



IP Phone User Manual

☒ H60P

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Preface

Welcome

Thanks for choosing the H60P Hotel IP phone! We hope you will make full use of this rich-feature IP phone, Contact us if you need any technical support: 0755-61919966.

About This Manual

This manual provides information about the introduction of the IP phone, and about how to install, configure or use it. Please read this document carefully before install the H60P Hotel IP phone.

Intended Audience

This manual is aimed primarily at the following people:

- Users
- Engineers who install, configure, and maintain the H60P Hotel IP phone.

Revision Record

Document Name	Version	Firmware Version
H60P Hotel IP Phone User Manual	V1.0	2.60.11.12.13

Conventions

The phone or device mentioned in this document refers to the H60P Hotel IP phone.
"Note" marked in the document is what users need to pay attention to.

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1 Overview

1.1 Introduction

The H60P Hotel Phone is a cost-effective, multifunctional HD IP voice phone specifically designed for the hospitality industry to deliver an exceptional communication experience to clients. Its sleek and user-friendly design ensures compatibility with various IP voice telephone systems. The H60P series not only offers superior HD voice quality but also supports a wide range of telephony features, making it adaptable to the diverse operational needs of hotels.

The H60P phone integrates HD audio technology to ensure clear and smooth call quality. It supports 100 Mbps Ethernet and is compatible with PoE power supply, effectively reducing wiring costs. Additionally, it is equipped with six programmable keys that can be used for personalized configuration of various hotel service hotlines. Moreover, the H60P supports customization of the phone faceplate according to customer needs enhancing experience. By perfectly the user integrating with IPPBX, it offers a wide range of call service functions. The H60P supports features such as speed dial, call waiting, and voicemail, which are suitable for hotels, helping hotels enhance service quality and customer satisfaction.

1.2 Feature

➤ Highlights

- HD Voice
- 2 SIP Account
- 8 programmable soft keys
- 5-way Conference
- Dual-port 10/100Mbps Ethernet
- Support for customized hotel specialty faceplate
- Full Compatible with 3CX, Asterisk, Yeastar, Xorcom, VoIP.ms, Broadsoft, FreeSWITCH Platform etc

➤ Call Features

- 2 SIP Accounts
- Call Waiting
- Call Forward
- Blind/attendant Transfer
- Call hold, Mute, DND
- Speed Dial, Hotline
- Redial, Call Return
- Auto-redial, Auto-answer
- Dial Plan

- 5-way Calling
- Peer Dialing

➤ **Audio**

- HD Voice
- Wideband Audio Codec: G.722, OPUS-WB
- Narrowband Audio Codec: PCMA, PCMU, G.729, G723, G726, AMR, OPUS, iLBC
- Echo Cancellation/Dynamic Jitter
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Full-duplex Speaker

➤ **Protocols & Standards**

- SIP v1 (RFC2543), v2 (RFC3261)
- SIP over TLS, SRTP
- TCP/IP/UDP
- RTP/RTCP, RFC2198, RFC1889
- HTTP/HTTPS/FTP/TFTP
- ARP/RARP/ICMP/NTP/DHCP
- DNS SRV/ A Query/NATPR Query
- STUN, Session Timer (RFC4028)
- DTMF: In-Band, RFC2833, SIP INFO
- LLDP
- OpenVPN, 802.1X

➤ **Phone Features**

- Adjustable Volume
- Selectable Ring Tones
- Action URL/URI
- Blacklist: 100 Groups
- Call Log: 1000 Logs
- Support 5 Remote Phonebook URLs
- LDAP
- Support Multiple Languages

➤ **IP-PBX Feature**

- SMS, Voicemail, MWI Message Notification
- Music On-hold, Intercom, Multicast
- Call Pickup, Call Pickup in Group

- Anonymous Call
- Anonymous Call Rejection

➤ **Management**

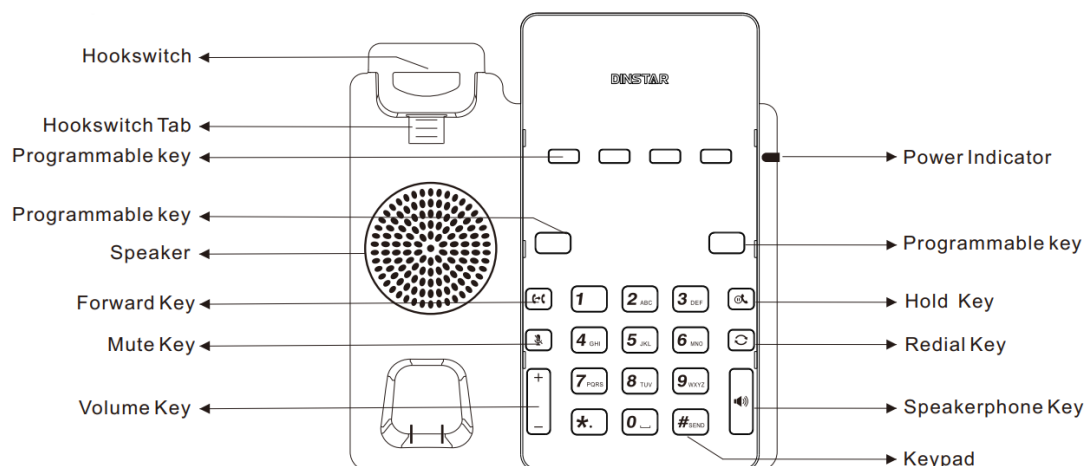
- Auto provisioning: FTP/TFTP/HTTP/HTTPS/PnP
- Configuration via HTTP/HTTPS web
- SNMP, TR069
- Network capture
- NTP/Daylight saving time
- Firmware upgrade via web
- Syslog
- Configuration backup/restore
- Equipment Management System (NMS)

➤ **Physical Features**

- Dimensions(approx.):180x192x36mm (W*L*D) w/o foot stand
- Weight(approx.): device 370g
- Input voltage: 5Vdc
- Power supply: 5Vdc power adapter or PoE
- Handset: 4P4C, RJ9
- Headset: 4P4C, RJ9
- Environmental:
 - Operation temperature: 0~40℃
 - Humidity: 10% to 95% (non-condensing)
- 2xRJ45 10/100M Ethernet Ports
- Installation method: Desktop

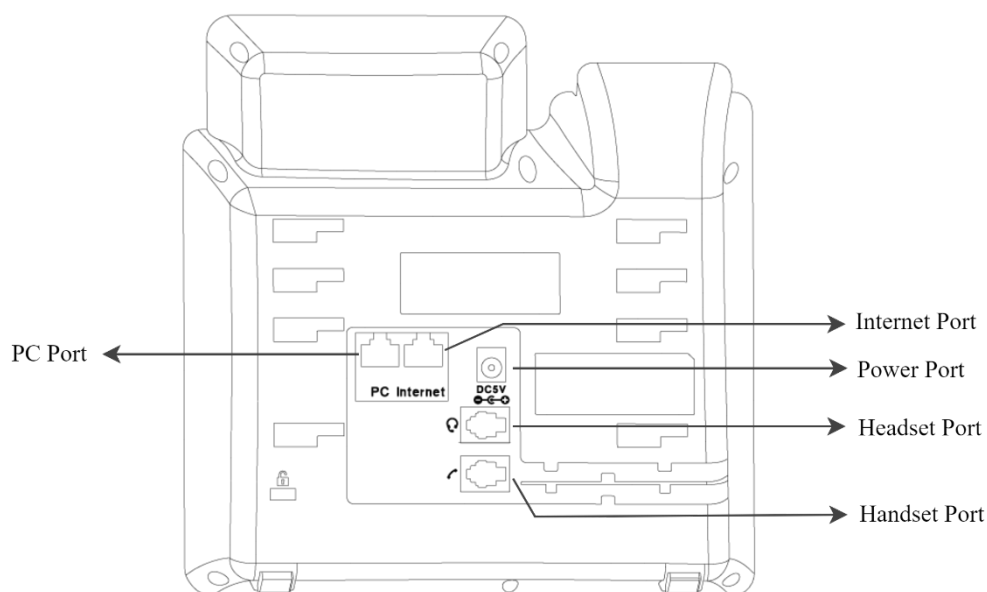
1.3 Hardware Introduction

1.3.1 Front View and Keypad Description



Name	Description
Power LED Indicator	The power indicator indicates the call, message and system status: Solid blue: The phone is idle. Fast-flashing blue: The phone is ringing. Slow-flashing blue: The phone receives a message.
Cancel/Reject Key	Cancel the current actions or used as the return key
Soft Keys	Used to complete the operations about the soft key labels. The labels include: History, Favorites, Redial, Call Return, Hot Desking, XML Browser, DND, Menu, MSG, Status, Book, FWD, Pickup, Intercom, Speed Dial, etc.
Hold Key	Press this key during an active call, and you can place a call on hold.
Mute Key	Toggles the microphone on or off
Speakerphone Key	Press this key to activate the speakerphone mode, and you can make the call hand-free.
Keypad	Provide digits, letters and special characters in context-sensitive application.
Volume -/+	Turn down or turn up the volume by pressing the “-” key or the “+” key.
Headset Key	Use the headset key to activate headset mode.
Speaker	Provides hands-free (speakerphone) audio output.
Hookswitch Tab	Secures the handset in the handset cradle when the IP phone is mounted vertically.
Hookswitch	- Picking up the handset from the handset cradle, the hookswitch bounces, and the phone connects to the line. - Put the handset down on the handset cradle, the phone disconnects from the line.

1.3.2 Rear View and Panel Descriptions



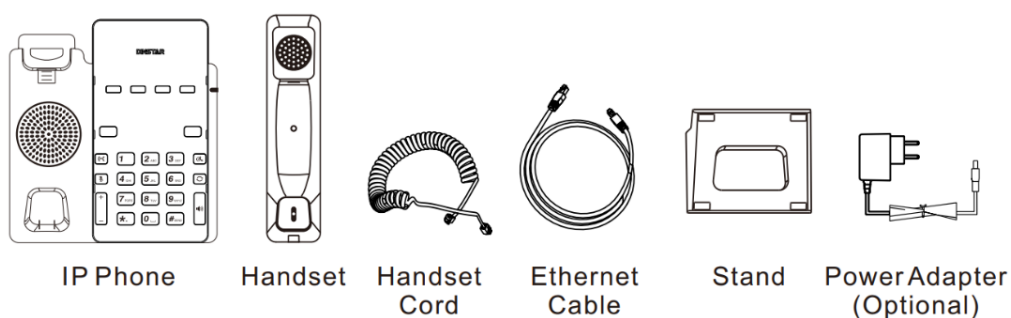
Name	Description
Power Port	Connect to power adapter (Input: 5V DC 1A)
PC Port	Connect to PC (10/100M)
Internet Port	Connect to Network (10/100M)
Handset Port	Connect to handset (Port type: RJ-9 connector)
Headset Port	Connect to headset (Port type: RJ-9 connector)

2 Installation

2.1 Inspection

Check the device and accessories with the device as per the list of packing (See the table below).

Note: If the IP phone is powered by PoE, no power adapter will be provided.

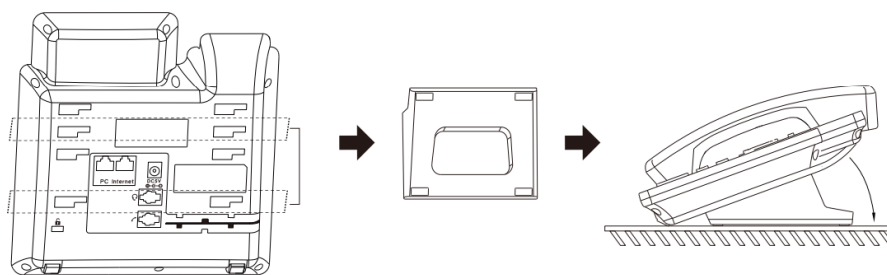


Name	Quantity
H60P IP Phone	1
Stand	1
Handset	1
Handset cord	1
Power Adapter (Optional)	1
Ethernet Cable	1

2.2 Installation

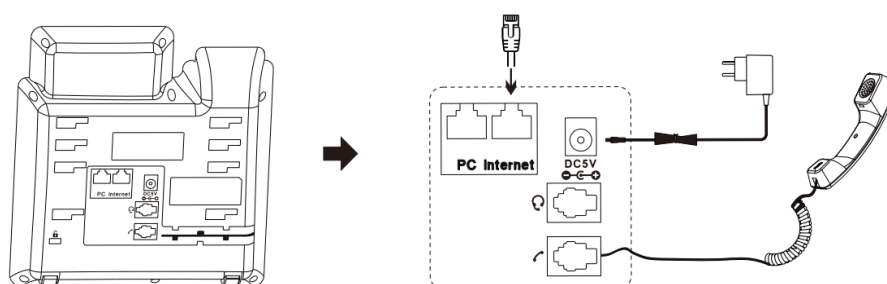
Step 1: Installing the stand

H60P IP phone supports desktop installation, suitable for hotel desk setups. as shown in the figure below. Align the stand with the slot and push it to the top evenly, making sure it is fixed.



Step 2: Connecting power, Internet and handset

- 1) Connect the provided power adapter to the power port, and plug the other end of the adapter into an available power socket. After the power is connected, the power indicator light of the phone will light up.
- 2) Connect one end of the Ethernet cable to the Internet port on the back of the H60P phone and the other end to the network jack.
- 3) Connect one end of the Handset cord to the handle, and the other end to the Handset port, as shown in the figure below.



Note:

- Never use a power adapter other than the one provided with Dinstar H60P.
- Internet port supports POE.

Step 3-Getting Started

After the phone is connected to the power and network, the phone will automatically start up and the common functions of the phone can be used normally (see section 3 for more details). After dial *158# to obtain the phone's IP address, you can customize more functions through the Web platform(see section 4 for more details).

3 Functions

3.1 Making a Call

3.1.1 Call Modes

You have the following three ways to make a call.

- 1) **Handset mode:** Pick up the handset, the  icon will be displayed in the upper left corner of the screen. And then enter a number or select a contact.
- 2) **Speakerphone mode:** Press the **Speakerphone Key** , the  icon will be displayed in the upper left corner of the screen. And then enter a number or select a contact.

You can also enter a number first, and then choose the method you will use to speak to the other party.

3.1.2 Methods to Make Calls

Making a call from the Dialer

According to the number you input from the dial pad, the screen will display the previously saved contacts. You can select the contact you want to call from the list, or enter the number you want to call directly.

Procedure

- 1) Do one of the following:
 - Enter the number from the dial
 - Press the Redial key to dial the last call number.
 - Set a programmable key as a speed dial and quickly dial the number by pressing the speed dial button
- 2) Press **Dial Key**  from keypad.

Note:

You can set the dial key through *Phone>Key>Soft Key*.

Making multiple calls

During the call, you can keep the current call and then make other calls.

Procedure

- 1) Press the **Hold Key**  to hold the current call;
- 2) Enter the number you want to call;
- 3) Press the **Dial Key**  from keypad.

Making a call with Speed Dial Key

You can use speed dial keys to call the number you dialed often or the number which it is hard to remembered. The steps for setting the quick dial key are as follows:

Procedure

- 1) Log in to the Web UI and navigate to *Phone> Key> Soft Key*;
- 2) Pick a soft key from the list
- 3) Select **Speed Dial** in the Type field;
- 4) Enter the number that will be set as the speed dial in the value field
- 5) In the account field, select the account that is set as a speed dial key and used for outgoing calls;
- 6) Click the **Submit** button to complete the setting.

Calling back the last call

You can use the redial key to quickly dial the last call.

Procedure

- 1) Log in to the Web UI and navigate to *Phone> Key> Soft Key*;
- 2) Pick a soft key from the list
- 3) Select **Callback** in the Type field;
- 4) Click the **Submit** button to complete the setting.

3.2 Answering the Call

Manually answer a call

The phone will ring when receiving an incoming call. You can answer the call manually.

Procedure

Do one of the following:

- Pick up the handset to answer;
- Press the Speakerphone Key  to answer;

Answering a new incoming call

You can answer a new incoming call while you are on a call. When there is a new incoming call, the phone emits a "beep".

Procedure

After selecting the **Answer** soft key to answer the new call, the previous call is put on hold

Auto answer

When the phone is idle, you can enable the speakerphone mode to automatically answer incoming calls.

Procedure

- 1) Log in to the Web UI and navigate to *Account> Sip Param> Service Param* ;
- 2) Select the account that you want to activate auto answer first;
- 3) In the auto answer bar, select **Enable** .
- 4) Click the **Submit** button to complete the setting.

3.3 Silencing or Rejecting the Call

You can ignore or reject the call when you receive an incoming call.

Silencing call

If you ignore the incoming call, the phone will no longer ring.

Procedure

Press the **Mute Key** .

Using Do Not Disturb (DND) to reject all calls

If you don't want to be disturbed by incoming calls, you can use the Do Not Disturb function to automatically reject all incoming calls or all incoming calls from a specified account. For details, please refer to 3.7 Do Not Disturb> Enable Do Not Disturb in Default Mode..

Using the Do Not Disturb (DND) to reject all calls from a specified account

By default, the DND mode applies to all accounts. You can set the mode through the web system to enable the DND function of a specific account. For the specific configuration, please refer to 3.7 DND> Enable Do Not Disturb in Custom Mode.

3.4 Ending Calls

You can end the current call at any time.

Procedure

Do one of the following:

- If the phone works in handset mode, press hang up the call;
- If the phone works in speakerphone mode, press the **Speakerphone** key .

3.5 Mute

When you enable the mute function during a call, the other party will not receive your voice message, but you can.

Procedure

- 1) Press the **Mute Key**  to enable the mute function, the other party will not hear you, but you can.
- 2) Press the **Mute Key**  again to disable the mute mode.

3.6 Holding and Resuming Call

You can put a call on hold and resume the call at any time. The held party may hear the music sent from the PBX server while waiting.

Procedure

- 1) During the call, press the **Hold Key**  .
- 2) Press the **Hold Key**  again or select **Resume** to resume the call.

3.7 DND

If you don't want to be disturbed by incoming calls, you can set the DND function. The DND feature is applicable to all accounts and can also be directed to a specific account. And we also provide you with a DND Timer Enable.

The DND setting depends on the phone mode: phone mode or custom mode (the default is phone mode). In the phone mode, the DND function will be enabled for all accounts of the phone by default; in the custom mode, Account 1 is selected by default, and the DND function can be set for a specific account. Both the mode needs to be set on the web system.

DND function is enabled in default mode

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Forward&DND**.
- 2) In the DND bar, select **Enable**.
- 3) (Optional) Input the value in the area of On code and Off code.
- 4) Click the **Submit** button to complete the setting.

DND function is enabled in custom mode

1. Setting custom mode on the Web page

You can log in to the phone with the account User, Navigate to **Phone> Forward & DND**, as shown in the figure below. Select the phone mode on this page.

Sync Feature Code	
Active	Disabled 
Mode	☐ Phone ☒ Custom

2. Enabling DND for specific accounts

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Forward & DND > DND**;
- 2) Select an account;
- 3) In the DND bar, select **Enable**;
- 4) (Optional) Input the value in the area of On code and Off code;
- 5) Click the **Submit** button to complete the setting.

DND timer enable

You can enable the DND timer according to your work and rest habits, and reject all calls during the period you set. There are three sets. The range of the three periods can overlap.

Log in to the phone with the account User, navigate to Phone> Forwarding & DND, set DND Timer to Enable, then set the period, and finally select Submit to save the configuration.

DND	
Account	Account 1
DND	Enabled
Return Code When DND	486 (Busy Here)
DND On Code	
DND Off Code	
DND Timer Enable	Enabled
DND Timer 1	0 Hour 0 Min - 23 Hour 59 Min
DND Timer 2	0 Hour 0 Min - 23 Hour 59 Min
DND Timer 3	0 Hour 0 Min - 23 Hour 59 Min

Disabling DND

When you need to answer a call, you can disable the DND function.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Forward & DND > DND**;
- 2) Select a specified account or select all accounts;
- 3) In the DND bar, select **Disable**;
- 4) Click the **Submit** button to complete the setting.

3.8 Call Waiting

You can enable the call waiting function so that you can answer new calls during the call, otherwise, the new call will be automatically rejected and a busy tone will be displayed. You can enable the call waiting tone function at the same time. When you receive a new call during a call, The phone will emit a "beep" tone to indicate a new call.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Call Feature> Call Waiting**.
- 2) Select **Enable** to active call waiting.
- 3) (Optional) Select to **Enable** tone.
- 4) Click the **Submit** button to complete the setting.

3.9 Call Completion

When you fail to call out, you can use the call completion function to monitor the called party and re-establish the call when the called party is free.

Note:

Not all servers support the call completion. Please consult your system administrator for further information.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Call Feature> Call Completion**.
- 2) Enable call completion.
- 3) Click the **Submit** button to complete the setting.

3.10 Call Forward

When you cannot handle the incoming call on time, you can forward the incoming call to an available number.

Setting the type of forwarding calls:

Always Forward: All incoming calls will be automatically forwarded to the pre-configured number

Busy Forward: When the phone is busy, the incoming call will be forwarded to the pre-configured number

No Answer Forward: When the phone does not answer within the set time, the incoming call will be forwarded to the pre-configured number.

Forwarding all calls

After enabling call forwarding, you can forward all incoming calls on the phone. The activation method is as follows:

Note:

Some servers need to set the enable code and the disable code for call forwarding. Please consult your system administrator for further information.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Forward & DND> Forward Transfer**;
- 2) Select a call forwarding type, and then select **Enable**;
- 3) Enter the number to be forwarded in the forward to area;
- 4) If you choose no answer forward, enter the time in the **No Answer Ring Time** bar;
- 5) (Optional) Input the value in the area of On code and Off code;
- 6) Click the **Submit** button to complete the setting.

Forwarding the call to the designated account

By default, enabling call forwarding applies to all accounts. You can enable call forwarding for specific account by setting the phone mode.

Setting the custom mode on the Web page

You can log in to the Web with the account user, navigate to **Phone> Forwarding & DND**, set the phone mode to Custom, and select Submit to save the configuration.

Sync Feature Code	
Active	Disabled ▼
Mode	☐ Phone ☒ Custom

Disabling call forward.

When you no longer need to forward incoming calls, you can choose to disable the call forwarding.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Forward & DND> Forward Transfer**;
- 2) (Optional) If the call forward works in a custom mode, you need to select an account;
- 3) Select the call forwarding type, and then select **Disable**;
- 4) Click the **Submit** button to complete the setting.

3.11 Call Transfer

You can transfer calls to other during a call. There are the following three transfer methods:

Blind transfer: Transfer the call to a third party directly without prior notice;

Semi-Attended Transfer: When you hear the back tone, transfer the call to a third party;

Attended Transfer: After the third party answers the call and consults, the call is transferred to the third party.

3.11.1 Blind Transfer

You can transfer directly without notifying the third party, as follows:

Procedure

- 1) Press the soft key **Trans** during the call;
- 2) Enter the number to be transferred;
- 3) Press the soft key **Trans** again.

Note: If the phone works in handle mode, you can directly put the handle back to complete the transfer.

3.11.2 Semi-Attended Transfer:

You can transfer the call to the third party after hearing the back tone.

Procedure

- 1) Press the soft key **Trans** during the call;
- 2) Enter the number to be transferred;
- 3) Press **“#”** key;

- 4) After hearing the ring back, press the soft key **Trans** again to complete the semi-attended transfer.

Note:

You can directly put the handle back to complete the transfer in the handset mode.

3.11.3 Attended Transfer

You can transfer the call to the third party after hearing the back tone.

Procedure

- 1) Press the soft key **Trans** during the call;
- 2) Enter the number to be transferred;
- 3) Press “#” key;
- 4) After the third-party answers and consults, press the soft key **Trans** again to complete the attended transfer

Note:

You can directly put the handle back to complete the transfer in the handset mode.

3.12 Call Park

You can use the call park function to maintain the current conversation, and then resume the call on another. According to different parking modes, it can be divided into code mode and call transfer mode.

Note:

Not every server supports call park and the support methods are also different. Please contact your administrator to confirm whether this feature is available.

3.13 Pickup

You can use the pickup feature to answer calls from others. There are following two pickup modes:

Directed Pickup: Answer incoming calls from designated number.

Group Pickup: Pickup calls from any member in the same group. Group pickup needs to set up group members in advance.

3.13.1 Directed Pickup

You can answer incoming calls from a designated number.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Key> Soft Keys**;
- 2) Select **Pickup** in the type bar;
- 3) Enter the pickup code in the value area and the account to be picked
- 4) Click the **Submit** button to complete the setting.

When the designated number receives an incoming call, press the pickup key to allow the user's incoming call to be answered on your phone.

3.13.2 Group Pickup

You can answer an incoming call in the same group. If the group receives multiple incoming calls, it will answer the first by default.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Key> Soft Keys**;
- 2) select **Group Pickup** in the type bar;
- 3) Enter the group pickup code in the value bar;
- 4) Click the **Submit** button to complete the setting.

When there is an incoming call in the group, press the group pickup key to make the incoming call in the group be answered on your phone.

3.14 Speed Dial

You can use the speed dial to quickly dial the specified number.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Soft Keys> Soft Key**;
- 2) select **Speed Dial** in the type bar;
- 3) Enter the number in the value area;
- 4) The account bar is to select an account to make a call;
- 5) Click the **Submit** button to complete the setting.

Press the speed dial key when idle, and the selected account will automatically dial the preset number.

3.15 Auto Redial

You can use the automatic redial feature when the called party does not answer.

Note:

Not all servers support the auto redial feature. Please consult your administrator for further information.

Procedure

- 1) Log in to the Web UI and navigate to **Phone>Call Features> Auto Redial**;
- 2) Select enable in the Auto Redial bar; fill in the time in the interval area; fill in the number of redial in the times bar;
- 3) Click the **Submit** button to complete the setting.

After the phone call fails, an automatic redial countdown for the auto redial. You can wait for the countdown to finish the auto redial.

3.16 Intercom

You can make an intercom call to a contact. When receiving an intercom call, the phone will trigger an alert tone and automatically answer it in Speakerphone mode. The intercom feature is mainly used as the configuration of the receiver.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Intercom**;
- 2) Select **Enable** in the active bar;
- 3) (Optional) Select **Enable** in the mute bar;
- 4) Click the **Submit** button to complete the setting.

If the intercom mute is enabled on the phone, that is, the automatically answered intercom call is in the mute, you must press the Mute key  to cancel.

When intercom supported by the server, you need to dial the code of the intercom feature + the extension number of the other party to initiate an intercom call. Take a 3CX server as an example: dial *998100, where *998 is the intercom code, and 100 is the extension number. For the codes of the intercom feature of other servers, please refer to the configuration guide documents of the servers.

3.17 Call Return

You can make the last call through the call return key.

Procedure

- 1) Log in to the Web UI and navigate to **Phone> Key> Soft Key**;
- 2) Select **CallBack** in the type bar;
- 3) Click the **Submit** button to complete the setting.

After the callback key is configured, press the callback key on the phone to automatically dial the last call.

4 Web Platform

4.1 Login & Logout

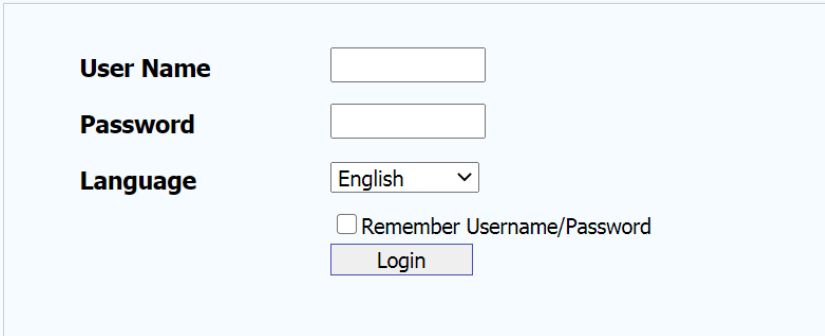
The Web system of the H60P is designed for users or administrators to configure or manage IP phones.

Checking the IP address of the phone and log in to the web system:

- 1) Enter ***158#** on the phone to broadcast the phone IP address.
- 2) Input the IP address of the phone (such as <http://172.31.2.167>) in the browser address box and press Enter.
- 3) If successfully connected to the IP phone, The login GUI will be displayed. On this page, enter the correct username and password. By default, username and password are as below:

User Name: admin

Password: admin

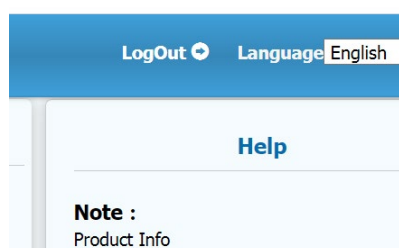


The screenshot shows a login interface with the following elements:

- User Name:** A text input field.
- Password:** A text input field.
- Language:** A dropdown menu currently set to "English".
- ☐ Remember Username/Password
- Login:** A button to submit the login information.

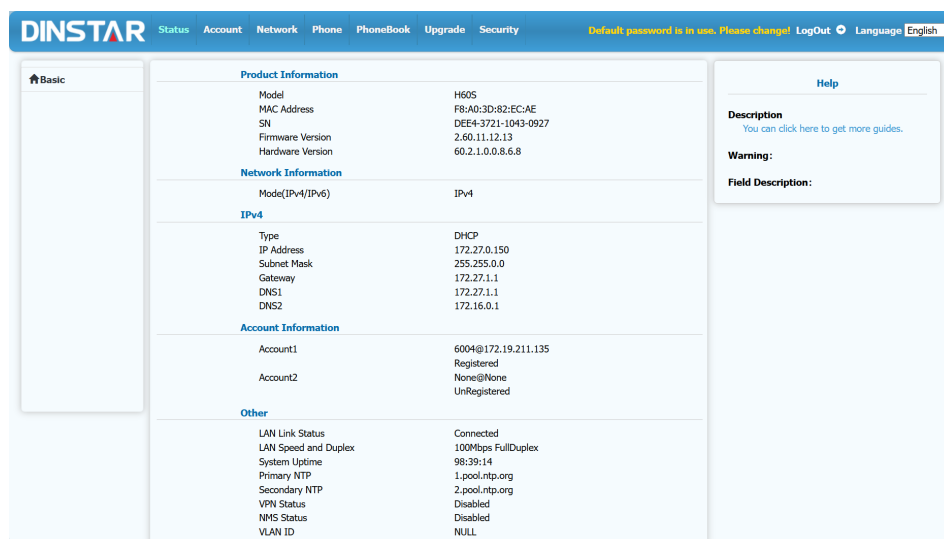
Log out:

You can log out of the web system by clicking log out in the upper right corner of the window.



4.2 Status

4.2.1 Status->Basic



Sections	Description
Product Information	It provides the device's information such as Model name, MAC address (IP device's physical address), Firmware version and Hardware version.
Network Information	Show you the network mode (LAN port) of the device.
IPv4	It provides the information of the LAN port IPv4 type (may be DHCP/static/PPPoE), IP address, subnet mask, gateway, preferred DNS server and alternative DNS server.
IPv6	It provides the information of the LAN port IPv6 type (may be DHCP/static/PPPoE), IP address, gateway, preferred DNS server and alternative DNS server.
Account Information	It provides the device's Account information and Registration status (account username, registered server's address, and Register result).
Others	Show you LAN port connection status, running time, primary NTP server, secondary NTP server (NTP server is used to automatically synchronize time from the Internet), VPN status and NMS status.

4.3 Account

4.3.1 Account->Basic

SIP Account	
Status	Registered
Account	Account1: 6004
Active	Enabled
Display Label	6004
Display Name	6004
Register Name	6004
Username	6004
Password	*****
SIP Server 1	
Server IP	172.19.211.135
Registration Expires	1800 (30~65535s)
Port	5060
SIP Server 2	
Server IP	
Registration Expires	1800 (30~65535s)
Port	5060
SIP Server 3	
Server IP	
Registration Expires	1800 (30~65535s)
Port	5060
Registration Params	
Registration Type	Concurrent
Registration Fallback	Disabled
SIP Registration Retry Timer	30 (1~1800s)
Outbound Proxy Server	
Active	Disabled
Server IP	
Backup Server IP	
Proxy Fallback Active	Disabled
Proxy Fallback Interval	3600 (35~65535s)
Port	5060
Transport Type	
Transport Type	UDP
NAT	
NAT Type	Disabled
Stun Server Address	
Static NAT Address	
Via Use Type	NAT Address
SDP Use Type	NAT Address
Port	3478
VPN	
VPN Preferred	Enabled
Submit Cancel	

Sections	Description
SIP Account	For the specified account settings. - Status: To display register result. - Account: SIP account - Active: To enable or disable SIP account

Sections	Description
	• Display Name: Which is sent to the other call party for display. • Register Name: Allocated by SIP server provider, used for authentication. • Username: the extension name assigned by the SIP server. • Password: Used for authorization.
SIP Server 1	For the primary SIP server settings. • Server IP: SIP server address, it could be an URL or IP address. • Registration Expires: The registration will expire after Registration period, the IP phone will re-register automatically within registration period.
SIP Server 2	For the secondary SIP server 2 settings. This is for redundancy, if registering to Primary SIP server fails, the IP phone will go to Secondary SIP server 2 for registering. Note: Secondary SIP server 2 is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.
SIP Server 3	For the secondary SIP server 3 settings. This is for redundancy, if registering to Primary SIP server and Secondary SIP server 2 fails, the IP phone will go to Secondary SIP server 3 for registering. Note: Secondary SIP server 3 is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.
Registration Param	• Registration Type: Set the registration type, can set Concurrent / Successive • Registration Fallback: When the registration type is successive, we can enable registration fallback; • SIP registration retry time: Set the SIP registration retry time (1~1800 seconds).

Sections	Description
Outbound Proxy Server	For the Outbound Proxy server settings. An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server. Note: If configured, all SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.
Transport Type	It configures Transport type for SIP message • UDP: UDP is an unreliable but very efficient transport layer protocol. • TCP: Reliable but less-efficient transport layer protocol. • TLS: Secured and Reliable transport layer protocol. • DNS-SRV: A DNS RR for specifying the location of services.
NAT	NAT (Net Address Translator) settings. • STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues. Note: By default, NAT is disabled.
VPN	For a specific account configure VPN options. • VPN preferred: If enabled, the designated account will preferentially use the VPN address for communication. If disabled, the specified account uses the local address to communicate. Note: By default, VPN priority is enabled.

4.3.2 Account->SIP Param

SIP Account		
Account	Account1: 6004 ▼	
Service Param		
Min Local SIP Port	20000	(1024~65535)
Max Local SIP Port	20299	(1024~65535)
Use New SIP Port When Registration Fail	Enabled ▼	
Call without Registration	Disabled ▼	
MWI Subscribe	Disabled ▼	
MWI Subscribe Period	1800	(120~65535s)
Voice Mail Number	*170*2#	
BLF Expire	1800	(120~65535s)
ACD Expire	1800	(120~65535s)
Caller ID Header	FROM ▼	
Auto Answer	Disabled ▼	
Ringtones	Default ▼	
PTime	20 ▼	
Anonymous Call	Disabled ▼	
Anonymous Call Rejection	Disabled ▼	
Privacy Number	None ▼	
Missed Call Display on LCD	Enabled ▼	
Trust SIP Server Only	Disabled ▼	
DNS Query Mode	A/AAAA ▼	
DNS Cache	Enabled ▼	
DNS Cache Time(0:Not Refresh)	0	(0~3600s)
Sync Time from SIP Server	Disabled ▼	
SIP Registration Expires (Percentage)	50	(10%~90%)
SIP Request Retry Count	0	(0~8)
Compatibility Param		
User-Agent		
PRACK	Disabled ▼	
PRACK Only for 18x with SDP	Disabled ▼	
Invite with user=phone	Disabled ▼	
INVITE with P-Preferred-Identity Header(RFC3325)	Disabled ▼	
TLS version	Adaptive ▼	
Session Timer	Disabled ▼	
Session Expire	1800	(90~7200s)
Session Refresher	UAC ▼	
Session Refresh Method	UPDATE ▼	
Use IPv6 Stateless Address	Enabled ▼	
Unregister When Reboot	Disabled ▼	
Escape special characters in URL	Enabled ▼	
Send MAC Info	Register ▼	
MAC Header Separator	:	
Displayname With Quotes	Enabled ▼	
RTP Mode in SDP when Call Holding	sendonly ▼	
Value of 'Refer To' refers to 'Contact'	Disabled ▼	
Third Party Do Not Send 18x Response	Disabled ▼	
Support P-Early-Media	Enabled ▼	
18x Response Preferred	18x Response with 1800s ▼	
Support eSpace Linkage	Disabled ▼	
Submit		Cancel

Sections	Description
SIP Account	Show you current Account settings or to select which account to display.
Service Param	• Min Local SIP Port • Max Local SIP Port • Use New SIP Port When Registration Fail • Call without Registration • MWI Subscribe • MWI Subscribe Period • Voice Mail Number • BLF Expire • ACD Expire • Caller ID Header • Auto Answer • Ringtones • PTime • Anonymous Call • Anonymous Call Rejection • Privacy Number • Missed Call Display on LCD • Trust SIP Server Only • DNS Query Mode • DNS Cache • DNS Cache Time • Sync Time from SIP Server • SIP Registration Expire(Percentage) • SIP Request Retry Count
Compatibility Param	• User-Agent • PRACK • PRACK Only for 18x with SDP • Invite with user=phone

Sections	Description
	• INVITE with P-Preferred-Identity Header (RFC3325) • TLS version • Session Timer • Session Expire • Session Refresher • Session Refresh Method • Use IPv6 Stateless Address • Unregister When Reboot • Escape special characters in URL • Send MAC Info • MAC Header Separator • Displayname With Quotes • RTP Mode in SDP when Call Holding • Value of 'Refer To' refers to 'Contact' • Third Party Do Not Send 18x Response • Support P-Early-Media • 18x Response Preferred

4.3.3 Account->Advanced

SIP Account	
Account	Account1: 7777 ▼
Hotline	
Active	Disabled ▼
Number	
Delay Time	4 (0~5s)
Keep Alive	
Active	Disabled ▼
Keep Alive Interval	15 (5~3600s)
Keep Alive Error Code	(100~699)
DTMF	
Type	RFC2833 ▼
DTMF Info Type	DTMF-Relay ▼
DTMF Payload	101 (96~127)
Encryption	
SRTP	Disabled ▼
NAT	
UDP Keep Alive Messages	Enabled ▼
UDP Alive Msg Interval	30 (5~60s)
Conference	
Type	Local ▼
Conference URI	
Music Server	
Active	Disabled ▼
Music Server	
BLF List	
BLF List URI	
BLF List Pickup Code	
BLF List Barge in Code	
Broadsoft	
AoC	Disabled ▼
Submit Cancel	

Sections	Description
SIP Account	Show you current Account settings or to select which account to display.
Hotline	The hotline allows users to automatically dial a defined number after hearing a dial tone without dialing any number. Function on: Enable or disable the hotline function. Number: Set the defined hotline number. Delay: Set the interval for automatic outgoing calls after hearing the dial tone.
Keep Alive	- Active: When enabled, it will send an Option message to the server; - Keep Alive Interval: The interval time between sending option messages; Keep Alive Error Code: A response code used to determine network failures.
DTMF	It configures DTMF feature. - Type: Support Inband, Info, RFC2833 or their combinations. - DTMF Info Type: Only available when DTMF Type is Info. - DTMF Payload: To configure payload type for DTMF. Note: By default, DTMF type is RFC2833 which is the standard. Type Inband uses inband frequency to indicate DTMF tone which is most used to be compatible with traditional telephone server. Type Info use SIP Info message to indicate DTMF message.
Encryption	It enables or disables SRTP feature. Voice Encryption (SRTP): If enabled, all audio signal(technically speaking it's RTP streams) will be encrypted for more security.
NAT	NAT-related settings. - UDP Keep Alive message: If enabled, IP phone will send UDP keep-alive message periodically to router to keep NAT port alive. - UDP Alive Msg Interval: Keep alive message interval. Rport: Remote Port, if enabled, it will add Remote Port into outgoing SIP message for designated account.
Conference	To select Local or network conference. Type: To select desired conference type

Sections	Description
	Conference URI: If network conference is selected, a network conference URI is needed.
Music Server Address	It configures third-party MOH (music-on-hold) server. - Active: To enable or disable this MOH server, If enabled, the IP phone will play MOH from configured server. - Music Server : To configure MOH server address.
BLF List	It configures BLF List URI address. - BLF List URI: BLF List is short for Busy Lamp Field List. - BLF List PickUp Code: To set the BLF pick up code. - BLF List Barge In Code: To set the BLF barge in code.
Broadsoft	It configures Broadsoft AOC feature. AoC: A feature used to be accounting on Broadsoft platform. Note: Please consult your administrator further information.

4.3.4 Account-> Audio Codecs

SIP Account

Account

Account1: 7777

Prefer Codec

Remote

Audio Codecs

Disabled Codecs

G723_53
G723_63
G726-16
G726-24
G726-32
G726-40
iLBC_13_3
iLBC_15_2
AMR
OPUS
OPUS-WB

>>

<<

Enabled Codecs

G722
PCMU
PCMA
G729

↑

↓

Submit

Cancel

Sections	Description
SIP Account	Show you current Account settings or to select which account to display.
Audio Codecs	To display and configure available/unavailable codecs list. Codec means coder-decoder which is used to transfer analog signal to digital signal or vice versa. Familiar codecs are PCMU(G711U), PCMA(G711A), G722(wide-bandth codecs), G729, G723_53, G723_63, G726_16, G726_24, G726_32, G726_40.

4.4 Network

4.4.1 Network->Basic

LAN Port Mode

Mode(IPv4/IPv6)

IPv4

LAN Port IPv4

☒ DHCP
 ☐ Static IP

IP Address

192.168.1.240

Subnet Mask

255.255.255.0

Default Gateway

192.168.1.1

Static DNS

☐ ON
 ☒ OFF

DNS1

8.8.8.8

DNS2

LAN Port IPv6

☐ DHCP
 ☐ Static IP

IP Address

Prefix Length

Default Gateway

Static DNS

☐ ON
 ☒ OFF

DNS1

DNS2

LAN PPPoE

☐ PPPoE

Username

Password

Speed and Duplex

LAN port

Auto Negotiation

PC port

Auto Negotiation

Submit

Cancel

Sections	Description
LAN Port Mode	It configures LAN port mode, IPv4, IPv6, IPv4&IPv6 are optional.
LAN Port IPv4	It configures the LAN port IPv4: DHCP: If enabled, the IP phone will automatically obtain the IP address, subnet mask, default gateway and DNS

Sections	Description
	server address from the DHCP server. Static IP: If enabled, you must manually set the IP address, subnet mask, default gateway and DNS server.
LAN Port IPv6	It configures the LAN port IPv6: - DHCP: If enabled, the IP phone will automatically obtain the IP address, prefix length, default gateway and DNS server addresses from the DHCP server. - Static IP: If enabled, you must manually set the IP address, prefix length, default gateway and DNS server.
LAN PPPoE	LAN PPPoE settings: PPPoE: Use PPPoE username & password to connect to PPPoE server.
PC Port	PC Port settings. As Bridge: If selected, IP phone will act as a switch to route all incoming and outgoing packets from PC port.
Speed and Duplex	Set the network full-duplex, half-duplex and speed

4.4.2 Network-Advanced

LLDP	
Active	Enabled v
Packet Interval	60 (10~3600s)
Local RTP	
Min RTP Port	11800 (1024~65535)
Max RTP Port	12000 (1024~65535)
Web Service	
Web Type	HTTPS v
HTTP Port	80 (80/(1024~65535))
HTTPS Port	443 (443/(1024~65535))
DHCP Option	
Option 12 (Hostname)	
TR069 Option	246 (0/128~254)
TR069 URL Prefix	acs.address=
Registration Random	
Registration Random Time	0 (0~60s)
SSH	
Active	Disabled v
OpenVPN	
Active	Disabled v
Upload(<50K)	Browse not selected
	Upload
Submit Cancel	

Sections	Description
LLDP	It configures LLDP feature. - Active: To enable or disable LLDP feature. - Packet Interval: To configure the interval for LLDP admin message. Note: LLDP stands for Link Layer Discovery Protocol, it's used to exchange device information between any two directly-connected devices. LLDP is often used to configure Voice VLAN automatically for IP phone.
Local RTP	Local RTP settings.

	- Max RTP Port: Determine the maximum port that RTP stream can use. - Min RTP Port: Determine the minimum port that RTP stream can use.
DHCP Option	Option 12 (Hostname): Custom phone host name (if blank, the host name is phone model).
SSH	To display and configure SSH settings: Active: enable or disable this function.
OpenVPN	VPN settings. - Active: To enable or disable VPN feature. - Upload: To upload VPN client configuration file which is used to connect to VPN server. Note: For now, IP phone can only support OpenVPN.

4.4.3 Network ->TR069

TR069

ACS

Active

Disabled

Version

1.0

URL

Username

Password

.....

Periodic Inform

Active

Disabled

Periodic Interval

1800

(3~24x3600s)

CPE

URL

Username

Password

.....

Submit

Cancel

Sections	Description
TR069	TR069 settings. - Active: To enable or disable TR069 feature. - Version: To select supported TR069 version (version 1.0 or 1.1). - URL: To configure URL address for ACS.

	• User name: To configure username for ACS. • Password: To configure Password for ACS. • Periodic Inform: To enable periodically inform. • Periodic Interval: To configure interval for periodic inform. • CPE: CPE is short for Customer-premise equipment as client side devices. • URL: To configure URL address for CPE. • User name: To configure username for CPE. • Password: To configure Password for CPE. Note: TR-069(Technical Report 069) is a technical specification entitled CPE WAN Management Protocol (CWMP).It defines an application layer protocol for remote management of end-user devices.
--	--

4.4.4 Network ->SNMP

SNMP	
Active	Disabled ▼
Version	v1/v2c ▼
Trusted IP	
Submit Cancel	

Sections	Description
SNMP	SNMP settings. • Active: To enable or disable SNMP feature. • Version: Select the supported SNMP version. • Trusted IP: To configure allowed SNMP server address, it could be an IP address or any valid URL domain name. Note: SNMP (Simple Network Management Protocols) is Internet-standard protocol for managing devices on IP networks.

4.4.5 Network ->NMS

NMS	
Active	Disabled ▼
Server Address	
Server Port	20006
Interface	Adaptive ▼
Submit Cancel	

Sections	Description
NMS	It configures NMS feature. - Active: Enable or disable NMS function. - Server Address: The address of the NMS server. - Server Port: The port of the NMS server. - Interface: Configure the local interface for accessing the NMS server, which is default in DHCP and static IP mode, PPPoE in PPPoE mode, and default, tap0 or tun0 as appropriate after the VPN is turned on.

4.4.6 Network ->VLAN&QoS

VLAN		
LAN Port	Active	Disabled ▼
	VID	1 (1~4094)
	Priority	0 ▼
PC Port	Active	Disabled ▼
	VID	1 (1~4094)
	Priority	0 ▼
DHCP Vlan	Active	Enabled ▼
	DHCP Option	132 (128~254)
QoS		
	SIP QoS	40 (0~63)
	Voice QoS	40 (0~63)
Submit Cancel		

Sections	Description
VLAN	VLAN settings. • LAN Port/PC Port: You can configure VLAN setting for both ports respectively. • Active: To enable or disable VLAN feature for designated port. • VID: To configure VLAN id for designated port. • Priority: To select VLAN priority for designated port. Note: Please consult your administrator for specific VLAN settings in your networking environment.
QoS	QoS settings. • SIP QoS: To configure QoS value for all SIP message. • Voice QoS: To configure QoS value for all audio stream (RTP streams).

4.4.7 Network ->802.1X

802.1X

802.1X Mode

Disabled

Provisioning Mode

Unauthenticated Pr

Anonymous ID

Identity

Password

.....

Submit
Cancel

CA Certificate

Index	Issue To	Issuer	Expire Time	
1				☐
2				☐
3				☐
4				☐
5				☐

Delete
Cancel

CA Certificate index Select

Index

1

CA Certificate Upload

Index

Auto

Browse not selected
Submit
Cancel

Device Certificate

Index	Issue To	Issuer	Expire Time	
1				☐
2				☐
3				☐
4				☐
5				☐

Delete
Cancel

Device Certificate index Select

Index

1

Device Certificate Upload

Index

Auto

Browse not selected
Submit
Cancel

Submit
Cancel

Sections	Description
802.1X	It configures 802.1X feature. 802.1X Mode: To enable or disable 802.1X. - Anonymous ID: Configure the anonymous ID used to hide the real 802.1X user. - Identity: Configure the username used for 802.1X. - Password: Configure the password used for 802.1X.
CA Certificate	Display or delete the CA certificate, which is used to verify the authenticity of the device certificate.
CA Certificate Upload	Upload the file that will be used as the CA certificate.
Device Certificate	Display or delete the device certificate, which is used to verify the identity of the device with the 802.1X server.
Device Certificate Upload	Upload the file that will be used as the device certificate.

4.4.8 Network ->Diagnosis

Diagnosis

Command
Destination Address

Ping

Start
Stop

Sections	Description
Diagnosis	Used to detect network connectivity

4.5 Phone

4.5.1 Phone->Basic

LCD Language	
LCD Language	简体中文 ▼
Audio Mode	
Audio Device	Speaker ▼
Ring From Headset	Disabled ▼
EHS Type	Disabled ▼
EHS Mic Impedance	15
Display	
Backlight Level	4 ▼
Backlight Time	30 ▼
Notification Popups	
Display Voice Mail Popup	Enabled ▼
Display Txt Message Popup	Enabled ▼
Display Missed Call Popup	Enabled ▼
Display Forward Call Popup	Enabled ▼
Submit Cancel	

Sections	Description
LCD Language	choose the phone language.
Audio Mode	Audio Mode Ring From Headset EHS Type EHS Mic Impedance
Display	Backlight Level: Adjust the backlight intensity of the phone interface. Backlight Time: Adjust the backlight timer, once it expires, the backlight on the phone interface will dim. Screensaver Time: Adjust the standby time of the screen saver on.
Notification Popups	- Notification Popups - Enable/disable display voice mail popup.

Sections	Description
	• Enable/disable display Txt message popup. • Enable/disable display missed call popup. • Enable/disable display forward call popup.

4.5.2 Phone-Time&Date

Format Setting

Time Format12Hour
Date FormatDD-MM-YYYY
Display ModeDay

Time&Date

☐ Manual

Date

Year
Mon
Day

Time

Hour
Min
Sec

☒ NTP

Time Zone+8 China(Beijing)
Primary Server0.pool.ntp.org
Secondary Server1.pool.ntp.org
Update Interval3600(>= 3600s)

Daylight Saving Time

ActiveAuto
Offset60(~300~300Minutes)
☒ By Date

Start Time1 Mon 1 Day 0 Hour
End Time12 Mon 31 Day 23 Hour

☐ By Week

Start MonthJan
Start Week Of MonthFirst In Month
Start Day Of WeekMonday
Start Hour0(0~23)
End MonthDec
End Week Of MonthFourth In Month
End Day Of WeekSunday
End Hour23(0~23)

Daily Reboot

ActiveDisabled
Reboot Time0 : 0

Submit
Cancel

Sections	Description
Format Setting	Time and date format settings. - Time Format: Determine what format to display on Phone UI(12 hour/24 hour). - Date Format: Determine what format to display on Phone UI for Date. - Display Mode: Determine what mode to display Time&Date on Phone UI.
Time&Date	To select how to configure time, it could be set manually or get from INTERNET automatically via NTP server. - Manual: To set Time and Date manually. Note: If you set time to be Manually, it only take effect till next reboot, after reboot, the phone will switch to Auto mode automatically, because there is no way for IP phone to record time during power off. - NTP: To get Time via NTP server. - Time Zone: To select local Time Zone for NTP server. - Primary Server: To configure primary NTP server address. - Secondary Server: To configure secondary NTP server address, it takes effect if primary NTP server is unreachable. - Update Interval: To configure interval between two consecutive NTP requests. Note: NTP, Network Time Protocol is used to automatically synchronized local time with INTERNET time, since NTP server only response GMT time, so that you need to specify the Time Zone for IP phone to decide the local time.
Daylight Saving Time	DST settings. Note: Here DST, is short for Daylight saving time, which stands for the time in the summer days when sun rises early will be adjusted forward to save daylight. The DST will take effects during the period that set by user (all the settings for DST are all self-explanatory, please consult with your administrator for local DST details).
Daily Reboot	Set the phone's daily restart time

4.5.3 Phone- Call Feature

Dial	
Dial Timeout Enable	Disabled ▾
Dial Timeout	4 (1~15s)
Call Waiting	
Active	Enabled ▾
CallWaiting Tone	Enabled ▾
CallWaiting Response Code	180 ▾
On Code	
Off Code	
Call Park	
Active	Disabled ▾
Call Park Code	
Group Park Code	
Auto Redial	
Auto Redial	Disabled ▾
Auto Redial Interval	10 (1~300s)
Auto Redial Times	3 (1~100)
BLF Pickup Alert	
BLF Pickup Alert	Disabled ▾
Remote Control	
Allowed Access IP List	
Call Completion	
Call Completion	Disabled ▾
Broadcast IP	
Broadcast IP	Enabled ▾
Broadcast IP Feature Code	*158#
Others	
Reject Response Code	486 (Busy Here) ▾
Auto Answer Delay	0 (0~5s)
Auto Hangup Time	2 (1~30s)
Early DTMF	Disabled ▾
Multicast Codec	PCMU ▾
User Info Show Type	Name+Number ▾
Display The Active Account	Disabled ▾
Rtp Check Time	Disabled ▾
Submit Cancel	

Sections	Description
----------	-------------

Call Waiting	It enables or disables Call Waiting. • Active: If enabled, it allows IP phones to receive a new incoming call when there is already an active call. • Call Waiting Tone: If enabled, it allows IP phones to play the call waiting tone to the waiting callee.
Auto Redial	Auto redial allows IP phones to redial an unsuccessful call for designated times within designated interval. • Auto Redial: To enable or disable auto redial feature. • Auto Redial Interval: Determine the interval between two consecutive attempts. • Auto Redial Times: Determine how many times to redial.
BLF Pick Up Alert	Visual BLF Pick Up Alert: To enable the BLF pick up function.
Remote Control	Remote Control allows specific host to interact with IP phone by sending HTTP or HTTPS requests. The specific action could be answering an incoming call, hangup an ongoing call and so on. • Allowed Access IP List: To configure the allowed host address. Note: For now, IP phone can only support IP address, IP address list and IP address pattern as allowed hosts.
Call Completion	It enables or disables Call Completion.
Broadcast IP	• Broadcast IP: To enable or disable Broadcast IP. • Broadcast IP Feature Code: To configure the Feature Code for Broadcast IP.
Others	• Reject Response Code: Allows user to assign specific code as return code to SIP server when an incoming call is rejected. • Auto Answer Delay: To configure delay time before an incoming call is automatically answered. • Auto Hangup Time: To configure the time from the busy tone to the standby state after the other party hangs up. • Early DTMF: To enable or disable early DTMF. • Multicast codec: To choose different multicast codec.

4.5.4 Phone-Voice

Echo Canceller	
Echo Canceller	Enabled ▼
VAD	Disabled ▼
CNG	Disabled ▼
Jitter Buffer	
Jitter Type	Adaptive ▼
Min Delay	0 (0~200ms)
Normal Delay	120 (0~200ms)
Max Delay	200 (0~200ms)
Talk Volume	
Handset Volume	6 (1~10)
Headset Volume	6 (1~10)
Handoff Volume	6 (1~10)
Tone Volume	
Handset Volume	5 (1~10)
Headset Volume	5 (1~10)
Handoff Volume	5 (1~10)
Mic Volume	
Handset Volume	5 (1~10)
Headset Volume	5 (1~10)
Handsfree Volume	5 (1~10)
Key Volume	
Volume	5 (0~10)
Ring Volume	
Volume	5 (0~10)
Submit Cancel	

Sections	Description
Echo Canceller	- Echo Canceller: To remove acoustic echo from a voice communication in order to improve the voice quality. - VAD (Voice Activity Detection): Allow IP phone to detect the presence or absence of human speech during a call. When detecting period of “silence” , VAD replaces that silence efficiently with special packets that indicate silence is occurring. It can facilitate speech processing, and deactivate some processes during non-speech section of an

Sections	Description
	audio session. It can avoid unnecessary coding or transmission of silence packets in VoIP applications, saving on computation and network bandwidth. • CNG (Comfort Noise Generation): Allow IP phone to generate comfortable background noise for voice communications during periods of silence in a conversation. It is a part of the silence suppression or VAD handling for VoIP technology. CNG, in conjunction with VAD algorithms, quickly responds when periods of silence occur and inserts artificial noise until voice activity resumes. The insertion of artificial noise gives the illusion of a constant transmission stream, so that background sound is consistent throughout the call and the listener does not think the line has released.
Jitter Buffer	Jitter buffer is a shared data area where voice packets can be collected, stored, and sent to the voice processor in even intervals. Jitter is a term indicating variations in packet arrival time, which can occur because of network congestion, timing drift or route changes. The jitter buffer, located at the receiving end of the voice connection, intentionally delays the arriving packets so that the end user experiences a clear connection with very little sound distortion. IP phones support two types of jitter buffers: fixed and adaptive. • Fixed: Add the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones. • Adaptive: Capable of adapting the changes in the network's delay. The range of the delay time for the dynamic jitter buffer added to packets can be also configured on IP phones. • Min Delay: Set the minimum delay time • Normal Delay: Set normal delay time • Max Delay: Set the maximum delay time
Mic Volume	Adjust microphone volume for headset, handset, and speaker mode.
Key Volume	Volume of keys.
Ring Volume	Volume when calling.

4.5.5 Phone->Key

Soft Key				
Key	Type	Label	Value	Account
Soft Key 1	Speed Dial ▾		6	Auto ▾
Soft Key 2	Speed Dial ▾	S2	1	Account 1 ▾
Soft Key 3	Speed Dial ▾	S3	2	Account 1 ▾
Soft Key 4	Speed Dial ▾	S4	3	Account 1 ▾
Soft Key 5	Speed Dial ▾	S5	4	Account 1 ▾
Soft Key 6	Speed Dial ▾	S6	5	Account 1 ▾

Keypad Lock	
Keypad Lock Type	Disabled ▾
Keypad Unlock PIN	•••••• (1~15)
Keypad Lock Timeout	0 (1~3600s)
Emergency Call	

Key As Send	
Key As Send	# ▾
Send # when First Dial Number is *	Disabled ▾

Others	
Redial Function	Direct Call ▾

Sections	Description
Soft Key	Allows user to assign specific feature to the designated soft keys. For soft key, the following is the available features: DND, Menu, SMS, Status, Book, Fwd, PickUp, Group PickUp, Intercom, Speed Dial, History, Favorites, Redial, Callback, Hot Desking, XML Browser, simple Menu, Voice Message, XML History, XML PhoneBook, ACD.
Keypad Lock	Keypad lock can temporarily lock the phone keypad when not in use. This feature helps prevent unauthorized use of the phone. - Keypad Lock Type: Choose to lock function keys or all keys.. - Keypad Unlock PIN: The password for lock. - Keypad Lock Timeout: The idle interval for locking the phone.
Key As Send	" Key As Send " allow you to disable the send key or assign the # key and * key as the send key.
Others	Redial function optional direct call or enter dialed history.

4.5.6 Phone-Ringtones

Ringtones

Upload(Max Total Size: 100K) Browse not selected

Submit Cancel

Uploaded Ringtones

Delete

Ring Tones Ring1.wav

Distinctive Ringers

Index	Keyword	Ringtone
0		Ring1.wav
1		Ring1.wav
2		Ring1.wav
3		Ring1.wav
4		Ring1.wav
5		Ring1.wav
6		Ring1.wav
7		Ring1.wav
8		Ring1.wav
9		Ring1.wav
10		Ring1.wav
11		Ring1.wav

Sections	Description
Ringtones	Allows user to upload and view ringtone files or delete uploaded ringtone files. Note: Ringtone files are suggested to be (8K 8bit U-law) .wav format files, please contact technical support team for instructions how to make ringtone files. system ringtone files cannot be deleted thus user can only delete uploaded ringtone.
Distinctive Ringers	Distinctive ringers allow different incoming calls to trigger distinctive ringtone. The IP phone will check "Alert-Info" header inside the incoming "invite" SIP message. And strip out the URL or keyword inside the "Alert-Info" header, from the specific URL or keyword to match or

Sections	Description
	download designated ringtone to play.

4.5.7 Phone-Tones

Tones

Select Country or Region

Default

Busy Tone

480+620/500,0/500

RingBack Tone

440+480/2000,0/4000

Dial Tone

350+440

Submit

Cancel

Sections	Description
Tones	Allows users to select tones (classified by country or region) or customize tones. ☐

4.5.8 Phone ->Dial Plan Rule

a) Phone-Dial Plan-Replace Rule

Rules

Replace Rule ▼

Index	Account	Prefix	Replace	
1				☐
2				☐
3				☐
4				☐
5				☐
6				☐
7				☐
8				☐
9				☐
10				☐

Page 1 ▼

Prev

Next

Add

Edit

Delete

Area Code

Code

Min Length

1

(1~15)

Max Length

1

(1~15)

Account

Auto ▼

Submit

Cancel

Sections	Description
Rules	Allows user to select Replace rule or Dial-now to display or edit. For replace rules, you can add, modify, or delete prefix for accounts.
Area Code	Area codes are also known as NPAs (Numbering Plan Areas). They usually indicate different geographical areas within one country. If entered numbers match the predefined area code rule, the IP phone will automatically prefix outgoing number with area code. Note: There is only one area code rule supported.

b) Phone-Dial Plan-Dial Now

Rules

Dial Now ▼

Index	Account	Dial Now Rule	
1			☐
2			☐
3			☐
4			☐
5			☐
6			☐
7			☐
8			☐
9			☐
10			☐

Add

Edit

Delete

Area Code

Code

Min Length

Max Length

Account

1 (1~15)

1 (1~15)

Auto ▼

Submit

Cancel

Sections	Description
Rules	Allows user to select Replace rule or Dial-now to display or edit. For Dial Now, you can add, modify, or delete prefix for accounts.
Dial Now Delay	Allows user configure dial now delay time for dial now. It means user can configure the IP phone to dial out the phone number automatically after the designated delay time if it matches any dial now rule.
Area Code	Area codes are also known as NPAs (Numbering Plan Areas). They usually indicate different geographical areas within one country. If entered numbers match the predefined area code rule, the IP phone will automatically prefix outgoing number with area code. Note: There is only one area code rule supported.

4.5.9 Phone-Action URL

Action URL

Active

Disabled

Setup Completed

Registered

Unregistered

Registered Failed

Off Hook

On Hook

Incoming Call

Outgoing Call

Established

Terminated

Open DND

Close DND

Open Always FWD

Close Always FWD

Open Busy FWD

Close Busy FWD

Open No Answered FWD

Close No Answered FWD

Transfer Call

Blind Transfer

Attended Transfer

Hold

UnHold

Mute

UnMute

Missed Call

IP Changed

FWD Incoming Call

Reject Incoming Call

Answer New Call

Transfer Finished

Transfer Failed

Idle To Busy

Busy To Idle

Submit

Cancel

Sections	Description
Action URL	Action URL settings.

Sections	Description
	• Active: Enable or disable ActionURL • Setup Completed: When the IP phone completes startup. • Registered: When the IP phone successfully registers an account. • Unregistered: When the IP phone logs off the registered account. • Register Failed: When the IP phone fails to register an account. • Off Hook: When the IP phone is off hook. • On Hook: When the IP phone is on hook. • Incoming Call: When the IP phone receives an incoming call. • Outgoing Call: When the IP phone places a call. • Established: When the IP phone establishes a call. • Terminated: When the IP phone terminates a call. • Open DND: When the IP phone enables the DND mode. • Close DND: When the IP phone disables the DND mode. • Open Always FWD: When the IP phone enables the always forward. • Close Always FWD: When the IP phone disables the always forward. • Open Busy FWD: When the IP phone enables the busy forward. • Close Busy FWD: When the IP phone disables the busy forward. • Open No Answer FWD: When the IP phone enables the no answer forward. • Close No Answer FWD: When the IP phone disables the no answer forward • Transfer Call: When the IP phone transfers a call. • Blind Transfer: When the IP phone blind transfers a call. • Attended Transfer: When the IP phone performs the semi-attended/attended transfer.

Sections	Description
	• Hold: When the IP phone places a call on hold. • UnHold: When the IP phone retrieves a hold call. • Mute: When the IP phone mutes a call. • UnMute: When the IP phone un-mutes a call. • Missed Call: When the IP phone misses a call. • IP Changed: When the IP address of the IP phone changes. • FWD Incoming Call: When the IP phone forwards an incoming call. • Reject Incoming Call: When the IP phone rejects an incoming call. • Answer New Call: When the IP phone answers a new call. • Transfer Finished: When the IP phone completes to transfer a call. • Transfer Failed: When the IP phone fails to transfer a call. • Idle To Busy: When the state of the IP phone changes from idle to busy. • Busy To Idle: When the state of phone changes from busy to idle.

4.5.10 Phone ->Multicast

Multicast Setting

Paging Barge

Disabled

Paging Priority Active

Enabled

Priority List

Priority	Listening Address	Label
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		

Submit

Cancel

Sections	Description
Multicast Setting	Multicast setting. - Paging Barge: Choose the multicast number, the range is 1-10. - Paging priority Active: Enable o disable the multicast.
Priority List	It configures the multicast parameters. - Listening Address: Enter the IP address you need to listen. - Label: Input the label for each listening address

4.5.11 Phone->Forward & DND

Sync Feature Code

Active
Disabled

Mode
☒ Phone
☐ Custom

Forward Transfer

Account
All Account

Always Forward
Disabled

Target Number

On Code

Off Code

Busy Forward
Disabled

Target Number

On Code

Off Code

No Answer Forward
Disabled

No Answer Ring Time
30

Target Number

On Code

Off Code

DND

Account
All Account

DND
Disabled

Return Code When DND
486 (Busy Here)

DND On Code

DND Off Code

DND Timer Enable
Disabled

DND Timer 1
0 Hour 0 Min

DND Timer 2
0 Hour 0 Min

DND Timer 3
0 Hour 0 Min

Submit
Cancel

Sections	Description
Sync Feature Code	- Active: enable or disable function synchronization. - Mode: select the desired mode.
Forward Transfer	Forward settings: ☐ Note: There are three types of forwarding: Always Forward,

Sections	Description
	Busy Forward, and No Answer Forward. • Account: Select the account that needs to enable forwarding • Always Forward: calls will be forwarded under any circumstances. • Target Number: Set the always forward target number; • On Code: After setting the On Code, the user can dial the On Code + target number to enable always forward; • Off Code: After setting the Off Code, the user can dial the Off Code + target number to disable the always forward; • Busy Forward: if the IP phone is busy, the call will be forwarded. • Target Number: Set the busy forward target number; • On Code: After setting the On Code, the user can dial the On Code + target number to enable busy forward; • Off Code: After setting the Off Code, the user can dial the Off Code + target number to disable the busy forward; • No Answer Forward: If no one answers after a certain time, the call will be forwarded. • No Answer Ring Time: Set the no answer ring time, if no one answers within the set time, the call will be forwarded; • Target Number: Set the no answer forward target number; • On Code: After setting the On Code, the user can dial the On Code + target number to enable no answer forward; • Off Code: After setting the Off Code, the user can dial the Off Code + target number to disable the no answer forward;
DND	Do not disturb allows the IP phone to ignore any incoming

Sections	Description
	call: - Account: Select the account that needs to enable the DND function - DND: Enable or disable DND - Return Code When DND: Determine what response code should be sent back to the server when Do not disturb is turned on and there is an incoming call. - DND On Code: The code used to turn on DND on the server (if configured), if you press DND when the DND is off, the IP phone will send a SIP message to the server to turn on DND on the server. - DND Off Code: The code used to turn off DND on the server (if configured), if you press DND when it is turned on, the IP phone will send a SIP message to the server to turn off DND on the server. - DND Timer Enable: Enable or disable DND timer. The DND timer can make DND take effect in a specified time period. Note: The DND timer can only be enabled when DND is turned on.

4.5.12 Phone->Intercom

Intercom

Active

Enabled

Intercom Mute

Disabled

Submit

Cancel

Sections	Description
----------	-------------

Intercom	Intercom allows users to directly calls with the other party: • Active: Enable or disable the intercom function. • Intercom Mute: After enabling, once a call is established, the phone will be muted by default.
-----------------	---

4.5.13 Phone->ACD

ACD

ACD Activated Auto

Disabled

ACD Activated Auto Timer

90

(0~180s)

Submit

Cancel

Sections	Description
ACD	ACD (Automatic Call Distribution) is the most common used in the call center. It allows IP phones to log in to the system in multiple states so that the system can distribute incoming calls to available users or agents. • ACD Activated Auto: If activated, the phone will automatically become available within a specified time (determined by the ACD automatic activation time). • ACD Activated Auto Timer: Configure the time interval for the phone to automatically become available.

4.5.14 Phone->Power LED

Power LED Status	
Ringling	Fast Flash ▼
Call Hold	Fast Flash ▼
Mute	OFF ▼
Calling or Dialing	OFF ▼
Miss Call	Slow Flash ▼
Miss Message	Slow Flash ▼
Default	OFF ▼
Submit Cancel	

Sections	Description
Power LED Status	• Ringing: optional off/always bright/slow flash/fast flash state. • Call Hold: optional off/always bright/slow flash/fast flash state. • Mute: Optional off/always bright/slow flash/fast flash state. • Calling or Dialing: Optional off/always bright/slow flash/fast flash state. • Miss call: Optional off/always bright/slow flash/fast flash state. • Miss Message: Optional off/always bright/slow flash/fast flash state. • Default: Optional off/always bright/slow flash/fast flash state.

4.6 PhoneBook

4.6.1 PhoneBook -> Local Book

Contact All Contacts ▼

Search Search Reset

Dial Auto ▼ Dial Hand Up

Sort Name ▼

Index	Name	Office Number	Mobile Number	Other Number	Group	Ring	Account	
1								☐
2								☐
3								☐
4								☐
5								☐
6								☐
7								☐
8								☐
9								☐
10								☐

Page 1 ▼ Prev Next Move To All Contacts ▼ Delete Delete All

Contact Setting

Name

Mobile Number

Group Default ▼

Account Auto ▼

Office Number

Other Number

Ring Auto ▼

Add Edit Cancel

Group

Index	Name	Ring	Description	
1				☐
2				☐
3				☐
4				☐
5				☐
6				☐
7				☐
8				☐
9				☐
10				☐

Page 1 ▼ Prev Next Delete Delete All

Group Setting

Name

Ring Auto ▼

Description

Add Edit Cancel

Import/Export

Export Format .XML ▼

Contact Browse not selected (.XML/.CSV/.VCF)

Import Export Cancel

Black List Browse not selected (.XML/.CSV/.VCF)

Import Export Cancel

White List Browse not selected (.XML/.CSV/.VCF)

Import Export Cancel

Sections	Description
Contact	To select the type for local book. - All Contacts: To display or edit all local contacts. - Favorites: To display or edit favorites contacts. - Black List: To display black list contacts. - White List: To display white list contacts.
Search	To search designated contacts from local phonebook.
Dial	Dial out a call or hangup an ongoing call from Web UI. Note: For this feature, you need to have the remote control privilege to control IP phone via Web UI. Please refer to section “Remote Control” in the Web UI-> Phone -> Call Feature page.
Sort	To configure the sorting method for the local book. The Index and name are optional.
Group Setting	Management group contacts.
Import/Export	Import or export the contact or blacklist file.

4.6.2 PhoneBook -> Remote Book

Remote Book

Index	Remote URL	Display Name
1		
2		
3		
4		
5		

Search Remote Phonebook Name

Enabled

Refresh Interval

3600

(120~2592000s)

Submit

Cancel

Sections	Description
----------	-------------

Remote Book	Remote Book settings. • Index: To select desired Remote Book item to display and configure. • Remote URL: To configure remote book server address. • Display Name: To configure display remote book name on Phone UI. • Search Remote Phonebook Name: To enable or disable search remote phonebook name. • Reflash Interval: To set interval (Range from 120s to 2592000s). Note: IP phone supports at most 5 remote books. Please refer to your administrator for how to establish a remote book server and how to create remote book xml file.
--------------------	---

4.6.3 PhoneBook -> Call History

Sections	Description
Direct Call By History	Enable/disable the direct call by history.
History Max Num	To configure the maximum number of call history displayed.
History Clear Time	To configure the clear time of the call history.
Call History	Available call history types are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls.

Sections	Description
	Hang Up: To click to hangup ongoing call on the IP phone. Note: For “HangUp” feature, you need to have the remote control privilege to control IP phone via Web UI. Please refer to section “Remote Control” in the Web UI -> Phone -> Call Feature page.

4.6.4 PhoneBook -> LDAP

LDAP

Name Filter

Number Filter

Encryption Mode

NULL

Server

Port

(1~65535)

Base

Username

Password

Protocol

Version3

Name Attribute

Number Attribute

Display Name

Ou Attribute

Hierarchy Display Name

Max Hits

(1~2000)

Search Delay Time

1000

(200~3000)ms

Lookup for Incoming Call

Disabled

Lookup for Dial

Disabled

Hierarchy View

Enabled

Submit

Cancel

Sections	Description
LDAP	LDAP phonebook settings. - Name Filter: The settings used to tell LDAP server what name attributes to search. - Number Filter: The settings used to tell LDAP server what number attributes to search. - Encryption Mode:

Sections	Description
	NULL: It is not encrypted. StartTLS: It is not encrypted at first, and can be switched to TLS encryption mode after negotiation. TLS: Transport Layer Security (TLS) is an encryption protocol that provides communication security over computer networks. • Server: To configure LDAP server's address. • Port: To configure LDAP server's port. • Base DN: To configure searching base DN on LDAP server. • User Name: To configure user name for accessing LDAP server. • Password: To configure password for accessing LDAP server. • Name Attribute: To configure which name attributes should be feedback from LDAP server. • Number Attribute: To configure which number attributes should be feedback from LDAP server. • Display Name: To configure display name on Phone UI when there is any searching result from LDAP server. • Max Hits: To configure the maximum size of result response from LDAP server. • Search Delay Time: To configure delay time before initiate LDAP searching request after you input a value from Phone UI. • When this function is enabled, the system will automatically match and display the incoming call information • When this function is enabled, the system will automatically match the outgoing number with the LDAP phonebook and display the contact information of the number • Set LDAP phone book level display When this function is enabled, the system will automatically match and display the incoming call information When this function is enabled, the system will automatically

Sections	Description
	match the outgoing number with the LDAP phonebook and display the contact information of the number Set LDAP phone book level display • Lookup for Incoming Call: If enabled, the incoming call will be matched with the LDAP phone book and displayed. • Lookup for Dial: If enabled, it will be matched with the LDAP phonebook and displayed when calling out or replaying. • Hierarchy View: If enabled, the LDAP phone book will display hierarchical directories, such as multiple departments of a company. Note: For setting details, please consult with your system administrator for further information.

4.6.5 PhoneBook -> Broadsoft

Broadsoft PhoneBook

PhoneBook Item

Item1

Display Name

Group

Server Address

Server Port

(1~65535)

Username

Password

••••••••

Submit

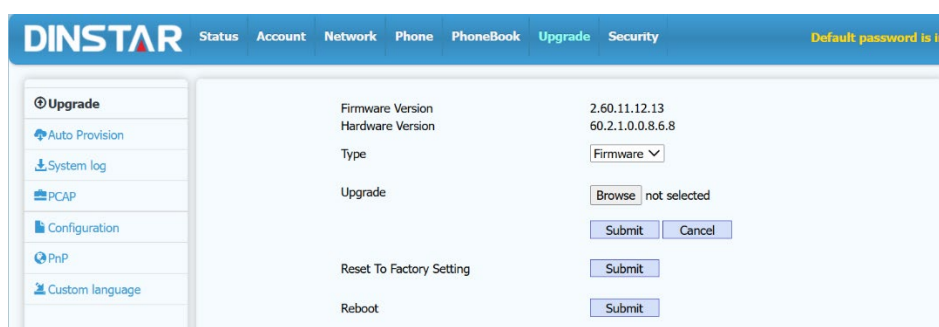
Cancel

Sections	Description
Broadsoft PhoneBook	Broadsoft PhoneBook settings. • PhoneBook Item: To select specific item to configure. • Display Name: The name displayed at IP phone's LCD screen when accessed via Phone UI. • Server Address: Broadsoft PhoneBook server's address.

Sections	Description
	- Server Port: Broadsoft PhoneBook server's port. - User Name: Username used to access Broadsoft PhoneBook server. - Password: Password used to access Broadsoft PhoneBook server. Note: IP phone supports at most 5 Broadsoft PhoneBook items. For Broadsoft PhoneBook's server address, port, username and password, you need to consult your Broadsoft service provider for further information.

4.7 Upgrade

4.7.1 Upgrade->Upgrade



Sections	Description
Firmware Version	Show you the firmware version
Hardware Version	Show you the hardware version.
Upgrade	To select upgrading rom file from local or a remote server automatically. Note: Please make sure it's right file format for right model.
Reset to Factory Setting	Allow you to reset IP phone's setting to factory settings.
Reboot	To reboot IP phone remotely from Web UI.

4.7.2 Upgrade-Auto Provision

DHCP Option	
DHCP Option 66	Enabled
DHCP Option 43	Enabled
Custom Option	(128~254)
Provisioning Server	
URL	
Username	
Password	
Common AES Key	
AES Key(MAC)	
Auto Provision	
Mode	Power On
Schedule	Sunday
Time	22 Hour 0 Min
Periodic Interval	60 Min(5~1440)
Idle Check Time	0 Min(0~60)
Clear MD5	Submit
Export Configuration Template	Export
Auto Provision Now	
Auto Provision Now	
Submit Cancel	

Sections	Description
DHCP Option	It displays and configures DHCP option. - DHCP Option 66: To enable or disable DHCP Option 66. - DHCP Option 43: To enable or disable DHCP Option 43. - Custom Option: If configured, IP Phone will use designated DHCP option to get Auto Provisioning server's address via DHCP. This setting requires DHCP server to support corresponding option.
Provisioning Server	Update server settings. - URL: Auto provisioning server address. - User name: Configure if server needs an username to access, otherwise left blank. - Password: Configure if server needs a password to access, otherwise left blank. - Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file. - AES Key (MAC): Used for IP phone to decipher MAC-

Sections	Description
	oriented auto provisioning configuration file (for example, file name could be 0c1105888888.conf if IP phone's MAC address is 0c1105888888). Note: AES is one of methods for encryption, it should be configured only configure filed is ciphered with AES, otherwise left blank.
Auto Provision	Auto Provisioning mode settings. This Auto Provisioning mode is actually self-explanatory. For example, mode “Power on” means IP phone will go to do Provisioning every time it powers on. - Schedule: Set automatic update time. - Clear MD5: Used to clear MD5 of configuration fil - Export Configuration Template: Export Auto Provisioning configuration template file.
Auto Provision Now	Auto upgrade now

4.7.3 Upgrade->System Log

System Log

Log Level

3

Export Log

Export

Remote Syslog

Disabled

Remote Syslog Server

Screenshot

Phone Screen

Save BMP

Submit

Cancel

Sections	Description
System Log	To display system log level and export log. Log Level: From level 0~7.The higher level means the more

Sections	Description
	specific syslog is saved to a temporary file. By default, it's level 3. - Export Log: Click to export temporary syslog file to local PC. - Remote System Log: To enable or disable Remote System Log. - Remote System Log Server: To input the syslog server address.
Screenshot	Phone Screen: To capture and save the phone screen

4.7.4 Upgrade->Network Capture

Network Capture

Interface

LAN ▼

Network Capture

Start

Stop

Mirror

Ethernet Port Mirror

Disabled ▼

Sections	Description
Network Capture	Network capture settings and export data packet: - Interface: Select the interface for network capture (LAN/VPN/PPPOE). - Start: To start capturing all the packets file sent or received from IP phone. - Stop: To stop capturing packets.
Mirror	Network port mirror settings: Ethernet Port Mirror: If enabled, the PC port can mirror the packets of the Ethernet port.

4.7.5 Upgrade->Configuration

Configuration	
Import Configuration File(.tgz/.conf/.cfg)	Browse not selected Import Cancel
Export Configuration File	Export

Sections	Description
Configuration	Export or import the configuration file of the IP phone

4.7.6 Upgrade->PnP

PnP	
Active	Enabled ▼
Server Address	www.commsns.com
Server Port	5060
Update Interval	1 Hour(1-99)
Submit Cancel	

Sections	Description
PnP	PnP setting for Auto Provisioning. - Active: Once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get Auto Provisioning server's address. By default, this SIP message is sent to multicast address 224.0.1.75. - Server Address: The address of the PNP server. - Server Port: The port of the PNP server. - Update Interval: Check whether the PNP server has an update interval.

4.7.7 Upgrade->Custom Language

Resource Files	
Web (.js)	Browse not selected Upload Cancel
Web Help (.js)	Browse not selected Upload Cancel
LCD (.txt)	Browse not selected Upload Cancel

Sections	Description
Resource Files	- Web: Upload and set Web language other than help information. - Web Help: Upload and set Web help information language. - LCD: Upload and set the system language of the phone.

4.8 Security

4.8.1 Security->Basic

Web Password Modify	
Username	admin
Current Password	
New Password	
Confirm Password	
Session Timeout	
Session Timeout	300 (60~14400s)
Submit Cancel	

Sections	Description
Web Password Modify	To modify Web password. • User Name: Select the user name (administrator or user) that needs to be modified • Current Password: The current password you used. • New Password: Input new password you intend to use. • Confirm Password: Repeat the new password.
Session Timeout	It configures session timeout. • Session Timeout: The value specifies the timeout for login retry in seconds.

4.8.2 Security->Advanced

Web Server Certificate

Index	Issue To	Issuer	Expire Time	Delete
1	Dinstar IP Phone WEB	Dinstar Co.,Ltd.	Sun Aug 17 06:34:00 2036	Delete

Web Server Certificate Upload

[Browse](#) not selected
 [Submit](#)
[Cancel](#)

Trusted Certificate

Index	Issue To	Issuer	Expire Time	
1				☐
2				☐
3				☐
4				☐
5				☐
6				☐
7				☐
8				☐
9				☐
10				☐

[Delete](#)
[Cancel](#)

Trusted Certificate Upload

Index

Auto ▾

[Browse](#) not selected
 [Submit](#)
[Cancel](#)

Only Accept Trusted Certificates

Disabled ▾

[Submit](#)
[Cancel](#)

Sections	Description
Web Server Certificate	To display or delete Certificate which is used when IP phone is connected from any incoming HTTPs request. Note: The default certificate could not be deleted.
Web Server Certificate Upload	Upload certificate file which will be used as server certificate.

Client Certificate	To display or delete Certificates which is used when IP phone is connecting to any HTTPs server.
Client Certificate Upload	Upload certificate files which will be used as client certificate. Only Accept trusted Certificates: If this option is enabled, only trusted certificates will be accepted.

4.8.3 Security->Firewall

Firewall

Active
Disabled

Submit
Cancel

Firewall Rule

Index	Rule Name	Input/Output	Action	Protocol	Source IP	Source MAC	Source Port	Destination IP	Destination Port	
1	rule0	Input	REJECT						20	☐
2										☐
3										☐
4										☐
5										☐
6										☐
7										☐
8										☐
9										☐
10										☐

Page 1
Prev
Next
Delete
Delete All

Firewall Rule Set

Rule Name

Input/Output
Input

Source IP

Destination IP

Source MAC

Protocol
tcp&udp

Source Port

Destination Port

Action
ACCEPT

Add
Edit
Cancel

Sections	Description
Firewall	Active: Enable or disable firewall
Firewall Rule	- Input/Output: Select the direction of the input and output of the network packet - Source IP: Set source IP

	• Destination IP: Set the destination IP • Source MAC: Set network source MAC • Protocol: Select the transmission protocol of the network packet • Source Port: Set the network source port • Destination Port: Set the network destination port • Action: To accept or reject network packets to pass
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5 Troubleshooting

Issue 1: No tone heard from the handset

- Check the connection cord between the handset and the phone. Make sure it is connected properly.

Issue 2: Cannot access the web system

- Check the connection between the PC port of the device and the network port of the computer. Make sure it is fine.
- Check whether the IP address of the device is correct.
- If it is LAN, please make sure there is no IP address conflict with other devices on the network.

Issue 3: Cannot call out

- Please see the network connection status of device, if it is exception, and then check the connection of network.
- If the network connection is normal, please check whether the device has registered successfully.
- If the network connection and the registered are both normal, please confirm whether the dial rule is correct, or please communicate with the service operator.

6 Appendix

6.1 Abbreviations

Abbreviation	Explanation
ACD	Automatic Call Distribution
BLF	Busy Lamp Field
CID	Caller Identity
DND	Do NOT Disturb
DTMF	Dual Tone Multi Frequency
DNS	Domain Name System
DHCP	Dynamic Host Configuration Protocol
NAT	Network Address Translation
LDAP	Lightweight Directory Access Protocol
LLDP	Link Layer Discovery Protocol
SNMP	Simple Network Management Protocol
TR069	Technical Report - 069
NMS	Network management system
PNP	Plug-and-Play
PPPoE	point-to-point protocol over Ethernet
NTP	Network Time Protocol
RTP	Real Time Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
802.1X	IEEE 802.1X
VPN	Virtual Private Network
WLAN	Wireless Local Area Network

6.2 Time Zones

Time Zone	Time Zone Name
-11	Samoa
-10	United States-Hawaii-Aleutian
-10	United States-Alaska-Aleutian
-9	United States-Alaska Time
-8	Canada(Vancouver,Whitehorse)
-8	Mexico(Tijuana,Mexicali)
-8	United States-Pacific Time
-7	Canada(Edmonton,Calgary)
-7	Mexico(Mazatlan,Chihuahua)
-7	United States-Mountain Time
-7	United States-MST no DST
-6	Canada-Manitoba(Winnipeg)
-6	Chile(Easter Islands)
-6	Mexico(Mexico City,Acapulco)
-6	United States-Central Time
-5	Bahamas(Nassau)
-5	Canada(Montreal,Ottawa,Quebec)
-5	Cuba(Havana)
-5	United States-Eastern Time
-4:30	Venezuela(Caracas)
-4	Canada(Halifax,Saint John)
-4	Chile(Santiago)
-4	Paraguay(Asuncion)
-4	United Kingdom-Bermuda(Bermuda)
-4	United Kingdom(Falkland Islands)
-4	Trinidad&Tobago
-4	Curaçao

Time Zone	Time Zone Name
-3:30	Canada-New Foundland(St.Johns)
-3	Denmark-Greenland(Nuuk)
-3	Argentina(Buenos Aires)
-3	Brazil(no DST)
-3	Brazil(DST)
-2	Brazil(no DST)
-1	Portugal(Azores)
0	GMT
0	Greenland
0	Denmark-Faroe Islands(Torshaven)
0	Ireland(Dublin)0 Portugal(Lisboa,Porto,Funchal)
0	Spain-Canary Islands(Las Palmas)
0	United Kingdom(London)
0	Morocco
+1	Poland(Warsaw)
+1	Albania(Tirane)
+1	Austria(Vienna)
+1	Belgium(Brussels)
+1	Caicos
+1	Chatham
+1	Croatia(Zagreb)
+1	Czech Republic(Prague)
+1	Denmark(Copenhagen)
+1	France(Paris)
+1	Germany(Berlin)
+1	Hungary(Budapest)
+1	Italy(Rome)
+1	Luxembourg(Luxembourg)
+1	Makedonia(Skopje)
+1	Netherlands(Amsterdam)

Time Zone	Time Zone Name
+1	Namibia(Windhoek)
+2	Estonia(Tallinn)
+2	Finland(Helsinki)
+2	Gaza Strip(Gaza)
+2	Greece(Athens)
+2	Israel(Tel Aviv)
+2	Jordan(Amman)
+2	Latvia(Riga)
+2	Lebanon(Beirut)
+2	Moldova(Kishinev)
+2	Russia(Kaliningrad)
+2	Romania(Bucharest)
+2	Syria(Damascus)
+2	Turkey(Ankara)
+2	Ukraine(Kyiv,Odessa)
+3	East Africa Time
+3	Iraq(Baghdad)
+3	Russia(Moscow)
+3	30 Iran(Teheran)
+4	Armenia(Yerevan)
+4	Azerbaijan(Baku)
+4	Georgia(Tbilisi)
+4	Kazakstan(Aqtau)
+4	Russia(Samara)
+5	Kazakstan(Aqtobe)
+5	Kyrgyzstan(Bishkek)
+5	Pakistan(Islamabad)
+5	Russia(Chelyabinsk)
+5	Russia(Yekaterinburg)
+5:30	India(Calcutta)

Time Zone	Time Zone Name
+6	Kazakstan(Astana,Almaty)
+6	Russia(Novosibirsk,Omsk)
+7	Russia(Krasnoyarsk)
+7	Thailand(Bangkok)
+8	China(Beijing)
+8	Singapore(Singapore)
+8	Australia(Perth)
+9	Korea(Seoul)
+9	Japan(Tokyo)
+9:30	Australia(Adelaide)
+9:30	Australia(Darwin)
+10	Australia(Sydney,Melbourne,Canberra)
+10	Australia(Brisbane)
+10	Australia(Hobart)
+10	Russia(Vladivostok)
+10:30	Australia(Lord Howe Islands)
+11	New Caledonia(Noumea)
+12	New Zealand(Wellington,Auckland)
+12:45	New Zealand(Chatham Islands)
+13	Tonga(Nukualofa)
-7	United States-Mountain Time

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