860	Status Account Network Dsskey Features Setti	Log Out English(English) v Directory Security
Preference	Replace Rule Dial Now Area Code Block Out	NOTE
Time & Date	BlockOut Number1 6543	Replace Rule: An alternative string that replaces the entered
Call Display	BlockOut Number2	numbers. Dial-now:Automatically dial out
Upgrade	BlockOut Number3	the entered numbers. Area Code:Automatically add the area code before the
Auto Provision	BlockOut Number4	numbers when dialing. Block Out:It prevents users
Configuration	BlockOut Number5 BlockOut Number6	from dialing out specific numbers.
Dial Plan	BlockOut Number7	".":represents any string. "x":represents any character. "-":match a range of characters
Voice	BlockOut Number8	within the brackets. ",":a separator within the
Ring	BlockOut Number9 BlockOut Number10	bracket. "[]":a character matches any of character sets. "()":combines two or more
Tones	Confirm	patterns. "\$":followed by the sequence

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

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

Emergency Number

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

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

Lock on page 30.

Note

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

The IP phone also supports the emergency dialplan, which allows users to make emergency calls if the phone is locked or unregistered. For more information, contact your system administrator.

To specify emergency numbers via web user interface:

- 1. Click on Features->Phone Lock.
- 2. Enter the emergency number in the Emergency field.

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

Yealink CP860	Status Account Network	Dsskey Features	Settings	Log Out English(English) • Directory Security
Forward&DND General Information Audio Intercom Transfer Call Pickup Remote Control Phone Lock	Phone Lock Enable Phone Lock Type Phone Unlock PIN(0~15 Digit) Phone Lock Time Out(0~3600s) Emergency Confirm	Disabled • All Keys • ••••••• • 0 • 112,911,110 Cancel		NOTE Phone Lock It is used to lock the IP phone to prevent if from unauthorized use. Once the IP phone is locked, a user must enter the password to unlock it. IP phones offer three types of phone lock Menu Key, Function Keys and All Keys. The IP phone will not be locked middely after the phone lock type is configured. Quot configured.

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

Note

Emergency number is configurable via web user interface only.

Live Dialpad

You can enable live dialpad on the CP860 IP conference phone, which enables the IP phone to automatically dial out a phone number without the pressing of the send key. You can also configure a delay, where the phone will dial out the phone number automatically after the designated period of time.

To enable live dialpad via web user interface:

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

Yealink	Status Account Network	k Dsskey Features	Log Out English(English) - Settings Directory Security
Preference	Live Dialpad Inter Digit Time(1~14s)	Enabled •	NOTE
Time & Date	Backlight Time(seconds)	30s 👻	Live Dialpad It allows IP phones to
Call Display	Contrast	6 🗸	automatically dial out the entered phone number after a
Upgrade	Watch Dog	Disabled 👻	specified period of time.
Auto Provision	Ring Type	Ring1.wav -	Backlight Specifies the brightness of the
Configuration	Upload Ringtone	Browse No file selected. Upload Cancel	LCD screen display. Contrast Specifies the contrast of the
Dial Plan	Confirm	Cancel	LCD screen display.

The default delay time is 4.

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

Note Live dialpad is configurable via web user interface only.

Hotline

You can dial a hotline number immediately upon pressing the off-hook key. You can also configure a delay, where the phone will dial out the hotline number automatically after the designated period of time.

To configure the hotline number via phone user interface:

- 1. Press Menu->Features->Hot Line.
- 2. Enter the desired number in the Hotline Number field.

Hot Line	
1. Hotline Number:	
1008	l [
🛛 🗛 🚺 🕺 🚺 🚺 Back 🚺 🛛 123 🚺 Delete 🚺 Sav	e

3. Enter the delay time (in seconds) in the Hotline Delay field.

The valid values range from 0 to 10 (seconds) and the default value is "4".

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

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

Using Your Phone with SIP/Cloud Account

Basic Call Features

The CP860 IP conference 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 CP860 IP conference phone. Topics include:

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

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

Placing Calls

You can press the off-hook key either before or after entering the number to place a call. You can also dial an entry from local directory, favorite directory, remote phone book or call history. For more information, refer to Contact Management on page 35 and Call History Management on page 49.

The call duration and far-site's information (name and phone number) are visible on the LCD screen. In the figure below, the call to "Bob" (the phone number: 1050) has lasted 7 seconds.

цþ	Talking		1/1
		Bob	
		1050	
		00:07 HD	
٦	iran 🚺	Hold Conf	EndCall

To place a call:

Do one of the following:

Press 🥜 to obtain a dial tone.

Enter the desired number using the keypad.

Press $| \mathbf{r} |$, $(\mathbf{o}\mathbf{k})$, $| \mathbf{\#}_{\text{sevo}} |$ or the **Send** soft key.

- Enter the desired number using the keypad.
 - Press $| \mathbf{r} |$, $(\mathbf{o}\mathbf{K})$, $| \mathbf{\#}_{\mathsf{seno}} \rangle$ or the **Send** soft key.

The **#** key is configured as a send key by default. You can also set the ***** key as the send key, or set neither. For more information, refer to the Key As Send on page 30.

Note You can also dial using the SIP URI or IP address. To obtain the IP address of a phone, press the **OK** key when the phone is idle. The maximum length of SIP URI or IP address length is 32 characters. For example, SIP URI: 3606@sip.com, IP: 192.168.1.15 or 192*168*1*15.

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

The CP860 IP conference phone can handle multiple calls at a time. However, only one active call (the call that has audio associated with it) can be in progress at any given time.

To place multiple calls:

You can have more than one call on your CP860 IP conference phone. To place a new call during an active call, do one of the following:

Press 🥜 . The active call is placed on hold.

Enter the desired number using the keypad.

- Press $| (\circ K) , | \#_{\text{sevo}}$ or the **Send** soft key.
- Press the **Hold** soft key to place the original call on hold.

Press the **NewCall** soft key.

Enter the desired number using the keypad.

Press $| \mathbf{c} |$, $(\mathbf{o}\mathbf{K})$, $| \mathbf{\#}_{sso} |$ or the **Send** soft key.

You can press \frown or \bigtriangledown to switch between calls, and then press the **Resume** soft key to retrieve the desired call.

Answering Calls

You can answer a call no matter whether you are in another call or not. If you want to answer a new incoming call when in another call, ensure that call waiting has been enabled. For more information on call waiting, refer to Call Waiting on page 87.

You can ignore incoming calls by pressing the **Reject** soft key or on-hook key. You can also activate Do Not Disturb mode to ignore the incoming calls without ringing on your phone. For more information, refer to Do Not Disturb (DND) on page 81.

You can forward incoming calls to someone else by pressing the **FWD** soft key. For more information, refer to Call Forward on page 82.

Answering When Not in Another Call

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

To answer a call:

Do one of the following:

- Press 🦿 .
- Press the Answer soft key.
- Press (ок).

Answering When in Another Call

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

- Press the **Answer** soft key.

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

• Press \mathbf{r} to access the new call.

Press $| \mathbf{r} |$, $(\mathbf{o}\mathbf{k})$ or the **Answer** soft key.

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

Note

Ending Calls

To end a call:

Do one of the following:

- Press the EndCall soft key.
- Press 🗖

Note

To end a call placed on hold, you can press the **EndCall** soft key to end the call directly, or press the **Resume** soft key to resume the call 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 $[\ column column]$ when the phone is idle.
- **2.** Press \frown or \bigtriangledown to select the desired entry from the placed calls list, and then
 - press \bigcirc , , # or the **Send** soft key.

Recent Call In Dialing

To view the placed calls list when the phone is on the dialing screen, you should enable recent call in dialing in advance.

To enable recent call in dialing via web user interface:

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

Yealink CP860							Er	Log Out nglish(English) 🔻
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security
Local Directory	Dire	ctory					NOTE	
Remote Phone Book		Disabled History	A	Enabled Local Directory	*		Directory It provides ea frequently use	isy access to ed lists.
Phone Call Info		LDAP Remote F	Phone Bool →	Favorite Director	γ t		Search Sour	ce in Dialing
LDAP			-				automatically from the sear	search entries
Multicast IP			-		L.			sults on the pre-
Setting			*		*			In Dialing s to view the t when the phone dialing screen.
	Sear	ch Source List In	Dialing					lick here to get
		Disabled	Phonebook 🔺	Enabled Local Directory			more guides.	
		LDAP	PHONEDOUK A	History				
			-		1			
			-		1			
			-		.			
		Recent Ca	all In Dialing Enal	oled	•			
		Conf	ìrm		Cancel			

2. Select Enabled from the pull-down list of Recent Call In Dialing.

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

Note

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

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

1. Press 🥜

The LCD screen displays the placed calls list.

nþ 1037:			1/1
Bob(1039)			П
Directory	123		Cancel

You can also enter a few continuous characters of the contact name or continuous numbers of the contact number (office, mobile or other number) to search from placed calls list.

nþ 1037:			1/1	
104				
Jessica(1040)			-
Send I	123	Delete	Cancel	

Auto Answer

You can enable auto answer to automatically answer an incoming call. You can also enable auto answer mute to mute the local microphone when an incoming call is answered automatically.

To configure auto answer and auto answer mute via phone user interface:

- 1. Press Menu->Features->Auto Answer.
- Press → to select Status, and then press the < or > soft key to select Enabled from the Status field.

	Aut	o Ans	wer—		Г
2. Status:					
Enabled				4	
Back	•		Þ	Save	

- 3. Press the \triangleleft or \triangleright soft key to select **Enabled** from the **Auto Answer Mute** field.
- 4. Press the Save soft key to accept the change or the Back soft key to cancel.

The icon AA appears on the LCD screen.



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

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

Auto Redial

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

To configure auto redial via phone user interface:

- 1. Press Menu->Features->Auto Redial.
- 2. Press the ◀ or ▶ soft key to select **Enabled** from the **Auto Redial** field.

Auto Redial	
nato nealai	
1. Auto Redial:	
Enabled	40
Back 📘 ┥ 📘 🕨	Save .

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

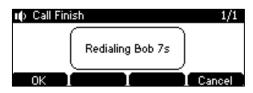
Auto redial is configurable via web user interface at the path Features->General Information.

To use auto redial:

II() Call Finish 1/1 Auto Redial? OK Cancel

1. Press the OK soft key to activate auto redial.

The LCD screen prompts the following:



2. Wait for the designated period of time or press the **OK** soft key to redial the phone number.

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

Call Completion

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

To configure call completion via phone user interface:

- 1. Press Menu->Features->Call Completion.
- 2. Press the \blacktriangleleft or \blacktriangleright soft key to select **Enabled** from the **Call Completion** field.

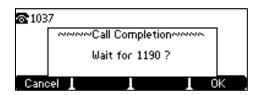
Call Completion	
1. Call Completion:	
Enabled	4
Back I 🖌 I 🕨	Save .

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

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

To use call completion:

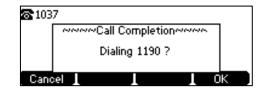
When the called party is busy, the following prompt will appear on the LCD screen of the phone:



When the called party is busy, the LCD screen prompts the following:

1. Press the **OK** soft key, the phone returns to the idle screen and call completion is activated.

When the called party becomes idle, the following prompt will appear on the LCD screen of the phone:



2. Press the **OK** soft key to redial the number.

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

ReCall

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

To configure a recall key via web user interface:

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

Yealink CP860							Log Out English(English)	
	Status	Account	Network	Dsskey	Features	Settings	Directory Security	
Programable Key	Кеу	Туре	Line	Value	Label	Extension	NOTE	
	SoftKey 1	History	Local History 🔻					
	SoftKey 2	Directory	• N/A •				Programmable Keys Customizes the soft keys,	
	SoftKey 3	DND	• N/A •				navigation keys and function keys.	
	SoftKey 4	Menu	N/A					
	Up	History	▼ Local History ▼				You can click here to get more guides.	
	Down	Local Favorite	• N/A •					
	ок	Status	N/A					
	MUTE	ReCall	N/A T				7	
		Confirm	Cancel		Reset To De	fault		

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

A recall key is configurable via web user interface only.

Call Mute

Note

You can mute the local microphone during an active call so that the other party cannot hear you, but you can still hear the other party. You can also mute the microphone while dialing so that the other party cannot hear you when the call is set up. It helps prevent the other party from hearing the informal discussion when auto answer is enabled on his/her phone.

Keep Mute

Normally, the mute feature is deactivated when the active call ends. Keep mute feature enables you to make the mute state of your phone persist across calls. When keep mute is enabled and you press the MUTE key, the phone stays in the mute state until you un-mute the microphone or until the phone restarts. It helps prevent the other party from hearing the noise coming from your room when auto answer is enabled on your phone. When you mute the phone in an idle state or any other states, the icon 32 appears on the status bar.



Note

Keep mute feature should be pre-configured by your system administrator.

Mute a Call

To mute a call:

1. Press [🐐 during an active call.

LED indicators illuminate solid red. The LCD screen indicates that the call is now muted.

цþ	Talking			1/1
		B	ob	
			50	
		<u>∕</u> ⊉ N	1ute HD	÷
	ran I	Hold	Conf	EndCall

To un-mute a call:

1. Press 🛛 🧔 again to un-mute the call.

To mute a call if you use expansion microphones:

1. Press [*****] on the phone or tap ***** on the top of the expansion microphone during an active call.

LED indicators on the phone and the mute indicator LED on the expansion microphone illuminate solid red.

To un-mute a call:

1. Press on the phone or tap on the top of the expansion microphone again to un-mute the call.

Mute While Dialing

To mute the microphone while dialing:

1. Press 🐐 on the pre-dialing or dialing screen.

The call is muted automatically when set up successfully.

To un-mute the microphone while dialing:

1. Press 🐐 again on the pre-dialing or dialing screen.

Note You can also mute the microphone when the IP phone is ringing.

Call Hold/Resume

You can place an active call on hold. Only one active call can be in progress at any time. Other calls can be made and received while placing the original call on hold. When you place a call on hold, your IP PBX may play music on hold to the other party while waiting.

To place a call on hold:

1. Press the Hold soft key during a call.

LED indicators flash green. The LCD screen indicates that the call is on hold.

🕪 Talking	1/1
Bob	
1190	
🛈 Hold 🛛 HD	
Tran 🛛 Resume 🚺 NewCall 🚺 End0	all

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

To resume a held call:

1. Press the Resume soft key.

Multiple Calls on Hold:

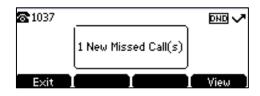
If multiple calls are placed on hold:

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

If more than one call is placed on hold, a numbered prompt appears on the LCD screen, for example "2/3", indicating that this is the second call out of three calls.

Do Not Disturb (DND)

You can use DND to reject incoming calls automatically on the phone. The prompt message "**n New Missed Call(s)**" ("n" indicates the number of missed calls, e.g., 1 New Missed Call(s)) will appear on the LCD screen, and callers will receive a busy message. All calls you receive while DND is activated are logged to your missed calls list.



Note

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

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

When DND feature is activated, the IP phone supports displaying a large DND icon on the idle screen. For more information, contact your system administrator.



To activate DND via phone user interface:

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

The icon **DND** appears on the status bar.



Note When DND and busy forward are activated, calls will be sent to the configured destination number. For more information on busy forward, refer to Call Forward on page 82.

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

To configure the DND authorized numbers via web user interface:

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

ealink cp860								glish(English)		
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security		
Forward&DND	F	orward					NOTE			
General Information	Forward Emergency Forward Authorized Numbers			Disabled	Disabled •			Call Forward It allows users to redirect an incoming call to a third party.		
Audio				1037	٣		Call Forward	Mode		
Intercom		Always Forward	1	◯ On ◉ Off			effective for t	nward feature is he IP phone.		
Transfer	Target On Code							Call forward feature nfigured for each or a		
Call Pickup	Off Code						Do Not Disturb (DND) It allows IP phones to ignore			
Remote Control	Busy Forward			◯ On . Off		incoming calls. DND Mode Phone: DND feature is effectiv for the IP phone.				
Phone Lock	Target									
ACD		On Code				Custom: DND feature can be configured for each or all accounts.				
SMS		Off Code		◯ On ● Off		You can click here to get				
Action URL		After Ring Tin		12 T			more guides.	lick here to get		
		Target	10(0-1200)	12						
Notification Popups		On Code								
		Off Code								
		ND								
		DND Emergency		Enabled	¥					
		DND Authorized I	Numbers	1002,2003						

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

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

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

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

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

To deactivate DND via phone user interface:

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

Call Forward

You can configure your phone to forward incoming calls to another party (static forwarding). You can also forward calls while your phone is ringing (dynamic forwarding).

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

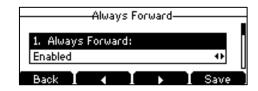
Static Forwarding

Three types of static forwarding are:

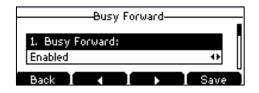
- Always Forward: Incoming calls are immediately forwarded.
- **Busy Forward**: Incoming calls are immediately forwarded if the phone is busy.
- No Answer Forward: Incoming calls are forwarded if not answered after a period of time.

To activate call forward via phone user interface:

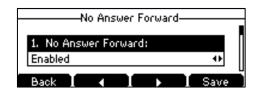
- 1. Press Menu->Features->Call Forward.
- **2.** Press \frown or \bigtriangledown to select the desired forwarding type, and then press the **Enter** soft key.
- **3.** Depending on your selection:
 - a) If you select Always Forward:
 - 1) Press the \blacktriangleleft or \blacktriangleright soft key to select **Enabled** from the **Always Forward** 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.
- b) If you select **Busy Forward**:
 - 1) Press the **d** or **b** soft key to select **Enabled** from the **Busy Forward** field.



- 2) Enter the destination number you want to forward all incoming calls to when the phone is busy in the **Forward to** field.
- (Optional.) Enter the busy forward on code or off code respectively in the On Code or Off Code field.
- c) If you select No Answer Forward:
 - 1) Press the **◄** or **▶** soft key to select **Enabled** from the **No Answer Forward** field.



2) Enter the destination number you want to forward all unanswered incoming calls

to in the Forward to field.

 Press the ◀ or ▶ soft key to select the ring time to wait before forwarding from the After Ring Time field.

The default ring time is 12 seconds.

- 4) (Optional.) Enter the no answer forward on code or off code respectively in theOn Code or Off Code field.
- 4. Press the Save soft key to accept the change or the Back soft key to cancel.

The icon $rac{d}{c}$ on the idle screen indicates that the call forward is activated.



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

Note

You can also enter the SIP URL or IP address in the **Forward to** field. For more information on using the SIP URL or IP address, refer to Placing Calls on page 72.

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

To configure the forward authorized numbers via web user interface:

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

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

Yealink CP860	Status Account Network Dsskey Features Settings	Log Out English(English) • Directory Security
Forward&DND	Forward	NOTE
Comment	Forward Emergency Enabled	
General Information	Forward Authorized Numbers 1002,1003	Call Forward It allows users to redirect an incoming call to a third party.
Audio	Account 1037	Call Forward Mode
	Always Forward 💿 On 💿 Off	Phone: Call forward feature is effective for the IP phone.
Intercom	Target 2	Custom: Call forward feature can be configured for each or all
Transfer	On Code	accounts.
Call Pickup	Off Code	Do Not Disturb (DND) It allows IP phones to ignore

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

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

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

To deactivate call forward via phone user interface:

- 1. Press Menu->Features->Call Forward.
- 2. Press \frown or \bigtriangledown to select the desired forwarding type, and then press the **Enter** soft key.
- 3. Press the \blacktriangleleft or \blacktriangleright soft key to select **Disabled** to deactivate the call forward.
- 4. Press the **Save** soft key to accept the change.

Dynamic Forwarding

To forward an incoming call to another party:

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

Forward to:			
1040			
Jessica(1040)			
Send I	123 I	Delete	Cancel

3. Press $| \mathcal{C} |$, $(\mathbf{o}\mathbf{K})$, $\#_{\text{see0}}$ or the **Send** soft key.

The LCD screen prompts a call forward message.

8 1037	
	Forward to: Jessica
Exit	View

Call Transfer

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

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

To perform a blind transfer:

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

🕪 Transfer	to:		2/2
1040			
Jessica(1040))		
Tran I	123	Delete	Cancel

3. Press the Tran soft key 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 the **Tran** soft key during a call.
- **2.** Do one of the following:
 - Enter the number you want to transfer the call to.
 - Press the **Directory** soft key, and then select **Local Directory**. Select the desired group and search for the contact (Directory should be configured in advance. Refer to Directory on page 36 for more information).
 - Press the **Directory** soft key, and then select **Favorite Directory**. Select the desired contact and search for the contact (Directory should be configured in advance. Refer to Directory on page 36 for more information).
 - Press the **Directory** soft key, and then select **History**. Select the desired list and then press or view of to select the entry (Directory should be configured in advance. Refer to Directory on page 36 for more information).
 - Press the **Directory** soft key, and then select **Remote Phone Book**. Select the desired group and search for the contact (Directory and Remote Phone Book should be configured in advance. Refer to Directory on page 36 and Remote Phone Book on page 48 for more information).
- **3.** Press $| \ (\circ \mathbf{K})$ or $| \ \#_{\text{SEND}}$ to dial out.
- 4. Press the Tran soft key to complete the transfer when receiving ringback.

To perform an attended transfer:

- 1. Press the Tran soft key during a call.
- 2. Do one of the following:
 - Enter the number you want to transfer the call to.
 - Press the **Directory** soft key, and then select **Local Directory**. Select the desired group and search for the contact (Directory should be configured in advance. Refer to Directory on page 36 for more information).
 - Press the **Directory** soft key, and then select **Favorite Directory**. Select the desired contact and search for the contact (Directory should be configured in advance. Refer to Directory on page 36 for more information).
 - Press the **Directory** soft key, and then select **History**. Select the desired list and then press in or intervention or intervention or intervention of the select the entry (Directory should be configured in advance.
 Refer to Directory on page 36 for more information).
 - Press the **Directory** soft key, and then select **Remote Phone Book**. Select the desired group and search for the contact (Directory and Remote Phone Book should be configured in advance. Refer to Directory on page 36 and Remote Phone Book on page 48 for more information).
- 3. Press C , OK or #sso to dial out.

4. After the party answers the call, press the **Tran** soft key to complete the transfer.

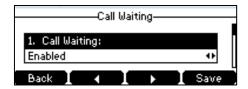
You can cancel the transfer before the call is connected by pressing the Cancel soft key.

Call Waiting

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

To configure call waiting via phone user interface:

- 1. Press Menu->Features->Call Waiting.
- 2. Press the < or > soft key to select Enabled from the Call Waiting field.



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

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

Conference

You can create a conference with other four parties using the phone's local conference. You can view and manage each participant in the conference call. The CP860 IP conference phone also supports network conference. The network conference URI should be predefined, contact your system administrator for more information.

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

Local Conference

The CP860 IP conference phone supports up to five parties (including yourself) in a conference call. The default conference type is Local Conference.

To set up a three-way local conference call:

1. Place a call to the first party.

2. When the first party answers the call, press the **Conf** soft key to place a new call.

The active call is placed on hold.

- **3.** Enter the number of the second party and then press \checkmark , $(\circ\kappa)$, $\#_{wwo}$ or the **Send** soft key.
- **4.** When the second party answers the call, press the **Conf** soft key again to join all parties in the conference.



You can create a conference between an active call and a call on hold by pressing the the **Conf** soft key.

To set up a conference call with an active call and a call on hold:

- **1.** Establish two calls on the phone.
- **2.** Press \frown or \bigtriangledown to select an active call.
- 3. Press the **Conf** soft key to join the two calls in the conference.

To join more parties in an established conference call:

- 1. Press the **Conf** soft key after the conference call is established.
- **2.** Enter the number of the new party and then press $| \mathcal{C} |$, (αK) , $| \mathcal{H}_{mo}$ or the **Send** soft key.
- 3. Press the **Conf** soft key when the party answers.
- 4. Repeat steps 1 to 3 to join more parties in the established conference call.

During the conference call, you can do the following:

- Press the **Hold** soft key to place the conference on hold.
- Press the **Conf** soft key to join more parties in an established conference call.
- Press the **Manage** soft key, and then press () or \bigvee to select the desired party:
 - Press the **FarMute** soft key to mute the party. The muted party can hear everyone, but no one can hear the muted party.
 - Press the **Remove** soft key to remove the selected party from the conference call.
 - Press the All Split soft key to split the conference call into individual calls on hold.
 - Press the **Back** soft key to return to the previous screen.
- Press 1/2 to mute the conference call, all other participants can hear each other, but they cannot hear you.
- Press
 or the EndCall soft key to drop the conference call.

Network Conference

You can use network conference on the CP860 IP conference 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.
- Remove a specific conference party.

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

To configure network conference via web user interface:

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

Yealink CP860	Status Account Network	Dsskey Features Settings	Log Out English(English) • Directory Security
Register Basic Codec Advanced	Keep Alive Type Keep Alive Interval(Seconds) RPort Subscribe Period(Seconds) DTMF Type DTMF Info Type DTMF Payload Type(96~127) Conference Type Conference URI ACD Subscribe Period(120~3600s) Early Media Unregister When Reboot VQ RTCP-XR Collector Name VQ RTCP-XR Collector Name VQ RTCP-XR Collector Port	Default 30 30 Disabled V 1800 RFC2833 V DTMF-Relay V 101 Network Conference V conference@example.com 3600 Disabled V Disabled V Cancel	NOTE Title Title Stand sent from the IP phone to the network, which is generated when pressing the IP phone's keypad during a call. Session Timer It allows a periodic refresh of SIP request, to determine whether a SIP session is still active. Busy Lamp Field/BLF List Montors a specific extension/a list of extensions for status changes on IP phones. Any IP phone can be used to originate or receive calls on the shared line. Network Conference Thoms multiple participants (more than three) to join in a call.

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

Note

Network conference is configurable via web user interface only.

To set up a network conference call:

- **1.** Place a call to the first party.
- 2. Press the **Conf** soft key to place a new call.

The active call is placed on hold.

3. Enter the number of the second party and then press

, $(o\kappa)$, $\#_{ssso}$ or the **Send** soft

C

key.

- **4.** When the second party answers the call, press the **Conf** soft key to add the second party to the conference.
- 5. Press the **Conf** soft key to place a new call.

The conference is placed on hold.

- **6.** Enter the number of the new party and then press [,], (ok), $(\#_{mon})$ or the **Send** soft key.
- **7.** When the new party answers the call, press the **Conf** soft key to add the new party to the conference.
- 8. Repeat steps 5 to 7 until you have added all intended parties.

The procedures to set up a network conference call on 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 an active call by pressing the **Park** soft key on the phone. If the call is parked successfully, there is a voice prompt confirming that the call was parked. You can retrieve the parked call by pressing the **Retrieve** soft key. If the parked call is not retrieved within a period of time defined by the system, the phone performing call park will receive the call back.

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

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

- **FAC mode**: park the call to the local extension or a desired extension through dialing the park code.
- **Transfer mode**: park the call to shared parking lot through performing a blind transfer to a call park number (call park code).

You can select to use any of above modes according to your server.

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

FAC Mode

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

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

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

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

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

Yealink CP860			Log Out English(English) ▼
	Status Account Network	Dsskey Features Settings	Directory Security
Forward&DND	Call Pickup Directed Call Pickup	Disabled v	NOTE
General Information	Directed Call Pickup Code		Directed Call Pickup Picks up an incoming call on a specific extension.
Audio	Group Call Pickup Group Call Pickup Code	Disabled •	Directed Call Pickup Picks up incoming calls within a pre-defined group.
Intercom Transfer	Call Park Call Park Mode	FAC	You can configure directed/group call pickup feature for the IP phone.
Call Pickup	Call Park	Enabled	Visual Alert for BLF Pickup
Remote Control	Call Park Code Park Retrieve Code	*68	It allows the supervisor's phone to display a visual prompt when the monitored user receives an incoming call.
Phone Lock ACD	Confirm	Cancel	Audio Alert for BLF Pickup It allows the supervisor's phone to play an alert tone when the

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

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

To park a call in FAC mode:

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



Press (o_{κ}) , $(\#_{sso})$ or the **Park** soft key.

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



2. Do one of the following:

- **a)** If you want to park the call against the local extension.
 - **1)** Press **#**_{seno}.

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

- **b)** If you want to park the call against desired extension.
 - 1) Enter an extension (e.g., 4605) where you want to park the call.
 - **2)** Press (οκ) or *#*_{seno}.

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

To retrieve a parked call in FAC mode:

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

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



- 2. Follow the voice prompt, do one of the following:
 - Press $(\#_{\text{seed}})$ on the phone where the call is parked.
 - Enter the desired extension followed by # (e.g., 4605#) on any phone.

Transfer Mode

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

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

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

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

ealink cp860	Status Account Network	Dsskey Features	Log 0 English(English) Settings Directory Security
Forward&DND	Call Pickup		NOTE
General	Directed Call Pickup	Disabled v	Directed Call Pickup
Information	Directed Call Pickup Code		Picks up an incoming call on a specific extension.
Audio	Group Call Pickup	Disabled •	Directed Call Pickup
Intercom	Group Call Pickup Code		Picks up incoming calls within a pre-defined group.
	Call Park		You can configure
Transfer	Call Park Mode	Transfer 🔹	directed/group call pickup feature for the IP phone.
Call Pickup	Call Park	Enabled 🔹	Visual Alert for BLF Pickup
Remote Control	Call Park Code	*01	It allows the supervisor's phon to display a visual prompt when the monitored user receives at
Phone Lock	Park Retrieve Code	*11	incoming call.
ACD	Confirm	Cancel	Audio Alert for BLF Pickup It allows the supervisor's phon to play an alert tone when the

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

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

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

To park a call in Transfer mode:

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

		1/1
	_	
123	Delete	Cancel
	123	

Press $(\mathbf{o}\mathbf{K})$, $\mathbf{\#}_{\mathbf{m}\mathbf{o}\mathbf{o}\mathbf{N}}$ or the **Park** soft key. The call will be transferred to the shared parking lot.

- If the call park code is configured, the call will be directly transferred to the shared parking lot.
- **Note** For some servers, the system will return a specific call park retrieve number (park retrieve code) from which the call can be retrieved after parking successfully.

To retrieve a parked call in Transfer mode:

1. Do one of the following:

- If the park retrieve code is not configured, dial the park retrieve code (e.g., *11).

ιφ 101:			1/1
* 11			
••			
Send	123	Delete	Cancel

If the park retrieve code is configured, press the **Retrieve** soft key on the dialing screen.

The phone will retrieve the parked call from the shared parking lot.

Call Pickup

You can use call pickup to answer someone else's incoming call on your phone. The CP860 IP conference phone supports directed call pickup and group call pickup. Directed call pickup is used for picking up a call that is ringing at a target phone number. Group call pickup is used for picking up a call that is ringing at any phone number in a certain group. The pickup group should be predefined, contact your system administrator for more information.

You can pick up an incoming call by using the **DPickup/GPickup** soft key. To use call pickup, you need to configure the call pickup code beforehand via web user interface.

Note If there are many incoming calls at the same time, pressing the **GPickup** soft key on the phone will pick up the call that rings first.

Directed Call Pickup

To enable directed call pickup and configure the directed call pickup code via web user interface:

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

Yealink CP860		Log Out English(English) ▼					
	Status	Account	Network	Dsskey	Features	Settings	Directory Security
Forward&DND		Call Pickup					NOTE
General		Directed Call Picku	qu	Enabled	T		Directed Call Pickup
Information		Directed Call Pickup Code		*97			Picks up an incoming call on a specific extension.
Audio		Group Call Pickup		Disabled	٣		Directed Call Pickup
Intercom		Group Call Pickup	Code				Picks up incoming calls within a pre-defined group.
	(Call Park					You can configure
Transfer		Call Park Mode		Transfer	Transfer 🔹		directed/group call pickup feature for the IP phone.
Call Pickup		Call Park		Enabled	T		Visual Alert for BLF Pickup

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

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

1. Click on Account->Advanced.

ealink cpsoo							Er	Log (nglish(English)		
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security		
Register	Kee	p Alive Type		Default	~		NOTE			
Basic		p Alive Interval(Seco	onds)	30 Disabled			DTMF	sent from the IP		
Codec	RPort Subscribe Period(Seconds) DTMF Type			1800				It is the signal sent from the IP phone to the network, which is generated when pressing the If		
Advanced				RFC2833			phone's keypad during a call.			
	DTM	IF Info Type		DTMF-Relay	\sim		Session Time	r odic refresh of S		
	DTMF Payload Type(96~127)			101			sessions through a re-INVITE request, to determine whether			
				:			SIP session is s	still active.		
	Directed Call Pickup Code		de	*97			Busy Lamp Fi Monitors a special list of extension	cific extension/a		
	Group Call Pickup Code			*98			changes on IP phones.			
	Dist	inctive Ring Tones		Enabled	\checkmark		Shared Call A	ppopropco		
	Unn	egister When Reboo	t	Disabled	\checkmark		(SCA)/ Bridge Appearance (e Line		
	VQ	RTCP-XR Collector N	lame					to share a SIP I		
	VQ	RTCP-XR Collector A	ddress					used to originate		
	VQ	RTCP-XR Collector P	Port	5060			line.	on the shared		
		Confi	rm		Cancel		Network Con It allows multip (more than thr call.	ole participants		

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

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

To pick up a call directly:

1. Press 🦿

The **DPickup** soft key appears on the LCD screen (You may need to press the **More** soft key).

up 1037:	1/1
Cancel I DPickup I	[More]

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

The call is answered on your phone.

Group Call Pickup

To enable group call pickup and configure the group call pickup code via web user interface:

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

3. Enter the group call pickup code in the Group Call Pickup Code field.

ealink cp860			Log English(English)
	Status Account Network	Dsskey Features	Settings Directory Security
Forward&DND	Call Pickup		NOTE
General	Directed Call Pickup	Enabled 🔹	Directed Call Pickup
Information	Directed Call Pickup Code		Picks up an incoming call on a specific extension.
Audio	Group Call Pickup	Enabled 🔹	Directed Call Pickup
	Group Call Pickup Code	*98	Picks up incoming calls within pre-defined group.
Intercom	Call Park	You can configure	
Transfer	Call Park Mode	Transfer •	directed/group call pickup feature for the IP phone.
Call Pickup	Call Park	Enabled 🔹	Visual Alert for BLF Pickup It allows the supervisor's pho

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

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

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

Yealink CP860	Log Out English(English) •							
	Status	Account	Network	Dsskey	Features	Settings	Directory	Security
Register Basic Codec Advanced	Keep RPor Subs DTM DTM DTM Grou Direc Grou Disti	o Alive Type o Alive Interval(Sec	onds) ds) ~127) de it iame	Default 30 Disabled 1800 RFC2833 DTMF-Relay 101 DTMF-Relay 101		Settings	NOTE DTMF It is the signal phone to then a generated whe phone's keypar Session Time It allows a per It allows a per or several IP per phone can be to phone can be t	sent from the IP etwork, which is in pressing the IP d during a call. and the sentence of the sentence d during a call. and the sentence ermine whether a still active. and of the sentence it active sentence it active sentence phones. and the size of the sentence (BLA) to share a SIP line whones. Any IP
	VQ RTCP-XR Collector Port	Port	5060			s on the shared		
		Confi	rm		Cancel		Network Con It allows multip	ple participants

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

To pick up a call in the group:

1. Press *c* .

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

∎ ∲ 2003:			1/1
1			
I			
Bob(1050)			П
Directory	123	GPickup C	ancel

2. Press the **GPickup** soft key on your phone when a phone in the group receives an incoming call.

The call is answered on your phone.

Note The directed call pickup code and group call pickup code are predefined on the system server. Contact your system administrator for more information.

The call pickup code configured on a per-line basis takes precedence over that configured on a global basis.

Anonymous Call

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

Note

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

To configure anonymous call via phone user interface:

- 1. Press Menu->Features->Anonymous Call.
- Press → to select Local Anonymous and then press the < or > soft key to select
 Enabled from the Local Anonymous field.

Anonymous Call-	
·	R
2. Local Anonymous:	
Enabled	••
Back I 🖌 I 🕨 I	Save

 (Optional.) Press the ◀ or ► soft key to select the desired value from the Send Anonymous Code field.

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

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

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

To place an anonymous call:

1. Use the specific phone to place a call to phone B.

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

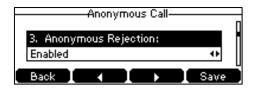


Anonymous Call Rejection

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

To configure anonymous call rejection via phone user interface:

- 1. Press Menu->Features->Anonymous Call.
- **2.** Press or v to scroll to the **Anonymous Rejection** field.
- 3. Press the ◀ or ▶ soft key to select **Enabled** from the **Anonymous Rejection** field.



4. (Optional.) Press in or it to select the desired value from the **Send Rejection Code** field.

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

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

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

Advanced Phone Features

This chapter provides operating instructions for the advanced features of the CP860 IP conference phone. Topics include:

- Recording Using a USB Flash Drive
- Intercom
- Multicast Paging
- Music on Hold
- Shared Call Appearance (SCA)

- Bridged Line Appearance (BLA)
- Messages
- Using PC or Mobile Device with the Conference Phone

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

Recording Using a USB Flash Drive

Call and Conference Recording

You can insert a USB flash drive into the USB port on your phone to manually record active calls and conferences, or automatically record once the call is set up. Recordings are stored in *.wav format. The filenames include a date & time stamp and the other party's number/IP address/name (or the first person's number/IP address/name you called), for example, 20161123-1630-1011 was created on November 23, 2016, at 16:30 and you have a call with 1011.

Note Recording feature is not available by default. For more information, contact your system administrator.

The size of a single recording file should be less than 2G.

When your phone is idle, and you insert a USB flash drive into the USB port on your phone, the phone will detect the USB flash drive and display the prompt message "USB Inserted" and the icon **G**---.



If the phone detects the USB flash drive successfully, the icon 🔫 will display on the idle screen.



Note Before recording any call, especially those involving PSTN, it is necessary to know about the rules and restrictions of any governing call-recording in the place where you are. It is also very important to have the consent of the person you are calling before recording the conversation.

The Start REC soft key controls the recording function, and is available:

When there are one or more calls connected to your phone

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

The Start REC soft key is not available when:

- There are no connected calls on your phone
- You place a new call
- The phone prompts you to answer an incoming call

The recording is not be paused when the following occurs:

- You place a call on hold
- You mute a call
- You set up a conference call
- You perform the attended or semi-attended transfer
- An incoming calls arrives on your phone

Call Recording

To record a call:

1. Press the More soft key and then press the Start REC soft key during a call.

If automatic recording feature is enabled, the call will be automatically recorded once it is set up.

The LCD screen prompts the approximate time remaining (depending on the free space of the USB flash drive), and displays the icon \bullet and recording duration.



If you press the **Hold** soft key while recording, only you are recorded. If you press while recording, only the callee is recorded.

If there is insufficient free space (less than 1 hour) on the USB flash drive during recording, the LCD screen prompts the following:



Press the **OK** soft key to return to the previous screen.

If there is insufficient free space (less than 2 minutes) on the USB flash drive when you press the **Start REC** soft key, or when the call is set up to automatically record, the recording is not started, and the LCD screen prompts the following:



If there is insufficient free space (less than 2 minutes) on the USB flash drive during recording, recording is automatically stopped, the LCD screen prompts the following:

🕪 Talking	1/1
Space is not enough Record file saved	
Tran Hold Conf	More

Note You can hold, transfer or set up a conference call while recording.

When you end a call while recording, the recording will be stopped and saved to USB flash drive automatically.

To stop recording while the phone records, do one of the following:

- Press the More soft key, and then press the Stop REC soft key.

The LCD screen prompts "Record file saved", and the recording icon and recording duration disappear.



- When there is only a call on the phone, press the **More** soft key and then press the **EndCall** soft key, or press

The LCD screen prompts "Record file saved", the recording icon and recording duration disappear, and the phone returns to the idle screen.

@ 1037		Å
	Record file saved	
History	Directory DND I Me	nu)

The recording will be stored as a new .wav file on the USB flash drive when the phone starts recording again.

Conference Recording

You can record conference calls in the same way as other calls with the following exceptions:

All conference participants are recorded while recording. If one of the participants presses the **Hold** soft key, only that participant is recorded. If one of the participants presses *(*)*, only that participant is not recorded.

Idle Recording

You can insert a USB flash drive into the USB port on your phone to record an important discussion when the phone is idle. Recordings are stored in *.wav format. The filenames include a date & time stamp and a keyword "idleREC", for example, 20161202-1043-idleREC was created on December 2, 2016, at 10:43 when the phone is idle.

Note

Idle recording is not available by default. For more information, contact your system administrator.

The size of a single recording file should be less than 2G.

To record a discussion when the phone is idle:

- 1. Press Menu->USB Record->Idle Record.
- 2. Press the Start REC soft key.

The LCD screen displays the icon
and recording duration.



During the recording, you can do the following:

- To pause the recording, press the Pause REC soft key. The Resume REC soft key appears on the LCD screen. The LCD screen displays the icon II and the duration stops counting.
 To continue the recording, press the Resume REC soft key.
- To stop the recording, press the Stop REC soft key. The LCD screen prompts "Record file saved".
- To go back to the previous screen, press the **Back** soft key. The phone returns to the USB Record screen. The recording continues.
- To record the remote party and local party, press the **Conf** soft key to place a new call.

Recorded Files Playback

If you insert a USB flash drive into the USB port on your phone, you can play back recorded files

on your phone.

You can also browse and delete the recorded files on the USB flash drive.

Note Playback can occur on either the phone itself or a computer using an application capable of playing .wav files.

To browse the recorded files:

1. Press Menu->USB Record->Browse Audio.

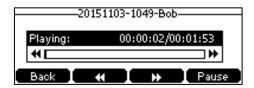
The LCD screen shows all recorded files on the USB flash drive. The recording duration and the size of the recorded file display after the filename.

Browse Audio
20161202-1009-idleRl 00:08:18(15.2MB)
20161123-1630-1011 00:00:04(149.4KB)
20161123-1628-1011 00:02:00(3.7MB)
Back Clear Delete Play

2. Press \frown or \bigtriangledown to scroll through the recorded files.

To play back a recording:

- 1. Press Menu->USB Record->Browse Audio.
- **2.** Press \frown or \bigvee to highlight the recording you want to play back.
- 3. Press the Play soft key.



During the recording playback, you can do the following:

To pause the playback, press the Pause soft key. The Play soft key appears on the LCD screen.

To continue the playback, press the **Play** soft key.

- To skip forward the playback, press the 🏓 soft key. Press once to skip forward 8 seconds.
- To rewind the playback, press the 🗲 soft key. Press once to rewind 8 seconds.
- To adjust the volume of the speakerphone, press (- - +).
- To stop the playback, press the **Back** soft key. The phone returns to the Browse Audio screen.

If you are playing back a recorded call and an incoming call arrives on your phone, the playback pauses and the phone rings. The playback will not continue until you press the **Play** soft key.

To delete a recorded file:

- 1. Press Menu->USB Record->Browse Audio.
- 2. Press or v to highlight the recording you want to delete.
- 3. Press the Delete soft key.

The LCD screen prompts the following warning:

	Browse Audio	
20161	~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~	в)
20161	Delete Focus Record ?	
20161	Delete Focus Necolu :	- в)
Cance		ОK

4. Press the OK soft key to confirm the deletion or the Cancel soft key to cancel.

You can also delete all recorded files by pressing the Clear soft key.

Intercom

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

Note

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

Outgoing Intercom Calls

To configure an intercom key via web user interface:

- 1. Click on Dsskey->Programable Key.
- 2. Select the desired programable key.
- 3. Select Intercom from the pull-down list of Type.
- 4. Enter the remote extension number in the Value field.

Yealink	Status	Account	Network	Dsskey	Features	Settings	Log Out English(English) V Directory Security
	Status	Account	Network	bookey	reatures	Securitys	Directory
Programable Key	Кеу	Туре	Line	Value	Label	Extension	NOTE
	SoftKey 1	History •	Local History 🔻				
	SoftKey 2	Directory •	N/A T				Programmable Keys Customizes the soft keys,
	SoftKey 3	DND •	N/A •	,			navigation keys and function keys.
	SoftKey 4	Menu 🔻	N/A 🔻				
	Up	History •	Local History 🔻	, 			You can click here to get more guides.
	Down	Local Favorite	N/A 🔻				
	ОК	Status 🔹	N/A •	,			
	MUTE	Intercom 🔻	Line 1 🔻	6008]
		Confirm	Cancel		Reset To De	efault	

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

Incoming Intercom Calls

The CP860 IP conference phone supports automatically to answer an incoming intercom call by default. The phone automatically plays a warning tone when it receives an incoming intercom call. In addition, you can enable the phone to mute the microphone when it automatically answers an incoming intercom call. You can also enable the phone to automatically answer an incoming intercom call while there is already an active call on the phone. The active call is then placed on hold.

Intercom features you need to know:

Intercom Feature	Description
Intercom Allow	Enable or disable the IP phone to answer an incoming
	intercom call.
Intercom Mute	Enable or disable the IP phone's microphone for intercom
Intercontinute	calls.
Intercom Tone	Enable or disable the IP phone to play a warning tone
Intercom Tone	when it receives an incoming intercom call.
	Enable or disable the IP phone to automatically answer an
Intercom Barge	incoming intercom call while there is already an active call
	on the phone.

Intercom Allow

You can enable or disable the phone to answer an incoming intercom call. If Intercom Allow is enabled, the phone will automatically answer an incoming intercom call. If Intercom Allow is disabled, the phone will handle an incoming intercom call like a normal call instead of directly rejecting it. Intercom Allow is enabled by default.

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

Intercom Mute

Note

You can mute or un-mute the phone's microphone for intercom calls automatically. If Intercom Mute is enabled, the microphone will be muted for intercom calls. If Intercom Mute is disabled, the microphone will work for intercom calls. Intercom Mute is disabled by default.

Intercom Tone

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

Intercom Barge

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

Note To enable the phone to receive a new incoming call when it has an active call, make sure that call waiting feature is enabled on the phone in advance. For more information, refer to Call Waiting on page 87.

To configure intercom features via phone user interface:

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

Intercom	
1. Intercom Allow:	
Enabled	•
Back 🖌 🖌 📘	Save

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

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

Using Intercom

To place an intercom call when the target phone is idle:

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

The called destination plays a warning tone and automatically answers the call in the hands-free (speakerphone) mode by default.

2. Press - or the EndCall soft key, to end the intercom call.

Multicast Paging

You can use multicast paging to quickly and easily broadcast time sensitive announcements to people who are listening to a specific multicast group and a specific channel. You can configure a paging key or paging list key on the phone, which allows you to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address(es) and channel(s) without involving SIP signaling. You can configure the phone to receive an RTP stream from pre-configured multicast listening address(es) and channel(s) without involving SIP signaling. You can specify up to 31 multicast listening addresses and channels.

Yealink IP phone supports the following 31 channels:

- **0**: You can broadcast audio to channel 0. Note that the Yealink IP phones running old firmware version (old paging mechanism) can be regarded as listening to channel 0. It is the default channel.
- **1 to 25**: You can broadcast audio to a specific channel. We recommend that you specify these channels when broadcasting with Polycom IP phones which have 25 channels you can listen to.
- **26 to 30**: You can broadcast audio to a specific channel. We recommend that you specify these channels when broadcasting with Yealink IP phones running new firmware version (new paging mechanism).

The IP phones will automatically ignore all incoming multicast paging calls on the different channel.

Sending RTP Stream

To configure a multicast paging key via web user interface:

- 1. Click on Dsskey->Programable Key.
- **2.** Select the desired programable key.
- 3. Select Paging from the pull-down list of Type.
- 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.
- 5. Enter the desired channel in the Extension field.

The valid channel ranges from 0 to 30.

ealink cps60			Log O English(English)				
	Status	Account	Network	Dsskey	Features	Settings	Directory Security
Programable Key	Кеу	Туре	Line	Value	Label	Extension	NOTE
	SoftKey 1	History •	Local History 🔻				
	SoftKey 2	Directory •	N/A T				Programmable Keys Customizes the soft keys,
	SoftKey 3	DND	N/A T				navigation keys and function keys.
	SoftKey 4	Menu 🔻	N/A 🔻				_
	Up	History •	Local History 🔻				You can click here to get more guides.
	Down	Directory •	N/A T				
	ОК	Status 🔻	N/A T				
	MUTE	Paging 🔻	N/A 🔻	224.5.6.20:10008		0	
		Confirm	Cancel		Reset To De	fault	

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

To configure a paging list key via web user interface:

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

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

ealink cps60	Log Ou English(English) ▼						
	Status	Account	Network	Dsskey	Features	Settings	Directory Security
Programable Key	Кеу	Туре	Line	Value	Label	Extension	NOTE
	SoftKey 1	History •	Local History 🔻				
	SoftKey 2	Directory •	N/A 🔻				Programmable Keys Customizes the soft keys,
	SoftKey 3	DND	N/A T				navigation keys and function keys.
	SoftKey 4	Menu 🔻	N/A 🔻				
	Up	History •	Local History 🔻				You can click here to get more guides.
	Down	Directory •	N/A				
	ок	Status •	N/A				
	MUTE	Paging List 🔻	N/A				
		Confirm	Cancel		Reset To D	efault	

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

To configure paging list via phone user interface:

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

If the paging list key is not configured, you can also press **Menu->Features->Paging List** to configure the paging list.

2. Press i or i to select a desired paging group.

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

	-Paging List
1. (Empty)	1
2. (Empty)	
3. (Empty)	
Back	I Option I Paging]

- 3. Press the **Option** soft key, and then press the **Edit** soft key.
- **4.** Enter the multicast IP address and port number (e.g., 224.5.6.20:10008) in the **Address** field.

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

F	dit Pao	ina	Address	_		
-		a		r		П
2. Address	5:					
224.5.6.20:	10008					
Back I	123		Delete		Save	

- 5. Enter the group name in the **Label** field.
- 6. Enter the desired channel in the **Channel** field.

The valid channel ranges from 0 to 30.

Edit Paging Address	П
4. Channel:	
0	
Back 123 Delete Save	

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

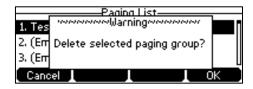
8. Repeat steps 2 to 7, you can add more paging groups.

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

To delete a paging group via phone user interface:

- 1. Press the paging list key when the phone is idle.
- **2.** Press \frown or \bigtriangledown to select a desired group.
- **3.** Press the **Option** soft key, and then select **Delete** soft key.

The LCD screen prompts "Delete selected paging group?".



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

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

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

To configure a default codec for multicast paging via web user interface:

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

The default codec is G722.

Yealink CP860	Status Account Networ	k Dsskey Features	Log Out English(English) • Settings Directory Security
Forward&DND	General Information		NOTE
General	Call Waiting	Enabled V	Call Waiting
Information	Call Waiting On Code		It allows IP phones to receive a new incoming call when there is
Audio	Call Waiting Off Code		already an active call.
Intercom	Auto Redial	Enabled V	Auto Redial It allows IP phones to
	Auto Redial Interval (1~300s)	10	automatically redial a busy number after the first attempt.
Transfer	Auto Redial Times (1~300)	10	Key As Send
Call Pickup		:	Assigns "#" or """ as the send key.
Remote Control		•	Hotline
Phone Lock	DTMF Repetition	3 🗸	IP phone will automatically dial out the hotline number when
	Multicast Codec	G722 🗸	lifting the handset, pressing the speakerphone key or the line
ACD	Play Hold Tone	Enabled V	key.
SMS	Return Code When Refuse	486 (Busy Here) 🗸	Call Completion It allows users to monitor the
Action URL	Hide Feature Access Codes	Disabled V	busy party and establish a call when the busy party becomes
Notification	Display Method on Dialing	User Name 🗸	available to receive a call.
Popups	Confirm	Cancel	You can click here to get more guides.

- 3. Click **Confirm** to accept the change.
- **Note** If G722 codec is used for multicast paging, the LCD screen will display the **HD**[®] icon to indicate that it is providing high definition voice.

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

Receiving RTP Stream

You can configure the phone to receive a Real Time Transport Protocol (RTP) stream from the pre-configured multicast address(es) and channel(s) without involving SIP signaling. You can specify up to 31 multicast addresses and channels that the phone listens to on the network. How the phone handles incoming multicast paging calls depends on Paging Barge and Paging Priority Active parameters configured via web user interface.

Paging Barge

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

- 1 to 31: Define the priority of the active call, 1 with the highest priority, 31 with the lowest.
- **Disabled**: The voice call in progress will take precedence over all incoming paging calls.

Paging Priority Active

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

To configure multicast listening addresses via web user interface:

- 1. Click on Directory->Multicast IP.
- 2. Select the desired value from the pull-down list of Paging Barge.
- 3. Select the desired value from the pull-down list of Paging Priority Active.
- **4.** Enter the multicast IP address(es) and port number (e.g., 224.5.6.20:10008) which the phone listens to for incoming RTP multicast in the **Listening Address** field.
- 5. (Optional.) Enter the label in the Label field.

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

6. Select the desired channel to listen from the pull-down list of Channel.

ealink cps60												Er	Log O Iglish(English)
	Status	Acco	unt	Network		Dsskey	Fea	ture	5	Setting	s	Directory	Security
Local Directory	Multicast Li	stening							_			NOTE	
Remote Phone Book		Paging B Paging P	Barge Priority Acti	ve	31 Er	abled		•				Multicast Pa Multicast pagi	ging ng allows IP nd/receive Real-
Phone Call Info	IP Add			ning Address		Label			Channe	l Priority		time Transpor streams to/fre	rt Protocol (RTP)
LDAP	1 IP Ac	ldress	224.5.6	5.20:10008		Product]	0 🔻] 1	Ê	without invol- Up to 10 liste	ving SIP signaling. ning multicast
Multicast IP	2 IP Ac	idress			E				0 🔹	2		addresses can the IP phone	be specified on
Setting	3 IP Ac	idress							0 •	3		7 You can o	lick here to get
Secury	4 IP Ac	idress							0 🔻	4	н.	more guides.	
	5 IP Ad	Idress			[0 •	5			
	6 IP Ac	Idress							0 🔻	6			
	7 IP Ac	idress			[0 •	7			
	8 IP Ac	idress							0 🔻	8			
	9 IP Ad	idress							0 •	9			
	10 IP A	ddress							0 •	10			

The default channel is 0.

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

Note The priorities of listening addresses are predefined: 1 with the highest priority, 31 with the lowest.

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

Multicast listening addresses are configurable via web user interface only.

Using Multicast Paging

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

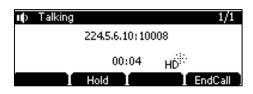
1. Press the multicast paging key when the phone is idle.

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

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

LED indicators illuminate solid green.

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



- 2. You can do the following:
 - To place the current multicast RTP session on hold, press the Hold soft key.

The sender's phone places the multicast RTP session on hold and receiver's phone releases the session.

To resume the held multicast RTP session, press the **Resume** soft key.

The multicast RTP session is established again.

• To end the multicast RTP session, press the EndCall soft key.

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

- **1.** Press the paging list key when the phone is idle.
- **2.** Press \frown or \bigtriangledown to select the desired paging group.
- 3. Press the Paging soft key to send RTP.
- 4. Your can do the following:
 - To place the current multicast RTP session on hold, press the Hold soft key.

The sender's phone places the multicast RTP session on hold and receiver's phone releases the session.

To resume the held multicast RTP session, press the **Resume** soft key.

The multicast RTP session is established again.

• To end the multicast RTP session, press the **EndCall** soft key.

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

Music on Hold

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

To configure music on hold server via web user interface:

1. Click on Account->Advanced.

ealink cps60			Log O English(English)
	Status Account Network	: Dsskey Features Sett	ings Directory Security
Register	Keep Alive Type	Default	NOTE
Basic	Keep Alive Interval(Seconds)	30	DTME
Dasic	RPort	Disabled 🔻	It is the signal sent from the IP phone to the network, which
Codec	Subscribe Period(Seconds)	1800	generated when pressing the phone's keypad during a call.
Advanced	DTMF Type	RFC2833	priorie's keypad during a call.
	DTMF Info Type	DTMF-Relay	Session Timer It allows a periodic refresh of
		:	SIP sessions through a re- INVITE request, to determine whether a SIP session is still active.
	SIP Server Type	Default	Busy Lamp Field/BLF List Monitors a specific extension/a list of extensions for status changes on IP phones.
	Music Server URI	sip:moh@sip.com	
	Directed Call Pickup Code		Shared Call Appearance (SCA)/ Bridge Line
	Group Call Pickup Code		Appearance (BLA) It allows users to share a SIP
	Distinctive Ring Tones	Enabled V	line on several IP phones. Any
	bisanceve rang rones		
	Unregister When Reboot	Disabled 🔻	IP phone can be used to originate or receive calls on the
		Disabled	IP phone can be used to
	Unregister When Reboot	Disabled •	IP phone can be used to originate or receive calls on the shared line.
	Unregister When Reboot VQ RTCP-XR Collector Name	Disabled	IP phone can be used to originate or receive calls on the shared line.

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

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

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

Note

For this feature to function, all involved parties cannot use encrypted RTP (SRTP).

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

Shared Call Appearance (SCA)

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

In the following figure, the registered line is shared:

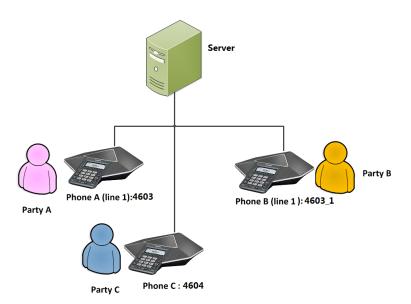


If two phones share a line, an incoming call to this extension will cause both phones to ring simultaneously. The incoming call can be answered on either phone but not both.

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

Configuring SCA Feature on the IP Phone

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



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

ealink _{CP860}	Status Account Network	Dsskey Features Settings	Log Out English(English) - Directory Security
Register	Account	Account1 -	NOTE
	Register Status	Registered	
Basic	Line Active	Enabled -	Account Registration Registers account(s) for the IP
Codec	Label	4603	phone.
Advanced	Display Name	4603	Server Redundancy It is often required in VoIP
	Register Name	4603	deployments to ensure continuity of phone service, for
	User Name	4603	events where the server needs
	Password	•••••	to be taken offline for maintenance, the server fails, or
	SIP Server 1		the connection between the IP phone and the server fails.
	Server Host	test.com Port 5060	NAT Traversal
	Transport	UDP 👻	A general term for techniques that establish and maintain IP
	Server Expires	3600	connections traversing NAT gateways. STUN is one of the
	Server Retry Counts	3	NAT traversal techniques.
	SIP Server 2		You can configure NAT traversa
	Server Host	Port 5060	for this account.
	Transport	UDP -	You can click here to get
	Server Expires	3600	more guides.
	Server Retry Counts	3	
	Enable Outbound Proxy Server	Enabled -	
	Outbound Proxy Server 1	10.1.8.11 Port 5060	
	Outbound Proxy Server 2	Port 5060	

1. Register the account 4603.

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

Yealink cpsoo	Status Account Netwo	rk Dsskey Features Settings	Log Out English(English) • Directory Security
Register Basic Codec Advanced	Keep Alive Type Keep Alive Interval(Seconds) RPort Subscribe Period(Seconds) DTMF Type DTMF Info Type PTime(ms) Shared I ine BLA Number BLA Subscription Period Unregister When Reboot VQ RTCP-XR Collector Name	Default ✓ 30 ✓ 1800 ✓ RFC2833 ✓ DTMF-Relay ✓ 20 ✓ Shared Call Appearance ✓ 300 ◯	NOTE DTMF It is the signal sent from the IP phone to the network, which is generated when pressing the IP phone's keypad during a call. Section Timer It alows a periodic refresh of SIP request, to determine whether a SIP session is still active. Busy Lamp Field/BLF List Montors a specific extension/a list of extensions for status changes on IP phones. Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA) It alows users to share a SIP line on several IP phones. Any IP
	VQ RTCP-XR Collector Address VQ RTCP-XR Collector Port Confirm	5060 Cancel	phone can be used to originate or receive calls on the shared line. Network Conference It alows multiple participants

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

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

1. Register the alternate account 4603_1.

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

ealink CP86		work Dsskey Fe	atures Settings	Log English(English) Directory Securit
Register	Account	Account1		NOTE
Register	Register Status	Registered		
Basic	Line Active	Enabled	•	Account Registration Registers account(s) for the
Codec	Label	4603 1		phone.
Advanced	Display Name	4603 1		Server Redundancy It is often required in VoIP
Advanced	Register Name	4603		deployments to ensure
	User Name	4603 1	_	continuity of phone service, events where the server nee
	Password		_	to be taken offline for maintenance, the server fails
	SIP Server 1			the connection between the phone and the server fails.
	Server Host	test.com	Port 5060	NAT Traversal
	Transport	UDP	-	A general term for technique that establish and maintain IF
	Server Expires	3600		connections traversing NAT gateways. STUN is one of th
	Server Retry Counts	3		NAT traversal techniques.
	SIP Server 2			
	Server Host		Port 5060	You can configure NAT trave for this account.
	Transport	UDP	•	You can click here to get
	Server Expires	3600		more guides.
	Server Retry Counts	3		
	Enable Outbound Proxy Server	Enabled	•	
	Outbound Proxy Server 1	10.1.8.11	Port 5060	
	Outbound Proxy Server 2		Port 5060	

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

							Log Out English(English) ▼	
		Status	Account	Network	Dsskey	Features	Settings	Directory Security
1	Register	Keep	Alive Type		Default	~		NOTE
	Basic	Кеер	Alive Interval(Seco	onds)	30			DTMF
	Dabro	RPort	t		Disabled	~		It is the signal sent from the IP phone to the network, which is
	Codec	Subs	cribe Period(Second	ds)	1800			generated when pressing the IP phone's keypad during a call.
	Advanced	DTM	F Туре		RFC2833	~		pircues region comigia can.
		DTM	F Info Type		DTMF-Relay	~		Session Timer It alows a periodic refresh of SIP
								sessions through a re-INVITE request, to determine whether a
					:			SIP session is still active.
		PTim	e(ms)		20	~		Busy Lamp Field/BLF List
		Share	ed Line		Shared Call Appe	earance 🗸		Monitors a specific extension/a list of extensions for status
		BLA I	Number					changes on IP phones.
		BLA S	Subscription Period		300			Shared Call Appearance
		Unre	gister When Reboo	t	Disabled	~		(SCA)/ Bridge Line Appearance (BLA)
		VQ R	TCP-XR Collector N	ame				It allows users to share a SIP line on several IP phones. Any IP
		VQ R	TCP-XR Collector A	ddress				phone can be used to originate or receive calls on the shared
		VQ R	TCP-XR Collector P	ort	5060			line.
			Confi	m		Cancel		Network Conference It allows multiple participants

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

Using SCA Feature on the IP Phone

This section provides you with detailed information on using the CP860 IP Conference phone in a SCA scenario.

You can do the following using CP860 IP Conference phone in a SCA scenario:

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

Placing Calls

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

To place a call on the shared line:

- 1. Enter the desired number using the keypad when the phone is idle.
- 2. Press | C |, (OK), #see or the Send soft key.

To place multiple calls on the shared line:

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

1. Press the **Hold** soft key. The original call is placed on hold.

- 2. Press the NewCall soft key to enter the dialing screen.
- 3. Enter the desired number using the keypad.
- 4. Press / (οκ), #see or the Send soft key.

Phone A will dial the entered number.

Answering Calls

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

To answer a call on the shared line:

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

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

- Press , (or) or the Answer soft key on phone A.
 Phone B stops ringing.
- Press $| \mathbf{c} |$, $(\mathbf{o}\mathbf{k})$ or the **Answer** soft key on phone B.

Phone A stops ringing.

To answer multiple calls on the shared line:

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



Note

Make sure call waiting feature is enabled on phone A. For more information, refer to Call Waiting on page 87.

Do one of the following on phone A:

- Press the Answer soft key. Phone B stops ringing.
- Press 🟹 to access the new call.

Press $\left| \begin{array}{c} c \end{array} \right|$, $\left(\circ \kappa \right)$ or the **Answer** soft key. Phone B stops ringing.

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

You can also answer the call on phone B:

1. Press $| \mathbf{r} |$, $(\mathbf{o}\mathbf{K})$ or the **Answer** soft key. Phone A stops ringing.

Placing a Call on Hold

To place a call on hold:

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

Retrieving a Held Call

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

To retrieve the held call on phone A:

1. Press the **Resume** soft key on phone A.

The conversation between phone A and phone C is retrieved.

Bridged Line Appearance (BLA)

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

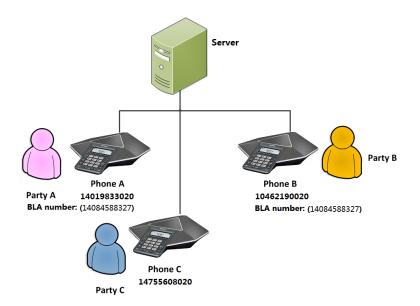
In the following figure, the registered account is shared:



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

Configuring BLA Feature on the IP Phone

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



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

alink CP86				Log English(English)
	Status Account	Network Dsskey	Features Sett	tings Directory Securit
Register	Account	Account1	•	NOTE
	Register Status	Registered		
Basic	Line Active	Enabled	•	Account Registration Registers account(s) for the
Codec	Label	14019833020		phone.
Advanced	Display Name	14019833020		Server Redundancy It is often required in VoIP
	Register Name	14019833020		deployments to ensure continuity of phone service,
	User Name	14084588327		events where the server ne
	Password	•••••		to be taken offline for maintenance, the server fail
	SIP Server 1			the connection between th phone and the server fails.
	Server Host	sip.test.com	Port 5060	NAT Traversal
	Transport	UDP	•	 A general term for technique that establish and maintain I
	Server Expires	3600		connections traversing NAT gateways. STUN is one of th
	Server Retry Counts	3		NAT traversal techniques.
	SIP Server 2			No
	Server Host		Port 5060	You can configure NAT trav for this account.
	Transport	UDP	•	You can click here to ge
	Server Expires	3600		more guides.
	Server Retry Counts	3		
	Serial really counts	9		_
	Enable Outbound Proxy S	erver Enabled	•	
	Outbound Proxy Server 1	sip114.test.com	Port 5060	
	Outbound Proxy Server 2		Port 5060	

1. Register the account 14019833020.

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

alink CP860						En	Log O glish(English)
	Status Accou	nt Network	Dsskey	Features	Settings	Directory	Security
Register	Keep Alive Type		Default	~		NOTE	
Basic	Keep Alive Interv	al(Seconds)	30			DTMF	
asic	RPort		Disabled	~		It is the signal	
odec	Subscribe Period	Seconds)	1800			generated whe	
dvanced	DTMF Type		RFC2833	~		phone's keypad	i during a call.
	DTMF Info Type		DTMF-Relay	\checkmark		Session Time	
						It allows a peri sessions throug	h a re-INVITE
			:			SIP session is s	termine whether still active.
	PTime(ms)		20	~		Busy Lamp Fi	old / PLE List
	Shared Line		Draft BLA	~		Monitors a spec list of extension	ific extension/a
	BLA Number		14084588327			changes on IP	
	BLA Subscription	Period	300			Shared Call A	ppearance
	BLA Subscription	Period	300			(SCA)/ Bridge Appearance (e Line
	Unregister When	Reboot	Disabled	~		It allows users on several IP p	to share a SIP I
	VQ RTCP-XR Coll	actor Name				phone can be u or receive calls	used to originate
	VQ RTCP-XR Coll	ector Address				line.	
	VO RTCP-XR Coll	ector Port	5060			Network Con	ference
						It allows multip (more than thr	le participants
		Confirm		Cancel		call.	

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

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

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

1. Register the account 10462190020.

Yealink CP860	Status Account Network	Dsskey Features Settings	Log Out English(English) • Directory Security
Register	Account	Account1 -	NOTE
register	Register Status	Registered	
Basic	Line Active	Enabled 🗸	Account Registration Registers account(s) for the IP
Codec	Label	10462190020	phone.
Advanced	Display Name	10462190020	Server Redundancy
Auvanceu	Register Name	10462190020	It is often required in VoIP deployments to ensure
	User Name	14084588327	continuity of phone service, for events where the server needs
			to be taken offline for maintenance, the server fails, or
	Password	••••••	the connection between the IP
	SIP Server 1		phone and the server fails.
	Server Host	sip.test.com Port 5060	NAT Traversal A general term for techniques
	Transport	UDP 👻	that establish and maintain IP
	Server Expires	3600	connections traversing NAT gateways. STUN is one of the
	Server Retry Counts	3	NAT traversal techniques.
	SIP Server 2		You can configure NAT traversal
	Server Host	Port 5060	for this account.
	Transport	UDP -	You can click here to get
	Server Expires	3600	more guides.
		3	
	Server Retry Counts	3	
	Enable Outbound Proxy Server	Enabled 👻	
	Outbound Proxy Server 1	sip114.test.com Port 5060	
	Outbound Proxy Server 2	Port 5060	

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

Yealink CP860			Log Out Englsh(Englsh) v
	Status Account Netw	work Dsskey Features	Settings Directory Security
Register	Keep Alive Type	Default	NOTE
	Keep Alive Interval(Seconds)	30	
Basic	RPort	Disabled 🗸	DTMF It is the signal sent from the IP
Codec	Subscribe Period(Seconds)	1800	phone to the network, which is generated when pressing the IP
Advanced	DTMF Type	RFC2833	phone's keypad during a call.
	DTMF Info Type	DTMF-Relay 🗸	Session Timer
			It allows a periodic refresh of SIP sessions through a re-INVITE
		:	request, to determine whether a SIP session is still active.
	PTime(ms)	20 🗸	
	Shared Line	Draft BLA 🗸	Busy Lamp Field/BLF List Monitors a specific extension/a
	BLA Number	14084588327	list of extensions for status changes on IP phones.
	BLA Subscription Period	300	Shared Call Appearance
	BLA Subscription Period	300	(SCA)/ Bridge Line Appearance (BLA)
	Unregister When Reboot	Disabled	It allows users to share a SIP line on several IP phones. Any IP
	VQ RTCP-XR Collector Name		phone can be used to originate or receive calls on the shared
	VQ RTCP-XR Collector Address		line.
	VQ RTCP-XR Collector Port	5060	Network Conference
			It allows multiple participants (more than three) to join in a
	Confirm	Cancel	call.

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

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

Using BLA Feature on the IP Phone

This section provides you with detailed information on using the CP860 IP phone in a BLA scenario.

You can do the following using CP860 IP phone in a BLA scenario:

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

Placing Calls

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

To place a call on the shared line:

- 1. Enter the desired number using the keypad when the phone is idle.
- **2.** Press $| \mathbf{r} |$, $(\mathbf{o}\mathbf{K})$, $| \mathbf{\#}_{\mathsf{seco}} |$ or the **Send** soft key.

To place multiple calls on the shared line:

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

1. Press the Hold soft key. The original call is placed on hold.

- 2. Press the **NewCall** soft key to enter the dialing screen.
- **3.** Enter the desired number using the keypad.
- **4.** Press $| \mathbf{r} |$, $(\mathbf{o}\mathbf{K})$, $| \mathbf{\#}_{ssso} |$ or the **Send** soft key.

Phone A will dial the entered number.

Answering Calls

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

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

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

Placing a Call on Hold

To place a call on hold:

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

Retrieving a Held Call

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

To retrieve the held call on phone A:

1. Press the **Resume** soft key on phone A.

The conversation between phone A and phone C is retrieved.

Messages

Short Message Service (SMS)

You can send and receive text messages using the CP860 IP phone. New text messages can be indicated both acoustically and visually. When receiving a new text message, the phone will play a warning tone. The LCD screen will prompt "n New Text Message(s)" ("n" indicates the number of unread text messages, e.g., 1 New Text Message(s)) and display an icon \sum .



Note When the phone receives a text message, the text message prompt window will pop up by default, if you want to disable the feature, contact your system administrator for more information.

You can store text messages in your phone's Inbox, Sentbox, Outbox or Draftbox. Each of the boxes can store up to 100 text messages. If the number of the text messages in one box is more than 100, the phone will directly delete the oldest text message in the box.

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

To read a text message:

1. Press Menu->Message->Text Message->Inbox.



2. Select the desired message and then press the View soft key.

Note If the phone prompts receiving new text messages, you can also press the **View** soft key to read the new messages directly.

To send a text message:

1. Press Menu->Message->Text Message->New Message.

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

		 -New	Message	e	
Hi				•	
	D			-	_
	Back	abc	Der	ete 📘	Send

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

Sending a text message is configurable via web user interface at the path Features->SMS.

To reply a text message:

- 1. Press Menu->Message->Text Message->Inbox.
- 2. Select the desired message and then press the **Reply** soft key.
- 3. Compose the new text message. You can press the **abc** soft key to change the input mode.

okļ	——T	o:1039			
Back	abc		elete	Send	

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

To delete a text message:

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

	—Inbox—		_
Delete			
Delete All			
Cancel		0K	

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

The LCD screen prompts "Delete the selected message?".



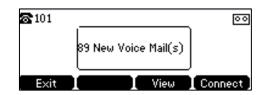
4. Press the **OK** soft key to delete this message or the **Cancel** soft key to cancel.

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

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

Voice Mail

You can leave voice mails for someone else using the CP860 IP conference phone. You can also listen to voice mails that are stored in voice mailbox. When receiving a new voice mail, the phone will play a warning tone. The LCD screen will prompt "n New Voice Mail(s)" ("n" indicates the number of unread voice messages, e.g., 89 New Voice Mail(s)) and display an icon \boxed{OO} .



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

NoteVoice Mail is not available on all servers.You can configure the phone not to display the pop-up prompt, contact your system
administrator for more information.

To leave a voice mail:

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

To configure voice mail access codes via phone user interface:

- 1. Press Menu->Message->Voice Mail->Set Voice Mail Code.
- 2. Press the **123** soft key to select the proper input mode and then enter the voice mail access code (e.g., *88).

	-Set Voi	ce Mail Code—	
1. 101			
Pack	123	Delete	Save

- 3. Press the Save soft key to accept the change or the Back soft key to cancel.
- **Note** Voice mail access codes must be predefined on the system server. Contact your system administrator for the more information.

To listen to voice mails:

- When the LCD screen prompts that the phone receives a new voice mail, you can press the Connect soft key to dial out the voice mail access code.
- 2. Follow the voice prompt to listen to your voice mails.

To view the voice mail via phone user interface:

1. Press Menu->Message->Voice Mail->View Voice Mail.

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

1.	101	7 New 647 Old Mail
	Back	Connect

2. Press the Connect soft key to listen to voice mails.

Message Waiting Indicator (MWI)

The CP860 IP conference 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 via an indicator message (including a voice mail icon) on the LCD screen. This will be cleared when you retrieve all voice mails or delete them.

The MWI service is unsolicited for some servers, so the CP860 IP conference phone only handles the MWI messages sent from the server. But for other servers, the MWI service is solicited, so the CP860 IP conference phone must enable subscription for MWI.

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

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

The MWI subscription parameters you need to know:

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

Note

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

To configure subscribe for MWI via web user interface:

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

ealink cps60					Log Ou English(English) ▼	
	Status Account Network	C Dsskey Fe	atures	Settings	Directory Security	
Register	Account	Account1	۲		NOTE	
Deale	Keep Alive Type	Default	Ŧ			
Basic	Keep Alive Interval(Seconds)	30			DTMF It is the signal sent from the IP	
Codec	Codec RPort		Disabled •		phone to the network, which is generated when pressing the IP phone's keypad during a call.	
Advanced	Subscribe Period(Seconds)	1800			priorie s keypad during a cail.	
	DTMF Type	RFC2833	•		Session Timer It allows a periodic refresh of	
	DTMF Info Type	DTMF-Relay	•		SIP sessions through a re- INVITE request, to determine	
	DTMF Payload Type(96~127)	101			whether a SIP session is still active.	
	Retransmission	Disabled	•			
	Subscribe Register	Disabled	Disabled 🔹		Busy Lamp Field/BLF List Monitors a specific extension/a	
	Subscribe for MWI	Enabled	•		list of extensions for status changes on IP phones.	
	MWI Subscription Period(Seconds)	3600			Shared Call Appearance	
	Subscribe MWI To Voice Mail	Disabled	*		(SCA)/ Bridge Line Appearance (BLA)	

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

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

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

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

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

alink cp860						Eng	glish(English)
	Status Accou	nt Network	Dsskey	Features	Settings	Directory	Security
Register	Account		Account1	•		NOTE	
	Keep Alive Type		Default	Ŧ			
lasic	Keep Alive Inter	val(Seconds)	30			DTMF It is the signal	
odec	RPort		Disabled	٣		generated whe	network, which en pressing the
dvanced	Subscribe Period	l(Seconds)	1800			phone's keypad during a ca	
	DTMF Type		RFC2833	Ŧ		Session Timer It allows a periodic refre	
	DTMF Info Type		DTMF-Relay	•		SIP sessions th	
	DTMF Payload Type(96~127)		101			whether a SIP session is still active.	
	Retransmission		Disabled	Ŧ			
	Subscribe Regist	er	Disabled	Ŧ			cific extension/a
	Subscribe for MV	NI	Enabled	•	list of extensions changes on IP pl		
	MWI Subscriptio	n Period(Seconds)	3600			Shared Call A	nnearance
	Subscribe MWI To Voice Mail		Enabled	T	(CCA)/ Bridge		e Line
	Voice Mail		*4			It allows users	
	Voice Mail Displa	у	Enabled	•		IP phone can b	

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

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

Note

MWI subscription is configurable via web user interface only.

Using PC or Mobile Device with the Conference Phone

You connect a PC or mobile device to your IP conference phone so you can use the conference phone's speaker to stream two-way audio from the PC or mobile device for hands-free conversations. For more information on how to connect the PC or mobile device, refer to Connecting the Optional PC or Mobile Device on page 16.

From your conference phone, you can do the following:

- Place a PC or mobile audio on hold.
- Set up a conference between the calls on your conference phone and the PC or mobile device.
- Answer calls from other phones while you connect the PC or mobile device to the conference phone.

Connecting the PC or Mobile Device to the Conference Phone

You can connect the PC or mobile audio when your conference phone is idle, when you have a call or conference call, or when you are placing a call.

To connect the PC or mobile audio when your conference phone is idle:

1. Connect the PC or mobile device to the conference phone.

The PC or mobile audio plays through your conference phone's speaker. For example, if the PC

or mobile device plays a video or music, you can listen to the audio on your conference phone. If an incoming call arrives on the PC or mobile device and the PC or mobile device answers it, you can listen and speak to the caller on your conference phone.



To connect the PC or mobile audio when there is a call or conference call on your conference phone:

1. Connect the PC or mobile device to the conference phone.

The original call or conference call is placed on hold. The PC or mobile audio plays through your conference phone's speaker.



To connect the PC or mobile audio when you are placing a call:

1. Connect the PC or mobile device to the conference phone.

The dialing is canceled. The PC or mobile audio plays through your conference phone's speaker.

					1/1
PC-I	Mot	oil	e M	ode	•
	Hold		Conf		

Placing the PC or Mobile Audio on Hold

You can place an active PC or mobile audio on hold. When you place the PC or mobile audio on hold, you cannot hear any audio associated with the PC or mobile device on your conference phone.

To place the PC or mobile audio on hold:

1. Press the Hold soft key.

LED indicators flash green. The LCD screen indicates that the audio is on hold.



To resume the PC or mobile audio:

1. Press the **Resume** soft key.

Muting or Un-muting the PC or Mobile Audio

You can mute the PC or mobile audio so that the other party cannot hear you, but you can still hear the other party.

To mute the PC or mobile audio:

1. Press 繢 .

LED indicators illuminate solid red. The LCD screen indicates that the audio is muted



To un-mute the PC or mobile audio:

1. Press 繢 again.

Creating Conference Calls with the PC or Mobile Audio

To create a conference call with the PC or mobile audio, do one of the following:

- If you have an active PC or mobile audio, press the **Conf** soft key, and then place a new call to the other party. Press the **Conf** soft key again when the party answers the call.
- If you have an active PC or mobile audio and a call or conference call on hold, press the Conf soft key.
- **Note** The conference phone can set up five-way conference call with the PC or mobile audio. For more information on how to set up a conference call, refer to Local Conference on page 87.

Adjusting the Volume of the PC or Mobile Audio

You can adjust the volume of the PC or mobile audio on your conference phone or on the PC or mobile device.

To adjust the volume of the PC or mobile audio:

When you connect the PC or mobile audio, press - + to adjust the volume of the conference phone's speaker.

Removing the PC or Mobile Audio

To remove the PC or mobile audio, disconnect the 3.5mm jack cable from the conference phone.

Using Your Phone with PSTN Account

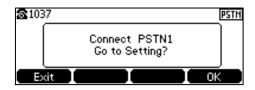
You can connect optional expansion PSTN box CPN10 to extend functions of the conference phone. Calls can be made using the Public Switched Telephone Network (PSTN). Up to 2 cascaded PSTN Boxes can be installed to an IP conference phone, which allows you to experience the local three-way conference conveniently in excellent speech quality with PSTN. For more information on how to connect the PSTN box, refer to Connecting the Optional Expansion PSTN Box CPN10 on page 15.

When using PSTN, some features are not available, such as transfer, DND, forward and so on. For more information, refer to Appendix C – Unavailable Features for PSTN on page 158.

Configuring the PSTN Account

When the first PSTN box CPN10 is connected to the phone successfully, the LCD screen prompts "Connect PSTN1 Go to Setting?". When the second PSTN box CPN10 is connected to the first PSTN box CPN10 successfully, the LCD screen prompts "Connect PSTN2 Go to Setting?".

When two PSTN boxes are connected simultaneously to the phone successfully, the LCD screen prompts "Connect PSTN1 PSTN2 Go to Setting?".



You can press the **OK** soft key to configure the PSTN account or press the **Exit** soft key to cancel. If you press the **Exit** soft key, you can also configure the PSTN account at the path **Menu->Settings->Advanced Settings** (default password: admin) ->**Accounts**.

To configure the PSTN account via phone user interface after pressing the OK soft key:

- 1. (Optional.) Press 📩 or 🔽 to select the desired PSTN account.
- 2. Press the Enter soft key.
- 3. Press the ◀ or ▶ soft key to select **Enabled** or **Disabled** from the **Active Line** field.

PSTN Line	
1. Active Line:	
Enabled	•
Back	Save

- 4. (Optional.) Enter the string that will appear on the LCD screen in the Label field.
- 5. Press the Save soft key to accept the change.

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

Default Account

To configure the default account via phone user interface:

- 1. Press Menu->Settings->Advanced Settings (default password: admin) ->Accounts.
- **2.** Press \frown or \bigtriangledown to select the desired account.

	Account-
1. 1037	Default Account
2. PSTN1	Registered
Back	Default Enter

3. Press the Default soft key.

Selecting the Country for PSTN Use

To select the country which your phone operates in via phone user interface:

- 1. Press Menu->Settings->Basic Settings->Country.
- **2.** Press \frown or \bigtriangledown to select the desired country.

	-Country	
9. Chile		R
🗸 10. China		
11. Columbia		
Back		Save

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

Flash Time

You can configure the flash time to adjust the length of time before a hook flash times out (or the call disconnects).

Flash time can be configured from the following options:

- Auto: It depends on the country that is configured for the phone (refer to Selecting the Country for PSTN Use). The flash time is automatically set for that country.
- **75msec**, **100msec**, **300msec** or **600msec**: The hook flash times out after the designated time (in milliseconds).

To configure flash time via phone user interface:

1. Press Menu->Settings->Advanced Settings (default password: admin) ->Flash Time.

2. Press \checkmark or \checkmark to select the desired value.

		Flash Time	
>	1.	Auto	
	2.	75msec	
	з.	100msec	U
	Ba	ck	Save

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

Note The required flash time may be different in different countries or for different gateways/PBX servers. For more information, contact your system administrator.

Basic Call Features

Placing Calls

You can press the off-hook key either before or after entering the number to place a call. You can also dial an entry from call history, local directory, favorite directory or remote phone book. For more information, refer to Contact Management on page 35 and Call History Management on page 49.

The call duration and far-site's information (name or phone number) are visible on the LCD screen. In the figure below, the call to 2248 has lasted 53 seconds.



To place a call:

Do one of the following:

Press to obtain a dial tone (You can press the Switch soft key to select the desired PSTN account).

• PSTN1:			1/1
"			
Directory	123	Switch	Cancel

Enter the desired number using the keypad.

Press	C	, (ОК),	#send	or	the	Send	soft	key
-------	---	-----	----	----	-------	----	-----	------	------	-----

- Enter the desired number using the keypad.

Press | $(\circ \kappa)$, $(\#_{sso})$ or the **Send** soft key.

The # key is configured as a send key by default. You can also set the * key as the send key, or

set neither. For more information, refer to the Key As Send on page 30.

The CP860 IP conference phone can handle multiple calls at a time. However, only one active call (the call that has audio associated with it) can be in progress at any given time. And one PSTN account can only support one call.

To place multiple calls:

You can have more than one call on your CP860 IP conference phone. To place a new call during an active call, do one of the following:

Press c . The active call is placed on hold.

Enter the desired number using the keypad.

, $(o\kappa)$, $\#_{sevo}$ or the **Send** soft key. Press C

Press the Hold soft key to place the original call on hold.

Press the NewCall soft key.

Enter the desired number using the keypad.

 $(\land (\circ \kappa), #_{sevo})$ or the **Send** soft key. Press

You can press (or) to switch between calls, and then press the **Resume** soft key to retrieve the desired call.

Answering Calls

You can answer a call no matter whether you are in another call or not. But one PSTN account can only support one call.

Answering When Not in Another Call

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

To answer a call:



Press $| \mathbf{r} |$, $(\mathbf{o}\mathbf{K})$ or the **Answer** soft key.

Answering When in Another Call

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

Press the **Answer** soft key.

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

Press vito access the new call.

Press $[\mathbf{r}]$, $(\mathbf{o}\mathbf{K})$ or the **Answer** soft key.

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

Ending Calls

To end a call:

Do one of the following:

- Press the EndCall soft key.

- Press	~
---------	---

Note

To end a call placed on hold, you can press the **EndCall** soft key to end the call directly, or press the **Resume** soft key to resume the call before ending it.

Redialing Numbers

For more information, refer to Redialing Numbers on page 74.

Recent Call In Dialing

For more information, refer to Recent Call In Dialing on page 74.

Auto Answer

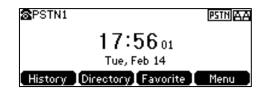
You can enable auto answer to automatically answer an incoming call. You can also enable auto answer mute to mute the local microphone when an incoming call is answered automatically.

To configure auto answer and auto answer mute via phone user interface:

- 1. Press Menu->Features->Auto Answer.
- 2. Press the **d** or **b** soft key to select the desired PSTN account from the Line ID field.
- 3. Press the \blacktriangleleft or \blacktriangleright soft key to select **Enabled** from the **Status** field.
- 4. Press the ◀or ▶soft key to select **Enabled** from the **Auto Answer Mute** field.

Auto Answer	
3. Auto Answer Mute:	
Enabled	•
Back 🖌 🕨	Save

Press the Save soft key to accept the change or the Back soft key to cancel.
 The icon AA appears on the LCD screen.



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

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

Auto Redial

For more information, refer to Auto Redial on page 76.

ReCall

For more information, refer to ReCall on page 78.

Call Mute

For more information, refer to Call Mute on page 78.

Call Hold/Resume

You can place a call on hold/resume a held call for PSTN account in the same way as SIP/Cloud account with the following exceptions: when you place a call on hold, the other party cannot hear music on hold and the LCD screen of the other party doesn't indicate that the call is being held.

For more information, refer to Call Hold/Resume on page 80.

Conference

As one PSTN account can only support one call. To set up a conference, you need to activate a hook flash. A hook flash is a quick off-hook/on-hook/off-hook cycle (just like picking up a handset, laying the handset down on the handset cradle, and then picking it up again). When you activate a hook flash, the message "Flashing" displays on the screen for about one second, as shown below:



Then the active call will be placed on hold, you are allowed to place a new call and connect three parties in a conference. You cannot view the last participant in the conference call.

You can also set up a conference with more than two parties when you have multiple accounts on the phone. You can view and manage each participant in the conference call.

Note The way to set up a conference call using hook flash may be different for different gateways/PBX servers.

To set up a conference call:

- 1. Place a call to the first party.
- **2.** When the first party answers the call, press the **Flash** soft key to activate a hook flash.

The message "Flashing" displays on the screen for about one second and the active call is placed on hold.

- **3.** Enter the number of the second party.
- **4.** When the second party answers the call, press the **Flash** soft key again to join two parties in the conference.

To remove this party from the conference call, press the **Flash** soft key.

📭 Talking	1/1
2228	
2220	
01:37	
Flash Hold Conf	EndCall

- 5. To add an additional party to the conference, press the **Conf** soft key (another account is needed).
- **6.** Enter the number of the new party and then press $|_{\mathcal{C}}|_{\mathcal{C}}$, $(\alpha \kappa)$, $\#_{mo}$ or the **Send** soft key.
- 7. Press the Conf soft key when the party answers.
- 8. Repeat steps 5 to 7 to join more parties in the established conference call.

You can create a conference between an active call and a call on hold by pressing the the **Conf** soft key.

To set up a conference call with an active call and a call on hold:

- 1. Establish two calls on the phone.
- **2.** Press \frown or \bigtriangledown to select an active call.
- 3. Press the **Conf** soft key to join the two calls in the conference.

During the conference call, you can do the following:

- Press the Hold soft key to place the conference on hold.
- Press the **Conf** soft key to join more parties in an established conference call.
- Press the **Manage** soft key, and then press (-) or (-) to select the desired party:
 - Press the **FarMute** soft key to mute the party. The muted party can hear everyone, but no one can hear the muted party.

- Press the **Remove** soft key to remove the selected party from the conference call.
- Press the **All Split** soft key to split the conference call into individual calls on hold.
- Press the **Back** soft key to return to the previous screen.
- Press (*) to mute the conference call, all other participants can hear each other, but they cannot hear you.

Advanced Phone Features

Recording Using a USB Flash Drive

You can use the recording feature when connecting the PSTN box CPN10. But you need to connect the USB flash drive to the USB port on the PSTN box CPN10 in advance.

For more information on how to use recording feature, refer to Call and Conference Recording on page 99.

Using PC or Mobile Device with the Conference Phone

For more information, refer to Using PC or Mobile Device with the Conference Phone on page 128.

Troubleshooting

This chapter provides general troubleshooting information to help you solve the problems you might encounter when using your CP860 IP conference phone.

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

General Issues

How do I find the basic information of the IP phone?

Press the OK key when the IP phone is idle to check the basic information of the IP phone, such as IP address and firmware version. For more basic information, refer to Phone Status on page 17.

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

Three ways to obtain the MAC address of a phone:

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

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

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

Why can't I send an SMS to any other phone?

SMS depends on support from a SIP server. Contact your system administrator for more information.

Display Issues

Why is the LCD screen blank?

- Ensure that the phone is properly plugged into a functional AC outlet.
- Ensure that the phone is plugged into a socket controlled by a switch that is on.

- If the phone is plugged into a power strip, try to plug it directly into a wall outlet instead.
- If your CP860 IP conference phone is powered from PoE, ensure that you use a PoE-compliant switch or hub.

Why does the phone display "Network unavailable"?

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

Why does the phone display "No Service"?

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

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

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

Password Issues

How to change the user password?

To change the user password via web user interface:

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

Yealink CP860	Status Account Ne	etwork Dsskey Features	Settings	Log Out English(English) • Directory Security
Password Trusted Certificates Server Certificates	User Type Old Password New Password Confirm Password	user		NOTE User Password/ Administrator Password When loging into the web user interface, you need to enter the user name and
	Confirm	Cancel		password. You can change the user/ administrator password for security reasons.

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

You can also contact your system administrator for help.

Note

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

User password is configurable via web user interface only.

Call Issues

Why can't I receive calls?

- Check the SIP registration with your system administrator.
- Check that DND (Do Not Disturb) mode is deactivated on your phone. Refer to Do Not Disturb (DND) on page 81.
- Check that call forward is disabled on the phone. Refer to Call Forward on page 82.
- Check whether the caller number is stored in the blacklist directory. Refer to Blacklist on page 45.

Why can't I record calls?

- Ensure that the USB flash drive is inserted to the USB port on the phone.
- Ensure that the USB flash drive inserted is compatible with the phone. For more information, contact your reseller.
- Check if there is enough free space (greater than 2 minutes) on the USB flash drive. Press Menu->USB Record->Storage Space.

Storge Media Properties					
1. Total:	7.6GB				
2. Used:	5.3MB (0.1% Used)				
3. Free:	7.6GB (99.9% Free)				
Back					

Audio Issues

Why doesn't the phone ring?

Check the ringer volume on the phone. To adjust the ringer volume setting, press the Volume key when the phone is on-hook and idle. For more information, refer to Volume on page 32.

Why does the phone play a tone when hold? How to disable it?

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

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

- 1. Click on Features->General Information.
- 2. Select the desired value from the pull-down list of Play Hold Tone.

3. Enter the desired time in the **Play Hold Tone Delay** field.

alink cp860	Status	Account	Network	Dsskey	Features	Settings	Directory Security	
Forward&DND	c	General Informati	on				NOTE	
General Information		Call Waiting Call Waiting On Co	de	Enabled	~		Call Waiting It allows IP phones to receive	
Audio		Call Waiting Off Co	de			new incoming call when already an active call.		
Intercom	Auto Redial Auto Redial Interval (1~300s) Auto Redial Times (1~300)			Enabled	~	Auto Redial It allows IP phones to automatically redial a b		
Transfer				10			number after the first attempt	
Call Pickup				:			Key As Send Assigns "#" or "*" as the send key.	
Remote Control	Multicast Codec			G722	~		Hotline IP phone will automatically dia	
Phone Lock		Play Hold Tone		Enabled	~	out the hotline num	out the hotline number when lifting the handset, pressing the	
ACD		Play Hold Tone De	lay	30		speakerphone key or t key.		
SMS		Allow Mute		Enabled	~		Call Completion	
Action URL		DHCP Hostname		CP860			It allows users to monitor the busy party and establish a call when the busy party becomes	
ACTOLI OKL		Reboot in Talking		Disabled	~		available to receive a call.	
Notification Popups		Hide Feature Acces	s Codes	Disabled	~		You can click here to get more guides.	

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

How to make a call using SRTP?

You can enable SRTP to encrypt the audio stream(s) of phone calls. The parties participating in the call should enable SRTP.

To enable SRTP via web user interface:

1. Click on Account->Advanced.

2. Select the desired type from the pull-down list of RTP Encryption(SRTP).

Yealink CP860			Log Out English(English) 🔻	
	Status Account Network	Dsskey Features Settings	Directory Security	
Register	Account	Account1	NOTE	
Basic	Keep Alive Type	Default 🔻	DTHE	
Dasic	Keep Alive Interval(Seconds)	30	DTMF It is the signal sent from the IP phone to the network, which is	
Codec	RPort	Disabled 🔹	generated when pressing the IP phone's keypad during a call.	
Advanced	Subscribe Period(Seconds)	1800	priorie's keypau during a cai.	
	DTMF Type	RFC2833	Session Timer It allows a periodic refresh of	
	DTMF Info Type	DTMF-Relay	SIP sessions through a re- INVITE request, to determine	
	DTMF Payload Type(96~127)	101	whether a SIP session is still active.	
	Retransmission	Disabled 🔻		
	Subscribe Register	Disabled 🔻	Busy Lamp Field/BLF List Monitors a specific extension/a	
	Subscribe for MWI	Disabled 🔻	list of extensions for status changes on IP phones.	
	MWI Subscription Period(Seconds)	3600	Shared Call Appearance	
	Subscribe MWI To Voice Mail	Disabled 🔻	(SCA)/ Bridge Line Appearance (BLA)	
	Voice Mail		It allows users to share a SIP line on several IP phones. Any	
	Voice Mail Display	Enabled •	IP phone can be used to originate or receive calls on the	
	Caller ID Source	FROM	shared line.	
	Session Timer	Disabled 🔻	Network Conference	
	Session Expires(30~7200s)	1800	It allows multiple participants (more than three) to join in a	
	Session Refresher	UAC 🔻	call.	
	Send user=phone	Disabled •	VQ-RTCPXR The VQ-RTCPXR mechanism,	
	RTP Encryption(SRTP)	Optional •	complaint with RFC 6035, sends the service quality metric	

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

NoteSRTP is not available on all servers. Contact your system administrator for more information.SRTP is configurable via web user interface only.

Network Issues

How to diagnostic the network?

The wrong network settings may result in inaccessibility of your phone and may also have an impact on your network performance. In this case, you can use the "Ping" or "Traceroute" method to check whether the network is connected, analyze and judge network fault.

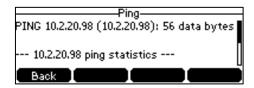
To check the network using "Ping" method via phone user interface:

- 1. Press Menu->Features->Diagnostics->Network->Ping.
- 2. Enter the desired IP address or URL in the Ping IP or URL field.

		Ping———	
Ping IP o	r URL		
10.2.20.98	}		
Back	123	Delete	Start

3. Press the Start soft key.

The screen displays the network status information.



You can use the ping statistics information to check the network connection situation.

The following table shows the network connection situation and ping statistics:

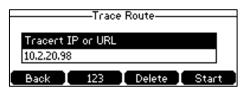
Ping Statistics	Network Connection Situation
4 packets transmitted, 4 packets received, 0% packet loss	The network connection between you and the target is strong.
4 packets transmitted, 1, 2 or 3 packets received, 25%, 50% or 75% packet loss	The network connection between you and the target is unstable or faulty.
4 packets transmitted, 0 packets received, 100% packet loss	The network connection between you and the target is unreachable.

4. Press the **Back** soft key to back to the preview screen.

If you find that the network connection between you and the target is unreachable, you can traceroute the target to see where the problem is. It can also display the path traffic between your IP phone and the target.

To check the network using "Traceroute" method via phone user interface:

- 1. Press Menu->Features->Diagnostics->Network->Trace Route.
- 2. Enter the desired IP address or URL in the Tracert IP or URL field.



3. Press the Start soft key.

The screen displays the network status information.

Trace Route
traceroute to 10.2.20.98 (10.2.20.98), 30 ho
ps max, 38 byte packets
1 10.2.20.151 (10.2.20.151) 3009.922 ms!
Back

The first line of result is information about what you are doing. It shows the target, IP address of the target, the maximum number of hops that will be allowed, and the size of the packets being

sent.

Then the following line describes each network device in the path traffic between the IP phone and the target.

A traceroute can end with one of several error indications indicating why the traceroute cannot proceed. The following table shows the possible error messages:

Error Message	Meaning
* * *	Target unreachable.
!H	Host unreachable.
!N	Network unreachable.

Press the **Back** soft key to back to the preview screen.

Log Issues

How to export PCAP trace?

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

To export a PCAP trace via web user interface:

- 1. Click on Settings->Configuration.
- 2. Click Start to begin capturing signal traffic.
- 3. Recreate the error to be documented in the trace.
- 4. Click **Stop** to stop the capture.

Yealink CP860	Status Account Network	Dsskey Features Settings	Log Out English(English) • Directory Security
Preference	Export Import Config	Browse No file selected.	NOTE
Time & Date		Import Export	Configuration IP phones can provide feedback
Call Display	Export CFG Configuration File	Static Settings Export	in a variety of forms such as log files, packets, status indicators
Upgrade	Export of a configuration rife	State Secangs	and so on, which can help an administrator more easily find the system problem and fix it.
Auto Provision	Import CFG Configuration File	Browse No file selected Import	· Log Files
Configuration			 Capturing Packets Configuration File
Dial Plan	Pcap Feature	Start Stop Export	(*.cfg/*.bin)

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

How to export local log?

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

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

- 1. Click on Settings->Configuration.
- 2. Select Enabled from the pull-down list of Enable Local Logging field.
- 3. Select 6 from the pull-down list of Local Log Level.

The default local log level is "3".

- 4. Select sys.log from the pull-down list of Export Local Log.
- 5. Click **Confirm** to accept the change.

Yealink cp860			Log Out English(English)
	Status Account Network	Dsskey Features	Settings Directory Security
Preference	Export Import Config	Browse No file selected.	NOTE
Time & Date		Import Export	Configuration IP phones can provide feedback
Call Display			in a variety of forms such as log files, packets, status indicators
Upgrade	Export CFG Configuration File	Static Settings Export	and so on, which can help an administrator more easily find
Auto Provision	Import CFG Configuration File	Browse No file selected Import	the system problem and fix it.
Configuration			Capturing Packets Configuration File
Dial Plan	Pcap Feature	Start Stop Export	(*.cfg/*.bin)
Voice	Local Log		You can click here to get more guides.
Ring	Enable Local Log	Enabled -	
Tones	Local Log Level	б 🔻	
Softkey Layout	Max Log File Size (1024-2048KB)	1024	
TR069	Export Local Log	sys.log	
	Syslog		

- 6. Reproduce the issue.
- 7. Click Export to open the file download window, and then save the file to your local system.

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

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

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

- 1. Click on Settings->Configuration.
- 2. Select Enabled from the pull-down list of Enable Local Logging field.
- 3. Select **boot.log** from the pull-down list of **Export Local Log**.
- 4. Click Export to open the file download window, and then save the file to your local system.

How to export all diagnostic files?

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

1. Click on Settings->Configuration.

alink cps60			English(English)	Log
	Status Account Network	Dsskey Features	Settings Directory Secu	rity
Preference	Export Import Config	Browse No file selected.	NOTE	
Time & Date		Import Export	Configuration	
Call Display		-	IP phones can provide fe in a variety of forms such	n as k
Upgrade	Export CFG Configuration File	Static Settings Export	t files, packets, status indi and so on, which can he administrator more easly the system problem and	lp an find
Auto Provision	Import CFG Configuration File	Browse No file selecter Impor		
Configuration			Capturing Packets Configuration File	
Dial Plan	Pcap Feature	Start Stop Export	(# cfa/# bia)	
Voice	Local Log		You can click here to more guides.	get
Ring	Enable Local Log	Enabled +		
Tones	Local Log Level	6 •		
Softkey Layout	Max Log File Size (1024-2048KB)	1024		
TR069	Export Local Log	sys.log • Export		
	Syslog			
Voice Monitoring	Enable Syslog	Enabled •		
SIP	Syslog Server	10.3.5.21 Port 514		
Power Saving	Syslog Transport Type	UDP -		
	Syslog Level	6 •		
	Syslog Facility	local use 0 (local0) ·		
	Sysiog Prepena MAL	usaoled -		
	Export All Diagnostic Files	Start Stop Export		

2. Click Start to begin capturing signal traffic.

The local log level will be automatically set to 6.

- **3.** Reproduce the issue.
- 4. Click **Stop** to stop the capture.

The local log level will be reset to 3.

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

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

Reboot & Upgrade & Reset Issues

How to reboot the phone?

To reboot the phone via web user interface:

1. Click on Settings->Upgrade.

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

ealink cp860			Log English(English)
	Status Account Netwo	rk Dsskey Features Setting	gs Directory Security
Preference			NOTE
	Version		
Time & Date	Firmware Version	37.81.0.10	Reset to Factory Setting Resets the IP phone to facto
Call Display	Hardware Version	37.0.0.0.0.0	configurations.
Upgrade	Reset		Reboot Reboots the IP phone.
	Reset to factory	Reset to factory	Upgrading Firmware
Auto Provision	Reboot	Reboot	Upgrades firmware manually.
Configuration	Select And Upgrade Firmware	Browse No file selected.	You can click here to get
Dial Plan		Upgrade	more guides.

Note

Any reboot of the phone may take a few minutes.

How to upgrade firmware?

To upgrade firmware via web user interface:

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

Yealink CP860	Status Account N	etwork Dsskey	Features	Settings	Log Out English(English) • Directory Security
	Status Account N	etwork DSSRey	reatures	Jettings	Directory Security
Preference	Version				NOTE
Time & Date	Firmware Version	37.81.0.10			Reset to Factory Setting Resets the IP phone to factory
Call Display	Hardware Version	37.0.0.0.0.0			configurations.
Upgrade	Reset				Reboot Reboots the IP phone.
Auto Provision	Reset to factory	Reset t	to factory		Upgrading Firmware
Configuration	Reboot	Reboot			Upgrades firmware manually.
Configuration	Select And Upgrade Firmware	Browse	No file selected.		You can click here to get
Dial Plan		Upgrade			more guides.

3. Click Upgrade to upgrade the firmware.

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

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

How to reset the phone?

Reset the phone to factory configurations after you have tried all troubleshooting suggestions but do not solve the problem. You need to note that all customized settings will be overwritten after a reset.

To reset the phone via phone user interface:

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

The LCD screen prompts the following warning:



3. Press the OK soft key.

The LCD screen prompts the following:



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

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

Regulatory Notices

Service Agreements

Contact your Yealink Authorized Reseller for information on 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 RICE PAID, AT YEALINK'S OPTION. THIS DISCLAIMER OF LIABILITY FOR DAMAGES WILL NOT BE AFFECTED IF ANY REMEDY PROVIDED HEREIN SHALL FAIL OF ITS ESSENTIAL PURPOSE.

Safety Instructions

Save these instructions. Read these safety instructions before use!

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

\land General Requirements

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

A Environmental Requirements

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

🕂 Operating Requirements

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

Cleaning Requirements

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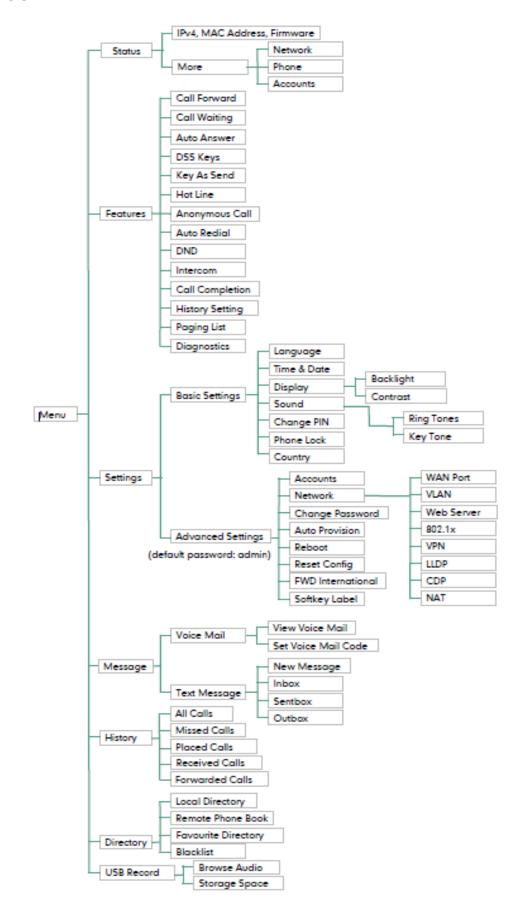
- Before cleaning the device, stop using it and disconnect it from the power supply.
- Use a piece of soft, dry and anti-static cloth to clean the device.
- Keep the power plug clean and dry. Using a dirty or wet power plug may lead to electric shock or other perils.

Appendix

Appendix A - Time Zones

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

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



Appendix B - Menu Structure

Appendix C – Unavailable Features for PSTN

The following lists the main features not available for PSTN:

- Call Completion
- DND
- Call Forward
- Call Transfer
- Call Waiting
- Call Park
- Call Pickup
- Anonymous Call
- Anonymous Call Rejection
- Intercom
- Multicast Paging
- Music on Hold
- SCA
- BLA
- Messages
- Programable Keys (Prefix, SMS, NewSMS, Forward, Direct Pickup, XML Group, XML Directory, Intercom, Group Pickup, Multicast Paging, Paging List, XML Browser and Zero Touch)

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