OpenVox Communication Co Ltd

C220
User Manual

OpenVox Communication Co Ltd

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Directory

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3 Safety Instruction

Please read the following safety notices before installing or using this unit. They are crucial for the safe and reliable operation of the device.

- Please use the external power supply that is included in the package. Other power supply may cause damage to the phone and affect the behavior or induce noise.
- Before using the external power supply in the package, please check the home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it because it may cause fire or electric shock.
- Do not drop, knock or shake the phone. Rough handling can break internal circuit boards.
- This phone is design for indoor use. Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure the phone to high temperature or below 0℃ or high humidity.
- Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

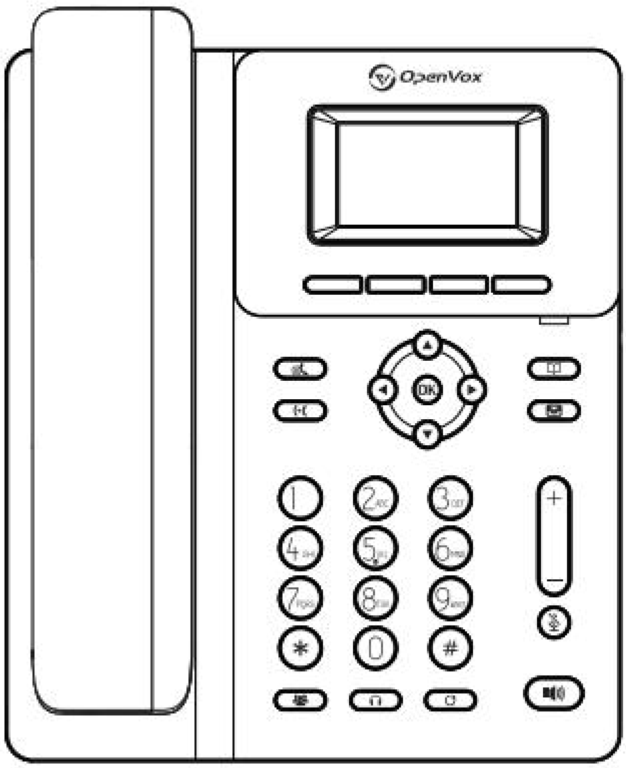
4 Overview

4.1 Overview

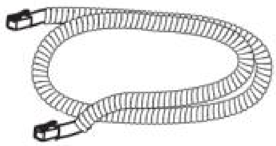
C220 is not only a basic level IP phone, but a delicate work of art, providing a smart and smooth communication experience for enterprises. As the basic level IP phone featuring 132x64 screen and necessary VoIP features and other extended features. It is a perfect combination of elegant outside and powerful inside.

In order to help some interested users to better understand the details of the product, this user manual can be used as a reference guide for the use of the device. This document may not apply to the latest version of the software. If you have any questions, you can use the help prompt interface that comes with the phone,

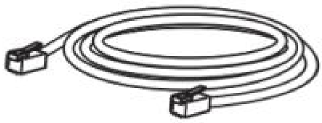
4.2 Packing Contents



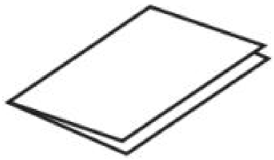
Phone



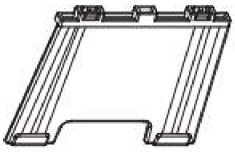
Handset　cable



Ethernet Cable

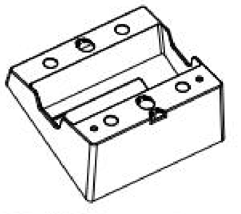


Quick Installation Guide



Stand

Power adapter (Optional)



Wall Stand (Buy separately)

5 Desktop Installation

5.1 PoE And the use of external power adapters

C220 called as ‘the device’ hereafter,supports two power supply modes, power supply from external power adapter or over Ethernet (PoE) complied switch.

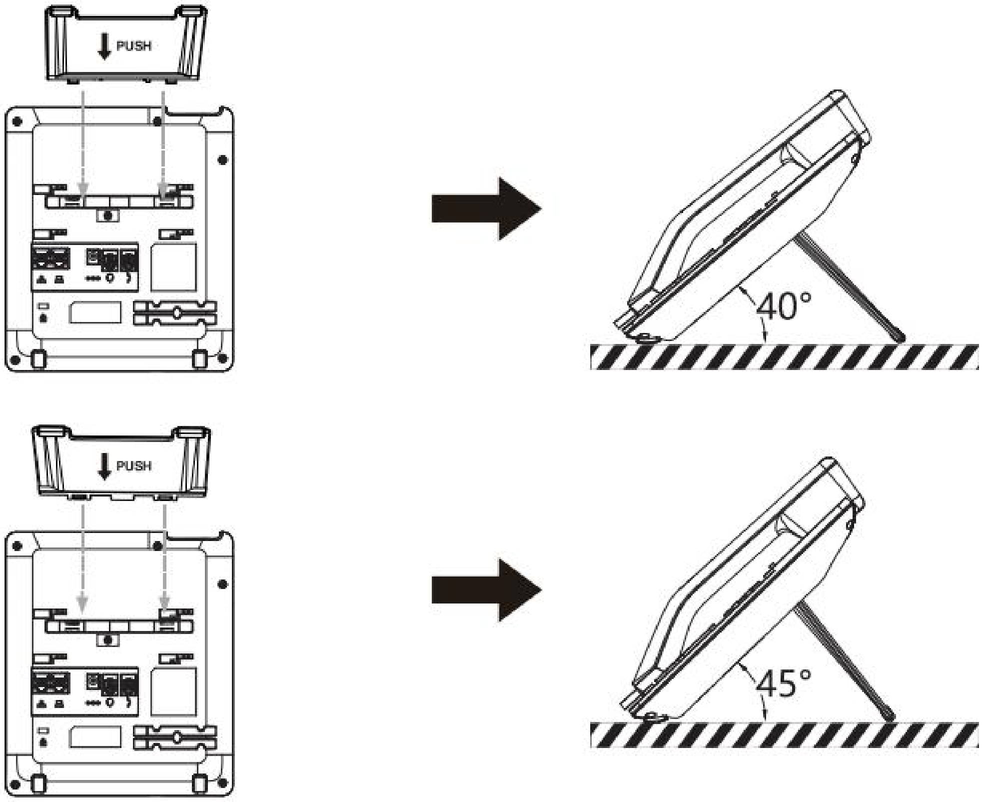
PoE power supply saves the space and cost of providing the device additional power outlet. With a PoE switch, the device can be powered through a single Ethernet cable which is also used for data transmission. By attaching UPS system to PoE switch, the device can keep working at power outage just like traditional PSTN telephone which is powered by the telephone line.

For users who do not have PoE equipment, the traditional power adaptor should be used. If the device is connected to a PoE switch and power adapter at the same time, the power adapter will be used in priority and will switch to PoE power supply once it fails.

Please use the power adapter supplied by and the PoE switch met the specifications to ensure the device work properly.

5.2 Desktop and wall mounted method

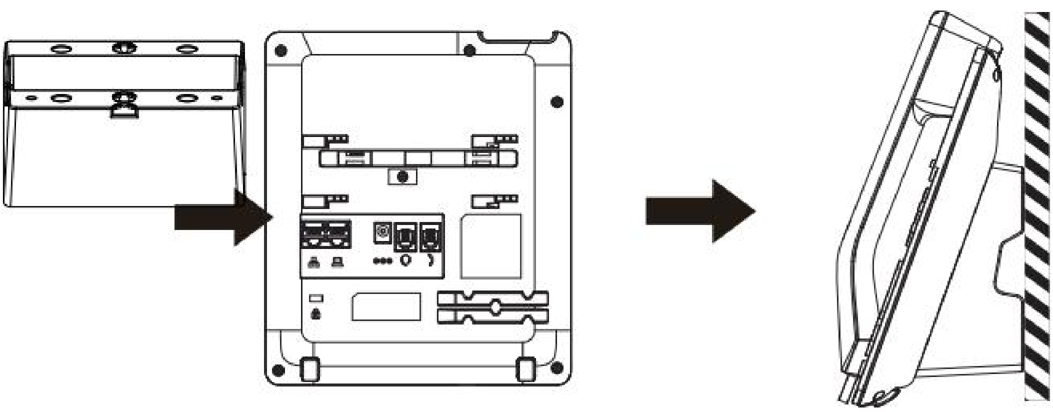
The device supports two installation modes, desktop and wall mounted.If the phone is on the desktop, please follow the instructions in the picture below to install the phone.



Picture 1 - Desktop installation

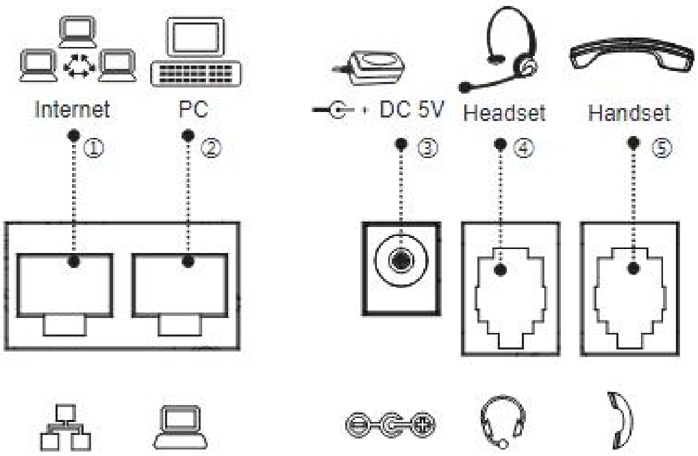
if the phone is mounted on the wall, please follow the instructions below to install it.

Note: wall hanging bracket needs to be purchased separately



Picture 2 - Wall-mounted installation

Connect the power adapter, network, PC, phone and earphone to the appropriate port as shown in the picture below.



Picture 3 - Connecting to the Device

Table 1 - Hardware Interface Description

Index	Interface	Description
①	Power Interface	Connecting Power Adapter
②	Network Interface	Connecting to LAN or Internet
③	PC Interface	Network Interface for Connecting Computer
④	Headset Interface	Connecting Headset
⑤	Receiver Interface	Connecting Microphone Receiver

6 Appendix Table

6.1 Appendix I - Icon

Table 2 - Keypad Icons

Icon	Description
	Redial key, access to redial the last record
	PhoneBook
	Hand-free key, activate/ deactivate hands free
	In idle mode: activate/deactivate silent mode In communication mode: mute/un-mute a call
	Volume down
	Volume up
	Hold
	Headset key, activate/deactivate headset
	MWI
	Conference
	Transfer

Table 3 - Status Prompt and Notification Icons

	Call out
	Call in
	Call Hold
	Network Disconnected
	Open VLAN
	Open VPN
	Keypad Locked
	Call forward activated
	Out calls
	In calls
	Missed calls
	SMS

	New voice message waiting
	Do-Not-Disturb inactivated on Phone
	Call forward activated
	Auto-answering activated
	Hands-free (HF) Mode
	Headphone (HP) Mode
	Handset (HS) Mode
	Mute Microphone
	The Voice encryption of calling
HD	HD Voice
	Voice recording
	Open SIP Hotline

6.2 Appendix II - Keyboard character query table

Table 4 - Look-up Table of Characters

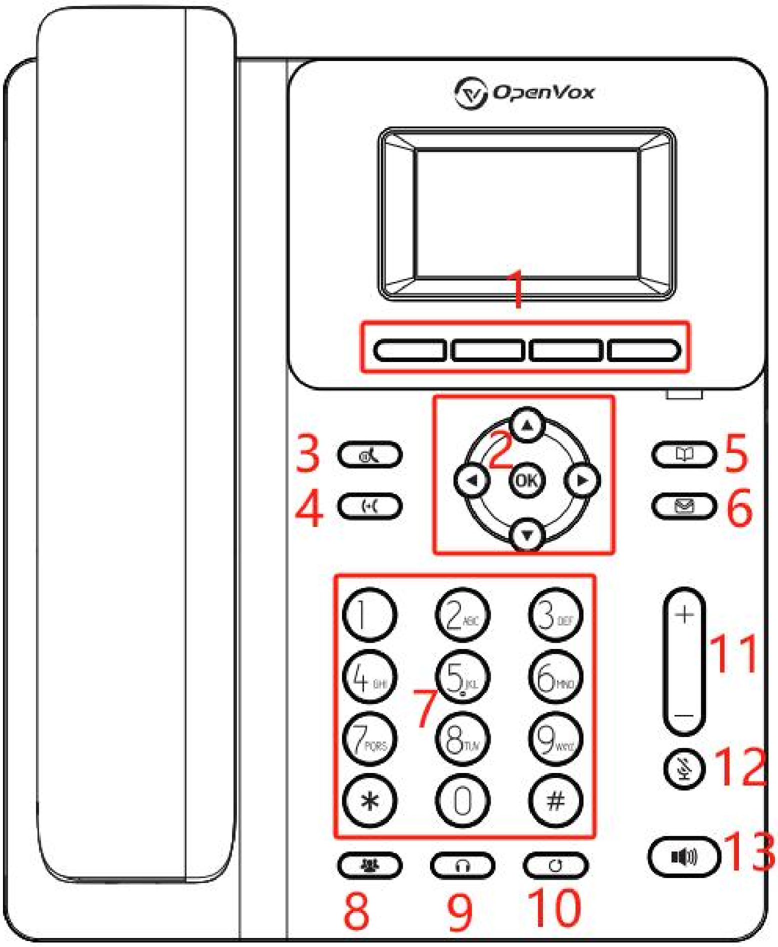
<table>
<tr><th>Mode Icon</th><th>Text Mode</th><th>Key Button</th><th>Characters Of Each Press</th></tr>
<tr><td rowspan="12">123</td><td rowspan="12">Numeric</td><td>1</td><td>1</td></tr>
<tr><td>2</td><td>2</td></tr>
<tr><td>3</td><td>3</td></tr>
<tr><td>4</td><td>4</td></tr>
<tr><td>5</td><td>5</td></tr>
<tr><td>6</td><td>6</td></tr>
<tr><td>7</td><td>7</td></tr>
<tr><td>8</td><td>8</td></tr>
<tr><td>9</td><td>9</td></tr>
<tr><td>0</td><td>0</td></tr>
<tr><td>*</td><td>*.:/@[],+-_=’?\”|;()<>{}</td></tr>
<tr><td>#</td><td>#</td></tr>
<tr><td rowspan="8">abc</td><td rowspan="8">Lower Case Alphabets</td><td>1</td><td>@:;()<>[]{}</td></tr>
<tr><td>2</td><td>a b c</td></tr>
<tr><td>3</td><td>d e f</td></tr>
<tr><td>4</td><td>g h i</td></tr>
<tr><td>5</td><td>j k l</td></tr>
<tr><td>6</td><td>m n o</td></tr>
<tr><td>7</td><td>p q r s</td></tr>
<tr><td>8</td><td>t u v</td></tr>
</table>

<table>
<tr><td rowspan="4"></td><td rowspan="4"></td><td>9</td><td>w x y z</td></tr>
<tr><td>0</td><td>(space)</td></tr>
<tr><td>*</td><td>.,*/+-:_= ’?\”|</td></tr>
<tr><td>#</td><td># ^!&$% £ ￥¤~¡ ¿ §</td></tr>
<tr><td rowspan="12">ABC</td><td rowspan="12">Upper Case Alphabets</td><td>1</td><td>@:;()<>[]{}</td></tr>
<tr><td>2</td><td>A B C</td></tr>
<tr><td>3</td><td>D E F</td></tr>
<tr><td>4</td><td>G H I</td></tr>
<tr><td>5</td><td>J K L</td></tr>
<tr><td>6</td><td>M N O</td></tr>
<tr><td>7</td><td>P Q R S</td></tr>
<tr><td>8</td><td>T U V</td></tr>
<tr><td>9</td><td>WZYX</td></tr>
<tr><td>0</td><td>(space)</td></tr>
<tr><td>*</td><td>.,*/+-:_= ’?\”|</td></tr>
<tr><td>#</td><td># ^!&$% £ ￥¤~¡ ¿ §</td></tr>
<tr><td rowspan="12">2aB</td><td rowspan="12">Mixed type input</td><td>1</td><td>1</td></tr>
<tr><td>2</td><td>2 a b c A B C</td></tr>
<tr><td>3</td><td>3 d e f D E F</td></tr>
<tr><td>4</td><td>4 g h I G H I</td></tr>
<tr><td>5</td><td>5 j k l J K L</td></tr>
<tr><td>6</td><td>6 m n o M N O</td></tr>
<tr><td>7</td><td>7 p q r s P Q R S</td></tr>
<tr><td>8</td><td>8 t u v T U V</td></tr>
<tr><td>9</td><td>9 w z y x W Z Y X</td></tr>
<tr><td>0</td><td>0</td></tr>
<tr><td>*</td><td>.*:/@[],+-_=’?\”|;()<>{}</td></tr>
<tr><td>#</td><td># ^!&$% £ ￥¤~¡ ¿ §</td></tr>
<tr><td rowspan="6">Abc</td><td rowspan="6">Initial capital letter</td><td>1</td><td>@:;()<>[]{}</td></tr>
<tr><td>2</td><td>A B C a b c</td></tr>
<tr><td>3</td><td>D E F d e f</td></tr>
<tr><td>4</td><td>G H I g h i</td></tr>
<tr><td>5</td><td>J K L j k l</td></tr>
<tr><td>6</td><td>M N O m n o</td></tr>
</table>

		7	P Q R S p q r s
		8	T U V t u v
		9	W X Y Z w x y z
		0	(space)
		*	.,*/+-:_= '?\"|
		#	# ^!&$% £ ¥ ¤~¡ ¿ §

7 Introduction to the User

7.1 Instruction of Keypad



Picture 4 - Instruction of Keypad

The picture above shows the keypad layout of the phone.Each button provides its own specific function.Users can refer to the instructions for the keys in the illustration in this section to operate the phone.

Table 5 - Instruction of Keypad

Number	The keypad names	Instruction

①	Soft-menu Buttons	These four buttons provide different functions corresponding to the soft-menu displayed on the screen.
②	Navigate/OK Keys	The user can press the up/down navigation key to change the line or move the cursor in the screen list. On some Settings and text editing pages, the user can press the left/right navigation key to change options or move the cursor in the screen list to the left/right. OK key: Default is equivalent to soft button confirmation; user can customize the function.
③	Hold Key	Press the "Hold" key during the call, the user can hold the call, and press it again to cancel the holding and restore the normal call state.
④	Transfer Key	Press the "Transfer" button, the user can transfer the current call to other numbers.
⑤	Phonebook key	Press the "Phonebook" button, and the user enters the interface of contact
⑥	Voicemail key	Press the "Voicemail" key, the user can enter voicemail interface or listen to the voicemail
⑦	Standard Telephone Keys	The 12 standard telephone keys provide the same function as standard telephones.At the same time, a long press on some keys can trigger a special function.Button #- Press and hold the button to lock the device (enable the keypad lock configuration).
⑧	Conference	Press the "Conference" button, the user can initiate a three-party conference.
⑨	Headset Key	Press the "Headset" button and the user can open the headset channel
⑩	Redial	Press the Redial key to redial the last number dialed
⑪	Volume key	In standby mode, on the ringing and ringing configuration interface, press this button to decrease/increaseRingtone volume; In the call or tone adjustment interface, press this button to decrease/increase the volume
⑫	Mute Key	During a call, the user can press this key to mute the microphone.
⑬	Hands free key	Users can press this button to activate the hands-free speaker audio channel.

7.2 Using Handset / Hands-free Speaker / Headphone

- **Using Handset**

To talk over handset, user can pick up the controller to dial a number, or dial the number first and then pick up the controller, and the number will be dialed out. When the speaker or headphone channel is turned on, users can switch the phone audio channel by picking up or dropping the handle.

- **Using Hands-free Speaker**

To talk over hands-free speaker, user should press the hands-free button then dial the number, or dial the number first then press the hands-free button. User can switch audio channel to the speaker from handset by pressing the hands-free button when audio channel is opened in handset.

- **Using Headphone**

To use headphone, by default, user should headset button which is defined by DSS key to turn on the headphone. Same as handset and hands-free speaker, user can dial the number before or after headphone turned on.

7.3 Idle Screen

Picture 5 - Screen layout/default home screen

The image above shows the default standby screen, which is the user interface most of the time.

The upper half of the home screen shows the status of the device, information and data that can be edited (such as voice messages, missed calls, auto answer, do not disturb, lock status, network connection status, etc.).

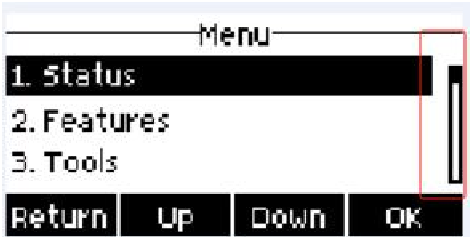
The lower half of the area is the function menu key, which is also the first layer of function menu keys, through which users can operate the phone.

Users can restore the phone to the default standby screen interface by picking up and dropping the handle.

The left and right part of the area shows default configuration of Side key, which dynamically displays the configuration of SIP information, message, headset, etc., which can be customized by users.

The icon description is described in 6.1 appendix I.

In some screens, there are many items or long text to be displayed which could not fit into the screen. They will be arranged in a list or multiple lines with a scroll bar. If user sees a scroll bar, user can use up/down navigator buttons to scroll the list.



Picture 6 - Scroll icon

7.4 Phone Status

The phone status includes the following information about the phone:

- Network Status:

  VLAN ID

  Mode(IPv4 or IPv6 status)

  ETH IP(IP Address)

- The Phone Device Information:

  ETH Mac Address

  SN

  Phone Mode

  Hardware Version number

  Software Version number

  Uboot Version number

  Phone Storage (RAM and ROM)

  System Running Time

- SIP Account Information:

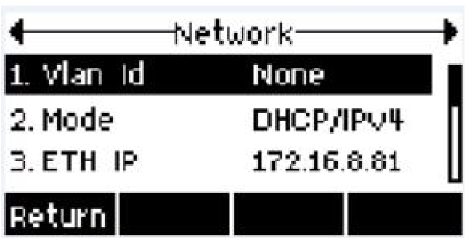
  SIP Account

SIP Account Status (register / Inactive/ trying / time out/Registration failed)

- TR069 Connect Status (Displays only in the phone interface state)

  The user can view the phone status through the phone interface and the web interface.

- Phone interface: When the phone is in standby mode, press **【Menu】** >> **【Status】** and select the option to view the corresponding information, as shown in the figure:



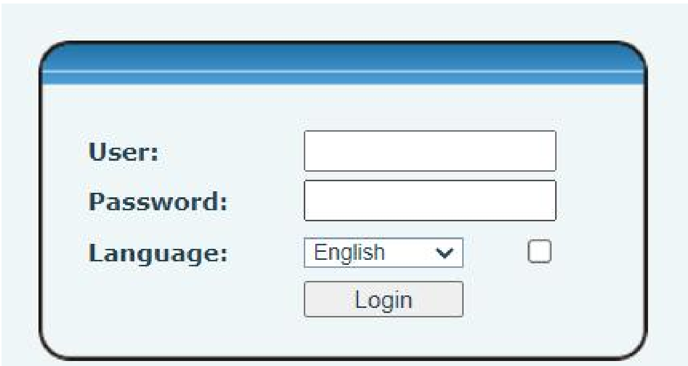
Picture 7 - The Phone status

- WEB interface: Refer to 7.5 Web Management to log in the phone page, enter the **【System】** >> **【Information】** page, and check the phone status, as shown in the figure:

Picture 8 - WEB phone status

7.5 Web Management

Phone can be configured and managed on the web page of the phone. The user first needs to enter the IP address of the phone in the browser and open the web page of the phone. The user can check the IP address of the phone by pressing [**Menu**] >> [**Status**].



Picture 9 - Landing page

Users must correctly enter the user name and password to log in to the web page. The default user name and password are "admin". For the specific details of the operation page, please refer to page 11 Web configuration.

7.6 Network Configurations

The device relies on IP network connection to provide service. Unlike traditional phone system based on a circuit switched wire technology, IP devices are connected to each other over the network and exchange data in packet basis based on the devices' IP address.

To enable this phone, you must first correctly configure the network configuration. To configure the network, users need to find the phone function menu button [**Menu**] >> [**System**] >> [**Network**] >> [**Network**].

The default password for advanced Settings is "123".

NOTICE! If user saw a ‘WAN Disconnected’ icon flashing in the middle of screen, it means the network cable was not correctly connected to the device’s network port. Please check the cable is connected correctly to the device and to the network switch, router, or modem.

The device supports three types of networks, IPv4/IPv6/IPv4&IPv6

There are three common IP configuration modes about IPv4

- Dynamic Host Configuration Protocol (DHCP) – This is the automatic configuration mode by getting network configurations from a DHCP server. Users need not to configure any parameters manually. All configuration parameters will be getting from DHCP server and applied to the device. This is recommended for most users.
- Static IP Configuration – This option allows user to configure each IP parameters manually, including IP Address, Subnet Mask, Default Gateway, the primary DNS, the secondary DNS server, and the DNS domain name. This is usually used in an office environment or by power users.
- PPPoE – This option is often used by users who connect the device to a broadband modem or router. To establish a PPPoE connection, user should configure username and password provided by the service provider.

The device is default configured in DHCP mode.

There are tow common IP configuration modes about IPv6

- DHCP - This is the automatic configuration mode by getting network configurations from a DHCP server. Users need not to configure any parameters manually. All configuration parameters will be getting from DHCP server and applied to the device. This is recommended for most users.
- Static IP configuration - this option allows users to manually configure each IP parameter, including IP address, Prefix length, gateway, primary and secondary DNS servers, DNS domain name. This usually applies to some professional network user environments.

Please see 10.6.2.1 network Settings for detailed configuration and use.

7.7 SIP Configurations

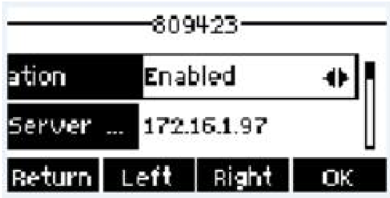
A line must be configured properly to be able to provide telephony service. The line configuration is like a virtualized SIM card. Just like a SIM card on a mobile phone, it stores the service provider and the account information used for registration and authentication. When the device is applied with the configuration, it will register the device to the service provider with the server's address and user's authentication as stored in the configurations.

The user can conduct line configuration on the interface of the phone or the webpage, and input the corresponding information at the registered address, registered user name, registered password and SIP user, display name and registered port respectively, which are provided by the SIP server administrator.

- Phone interface: To manually configure a line, the user can press the line key for a long time, or press the button in the function menu [**Menu**] >> [**System**] >> [**Accounts**] >> [**Line 1**] / [**Line 2**]>>[**Basic**]configuration, click OK to save the configuration.

NOTICE! User must enter correct PIN code to be able to advanced settings to edit line configuration. (The default password is 123)

The parameters and screens are listed in below pictures.



Picture 10 - Phone line SIP address and account information

- WEB interface: After logging into the phone page, enter [**Line**] >> [**SIP**] and select **SIP1/SIP2** for configuration, click apply to complete registration after configuration, as shown below:

Picture 11 - Web SIP registration

8 Basic Function

8.1 Making Phone Calls

■ **Default Line**

The device provides two line services. If both lines are configured, user can make or receive phone calls on either line. If default line is configured by user, there will be a default line to be used for making outgoing call which is indicated on the top left corner. To change the default line, user can press left/right navigator buttons to switch between two lines. Enable or disable default line, user can press [**Menu**] >> [**Features**] >> [**General**] >> [**Default Line**] or configure from Web Interface (Web / PHONE / Features / Basic Settings).

Picture 12 - Default line

■ **Dialing Methods**

User can dial a number by,

- Entering the number directly
- Selecting a phone number from phonebook contacts (Refer to 10.2.1 Local contacts)
- Selecting a phone number from cloud phonebook contacts (Refer to 10.2.3 Cloud Phone Book)
- Selecting a phone number from call logs (Refer to **10.3 Call Log**)
- Redialing the last dialed number

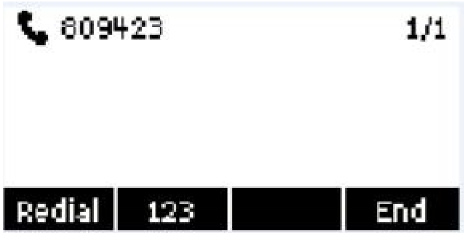
■ **Dialing Number then Opening Audio**

To make a phone call, user can firstly dial a number by one of the above methods. When the dialed number is completed, user can press [**Dial**] button on the soft-menu, or press hand-free button to turn on the speaker or headphone, or lift the handset to call out with the current line, or user can press line key(Configured by DSS Keys) to call out with specified line.

Picture 13 - Enable voice channel dialing

■ **Opening Audio then Dialing the Number**

Another alternative is the traditional way to firstly open the audio channel by lifting the handset, turning on the hands-free speaker or headphone by pressing hands-free button, or line key, and then dial the number with one of the above methods. When number dialed completed, user can press [**Dial**] button or [**OK**] button to call out, or the number will be dialed out automatically after timeout.



Picture 14 - Open the voice channel and dial the number

- **Cancel Call**

While calling the number, user can press end the audio channel by putting back the handset or pressing the hands-free button/headset button to drop the call.

Picture 15 - Call number

8.2 Answering Calls

When there is an incoming call while the device is idle, user will see the following incoming call alerting screen.

Picture 16 - Answering calls

User can answer the call by lifting the handset, open headphone or speaker phone/headset by pressing the hands-free/headset button, or the [Answer] button. To divert the incoming call, user should press [**Divert**] button. To reject the incoming call, user should press [**Reject**] button.

8.2.1 Talking

When the call is connected, user will see a talking mode screen as the following figure.

Picture 17 - Talking interface

Table 6 - Talking mode

Number	Name	Description
①	Voice channel	The icon shows the voice channel mode being used.
②	Call status	The call status of the current call-calling
③	Call the other end	The name (front) and number (back) of the other party in the call.
④	Numbers of line	Shows how many calls are present on the current device
⑤	Call duration	The duration of a call after it has been established.

8.2.2 Make / Receive Second Call

The device can support up to two concurrent calls. When there is already a call established, user can still answer another incoming call on either lines or make a second call on either lines.

- **Second Incoming Call**

When there is another incoming call during talking a phone call, this call will be waiting for user to answer it. User will see the call message in the middle of current screen. The device will not be ringing but playing call waiting tone in the audio channel of the current call and the LED will be flashing in green. User can accept or reject the call as same as normal incoming call. When the waiting call is answered, the first call will be put on hold automatically.

Picture 18 - The second call interface

- **Second Outgoing Call**

To make a second call, user may press [**Xfer**] / [**Conf**] button to make a new call on the default line or press the line key to make new call on specific line. Then dial the number the same way as making a phone call. Another alternative for making second call is to pressing DSS Keys dial out from the configured Keys (BLF/Speed Dial). When the user is making a second call with the above methods, the first call could be placed on hold manually first or will be put on hold automatically at second dial.

User can press up/down navigator buttons to switch screen page, and switch call focus by pressing [**Resume**] button.

■ **Ending One Call**

User may hang up the current talking call by closing the audio channel or press [**End**] button. The device will return to single call mode in holding state.

8.3 End of the Call

After the user finishes the call, the user can put the handle back on the phone, press the hands-free button or Softkey [**End**] key to close the voice channel and end the call.

8.4 Redial

- Redial the last outgoing number:
  When the phone is in standby mode, press the redial button and the phone will call out the last number dialed.
- Call out any number with the redial key:
  Enter the number, press the redial key, and the phone will call out the number on the dial.
- Press the redial key to enter the call record:
  Log in the phone page, enter [**Phone Settings**] >> [**Features**] >> [**Redial Settings**], check redial to enter the call record, press the redial button when standby to enter the call record page, and press again to call out the currently located number.

Picture 19 - Redial settings

8.5 Auto-Answering

User may enable auto-answering feature on the device and any incoming call will be automatically answered (not including call waiting). The auto-answering can be enabled on line basis.

The user can start the automatic answer function in the telephone interface or the webpage interface.

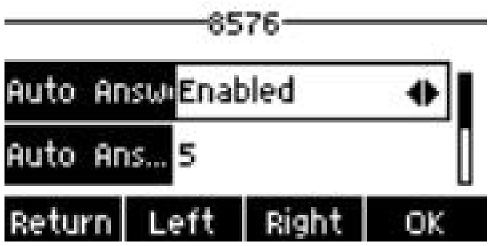
- **Phone interface:**

Press [**Menu**] >> [**Features**] >> [**Auto Answer**] button.

Press the button to select the line, use the left/right navigation key to turn on/off the auto answer option, and set the auto answer time to 5 seconds by default.

After completion, press [**OK**] key to save.

The icon in the upper right corner of the screen indicates that auto answer is enabled.

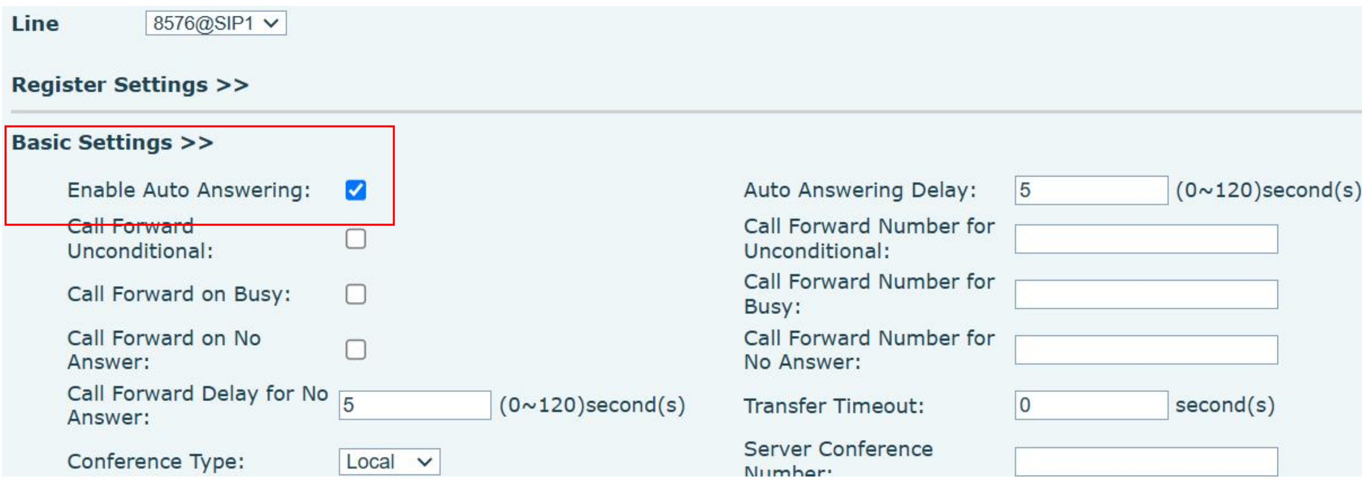


Picture 20 - Line 1 enables auto-answering

Picture 21 - The line has enabled auto-answering

- **WEB interface:**

  Log in the phone page, enter [**Line**] >> [**SIP**], select [**Basic settings**], start auto-answering, and click apply after setting the automatic answering time.



Picture 22 - Web page to start au wering

8.6 Mute

You can turn on mute mode during a call and turn off the microphone so that the local voice is not heard. Normally, mute mode is automatically turned off at the end of a call. You can also turn on mute on any screen (such as the free screen) and mute the ringtone automatically when there is an incoming call.

Mute mode can be turned on in all call modes (handles, headphones or hands-free).

8.6.1 Mute the Call

- During the conversation, press the mute button on the phone:

The call interface displays a mute icon, as shown in the figure:

Picture 23 - Mute the call

- Cancel mute: press cancel mute on the phone again. The mute icon is no longer displayed in the call screen.

8.6.2 Ringing Mute

- Mute: press the mute button when the phone is in standby mode:
  The top right corner of the phone shows the bell mute icon , Mute button red light is always on, when there is an incoming call, the phone will display the incoming call interface but will not ring.

Picture 24 - Ringing mute

- Cancel ring tone mute: On the standby or incoming call screen, press the mute button again or volume up cancel ring tone mute, no longer shows mute icon in upper right corner after cancel .The phone mute is off

8.7 Call Hold/Resume

The user can press the [**Hold**] button to maintain the current call, and this button will become the [**Resume**] button, and the user can press the "resume" button to restore the call.

Picture 25 - Call hold interface

8.8 DND

User may enable Do-Not-Disturb (DND) feature on the device to reject incoming calls (including call waiting). The DND can be enabled on line basis.

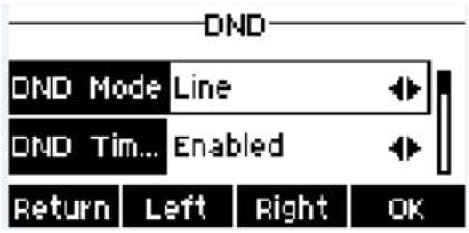
Enable/Disable phone all lines DND, Methods the following:

- Phone interface: Default standby mode，
  1) Press [**DND**] button to enter the DND setting interface, select line or phone to enable DND.
  2) Press [**DND**] button to enter the DND setting interface and disable DND.

Picture 26 - Enable DND

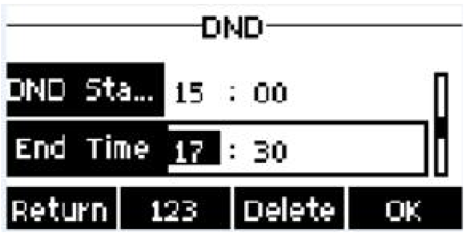
If the user wishes to enable/disable the uninterrupted function on a specific line, the user can set the uninterrupted function on the page of configuring the line.

1) Press [**Menu**] >> [**Features**] >> [**DND**] button, Enter the [**DND**] editing interface.
2) Click the left/right navigation button to select the line to adjust the mode and state of "do not disturb", and then press the [**OK**] button to save.
3) The user will see the DND icon turn red, and the sip-line has enabled the mode of "DND".



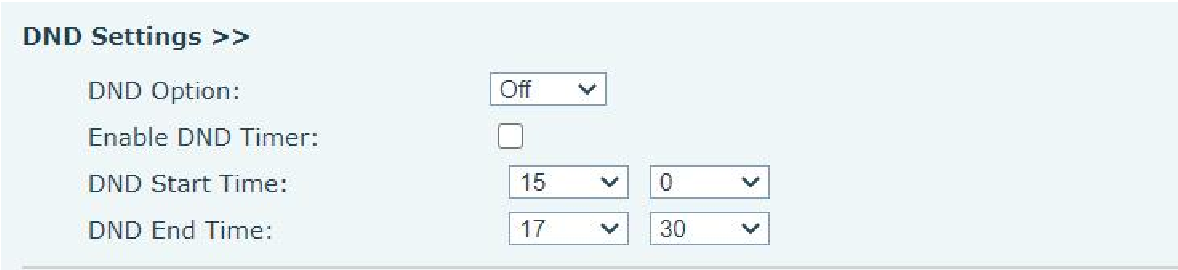
Picture 27 - DND setting interface

The user can also use the DND timer. After the setting, the DND function will automatically turn on and the



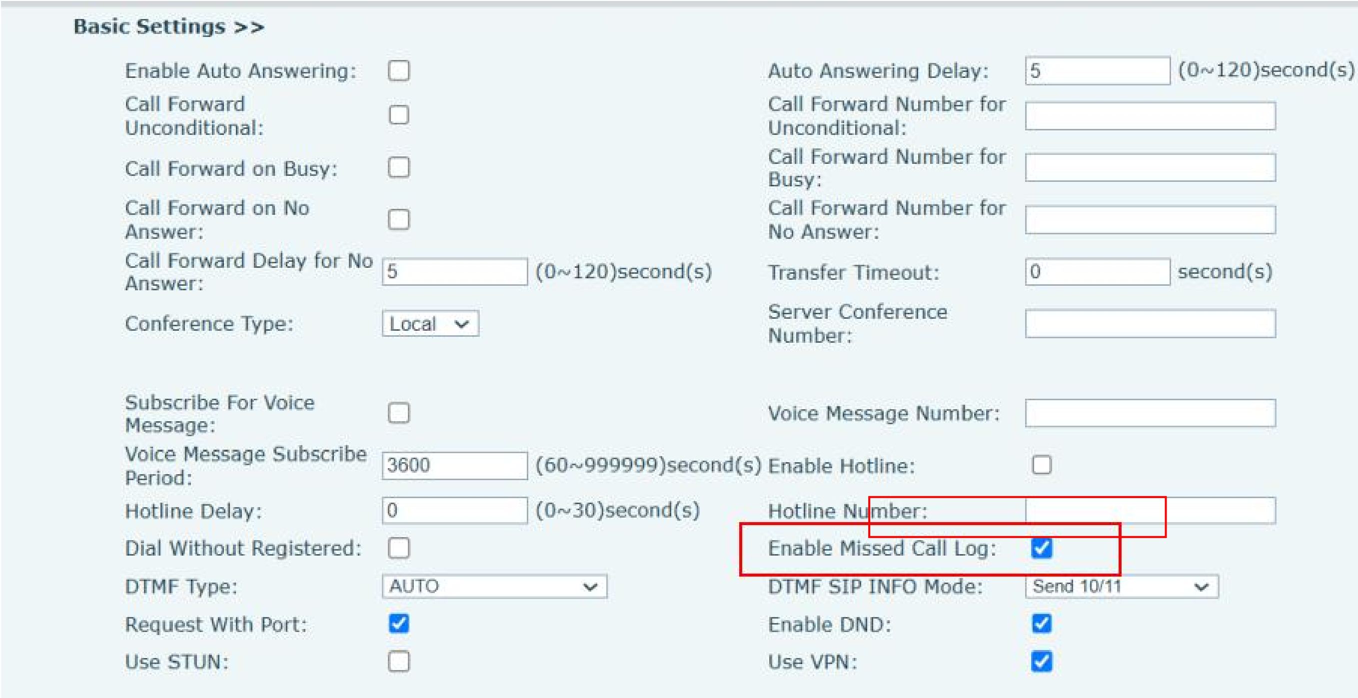
Picture 28 - DND timer

- WEB interface: Enter [**Phone setting**] >> [**Features**] >> [**DND Settings**]，set the DND type (off, phone, line), and DND timing function.



Picture 29 - DND Settings

The user turns on the DND for a specific route on the web page：Enter [**Line**] >> [**SIP**], select a [**Line**] >> [**Basic settings**], and enable DND.



Picture 30 - Line DND

8.9 Call Forward

Call forward is also known as ‘Call Divert’ which is to divert the incoming call to a specific number based on the conditions and configurations. User can configure the call forward settings of each line.

There are three types

- **Unconditional Call Forward –** Forward any incoming call to the configured number.
- **Call Forward on Busy –** When user is busy, the incoming call will be forwarded to the configured number.
- **Call Forward on No Answer –** When user does not answer the incoming call after the configured delay time, the incoming call will be forwarded to the configured number.
- Phone interface: Default standby mode
  1) Press [**Menu**] >> [**Features**] >> [**Call Forward**] button, select the line by up/down navigation key,

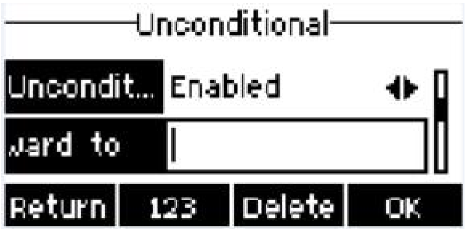
press [**OK**] button to set call forward.

Picture 31 - Select the line to set up call forwarding

2) Select the call forward type by pressing the up/down navigation button. Click [**OK**] to configure call forwarding.

Picture 32 - Select call forward type

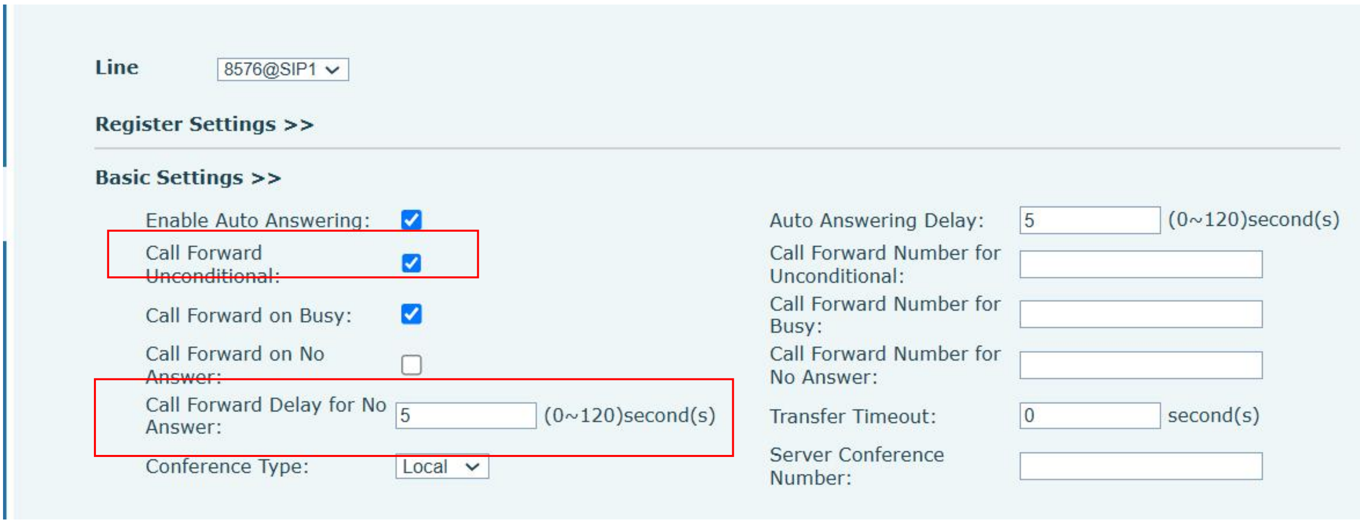
3) Select enable/disable by pressing the left/right navigation button.



Picture 33 - Enable call forward and configure the call forwarding number

4) Browse the parameters set by the up/down navigation key and enter the required information. When finished, press the [**OK**] button to save the changes.

- WEB interface: Enter [**Line**] >> [**SIP**], Select a [**Line**] >> [**Basic Settings**], and set the type, number and time of forward forwarding.



Picture 34 - Set call forward

8.10 Call Transfer

When the user is talking with a remote party and wish to transfer the call to another , there are three way to transfer the call, blind transfer, attended transfer and Semi-Attended transfer.

- Blind transfer: Do not need to negotiate with the other side, directly transfer the call to the other side.
- Semi-Attended transfer: When you hear the ring back, transfer the call to the other party.
- Attended transfer: When the caller answers the call, transfer the call to the caller.

Note！ For more transfer Settings, please refer to 12.6 Line >> Dial Plan。

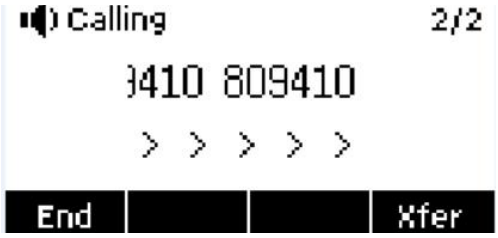
8.10.1 Blind transfer

During the call, the user presses the 'Transfer' button, enters the number to be transferred, and presses the transfer button again. After the third-party rings, the phone displays' Transfer Successful 'and hangs up.

Picture 35 - Transfer interface

8.10.2 Semi-Attended transfer

During the call, the user presses the 'Transfer' button on the phone, enters the number to be transferred, presses the call button, and if the third party does not answer, presses the transfer button on the call interface to perform a half attendance transfer or press the end button to cancel the half attendance transfer.



Picture 36 - Semi-Attended transfer

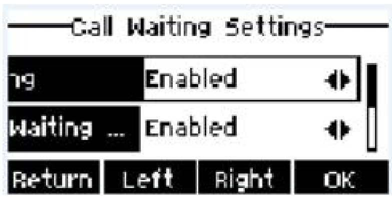
8.10.3 Attended transfer

Attendance transfer is also known as "courtesy mode", which is to transfer the call by calling the other party and waiting for the other party to answer the call.
Calling is the same procedure. In dual call mode, press the "transfer" button to transfer the first call to the second call.

Picture 37 - Attended transfer

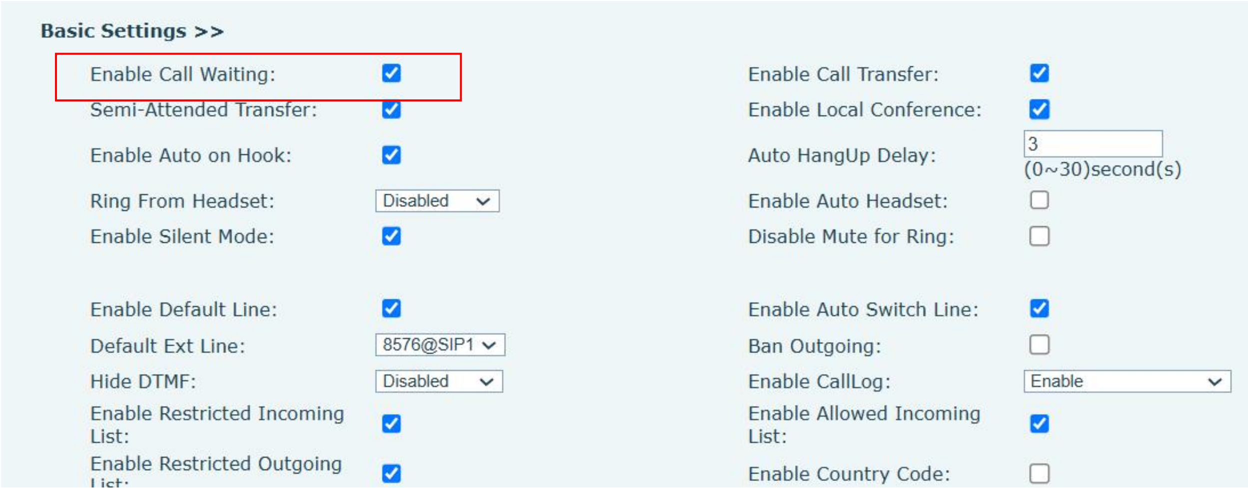
8.11 Call Waiting

- Enable call waiting: new calls can be accepted during a call.
- Disable call waiting: new calls will be automatically rejected and a busy tone will be prompted.
- Enable call waiting tone: when you receive a new call on the line, the tone will beep.
  The user can enable/disable the call waiting function in the phone interface and the web interface.
- Phone interface: Press [**Menu**] >> [**Features**] >> [**Call waiting**] >> [**Call Waiting Settings**], the navigation key left/right button to enable/disable call waiting and call waiting tone.

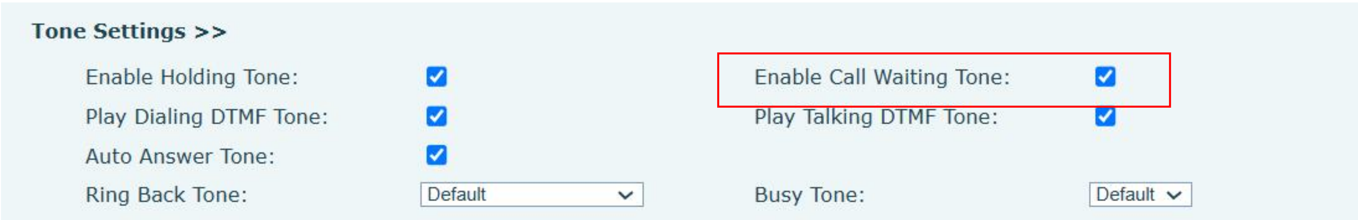


Picture 38 - Call waiting setting

- WEB interface: Enter [**Phone Settings**] >> [**Features**] >> [**Basic Settings**], enable/disable call waiting and call waiting tone.



Picture 39 - Web call waiting setting



Picture 40 - Web call waiting tone setting

8.12 Conference

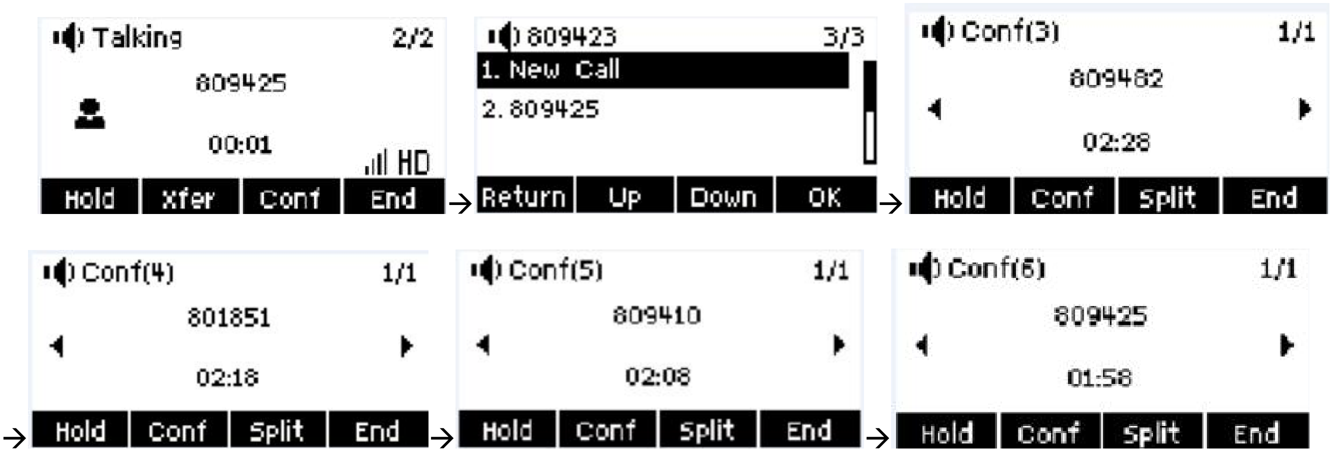
8.12.1 Local Conference

To conduct local conference, the user needs to log in the webpage and enter [**Line**] >> [**SIP**] >> [**Basic Settings**]. The meeting mode is set as local (the default is local mode), as shown in the figure:

Picture 41 - Local conference setting

Two ways to create a local conference:

1） If the device has two calls, press the conference key on the call interface, select another existing number when selecting the conference number, and press the confirm key to establish a local three-way conference. When the device is in a three-way conference, another call is made. Answer Press Conference to join a 4-party conference. In the same way, you can join the 5-party conference and the 6-party conference.



Picture 42 - Local conference (1)

2）There is a call on the device, press the conference button on the call interface, enter the number to join the conference, and press the call; after the opposite end answers, press the conference button again to establish a local three-party conference, and join the four-party conference in the same way. In the same way, you can join the five-party conference and the six-party conference. As shown in the figure:

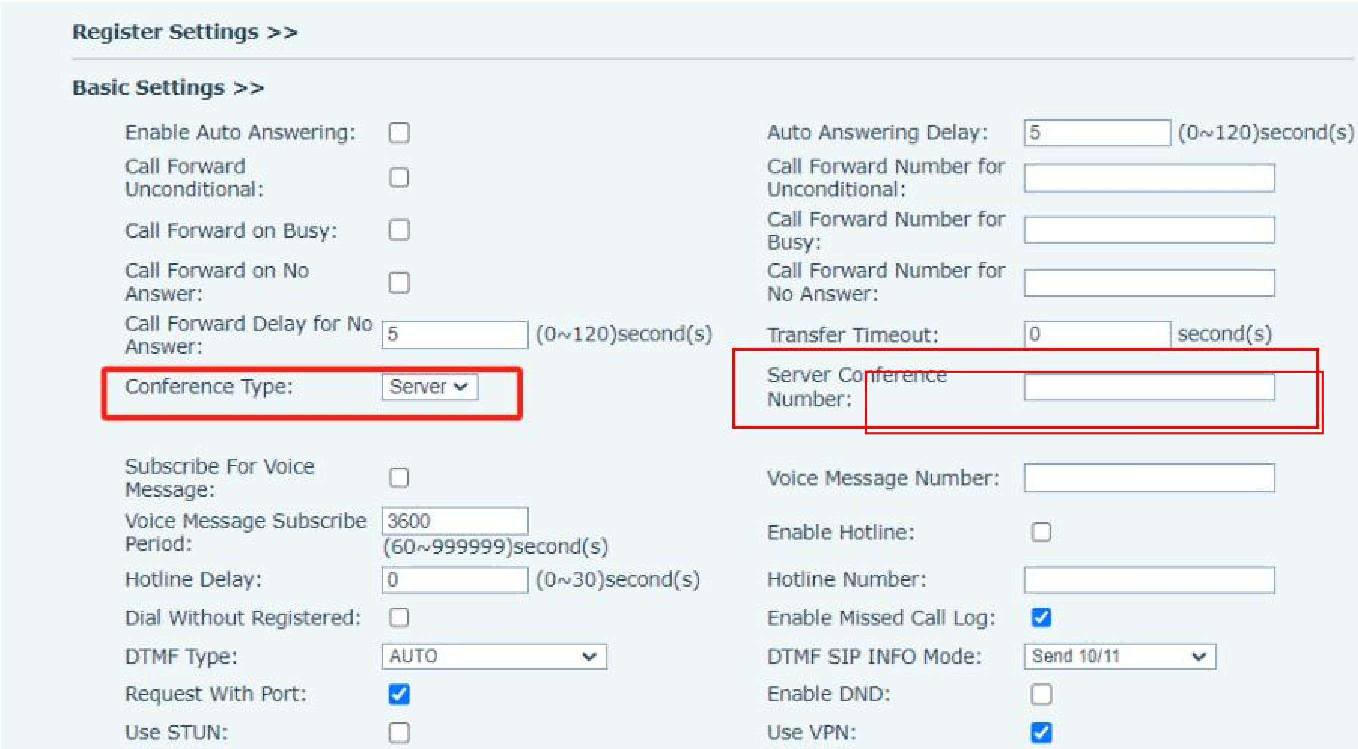
Picture 43 - Local conference (2)

Note: During the conference, press the split button to split the conference and press the end button to end the call.

8.12.2 Network Conference

Users need server support for network conference.

Log in the web page, enter [**Line**] >> [**SIP**] >> [**Basic settings**], set the conference mode as server mode (default is local mode), set the server conference room number (please consult your system administrator), as shown in the figure:



Picture 44 - Network conference

Method to join a network conference:

- Multi-party call number of network conference room and enter the password then all enter the conference room.
- The two phones have established common calls. Press the conference button to invite new members to the conference. Follow the voice prompt to operate.

Note: the upper limit of the number of participants in the network conference varies according to the server.

8.13 Call Park

Call park requires server support. Consult your system administrator for support.

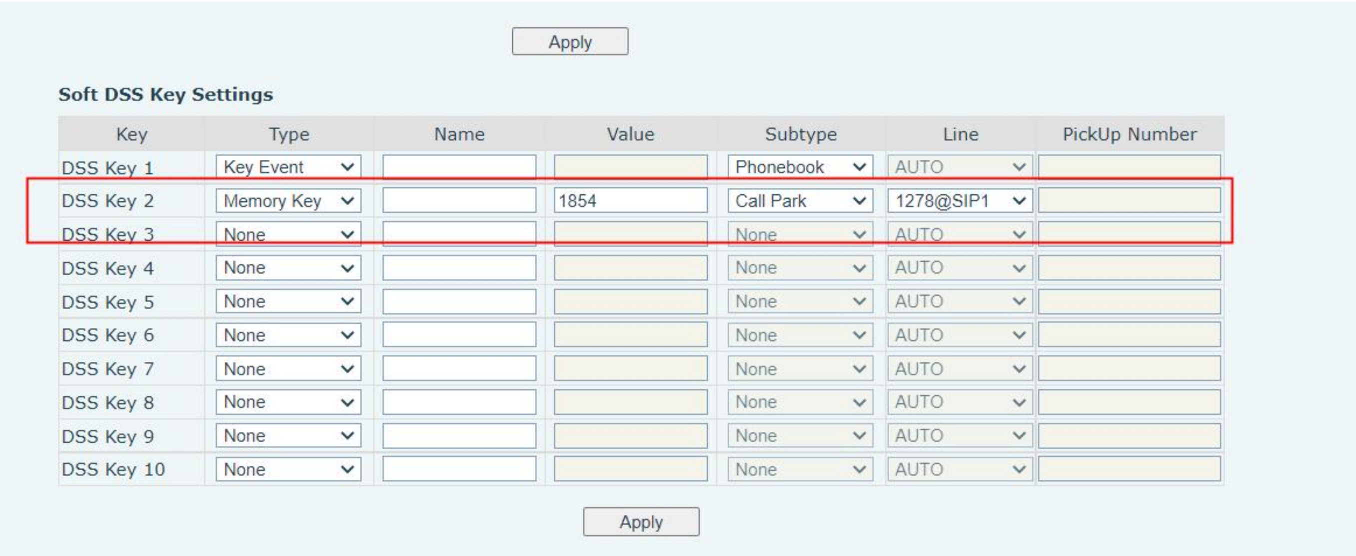
When you are on the call, if it is not convenient to answer the phone at this time, you can press the configured park button to hold the call. After a successful park, you can resume the call by pressing the configured park button on other devices.

Set the call park button:

- Phone interface: Through the [**Menu**] >> [**Basic**] >> [**Keyboard**] >>[**Soft DSS key Settings**]enter the function key Settings interface, Set function key type as memory and subtypes to call park, reside values for the server calls park number, set up corresponding SIP lines.

Picture 45 - Phone set call park

- WEB interface: log in the phone page, enter the [**Function Key**] >> [**Softkey**] >>[**Soft DSS Key Settings**] page, select a Softkey, set the function key type as memory key, the subtype as call park, and the value as the call park number of the server, and set the corresponding SIP line.



Apply

Soft DSS Key Settings

Key	Type	Name	Value	Subtype	Line	PickUp Number
DSS Key 1	Key Event			Phonebook	AUTO	
DSS Key 2	Memory Key		1854	Call Park	1278@SIP1	
DSS Key 3	None			None	AUTO	
DSS Key 4	None			None	AUTO	
DSS Key 5	None			None	AUTO	
DSS Key 6	None			None	AUTO	
DSS Key 7	None			None	AUTO	
DSS Key 8	None			None	AUTO	
DSS Key 9	None			None	AUTO	
DSS Key 10	None			None	AUTO	

Apply

Picture 46 - WEB set call park

8.14 Pick Up

Pick up requires server support. Consult your system administrator for support.

You can use the Pick Up function to answer incoming calls from other users. The phone can pick up incoming calls by configuring Soft DSS Key for BLF and setting the Pick Up Number.

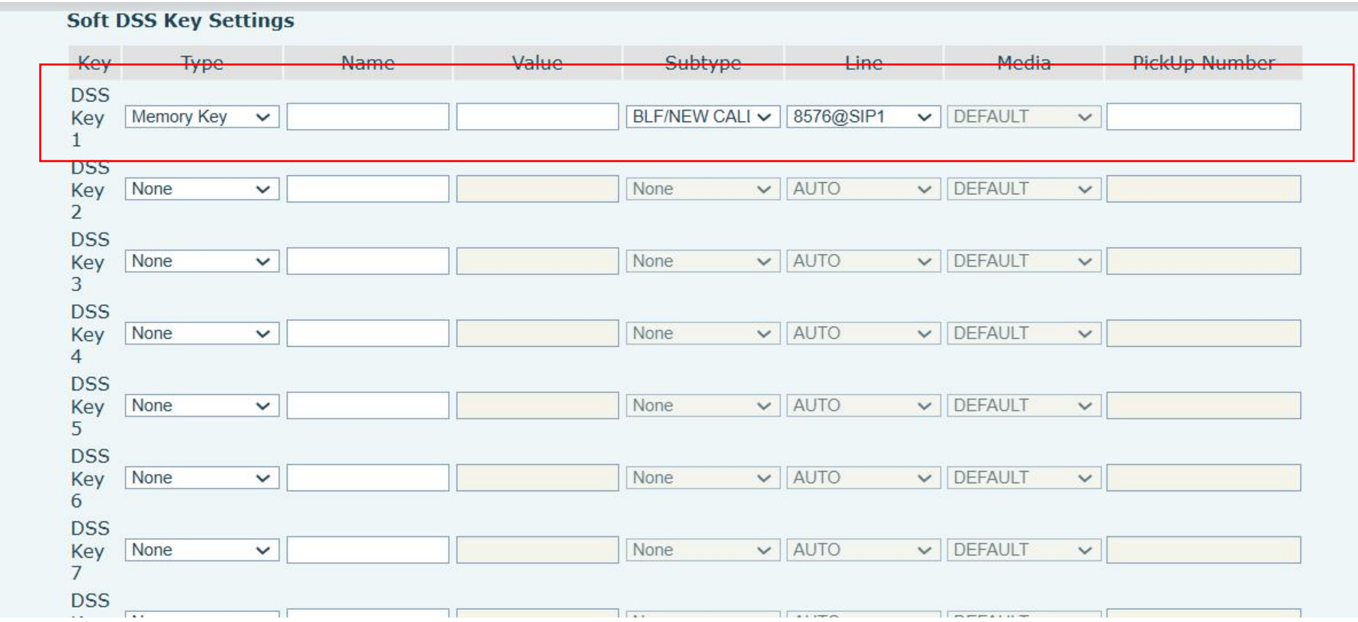
Phone interface: press [**Menu**] >> [**Basic**] >> [**Keyboard**] >>[**Soft DSS key Settings**], select the Softkey to set.

- Set the line, function key type as memory key, subtype as BLF/NEW CALL, set subscription number, and pick up code
- Other phones call the subscription number, and the opposite end is in the incoming ring.
- The caller picks up the call and speaks to it.

Picture 47 - Phone pick up setting

WEB interface: Log in the phone webpage, enter the [**Function Key**] >> [**Softkey**] >>[**Soft DSS Key Settings]** page, select a DSSkey, set the Function key type as memory key, the subtype as BLF/NEW

CALL, and set the corresponding SIP line and pick up codes.



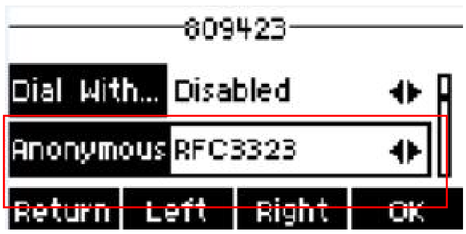
Picture 48 - WEB pick up setting

8.15 Anonymous Call

8.15.1 Anonymous Call

The phone can set up anonymous calls to hide the calling number and the calling name.

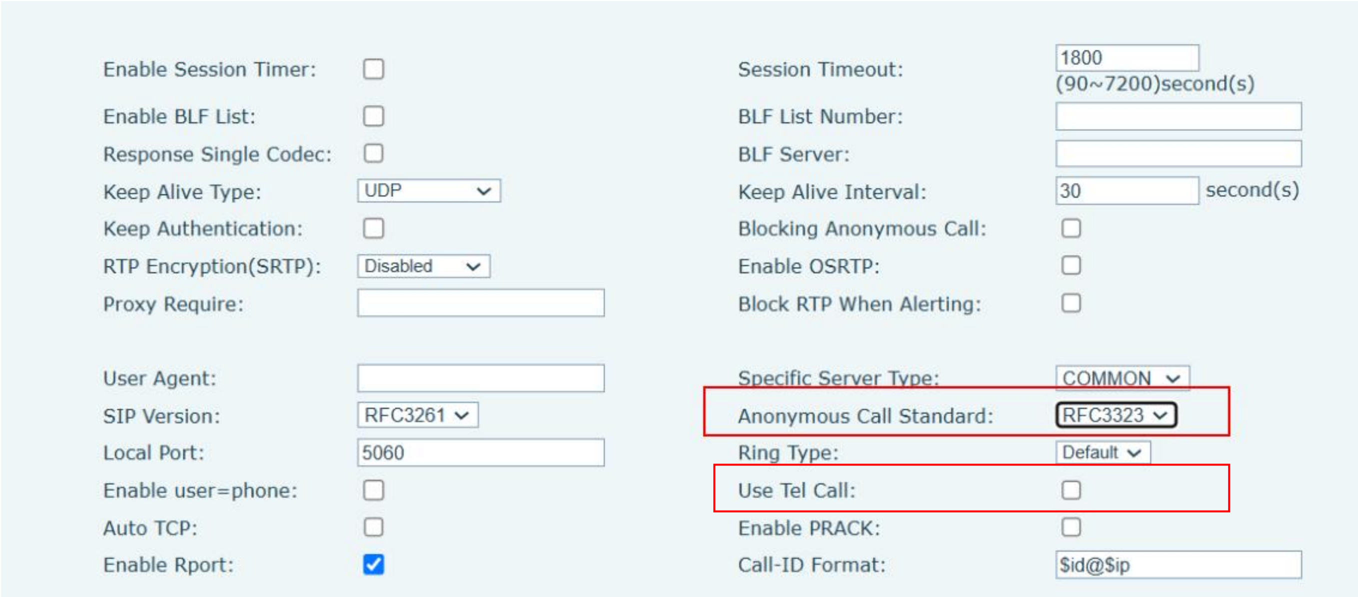
- You can see anonymity in the context of [**Menu**] >> [**Advanced**](default password:123) >> [**Accounts**] >> [**SIP**]>>[**Advanced**].
- The default is none, which is off, and RFC3323 and RFC3325 are optional.
- Select any one to open the anonymous call.



Picture 49 - Enable anonymous call

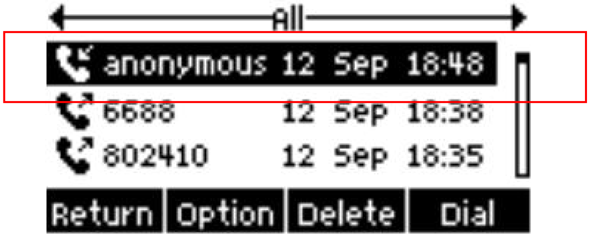
- On the web page [**Line**] >> [**SIP**] >> [**Advanced Settings**] can also open the anonymous call.

Note:Setting to enable anonymous calls also corresponds to the SIP line. That is, the setting under the SIP1 page can only take effect on the SIP1 line.



Picture 50 - Enable Anonymous web page call

The following is a transcript of an anonymous call received by the phone.



Picture 51 - Anonymous call log

8.15.2 Ban Anonymous Call

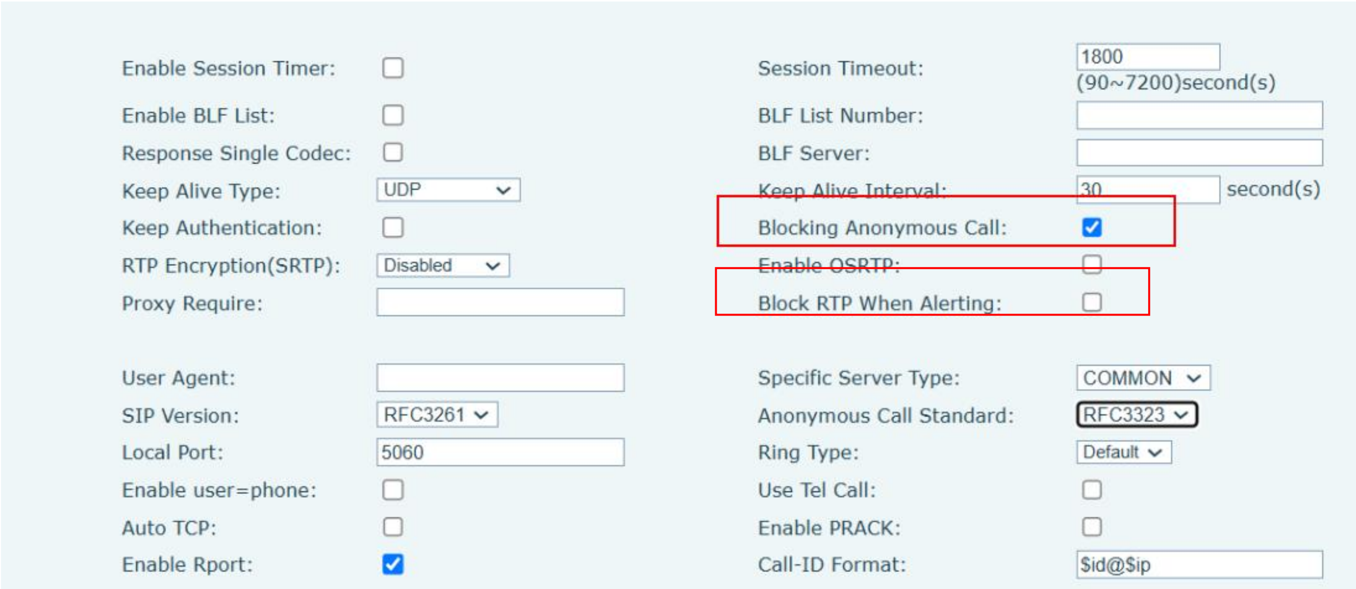
The device can be set to prohibit anonymous calls, that is anonymous calls to the number will be directly rejected.

- In the phone [**Menu**] >> [**Features**] >> [**Ban Anonymous Call**], click to enter and all SIP lines will be displayed.
- Click Softkey [**Switch**] or [**<**] [**>**] to switch the SIP line and enable anonymous call.

Picture 52 - Anonymous calls are not allowed on the phone

- On the web page [**Line**] >> [**SIP**] >> [**Advanced Settings**], also can disable anonymous calls.

Note:The setup to disable anonymous calls also corresponds to the SIP line. That is, the setting under the SIP1 page can only take effect on the SIP1 line.

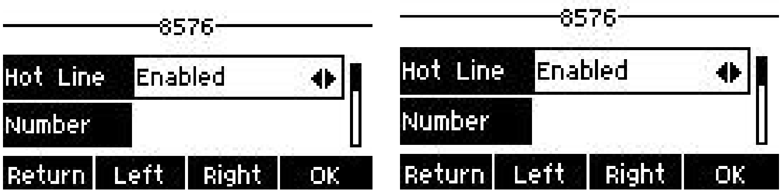


Picture 53 - Page Settings blocking anonymous call

8.16 Hotline

The device supports hotline dialing. After setting up the hotline dialing, directly pick up the handset, hands-free, earphone, etc., and the phone will automatically call according to the hotline delay time.

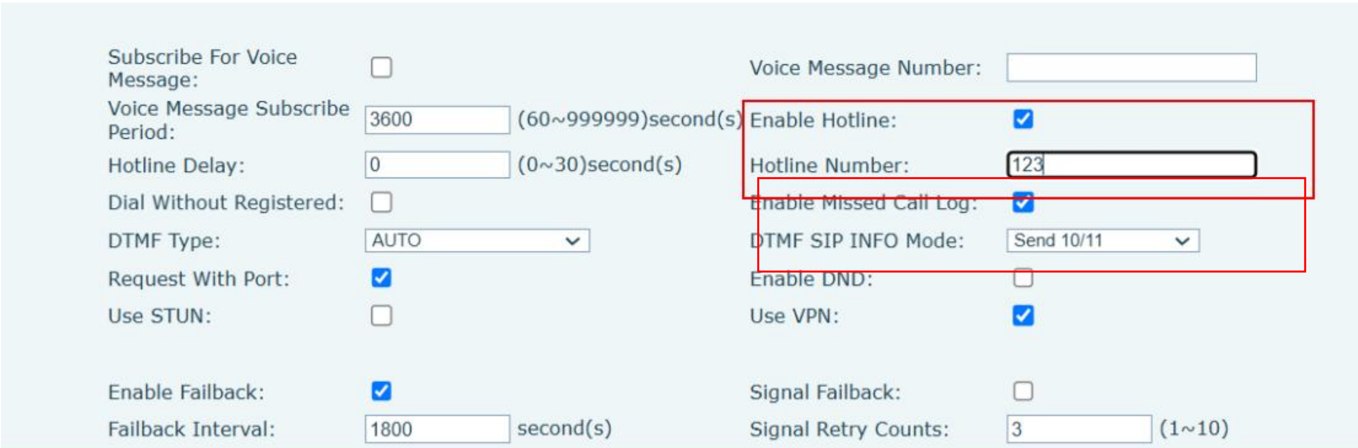
- In the phone [**Menu**] >> [**Features**] >> [**Advanced**] >> [**Hot Line**], click to enter and all SIP lines will be displayed.
- Then set the hotline for each SIP line, which is off by default.
- Open the hotline, set the hotline number, set the delay time of the hotline.



Picture 54 - Phone hotline setting interface

- On the website [**Line**] >> [**SIP**] >> [**Basic Settings**], can also set up a hotline.

Note:The setup hotline also corresponds to the SIP line. That is, the hotline set in the SIP1 webpage can only be activated in the SIP1 line.

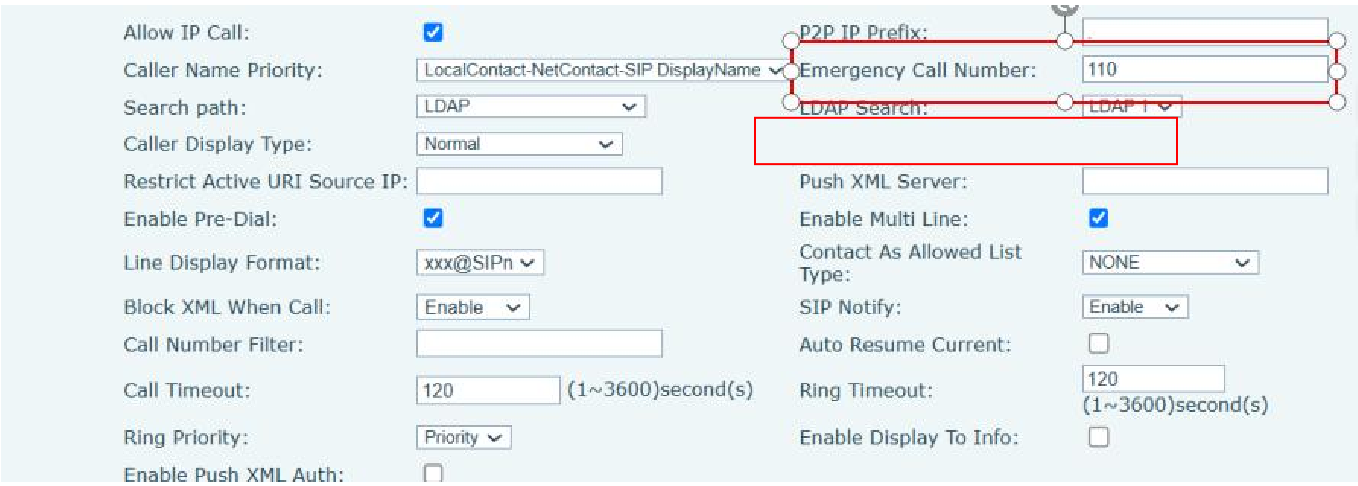


Picture 55 - Hotline set up on webpage

8.17 Emergency Call

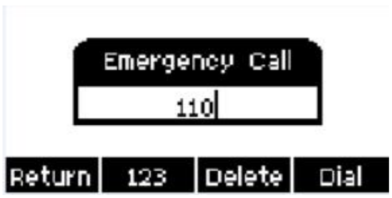
The emergency call function is used to enable the keypad lock. Users can set the corresponding emergency call number on the phone. You can also call emergency services when your phone is locked.

1) Configure the emergency call number: log in the phone page, enter the [**Phone Settings**] >> [**Features**]>> [**Basic Settings**]page, set up the emergency call code, if you need to set up more than one emergency call code, please use ", "to separate.



Picture 56 - Set up an emergency call number

2) When the phone set the keyboard lock, you can call the emergency call number without unlocking, as shown in the figure:



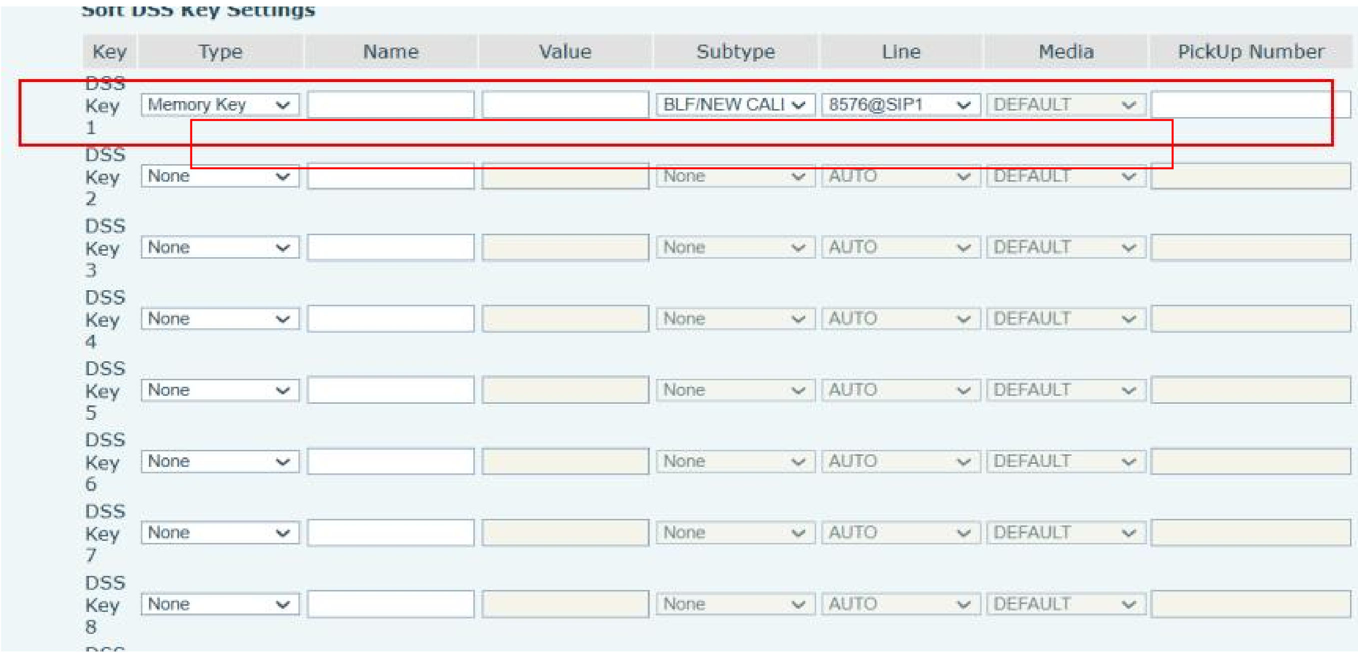
Picture 57 - Dial the emergency number

9 Advance Function

9.1 BLF (Busy Lamp Field)

9.1.1 Configure the BLF Functionality

- Page interface: log in the phone page, enter the [**Function Key**] >> [**Softkey**] >>[**Soft DSS Key Settings**] page, select a Softkey, set the function key type as memory key, choose subtype among BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, set BLF/DTMF value as the number to be subscribed, set the corresponding SIP line. The pickup number is provided by the server.The specific use of reference 8.16 Pick up,It can be left blank..



Picture 58 - Web page configuration BLF function key

- Phone interface: Going to the [**Menu**] >> [**Basic**] >> [**Keyboard**] >>[**Soft DSS key Settings**] to enter the function key settings interface, key function key types of memory, a subtype of BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, BLF/DTMF, the values to be subscription number, and set up corresponding SIP lines.

Picture 59 - Phone configuration BLF function key

Table 7 - BLF Function key subtype parameter list

Subtype	Standby is described	Calling is described
BLF/NEW CALL	Pressing the BLF key while standby to dial the subscriber number.	When you press this BLF key while talking to another user, you create a new call along with the subscribed number.
BLF/BXFER	Pressing the BLF key while standby to dial the subscriber number.	When you press this BLF key while talking to another user, you blind transfer the call to the subscribed number.
BLF/AXFER	Pressing the BLF key while standby to dial the subscriber number.	When you press this BLF key while talking to another user, you attendance transfer the call to the subscribed number.
BLF/CONF	Pressing the BLF key while standby to dial the subscriber number.	When you press this BLF key while talking to another user, you invite the subscriber number to join the meeting.
BLF/DTMF	Pressing the BLF key while standby to dial the subscriber number.	When the BLF key is pressed while talking to another user, the phone automatically sends the DTMF corresponding to the BLF key number.

9.1.2 Use the BLF Function

The BLF, also known as a "busy light field," notifies the user of the status of the subscribed object and is used by the server to pick up the call. BLF helps you monitor the other person's status (idle, ringing, talking).

BLF function:

- Call the subscribed number.
- Transfer calls/calls to the subscribed number.
- Pickup incoming calls from subscribed number.

1） Call the subscribed number.

When the phone is in standby mode, press the configured BLF key to call out the subscribed number.

2） Transfer calls/calls to the subscribed number.

Refer to Table 9.1.1-blf function key subtype parameter list, the BLF key can be used for blind transfer,attention-transfer and semi-attention-transfer of the current call, and also can invite the subscribed number to join the call and send DTMF, etc.

3） Pickup incoming calls from subscribed phones.

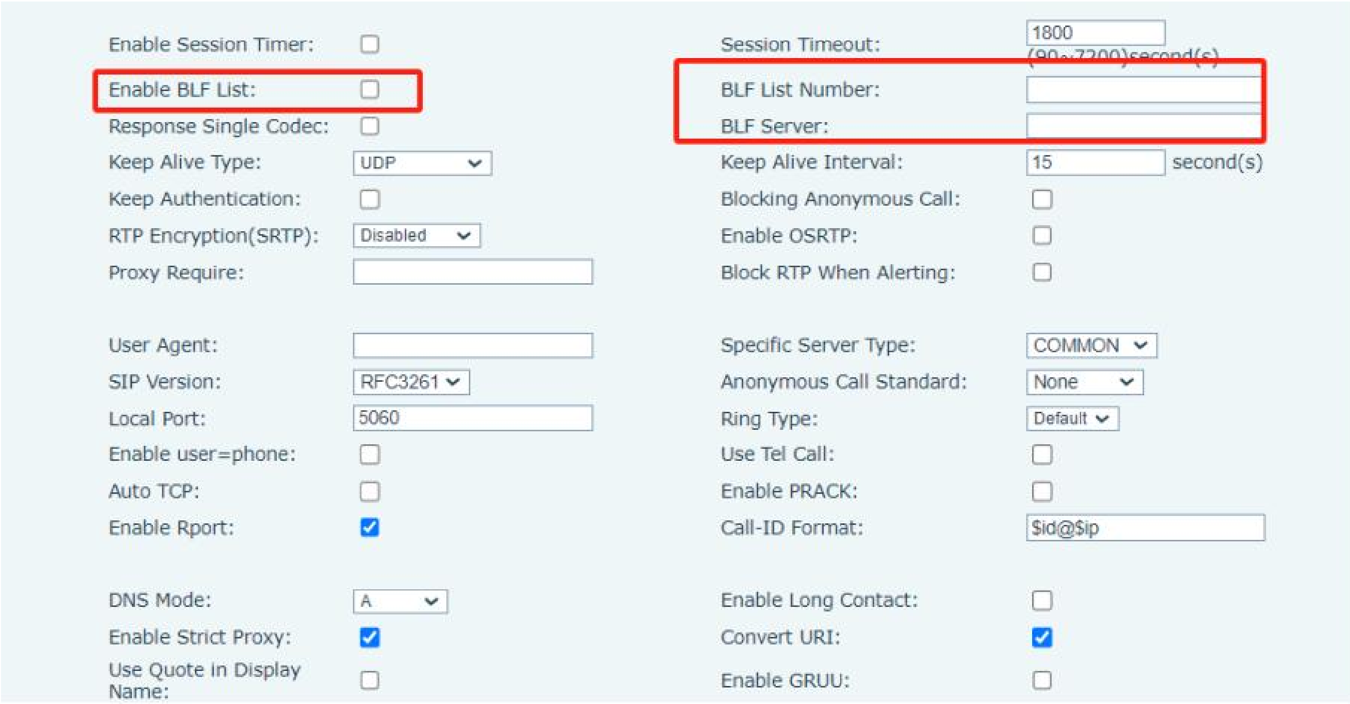
When configuring BLF function key, configure the pickup number.

9.2 BLF List

BLF List Key is to put the number to be subscribed into a group on the server side, and the phone uses the URL of this group to make unified subscription. The specific information, number, name and status of each number can be resolved based on notify sent from the server. The unoccupied Memory Key is then set to the

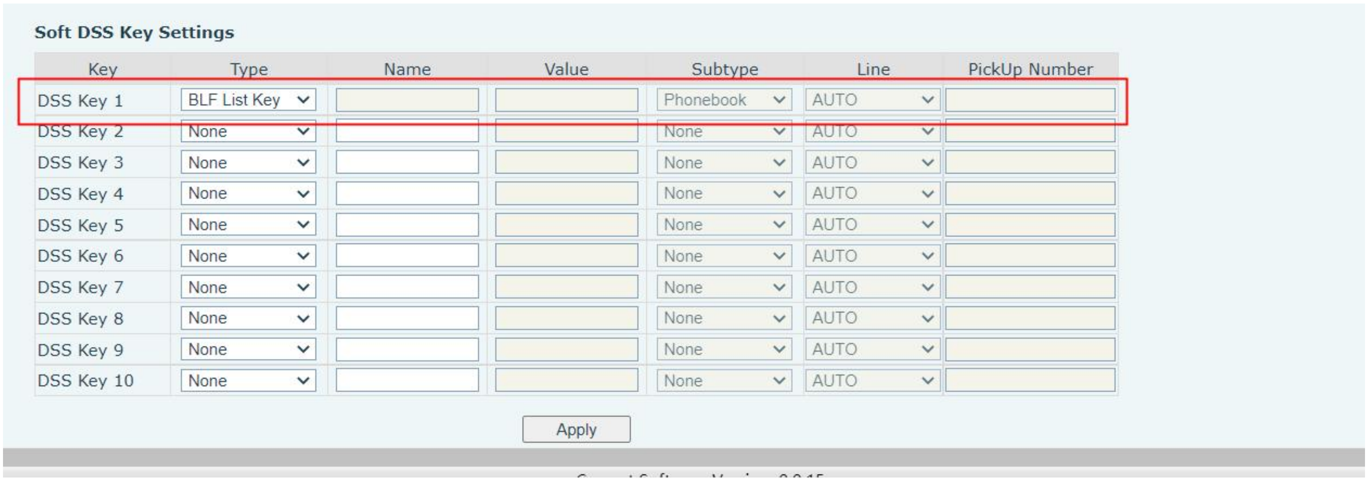
BLF List Key.

Configure BLF List function: log in the phone page, enter the [**Line**] >> [**SIP**] >> [**Advanced Settings**] page, open the BLF List, and configure the BLF List number.



Picture 60 - Configure the BLF List functionality

Use the BLF List function: when the configuration is complete, the phone will automatically subscribe to the contents of the BLF List group. Users can call and transfer the corresponding number by pressing the BLF List key.



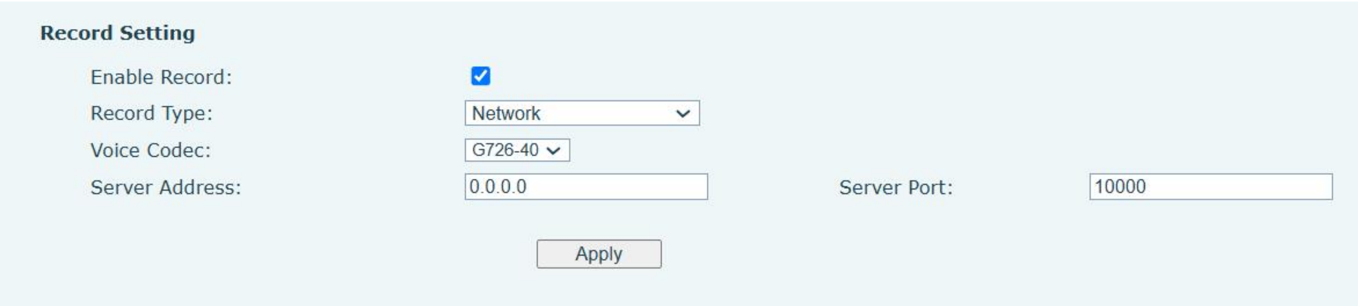
Picture 61 - BLF List number display

9.3 Record

The device supports recording during a call.

9.3.1 Server Record

When using the network server to record, it is necessary to open the recording in the phone web page [**Application**] >> [**Manage Recording**]. The type is selected as network, and the address and port of the recording server are filled in and the voice coding is selected. The web is as follows:

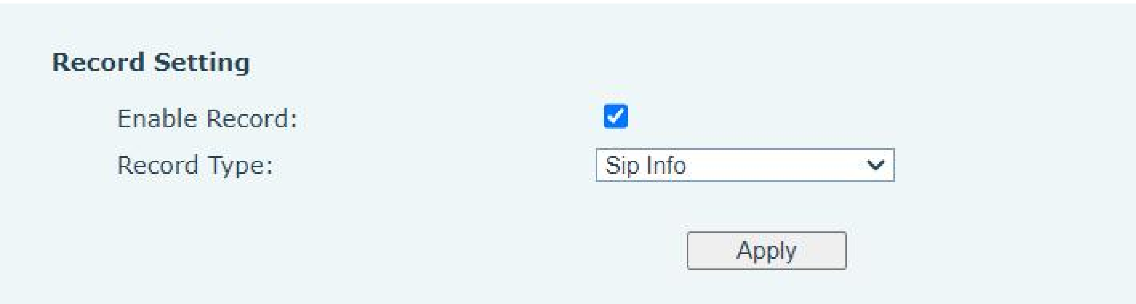


Picture 62 - Web server recording

Note: It needs to be used in conjunction with recording software.

9.3.2 Sip Info Record

The phone is registered with a server that supports SIP INFO recording. After registering the account, check the recording module of [**Application**] >> [**Manage Recording**] to open the recording, and the recording type is Sip Info



Picture 63 - Web SIP info recording

9.4 Agent

Agent (Agent function) of the phone can be realized: when multiple people use a device for Agent services at different times, he or she can quickly register his or her SIP account on the same server. The Agent functions of the phone can be divided into Normal and Hotel Guest. The Hotel Guest mode requires server support.

Normal Mode:

Configure agent function: set a DSSkey as agent, press the function key or enter the [**Menu**] >> [**Features**] >> [**Agent**] to enter the agent page. The SIP server needs to be configured before the account can be configured.

Picture 64 - Configure the agent account in normal mode

Picture 65 - Configure the proxy account-hotel Guest mode

Table 8 - Agency mode

Parameter	Description
Normal mode	
Number	Set the proxy account number.
User	Set the proxy account number to verify the user name.
Password	Set the proxy account number to verify the password.
Line	Select the SIP line.
CallLog	Users can choose to save all types, or delete all.
Hotel Guest mode	
Number	Set the proxy account number.
Password	Set the proxy account number to verify the password.
Line	Select the SIP line.
CallLog	Users can choose to save all types, or delete.
Status	Users can choose the status of the number, which can be:Logon/ Logoff/ Unavailable/ Available/ Wrap-up

Using agent functions:

1） When he phone has been configured on SIP server, fill in the correct number and user name password, click login and then the phone can be registered to the SIP server;

2） After registration, click logout and the phone can delete the user name and password, and log out of the SIP account.

3） Click Unregister and the phone retain the user name and password, and logs out of the SIP account.

9.5 Intercom

When the Intercom is enabled, it can automatically receive calls from the intercom.

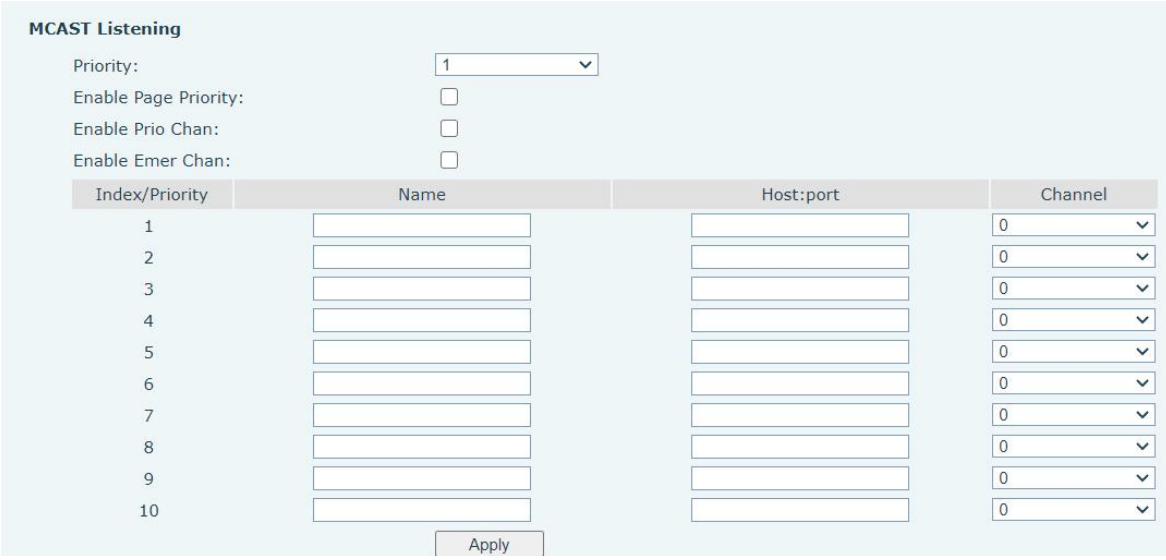
Picture 66 - Web Intercom configure

Table 9 - Intercom configure

Parameter	**Description**
Enable Intercom	When intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after specific delay.
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone
Enable Intercom Barge	Set whether it is possible to automatically answer the second intercom mode call when there is already one intercom mode call

9.6 MCAST

This feature allows user to make some kind of broadcast call to people who are in multicast group. User can configure a multicast DSS Key on the phone, which allows user to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address without involving SIP signaling. You can also configure the phone to receive an RTP stream from pre-configured multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.



Picture 67 - Multicast Settings Page

Table 10 - MCAST Parameters on Web

Parameters	**Description**
MCAST Send DTMF Mode	Set the DTMF mode sent by MCAST
Priority	Define the priority of the active call, 1 is the highest priority, 10 is the lowest.
Enable Page Priority	The voice call in progress shall take precedence over all incoming paging calls.
Enable Prio Chan	Set the priority to enable multicast listening on the current channel
Enable Emer Chan	The multicast of each channel is not affected by the order, and other multicasts can be interrupted at will
Index/Priority	Set the priority of the curent multicast
Name	Listened multicast server name
Host:port	Listened multicast server's multicast IP address and port.
Channel	Set the multicast channel

Multicast:

Go to web page of [**Function Key**] >> [**Softkey**] >>[**Soft DSS Key Settings**] , select the type to multicast, set the multicast address, and select the codec.

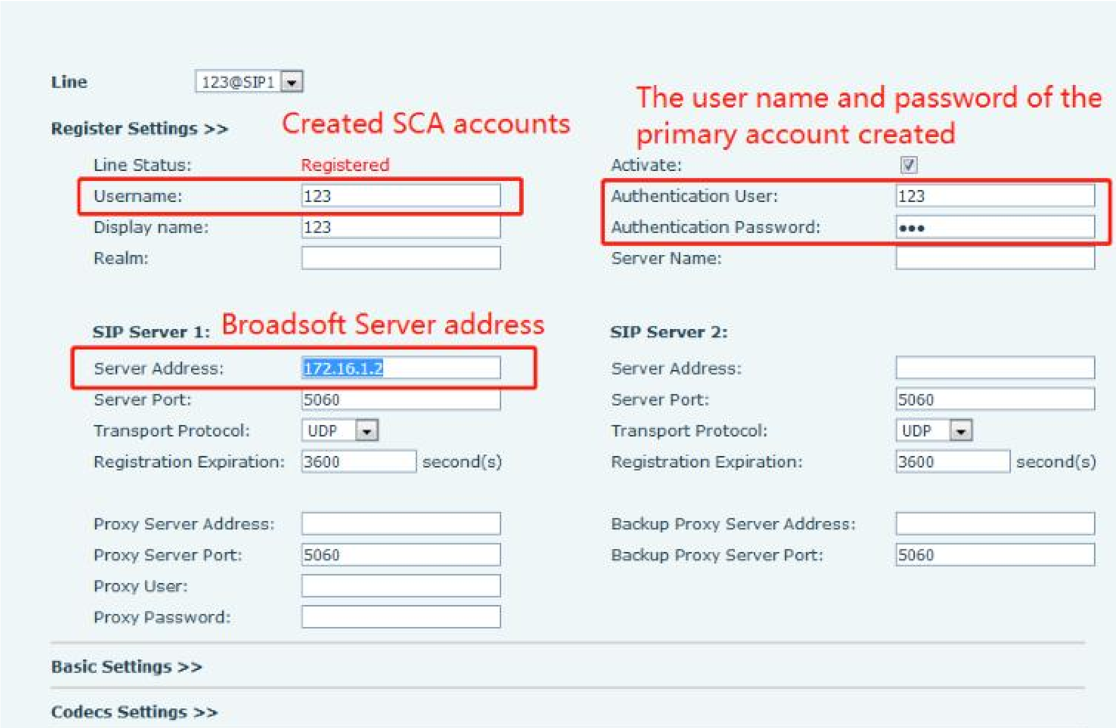
- After setting up, move the multicast key to the selected Softkey on the desktop display page
- Set up the name, host and port of the receiving multicast on the web page of [**Phone Settings**] >> [**MCAST**].
- Press the DSSKY of Multicast Key which you set.
- Receive end will receive multicast call and play multicast automatically.

9.7 SCA (Shared Call Appearance)

Users need the support of server end to use SCA function. You can refer to broadSoft SCA Server and Terminal Configuration Description

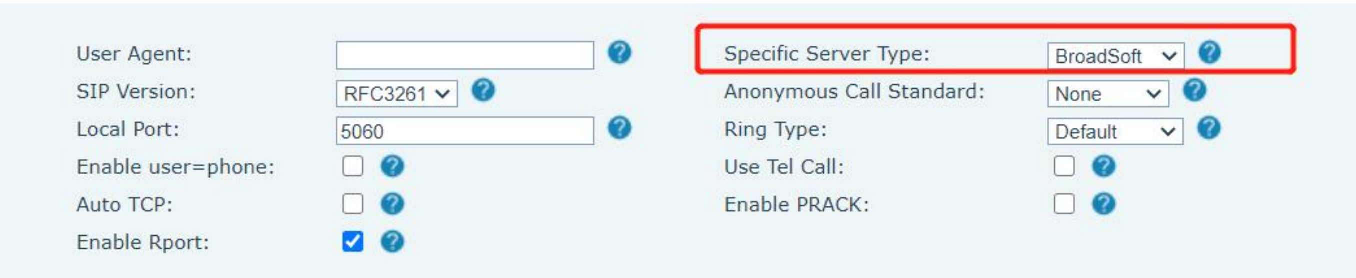
1） Configure on Phone

- When registering with the BroadSoft server, Phone can register the account created previously on multiple terminals.



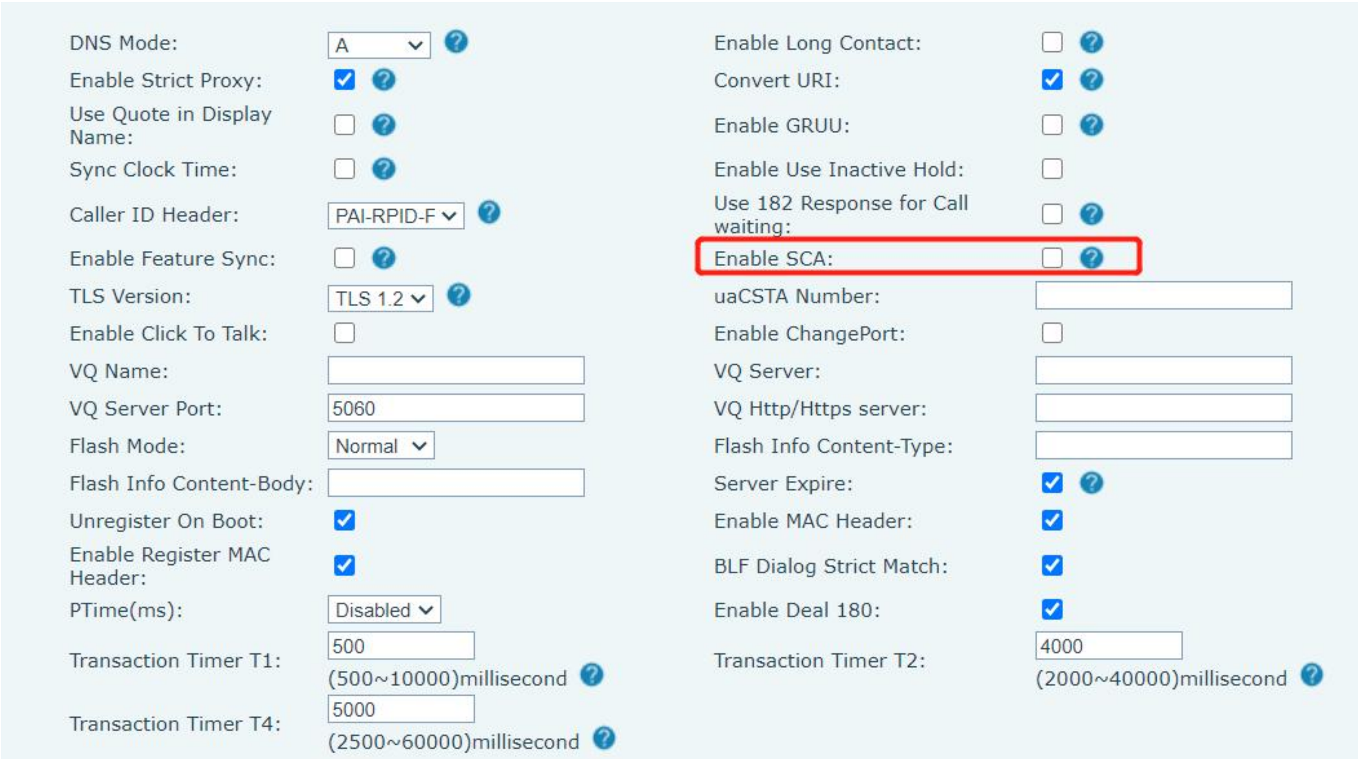
Picture 68 - Register BroadSoft account

- After the phone set registers with the BroadSoft server, a server type needs to be set. Specifically, log in to the webpage of the phone set, choose [**Line**] >> [**SIP**] >> [**Advanced Settings**] and set Specific Server Type to BroadSoft, as shown in the following figure.



Picture 69 - Set BroadSoft server

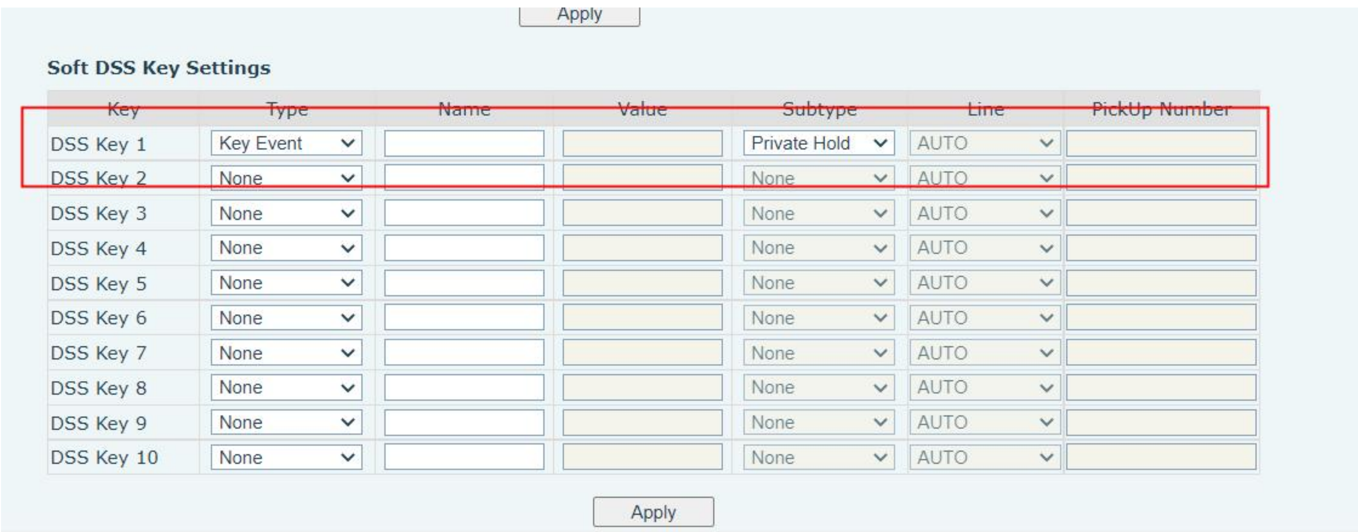
- If phone set needs to use the SCA function, enable it for the phone set. Specifically, log in to the webpage of the phone set, choose [**Line**] >> [**SIP**] >> [**Advanced Settings**], and select Enable SCA. If SCA is not enabled, the registered line is private line.



Picture 70 - Enable SCA

After configuring the account and successfully registering, in order to facilitate viewing the status of group calls, you can go to the shortcut key page to configure the shortcut key to enable the Shared Call Appearance Line. Each line key can represent a Call Appearance, and you can refer to Appendix III of 6.3 for the Line key light to understand the call status.

To facilitate private hold, configure keys whose DSS Key is Private Hold on the Function Key page. Pay attention that the public hold key is the softkey-hold key during a call.



Picture 71 - Set Private Hold Function Key

After each phone set registered with the BroadSoft server is configured as above, the SCA function can be used.

2） Shared Call Appearance(SCA)

The following lists a couple of instances to facilitate understanding.

In the following scenarios, the manager and secretary register the same SCA account and the account is configured based on the preceding steps.

Scenario 1: When this account receives an incoming call, the phone sets of both the manager and the secretary will receive the call and ring. If the manager is busy, the manager can reject the call and the manager's phone set stops ringing but the secretary's phone set keeps ringing until the secretary rejects/answers the call or the call times out.

Scenario 2: When this account receives an incoming call, if the secretary answers the call first and the manager is required to answer the call, the secretary can press the Public Hold key to hold this call and notify the manager. The manager can press the line key corresponding to the SCA to answer the call.

Scenario 3: The manager is in an important call with a customer and needs to leave for a while. If the manager does not want others to retrieve this call, the manager can press the Private Hold key.

Scenario 4: The manager is in a call with a customer and requires the secretary to join the call to make records. The secretary can press the corresponding SCA line key to barge in this call.

9.8 Message

9.8.1 SMS

If the service of the line supports the function of the short message, when the other end sends a text message to the number, the user will receive the notification of the short message and display the icon of the new SMS on the standby screen interface.

Picture 72 - SMS icon

Send messages:

- Go to the 'SMS Service' section under 'Messages'Go to [**Menu**] >> [**Message**] >> [**SMS**].Users can create new messages, select lines and send numbers.
- After editing is complete, click Send.

View SMS:

- If there is any unread message, click "OK" directly on the standby page to enter the SMS inbox interface. Click "OK" to read the unread message
- If you want to view the read information, select the message under "Menu", enter "SMS" under "Messages", press "OK" to enter the SMS page, select "Inbox" to view all the information

Reply to SMS:

- If there are unread messages, click "OK" directly on the standby page to enter the SMS inbox interface. Click "OK" to read the unread messages, then select "reply", edit, and click "send" to proceed
- Select the message under "Menu", enter "SMS" under "Messages", press "OK" to enter the SMS service page, select "Inbox", select the message you want to reply to, select "Reply", edit it, and click "Send" to send it

9.8.2 MWI (Message Waiting Indicator)

If the service of the lines supports voice message feature, when the user is not available to answer the call, the caller can leave a voice message on the server to the user. User will receive voice message notification from the server and device will prompt a voice message waiting icon on the standby screen.

Picture 73 - New Voice Message Notification

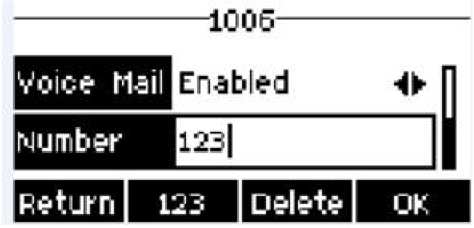
Voice message icon

To listen to a voice message, the user must first configure the voicemail number. After the voicemail number is configured, the user can retrieve the voicemail of the default line.

When the phone is in the default standby state.

- The phone buttons are pre-set with voice message shortcut keys - buttons.
- Press to open the voice message configuration interface, and select the desired route by pressing the up/down navigation button.
- Press the [**Edit**] button to edit the voice message number. When finished, press the [**OK**] button to save the configuration.
- In the following picture, “15” in front of line brackets represents unread voice messages, and “16” represents the total number of voice messages.

Picture 74 - Voice message interface



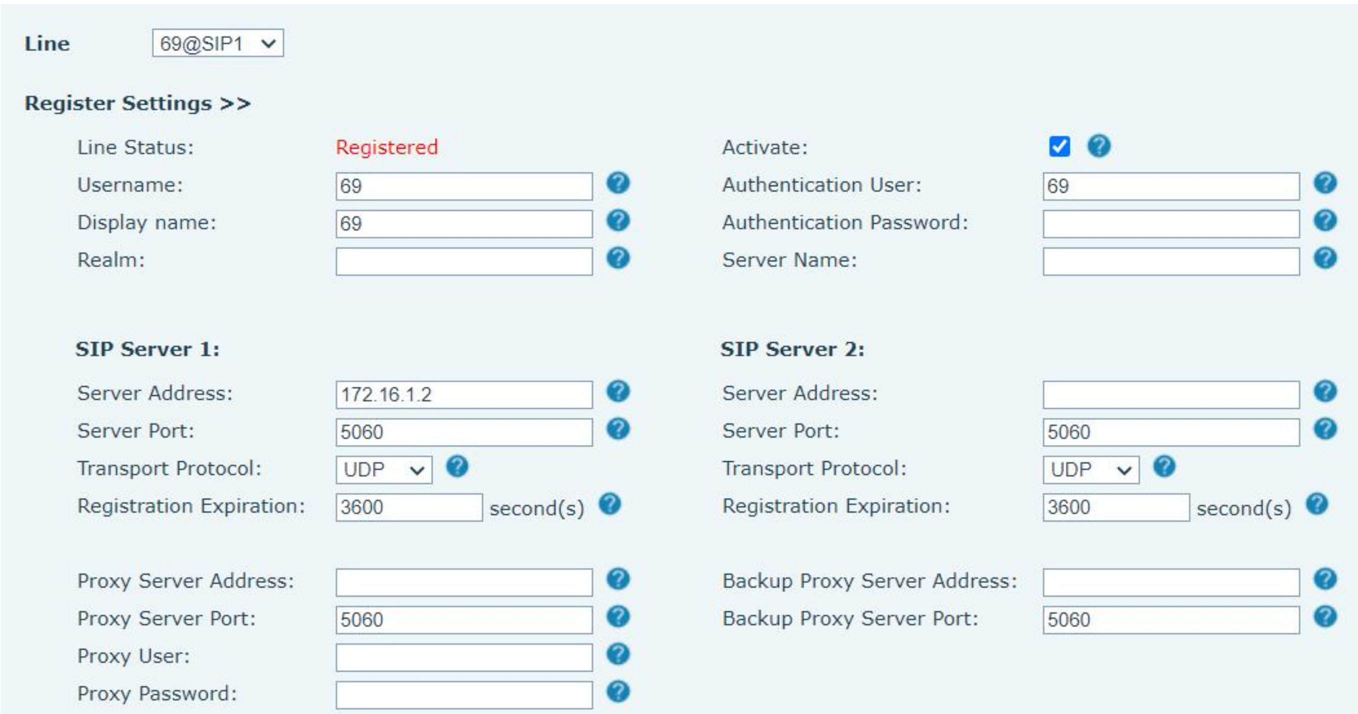
Picture 75 - Configure voicemail number

9.9 SIP Hotspot

SIP hotspot is a simple but practical function. With simple configurations, the SIP hotspot function can implement group ringing. SIP accounts can be expanded.

Phone set functions as a SIP hotspot and other phone sets (B and C) function as SIP hotspot clients. When somebody calls phone set A, phone sets A, B, and C all ring. When any phone set answers the call, other phone sets stop ringing. The call can be answered by only one phone set. When B or C initiates a call, the SIP number registered by phone set A is the calling number.

To set a SIP hotspot, register at least one SIP account.



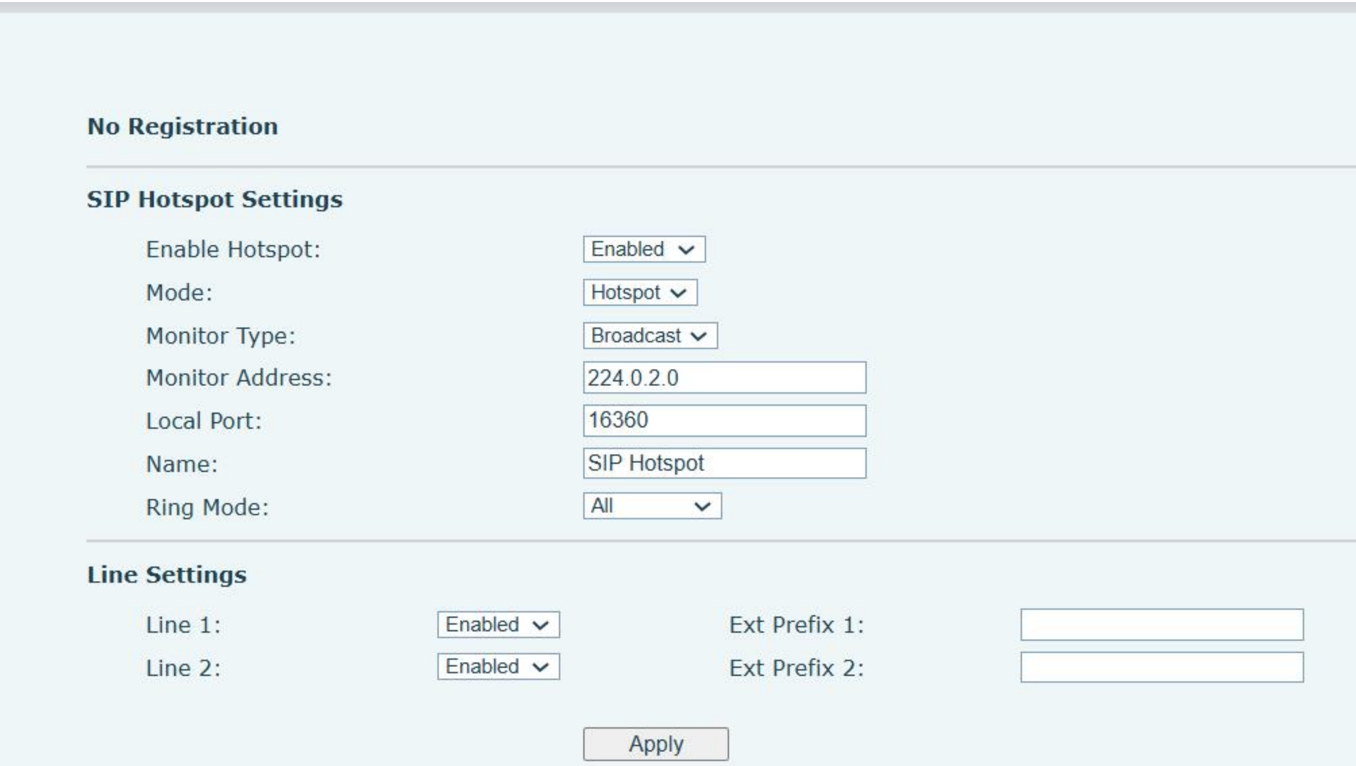
Picture 76 - Register SIP account

Table 11 - SIP hotspot Parameters

Parameters	Description
Hotspot Table	If your phone is set to “SIP hotspot server”, Device Table will display as Client Device Table which connected to your phone. If your phone is set to “SIP hotspot client”, Device Table will display as Server Device Table which you can connect to.

SIP hotspot	
Enable hotspot	Set it to be Enable to enable the feature.
Name	Fill in the name of the SIP hotspot. This configuration is used to distinguish different hotspots in the network and avoid connection conflicts
Mode	Choose hotspot, phone will be a “SIP hotspot server”; Choose Client, phone will be a “SIP hotspot Client”
Monitor Type	Either the Multicast or Broadcast is ok. If you want to limit the broadcast packets, you’d better use broadcast. But, if client choose broadcast, the SIP hotspot phone must be broadcast.
Monitor Address	The address of broadcast, hotspot server and hotspot client must be same.
Local Port	Type the Local port number.

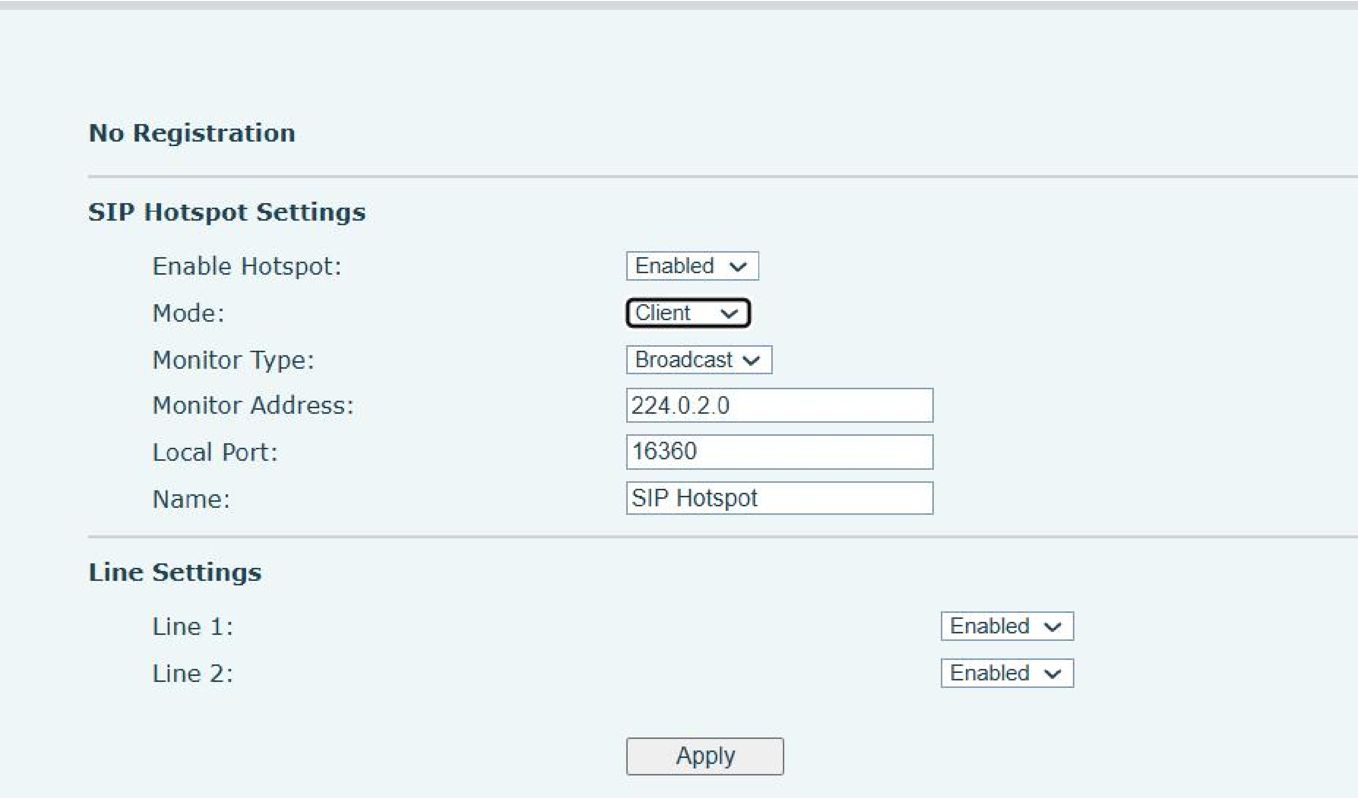
Configure SIP hotspot server:



Picture 77 - SIP hotspot server configuration

Configure SIP hotspot client:

To set as a SIP hotspot client, no SIP account needs to be set. The Phone set will automatically obtain and configure a SIP account. On the SIP Hotspot tab page, set Mode to Client. The values of other options are the same as those of the hotspot.



Picture 78 - SIP hotspot client configuration

As the hotspot server, the default extension number is 0. When the phone is used as the client, the extension number is increased from 1, you can view the extension number through the [**Line-SIP Hotspot**] page.

Call extension number:

- The hotspot server and the client can dial each other through the extension number.
- For example, extension 1 dials extension 0.

10 Phone Settings

10.1 Basic Settings

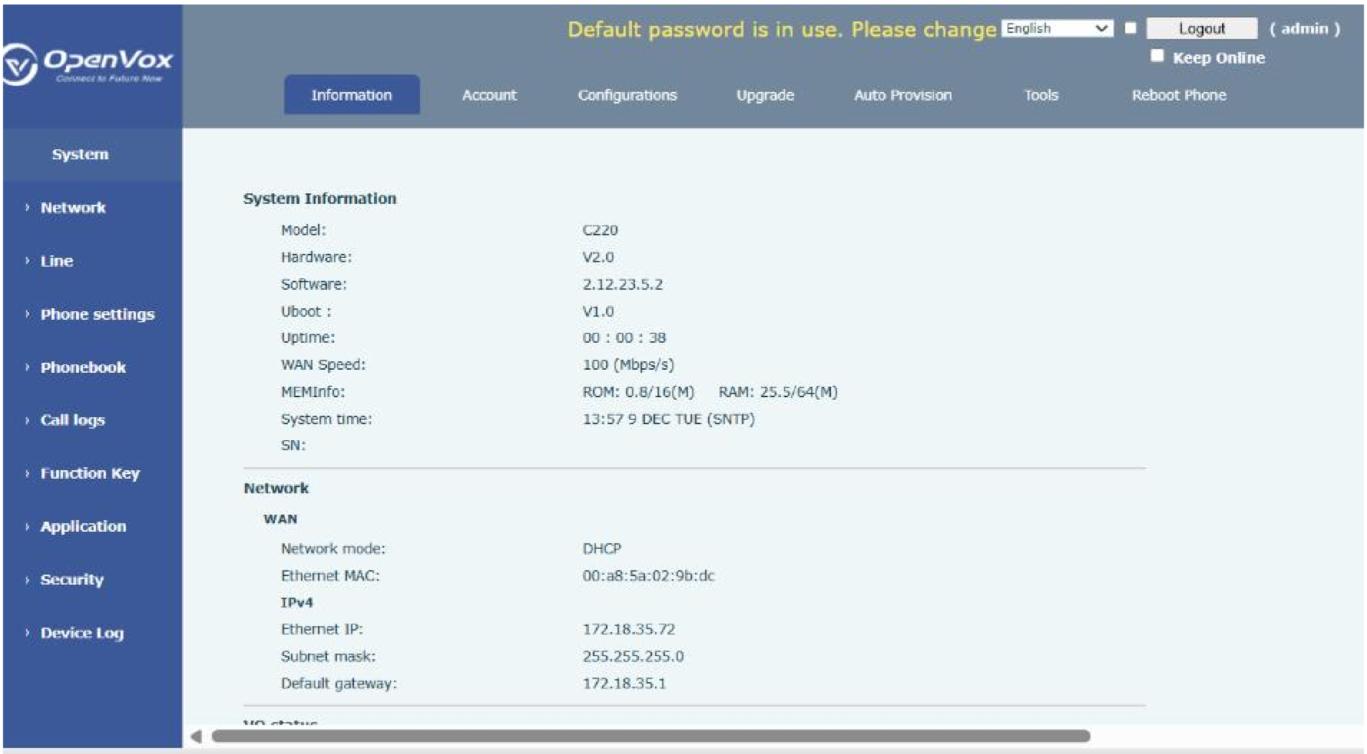
10.1.1 Language

The user can set the phone language through the phone interface and web interface.

- Phone end: After resetting the factory settings, the user needs to set the language; when setting the language during standby, go to [**Menu**] >> [**Basic**] >> [**Language**] Settings, as shown in the figure.

Picture 79 - Phone language setting

- Web interface: Log in to the phone webpage and set the language in the drop-down box at the top right corner of the page, as shown in the figure:



Picture 80 - Language setting on Web page

- The function box on the right side of the web interface language setting box is “Synchronic language to phone”; if selected, the phone language will be synchronized with the webpage language. If it is not selected, it will not be synchronized.

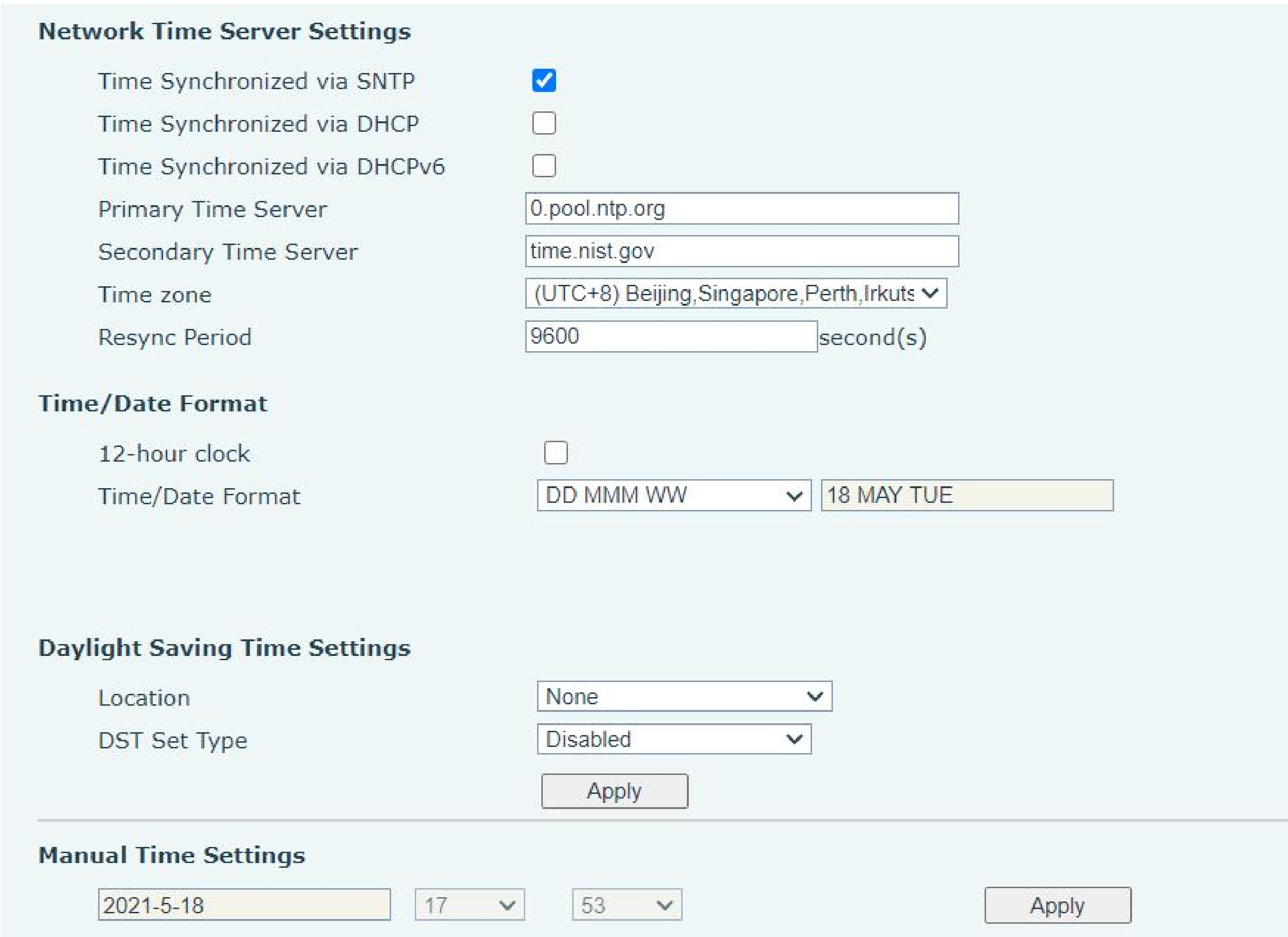
10.1.2 Time & Date

Users can set the phone time through the phone interface and web interface.

- Phone end: When the phone is in the default standby state, press the [**Menu**] >> [**Basic**] >> [**Time & Date**] , use the up/down navigation button to edit parameters, press the [**OK**] to save after completion, as shown in the figure:

Picture 81 - Set time & date on phone

- Web end: Log in to the phone webpage and enter [**Phone Settings**] >> [**Time/Date**] , as shown in the figure:



Picture 82 - Set time & date on webpage

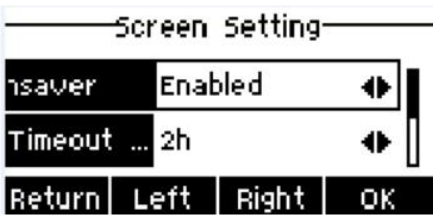
Table 12 - Time Settings Parameters

Parameters	Description
Mode	Auto/Manual Auto: Enable network time synchronization via SNTP protocol, default enabled. Manual: User can modify data manually.
SNTP Server	SNTP server address
Time zone	Select the time zone
Time/Date Format	Select time format from one of the followings: ■ DD MMM WW ■ MMM DD WW ■ WW DD MMM ■ WW MMM DD ■ DD MM YY ■ DD MM YYYY ■ MM DD YY ■ MM DD YYYY ■ YY MM DD ■ YYYY MM DD
Data Separator	Choose the separator between year and moth and day
12-Hour Clock	Display the clock in 12-hour format
Daylight Saving Time Settings	Enable or Disable the Daylight Saving Time.If the country or region does not have daylight saving time, there is no need to set it

10.1.3 Screen

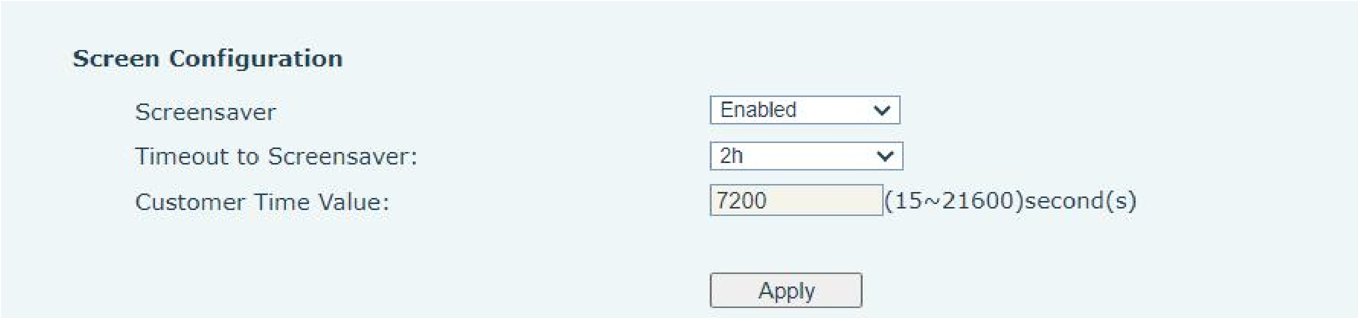
The user can set the phone screen parameters through both of the phone interface and web interface.

- Phone: When the phone is in the default standby state, go to [**Menu**] >> [**Basic**] >> [**Screen Settings**] to edit the screen parameters. After editing, click [**OK**] to save, as shown in the figure:



Picture 83 - Set screen parameters on phone

- Web : Go to [**Phone Settings**] >> [**Advanced**] Advanced, edit the screen parameters, and click Apply to save.



Picture 84 - Page screen Settings

10.1.3.1 Screen Saver

- Press [**Menu]**>>[**Setting**]>> [**Screen Settings**] to find the [**Screen protection**] button, press [**left**] / [**right**] button to open/close the screen protection, set the timeout time, the default is 15S, after completion, press [**OK**] button to save.
- After saving, return to standby mode and enter the screen saver after 15s, as follows:

Picture 85 - Phone screen saver

10.1.4 Ring

When the device is in the default standby mode,

- Press soft-button [**Menu**] till you find the [**Basic**] item.
- Enter [**Basic**] item till you find [**Ring**] item.
- Enter [**Ring**] item and you will find [**Headset**] or [**Handfree**] item, press left / right navigator keys to adjust the ring volume, save the adjustment by pressing [**OK**] when done.
- Enter [**Ring type**] item, press left / right navigator keys to change the ring type, save the adjustment by pressing [**OK**] when done.

10.1.5 Voice Volume

When the device is in the default standby mode,

- Press soft-button [**Menu**] till you find the [**Basic**] item.
- Enter [**Basic**] item till you find [**Voice Volume**] item.
- Enter [**Voice Volume**] item and you will find [**Handset**], [**Handfree**] and [**Headset**] item.
- Enter [**Handset**] or [**Handfree**] or [**Headset**] item, press Left / Right navigator keys to adjust the audio volume for different mode.
- Save the adjustment by pressing [**OK**] when done.

10.1.6 Greeting Words

When the device is in the default standby mode,

- Press soft-button [**Menu**] till you find the [**Basic**] item.
- Enter [**Basic**] item till you find [**Greeting Words**] item.
- Press [**OK**] to enter the setting interface to edit the Greetings Words.
- Save the adjustment by pressing [**OK**] when done.

NOTICE! The welcome message can only be displayed in the upper left corner of standby mode when the default option is disabled.

10.1.7 Reboot

When the device is in the default standby mode,

- Press soft-button [**Menu**] till you find the [**Basic**] item.
- Enter [**Basic**] item till you find [**Reboot**] item.
- Press [**OK**] a prompt message, "restart now," prompts the user.
- Press [**OK**] to restart the phone or [**Cancel**].

  The phone is in standby mode,
- Press [**OK**] to restart the phone or [**Cancel**] to exit.

10.2 Phone Book

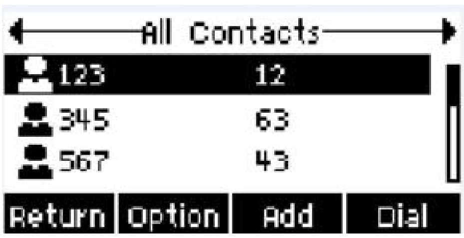
10.2.1 Local Contact

User can save contacts' information in the phone book and dial the contact's phone number(s) from the phone book. To open the phone book, user can press soft-menu button [**Contact**] in the default standby screen or keypad.

By default the phone book is empty, user may add contact(s) into the phone book manually or from call logs.

Picture 86 - Phone book screen

NOTICE! The device can save up to total 1000 contact records.



Picture 87 - Local Phone book

When there are contact records in the phone book, the contact records will be arranged in the alphabet order.

User may browse the contacts with up/down navigator keys. The record indicator tells user which contact is currently focused.

10.2.1.1 Add / Edit / Delete Contact

To add a new contact, user should press [**Add**] button to open Add Contact screen and enter the contact information of the followings,

- Name
- Office Number
- Mobile
- Other Number
- Line
- Ring Type
- Group

Picture 88 - Add New Contact

User can edit a contact by pressing [**Option**] >> [**Edit**] button.
To delete a contact, user should move the record indicator to the position of the contact to be deleted, press [**Option**] >> [**Delete**] button and confirm with [**OK**].

10.2.1.2 Add / Edit / Delete Group

By default, the group list is blank. User can create his/her own groups, edit the group name, add or remove contacts in the group, and delete a group.

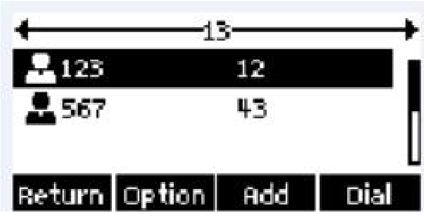
- To add a group, press [**Add Group**] button.
- To delete a group, press [**Option**] >> [**Delete**] button.
- To edit a group, press [**Option**] >> [**Edit**] button.

The Number behind the group name means the total contacts number of selected groups.

Picture 89 - Group List

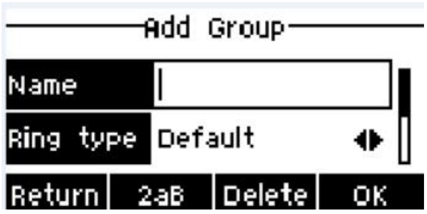
10.2.1.3 Browse and Add / Remove Contacts in Group

User can browse contacts in a group by opening the group in group list with [**OK**] button.



Picture 90 - Browsing Contacts in a Group

When user is browsing contacts of a group, user can also add contacts in that group by pressing [**Add**] button to enter the group contacts management screen, then press [**OK**] button to save the contact. The contact will also be added in local phonebook. User can delete contact from group by [**Option**] >> [**Delete**].

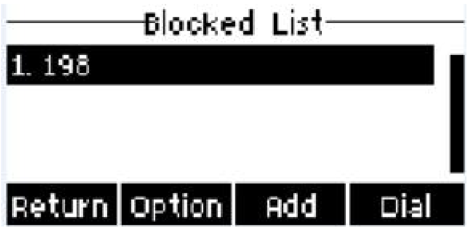


Picture 91 - Add Contacts in a Group

10.2.2 Blocked List

The device Support blocked list, such as the number added to the blocked list, the number of calls directly refused to the end, the end of the phone shows no incoming calls. (Prevent the number in the call list from being called out normally)

- There are multiple ways to add a number to Blocked List on X210 device. It can be added directly on [**Menu**] >> [**Contact**] >> [Blocked List].
- Select any number in the phone book (both local and network) for configuration addition.
- Select any number in the call log for configuration addition.



Picture 92 - Add Blocked List

- There are various ways to add number to the blocked list on web page, which can be added in the **[Phone book]** >> **[Call list]** >> **[Restricted Incoming Calls]**.
- Select any number in the phone book (both local and network) for configuration addition.
- Select any number in the call log for configuration addition.

Picture 93 - Web Blocked List

10.2.3 Cloud Phone Book

10.2.3.1 Configure Cloud Phone book

Cloud phonebook allows user to configure the device by downloading a phonebook from a cloud server. This is convenientfor office users to use the phonebook from a single source and save the effort to create and maintain the contact list individually. It is also a useful tool to synchronize his/her phonebook from a personal mobile phone to the device with Cloud Phonebook Service and App which is to be provided publicly soon.

NOTICE! The cloud phonebook is ONLY temporarily downloaded to the device each time when it is opened on the device to ensure the user get the latest phonebook. However, the downloading may take a couple seconds depending on the network condition. Therefore, it is highly recommended for the users to save important contacts from cloud to local phonebook for saving download time.

Open cloud phonebook list, press [**Menu**] >> [**PhoneBook**] >> [**Cloud Contacts**] in phonebook screen.

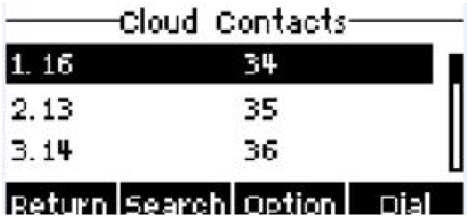
TIPS! The first configuration on cloud phone should be completed on Web page by selecting [PhoneBook] >> [Cloud Contacts]. The setting of addition/deletion on device could be done after the first setting on Web page.

Picture 94 - Cloud phone book list

10.2.3.2 Downloading Cloud Phone book

In cloud phone book screen, user can open a cloud phone book by pressing [**OK**] / [**Enter**] button. The device will start downloading the phone book. The user will be prompted with a warning message if downloading failed.

Once the cloud phone book is downloaded completely, the user can browse the contact list and dial the contact number same as in local phonebook.



Picture 95 - Downloading Cloud Phone book

10.3 Call Log

The device can store up to 1000 call log records and user can open the call logs to check all incoming, outgoing, and missed call records by pressing soft-menu button [**CallLog**] .

In the call logs screen, user may browse the call logs with up/down navigator keys.

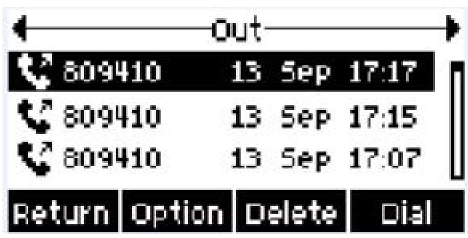
Each call log record is presented with 'call type' and 'call party number / name'. User can check further call log detail by pressing [**OK**] button and dial the number with [**Dial**] button, or add the call log number to phonebook with pressing [**Option**] >> [**Add to Contact**] .

User can delete a call log by pressing [**Delete**] button and can clear all call logs by pressing [**Delete All**] button.

Picture 96 - Call Log

Users can also filter the call records of specific call types to narrow down the scope of search records, and select a call record type by left and right navigation keys.

- Missed Call Logs
- Incoming Call Logs
- Outgoing Call Logs
- Forward Call Logs



Picture 97 - Filter call record types

10.4 Headset

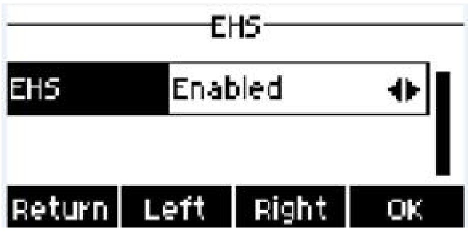
10.4.1 Wired Headset

- The device supports wired earphone with RJ9 interface, which can play incoming call sound and talk with earphone.
- After the phone is connected to the headset, the default DSS key of headset will be green light which indicating that the headset can be used normally.
- On the webpage [**Phone settings**] >> [**Features**], you can set the headset answering function, and the ring tone for headset.

Picture 98 - Headset function settings

10.4.2 EHS Headset

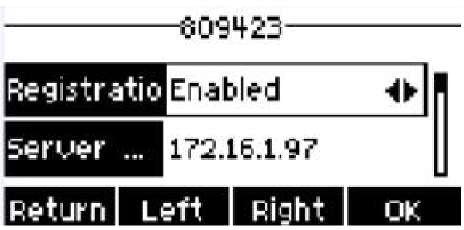
Phone into [**Menu**] >> [**Features**] >> [**Advanced**], Select [**EHS**] , can open EHS Headset (default closed EHS Headset).



Picture 99 - EHS Headset setting

10.5 Advanced

10.5.1 Line Configurations



Picture 101 - SIP address and account information

Save the adjustment by pressing [**OK**] when done.

For users who want to configure more options, user should use web management portal to modify or Advanced Settings in accounts on the individual line to configure those options.

Picture 102 - Configure Advanced Line Options

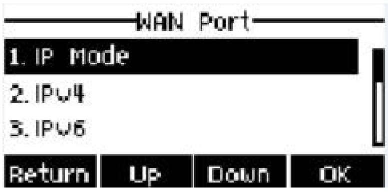
10.5.2 Network Settings

10.5.2.1 Network Settings

■ **IP Mode**

There are 3 network protocol mode options, IPv4, IPv6 and IPv4 & IPv6.

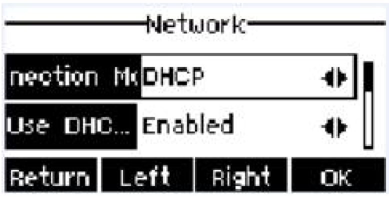
User could select available mode via “<” or “>”. The selected IP mode will be activated after pressing [OK] button.



Picture 103 - Network mode Settings

■ **IPv4**

In IPv4 mode, there are 3 connection mode options: DHCP, PPPoE and Static IP.



Picture 104 - DHCP network mode

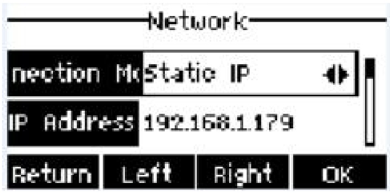
When using DHCP mode, phone will get the IP address from DHCP server (router).

- Use DHCP DNS: It is enabled as default. “Enable” means phone will get DNS address from DHCP server and “disable” means not.
- Use DHCP time: It is disabled as default. “Enable” to manage the time of get DNS address from DHCP server and “disable” means not.

Picture 105 - PPPoE network mode

When using PPPoE, phone will get the IP address from PPPoE server.

- Username: PPPoE user name.
- Password: PPPoE password.



Picture 106 - Static IP network mode

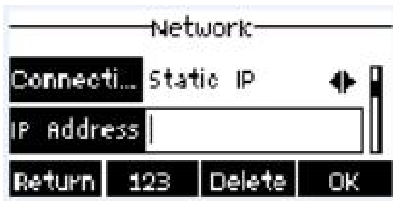
When using Static IP mode, user must configure the IP address manually.

- IP Address: Phone IP address.
- Mask: sub mask of your LAN.
- Gateway: The gateway IP address. Phone could access the other network via it.
- Primary DNS: Primary DNS address. The default is 8.8.8.8, Google DNS server address.
- Secondary DNS: Secondary DNS. When primary DNS is not available, it will work.

■ **IPv6**

In IPv6, there are 2 connection mode options, DHCP and Static IP.

- DHCP configuration refers to IPv4 introduction in last page.
- Static IP configuration is almost same as IPv4's, except the IPv6 Prefix.
- When using the phone to obtain an IPv6 address, parentheses need to be added to both accessing the web page and ping the IP, such as [fe80:: e38: 3eff: fe4f: 7daf]



Picture 107 - IPv6 Static IP network mode

10.5.2.2 QoS & VLAN

■ **LLDP**

Link Layer Discovery Protocol. LLDP is a vendor independent link layer protocol used by network devices for advertising their identity, capabilities to neighbors on a LAN segment.

Phone could use LLDP to find the VLAN switch or other VLAN devices and use LLDP learn feature to apply the VLAN ID from VLAN switch to phone its self.

■ **CDP**

Cisco Discovery Protocol. CDP is a not-for-profit charity that runs the global disclosure system for investors, companies, cities, states and regions to manage their environmental impacts. According to the CDP, Cisco devices could share the OS version, IP address, hardware version and so on.

Table 13 - QoS & VLAN

Parameters	Description
LLDP setting	
Enable LLDP	Enable LLDP
Packet Interval	LLDP requests interval time
Enable Learning Function	apply the learned VLAN ID to the phone configuration
QoS	
QoS Mode	configure Signal DSCP and audio DSCP
WAN VLAN	
WAN VLAN	WAN port VLAN configuration
LAN VLAN	
LAN VLAN	LAN port VLAN configuration
CDP	
CDP	CDP enable/disable ， CDP interval time

10.5.2.3 VPN

Virtual Private Network (VPN) is a technology to allow device to create a tunneling connection to a server and becomes part of the server's network. The network transmission of the device may be routed through the VPN server.

For some users, especially enterprise users, a VPN connection might be required to be established before activate a line registration. The device supports two VPN modes, Layer 2 Transportation Protocol (L2TP) and OpenVPN.

The VPN connection must be configured and started (or stopped) from the device web portal.

- **L2TP**

To establish a L2TP connection, users should log in to the device web portal, open webpage [**Network**] >> [**VPN**]. In VPN Mode, check the "Enable VPN" option and select "L2TP", then fill in the L2TP server address, Authentication Username, and Authentication Password in the L2TP section. Press "Apply" then the device will try to connect to the L2TP server.

When the VPN connection established, the VPN IP Address should be displayed in the VPN status. There may be some delay of the connection establishment. User may need to refresh the page to update the status. Once the VPN is configured, the device will try to connect with the VPN automatically when the device boots up every time until user disable it. Sometimes, if the VPN connection does not establish immediately, user may try to reboot the device and check if VPN connection established after reboot.

- **OpenVPN**

To establish an OpenVPN connection, user should get the following authentication and configuration files from the OpenVPN hosting provider and name them as the following,

OpenVPN Configuration file: client.ovpn

CA Root Certification: ca.crt

Client Certification: client.crt

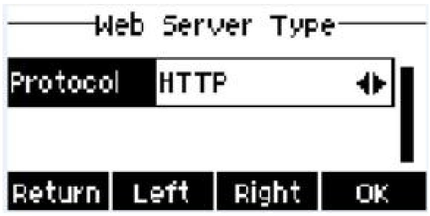
Client Key: client.key

User then upload these files to the device in the web page [**Network**] >> [**VPN**], select OpenVPN Files. Then user should check “Enable VPN” and select “OpenVPN” in VPN Mode and click “Apply” to enable OpenVPN connection.

Same as L2TP connection, the connection will be established every time when system rebooted until user disable it manually.

10.5.2.4 Web Server Type

Configure the Web Server mode to be HTTP or HTTPS and will be activated after the reboot. Then user could use http/https protocol to access pone web page.



Picture 108 - The phone configures the web server type

10.5.3 Set The Secret Key

When the device is in the default standby mode,

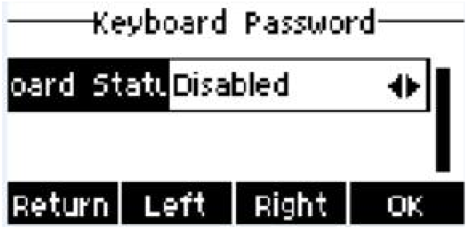
- Select [**Menu**] >> [**Advanced**]>>[**Security**], and enter it via [Confirm] or [OK] button.
- As default, the Advance setting password is 123.
- User will see the follow page after menu – Advanced setting – Security.

Picture 109 - Keypad lock password

Menu password is the permission for accessing the advanced setting.

- [**Current password**] is the password user configured before. If no configuration before, the default password is 123.
- [**New password**] is the new password user to use.
- After configuring the menu password, it will work immediately.

Keyboard password is used to unlock the phone once it’s locked.

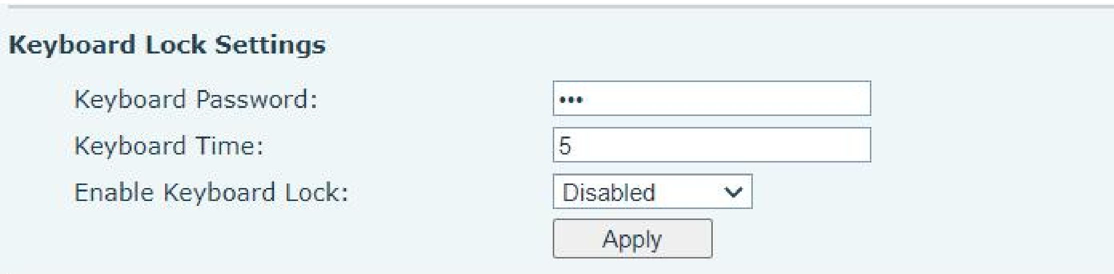


Picture 110 - Set keyboard lock password

User could only set to enable or disable the keyboard password in LCD screen.

- Enter [Keyboard password] setting by pressing [confirm] or [OK] button after password entered. If no menu password configuration before, it is 123 as default.
- If the menu password is correct, phone will go to keyboard password interface. As default, the keyboard password is disabled. When it is enabled, the keyboard will be locked after timeout.
- If the timeout time is not set (default is 0), you need to long press [#] on the standby interface to lock the keyboard. There will be a lock icon at the top of the phone, and pressing any button will prompt you to enter the password.

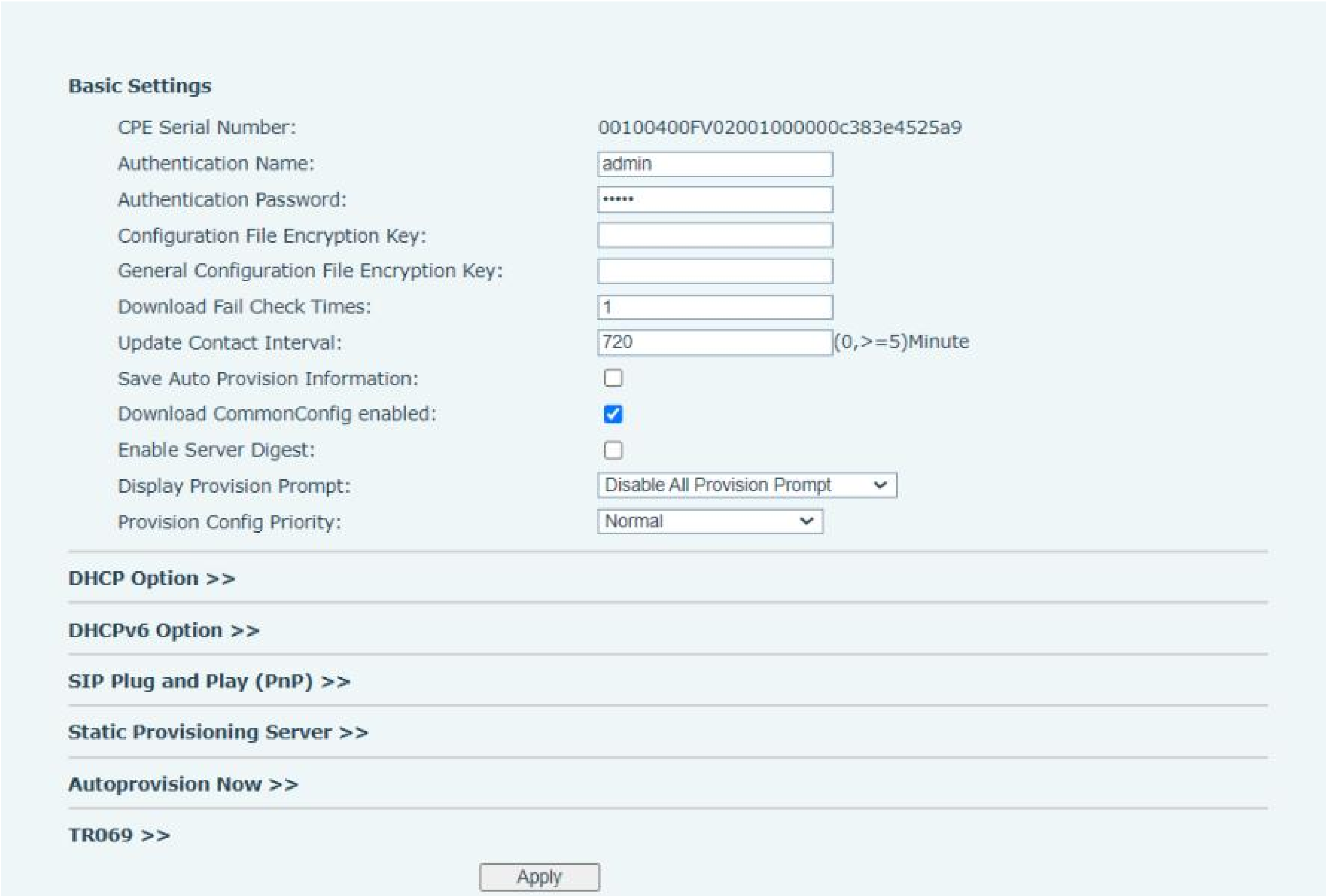
Picture 111 - Phone keypad lock password input interface



Picture 112 - Web keyboard lock password Settings

10.5.4 Maintenance

Phone Webpage: Login and go to [**System**] >> [**Auto provision**].



Picture 113 - Page auto provision Settings

LCD: [**Menu**] >> [**Advanced**] >> [**Maintenance**] >> [**Auto Provision**].

Picture 114 - Phone auto provision settings

The devices support SIP PnP, DHCP options, Static provision, TR069. If all of the 4 methods are enabled, the priority from high to low as below:

PNP>DHCP>TR069> Static Provisioning

Transferring protocol: FTP, TFTP, HTTP, HTTPS

Table 14 - Auto Provision

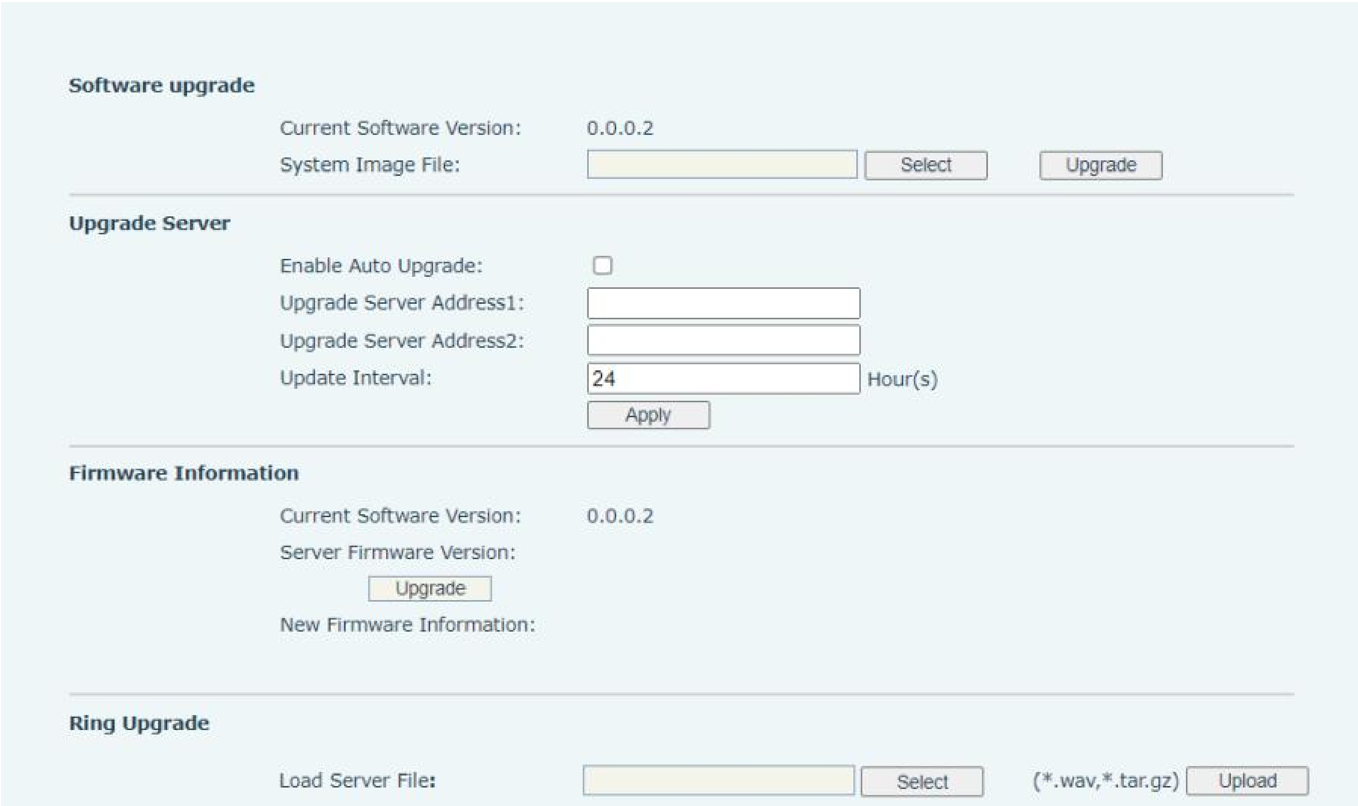
Parameters	**Description**
Basic settings	
CPE Serial Number	Display the device SN
Authentication Name	The user name of provision server

Authentication Password	The password of provision server
Configuration File Encryption Key	If the device configuration file is encrypted , user should add the encryption key here
General Configuration File Encryption Key	If the common configuration file is encrypted, user should add the encryption key here
Download Fail Check Times	If there download is failed, phone will retry with the configured times.
Update Contact Interval	Phone will update the phonebook with the configured interval time. If it is 0, the feature is disabled.
Save Auto Provision Information	Save the HTTP/HTTPS/FTP user name and password. If the provision URL is kept, the information will be kept.
Download Common Config enabled	Whether phone will download the common configuration file.
Enable Server Digest	When the feature is enable, if the configuration of server is changed, phone will download and update.
Display Provision Prompt	Configure if the phone display the provision prompt.
Provision Config Priority	During auto provision, the configuration file preferentially uses the local configuration of the phone or the configuration obtained by the server
DHCP Option	
Option Value	Confiugre DHCP option, DHCP option supports DHCP custom option | DHCP option 66 | DHCP option 43, 3 methods to get the provision URL. The default is Option 66.
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be same as server define.
Enable DHCP Option 120	Use Option120 to get the SIP server address from DHCP server.
DHCPv6 Option	
Option Value	Confiugre DHCPv6 option
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be same as server define.
SIP Plug and Play (PnP)	
Enable SIP PnP	Whether enable PnP or not. If PnP is enable, phone will send a SIP SUBSCRIBE message with broadcast method. Any server can support the feature will respond and send a Notify with URL to phone. Phone could get the configuration file with the URL.
Server Address	Broadcast address. As default, it is 224.0.0.0.
Server Port	PnP port
Transport Protocol	PnP protocol, TCP or UDP.
Update Interval	PnP message interval.

Static Provisioning Server	
Server Address	Provisioning server address. Support both IP address and domain address.
Configuration File Name	The configuration file name. If it is empty, phone will request the common file and device file which is named as its MAC address. The file name could be a common name, $mac.cfg, $input.cfg. The file format supports CFG/TXT/XML.
Protocol Type	Transferring protocol type ， supports FTP、TFTP、HTTP and HTTPS
Update Interval	Configuration file update interval time. As default it is 1, means phone will check the update every 1 hour.
Update Mode	Provision Mode. 1. Disabled. 2. Update after reboot. 3. Update after interval.
Autoprovision Now	**Configure the above three upgrade methods, click Autoprovision Now, and it will take effect immediately, without restart.**
TR069	
Enable TR069	Enable TR069 after selection
ACS Server Type	There are 2 options Serve type, common and CTC.
ACS Server URL	ACS server address
ACS User	ACS server username (up to is 59 character)
ACS Password	ACS server password (up to is 59 character)
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.
TLS Version	TLS version (TLS 1.0, TLS 1.1, TLS 1.2,TLS1.3)
INFORM Sending Period	INFORM signal interval time. It ranges from 1s to 999999s
STUN Server Address	Configure STUN server address
STUN Enable	To enable STUN server for TR069
Option Value	Confiugre option value or disable it.
DHCP Option ACS	Custom Option value is allowed from 128 to 254.

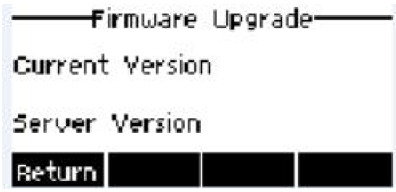
10.5.5 Firmware Upgrade

- Web page: Login phone web page, go to [**System**] >> [**Upgrade**].



Picture 115 - Web page firmware upgrade

- LCD interface: go to [**Menu**] >> [**Advanced**] >> [**Firmware Upgrade**] .



Picture 116 - Firmware upgrade information display

Table 15 - Firmware upgrade

Parameter	**Description**
Upgrade server	
Enable Auto Upgrade	Enable automatic upgrade, If there is a new version txt and new software firmware on the server, phone will show a prompt upgrade message after Update Interval.
Upgrade Server Address1	Set available upgrade server address.
Upgrade Server Address2	Set available upgrade server address.
Update Interval	Set Update Interval.
Firmware Information	
Current Software Version	It will show Current Software Version.
Server Firmware Version	It will show Server Firmware Version.

[Upgrade] button	If there is a new version txt and new software firmware on the server, the page will display version information and upgrade button will become available; Click [Upgrade] button to upgrade the new firmware.
New Firmware Information	When there is a corresponding TXT file and version on the server side, the TXT and version information will be displayed under the new version description information.

- The file requested from the server is a TXT file called vendor_model_hw10.txt.Hw followed by the hardware version number, it will be written as hw10 if no difference on hardware. All Spaces in the filename are replaced by underline.
- The URL requested by the phone is HTTP:// server address/vendor_Model_hw10 .txt: The new version and the requested file should be placed in the download directory of the HTTP server

- TXT file format must be UTF-8
- vendor_model_hw10.TXT　The file format is as follows：
  Version=0.0.0.2 #Firmware
  Firmware=xxx/xxx.z　#URL，Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers.
  BuildTime=2025.01.11 20:00
  Info=TXT|XML
  Xxxxx
  Xxxxx
  Xxxxx
  Xxxxx
- After the interval of update cycle arrives, if the server has available files and versions, the phone prompts software upgrade, click View to view version information and upgrade.

10.5.6 Factory Reset

The phone is in default standby mode.

- Press [**Menu**] to find [**Advanced Settings**], and press [**OK**].
- Press [**Advanced Settings**] to enter the password (default password is 123) to enter the interface.
- Press the [**Factory Reset**] button to select the file to be cleared.

2）In standby, press and hold the　[**OK**]　button for 6S to perform the reset operation

11 Web Configurations

11.1 Web Page Authentication

The user can log into the web page of the phone to manage the user's phone information and operate the phone. Users must provide the correct user name and password to log in.

11.2 System >> Information

User can get the system information of the device in this page including,

- Model
- Hardware Version
- Software Version
- Uboot
- WAN Speed
- SN
- MEMinfo
- System time

And summarization of network status,

- Network Mode
- Ethernet MAC
- Ethernet IP
- Subnet Mask
- Default Gateway

And information about VQ status,

- Start time
- Stop time
- Local user
- Remote user
- Local IP
- Remote IP
- Local Port
- Remote port
- Local codec
- Remote codec
- Jitter
- JitterBufferMax
- Packets lost
- NetworkPacketLossRate

- MOS-LQ
- MOS-CQ
- RoundTripDelay
- EndSystemDelay
- SymmOneWayDelay

JitterBufferRat

Besides, summarization of SIP account status,

- SIP User
- SIP account status (Registered / Unapplied / Trying / Timeout/Registration failed)

11.3 System >> Account

On this page the user can change the password for the login page.

Users with administrator rights can also add or delete users, manage users, and set permissions and passwords for new users.

11.4 System >> Configurations

On this page, users with administrator privileges can view, export, or import the phone configuration, or restore the phone to factory Settings.

- **Clear Configuration**

Select the module in the configuration file to clear. (Note: All basic configurations are cleared on a fixed basis; You can choose whether or not to keep/purge the following)

SIP: account configuration.

AUTOPROVISION: automatically upgrades the configuration

TR069:TR069 related configuration

MWI: MWI module, including authentication user information, web access protocol, etc.

DSS Key: DSS Key configuration

BASIC NETWORK: This includes the basic configuration of network

- **Clear Userdata**

Select the local data table to be cleared, all selected by default.

- **Clear ETC**

Select the ETC file to be cleared, all selected by default.

- **Reset Phone**

The phone data will be cleared, including configuration and database tables.

11.5 System >> Upgrade

Upgrade the phone software version, customized ringtone, background, DSS Key icon, etc., can also be upgraded to delete the file.Select the upgrade file in the corresponding location and click Upgrade. Ringtone uploads support .wav formats, and package uploads in .tar.gz formats. (The format supported by the upgrade file can be viewed at the corresponding location on the webpage)

11.6 System >> Auto Provision

The Auto Provision settings help IT manager or service provider to easily deploy and manage the devices in mass volume. For the detail of Auto Provision, please refer to this link Auto Provision Description。

11.7 System >> Tools

Tools provided in this page help users to identify issues at trouble shooting. Please refer to 13 Trouble Shooting for more detail.

11.8 System >> Reboot Phone

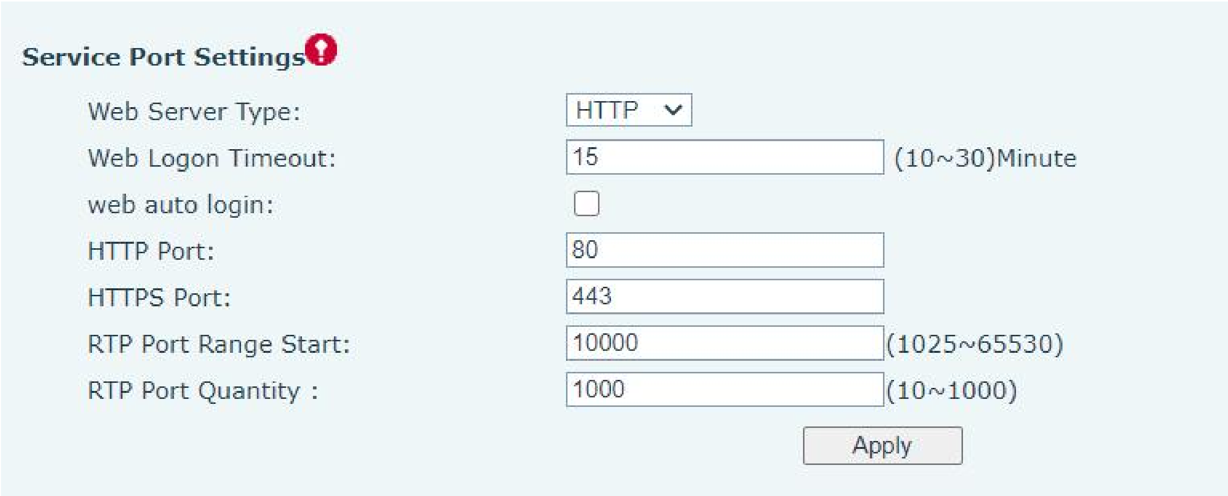
This page can restart the phone.

12 Network >> Basic

This page allows users to configure network connection types and parameters.

12.1 Network >> Service Port

This page provides settings for Web page login protocol, protocol port settings and RTP port.



Picture 119 - Service Port Settings

Table 16 - Service port

Parameters	**Description**
Web Server Type	Reboot to take effect after settings. Optionally, the web page login is HTTP/HTTPS.
Web Logon Timeout	Default as 15 minutes, the timeout will automatically exit the login page, need to login again.
Web auto login	After the timeout does not need to enter a user name password, will automatically login to the web page.
HTTP Port	The default is 80. If you want system security, you can set ports other than 80. Such as :8080, webpage login: HTTP://ip:8080
HTTPS Port	The default is 443, the same as the HTTP port.
RTP Port Range Start	The value range is 1025 - 65530. The value of RTP port starts from the initial value set. For each call, the value of voice and video port is added 2.
RTP Port Quantity	Number of calls.

12.2 Network >> VPN

Users can configure a VPN connection on this page. See 10.6.2.3 VPN for more details.

- **Link Layer Discovery Protocol (LLDP) Settings**
  - Enable LLDP: Select whether to enable LLDP
  - Packet Interval: Set the packet period, the range is 1~3600 seconds
  - Enable Learning Function: Select whether to enable learning function
- **Cisco Discovery Protocol(CDP)**
  - Enable CDP: Select whether to enable CDP
  - Packet interval: Set the packet delivery interval in the range of 1~3600 seconds
- **DHCP VLAN settings**
  - Select a parameter value: Select a custom parameter or disable the DHCP VLAN
  - Option Value Data Type: Select the Option value data type
  - DHCP Option VLAN (128-254): Set DHCP VLAN parameters
- **QoS setting**
  - Enable DSCP: Select whether to enable DSCP
  - Signal DSCP: Set DSCP signal, the range is 0~63
  - Audio DSCP: Set DSCP audio, the range is 0~63
- **ARP cache Life**
  - Set the ARP cache period
- **WAN VLAN settings**
  - Enable VLAN: Select whether to enable VLAN
  - WAN VLAN ID: Set the WAN VLAN ID, the range is 0~4095
  - 802.1p signal priority: Select 802.1p signal priority, 0~7
- **LAN VLAN settings**
  - LAN VLAN Mode: Select LAN VLAN mode, or disable LAN VLAN
  - LAN VLAN ID: Set the LAN vlan id, the range is 0~4095
  - Virtual LAN Priority: Select the virtual LAN priority, 0~7
- **802.1x setting**
  - 802.1x mode: Select 802.1x authentication mode, or disable authentication
  - Identity: Set the authentication username
  - Password: Set an authentication password
  - CA Certificate: Upload the CA certificate
  - Device Certificate: Upload the device certificate
- **Certificate documents**
  - You can upload an HTTPS certificate file

12.3 Network >> Advanced

Advanced network Settings are typically configured by the IT administrator to improve the quality of the phone service. For configuration, query the 10.6 advanced Settings.

12.4 Line >> SIP

Configure the Line service configuration on this page.

Table 17 - Line configuration on the web page

Parameters	**Description**
Register Settings	
Line Status	Display the current line status at page loading. To get the up to date line status, user has to refresh the page manually.
Activate	Whether the service of the line is activated
Username	Enter the username of the service account.
Authentication User	Enter the authentication user of the service account
Display Name	Enter the display name to be sent in a call request.
Authentication Password	Enter the authentication password of the service account
Realm	Enter the SIP domain if requested by the service provider
Server Name	Input server name.
SIP Server 1	
Server Address	Enter the IP or FQDN address of the SIP server
Server Port	Enter the SIP server port, default is 5060
Transport Protocol	Set up the SIP transport line using TCP or UDP or TLS.
Registration Expiration	Set SIP expiration date.
SIP Server 2	
Server Address	Enter the IP or FQDN address of the SIP server
Server Port	Enter the SIP server port, default is 5060
Transport Protocol	Set up the SIP transport line using TCP or UDP or TLS.
Registration Expiration	Set SIP expiration date.
SIP Proxy Server Address	Enter the IP or FQDN address of the SIP proxy server.
Proxy Server Port	Enter the SIP proxy server port, default is 5060.
Proxy User	Enter the SIP proxy user.
Proxy Password	Enter the SIP proxy password.
Backup Proxy Server Address	Enter the IP or FQDN address of the backup proxy server.
Backup Proxy Server Port	Enter the backup proxy server port, default is 5060.
Basic Settings	

Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the delay timeAuto Answering Delay
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
Call Forward Unconditional	Enable unconditional call forward, all incoming calls will be forwarded to the number specified in the next field
Call Forward Number for Unconditional	Set the number of unconditional call forward
Call Forward on Busy	Enable call forward on busy, when the phone is busy, any incoming call will be forwarded to the number specified in the next field.
Call Forward Number for Busy	Set the number of call forward on busy .
Call Forward on No Answer	Enable call forward on no answer, when an incoming call is not answered within the configured Call Forward Delay time for No Answerthe call will be forwarded to the number specified in the next field.
Call Forward Number for No Answer	Set the number of call forward on no answer.
Call Forward Delay for No Answer	Set the delay time of not answered call before being forwarded.
Transfer Timeout	Set the timeout of call transfer process.
Conference Type	Set the type of call conference, Local=set up call conference by the device itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server
Server Conference Number	Set the conference room number when conference type is set to be Server
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Voice Message Number	Set the number for retrieving voice message
Voice Message Subscribe Period	Set the interval of voice message notification subscription
Enable Hotline	Enable hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or turn on hands-free speaker or headphone
Hotline Delay	Set the delay for hotline before the system automatically dialed it
Hotline Number	Set the hotline dialing number
Dial Without Registered	Set whether to call out by proxy without registration
Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.

DTMF Type	Set the DTMF type to be used for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Use VPN	Set the line to use VPN restrict route
Use STUN	Set the line to use STUN for NAT traversal
Enable Failback	Whether to switch to the primary server when it is available.
Failback Interval	A Register message is used to periodically detect the time interval for the availability of the main Proxy.
Signal Failback	Multiple proxy cases, whether to allow the invite/register request to also execute failback.
Signal Retry Counts	The number of attempts that the SIP Request considers proxy unavailable under multiple proxy scenarios.
Codecs Settings	Set the priority and availability of the codecs by adding or remove them from the list.
Advanced Settings	
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field. (Note: The status of the function needs to be changed before the feature code is dialed to the server.)
Enable DND	Set the feature code to dial to the server
Disable DND	Set the feature code to dial to the server
Enable Call Forward Unconditional	Set the feature code to dial to the server
Disable Call Forward Unconditional	Set the feature code to dial to the server
Enable Call Forward on Busy	Set the feature code to dial to the server
Disable Call Forward on Busy	Set the feature code to dial to the server
Enable Call Forward on No Answer	Set the feature code to dial to the server
Disable Call Forward on	Set the feature code to dial to the server

No Answer	
Enable Blocking Anonymous Call	Set the feature code to dial to the server
Disable Blocking Anonymous Call	Set the feature code to dial to the server
Call Waiting On Code	Set the feature code to dial to the server
Call Waiting Off Code	Set the feature code to dial to the server
Send Anonymous On Code	Set the feature code to dial to the server
Send Anonymous Off Code	Set the feature code to dial to the server
SIP Encryption	Enable SIP encryption such that SIP transmission will be encrypted
RTP Encryption	Enable RTP encryption such that RTP transmission will be encrypted
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if there is not new session timer event update received after the timeout period
Session Timeout	Set the session timer timeout period
Enable BLF List	Enable/Disable BLF List
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Response Single Codec	If setting enabled, the device will use single codec in response to an incoming call request
BLF Server	The registered server will receive the subscription package from ordinary application of BLF phone. Please enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened
Keep Alive Interval	Set the keep alive packet transmitting interval
Keep Authentication	Keep the authentication parameters from previous authentication
Blocking Anonymous Call	Reject any incoming call without presenting caller ID
User Agent	Set the user agent, the default is Model with Software Version.
Specific Server Type	Set the line to collaborate with specific server type
SIP Version	Set the SIP version
Anonymous Call Standard	Set the standard to be used for anonymous
Local Port	Set the local port
Ring Type	Set the ring tone type for the line
Enable user=phone	Sets user=phone in SIP messages.

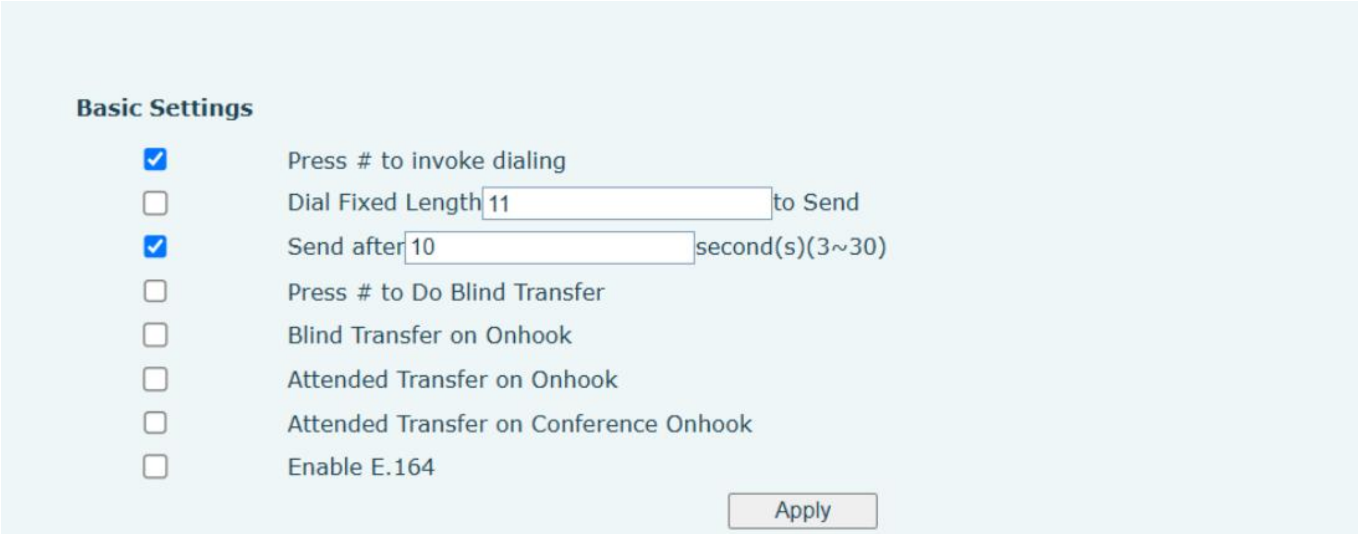
Use Tel Call	Set use tel call
Auto TCP	Using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes
Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
DNS Mode	Select DNS mode, A, SRV, NAPTR
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server, it will use the source IP address, not the address in via field.
Convert URI	Convert not digit and alphabet characters to %hh hex code
Use Quote in Display Name	Whether to add quote in display name, i.e. “VoIP” vs VoIP
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
Sync Clock Time	Time Sync with server
Enable Inactive Hold	With the post-call hold capture package enabled, you can see that in the INVITE package, SDP is inactive.
Caller ID Header	Set the Caller ID Header
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Enable Feature Sync	Feature Sync with server
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
CallPark Number	Set the CallPark number.
Server Expire	Set the timeout to use the server.
TLS Version	Choose TLS Version.
uaCSTA Number	Set uaCSTA Number.
Enable Click To Talk	With the use of special server, click to call out directly after enabling.
Enable Changeport	Set whether to enable changeport
VQ Name	Set the VQ name
VQ Server	Set the VQ server address
VQ Server Port	Set the VQ server port
VQ Http/Https Server	Set the VQ Http/Https server
Flash mode	Chose Flash mode, normal or SIP info.
Flash mode	Chose Flash mode，normal or SIP info.
Flash Info Content-Type	Set the SIP info content type.
Flash Info Content-Body	Set the SIP info content body.
PickUp Number	Set the scramble number when the Pickup is enabled.
JoinCall Number	Set JoinCall Number.

Intercom Number	Set Intercom Number.
Unregister On Boot	Whether to enable logout function.
Enable MAC Header	Whether to open the registration of SIP package with user agent with MAC or not.
Enable Deal 180	Set whether the phone rings when receiving a 180 SIP message
Transaction Timer T1	Configure the transaction timer time T1, the range of T1 value is from 500 to 10000 milliseconds.
Transaction Timer T2	Configure the transaction timer time T2, the range of T2 value is from 2000-40000 milliseconds.
Transaction Timer T4	Configure the transaction timer time T4, the range of T4 value is from 2500 to 60000 milliseconds.
CallPark Number	Set up supported callpark numbers by the server
PickUp Number	Set up the pick up number
JoinCall Number	Set up joincall numbers to join meetings
Retrieve Number	Set up the retrieve number to input when trying to retrieve the parked call
Enable Register MAC Header	Whether to open the registration is user agent with MAC or not.
BLF Dialog Strict Match	Whether to enable accurate matching of BLF sessions.
PTime(ms)	Set whether to bring ptime field, default no.
SIP Global Settings	
Strict Branch	Set up to strictly match the Branch field.
Enable Group	Set open group.
Enable RFC4475	Set to enable RFC4475.
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Set the registration failure retry time.
Local SIP Port	Modify the phone SIP port.
Enable uaCSTA	Enable uaCSTA function

12.5 Line >> SIP Hotspot

Please refer to 9.9 SIP Hotspot.

12.6 Line >> Dial Plan

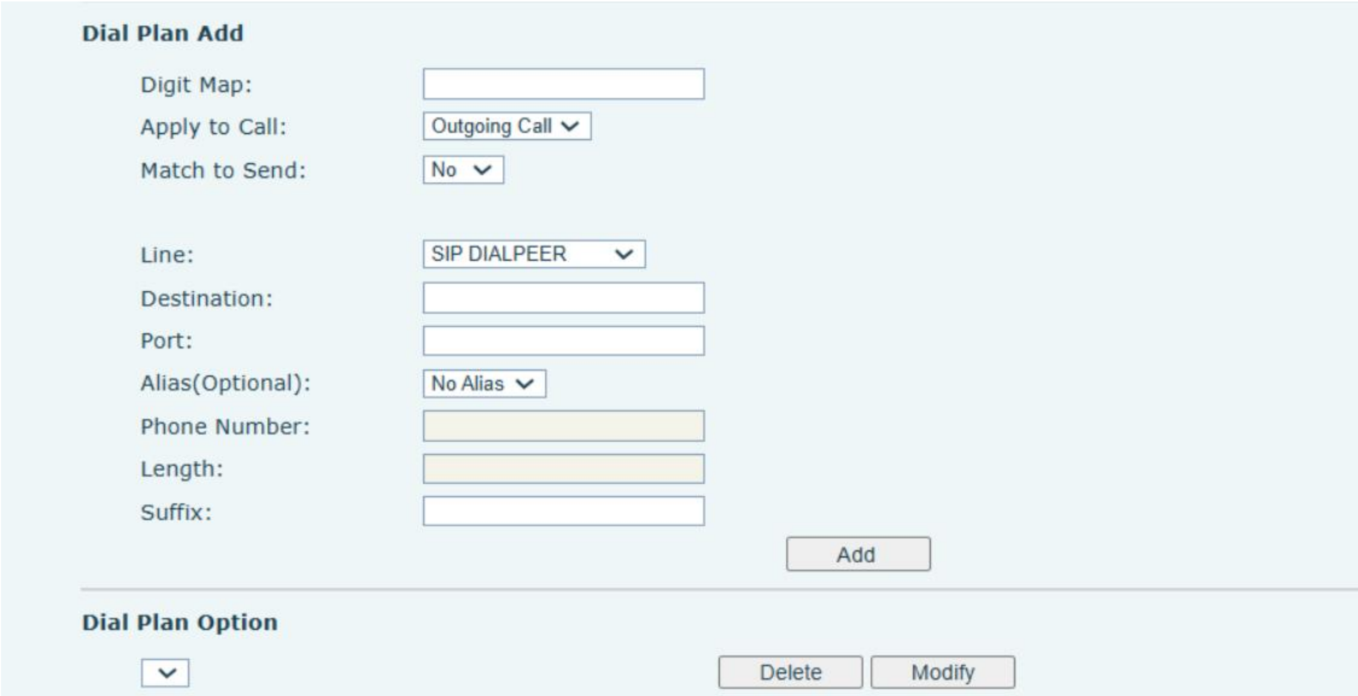


Picture 120 - Dial plan settings

Table 18 - Phone 7 dialing methods

Parameters	Description
Press # to invoke dialing	The user dials the other party's number and then adds the # number to dial out;
Dial Fixed Length	The number entered by the user is automatically dialed out when it reaches a fixed length
Timeout dial	The system dials automatically after timeout
Press # to Do Blind Transfer	The user enters the number to be transferred and then presses the "#" key to transfer the current call to a third party
Blind Transfer on Onhook	After the user enters the number, hang up the handle or turn off the hands-free function to transfer the current call to a third party.
Attended Transfer on Onhook	Hang up the handle or press the hands-free button to realize the function of transfer, which can transfer the current call to a third party.
Attended Transfer on Conference Onhook	During a three-way call, hang up the handle and the remaining two parties remain on the call.
Enable E.164	Please refer to e. 164 standard specification

Add dialing rules:



Picture 121 - Custom setting of dial - up rules

Table 19 - Dial - up rule configuration table

Parameters	Description
Dial rule	There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules. In prefix matching, only part of the number is entered followed by T. The mapping with then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.
Note: Two different special characters are used. ■ x -- Matches any single digit that is dialed. ■ [ ] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.	
Apply to Call	Configure dialing rule application scenarios: outbound, inbound, or both.
Match to Send	Enable precise matching.
Destination	Set Destination address. This is for IP direct.
Port	Set the Signal port, and the default is 5060 for SIP.
Alias(Optional)	Set the Alias. This is the text to be added, replaced or deleted. It is an optional item.
Note: There are four types of aliases.	

■ all: xxx – xxx will replace the phone number. ■ add: xxx – xxx will be dialed before any phone number. ■ del –The characters will be deleted from the phone number. ■ rep: xxx – xxx will be substituted for the specified characters.	
Phone Number	Configure dialing rule telephone number aliases.
Suffix	Characters to be added at the end of the phone number. It is an optional item.
Length	Set the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. It is an optional item.

This feature allows the user to create rules to make dialing easier. There are several different options for dial rules. The examples below will show how this can be used.

Example 1: All Substitution -- Assume that it is desired to place a direct IP call to IP address 172.168.2.208. Using this feature, 123 can be substituted for 172.168.2.208.

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media
1	"123"	Out	No	SIP DIALPEER(172.16.1.15:5560)			Default

Picture 122 - Dial rules table (1)

Example 2: Partial Substitution -- To dial a long distance call to Beijing requires dialing area code 010 before the local phone number. Using this feature 1 can be substituted for 010. For example, to call 62213123 would only require dialing 162213123 instead of 01062213123.

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix
1	"1T"	Out	No	AUTO		

Picture 123 - Dial rules table (2)

Example 3: Addition -- Two examples are shown. In the first case, it is assumed that 0 must be dialed before any 11 digit number beginning with 13. In the second case, it is assumed that 0 must be dialed before any 11 digit number beginning with 135, 136, 137, 138, or 139. Two different special characters are used.

x -- Matches any single digit that is dialed.

[] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.

12.7 Line >> Basic Settings

Set up the register global configuration.

Table 20 - Set the line global configuration on the web page

Parameters	**Description**
STUN Settings	

Server Address	Set the STUN server address
Server Port	Set the STUN server port, default is 3478
Binding Period	Set the STUN binding period which can be used to keep the NAT pinhole opened.
STUN effective time	Set the STUN binding cycle to ensure NAT traversal is enabled.
SIP Waiting Time	Set the timeout of STUN binding before sending SIP messages
SIP P2P Settings	
Enable automatic answering	Enable or disable the automatic answering function for IP calls
Automatic answer waiting time	Configure the automatic response time for IP calls.
DTMF type	Set DTMF type.
DTMF SIP INFO mode	When the DTMF type is SIP Info, the DTMF value sent when entering DTMF "*" and "#" or "10/11".
Using VPN	Configure OpenVPN functionality.
Call ID format	The format of the Call ID field for IP calls.
Prohibit sending RTP configuration items in ringback tones	Prohibit sending RTP configuration items in ringback tones

12.8 Line >> Action Plan

Action Plan application: a technical implementation defined and designed by for remote control and behavior linkage between terminal equipment and other equipment. That is, when an event occurs on the terminal, the terminal can perform an action, and this action is completed according to a Plan rule.

Log in to the phone web page, access [**Line**] -> [**Action Plan**], and configure the linkage plan rules.

Table 21 - Action Plan

Action	
Description	Actions triggered by the rules of number configuration.
Options	Default: when the rule is triggered, the phone displays video or converts multicast according to the RTSP URL or multicast address port set by the website. MCAST-XFER: when the rule is triggered, the phone converts the incoming call or multicast into multicast and sends it to the set multicast address port. Record: the phone automatically turns on the recording function when the rule is triggered.

	Mute: the phone will mute automatically when the rule is triggered. Answer: when the rule is triggered, the phone automatically answers the incoming call.
Default	Default
Number	
Description	The calling number corresponding to each linkage plan; supports number expressions identical to the receive number rules
Options	123; 1xx; 1.; 1[3,5,7,8]xxxxxxxxx; 5753[5-6]xxxx X represents a match for any single digit; . represents a match for any single digit; [] represents matching rules for a specific digit.
Default	None
Type	
Description	Types of time periods for rule triggering execution
Options	Connected: Triggered and executed after call establishment.
Direction	
Description	The corresponding behavior handling for the configured rule
Options	Both: Triggered for both incoming and outgoing calls simultaneously; Outgoing Call: Triggered for outgoing calls only; Incoming Call: Triggered for incoming calls only.
Default	Both
Line	
Description	The selected rule corresponds to the matched SIP line
Options	Auto,SIP1~SIP4
Default	Auto
MCAST Codec	
Description	The multicast encoding sent when the multicast conversion rule is triggered
Options	PCMU,PCMA,G726-16,G726-24,G726-32,G726-40,G729,G723,iLBC,opus,G722 Note: The supported encodings may vary depending on the model, and the phone encoding used in practice should prevail.”
Default	PCMU
URL	
Description	When triggered by the default and conversion multicast rules, execute the URL.
Options	It supports HTTP/HTTPS/RTSP and multicast address and port. 1. When set to Default, it supports sending Action URL, as well as configuring the RTSP video stream, multicast address, and port. When triggered by the conversion multicast rule, it supports configuring the multicast address and port. The configuration formats are:

	For Default action selection, the configuration format is: "mcast://multicast address:port". For Conversion Multicast action selection, the configuration format is: "mcast://multicast address:port"."
Default	None

12.9 Line >> Basic Settings

Set up the register global configuration.

Table 22 - Set the line global configuration on the web page

Parameters	Description
STUN Settings	
STUN NAT Traversal	Display whether STUN penetration is successful.
Server Address	Set the STUN server address
Server Port	Set the STUN server port, default is 3478
Binding Period	Set the STUN binding period which can be used to keep the NAT pinhole opened.
SIP Waiting Time	Set the timeout of STUN binding before sending SIP messages
SIP P2P Settings	
Enable Auto Answering	Turn on Auto Answering.
Auto Answering Delay	After the set time has elapsed, the device automatically answers the incoming call.
DTMF Type	Set the DTMF type.
DTMF SIP INFO Mode	Set the expression of #/* when SIP INFO is used as DTMF Send Type.
Use VPN	Turn on VPN.
Call-ID Format	Set the format of the SIP message Call-ID field, which is $id@$ip by default

12.10 Line >> RTCP-XR

RTCP-XR mode is based on RFC3611 (RTP Control Extended Report), which can measure and evaluate network packet loss, delay and voice quality by sending RTCP-XR packets.

Table 23 - VQ RTCP-XR Settings

Parameters	Description
VQ RTCP-XR Settings	
VQ RTCP-XR Session Report	VQ report on whether session mode is enabled or not.
VQ RTCP-XR Interval Report	Whether to turn on Interval mode for VQ report sending.

Period for Interval Report(5~99)	The time interval at which VQ reports are sent periodically.
Warning threshold for Moslq(15~40)	When the phone calculated the Moslq value x10 below the set threshold, a warning was issued.
Critical threshold for Moslq(15~40)	When the phone calculates the Moslq value x10 below the set threshold, the critical report is issued.
Warning Threshold for Delay(10~2000)	When the one-way delay of the phone is greater than the set threshold, warning is issued.
Critical Threshold for Delay(10~2000)	When the phone computes that the one-way delay is greater than the set threshold, the critical report is issued.
Display Report Options on Phone	Whether to display the VQ report data of the last call on the phone
Display Report Options on web	Whether to display the VQ report data for the last call through the web page.
Display Report Options on Phone	Choose whether the following options should be enabled

12.11 Phone settings >> Features

Configuration phone features.

Table 24 - General function Settings

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
Enable Call Transfer	Enable Call Transfer.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it
Enable Auto Onhook	The phone will hang up and return to the idle automatically at hands-free mode
Auto HangUp Delay	Specify Auto Onhook time, the phone will hang up and return to the idle automatically after Auto Hand down time at hands-free mode, and play dial tone Auto Onhook time at handset mode
Ring From Headset	Enable Ring for Handset by selecting it, the phone plays ring tone from handset.
Enable Auto Headset	Enable this feature, headset plugged in the phone, user press ‘answer’ key or line key to answer a call with the headset automatically.
Enable Silent Mode	When enabled, the phone is muted, there is no ringing when calls, you can use the volume keys and mute key to unmute.
Disable Mute for Ring	When it is enabled, you can’t mute the phone

Enable Default Line	If enabled, user can assign default SIP line for dialing out rather than SIP1.
Enable Auto Switch Line	Enable phone to select an available SIP line as default automatically
Default Ext Line	Select the default line to use for outgoing calls
Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out any number.
Hide DTMF	Configure the hide DTMF mode.
Enable DTMF/Transfer	Set the DTMF value for the server upon receiving transfer operation
Enable DTMF/Hold	Set the DTMF value for the server upon receiving hold operation
Enable DTMF/Conference	Set the DTMF value for the server upon receiving conference operation
Enable CallLog	Select whether to save the call log.
Enable Restricted Incoming List	Whether to enable restricted call list.
Enable Allowed Incoming List	Whether to enable the allowed call list.
Enable Restricted Outgoing List	Whether to enable the restricted allocation list.
Enable Country Code	Whether the country code is enabled.
Country Code	Fill in the country code.
Area Code	Fill in the area code.
Enable Number Privacy	Whether to enable number privacy.
Match Direction	Matching direction, there are two kinds of rules from right to left and from left to right.
Start Position	Open number privacy after the start of the hidden location.
Hide Digits	Turn on number privacy to hide the number of digits.
Allow IP Call	If enabled, user can dial out with IP address
P2P IP Prefix	Prefix a point-to-point IP call.
Caller Name Priority	Change caller ID display priority.
Emergency Call Number	Set Emergency Call Number
Search path	Select the search path.
LDAP Search	Select from with one LDAP for search
Emergency Call Number	Configure the Emergency Call Number. Despite the keyboard is locked, you can dial the emergency call number
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP address.
Push XML Server	Configure the Push XML Server, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.

Enable Pre-Dial	Disable this feature, user enter number will open audio channel automatically. Enable the feature, user enter the number without opening audio channel.
Enable Multi Line	If enabled, up to 10 simultaneous calls can exist on the phone, and if disabled, up to 2 simultaneous calls can exist on the phone.
Line Display Format	Custom line format: SIPn/SIPn: xxx/xxx@SIPn
Contact As White List Type	NONE/BOTH/DND White List/FWD White List
Block XML When Call	Disable XML push on call.
SIP notify	When enabled, the phone displays the information when it receives the relevant notify content.
Call Number Filter	Set the characters filtered by the phone when dialing; if the dialed number contains configured characters, the phone will automatically filter these characters when dialing.
Auto Resume Current	When the configuration is enabled, the phone will automatically restore the current call.
Call Timeout	Set the call timeout period, after which the phone cancels the current call
Ring Timeout	Set the ring timeout period, after which the phone rejects the current call when the incoming call rings timeout.
Enable Push XML Auth	Enable XML push authentication.
Ring Priority	Set the incoming call priority, configure whether to prioritize displaying the incoming call interface.
Enable push XML authentication	Enable push XML authentication.
Enable Display To Info	Set whether to display the “to” information.
Tone Settings	
Enable Holding Tone	When turned on, a tone plays when the call is held
Enable Call Waiting Tone	When turned on, a tone plays when call waiting
Play Dialing DTMF Tone	Play DTMF tone on the device when user pressed a phone digits at dialing, default enabled.
Play Talking DTMF Tone	Play DTMF tone on the device when user pressed a phone digits during taking, default enabled.
Auto Answer Tone	Upon activation, you will hear a “beep beep” prompt when auto-answering.
Ring Back Tone	Customize the ringing tone for outgoing calls.
Busy Tone	Customize the tone for hanging up calls.
DND Settings	
DND Option	Select to take effect on the line or on the phone or close.
Enable DND Timer	Enable DND Timer, If enabled, the DND is automatically turned on from the start time to the off time.

DND Start Time	Set DND Start Time
DND End Time	Set DND End Time
Intercom Settings	
Enable Intercom	When intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after specific delay.
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers the intercom call during a call. If the current call is intercom call, the phone will reject the second intercom call
Response Code Settings	
DND Response Code	Set the SIP response code on call rejection on DND
Busy Response Code	Set the SIP response code on line busy
Reject Response Code	Set the SIP response code on call rejection
Password Dial Settings	
Enable Password Dial	Enable Password Dial by selecting it, When number entered is beginning with the password prefix, the following N numbers after the password prefix will be hidden as *, N stand for the value which you enter in the Password Length field. For example: you set the password prefix is 3, enter the Password Length is 2, then you enter the number 34567, it will display 3**67 on the phone.
Encryption Number Length	Configure the Encryption Number length
Password Dial Prefix	Configure the prefix of the password call number
Power LED	
Ringing	Power lamp status when there is an incoming call, including off/on/slow flash/quick flash, default flash.
Hold/Held	The power lamp state, including off/on/slow flash/quick flash, is turned off by default when left/retained.
Mute	Power lamp status in mute mode, including off/on/slow flash/quick flash, off by default.
Talk/Dial	In the talk/dial state, the power lamp state, off is off, on is always red bright, the default is off.
Missed call	The state of the power lamp when there is a missed call, including off/on/slow flash/quick flash, the default slow flash.
SMS/Voice Mail	The status of power lamp when there is unread short message/voice message, including off/on/slow flash/quick flash, default slow flash.
Registration Failed	Power indicator light status on registration failure, including off/on/slow

	flashing/fast flashing, default slow flashing.
Phone Silent	Power indicator light status when in silent mode during standby, including off/on/slow flashing/fast flashing, default slow flashing.
Common	Standby power lamp state, off when off, open is always bright red. Off by default.
Power Saving	Whether to enable energy-saving mode, including off/on, default on.
Notification Popups	
Display Missed Call Popup	No incoming call popup prompt after opening, no popup prompt when closing, open by default.
Display Voice Mail Popup	Voice message popup prompt is not answered after opening, and it is opened by default if there is no popup prompt when closing.
Display SMS Popup	There is popup prompt for unread messages after opening, and there is no popup prompt when closing. It is opened by default.
Display Other Popup	When the handle is not hung back after opening, registration fails, IP acquisition fails, Tr069 connection fails and other abnormalities, there will be popup prompt when it is opened; otherwise, there will be no prompt when it is closed, and it will be opened by default.
Redial Settings	
Call completed	When the configuration option is enabled, after initiating a call, the other end replies with a response code of 486. After the other end phone ends the current call, the local end will prompt "call completed". Click confirm and redial the other end number
Allow automatic redialing	Configure whether to enable automatic redialing function. After the call ends, when the phone answers 486 on the other end (default busy response code), the phone will automatically redial according to the set number and time interval.
Automatic redial interval	Set the time interval for automatic redialing.
Automatic redial frequency	Set the number of automatic redials.
Redial to enter call history	Configure whether to directly redial the most recently called number or enter the call log when pressing the redial button.
Pick up & Park	
Display BLF PickUp Popup	Configure whether the phone LCD displays a prompt when there is an incoming call for the BLF-subscribed number.
Play BLF PickUp Tone	Configure whether to provide an audio prompt when there is an incoming call for the BLF-subscribed number.
Ring Type For BLF PickUp	Select the type of ringtone for call pickup configuration. .
Display Call Park Popup:	Configure whether to display a prompt pop-up window for call parking.
Play Call Park Tone:	After call parking, configure whether to provide an audio prompt.

Ring Type For Call Park:	Configure the ringtone for call parking

12.12 Phone settings >> Media Settings

Change voice Settings.

Table 25 - Voice settings

Parameters	**Description**
Codecs Settings	Select enable or disable voice encoding: G.711A/U, G.722, G.729,G.729AB, G.726-16, G726-24, G726-32, G.726-40, iLBC, Opus
Media Settings	
Handset Volume	Set the Handset volume, the value must be 1~9
Default Ring Type	Configure default ringtones. If no special ringtone is set for the phone number, the default ringtone will be used.
Speakerphone Volume	Set the hands-free volume to 1-9.
Headset Ring Volume	Set the volume of the earphone ringtone to 1~9.
Headset Volume	Set the volume of the headset to 1~9.
Speakerphone Ring Volume	Set the volume of hands-free ringtone to 1~9.
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
Headset Mic Gain	Set the earphone's radio volume gain to fit different models of earphones.
Opus playload type	Set Opus load type, range 96~127.
OPUS Sample Rate	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb (16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length
Enable VAD	Whether voice activity detection is enabled.
Enable Voice Mail Tone	Turn on voice message dial tone. When there is a new voice message message, the phone will activate a special dial tone
Onhook Time	Configure a minimum response time, which defaults to 200ms
Enable Hookflash	Whether to enable hookflash
EHS Type	EHS headset is available after enabling.
RTP Control Protocol(RTCP) Settings	
CNAME user	Set CNAME user
CNAME host	Set CNAME host
RTP Settings	
RTP keep alive	Hold the call and send the packet after 30s
RTP Relay	Set the RTP Relay
Alert Info Ring Settings	

Value	Set the value to specify the ring type.
Line	Set the corresponding line for incoming calls
Ring Type	None/Default/1.wav-10.wav

12.13 Phone settings >> MCAST

This feature allows user to make some kind of broadcast call to people who are in multicast group. User can configure a multicast DSS Key on the phone, which allows user to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address without involving SIP signaling. You can also configure the phone to receive an RTP stream from pre-configured multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.

Table 26 - Multicast parameters

Parameters	Description
MCAST Settings	
MCAST Send DTMF Mode	Set the DTMF mode sent by MCAST
MCAST Listening	
Priority	Define the priority of the active call, 1 is the highest priority, 10 is the lowest.
Enable Page Priority	The voice call in progress shall take precedence over all incoming paging calls.
Enable Prio Chan	Set the priority to enable multicast listening on the current channel
Enable Emer Chan	The multicast of each channel is not affected by the order, and other multicasts can be interrupted at will
Index/Priority	Set the priority of the curent multicast
Name	Listened multicast server name
Host:port	Listened multicast server's multicast IP address and port.
Channel	Set the MCAST channel
MCAST Dynamic	
Auto Exit Expires	Set the timeout for automatic cancellation of group multicast to default 60
MCAST Dynamic List	Including index, priority, MCAST IP, and port.

12.14 Phone settings >> Action

Action URLfor IPPBX system to submit phone events.

12.14.1 Action URL Description

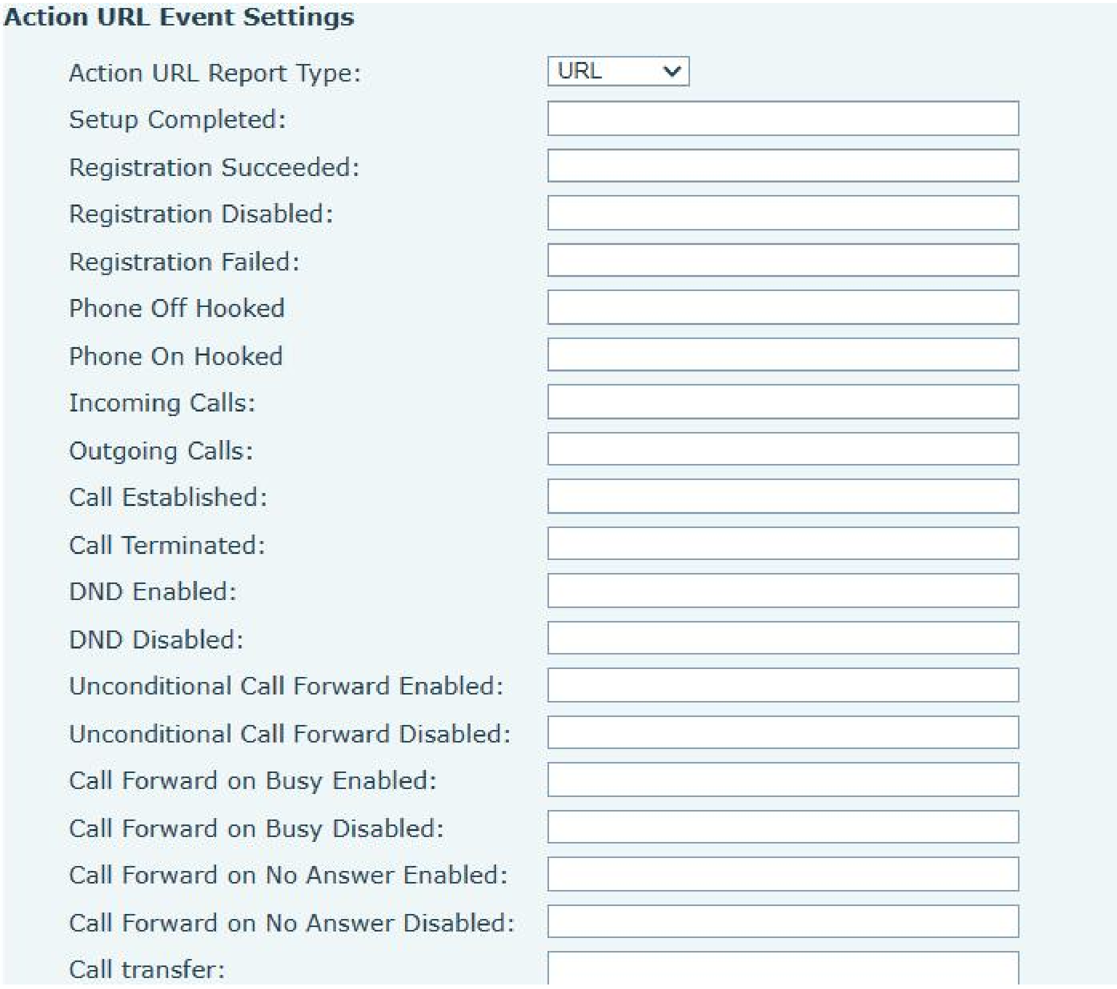
The Action URL is used by the phone itself to initiate an HTTP Get request to a remote control panel when the phone's own state changes. This event is then sent to the remote control panel, which can perform corresponding phone operations based on this state change.

12.14.2 Protocal Description

The HTTP URL format of the Action URL is defined by the HTTP Server end of the console. The phone is only responsible for initiating an HTTP Get request when the corresponding state changes. In general, the HTTP URL format of the Active URL is: "http://192.168.1.100/newcall.xml?num=$call_id"
"192.168.1.100" is the IP address of the remote control panel;
"newcall.xml?" is the processing method corresponding to various states defined by the remote control panel's HTTP Server;
"$call_id": the variables supported internally by the phone are automatically replaced with the current real values of the system before initiating an HTTP Get request. Internal variables start with "$"

12.14.3 Action URL Event Settings

Access the web management page of the phone, go to [Phone Settings] > [Action] > [Action URL Event Settings], and enter the corresponding URL in each input box for each event. For example, after the Incoming Call event, input http://192.168.1.100/newcall.xml?num=$call_id. After configuration, if there is an incoming call with the number 1234, the phone will initiate an HTTP Get to http://192.168.1.100/newcall.xml?num=1 (the call number).



Picture 124 - Action URL

12.14.4 Event List

Users can choose the Action URL notification type, which includes URL, SIP Notify, SIP Info, with the default set to URL.

Table 27 - Event List

Event	Description
Setup Completed	Phone startup completed
Registration Succeeded	Account registration successful
Registration Disabled	Account unregistered
Registration Failed	Account registration failed
Phone Off Hooked	Phone picked up
Phone On Hooked	Phone hung up
Incoming Calls	New incoming call
Outgoing Calls	Outgoing call
Call Established	Call established
Call Terminated	Call ended

DND Enabled	Do Not Disturb activated
DND Disabled	Do Not Disturb deactivated
Unconditional Call Forward Enabled	Unconditional Call Forwarding activated
Unconditional Call Forward Disabled	Unconditional Call Forwarding deactivated
Call Forward on Busy Enabled	Call Forward on Busy activated
Call Forward on Busy Disabled	Call Forward on Busy deactivated
Call Forward on No Answer Enabled	Call Forward on No Answer activated
Call Forward on No Answer Disabled	Call Forward on No Answer deactivated
Call transfer	Call transfer
Call hold	Call on hold
Call resume	Call resumed
Phone Silent	Phone muted
Phone Unsilent	Phone unmuted
Call Mute	Call muted
Call Unmute	Call unmuted
Missed Calls	Missed call
IP Changed	Change phone IP address
Phone State Idle	Phone transitioning from other interface to standby page
Phone State Talking	Phone in active call state
Phone State Ringing	Phone ringing
Voice Mail	Voicemail
SMS	Short message
Start Reboot	Control phone reboot
Web API Auth Changed	Web API authentication identity change
Reset Phone	Control phone factory reset
Insufficient ROM	Insufficient ROM space
Received Sip Message	SIP message received
Function keys	This configuration item is used for reporting URLs for function keys
Audio Mode Changed	Switch audio channel
Reject Incoming Call	Reject incoming calls
Cancel Call	Cancel outgoing call
Remote Call Cancel	Cancel remote outbound call
Handfree On	Handfree On
Handfree Off	Handfree Off

12.14.5 Parameter List

Table 28 - Parameter List

Parameter	Description
$mac	Device MAC Address
$ip	Currently available IP address
$model	Phone model
$firmware	Software version number
$active_uri	SIP URI of the current active SIP account (effective only during incoming, outgoing, and ongoing calls)
$active_user	User account part of the SIP URI of the current active account (effective only during incoming, outgoing, and ongoing calls)
$active_host	Server part of the SIP URI of the current active account (effective only during incoming, outgoing, and ongoing calls)
$local	Local SIP URI (effective during incoming, outgoing, and ongoing calls)
$remote	Remote SIP URI (effective during incoming, outgoing, and ongoing calls)
$display_local	Local display name (displays the number when there is no display name) (effective only during incoming and outgoing calls)
$display_remote	Remote display name (displays the number when there is no display name) (effective only during incoming and outgoing calls)
$call_id	Call ID (effective only during incoming, outgoing, and ongoing calls)
$duration	Call duration (effective only at the end of the call)
$date_time	Timestamp
$memory_free	Memory
$line	Line used for the call (effective during incoming, outgoing, ongoing, and registration)
$local_user	Local users in the call (effective during incoming, outgoing, and ongoing calls)
$local_server	Server used for SIP calls (effective during incoming, outgoing, and ongoing calls)
$local_domain	Domain for SIP calls (effective during incoming, outgoing, and ongoing calls)
$local_number	Local call number (effective during incoming, outgoing, and ongoing calls)
$local_displayname	Local call display name (effective during incoming, outgoing, and

	ongoing calls)
$remote_number	Remote number in the call (effective during incoming, outgoing, ongoing, and missed calls)
$remote_displayname	Display name of the remote number in the call (effective during incoming, outgoing, and ongoing calls)

Note:

1) Effective only during incoming calls means that the variable will be replaced with the corresponding information only when the variable is filled in the Incoming call options.

2) Effective only during outgoing calls means that the variable will be replaced with the corresponding information only when the variable is filled in the Outgoing call options.

3) Effective only during ongoing calls means that the variable will be replaced with the corresponding information only when the variable is filled in options related to calls such as Call established, Call terminated, Transfer call, Blind transfer call, Attended transfer call, Hold, Unhold, Mute, Unmute, etc.

Note! Action urls are used for IPPBX systems to submit phone events.

12.15 Phone settings >> Time/Date

The user can configure the time Settings of the phone on this page.

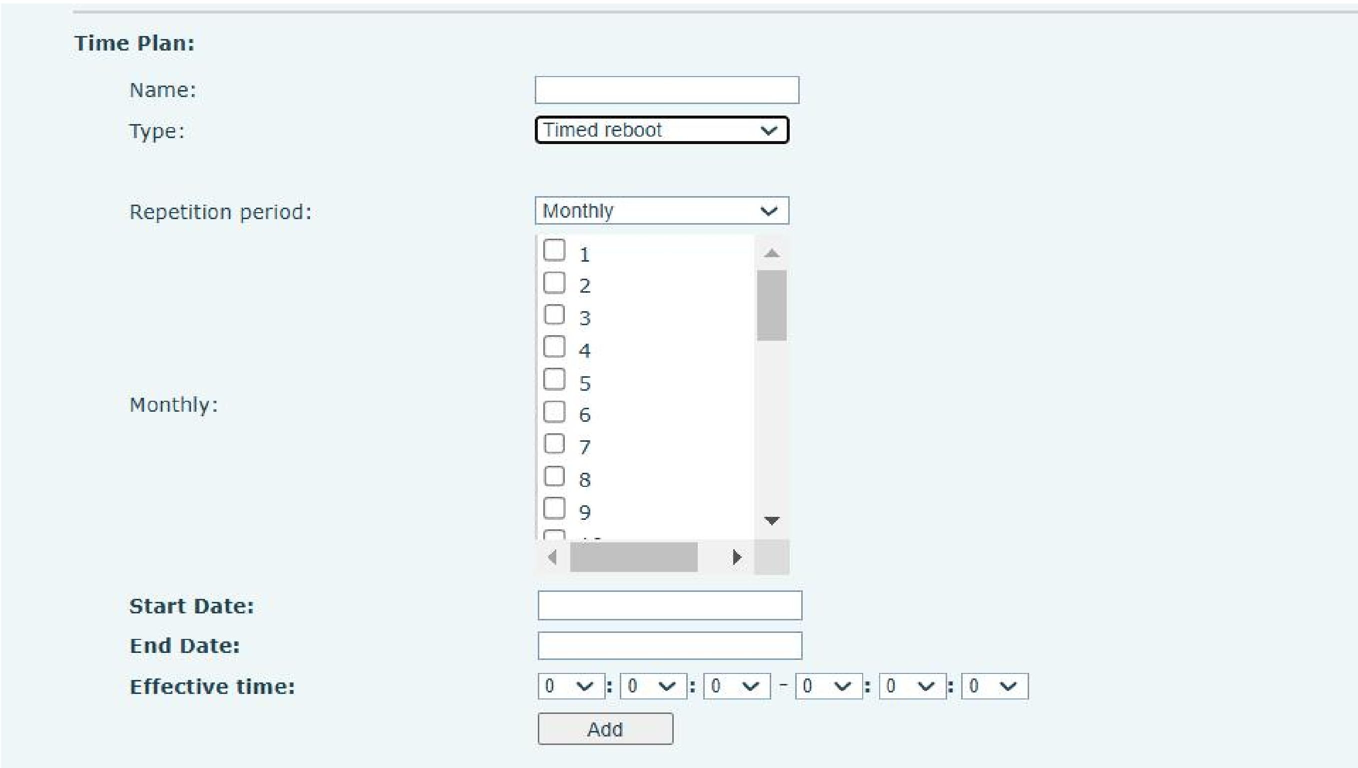
Table 29 - Time&Date settings

Parameters	Description
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Time Synchronized via DHCPv6	Enable time-sync through DHCPv6 protocol
Primary Time Server	Set primary time server address
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization.
Time Zone	Select the time zone
Resync Period	Time of re-synchronization with time server
Time/Date Format	
12-Hour Clock	Set the time display in 12-hour mode
Time/Date Format	Select the time/date display format
Daylight Saving Time Settings	
Location	Choose your local, phone will set daylight saving time automatically based on the local

DST Set Type	Choose DST Set Type, if Manual, you need to set the start time and end time.
Fixed Type	Daylight saving time rules are based on specific dates or relative rule dates for conversion. Display in read-only mode in automatic mode.
Offset	The offset minutes when DST started
Month Start	The DST start month
Week Start	The DST start week
Weekday Start	The DST start weekday
Hour Start	The DST start hour
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Manual Time Settings	You can set your time manually

12.16 Phone settings >> Time Plan

Time Plan (time management) settings can set a time period. The time point is to perform an action at a certain time, and the time period is to perform an action for a certain period of time.



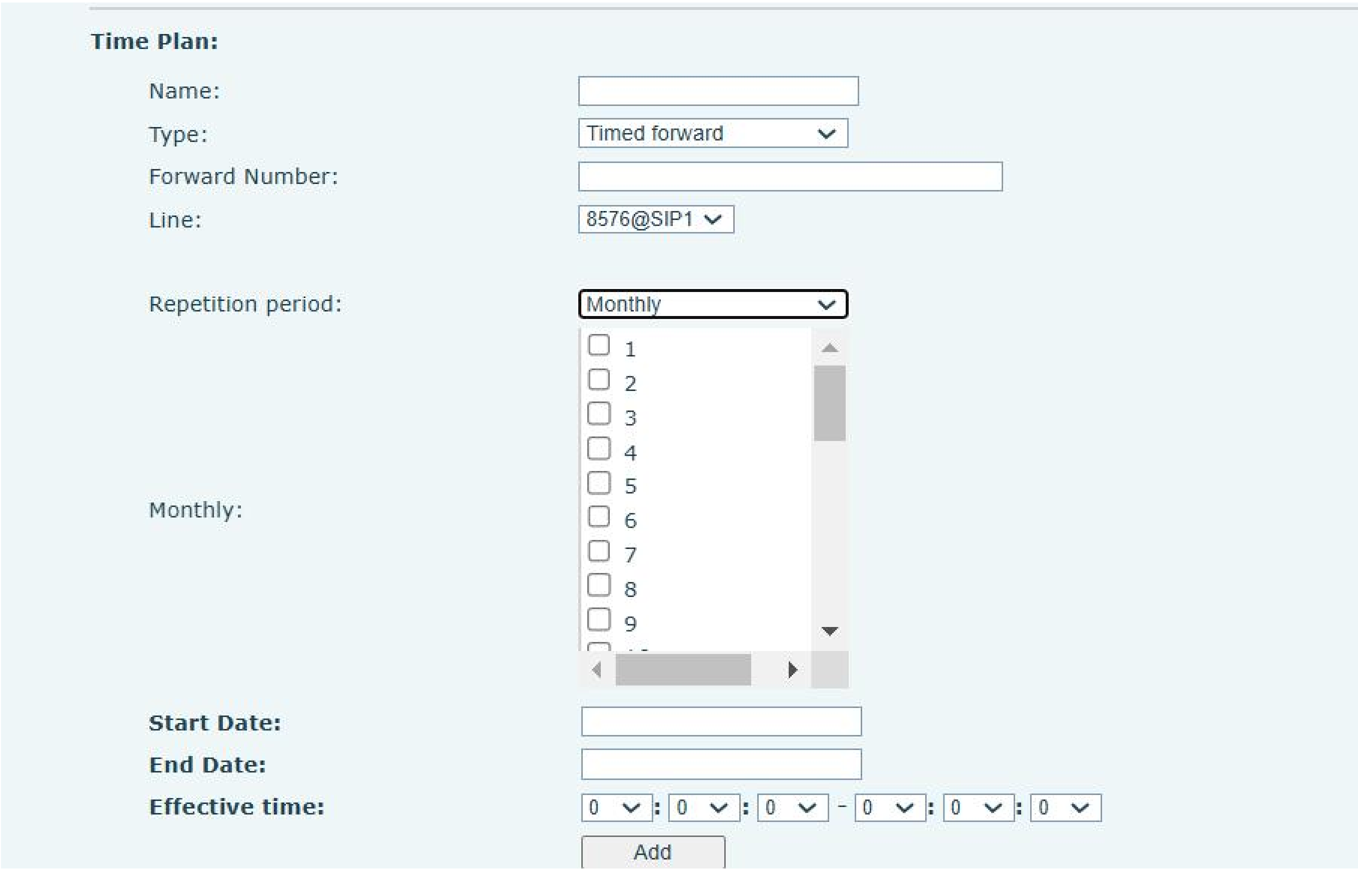
Picture 125 - Time Plan (1)

Table 30 - Time Plan

configure	Value	Description
Name	Custom	Set the name of time management
Time plan Type	1：Timed reboot 2：Timed upgrade 3：Timed forward 4：Timed config	Type，Action performed at a time period
Repetition period	0：No repetition 1：Daily 2：Weekly 3：Monthly	Repeat Type
No repetition/ Daily/Weekly/Monthly	When selecting the repetition period as' weekly ' 0-6：Sunday-Saturday, supports multiple separated by ";" When selecting the repetition period as' monthly ' 1-31：1-31 day When the repetition type is daily/non-repeating, the value is empty	Choose the exact repetiton period
Effective time	xx:xx:xx-xx:xx:xx	Start time - End time

■ Timed forward

When type is set to timed forward, the webpage will prompt for input of the forwarding number and line.



Picture 126 - Time Plan (2)

Forwarding Number: Configure the forwarding number to forward to the number within the set time period.

Line: Forward the specified line, when the line is set to a certain line, it will only take effect for this line.

1. Timed forwarding rules:

1) When there is forwarding under the line, the forwarding number under the line is used; when there is no forwarding number under the SIP line, when there is an incoming call within the time period set by the scheduled forwarding, the phone will be forwarded to the specified scheduled forwarding number; when outside the time period, no forwarding is performed. That is, the priority Line>Time Plan.

2) All scheduled forwarding types are unconditional forwarding.

■ Timed upgrade

The administrator configures the software version URL in the webpage [Auto Provision], and then goes to [Phone Settings] >> [Time Plan] to select the type as “Timed upgrade”, configure the repetition period, effective time, etc. Within the set time period, the phone will automatically upgrade its version.

■ Timed reboot

When the type is set to timed reboot, configure the repetition period, effective time, etc. Within the set time period, the phone will automatically reboot.

■ Timed config

You can fill in the configuration items that need to be modified and their corresponding configuration values at the “config” location, and set the repetition period and effective time. Within the set time period, the phone will automatically update the corresponding configuration.

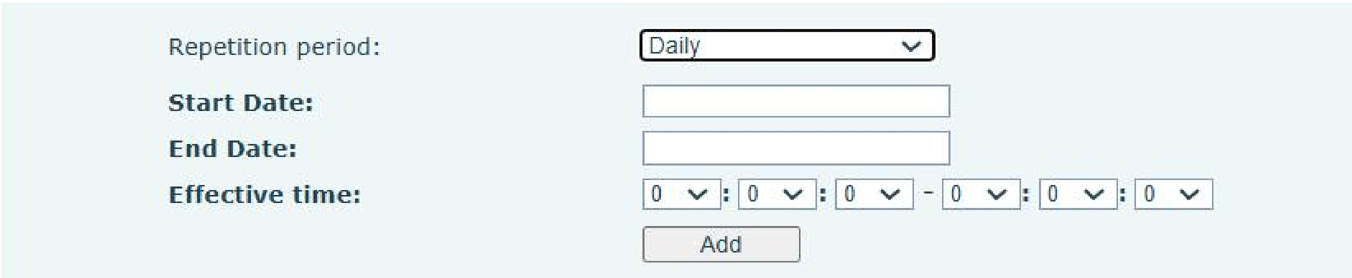
12.16.1 Repeat Period Select Daily

Select daily as the repetition period, and enter any time in the date format from 00:00:00-23:59:59 in the effective time input box.

The first and third input boxes only allow the input of any integer from 00 to 23, and do not automatically add 0 before entering integers less than 10.

The second and fourth input boxes only allow the input of any integer from 00 to 59, and do not automatically add 0 before entering integers less than 10.

The third and sixth input boxes only allow the input of any integer from 00 to 59, and do not automatically add 0 before entering integers less than 10.

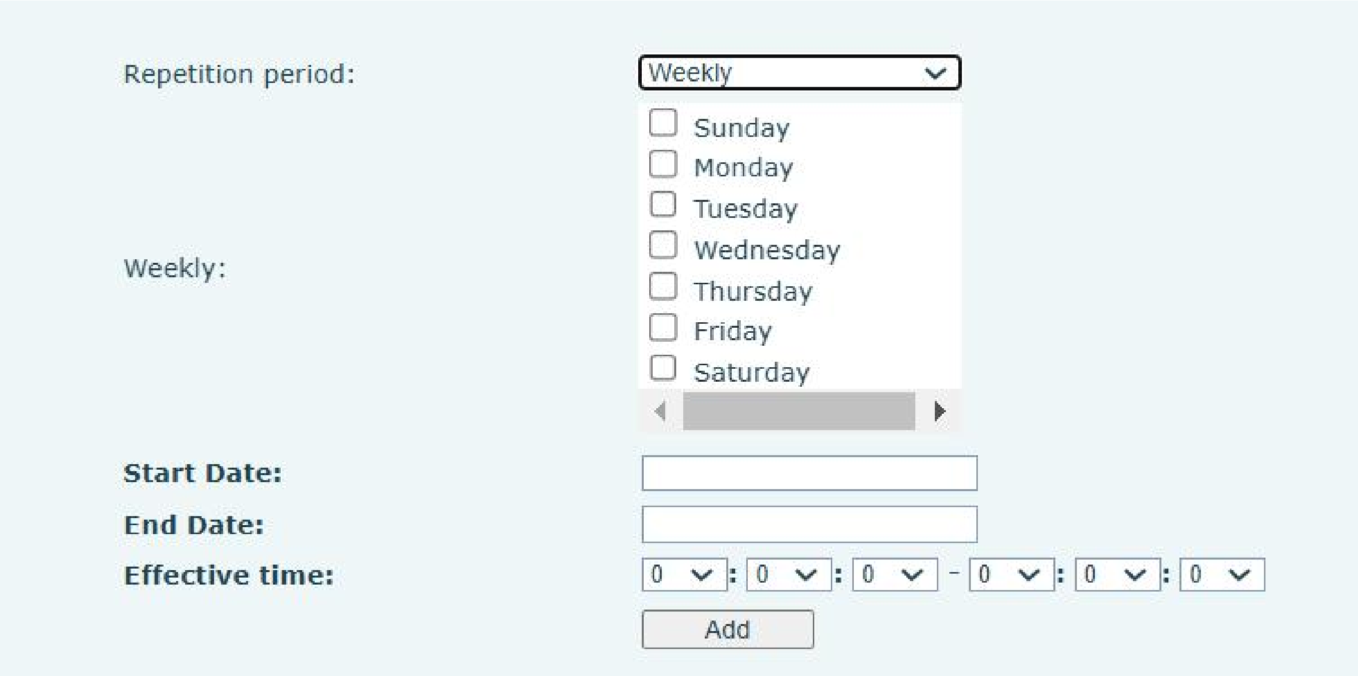


Picture 127 - Time Plan (3)

12.16.2 Repeat Period Select Weekly

Day of the week selection box, check it to take effect.

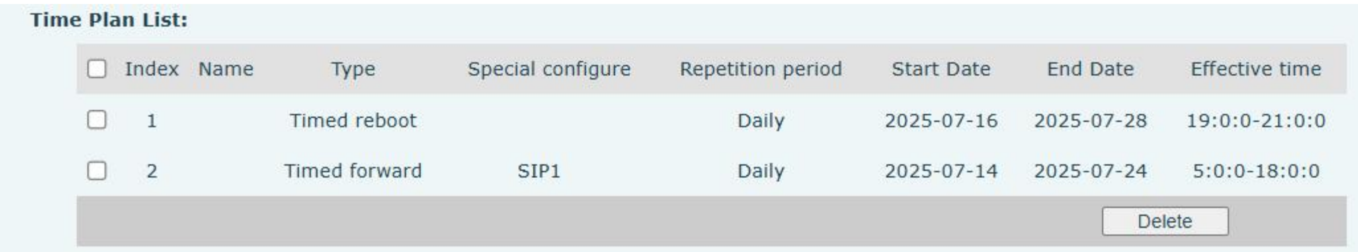
The final effective time is the combination of the day of the week and the set time.



Picture 128 - Time Plan (4)

12.16.3 Time Plan List

All configurations submitted after the configuration is submitted are displayed in a list, and the order is sorted by week (day, Monday, Tuesday...), and if the week is the same, it is sorted by time (time from small to large). The function sequence is restarted first and then upgraded.



Time Plan List:

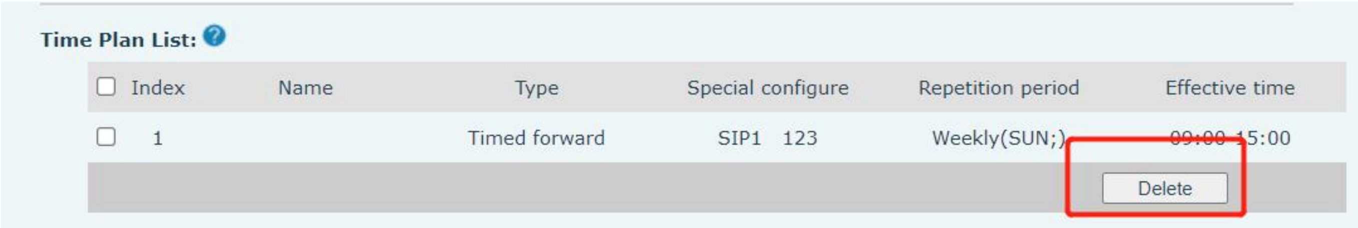
Index	Name	Type	Special configure	Repetition period	Start Date	End Date	Effective time
1		Timed reboot		Daily	2025-07-16	2025-07-28	19:0:0-21:0:0
2		Timed forward	SIP1	Daily	2025-07-14	2025-07-24	5:0:0-18:0:0

Delete

Picture 129 - Time Plan (5)

12.16.4 Delete

Check the box before the serial number, click to select all configuration items in the list.
Click Delete to delete the checked configuration in the configuration list, and it will become invalid after deletion.

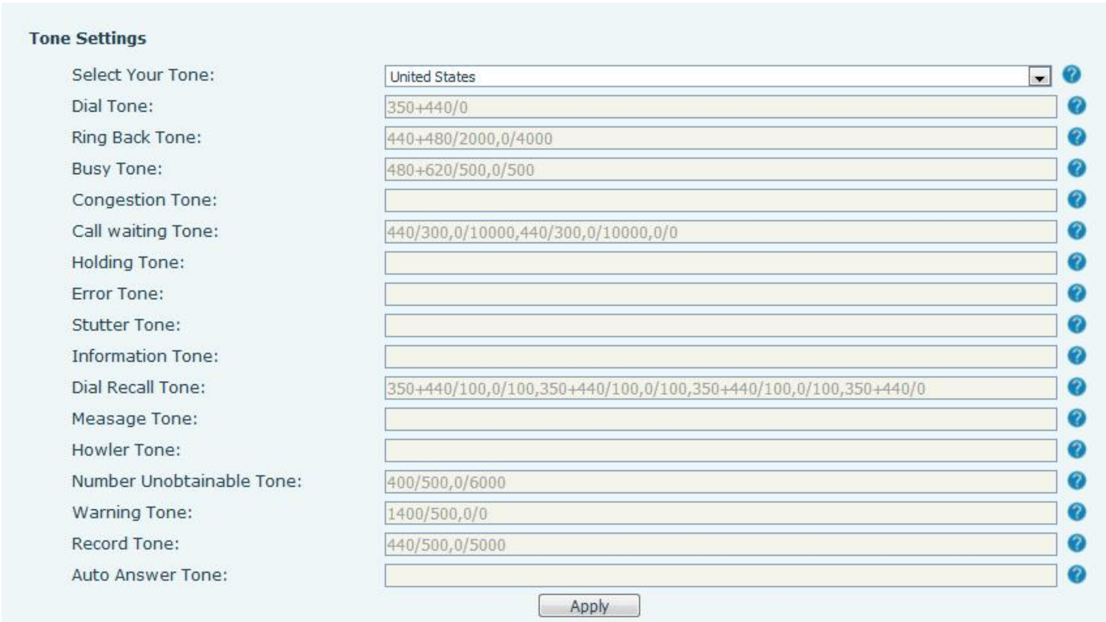


Picture 130 - Time Plan (6)

12.17 Phone settings >> Tone

This page allows users to configure a phone prompt.

You can either select the country area or customize the area. If the area is selected, it will bring out the following information directly. If you choose to customize the area, you can modify the button tone, call back tone and other information.



Picture 131 - Tone settings on the web

12.18 Phone settings >> Advanced

User can configure the advanced configuration settings in this page.

- Screen Configuration.
  - Screensaver
  - Timeout to Screensaver
  - Customer Time Value
- LCD Menu Password Settings.

The password is 123 by default.

- Keyboard Lock Settings.
  - Keyboard Password

- Keyboard Time
- Keyboard Lock Type

- Greeting Words

The greeting message will display on the top left corner of the LCD when the device is idle, which is limited to 12 characters. The default chars are 'VoIP Phone'.

Note: Only when the default line function is disabled can the Greeting Words be displayed in the upper left corner of the standby screen. (You can disable the default line in the [Menu] >> [Features] >> [General] interface.)

12.19 Phonebook >> Contacts

User can add, delete, or edit contacts in the phonebook in this page. User can browse the phonebook and sorting it by name, phones, or filter them out by group.
To add a new contact, user should enter contact's information and press "Add New contact" button to add it.
To edit a contact, click on the edit button after the contact to enter the editing box. After completing the editing, press the "Confirm" button. If you want to add a contact to a group, you need to ensure that the phone already has a contact group. Otherwise, you need to add the group first, then edit the contact, and move the contact to the group.
To delete one or multiple contacts, check on the checkbox in front of the contacts wished to be deleted and click the "Delete" button, or click the "Delete All" button with selecting any contacts to clear the phonebook.
User can also add multiple contacts into a group by selecting the group in the dropdown options in front of "Add to Group" button at the bottom of the contact list, selecting contacts with checkbox and click "Add to Group" to add selected contacts into the group.
Similarly, user can select multiple users and add them into blocked list by click "Add to Blocked List" button.

12.20 Phonebook >> Cloud phonebook

Cloud Phonebook

User can configure up to 8 cloud phonebooks. Each cloud phonebook must be configured with an URL where an XML phonebook is stored. The URL may be based on HTTP/HTTPs or FTP protocol with or without authentication. If authentication is required, user must configure the username and password.
To configure a cloud phonebook, the following information should be entered,

- Cloud Phonebook name (must)
- Cloud Phonebook URL (must)
- Access username (optional)
- Access password (optional)

Import XML Contact

Select the configured XML phone book to import into the phone.

LDAP Settings

The cloud phonebook allows user to retrieve contact list from a LDAP Server through LDAP protocols. User must configure the LDAP Server information and Search Base to be able to use it on the device. If the LDAP server requests an authentication, user should also provide username and password.

To configure a LDAP phonebook, the following information should be entered,

- Display Title (must)
- LDAP Server Address (must)
- LDAP Server Port (must)
- Search Base (must)
- Access username (optional)
- Access password (optional)

Note! Refer to the LDAP technical documentation before creating the LDAP phonebook and phonebook server.

Broadsoft Call logs Settings

The Broadsoft server itself can store call records and contact information. Users can configure the Broadsoft server address, username, and password to view relevant information on the LCD screen.

- Display Title (optional)
- Server Address (must)
- Username (optional)
- Password (optional)
- SIP Line (optional)

Broadsoft Directory Settings

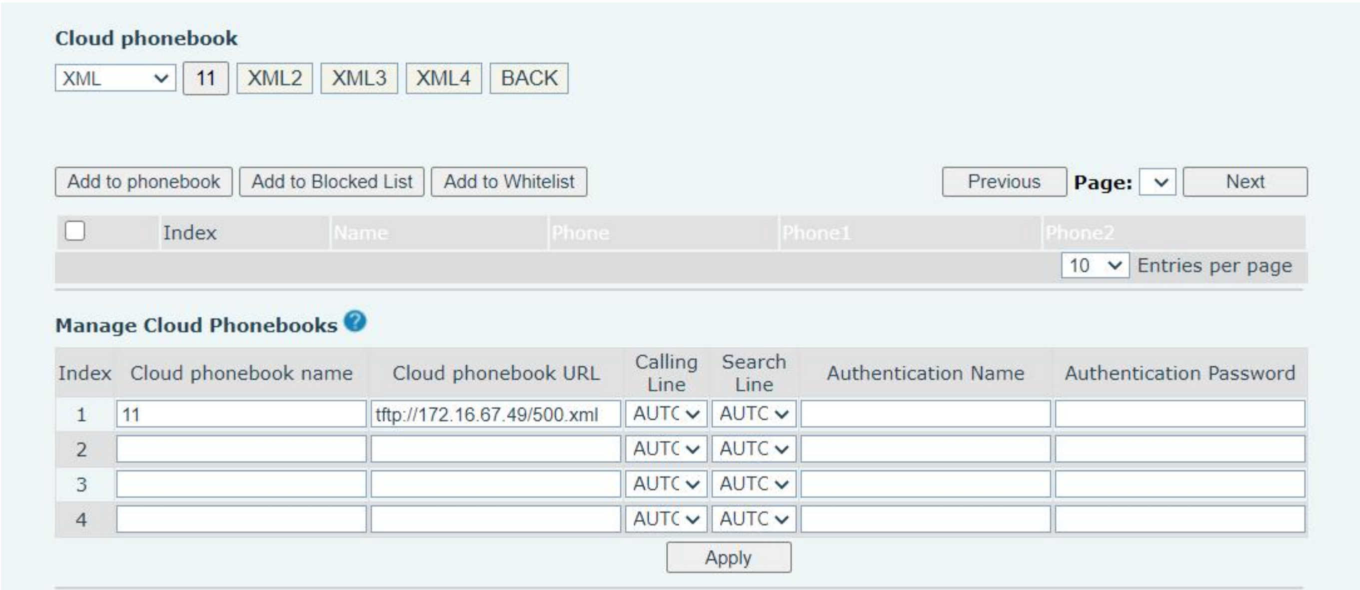
To configure a Broadsoft directory, the following information should be entered,

- Display Title (optional)
- Server Address (must)
- Username (optional)
- Password (optional)
- SIP Line (optional)

Web page preview

Phone page supports preview of Internet phone directory and contacts

- After setting up the XML Voip directory or LDAP,
- Select [**Phone book**] >> [**Cloud phone book**] >> [**Cloud phone book**] to select the type.
- Click the set XML/LDAP to download the contact for browsing.



Picture 132 - Web cloud phone book Settings

12.21 Phonebook >> Call List

■ Restricted Incoming Calls:

It is similar like a blocked list. Add the number to the blocked list, and the user will no longer receive calls from the stored number until the user removes it from the list.

Users can add specific Numbers to the restricted incoming calls list or add specific prefixes to the blocked list to block calls with all Numbers with this prefix.

■ Allowed Incoming Calls:

Allowed list type includes ALL, DND, FWD.When DND is enabled, the incoming call number can still be called.

■ Restricted Outgoing Calls:

Adds a number that restricts outgoing calls and cannot be called until the number is removed from the table.

12.22 Phonebook >> Web Dial

Use web pages for call, reply, and hang up operations.

12.23 Phonebook >> Advanced

Users can export the local phone book in XML, CSV, and VCF format and save it on the local computer. Users can also import contacts into the phone book in XML, CSV, and VCF formats.

Attention! If the user imports the same phone book repeatedly, the same contact will be ignored. If the name is the same but the number is different, the contact is created again.

Users can delete groups or add new groups on this page. Deleting a contact group does not delete contacts in that group.

12.24 Call Logs

The user can browse the complete call record in this page. The call record can be sorted by time, call number, contact name or line, and the call record can be screened by call record type (incoming call, outgoing call, missed call, forward call).
The user can also save the number in the call record to his/her phone book or add it to the blocked list\allowed list.
Users can also dial the web page by clicking on the number in the call log.
Users can also download call records conditionally and save them locally.

12.25 Function Key >> Softkey

The user can set the mode, display style and display page.

Users can also customize the DSSkey and use it on an appropriate page based on the actual application scenario.

Note: The function options of the DSSkey are included on all pages. Therefore, some function options may not be available on some pages

Table 31- Softkey configuration

Parameter	Description
Softkey mode	
Softkey mode	Disabled and more, the default is disabled
Exit Display style	
Exit Display style	Exit on the Left and Exit on the Right
Screen	
Call Dialer	Redial/2aB/Delete/Exit/callback/Dial/Join /Voice Mail /local contacts/Pickup/CallLog/Missed/Clear/In/Out/ Pause/Next line /Prev line/Headset/None/Missed/Cloud Phonebook/Save/ DSSkey
Conference(Conf)	Hold/Conference(Conf)/Split/End/Release/mute /DSSkey/ Headset
Desktop	CallLog/Menu/Local contacts/Previous Account /Next Account/Blocked List/Callback/Call Forward/Lock Phone/Memo/Missed /Voice Mail/ out/Redial/SMS/status/Headset/Network /DSSkey/In/DND/reboot/Cloud Phonebook
Divert Dialer	2aB/ Delete/Exit / Local contacts /CallLog/Clear/Missed/Out/Headset/Dial/Divert Dialer/None/In/Cloud Phonebook/DSSkey
Ending	Redial/End/Headset/Release /DSSkey/None
Predictive Dialer	Dial /2aB/ Delete/Exit/Callback/Local contacts/Redial/Pickup /Voice Mail/ Join /CallLog/Clear/Missed/Pause/Dial/Headset//DSSkey/In/Next line/Prev

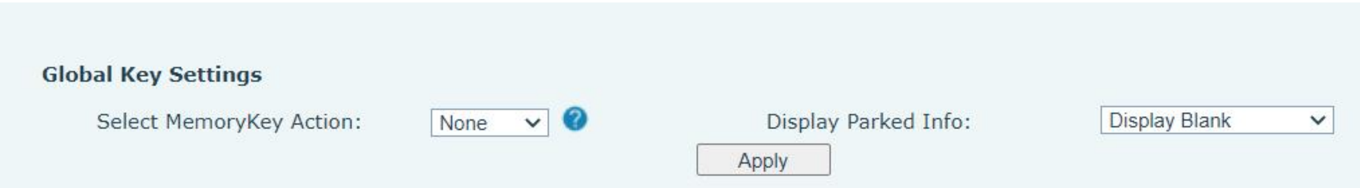
	line/Save/None/Out/Cloud Phonebook
Ringing	Answer / Reject/Mute/Release/Headset/DSSkey/None/Divert dialer
Talking Audio	Hold/Transfer/End/Release /New Call/ Local contacts /Listen/ CallLog/Next call/Prev call/ Private hold/Headset/DSSkey/Conference/RTP/None
Transfer Alerting	End/Transfer/Headset/Release /DSSkey/None
Transfer Dialer	Delete/Exit/2aB/Dial/Local contacts/ Transfer/CallLog/Clear/ Missed/Out/ Pause/Headset/DSSkey/None/In/Cloud PhoneBook
Trying	End/Release/Headset /DSSkey/None/Transfer
CallLog	Hold/Transfer/Meeting/End/Answer /Forward/ Microphone mute /Next call/New call/Prev call/ Reject/Release/Headset /Listen/ DSSkey/Exit/Option/ Delete/Dial

12.26 Function Key >> Advanced

- **Global Key Settings**

Select MemoryKey Action: for example, the phone set the memory key value to 4370. When 4370 calls, press this key to hold the call or hang up.

Display Parked Info: Select the resident information to be displayed, including blank display, first name display, and only number and standard display.



- **Programmable key Settings**

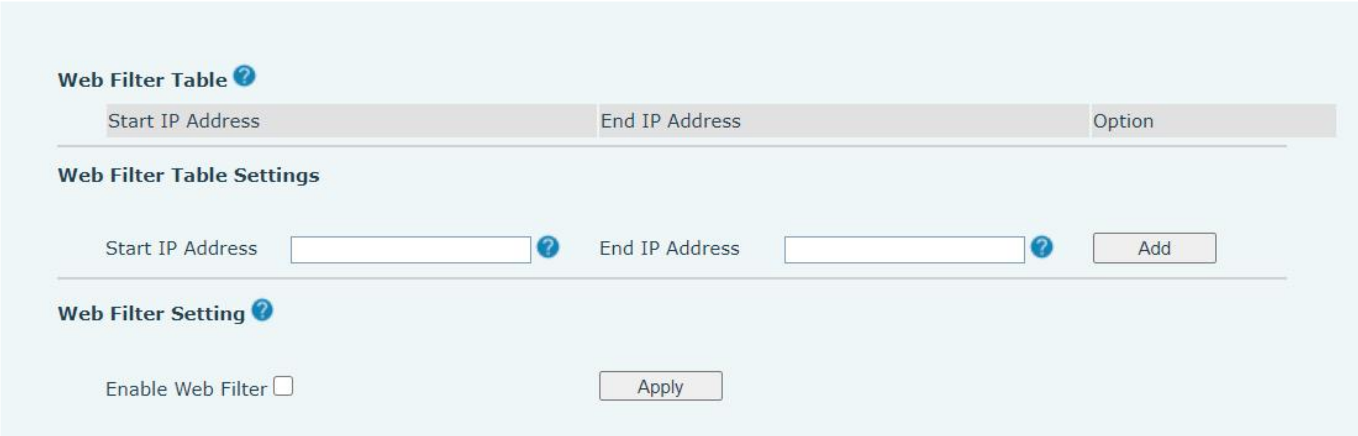
Please refer to the Table 26 Softkey configuration

12.27 Application >> Manage Recording

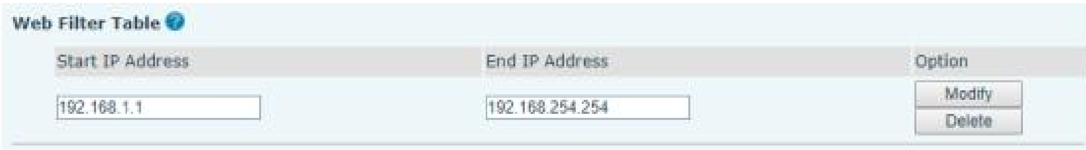
See 9.3 Record for details of recording.

12.28 Security >> Web Filter

The user can set up a configuration management phone that allows only machines with a certain network segment IP access.



Picture 133 - Web Filter settings



Picture 134 - Web Filter Table

Add and remove IP segments that are accessible; Configure the starting IP address within the start IP, end the IP address within the end IP, and click [**Add**] to submit to take effect. A large network segment can be set, or it can be divided into several network segments to add. When deleting, select the initial IP of the network segment to be deleted, and then click [**Delete**] to take effect.

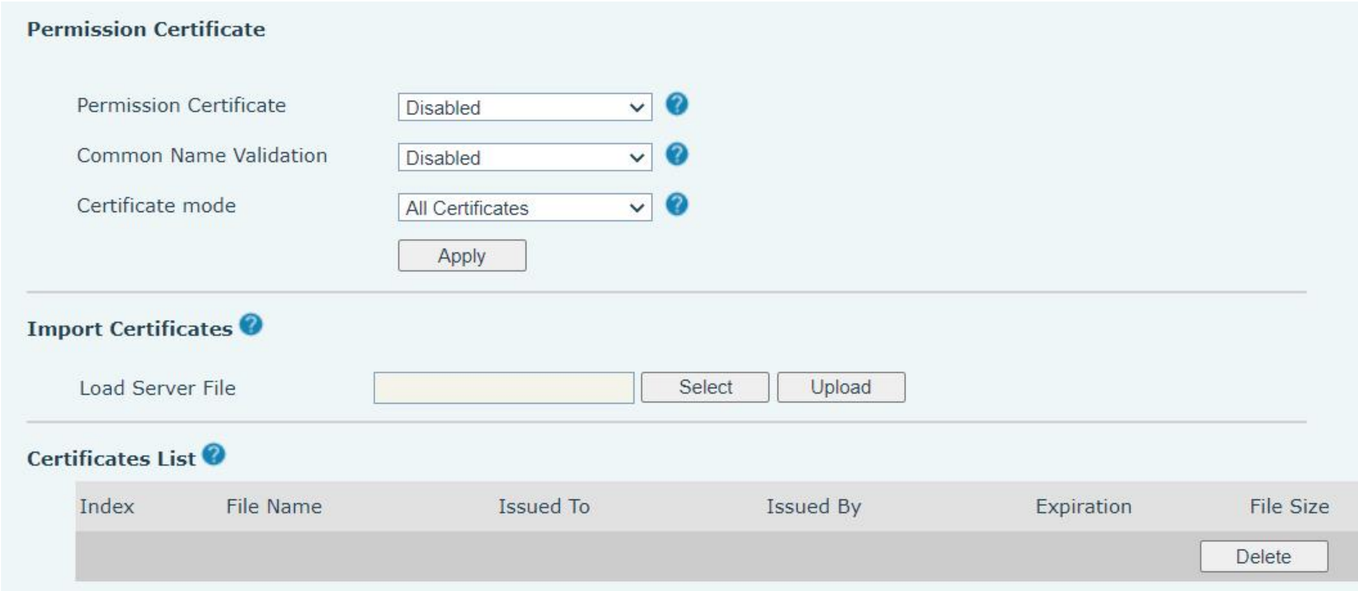
Enable web page filtering: configure enable/disable web page access filtering; Click the "Apply" button to take effect.

Note: if the device you are accessing is in the same network segment as the phone, please do not configure the filter segment of the web page to be outside your own network segment, otherwise you will not be able to log in the web page.

12.29 Security >> Trust Certificates

Set whether to open license certificate and general name validation, select certificate module.

You can upload and delete uploaded certificates.

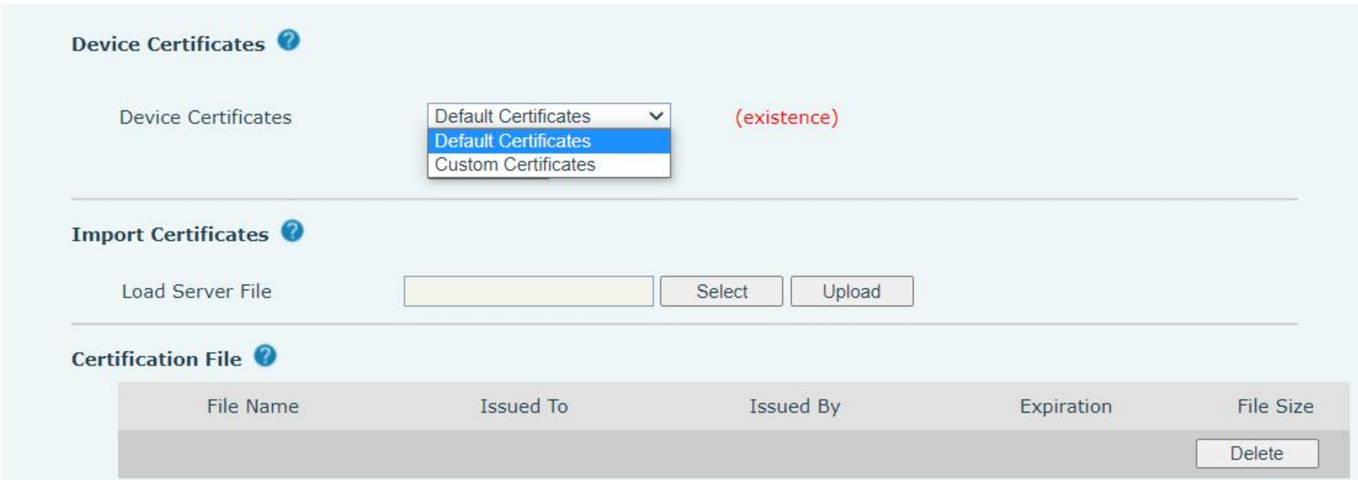


Picture 135 - Certificate of settings

12.30 Security >> Device Certificates

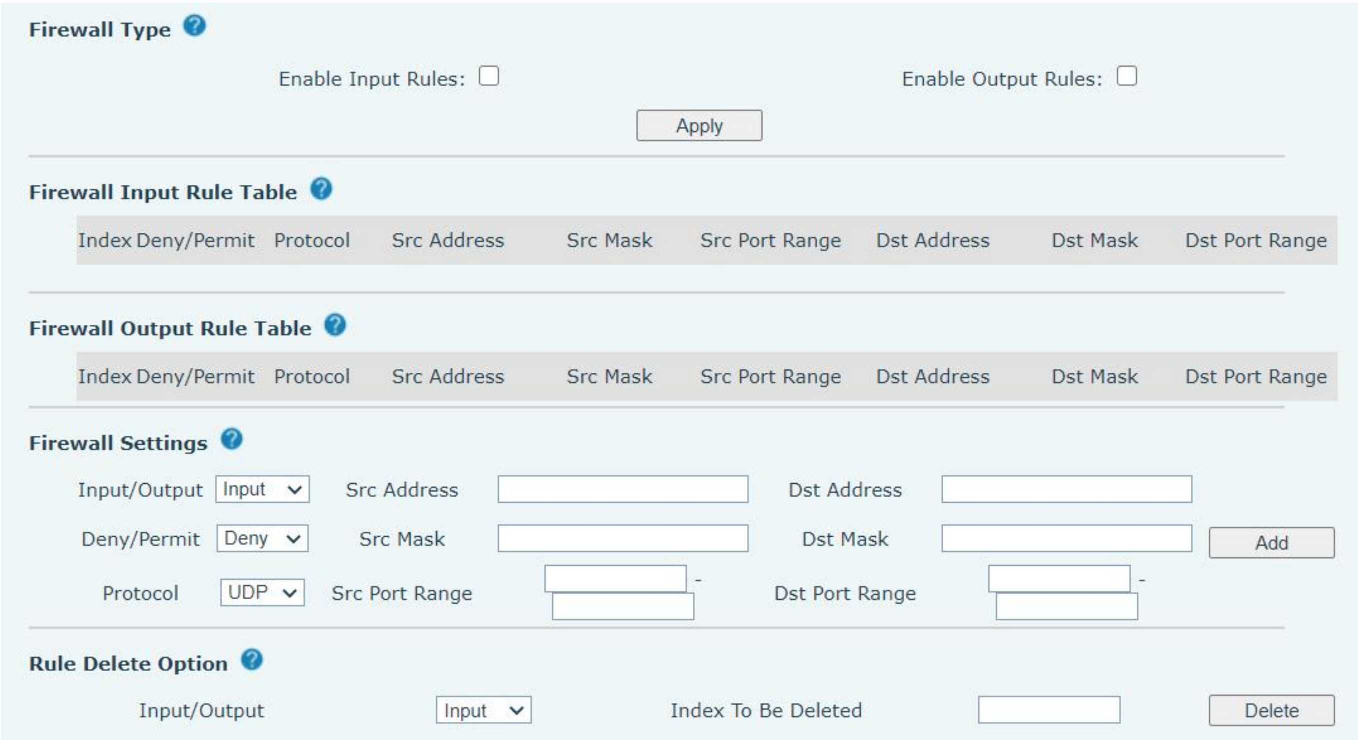
Select the device certificate as the default and custom certificate.

You can upload and delete uploaded certificates.



Picture 136 - Device certificate setting

12.31 Security >> Firewall



Picture 137 - Network firewall Settings

Through this page can set whether to enable the input, output firewall, at the same time can set the firewall input and output rules, using these Settings can prevent some malicious network access, or restrict internal users access to some resources of the external network, improve security.

Firewall rule set is a simple firewall module. This feature supports two types of rules: input rules and output rules. Each rule is assigned an ordinal number, allowing up to 10 for each rule.

Considering the complexity of firewall Settings, the following is an example to illustrate:

Table 32 - Network Firewall

Parameter	Description
Enable Input Rules	Indicates that the input rule application is enabled.
Enable Output Rules	Indicates that the output rule application is enabled.
Input/Output	To select whether the currently added rule is an input or output rule.
Deny/Permit	To select whether the current rule configuration is disabled or allowed;
Protocol	There are four types of filtering protocols: TCP | UDP | ICMP .
Src Port Range	Filter port range
Src Address	Source address can be host address, network address, or all addresses 0.0.0.0; It can also be a network address similar to *.*.*.0, such as: 192.168.1.0.
Dst Address	The destination address can be either the specific IP address or the full

	address 0.0.0.0; It can also be a network address similar to *.*.*.0, such as: 192.168.1.0.
Src Mask	Is the source address mask. When configured as 255.255.255.255, it means that the host is specific. When set as 255.255.255.0, it means that a network segment is filtered.
Dst Mask	Is the destination address mask. When configured as 255.255.255.255, it means the specific host. When set as 255.255.255.0, it means that a network segment is filtered.

After setting, click [**Add**] and a new item will be added in the firewall input rule, as shown in the figure below:

Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range
1	deny	icmp	192.168.1.0	192.168.1.154	0-9	255.255.255.0	255.255.255.0	0-9

Picture 138 - Firewall Input rule table

Then select and click the button [**Apply**].

In this way, when the device is running: ping 192.168.1.118, the packet cannot be sent to 192.168.1.118 because the output rule is forbidden. However, other IP of the ping 192.168.1.0 network segment can still receive the response packet from the destination host normally.

Picture 139 - Delete firewall rules

Select the list you want to delete and click [**Delete**] to delete the selected list.

12.32 Device Log >> Device Log

You can grab the device log, and when you encounter an abnormal problem, you can send the log to the technician to locate the problem. See 13.6 Get log information.

13 Trouble Shooting

When the phone is not in normal use, the user can try the following methods to restore normal operation of the phone or collect relevant information and send a problem report to technical support mailbox.

13.1 Get Device System Information

Users can get information by pressing the [**Menu**] >> [**Status**] option in the phone.The following information will be provided:

The network information

Equipment information (model, software and hardware version), etc.

13.2 Reboot Device

Users can reboot the device from soft-menu, [**Menu**] >> [**Basic**] >> [**Reboot System**], and confirm the action by [**OK**]. Or, simply remove the power supply and restore it again.

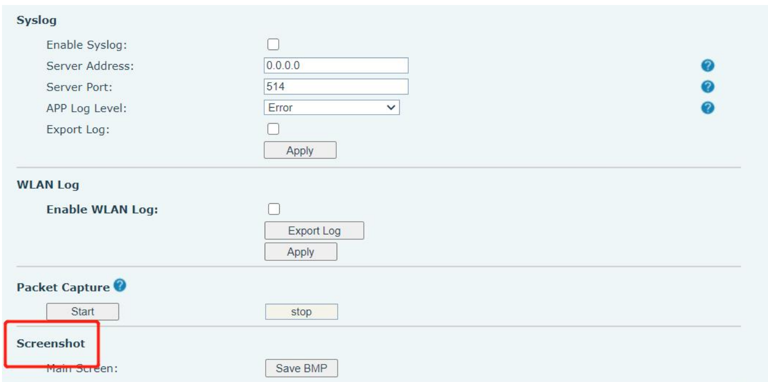
13.3 Reset Device to Factory Default

Reset Device to Factory Default will erase all user's configuration, preference, database and profiles on the device and restore the device back to the state as factory default.

To perform a factory default reset, user should press [**Menu**] >> [**Advanced**] , and then input the password （Default password 123） to enter the interface.Then choose [**Factory Reset**] , press [**Enter**], choose different restore modes and confirm the action by [**OK**].The device will be rebooted into a clean factory default state.

13.4 Screenshot

If there is a problem with the phone, screenshots can help technicians locate the function location and clarify the problem phenomenon. To obtain a screenshot, log in to the phone webpage [System]>>[Auxiliary Tools]. The screenshot can capture the main screen image (which can be captured on the problematic interface).



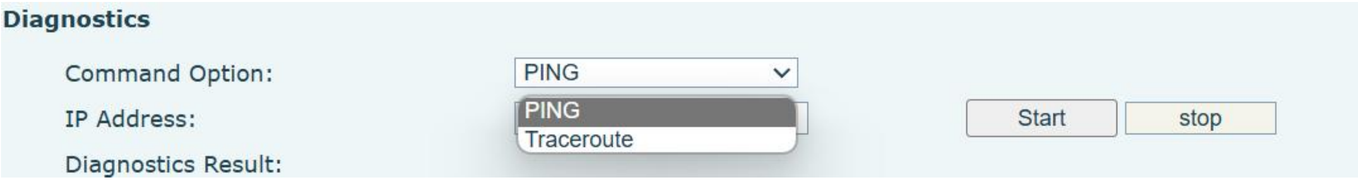
Picture 140 - Screenshot

13.5 Watch dog

When encountering an exception, the watchdog can detect the exception and restart the phone. You can log in to the web page [**System**] >> [**Tools**] >> [**Watchdog**] to enable or disable the watchdog.. If the watchdog is not detected within a short period of time, it will run again. Will reboot the entire system.

13.6 Diagnosis

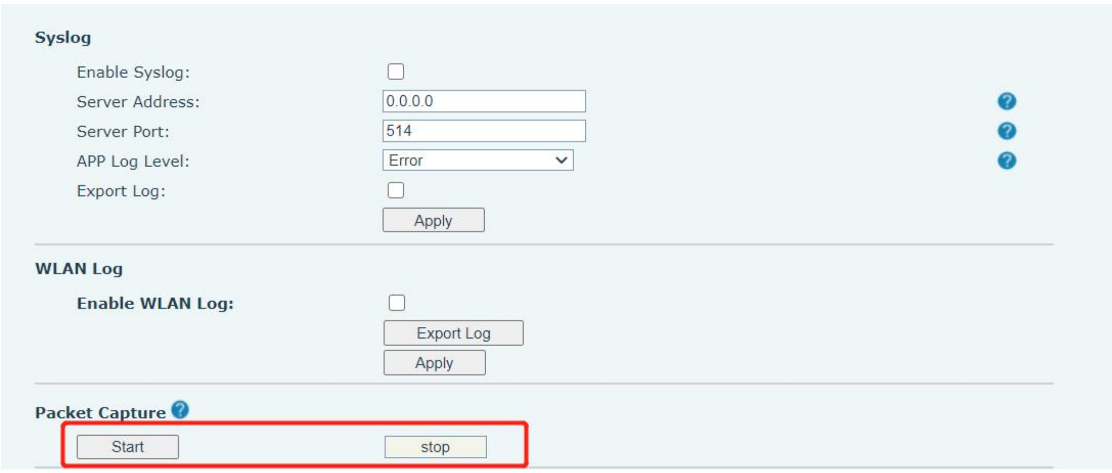
The device diagnosis function of the web page [**System**] >> [**Tools**] >> [**Diagnosis**] can be used to diagnose the connection between the phone and the network. You can select PING or Traceroute. After selecting PING or Route tracing, enter the computer ip address, phone ip address, or public ip address, and click Start. You can check whether the connection is normal.



Picture 141 - Web diagnostics

13.7 Network Packets Capture

Sometimes it is helpful to dump the network packets of the device for issue identification. To get the packets dump of the device, user needs to log in the device web portal, open page [**System**] >> [**Tools**] and click [**Start**] in “**LAN Packet Capture**” section. A pop-up message will be prompt to ask user to save the capture file. User then should perform relevant operations such as activate/deactivate line or making phone calls and click [**Stop**] button in the web page when operation finished. The network packets of the device during the period have been dumped to the saved file.



Picture 142 - Web capture

User may examine the packets with a packet analyzer or send it to support mailbox.

13.8 Get Log Information

Log information is helpful when encountering an exception problem. In order to get the log information of the phone, the user can log in the phone web page, open the page [**Device log**], click the [**Start**] button, follow the steps of the problem until the problem appears, and then click the [**Stop**] button, [**Save**] to local analysis or send the log to the technician to locate the problem.

13.9 Common Trouble Cases

Table 33 - Trouble Cases

Trouble Case	Solution
Device could not boot up	1. The device is powered by external power supply via power adapter or PoE switch. Please use standard power adapter provided by or PoE switch met with the specification requirements and check if device is well connected to power source. 2. If you saw "POST MODE" on the device screen, the device system image has been damaged. Please contact location technical support to help you restore the phone system.
Device could not register to a service provider	1. Please check if device is well connected to the network. The network Ethernet cable should be connected to the [Network] port NOT the [PC] port. If the cable is not well connected to the network icon [WAN disconnected] will be flashing in the middle of the screen. 2. Please check if the device has an IP address. Check the system information, if the IP displays "Negotiating…", the device does not have an IP address. Please check if the network configurations is correct. 3. If network connection is fine, please check again your line configurations. If all configurations are correct, please kindly contact your service provider to get support, or follow the instructions in "13.5 Network Packet Capture" to get the network packet capture of registration process and send it to support to analyze the issue.
No Audio or Poor Audio in Handset	1. Please check if Handset is connected to the correct Handset ( ) port NOT Headphone ( ) port. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Poor Audio or Low Volume in Headphone	1. There are two Headphone wire sequence in the market. Please use the Headphone provided by, or consult the wire sequence if you wish to use a third-party headphone.

	2. The network bandwidth and delay may be not suitable for audio call at the moment.
Audio is chopping at far-end in Hands-free speaker mode	This is usually due to loud volume feedback from speaker to microphone. Please lower down the speaker volume a little bit, the chopping will be gone.