A330 & A330i
User Manual

Software Version: 2.20.3
Release Date: 2026.2.25

Directory

1 Figure

2 Table

3 Overview

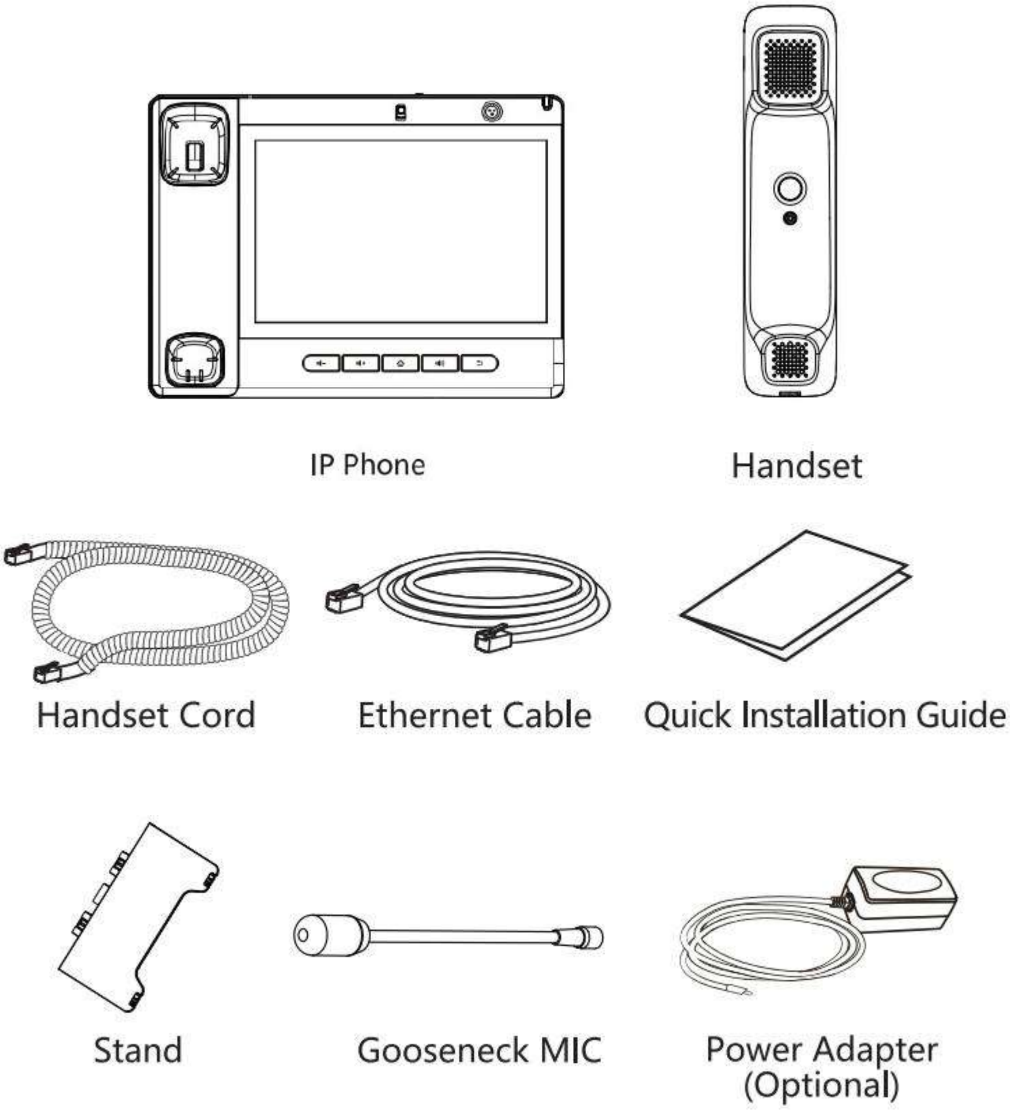
Fanvil A330 & A330i is a professional IP communication console designed for centralized communication, paging, and operational coordination in enterprise and institutional environments. Featuring a 10.1-inch 1280×800 touchscreen, it delivers MP3-quality audio with background noise reduction and full-duplex voice communication for clear interactions. With a 13MP adjustable camera and privacy shutter, the A330 / A330i supports the H.264 video codec to deliver smooth video calls and real-time monitoring. Comprehensive call handling functions - including call recording, call forwarding, and audio/video conferencing, while dual Gigabit Ethernet, Wi-Fi 6 (2.4 GHz / 5 GHz), and Bluetooth 5.4 ensure reliable connectivity across wired and wireless networks.

A330 & A330i with the built-in broadcast and intercom system enables automatic device discovery, rapid deployment and centralized management of up to 300 endpoints. An advanced task management framework allows one-touch live paging / pre-recorded audio broadcasting / rapid conferencing, and the tasks could be executed on schedule, triggered via DTMF, network commands, and alarm input ports for diverse communication scenarios. The Unattended Management Mode automatically forwards calls to designated on-duty personnel to ensure critical calls and emergency events are not missed, while video decoding with HDMI output enables video tour mode and extended monitoring.

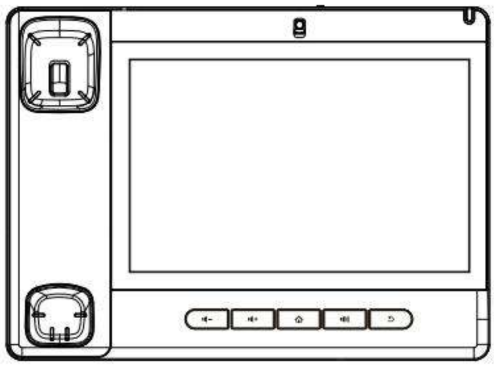
To help users gain a better understanding of product details, this user manual serves as a comprehensive reference guide for the A330 & A330i IP phones. Please note that this document may not apply to the latest software version. If you have any questions, you can access the built-in help and prompt interface on your A330 & A330i IP phone or download the latest user manual from the official website.

3.1 Packing Contents

3.1.1 A330i Packing Contents

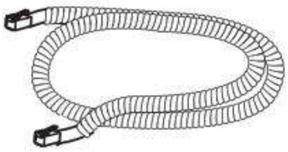


3.1.2 A330 Packing Contents

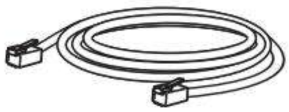


IP Phone

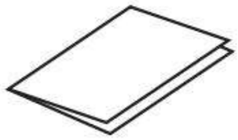
Handset



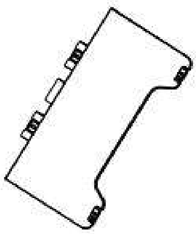
Handset Cord



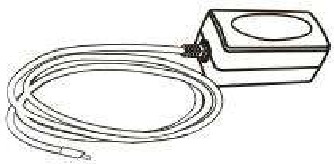
Ethernet Cable



Quick Installation Guide



Stand



Power Adapter
(Optional)

4 Install Guide

4.1 Use PoE or External Power Adapter

The device supports two power supply methods: external power adapter and Power over Ethernet (PoE) switch power supply.

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 adapter 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 us and the PoE switch met the specifications to ensure the device work properly.

4.2 Desktop Installation

4.2.1 A330i&A330 Desktop Installation

The device supports desktop use. If the phone is placed on the desktop for use,please follow the instructions in the picture below to install the phone.

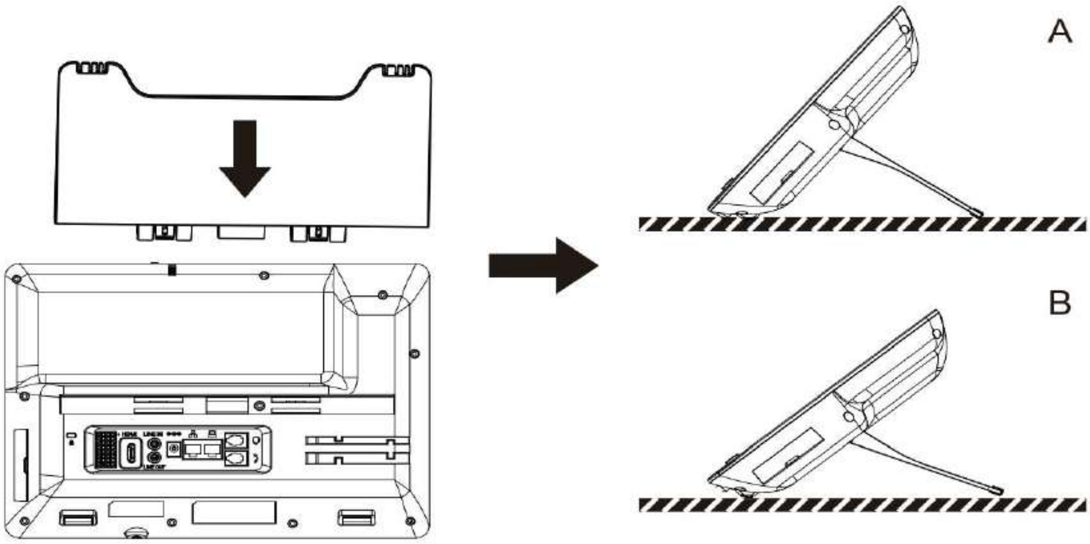


Figure 1 - Desktop Phone Installation

Please connect power adapter, network, PC, handset and headset to the corresponding ports as described in the picture below.

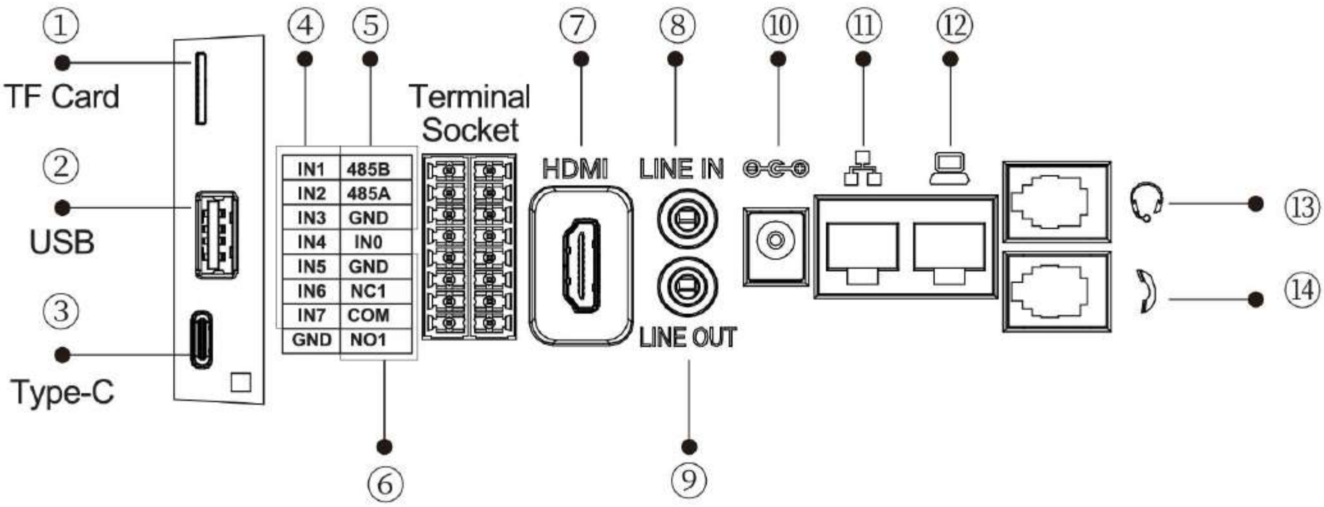


Figure 2 - Connecting to the Device

Table 1 - Hardware Interface Description

Index	**Description**
①TF Card Slot	Supports external SD cards up to 128 GB.
②USB Port	Connects USB devices (USB disk).
③Type-C Port	Compatible with multiple devices.
④Alarm Input	Connects alarm sensors to trigger preset alarms.
⑤RS-485 Port	GND, RS-485A, RS-485B.
⑥1 Short-Circuit Output	Controls electric locks, alarms, etc. NC: Connected when idle; COM: Relay contact; NO: Disconnected when idle.
⑦HDMI Port	Connects external displays for HD video/audio.
⑧LINE IN	Connects broadcast servers, external audio sources, etc.
⑨LINE OUT	Connects power amplifiers, etc.
⑩Power Port	Connects power adapter, DC 12V, 3A.
⑪Network Port	Connects to LAN or Internet
⑫PC Port	Connects to computer's network port.
⑬Headphone Port	Connects headphones.
⑭Handset Port	Connects IP Phone handsets.

4.2.2 A330i&A330 Camera Adjustment Method

Moving the slider covers the camera lens, and rotating the knob adjusts the vertical viewing angle of the camera.

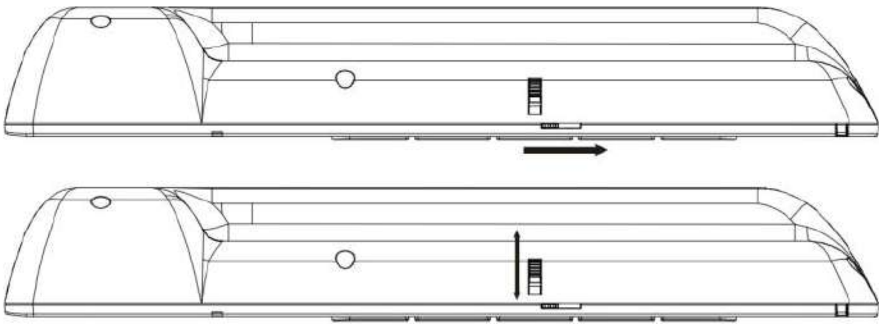


Figure 3 - Camera Adjustment

4.2.3 A330i Gooseneck MIC Installation

After aligning the gooseneck microphone with the port, load it and tighten the nut.

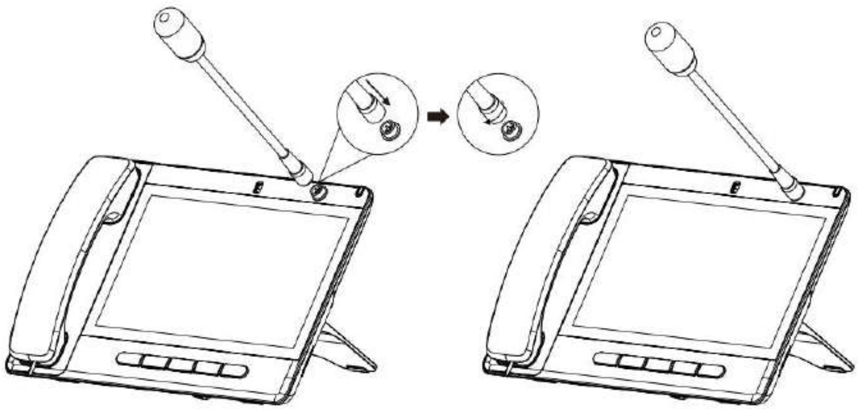


Figure 4 - A330i Gooseneck MIC Installation

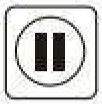
5 Appendix Table

You can view the icons and corresponding functions in the table below.

Table 2 - Keypad Icons

	Volume down
	Volume up
	Home key
	Hands-free (HF) speaker
	Redial

Table 3 - Status Prompt and Notification Icons

	In hands-free mode
	Call is on hold
	New SMS
	Dialed call
	In headset mode
	Auto-answering activated
	New VM Messages
	Missed call (s)

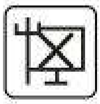
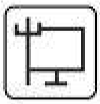
	In handset mode
	Call Forward activated
	Do not disturb activated
	Forward call
	Enable Blocked List
	Missed call (Status bar)
	Internet is disconnected
	Mute activated
	Ringer off
	Internet is connected
	Received call
	Enable Allowed List

Table 4 - Dsskey Icons

Icon	Description
	Line
	BLF & Call Park
	Speed Dial
	Intercom & Key Event/Intercom
	Voice Message& Key Event/Voice Message
	Key Event/DND
	Key Event/Call Hold
	Key Event/Call Transfer
	Key Event/Phonebook
	Key Event/Redial
	Key Event/Pickup
	Key Event/Join
	Call Forward & Key Event/Call Forward
	Key Event/Call Logs
	Key Event/Flash
	Key Event/Memo
	Key Event/Headset
	Key Event/Release

	Key Event/Lock Phone
	Key Event/SMS
	Key Event/Call Back
	Key Event/Power Light
	Key Event/Prefix
	Key Event/End
	Key Event/Handfree
	Key Event/Answer Key
	Key Event/Local Contact
	Record
	Auto Headset
	URL & Action URL
	DTMF
	Multicast
	MCAST Listening
	Unfold
	Collapse
	Mission
	Door Phone & URL/Open Door

	URL/IP Camera

6 Getting Started

6.1 Instructions of Keypad

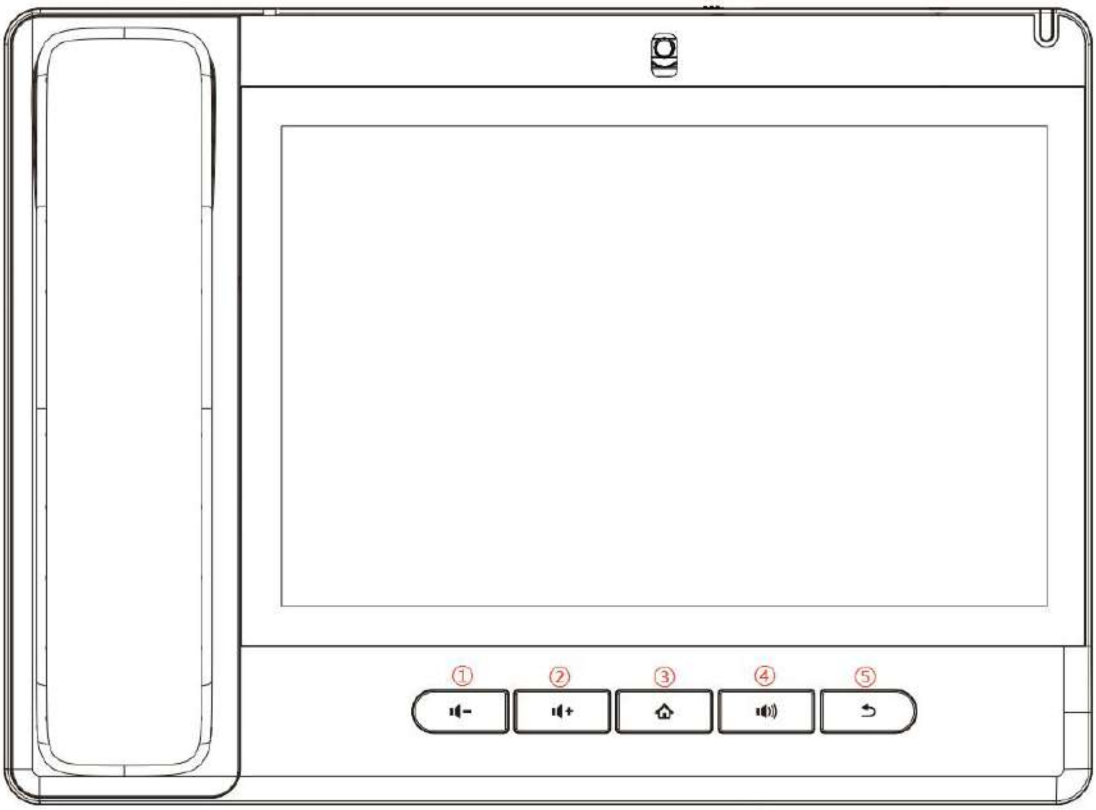


Figure 5 - Button Description

The figure above shows the keypad layout of the phone. Each key has its own specific function. Users should refer to the key descriptions in Figure 5 to operate the phone.

Table 5 - Button Description

Index	Name	Description
①	Volume down	decrease volume
②	Volume up	increase volume
③	Home Keys	Press this key to return the phone to the standby interface.
④	Hands-free Key	Press this key to activate the speakerphone audio channel.
⑤	Return key	Press to return to the previous page

6.2 Using Handset / Hands-free Speaker / Headset

- **Using Handset**

To talk over handset, user should lift the handset off the device and dial the number, or dial the number first, then lift the handset and the number will be dialed. User can switch audio channel to handset by lifting the handset when audio channel is opened in speaker or headset.

- **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 Headset**

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

- **Using Line Keys(Defined by dsskey)**

User can use line key to make or answer a call on specific line. If handset has been lifted, the audio channel will be opened in handset. Otherwise, the audio channel will be opened in hands-free speaker or headset.

6.3 Idle Screen

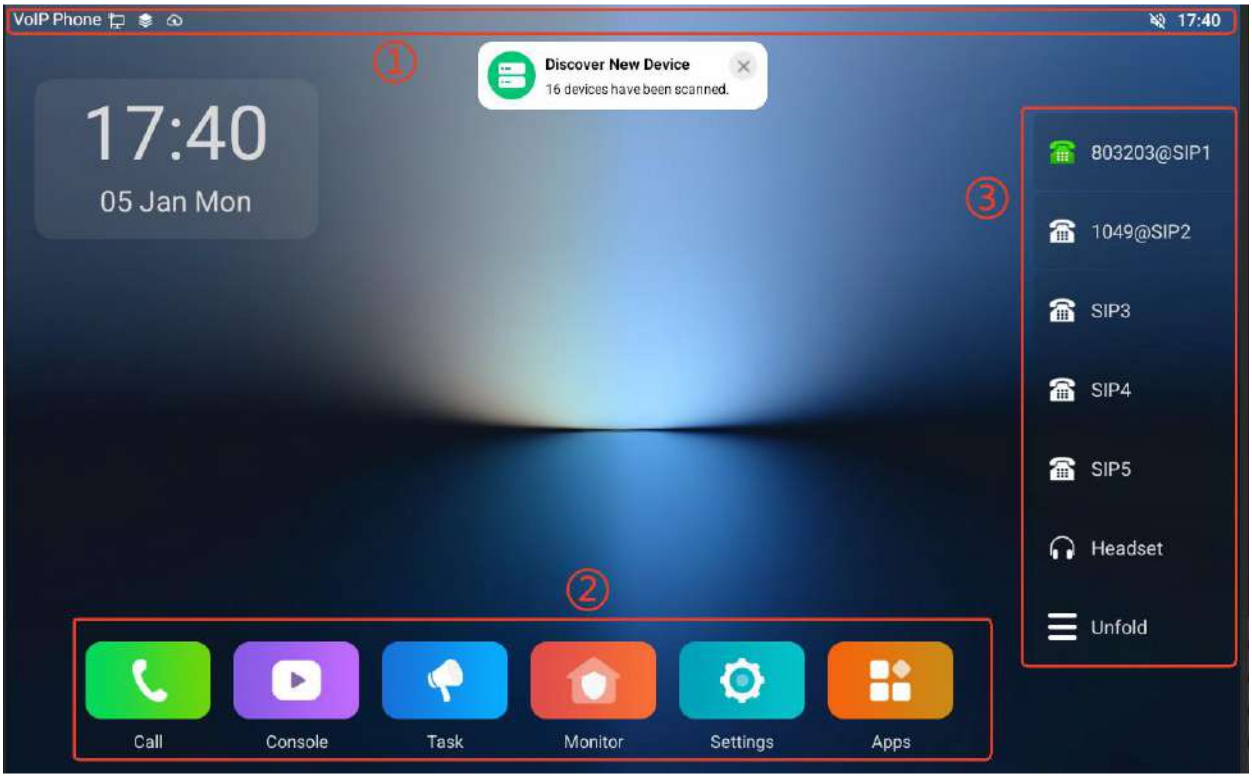


Figure 6 - Default Idle Screen

The figure above shows the default idle screen interface, which is the state of the user interface most of the time.

In the top area ① of the main screen, status indicators and notification icons are displayed. Swipe down from the top to open the drop-down status bar, where you can view the device's status, messages, and editable settings (such as voice messages, missed calls, auto-answer, Do Not Disturb, network connection status, etc.).

The lower area ② contains function menu buttons. Users can operate the phone via these buttons. On the right side, ③ is the shortcut key list. Descriptions of the shortcut key icons are provided in 5 Appendix Table.

6.4 Screen Touch Instructions

The device can be configured and operated through a series of touch screen actions.

- **Tap**

On any interface of the device, you can enter the settings and operation interfaces by tapping.The device supports multi-touch.

- **Long press**

Tap the [**Apps**] menu button on the idle screen to enter the app list interface. Long press an app icon to uninstall it.

- **Swipe**

Swipe down from the top of the idle home screen to open the drop-down status bar and view device information such as network connection status and date and time; swipe up to exit the information interface.

6.5 Phone Status

Users can check the phone status via two methods: the phone interface and the web interface.

- **Phone Interface:** From the idle screen, navigate to [**Settings**] >> [**Status**] to view information such as network and account configurations. For details, refer to section 14.1 Status.

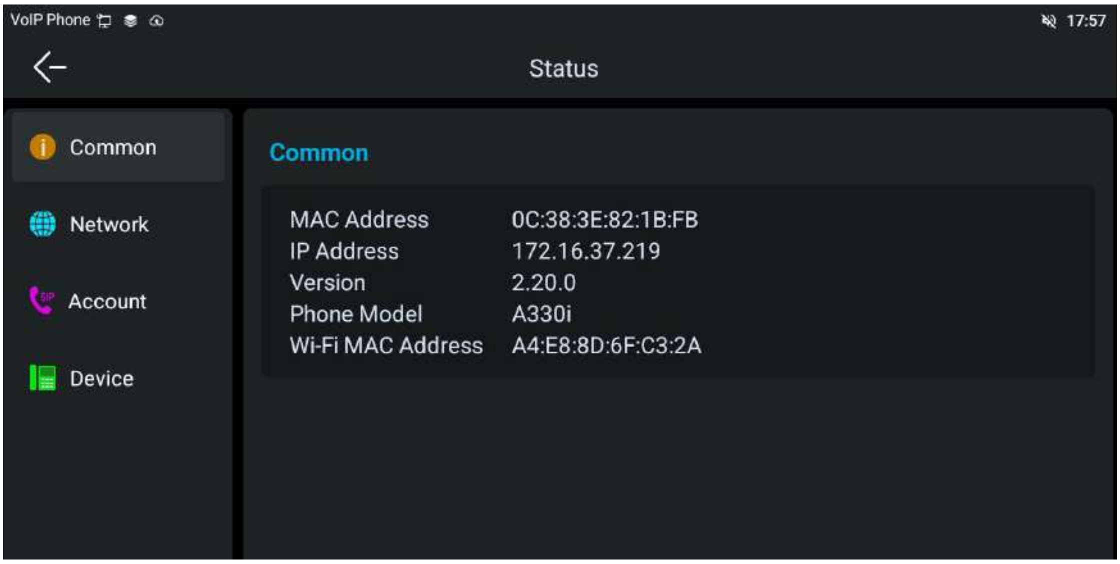


Figure 7 - Phone Status

- **Web Interface:** Refer to section 6.6 Web Management to log in to the phone's web interface, then navigate to the [**System**] >> [**Information**] page to check the phone status, as shown in the figure below.

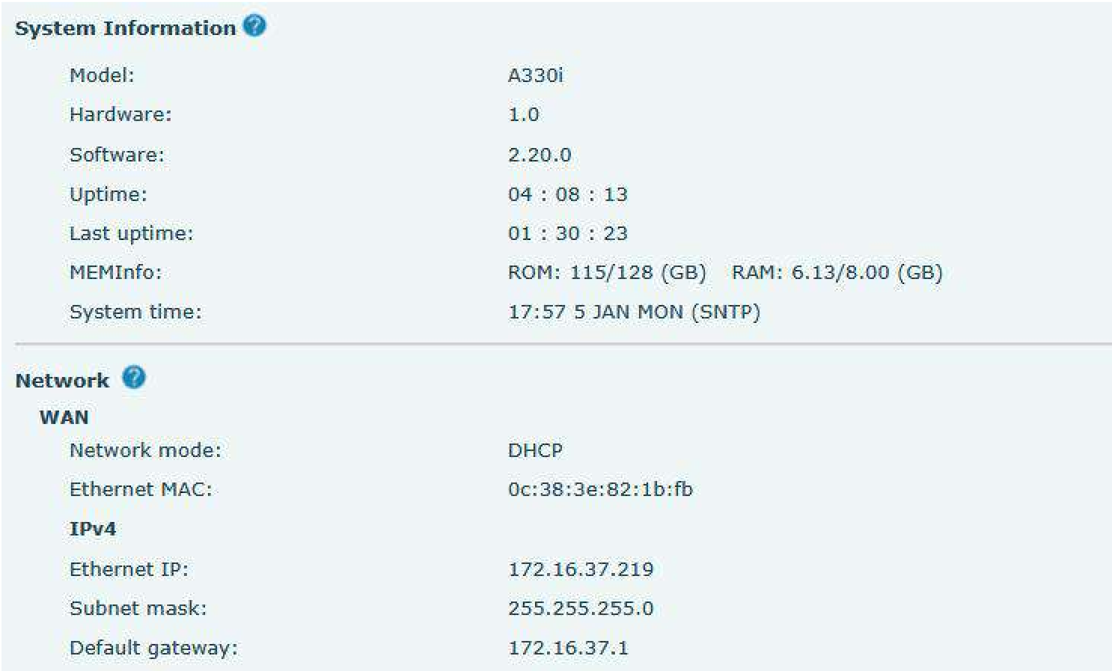


Figure 8 - Phone Status on Web

6.6 Web Management

Users can manage and operate the phone via its web management interface. First, users need to enter the phone’s IP address in a web browser to access the interface. The phone’s IP address can be found by navigating to [**Settings**] >> [**Status**] on the phone.

Figure 9 - Web Login Page

Open a web browser, enter the phone’s IP address, and access the phone’s web interface. The login page will be displayed first. Users must enter the correct username and password to log in. **The default username and password are both "admin"**. For detailed instructions on operating the web interface, please refer to section 17 Web Configurations.

6.7 Network Settings

The phone supports two network connection methods: wired network connection and wireless network connection. Users need to select the appropriate connection method based on their actual needs. This section describes the wired network connection method; for wireless network connection, refer to section 14.6.10 Wi-Fi.

The phone provides services via an IP network connection. Unlike traditional phones based on circuit-switched technology, IP phones connect to each other and exchange data packets over the network using their unique IP addresses.

To activate the phone, the network settings must first be configured correctly. To configure the network, users need to access the [**Apps**] >> [**Settings**] >> [**Network & Internet**] menu via the phone’s function menu button.

Note! If the user sees a "Network Disconnected" icon at the top of the screen, it means the network cable is not connected to the phone’s network port. Please check that the network cable is properly connected between the phone and the network switch, router, or modem.

The network type supported by the phone is as follows: IPv4.
There are two common IP configuration types for IPv4:

- **DHCP** – This is a mode that automatically obtains network configurations from a server. Users do not need to manually configure any parameters. It is suitable for most users.

- **Static IP Configuration** – This option allows users to manually configure each IP parameter, including IP address, Subnet Mask, Gateway and DNS server. This is usually suitable for environments with professional network users.

The phone’s default network configuration is DHCP mode.

6.8 SIP Configuration

The phone can provide call services only when at least one line has been correctly configured. The function of line configuration is similar to that of a SIM card in a mobile phone—both are used to store service provider information and account authentication data, with the key difference that line configuration is virtualized. When the phone applies these configurations, it will automatically register with the stored service provider, just as you can insert a SIM card into any mobile phone, and the phone will access services based on the information in the SIM card rather than the phone itself.

Users can configure lines via either the phone local interface or the web management interface. Enter the corresponding information in the fields of Registration Address, Registration Username, Registration Password, SIP User, Display Name, and Registration Port—these details are provided by the SIP server administrator.

Phone Interface: To manually configure a line, users can navigate to [**Settings**] >> [**Advanced Settings**] >> [**SIP Settings**] via the function menu button to configure each line, as shown in the figure below.

Note! Users must pass password authentication to access SIP Settings. The default PIN is "admin".

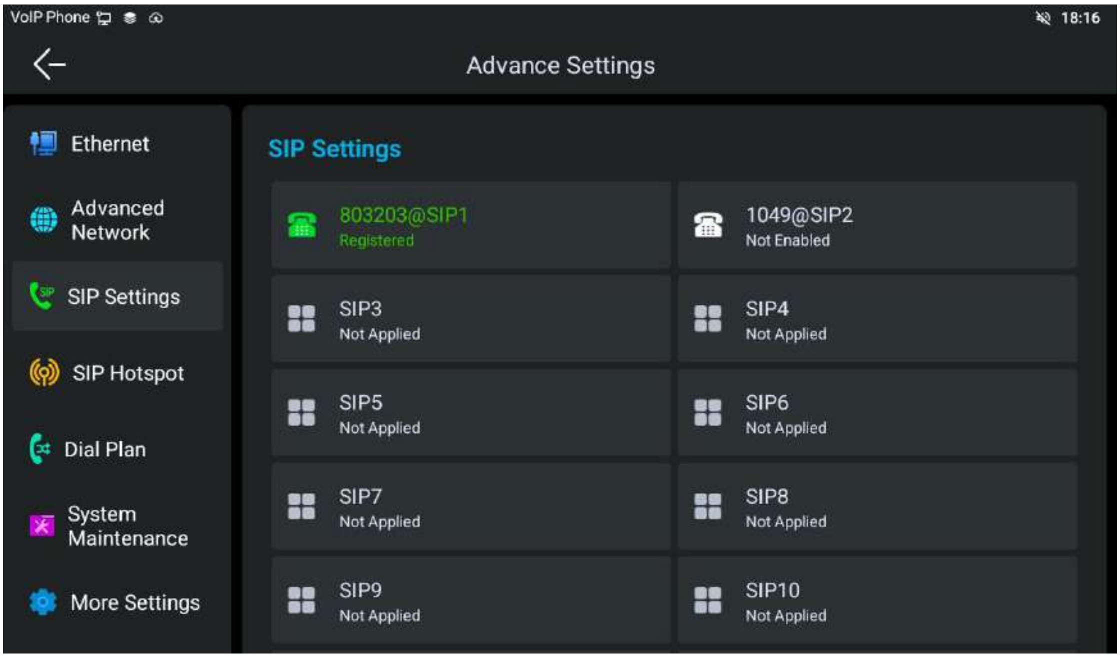


Figure 10 - Phone Line SIP Settings Interface

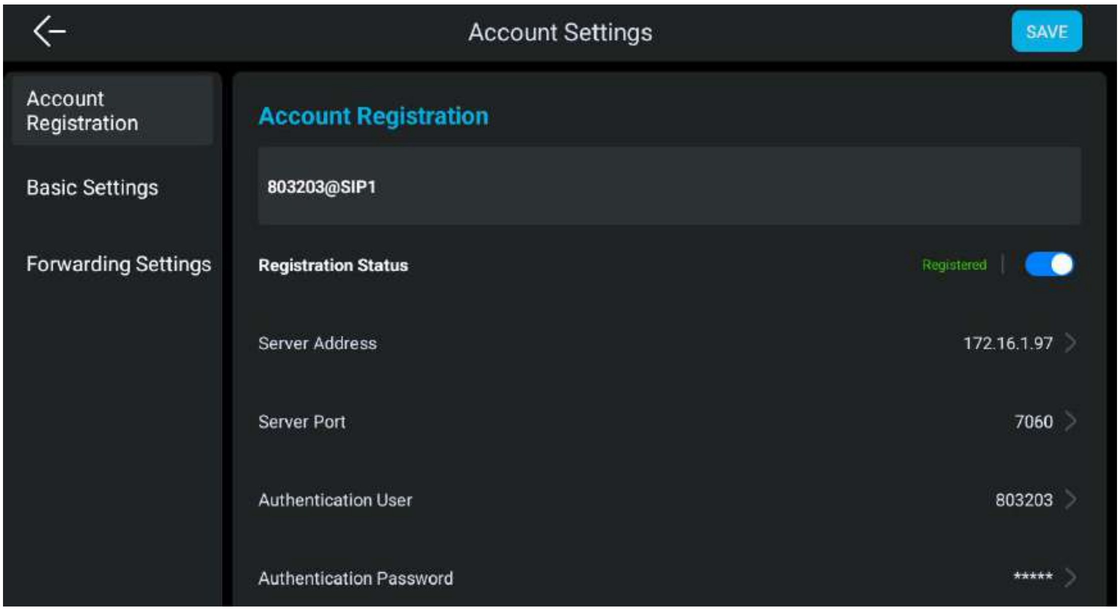


Figure 11 - Display Name and Port

Web Interface: After logging in to the phone's web interface, navigate to [**Line**] >> [**SIP**], select the SIP line to configure, and click Submit to complete registration after configuration, as shown in the figure below.

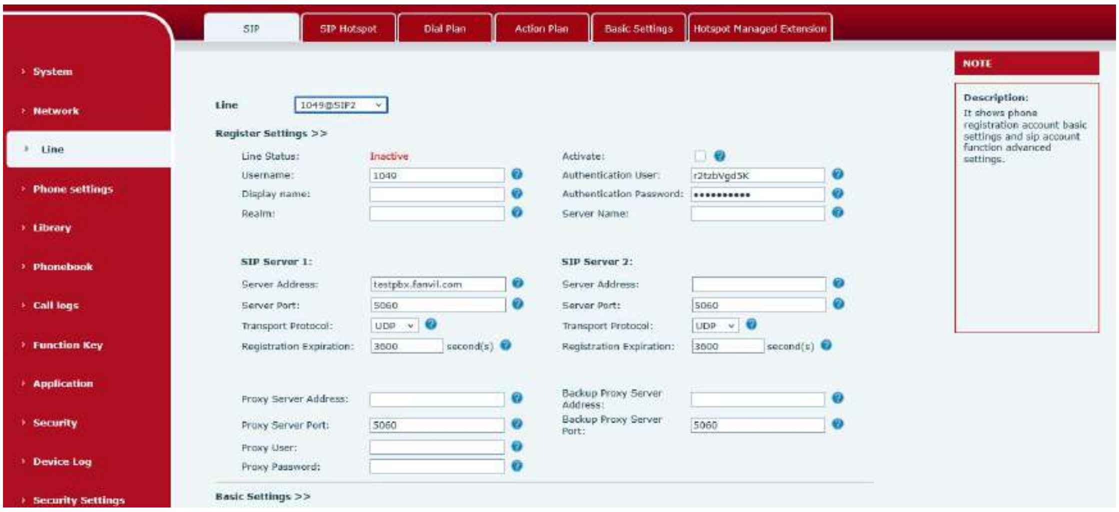


Figure 12 - SIP Settings on Web

7 Broadcast Configuration Process

Fanvil Broadcast & Intercom System supports automatic discovery and configuration of extensions within the same network segment, so that it can be used to build a small PA system quickly.

How to deploy

Step 1：Determine if there is a DHCP server in the network? If not, modify the paging phone (like A320i) network to static IP address mode

Step 2：The terminal device is connected to the same network and powered up

Step 3：The scanning terminal in the automatic network of the broadcast intercom system and the automatic configuration

Step 4：For management convenience, you can manually customize the name of the device

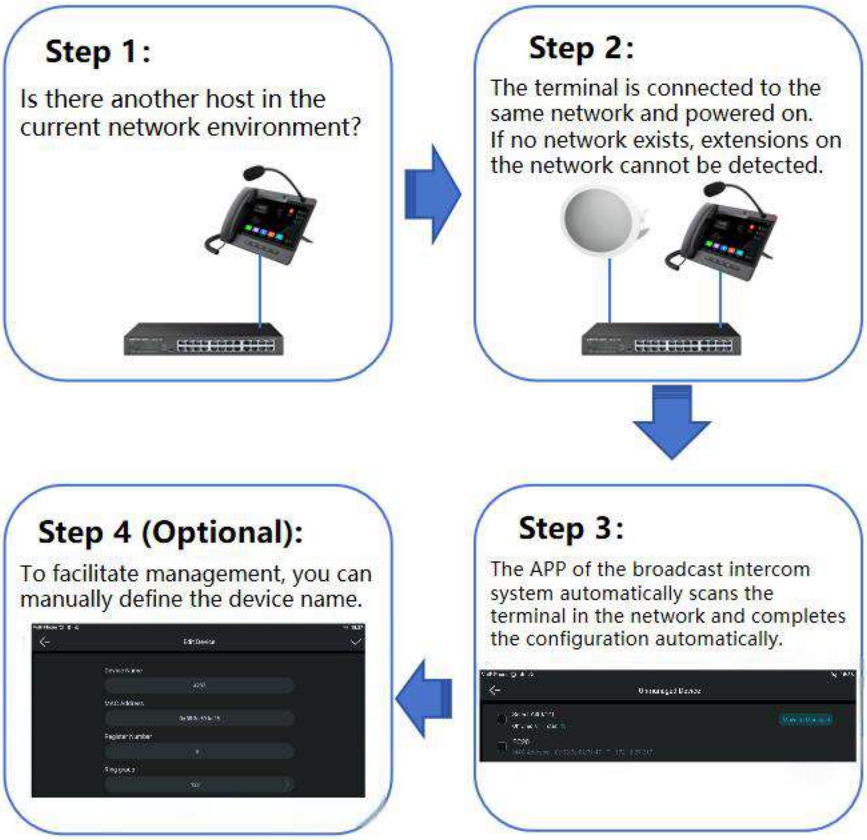


Figure 13 - Configuration Process

7.1 Discover and Configure Extension Devices Automatically

1. After the device is powered on, it will automatically scan and configure the extensions on the current network segment. When the host detects the extensions, a prompt box will pop up as shown in Figure 15. Users can click the box to view the automatically scanned extensions.

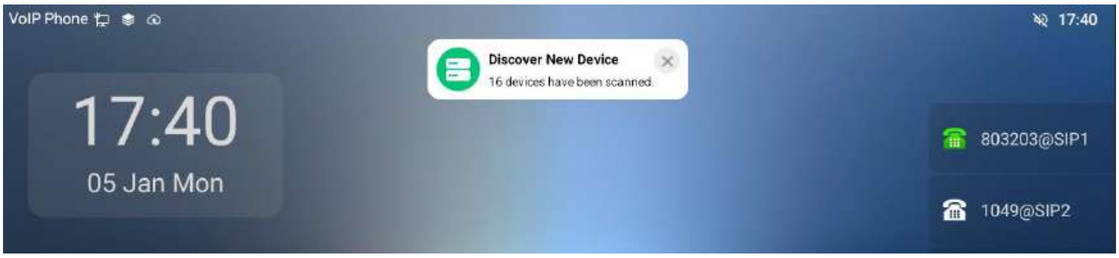


Figure 14 - Auto-Scan Prompt Box

2. Navigate to the Scanned Devices interface, where users can view the model, MAC address, and IP address of the scannable devices, as well as the management and deployment status. If unmanaged devices appear in the list, users can select the device(s) and click "**Move to Managed**" in the upper right corner to add them to the management and deployment.

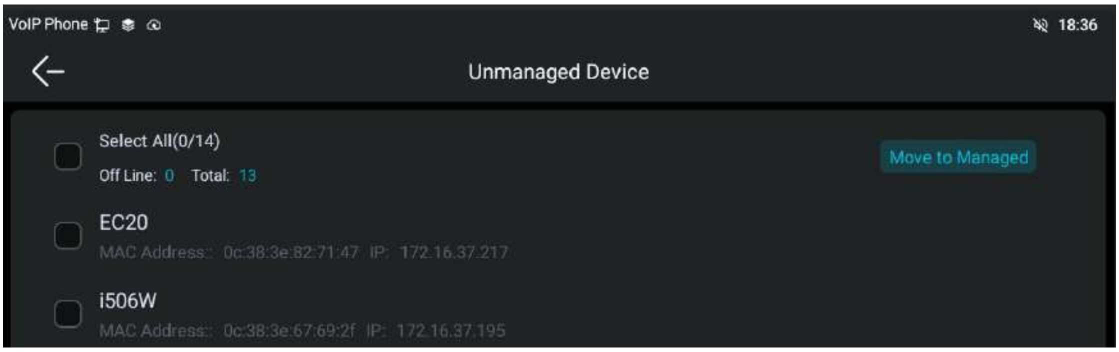


Figure 15 - Scanned Devices

7.2 Configure Extension Information

After the host has automatically scanned and configured the extensions, users can navigate to [**Settings**] >> [**Broadcast Settings**] >> [**Extension Management**]. Click Icon ① on the right of a device in the list to display the detailed information of the extension device. Then click Edit to customize the device name, extension number and ring group information.

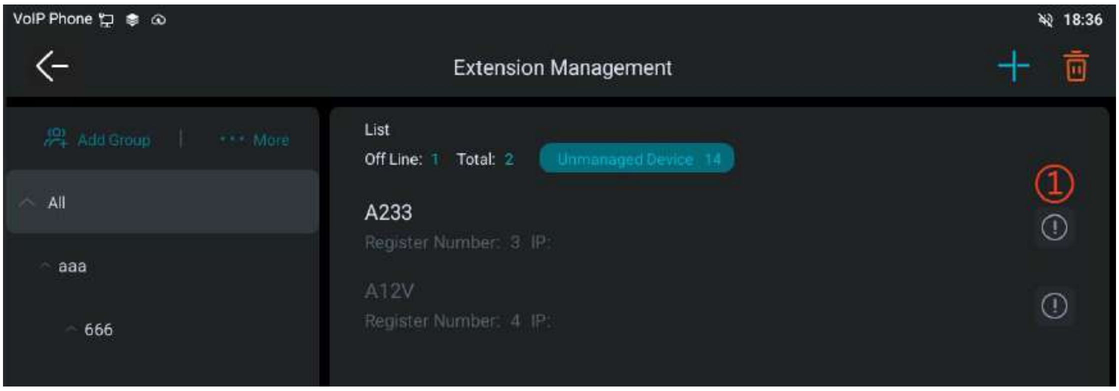


Figure 16 - Extension Management

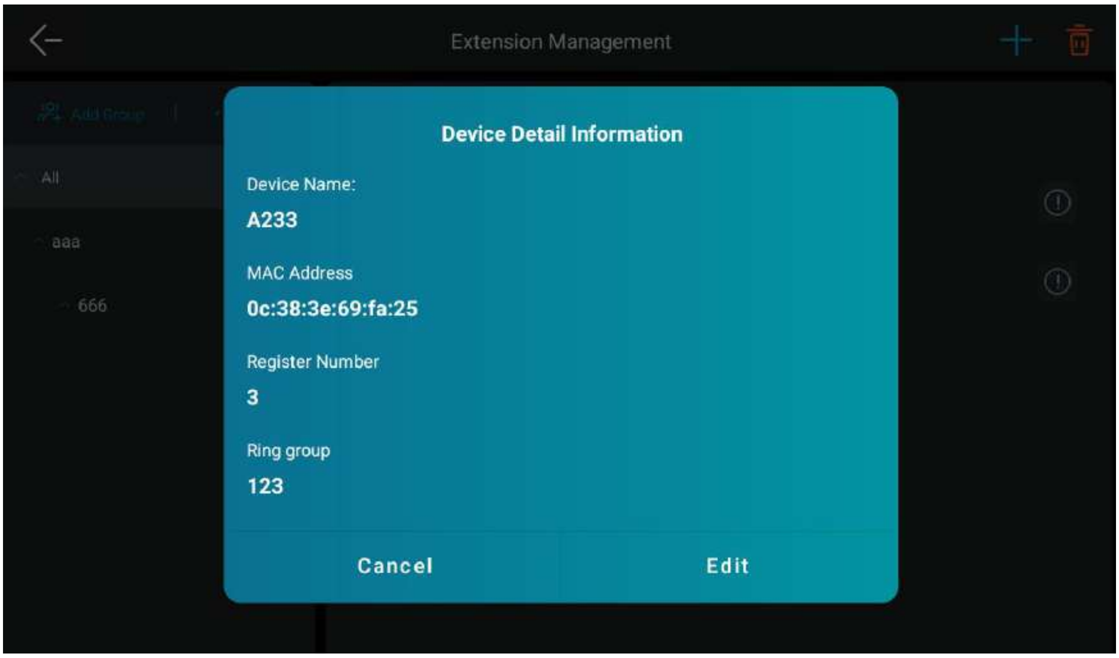


Figure 17 - Device Detail Information

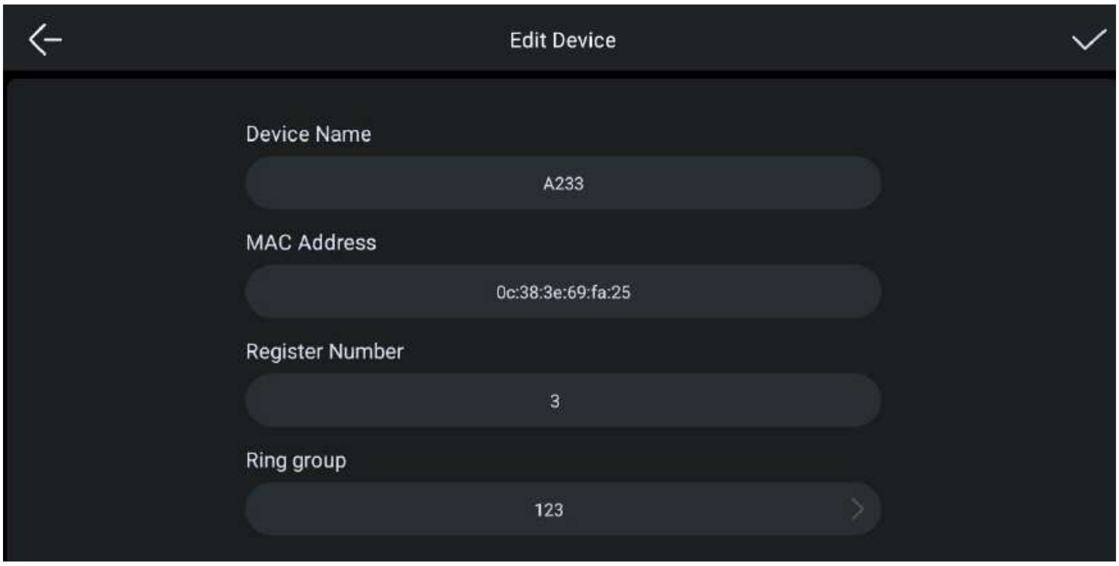


Figure 18 - Edit Device

8 Basic Functions

8.1 Making Phone Calls

- **Default line**

The phone supports up to 20 SIP line services. If all 20 lines are configured successfully, users can make or receive calls using any of them. If no specific line is selected for an outgoing call, the call will be placed via the default line (the default line can be customized). The detailed steps are as follows:

Navigate to [**Phone Settings**] >> [**Feature**] >> [**Basic Settings**] on the web interface, tick "**Enable Default Line**", and select the default line in the "**Default Ext Line**" configuration.

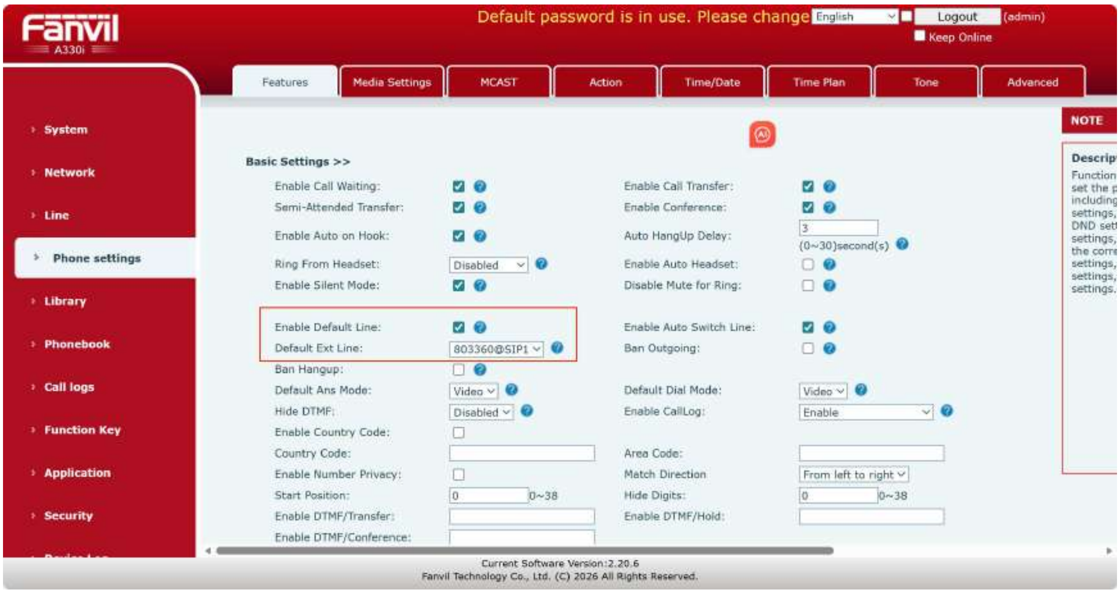


Figure 19 - Default Line

- **Dialing method**

Users can dial a number by,

- Go to the [**Call**] >> [**Dial**] interface and entering the number directly
- Selecting a phone number from phonebook contacts (Refer to 10.2.1 Local Contact)
- Selecting a phone number from cloud phonebook contacts (Refer to 10.2.3 Network Phonebook)
- Selecting a phone number from call logs (Refer to 10.3 Call Log)
- Redialing the last dialed number

- **Dial Number then Open Audio**

When making an outgoing call, users can dial the number using one of the methods described above. After confirming the number is correct, users can press the [**Audio**] or [**Video**] button, press the hand-free button (to enable the speaker or headset), lift the handset, or press a line key (configured via dsskeys) to select a line directly and start the call.

- **Open Audio then Dial the Number**

The other method is the traditional approach: first activate the phone's voice channel (by lifting the handset/pressing the hand-free button/switching to headset mode/pressing a line key), then dial the number. Once dialing is completed, users can press the [**Audio**] or [**Video**] button to place the call.

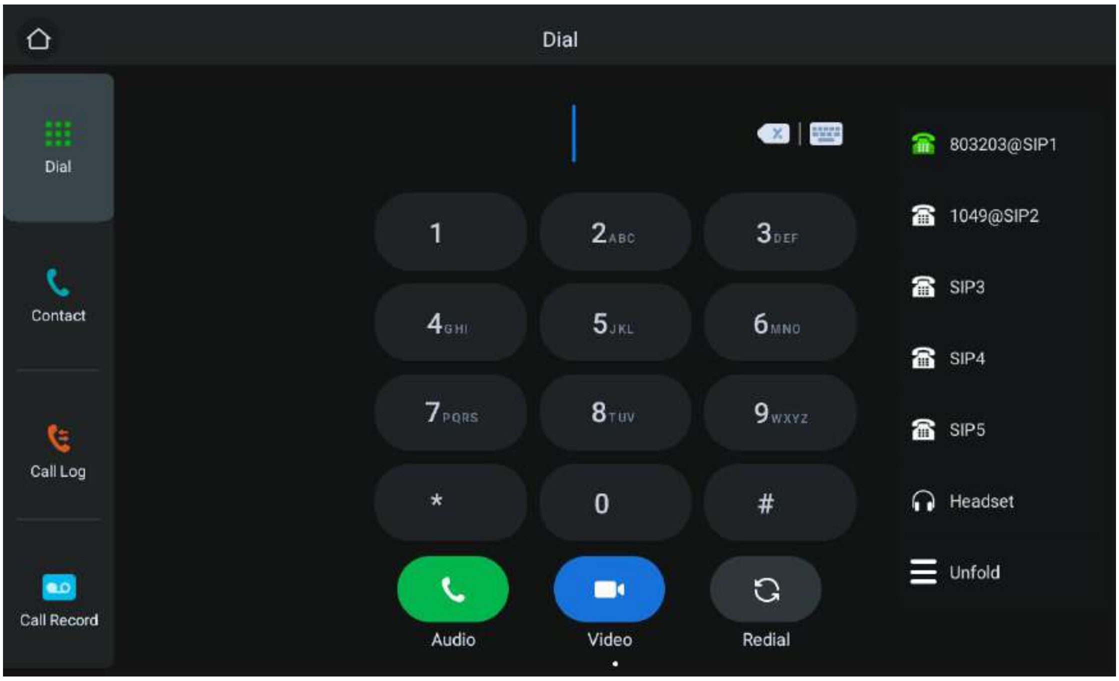


Figure 20 - Open the Voice Channel and Dial the Number

- **Cancel the call**

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

Figure 21 - Call Number

8.2 Answering Calls

When the phone is idle and there is a call, the user will see the call reminder screen as belowed.

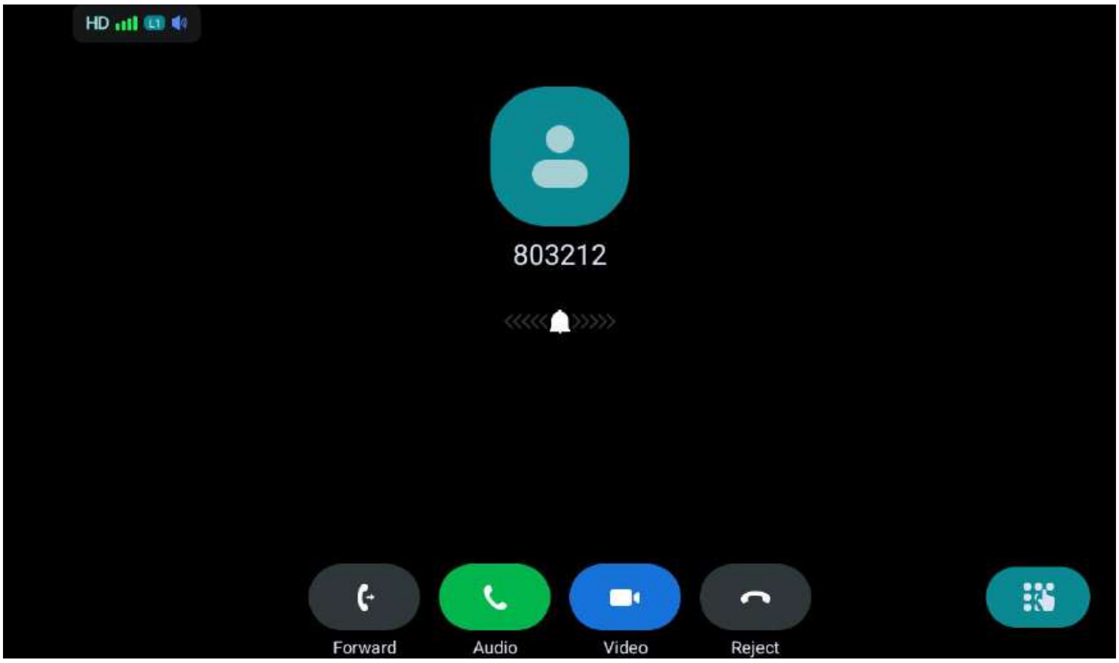


Figure 22 - Answering Calls

Users can answer a call by lifting the handset, pressing the [**Audio**] or [**Video**] button, or pressing the hand-free button to activate the handset/headset/speakerphone channel. To reject an incoming call, users should press the [**Reject**] button.

8.2.1 Talking

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

Figure 23 - Talking Interface

Table 6 - Talking Interface

Index	Name	Description
①	The current line	The line currently used by the phone.
②	User avatar	The avatar displayed by default.
③	Calls to end	The name or number of the person on the other end of the call.
④	Call duration	The duration of a call after it has been established.
⑤	Video icon	Click to initiate video call.

8.2.2 Make / Receive the Second Call

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

- **The Second Incoming Call**

When there is another incoming call during an active 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 red. 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.

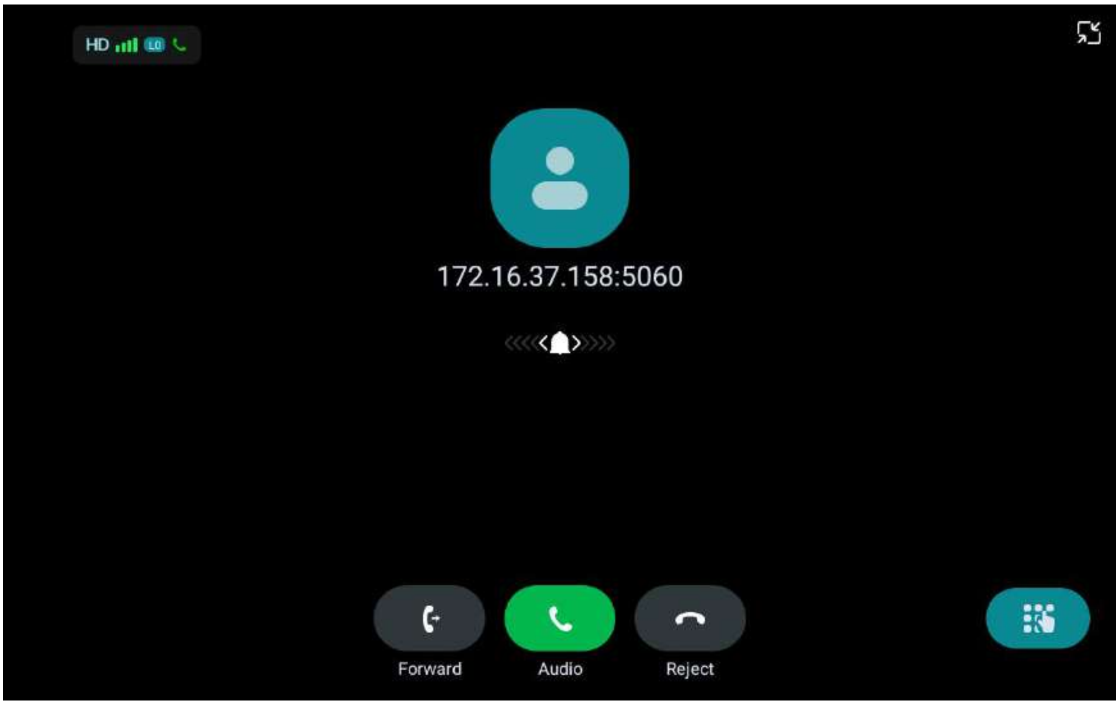


Figure 24 - The Second Call Interface

- **Second Outgoing Call**

To make a second call, user may press [**Transfer**] or [**More**]>>[**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 press dsskeys 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.

- **Switch between two calls**

When there are two calls established, user will see a dual calls screen as the following picture.

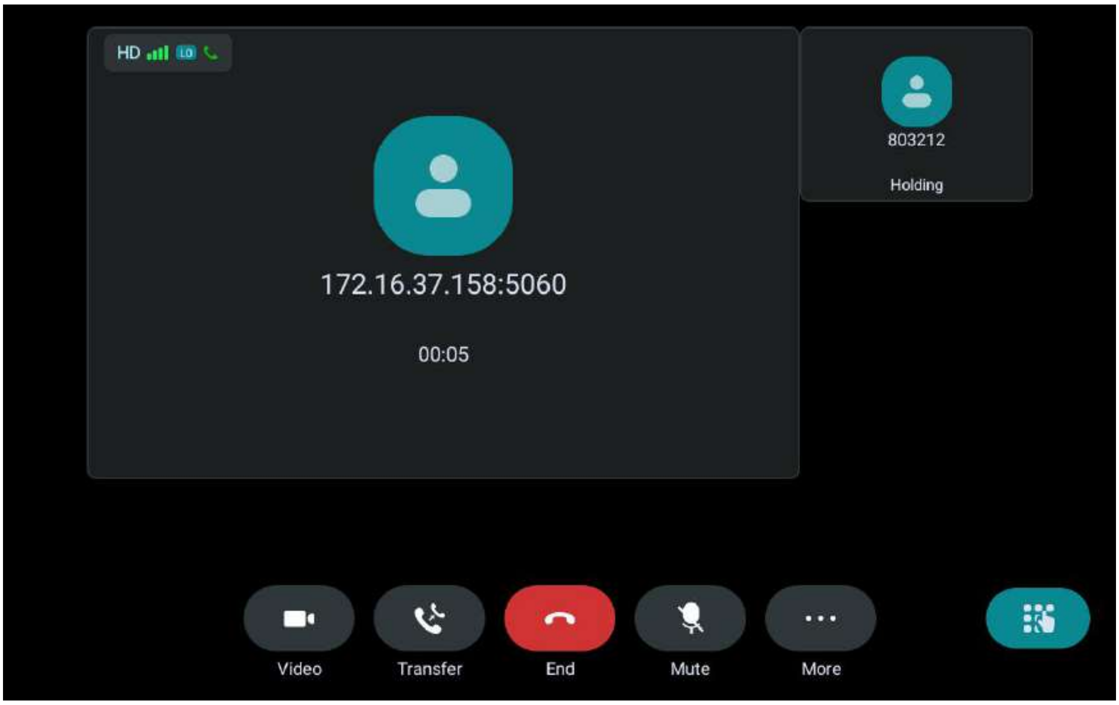


Figure 25 - Two Way Calling

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

- **Ending one call**

Users can end the current call by pressing the [**End**] key. The phone will then return to the hold state in single-call mode. Users can also resume the current call by pressing the [**Resume**] key.

8.3 End of 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.

Note: When the phone is on hold, the user must press the 【Resume】 key to return to the call state before ending the call by placing the handset back on the phone.

8.4 Video Call

A330i&A330 support a variety of video formats QVGA,CIF,VGA,4CIF,720P,1080P.

- The default dialing mode is video. When the device dials, it uses video mode to call out by default. If the end device supports sending video, both sides establish a video call.
- The default dialing mode is voice. The above operation establishes voice call.Users can press the [**Video**] button to send a video call request.

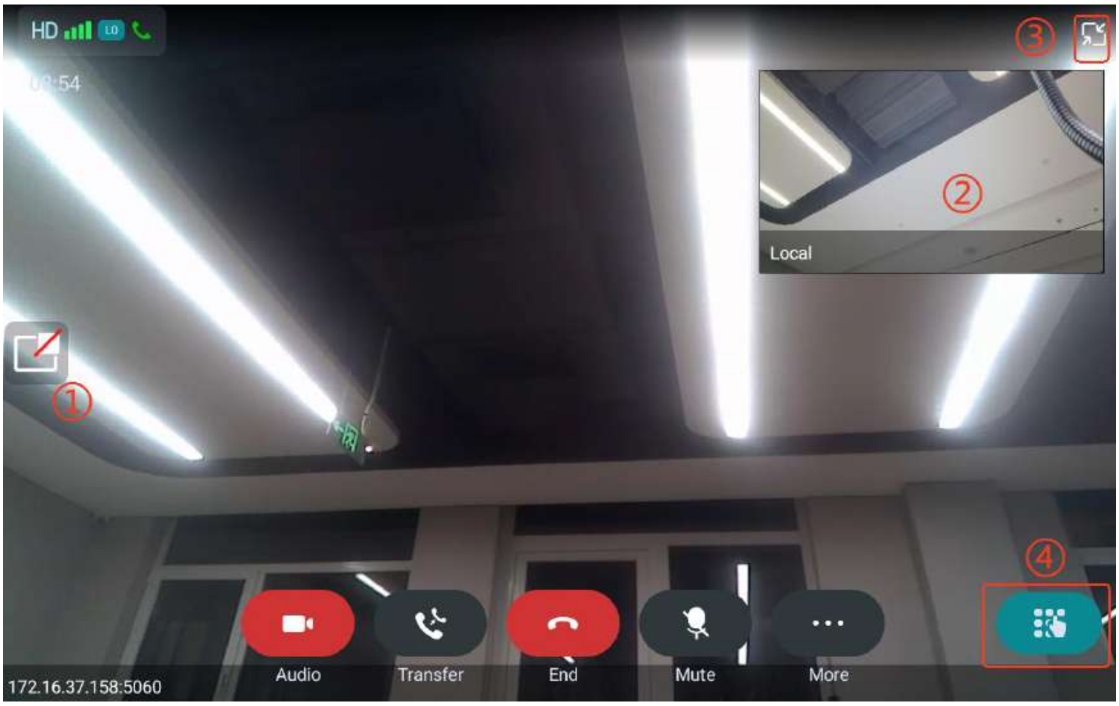


Figure 26 - Video Interface

Table 7 - Video Interface

Index	Function
①	Click this button to hide/display the small window video screen in the upper right corner.
②	Click on this area to swap the display positions of the local and remote videos on the interface.
③	Click to exit the full-screen mode.
④	Click this button to expand the function key list.

WEB interface: enter [**Phone Settings**] >> [**Features**] >> [**Basic Settings**], and choose to configure the “Default Dial Mode” and “Default Ans Mode”.

Figure 27 - Video Settings

8.5 Redial

- Redial the last outgoing number:

  Configure the shortcut key as redial key. When the phone is in standby mode, click the [**Unfold**] button on the right to expand the list of shortcut keys. Then press the redial button to call out the last dialed number.

- Call out any number with the redial key:

  Enter the number when the phone is off-hook, press the redial button, and the phone will call out the number on the dial pad.

- 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.

Figure 28 - Redial Set

8.6 Dial-up Query

Phone is defaulted to open the dial-up inquiry function, dial-out, enter two or more Numbers, dial the interface will automatically match call records, contacts in the number list.Click on the corresponding number, and press the [**Audio**] or [**Video**] button to call out.

8.7 Auto-Answering

User may enable the 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 web interface.

- **Phone interface:**

Press [**Settings**] >> [**Advanced Settings**] >> [**SIP Settings**].;

Select the corresponding line, then in [**Basic Settings**], you can enable/disable the auto-answer option and set the auto-answer delay time (default: 5 seconds). After making changes, click the Save button in the upper right corner.

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

- **Web interface:**

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

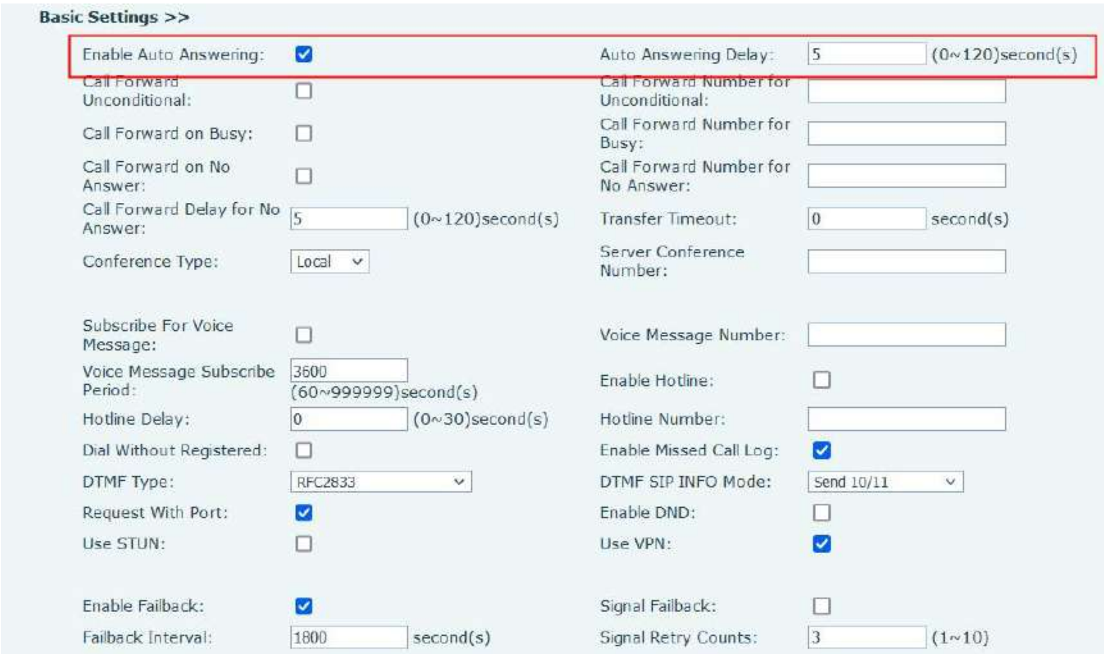


Figure 29 - Enable Auto-answering on Web

8.8 Call Back

The user can dial back the last call. If there is no call history, press the [**Call back**] button and the phone will say "can't process".

- Set the callback key through the phone interface:

  In standby mode, click the [**Unfold**] button, then long-press the function key to be set—this will automatically open the configuration interface. Select Type as key event type, subtype as call back, you can set the call back key name in the title input box, and finally press the [**SAVE**] button to save the settings.

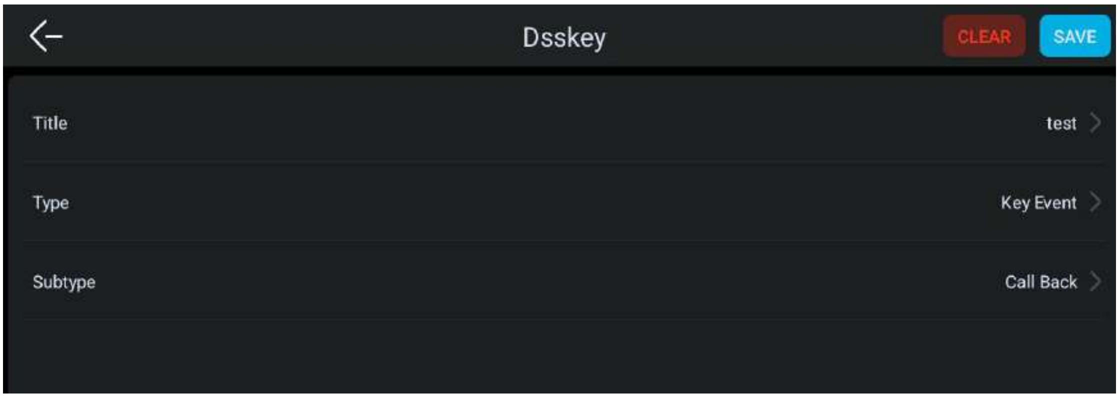


Figure 30 - Set the Call Back Key

- Set the callback key through the web interface:

  Log in the phone page, enter the [**Function Key**] >> [**Function Key**] page, select the function Key, set the type as the function Key, and set the subtype as the call back, as shown in the figure:

Figure 31 - Set the Call Back Key on Web

8.9 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 will be automatically turned off at the end of a call. You can also turn on mute on any screen (such as the Idle screen) and mute the ringtone automatically when there is an incoming call.

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

8.9.1 Mute the Call

- During the conversation, press [**mute**] button  on the phone:

The softkey switches to the red Unmute icon, and the mute indicator is displayed in the upper left corner of the call interface.

Mute icon is displayed in the call interface, as shown in the figure:

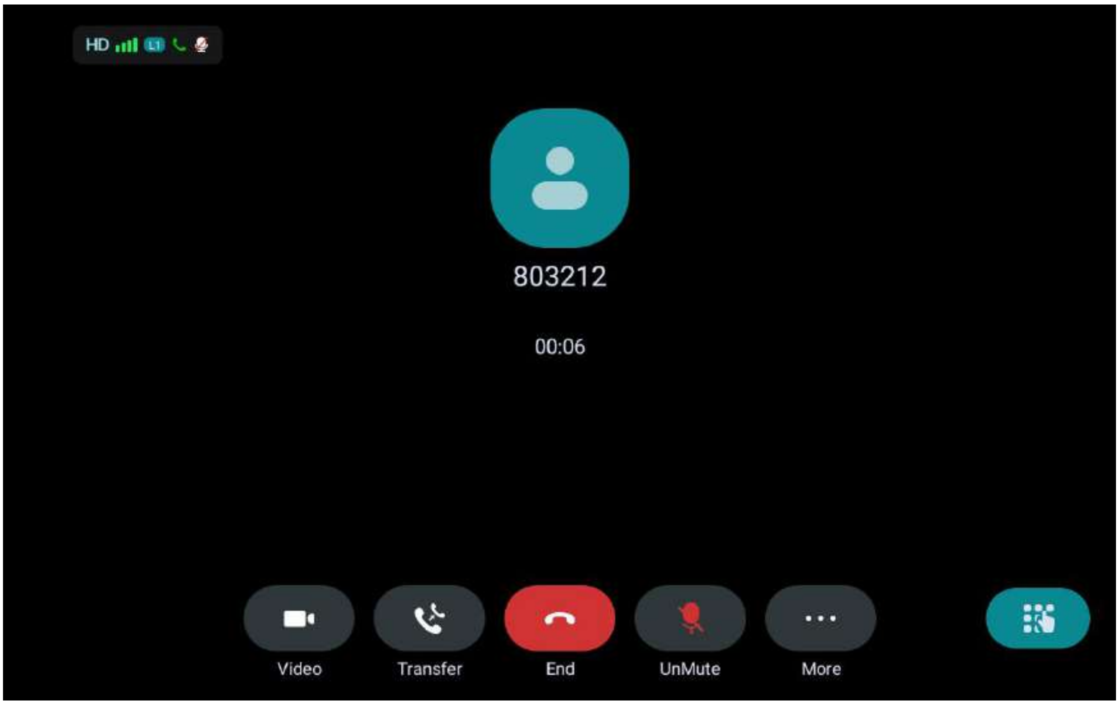


Figure 32 - Mute the Call

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

8.9.2 Ringing Mute

- **Enable Ringing Silent Mode**: Press the volume down key to set the volume to 0 when the phone is in standby mode. You can also enable silent mode or adjust “**Ring Volume**” to 0 in [**Settings**] >> [**System Settings**] >> [**Volume**].

- The ringer silent icon will be displayed in the upper right corner of the phone. When there is an incoming call, the phone will show the incoming call interface but will not ring.

- **Disable Ringing Silent Mode**: Press the volume up key to disable ringing silent mode in idle screen or on the incoming call interface. After disabling, the silent icon will no longer be displayed in the upper right corner.

8.10 Call Hold/Resume

The user can press the [**More**] >> [**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.

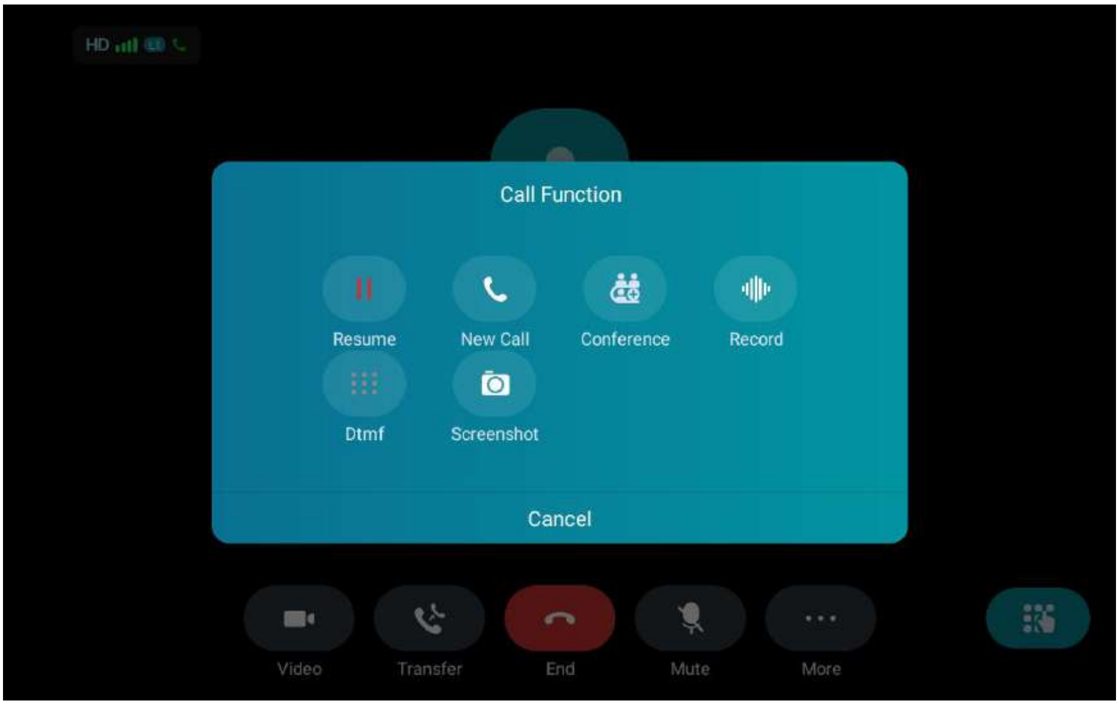


Figure 33 - Call Hold

8.11 DND

Users may enable the Do-Not-Disturb (DND) feature on the device to reject incoming calls (including call waiting). Do Not Disturb (DND) can be enabled or disabled for each SIP line individually.

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

- **Phone interface**：Default standby mode,
  1) In [**Settings**]>> [**System Settings**]>> [**Phone DND**] interface. Set the “**Do Not Disturb Mode**” configuration to “**Phone**”, click “**SAVE**” in the upper right corner, and the DND icon  will appear in the status prompt bar at the top of the phone.
  2) Set the ”**Do Not Disturb Mode**" configuration to “**off**”, and the DND icon in the status prompt bar at the top of the phone disappears.

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

1) In [**Settings**]>> [**System Settings**]>> [**Phone DND**] interface. Set the “**Do Not Disturb Mode**” configuration to “**Lines**”, and after selecting the line you want to turn on do not disturb, click “**SAVE**” in the upper right corner.
2) A DND icon will appear in the status prompt bar at the top of the phone.

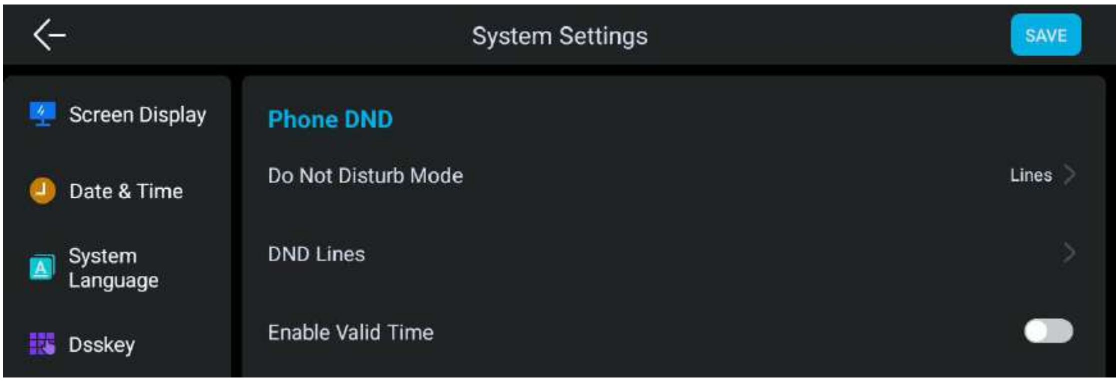


Figure 34 - DND Setting

The user can also use the DND timer. After enabling the valid time range, the Do Not Disturb function will be automatically activated within the time range.

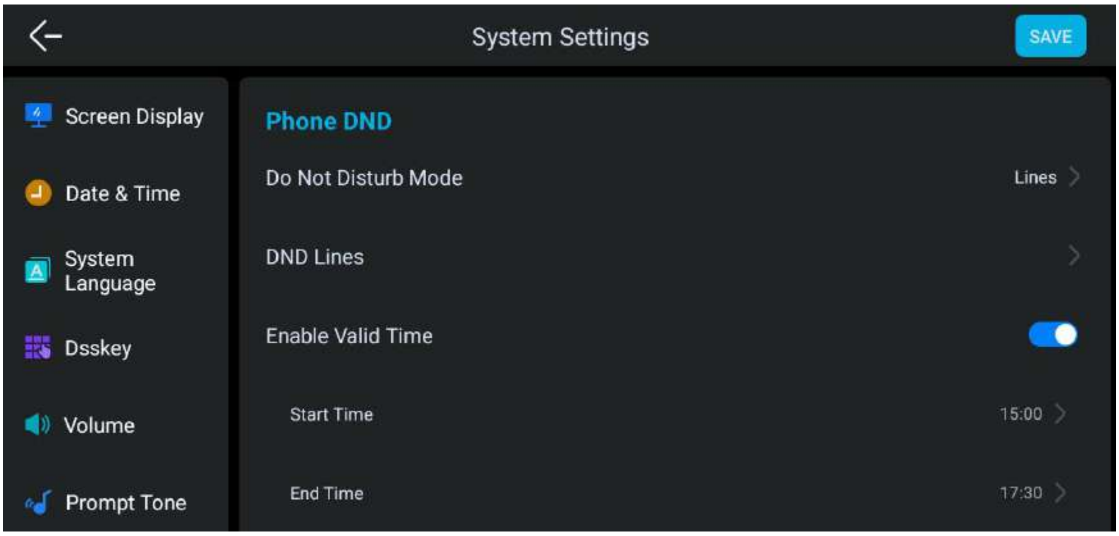


Figure 35 - DND Timer

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

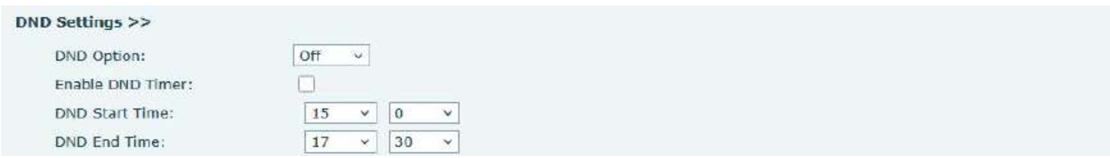


Figure 36 - DND Settings on Web

The user turns on the DND for a specific route on the web page：Enter [**Line**] >> [**SIP**], select a line and click [**Basic Settings**] , tick “**Enable DND**”.

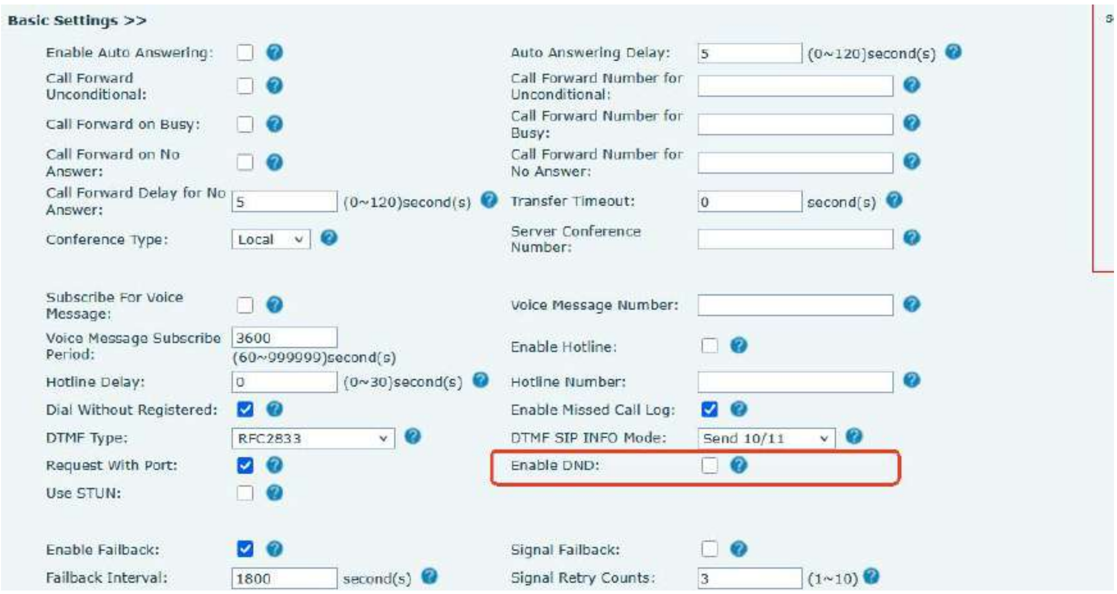


Figure 37 - Line DND

8.12 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) Go to [**Settings**]>>[**Advanced Settings**]>> [**SIP Settings**], click on corresponding line,and enter the [**Forwarding Settings**] interface.

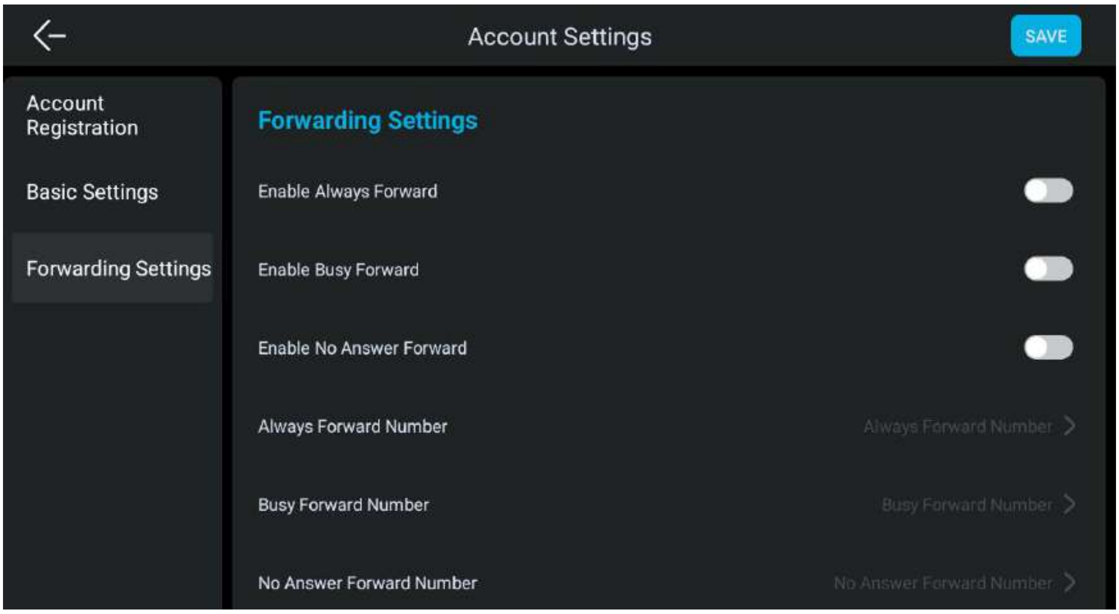


Figure 38 - Call Forward Settings

2) Select Enable/Disable by tapping the toggle switch.
3) Configure the parameters and enter the forward number. After completion, press the [Save] button to save the changes made.

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

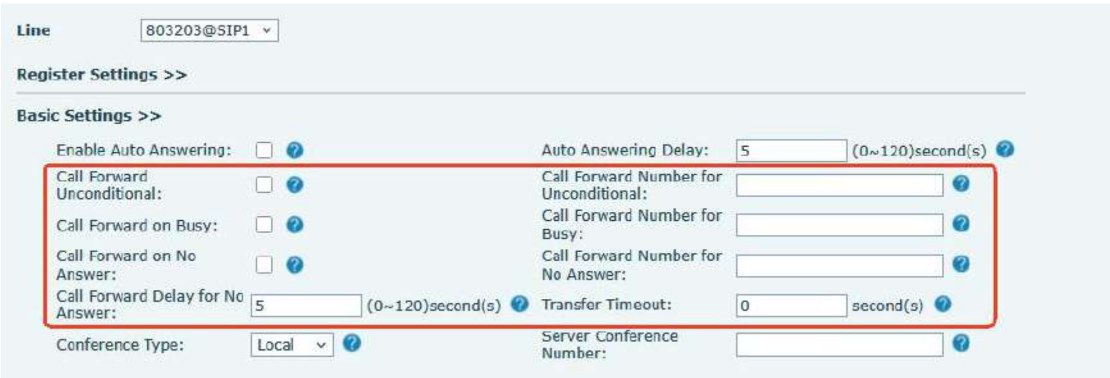


Figure 39 - Call Forward Settings on Web

8.13 Call Transfer

When the user is talking with a remote party and wish to transfer the call to another remote party, there are three ways 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 [17.15 Line >> Dial Plan]。

8.13.1 Blind Transfer

During a call, the user can press the [**Transfer**] button, then enters the target transfer number, or presses the [**Contact**] or [**Call Log**] button to select a number. Press the [**Transfer**] button again to perform a blind transfer to the third party. After the third party's phone rings, the phone will display "Transfer Success" and hang up automatically.

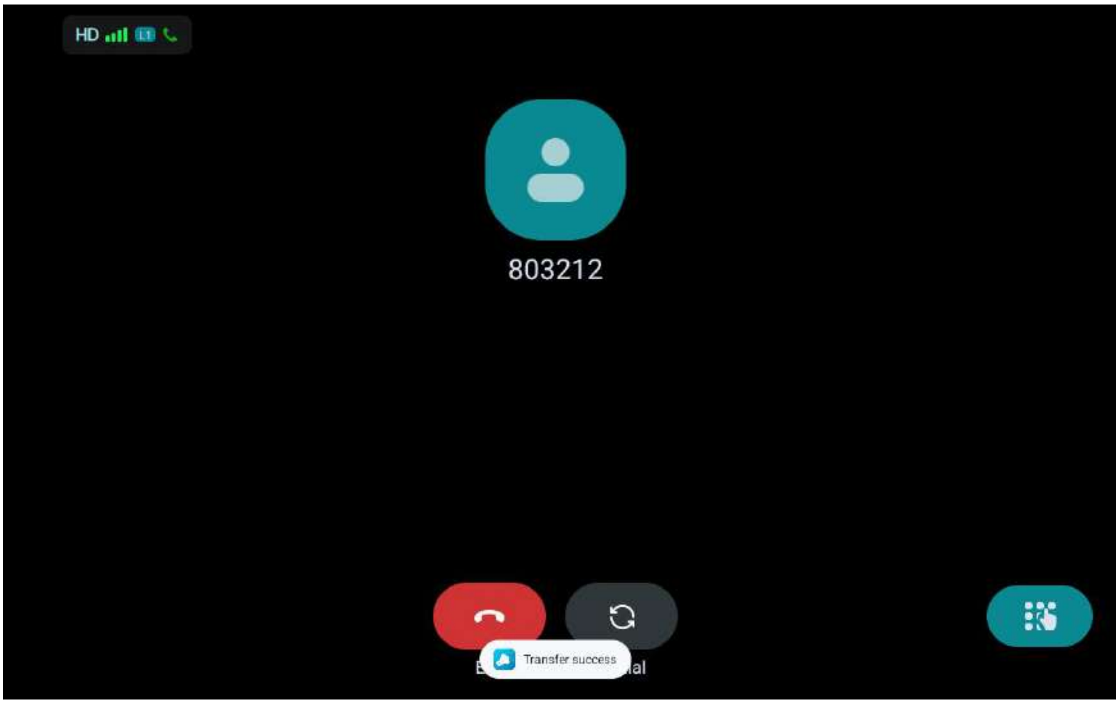


Figure 40 - Blind Transfer

8.13.2 Semi-Attended Transfer

During the call, the user presses the function menu button [**Transfer**] to input the number to be transferred or press [**Contact**] or [**Call Log**] button to select the number, and then press the [**Audio**] or [**Video**] button. When the third party is not answered, press "**Transfer**" on the call interface to make the semi-attended transfer or press the end button to cancel the semi-attended transfer.

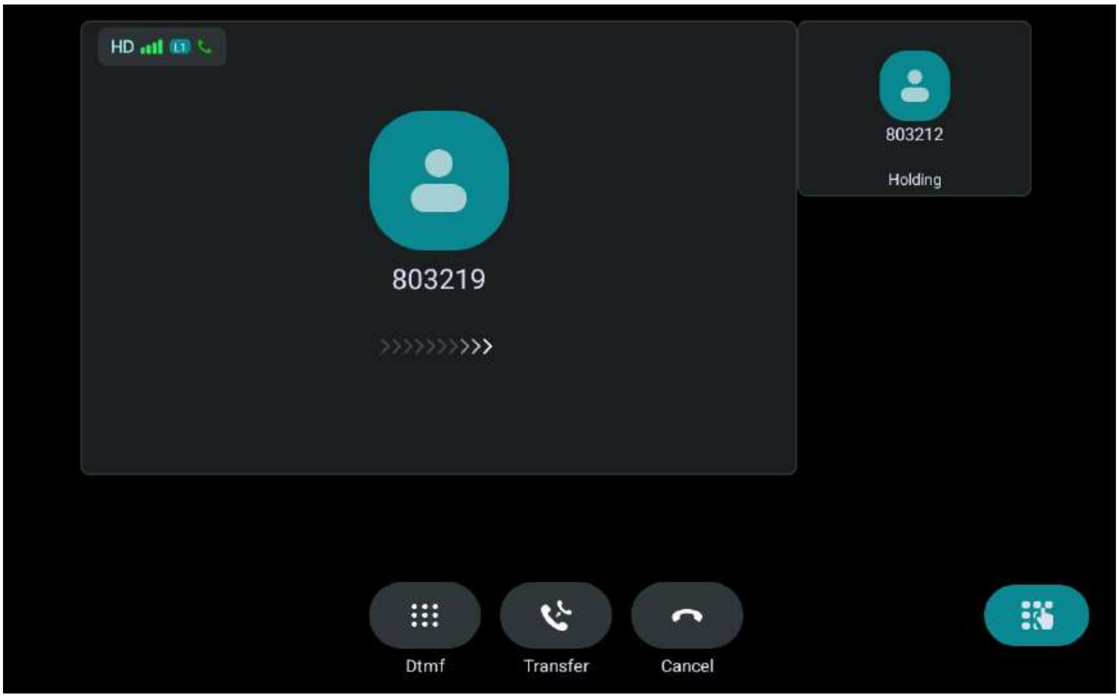


Figure 41 - Semi-Attended Transfer

8.13.3 Attended Transfer

Attended 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.

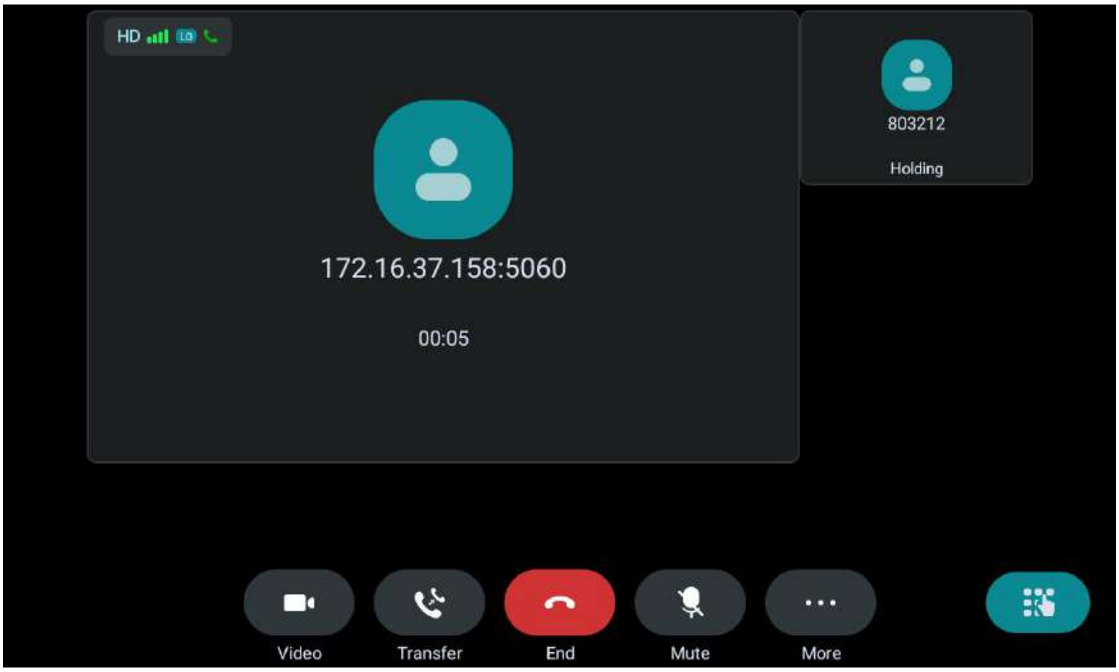


Figure 42 - Attended Transfer

8.14 Call Waiting

- **Enable call waiting:** new calls can be accepted during a call.
- **Disable call waiting:** During a call, 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 [**Settings**]>>[**System Settings**]>>[**Call Settings**], user can choose to ena ble/disable the call waiting configuration.

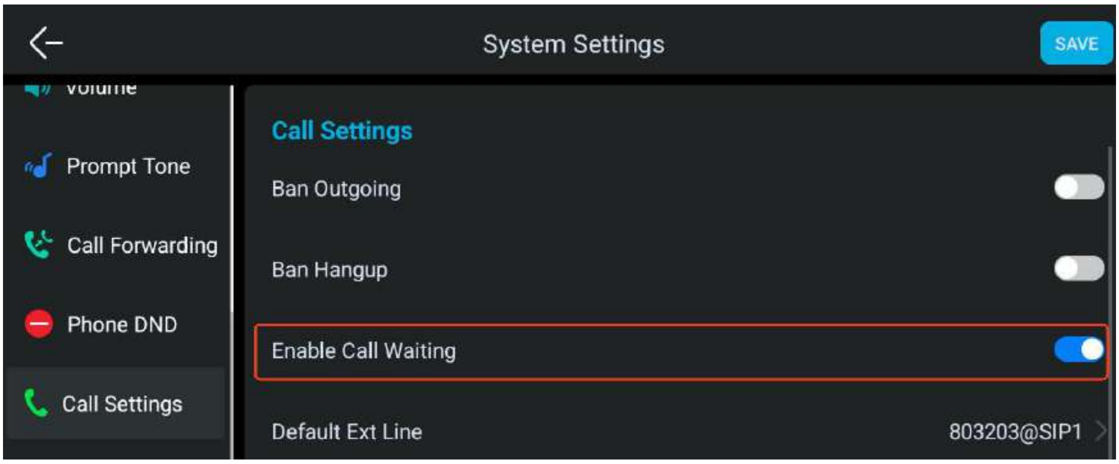


Figure 43 - Call Waiting Setting

Press [**Settings**]>>[**System Settings**]>>[**Prompt Tone**], you can turn on/off the call waiting tone.

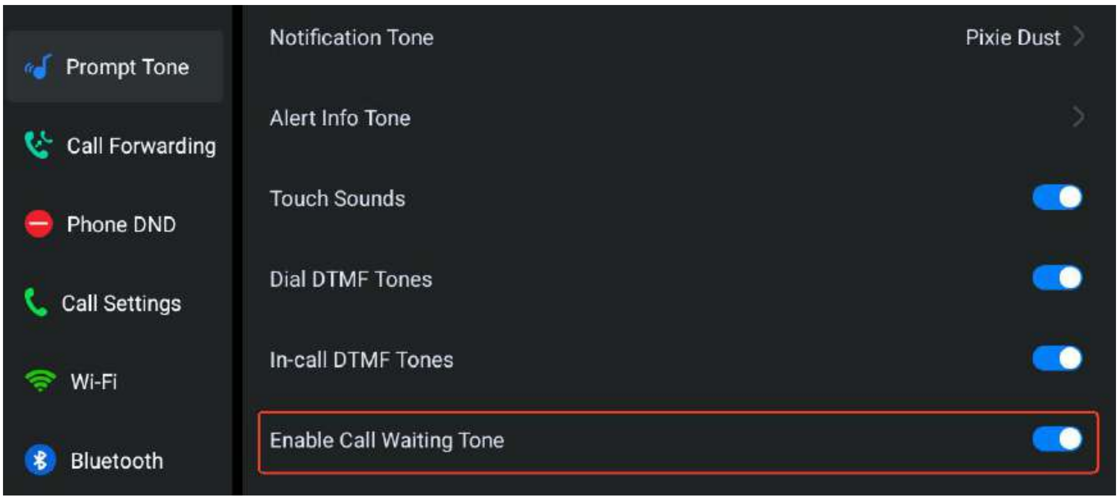


Figure 44 - Call Waiting Tone Setting

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

Figure 45 - Call Waiting Setting on Web

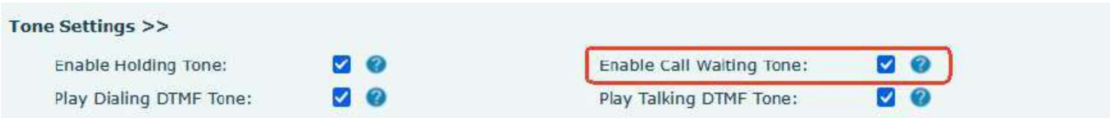


Figure 46 - Call Waiting Tone Setting on Web

8.15 Conference

8.15.1 Local Conference

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

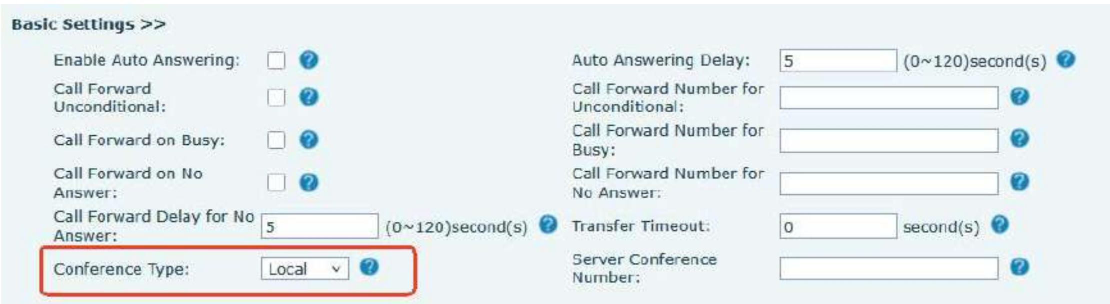


Figure 47 - Local Conference Setting

Two ways to create a local conference：

1） If the device has two channels of communication, press the [**More**] >> [**Conference**] button on the call interface. When selecting the conference number, select the other number that already exists.

Figure 48 - Local Conference(1)

2） If the device has a call all the way, press the conference key in the call interface, enter the number to join the meeting and press the call; After the opposite end is answered, press the conference button again to set up the local tripartite conference：

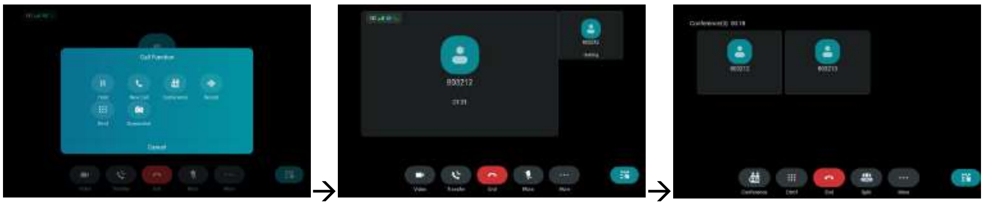


Figure 49 - Local Conference(2)

Note! During the conference, press the split button to split the conference and press the end button to end the cal.

8.15.2 Network Conference

Users need server support for network conference.

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

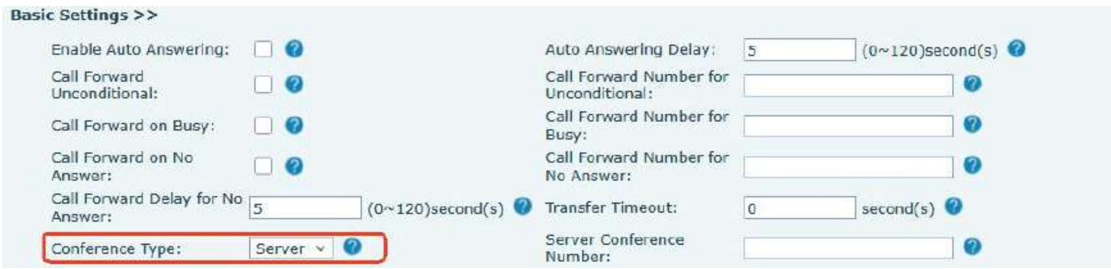


Figure 50 - Network Conference

Method to join a network conference：

- Call the numbers of network conference and when they enter the password then will 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 maximum number of participants in server meetings varies from server to server.

8.16 Record

The device supports recording during a call.

8.16.1 Local Record

When using local recording, you first need to enable recording in [**Application**] >> [**Manage Recording**] on the phone's web interface (enabled by default). Select Local as the recording type and set the voice codec. The supported voice codecs are PCMU and PCMA. The web interface is shown below:

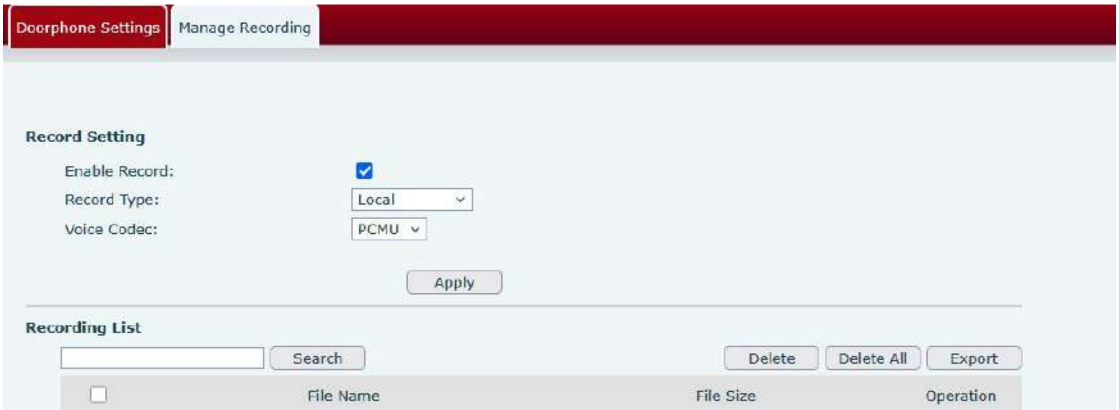


Figure 51 - Local Record

Local Recording Steps:

- Enable recording on the web interface and set the record type to Local.
- During a call, tap [**More**] >> [**Record**] button.

When recording is in progress, the Record button will display in red, and a recording icon  will appear in the upper left corner of the call interface. Recording will stop when you tap the [**Record**] button again or end the call. The recorded audio can be viewed and played back in [**Call Log**] and [**Call Record**] (please refer to 10.3 Call Log and 10.4 Call Record).

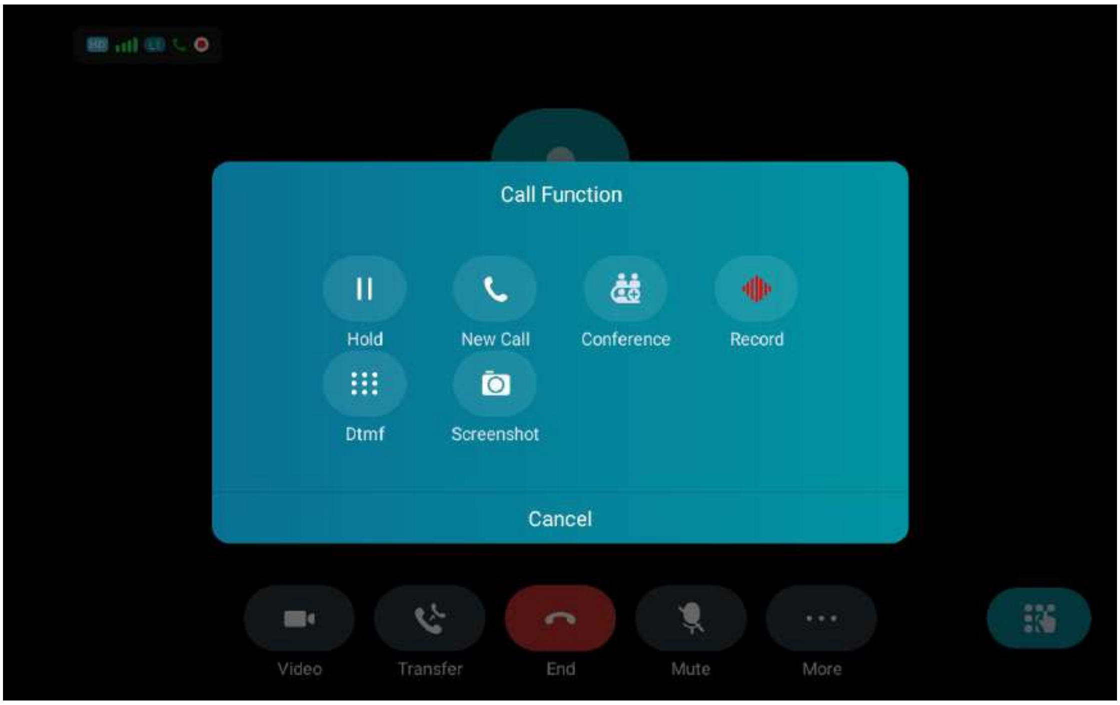


Figure 52 - Recording in Progress

8.16.2 Server Record

When using network server recording, you must first enable recording in [**Application**] >> [**Manage Recording**] on the phone's web interface. Select Network as the record type, enter the recording server address and port, and select the voice codec. The web interface is shown below:

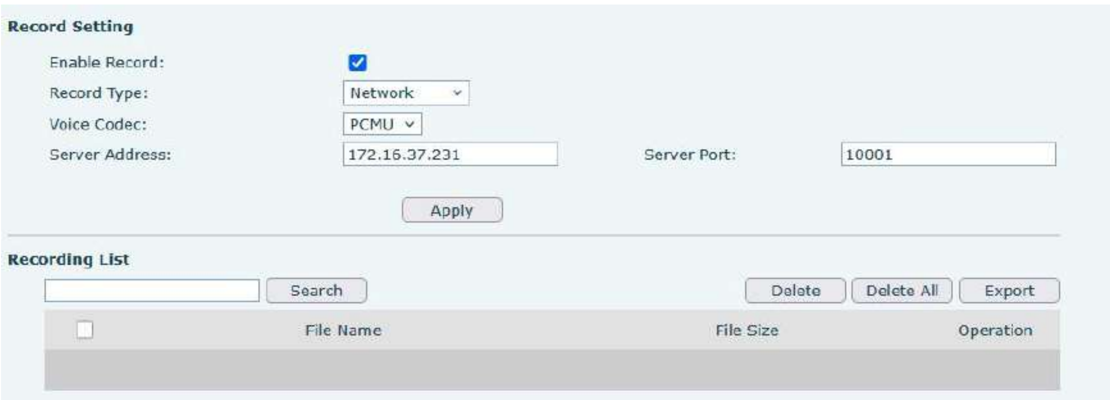


Figure 53 - Server Record

Note! This function must be used with recording software.

Please refer to the official document for specific usage instructions: **Call Recording Configuration and Use Description.**

8.16.3 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 record type is Sip info.

Please refer to the official document for specific usage instructions:**Call Recording Configuration and Use Description.**

Figure 54 - SIP Info Record

8.17 Capture

After the user establishes a video call, tap the [**More**] >> [**Capture**] button to save the remote video frame. The captured photos can be viewed in [**Call Log**] (please refer to 10.3 Call Log).

8.18 Screenshot

After the user establishes a video call, tap the [**More**] >> [**Screenshot**] button to save the local video frame. The screenshot photos can be viewed in [**Call Log**] (please refer to 10.3 Call Log).

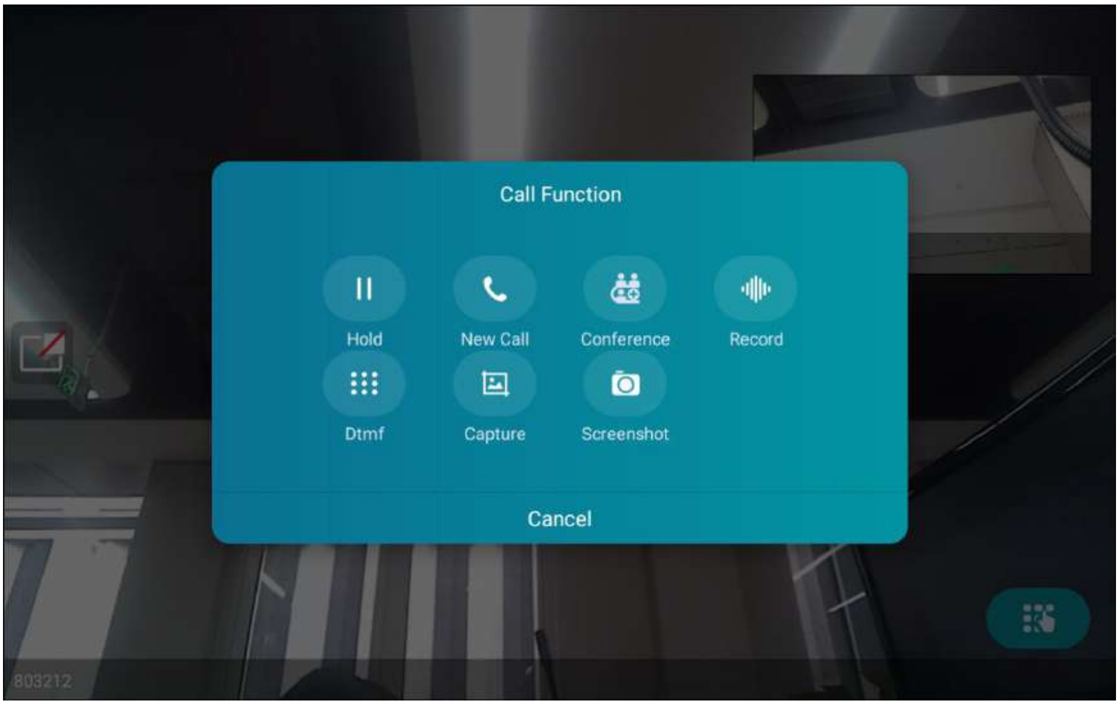


Figure 55 - Capture/Screenshot Button

8.19 Call Park

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

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

Set the call park button：

- **Phone interface:** In standby mode, click the unfold button and long press an editable key to enter the function key setting interface. Set function key type as memory and subtypes to call park, enter values for the server calls park number, set up corresponding SIP lines.

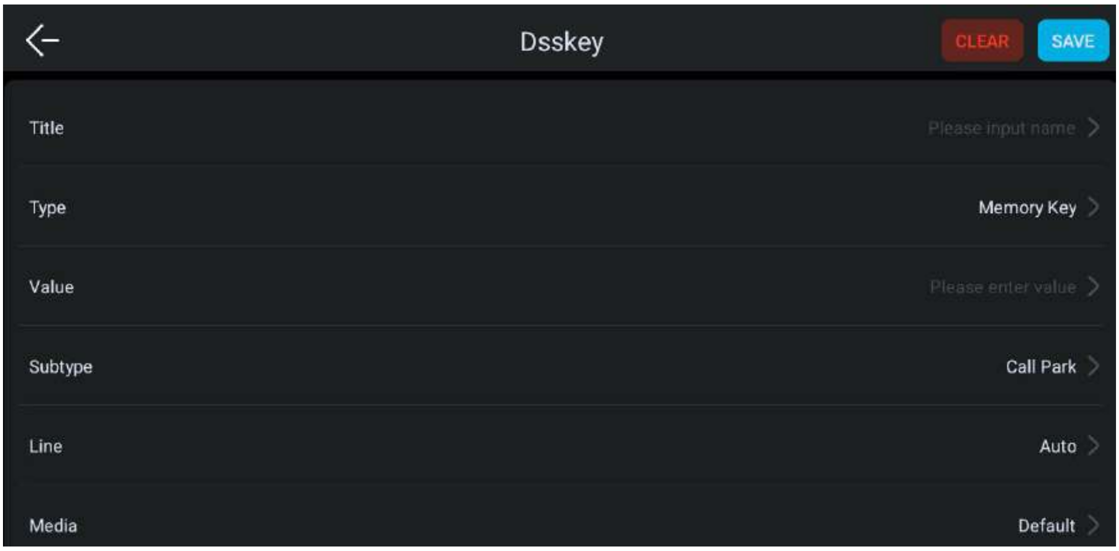


Figure 56 - Set Call Park key

- **Web interface:** log in the phone page, enter the [**Function Key**] >> [**Function Key**] page, select a dsskey, 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.

Figure 57 - Set Call Park key on Web

8.20 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 dsskey for BLF and setting the Pick Up Number.

- **Phone interface:** In standby mode, click the "unfold" button and long press an editable key to enter the interface of function key setting. Set the function key type as memory key and the subtype as BLF/NEW CALL, and set the corresponding SIP line. Finally fill in the Pickup number.
  - 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.
  - Press the dsskey to pick up the phone.
  - The caller picks up the call and speaks to it.

- **WEB interface:** Log in the phone web, enter the [**Function Key**] >> [**Function Key**] page, select a dsskey, set the memory key type as memory key, the subtype as BLF/NEW CALL, and set the corresponding SIP line and PickUp Number.

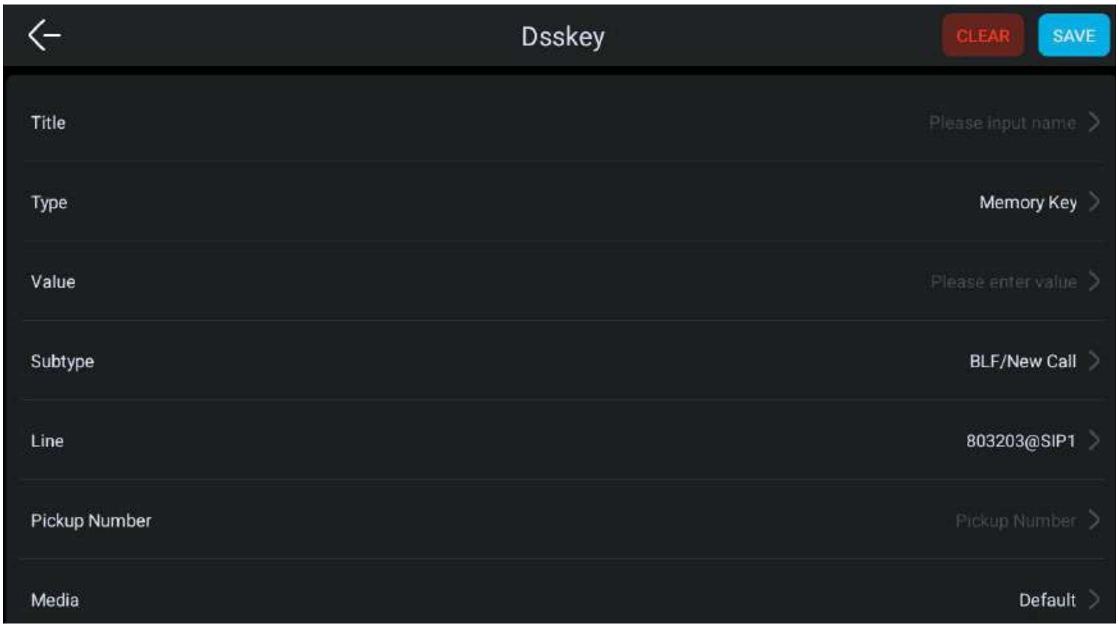


Figure 58 - Pick Up Key Setting

Figure 59 - Pick Up Key Setting on Web

8.21 Anonymous Call

8.21.1 Anonymous Call

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

- In [**Line**] >> [**SIP**] >> [**Advanced Settings**] on the web interface, you can find the configuration item for "**Anonymous Call Standard**".
- The default is none, which is off, and RFC3323 and RFC3325 are optional.
- Select any one to open the anonymous call.

Figure 60 - Enable Anonymous Call

The following is the call log of the phone receiving an anonymous call:

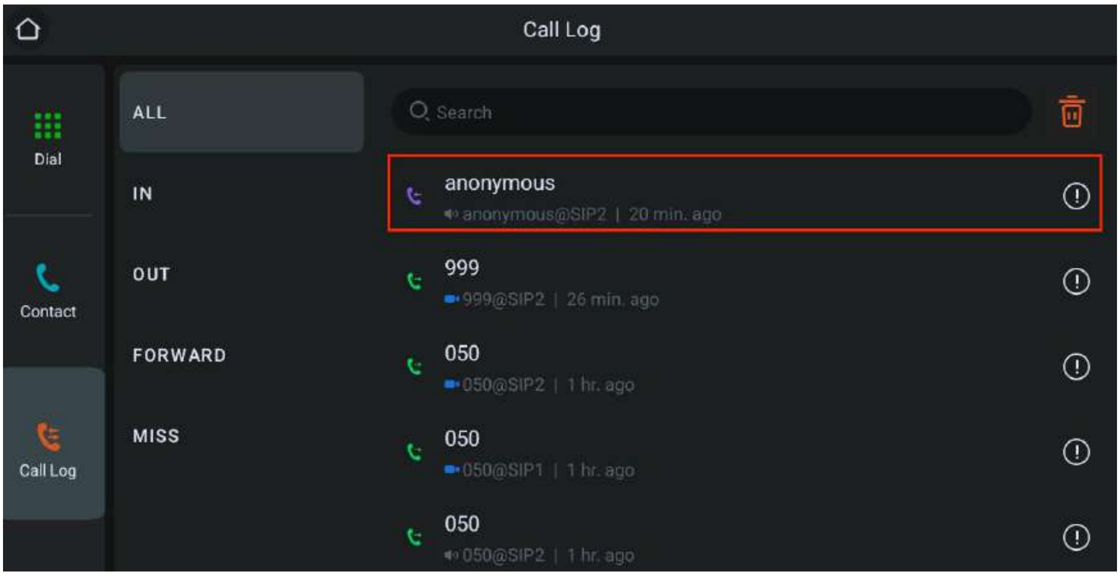


Figure 61 - Anonymous Call Log

8.21.2 Ban Anonymous Call

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

- On the web, [**Line**] >> [**SIP**] >> [**Advanced Settings**], check “**Blocking Anonymous Call**”.
- 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.

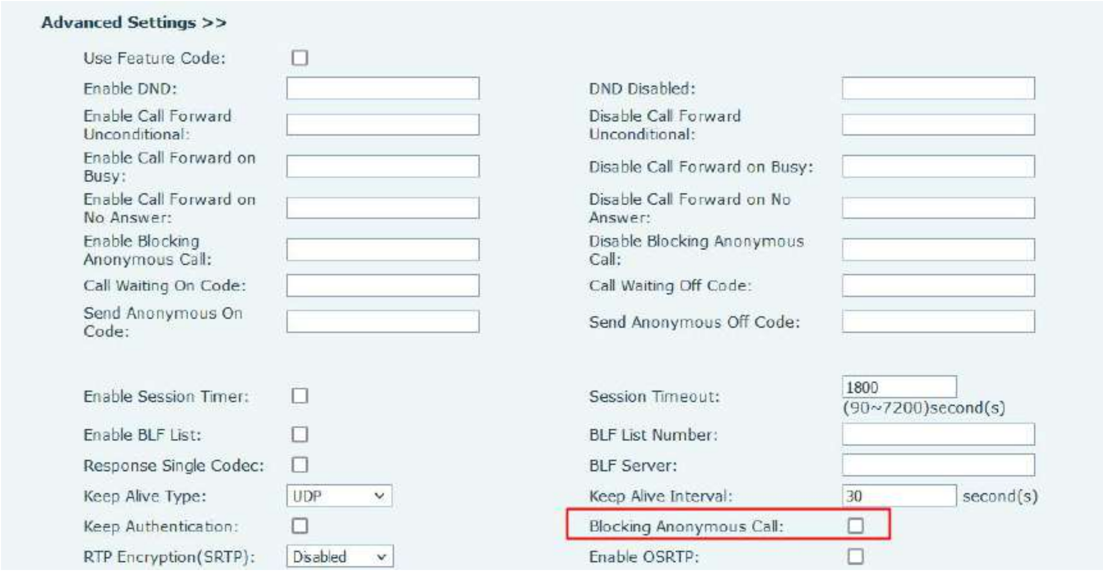


Figure 62 - Blocking Anonymous Call on Web

8.22 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.

- **Phone interface:**
  - Go to [**Settings**] >> [**Advanced Settings**] >> [**SIP Settings**] on the phone and select the line for which you want to enable the hotline.
  - Navigate to [**Account Settings**] >> [**Basic Settings**] and tap "**Enable Hotline**".
  - Set the hotline number and hotline delay time, then tap the [**SAVE**] button.

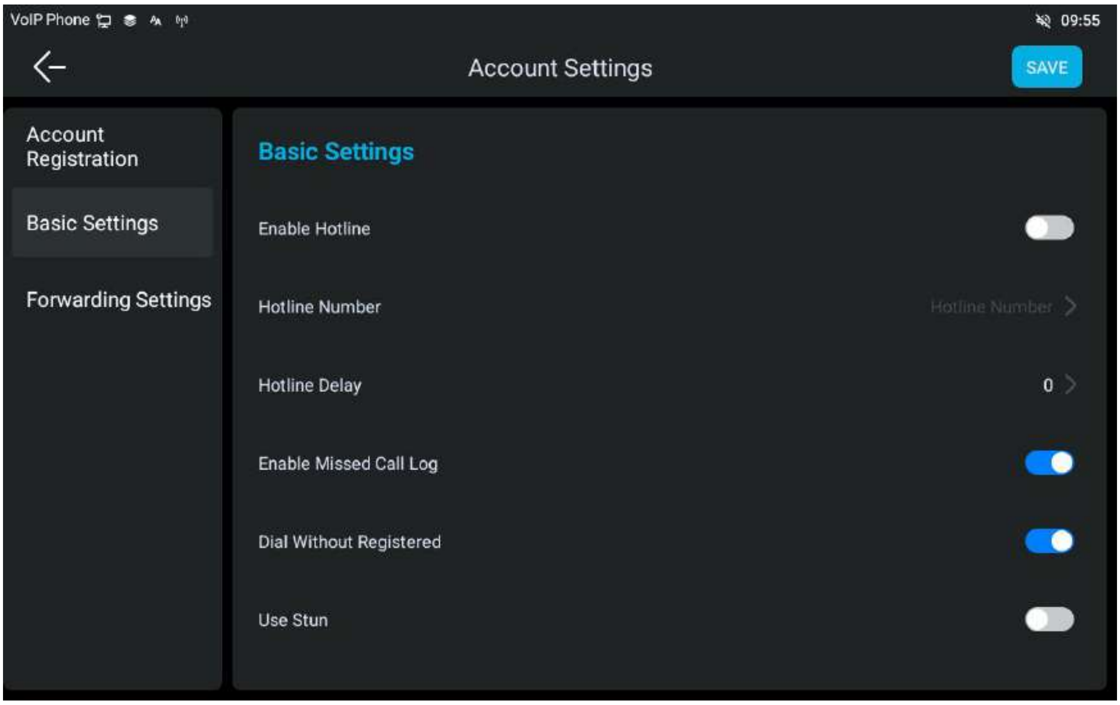


Figure 63 - Hotline Settings

■ **WEB interface:**

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

■ 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.

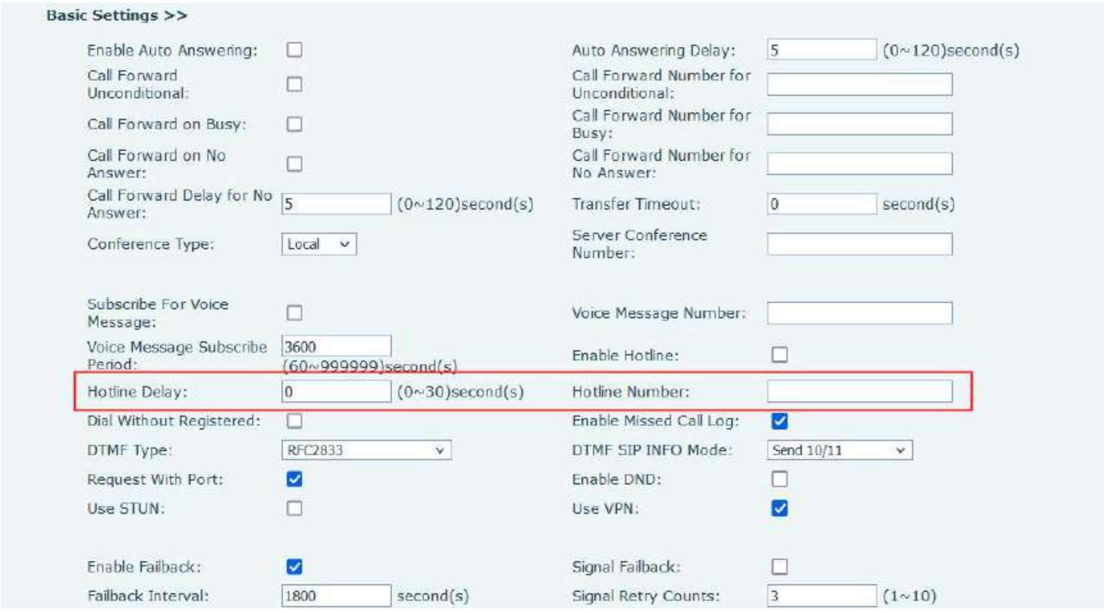


Figure 64 - Hotline Settings on Web

9 Advanced Function

9.1 BLF (Busy Lamp Field)

9.1.1 Configure the BLF Functionality

- **Web Interface:** log in the phone page, enter the [**Function key**] >> [**Function key**] page, select a dsskey, 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 which is subscribed, set the corresponding SIP line. The pickup number is provided by the server. The specific use of reference 8.20 Pick Up.

Figure 65 - Set BLF Function Key on Web

- **Phone interface:** Click unfold, long press a function key to enter the function key Settings interface, key function key types of memory, a subtype of BLF/New Call, BLF/Blind Transfer, BLF/Attended Transfer, BLF/Conference, BLF/DTMF, the values to be subscription number, and set up corresponding SIP lines.

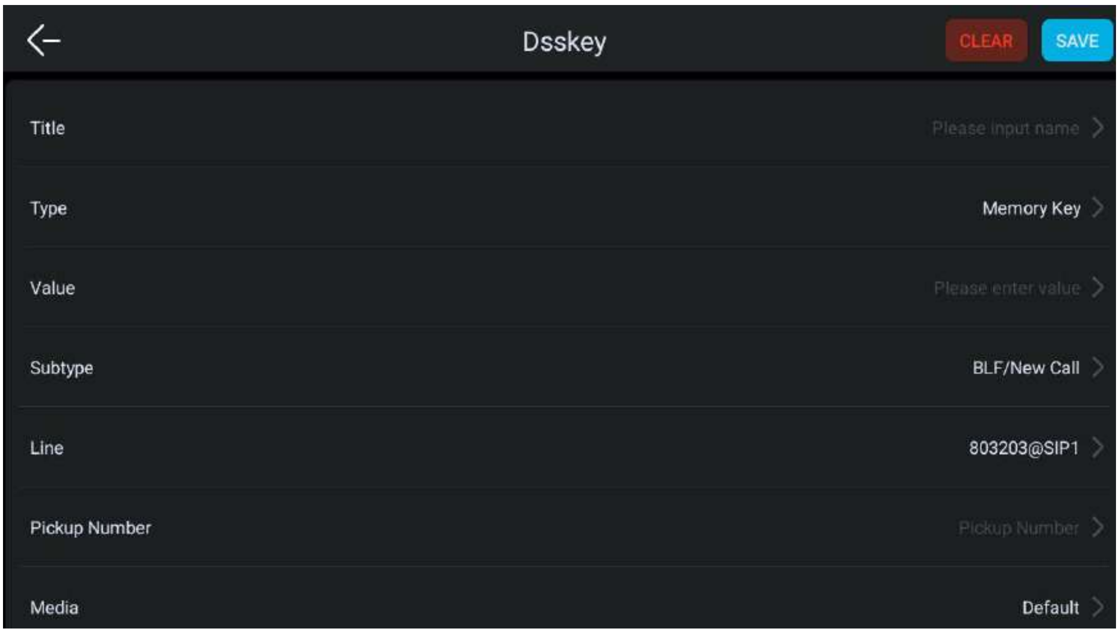


Figure 66 - Set BLF Function Key

Table 8 - 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 attended transfer the call to the subscribed number.
BLF/Conference	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, off).

BLF function：

- Monitor the status of subscribed phones.
- Call the subscribed number.
- Transfer calls to the subscribed number.
- Pick up incoming calls from subscribed number.

1）Monitors the status of subscribed phones.

Configure a BLF function key. When the status of the subscribed number (Idle, Ringing, In a Call) changes, the icon status of the function key will change accordingly.

2）Call the subscribed number.

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

3）Transfer calls/calls to the subscribed number.

Refer to Table 8 - BLF Function Key Subtype Parameter List. You can use the BLF key to perform blind transfer, attended transfer, and semi-attended transfer on the current call. You can also invite the subscribed number to join the call, send DTMF tones, etc.

4）Pickup incoming calls from subscribed phones.

Configure the pickup number when setting up the BLF function key.When the subscribed number has an incoming call ringing, the function key icon will display as a fast red flash. Press the BLF key at this time to answer the incoming call of the subscribed number.

9.2 Intercom

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

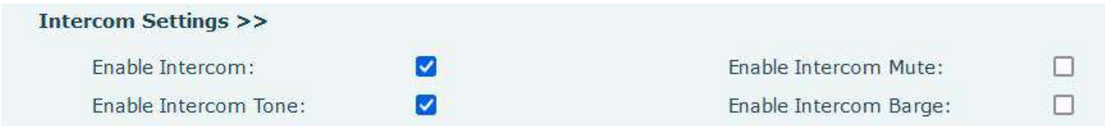


Figure 67 - Intercom Settings on Web

Table 9 - Intercom Configuration

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	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

9.3 Multicast

This feature allows users to make some kind of broadcast call to people who are in multicast group. User can configure a multicast dsskey 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.

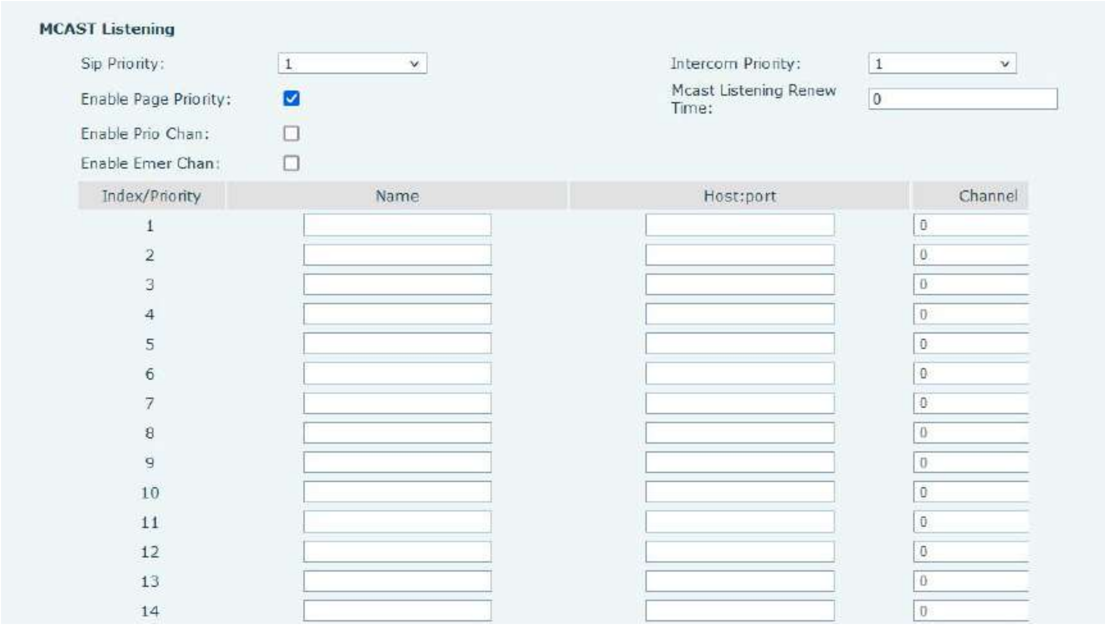


Figure 68 - Multicast Settings on Web

Table 10 - Multicast Settings

Parameter	Description
Sip Priority / Intercom 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.
Mcast Listening Renew Time	Configure mcast Listening Renew Time.
Enable Prio Chan	When enabled, only calls with the same port and channel can be connected. Channel 24 is the priority channel (higher than channels 1–23); a channel value of 0 means the channel function is not used.
Enable Emer Chan	When enabled, channel 25 has the highest priority.
Name	Listened multicast server name
Host: port	Listened multicast server's multicast IP address and port.

Multicast：

- Go to web page of [**Function Key**] >> [**Function Key**] , select the type to multicast, set the multicast address, and select the codec.
- Click Apply.
- Set up the name, host and port of the receiving multicast on the web page of [**Phone Settings**] >> [**MCAST**].
- Press the dsskey of Multicast Key which you set.
- Receiver will receive multicast call and play multicast automatically.

Figure 69 - Set Multicast Function Key on Web

MCAST Dynamic:

Multicast configuration information is delivered via the SIP Notify signaling. After receiving the information, the device configures it into the system to start or stop multicast listening. If a multicast call does not end normally, but the device fails to receive multicast RTP packets due to certain reasons, the listening function will be automatically disabled after the timeout period for Auto Exit Expires.

Figure 70 - MCAST Dynamic

9.4 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.

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

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

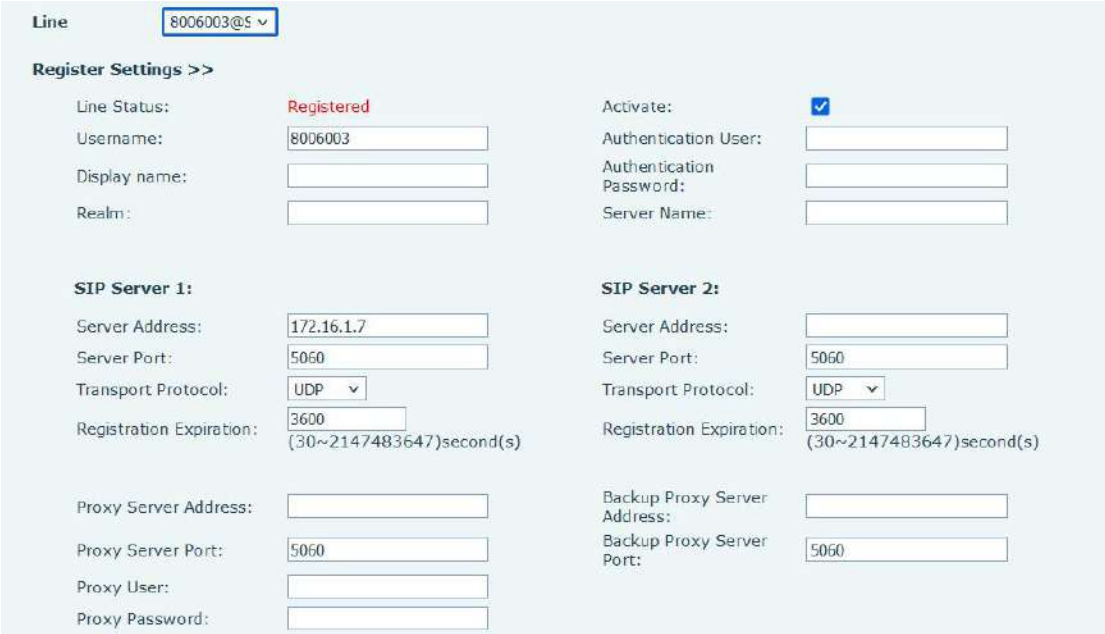


Figure 71 - Register SIP Account

- **Configure SIP hotspot server:**

The Hotspot feature can be configured in [**Settings**] >> [**Advanced Settings**] >> [**SIP Hotspot**] on the phone (a password is required; the default password is **admin**), or in [**Line**] >> [**SIP Hotspot**] on the web. The Hotspot feature is enabled by default on the A330 & A330i models.

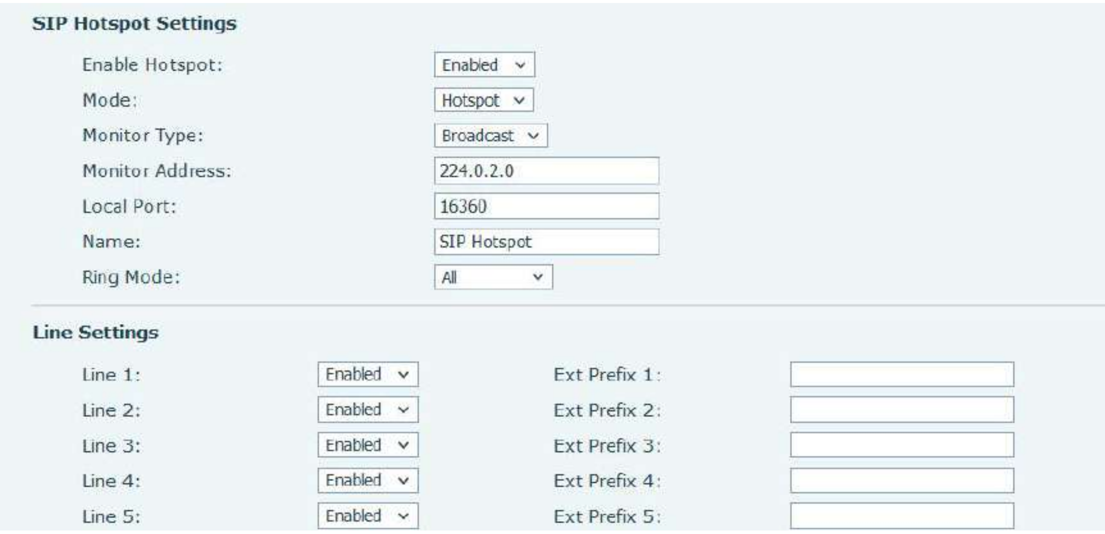


Figure 72 - SIP Hotspot Server Configuration

Table 11 - SIP Hotspot Settings

Parameter	Description
Enable Hotspot	Set the "**Enable Hotspot**" option in the SIP Hotspot configuration to "**Enabled**" .
Mode	Select **Hotspot** to configure the phone as a SIP hotspot.

	Select **Client** to configure the phone as a client.
Monitor Type	Both "**Broadcast**" and "**Multicast**" are supported. Select multicast to limit broadcast packets in the network.Note: The monitor type must be consistent between the server and client. For example, if the client uses Multicast, the SIP hotspot server must also use Multicast.
Monitor Address	When multicast is selected as the monitor type, enter the multicast communication address for the client and server. If broadcast is used, this field does not need to be configured—the system will use the broadcast address of the phone's WAN port IP by default.
Local Port	Enter a custom communication port for the hotspot. The port must be consistent between the server and client.
Name	Enter a name for the SIP hotspot to distinguish different hotspots in the network and avoid connection conflicts.
Ring Mode	Set the ringing rule for the hotspot host and extensions when there is an incoming call. The dropdown options are: ● **All**: Both the hotspot host and extensions ring ● **Hotspot**: Only the hotspot host rings ● **Extensions**: Only extensions ring
Line Settings	Set whether to associate and enable the SIP Hotspot feature on each SIP line

When the Hotspot feature is enabled and the mode is set to Hotspot, the phone will automatically connect to extensions whose models are in the management list (see 14.2.3 Management Device Model), and assign extension numbers (0, 1, 2, ...). You can manage the connected extensions in [**Line**] >> [**Hotspot Extension Management**] on the web.

When the phone is in management mode, only devices in the Managed Extensions list can connect to the phone's hotspot. To add a managed extension:

- Click the" **Add**" button to add one manually;
- Or select a device from the Unmanaged Extension List, then click "**Move to Managed**".


Managed Extension Information

Add | Delete | ReProvision | Reboot | Add to Group | Move To UnManaged

☐	Index	Extension Name	Mac	Model	SoftVersion	Ip	Ext	Group	Status	Registration Number	Edit
☐	1	A12V	0c:38:3e:7d:2f:50	A12V	2.12.49.8	172.16.37.214	1		OffLine		Edit
☐	2	i16SV	0c:39:3e:41:21:ab	i16SV	2.12.50.14		3		OffLine		Edit
☐	3	A12V-02	0c:38:3e:68:53:b8	A12V-02	2.12.49.8		4		OffLine		Edit
☐	4	i16S	00:30:1b:ba:02:db	i16S	2.12.50.14	172.16.37.174	5		OffLine		Edit

UnManaged Extension Information

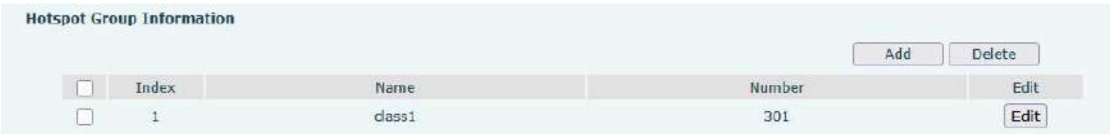
Delete | Move To Managed

☐	Index	Mac	Model	SoftVersion	Ip	Ext	Status	Registration Number
☐	1	00:d8:4a:0a:f9:7d	Y501W-Y	2.12.58.15			OffLine	

Figure 73 - Hotspot Extension Management

In the webpage Hotspot Extension Management, you can add hotspot groups. The configuration rules and operation instructions are as follows:

- A dedicated group number (cannot be between 0 and 300) must be set for the group. This number can be regarded as a virtual hotspot extension number. When the hotspot host or other extensions call this number, all devices in the group will ring simultaneously. When any device answers the call, the ringing of the remaining devices will stop automatically. (Note: A hotspot group is the same as a ring group on the phone interface. For details, see 14.2.4 Ring Group.)
- Add Hotspot hosts and managed extensions to the group: Click Edit to add them to the group one by one, or select multiple devices and click Add to Group to add them in batches.


Hotspot Group Information

Add | Delete

☐	Index	Name	Number	Edit
☐	1	class1	301	Edit

Figure 74 - Hotspot Group

- **Configure SIP hotspot client:**

As a SIP hotspot client, no SIP account needs to be set. The Phone set will automatically obtain and be configured 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.

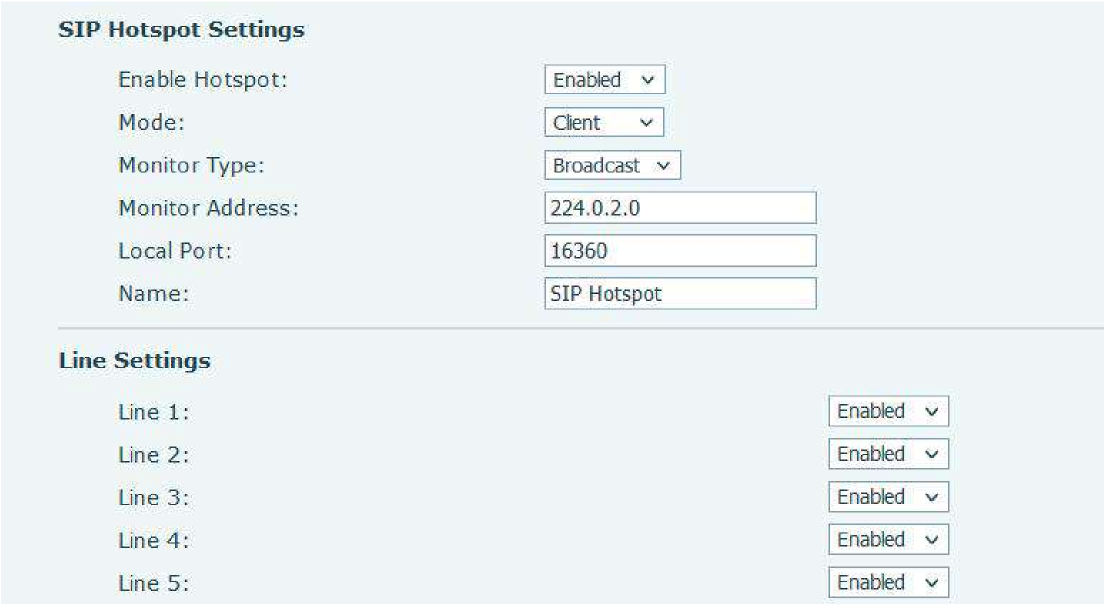


Figure 75 - SIP Hotspot Client Configuration

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 [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 Call

Tap the [**Call**] menu button at the bottom of the standby home screen to enter the [**Dial**] interface. The left navigation bar of this interface includes four functional options: Dial, Contact, Call Log, and Call Record.

10.1 Dial

1. Enter [**Call**]>>[**Dial**] in the Broadcast Intercom System.
2. Enter the number to dial, and then press the [**Audio**] or [**Video**] button to call out.

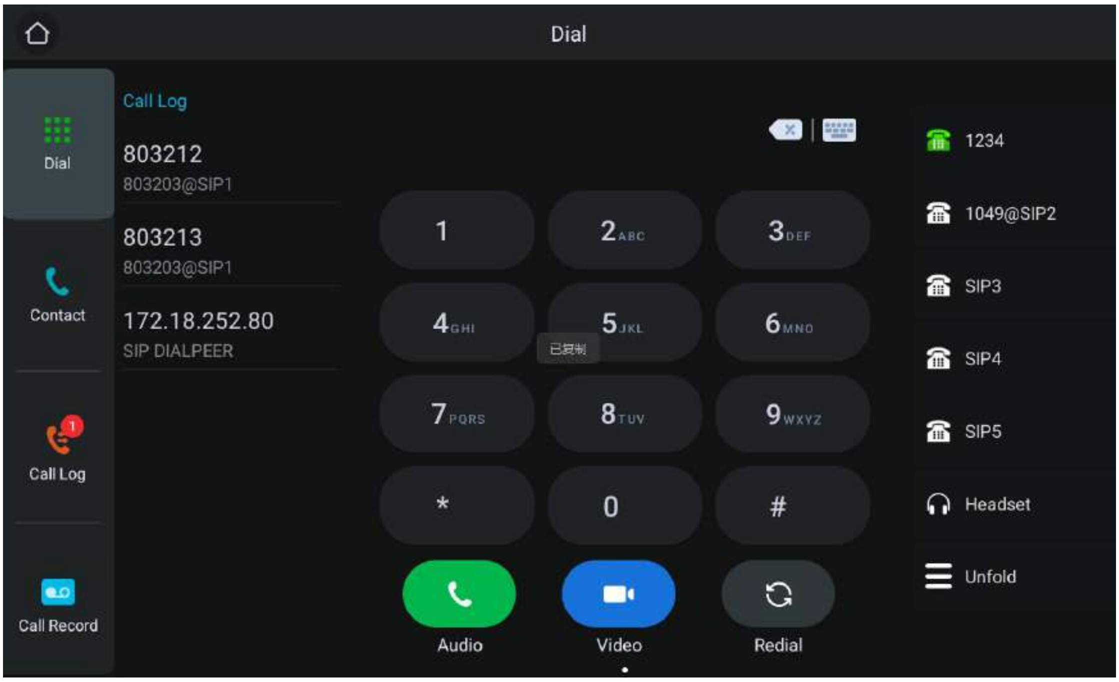


Figure 76 - Manual Dialing

10.2 Contact

In the [**Call**] >> [**Contact**] interface, you can display the information of local contacts, doorphones, and network phonebook contacts, and initiate calls. The address book is empty by default.

Note! The user account of the phone can store a maximum of 2000 contact entries.

10.2.1 Local Contact

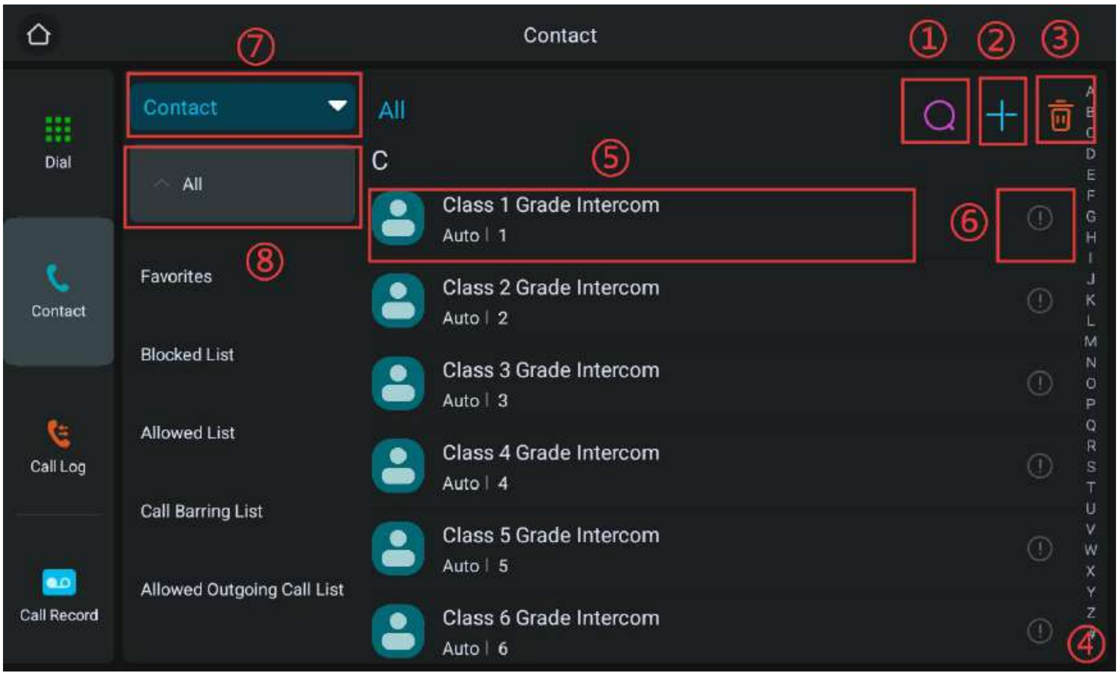


Figure 77 - Phonebook Local Contact

Table 12 - Phonebook Local Contact

Index	Function
①	Click to enter keywords and quickly find contacts in the phonebook.
②	Used to add contacts to the phonebook.
③	You can select multiple contacts in batches and then perform the delete operation.
④	Click the corresponding letter to quickly locate contacts starting with that letter.
⑤	Click to select the corresponding contact's number and initiate a call.
⑥	View the detailed information of the corresponding contact.
⑦	You can switch the classification type of the address book (Phonebook Contacts / Access Control Devices) and select different contact categories.
⑧	Click to display contact groups.

10.2.1.1 Add/ Edit / Delete Local Contacts

Add a Contact in the [**Call**] >> [**Contact**] Interface: Click Icon ② (see Figure 77), fill in information such as name, phone number, group, and ring, then click the √ icon in the upper right corner to add the contact successfully.

After creating a contact, you can click Icon ⑥ (see Figure 77) on the right side of the corresponding contact in the list to view the contact details, then click the "Edit" button in the upper right corner to edit the contact information.

Local contacts can also be added or imported in batches through the [**Phonebook**] >> [**Contacts**] / [**Advanced**] on the web page, or local contacts can be added from the call log and cloud phonebook.

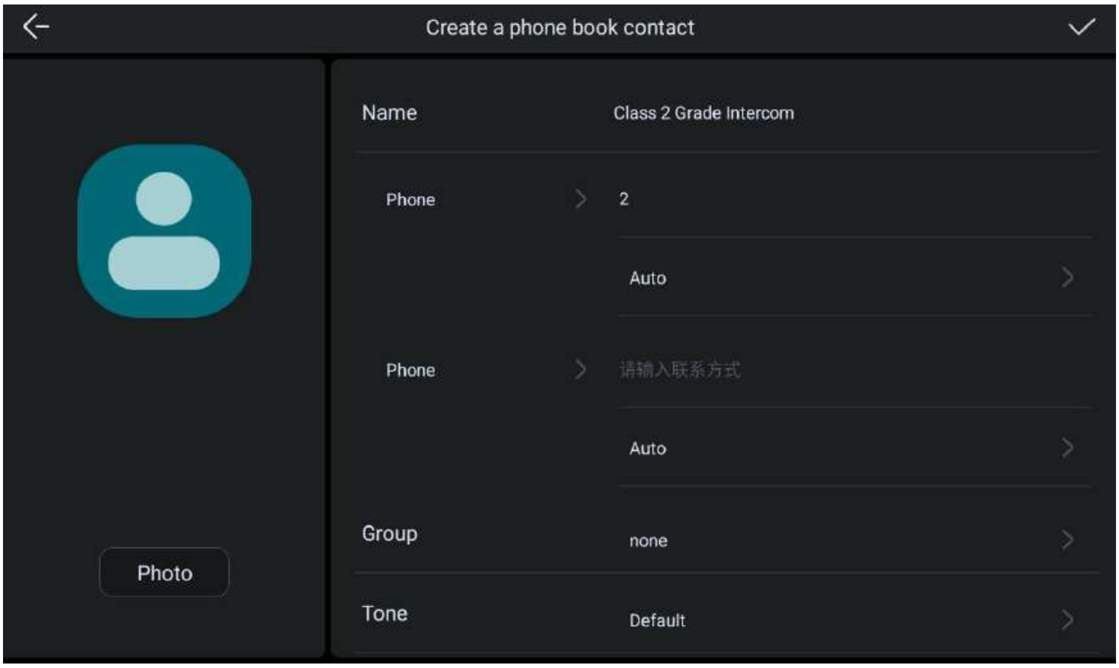


Figure 78 - Create/Edit Contact

Click the icon ③ in the upper right corner (see Figure 77) to select the contacts you want to delete.

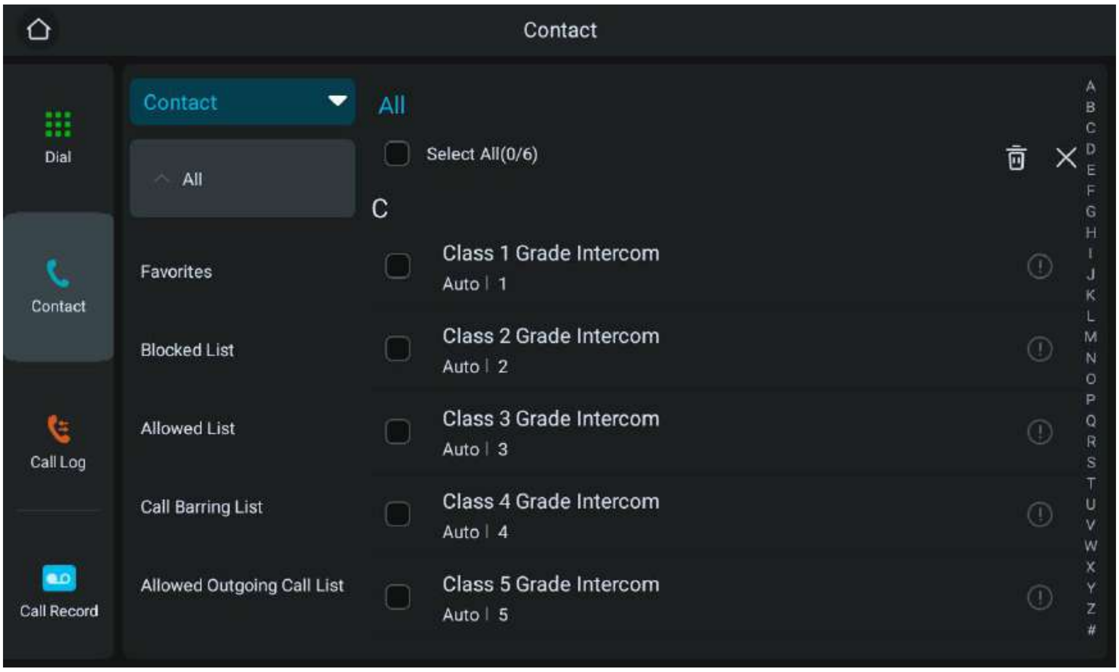


Figure 79 - Delete Contact

10.2.1.2 Call List

The device supports the Blocked Incoming Calls List. If a number is added to the Blocked Incoming Calls List, incoming calls from this number will be directly rejected, and the local phone will display a missed call.

- There are multiple ways to add numbers to the Blocked Incoming Calls List on the phone. You can add them directly in [**Call**] >> [**Contact**] >> [**Blocked List**].
- You can select any number in the phonebook (both local and network phonebooks are supported) to configure and add it to the list.
- You can select any number in the call log to configure and add it to the list.

Note! Numbers in the Blocked Incoming Calls List can be called out normally.

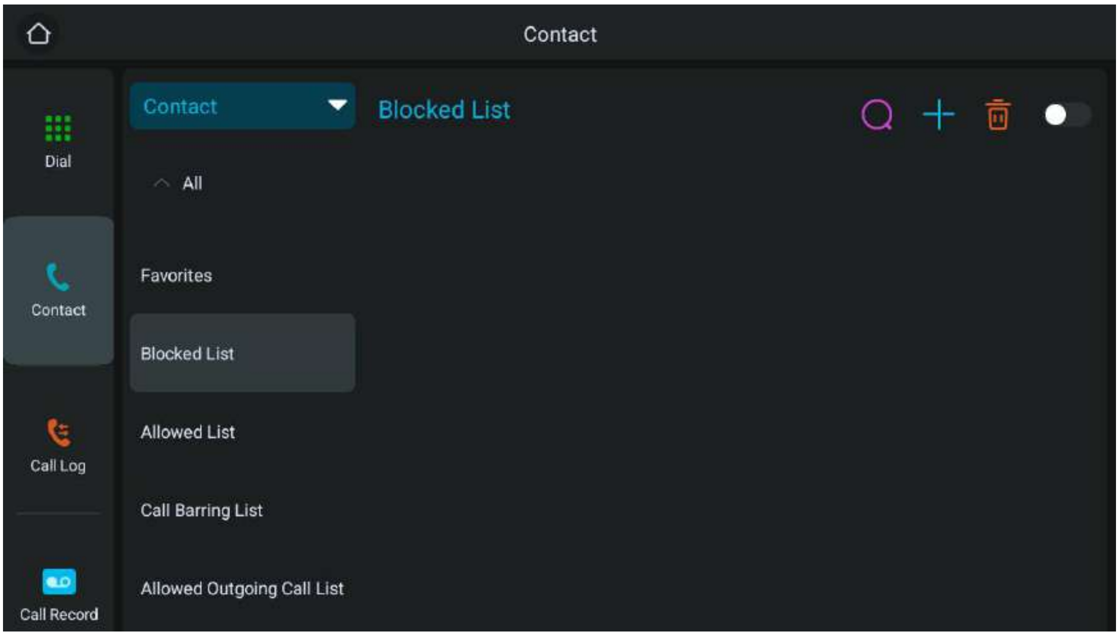


Figure 80 - Blocked List

- There are multiple ways to add numbers to the Blocked Incoming Calls List via the web page. You can add them in [**Phonebook**] >> [**Call List**] >> [**Restricted Incoming Calls**].
- You can select any number in the phonebook (both local and network) to configure and add it to the list.
- You can select any number in the call history to configure and add it to the list.

Note! After adding restricted incoming numbers, you need to click the button in the upper right corner of the [Call] >> [Contact] >> [Blocked List] interface on the phone, or enable the restricted incoming numbers list in [Phonebook] >> [Call List] >> [Calllist Settings] on the web page.

Figure 81 - Enable Restricted Incoming List

- Allowed Incoming Calls List:

When DND (Do Not Disturb) is enabled, numbers in the Allowed Incoming Calls List can still call in.

- Outgoing Call Restrictions List:

Numbers added to this list will be restricted from outgoing calls until they are removed from the list.

- Allowed Outgoing Calls List:

After enabling the Allowed Outgoing Calls List, the device only allows outgoing calls to numbers in this list.

10.2.2 Doorphone

Exclusive contacts bound to the door access function. It will display an [**Open Door**] button during a call with such contacts; clicking it will trigger the access control device to open the door. In addition, you can configure function keys and emergency broadcasts to achieve one-click door opening for doorphone in a specific group or all doorphones (see 10.2.2.3 One-Click Door Opening and 14.2.1 Emergency Broadcast for details).

10.2.2.1 Add Door Access Contact

Go to the [**Call**] >> [**Contact**] interface, click "Phonebook Contacts" (Icon ⑦ in Figure 77) to switch to the doorphone list. Click the "" button in the upper right corner and fill in the name, number, access code, password, line, and other information.

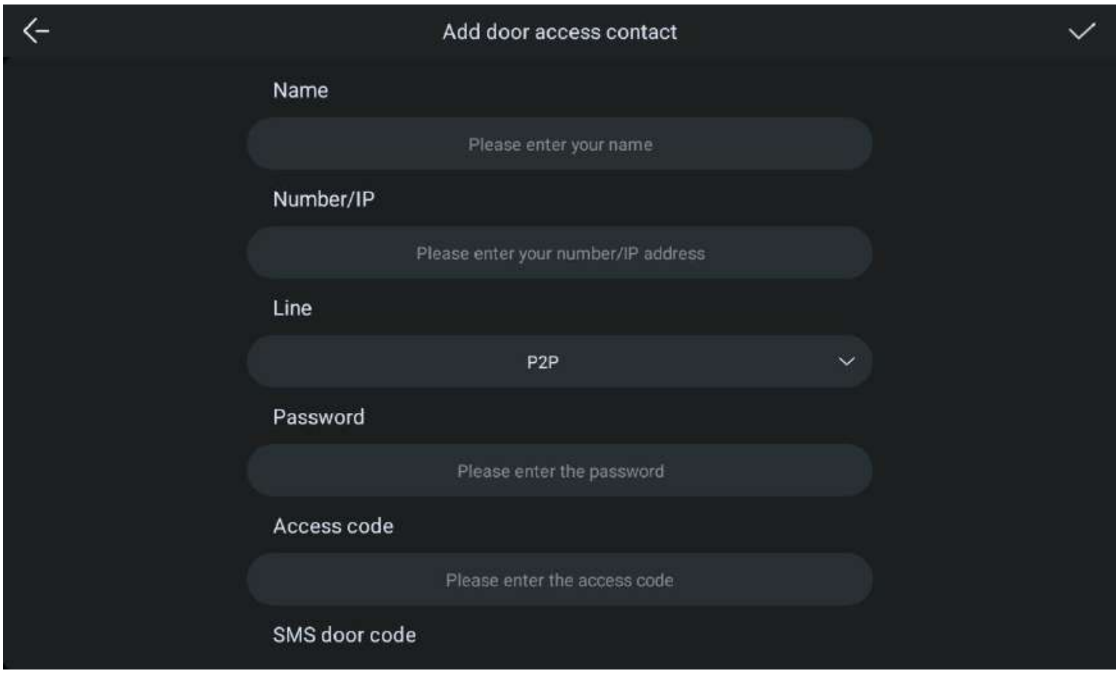


Figure 82 - Add Door Access Contact

Table 13 - Add Door Access Contact

Parameter	Description
Name	Name of the Door Access Contact.
Number/IP	SIP number or IP of the Door Access Contact.
Line	Select the communication line. When making a call using an IP number, the line must be set to P2P.
Password	Door opening password. When the doorphone calls the phone, whether the phone is answered or not (i.e., on the calling interface and call interface), click the "Open Door" button to send the door opening password saved in the contact to the doorphone via DTMF. The doorphone will open the door after successful identification.
Access code	When the phone calls the doorphone, click the "Open Door" button to send the access code saved in the contact to the doorphone via DTMF. The doorphone will open the door after successful identification.
SMS door code	After configuration, click the corresponding doorphone type dsskey to send the door open code, which is used for the door opening operation of the doorphone.
SMS door closing code	After configuration, click the corresponding doorphone type dsskey to send the door close code, which is used for the door closing operation of the doorphone.
Grouping	Assign to doorphone group. When using an doorphone type dsskey, the doorphone in the group will perform door opening or closing

	operations.

10.2.2.2 Opening the Door During Ringing/Call

After configuring the door access contact, when the doorphone calls the phone using the corresponding number, an [**Open Door**] button will be displayed during ringing and active call states. Pressing this button will control the doorphone to perform the door opening operation.

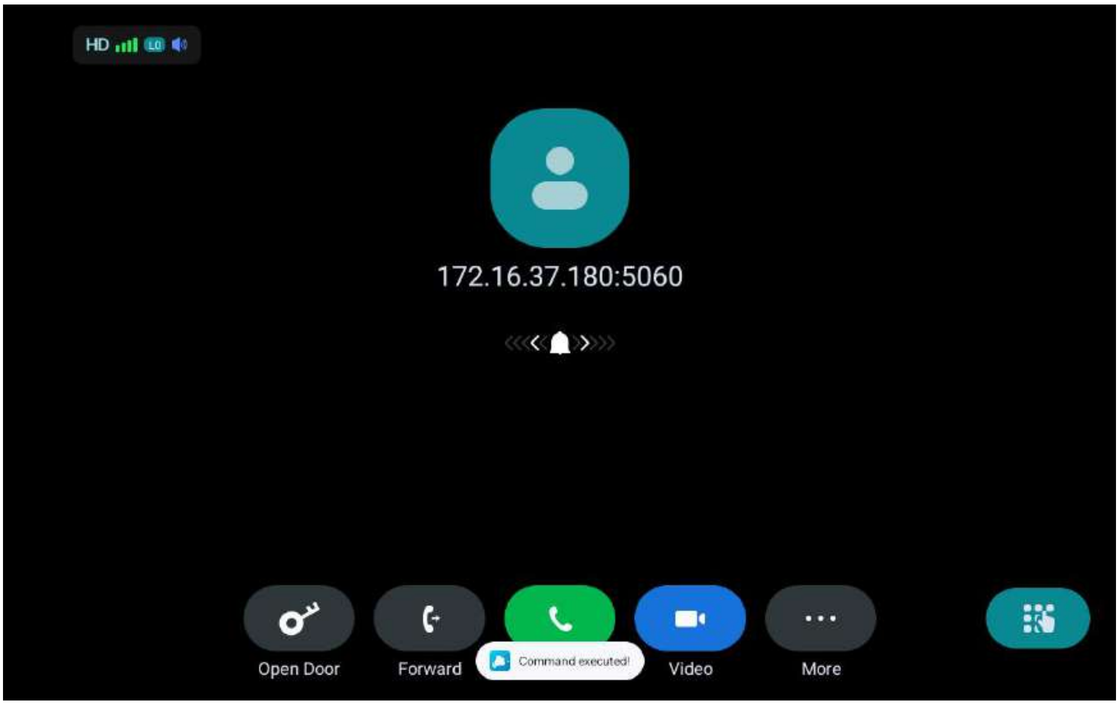


Figure 83 - Open the Door When Ringing

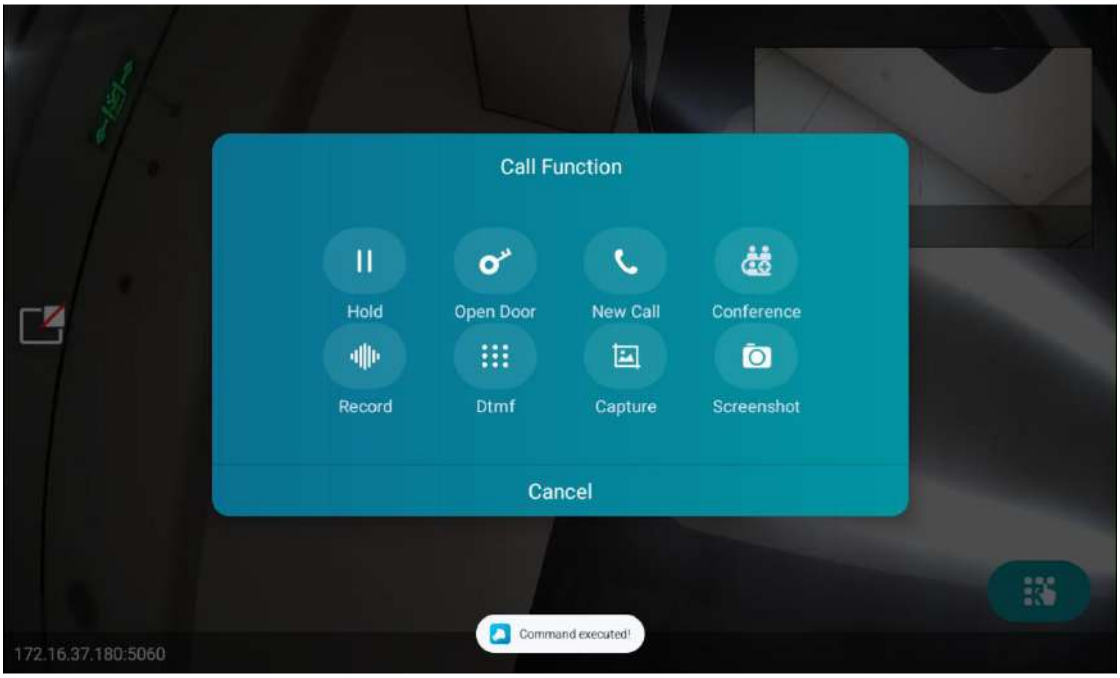


Figure 84 - Open the Door During a Call

10.2.2.3 One-Click Door Opening

1. Add an door access contact, configure the corresponding SMS door code and SMS door closing code according to the device's SMS trigger message and reset message.

2. Assign the device to a group. Subsequently, you can configure an access control type function key and associate it with the corresponding group to achieve one-touch door opening and closing for all devices in the group.

You can also enable the one click door opening configuration in an emergency broadcast task; when the task is executed, all doorphone will be opened uniformly. For details on creating an emergency broadcast task, please refer to 14.2.1 Emergency Broadcast.

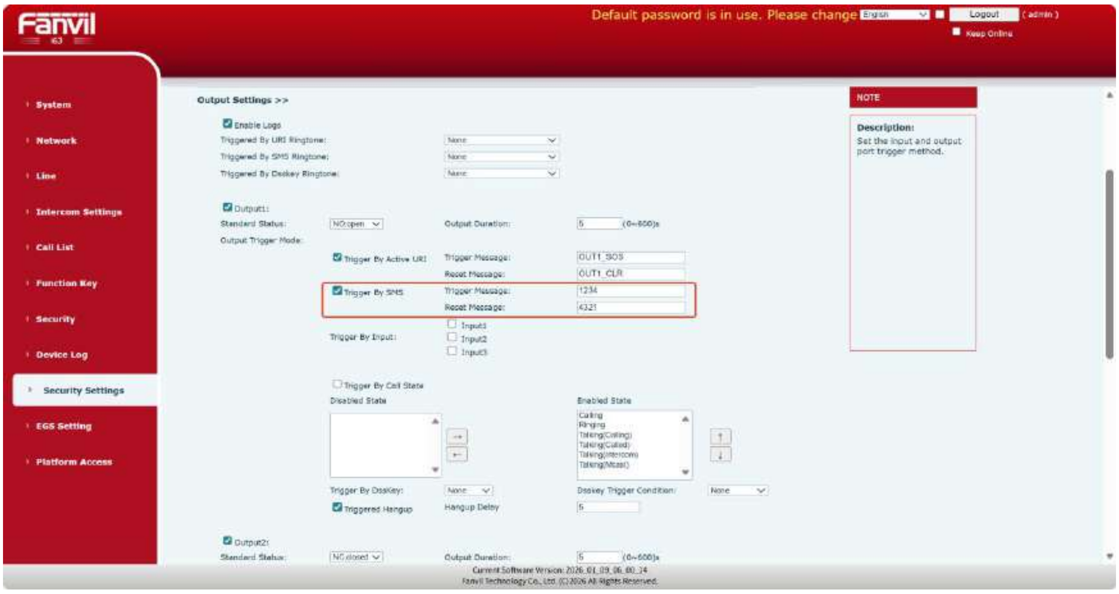


Figure 85 - Fanvil Doorphone SMS Trigger Message And Rest Message

Figure 86 - Doorphone Settings on Web

Configure One-click Door Opening Function Keys:

On the Idle screen , click "**Unfold**" in the lower right corner, long-press a function key to edit, select doorphone as the type, and then select the doorphone group and subtype. The subtypes include One-Click Door Opening, and One-Click Door Closing, and One-Click Always Door Opening.

- **One-Click Door Opening:** After the doorphone in the group are opened, they will automatically close when the door opening time configured on the doorphone side is reached.
- **One-Click Door Closing:** When the doorphone are in the open state, press the key to manually close the doors.
- **One-Click Always Door Opening:** After doorphones in the group are opened, they will remain in the open state and will not close automatically.

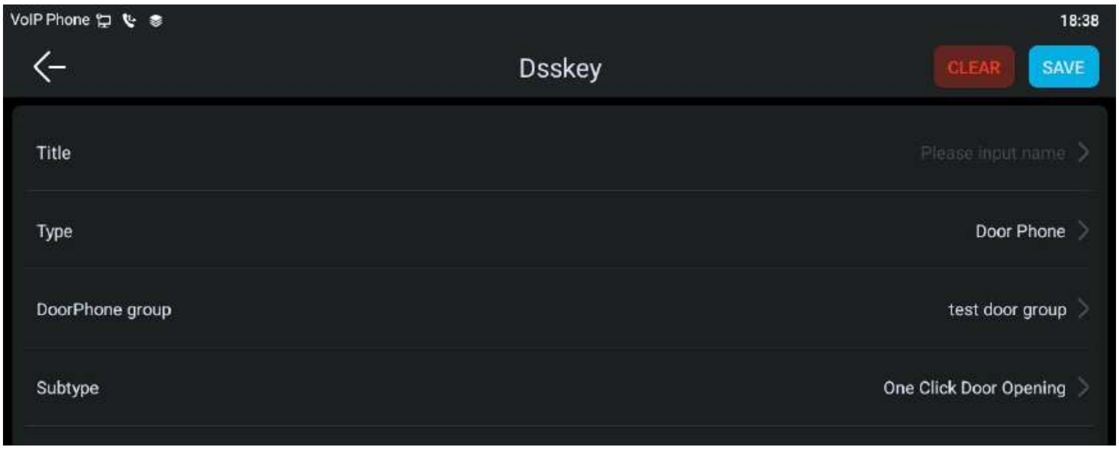


Figure 87 - Set Doorphone Function Key

After configuring the shortcut key, you can click it to perform door opening and closing operations. The operation result of each doorphone in the group will be displayed in real time.

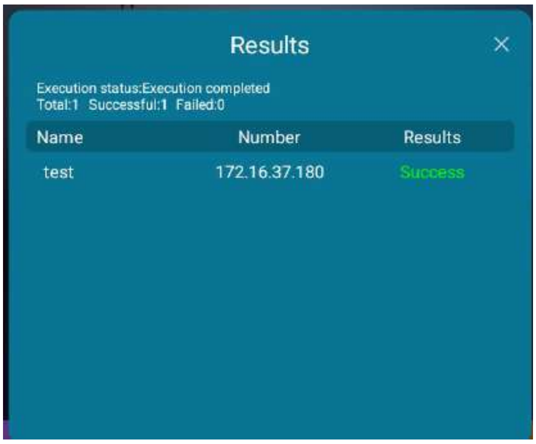


Figure 88 - One-Click Door Opening Result

10.2.3 Network Phonebook

The network phonebook allows users to download the phonebook from the cloud to the phone. This is very convenient for office users when using the phonebook, as the phonebook can be directly downloaded from the network phonebook server, making it extremely easy to create and maintain contact lists.

10.2.3.1 Configure the Network Phonebook

Go to the [**Calls**] >> [**Contact**] interface, click "**Contact**" (Icon ⑦ in Figure 77) to switch to "Network". Click the "+" icon in the upper right corner to configure a network phonebook. A maximum of 4 network phonebooks can be configured.

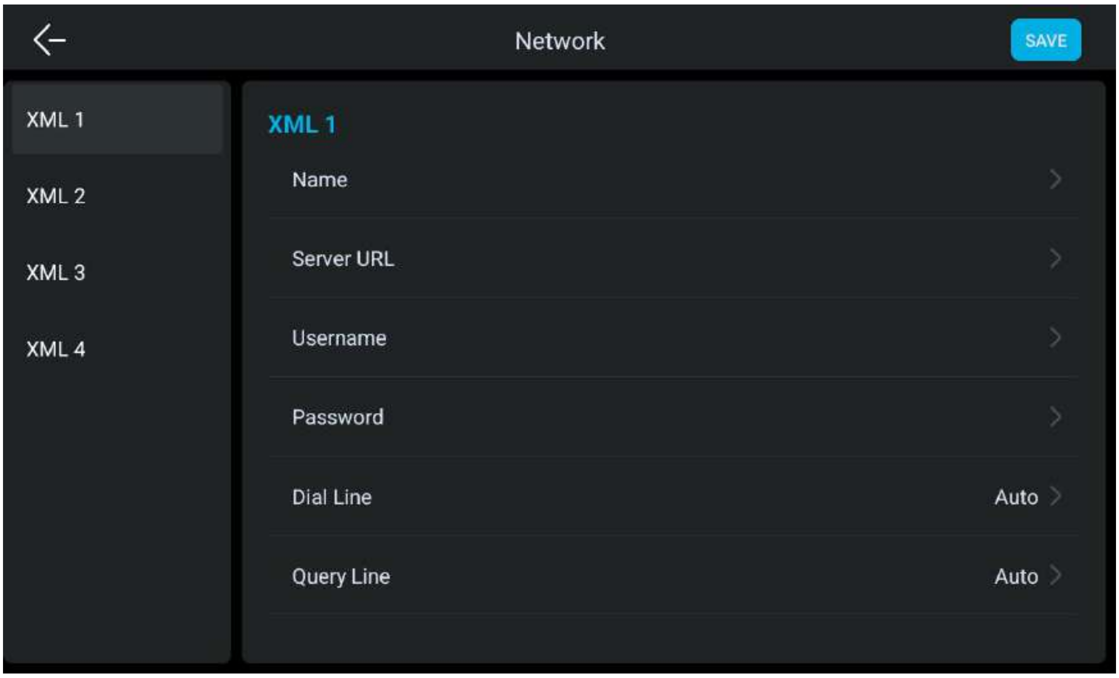


Figure 89 - Network Phonebook Settings

Table 14 - Network Phonebook Settings

Parameter	Description
Name	Set the identification name of the current network phonebook.
Server URL	Fill in the server address corresponding to the network phonebook.
Username	Enter the authentication username for accessing the network phonebook service.
Password	Enter the authentication password for accessing the network phonebook service.
Dial Line	Select the default line for dialing via this phonebook.
Query Line	When dialing a number from the network phonebook via the corresponding line, the system will match the phonebook associated with this line and automatically display the contact

	name during the call.
Phonebook type	Select the phonebook type; only XML phonebook is supported currently.

You can also log in to the device's web page and go to [**Phonebook**] >> [**Cloud Phonebook**] >> [Cloud Phonebook Management] to configure the network phonebook.

Figure 90 - Cloud Phonebook

10.2.3.2 Download the Network Phonebook

After you configure the network phonebook, you can select the corresponding phonebook name in the Network Phonebook interface to complete the download, and the contact information will be directly displayed in the current list.

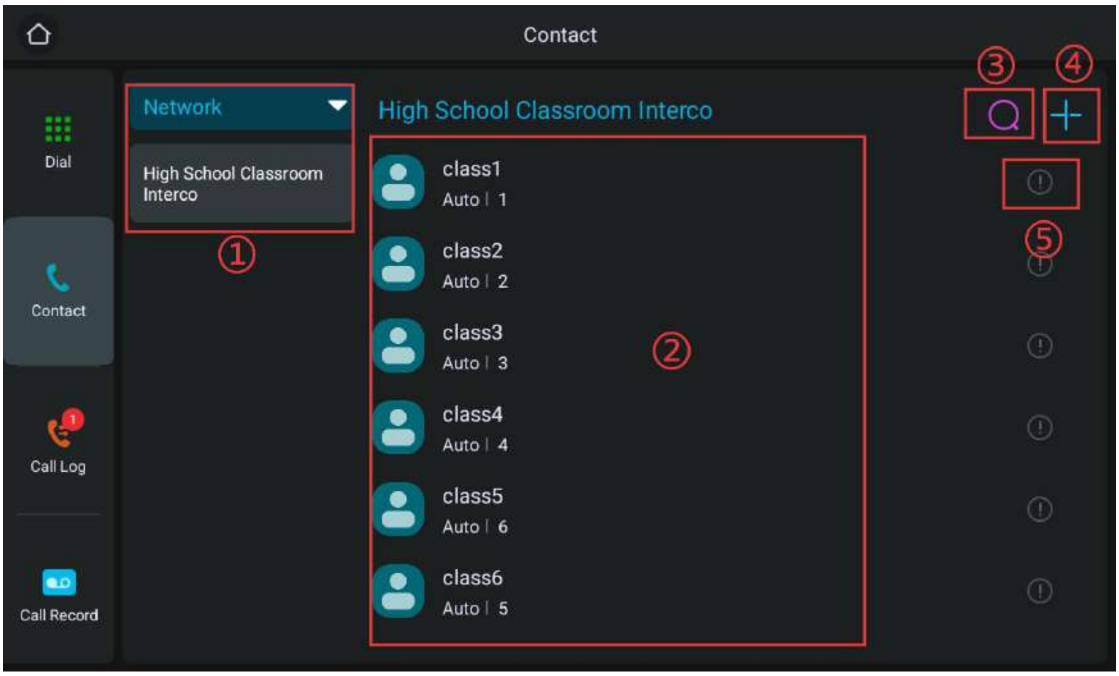


Figure 91 - Network Phonebook Contact

Table 15 - Network Phonebook Contact

Index	Function
①	Name of the network phonebook; click to download and view the corresponding

	contacts in the phonebook.
②	Contact in the network phonebook; click to call the contact's number.
③	Supports fuzzy matching search; enter any segment of the contact's name or number to quickly match the corresponding contact.
④	Click to configure the XML phonebook (see configuration in Figure 89).
⑤	Click to view the detailed information of the contact in the network phonebook.

In the Network Phonebook interface, click Icon ⑤ to view the detailed information of the contact, as shown in the figure below:

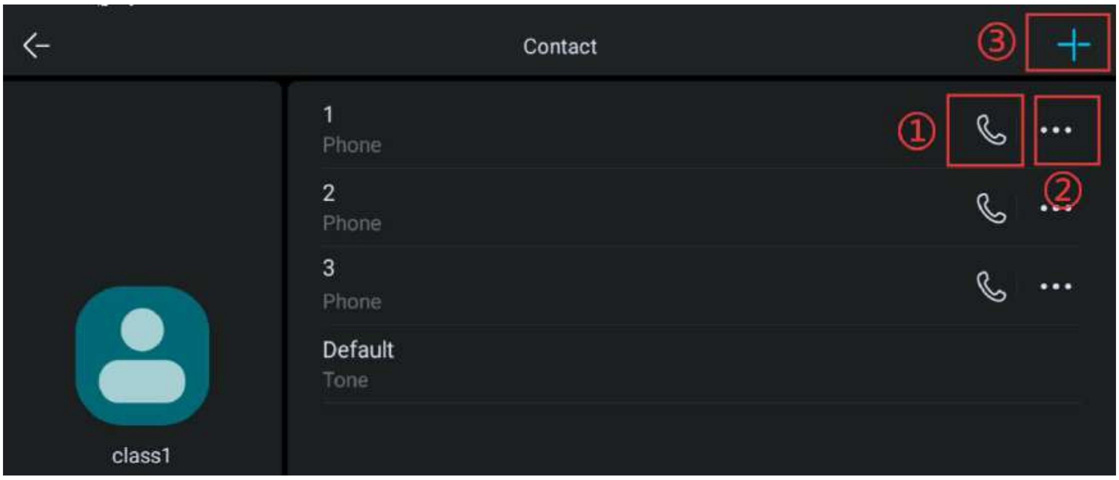


Figure 92 - Network Phonebook Contact Details

Table 16 - Network Phonebook Contact Details

Index	Function
①	Click to directly dial the contact's number.
②	Click to pop up the operation menu. The following operations are available:Copy the number to the clipboard / Extract to the dial pad / Add to the call list
③	Click to save the cloud phonebook contact to the local phonebook.

10.3 Call Log

Calls made on the broadcast intercom system by the user are saved in the call log. Users can view incoming/outgoing/forwarded/missed call records. The format of call records is as follows: the name is displayed at the top, and the number is displayed at the bottom. On the left side, users can filter call records by specific call types to narrow down the search scope.

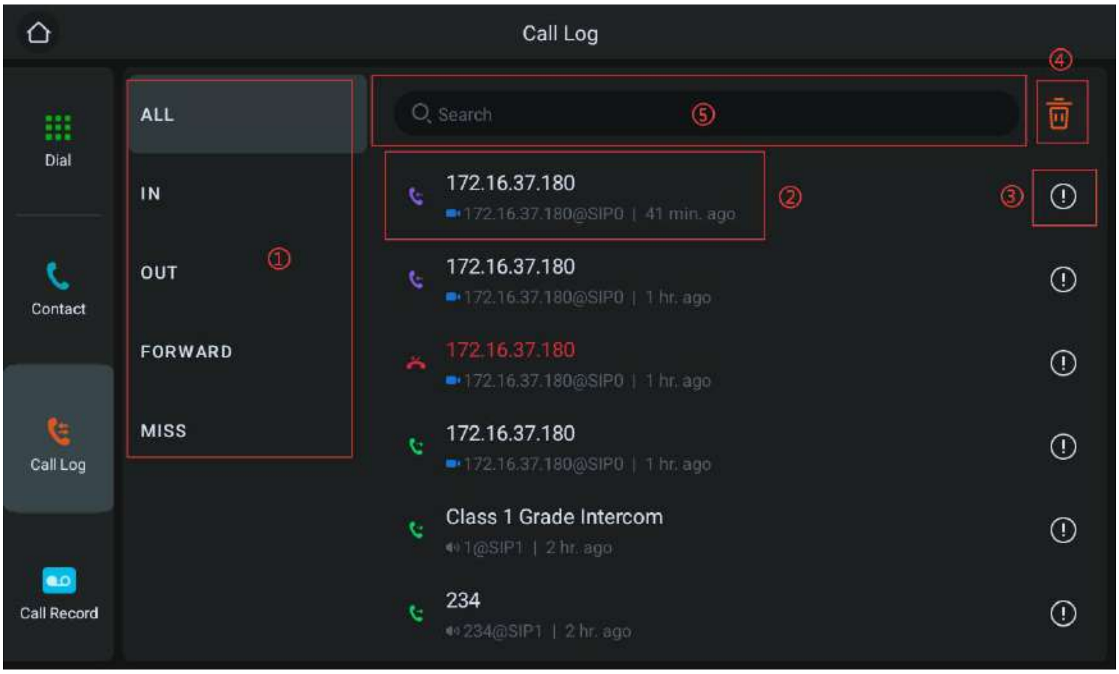
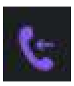
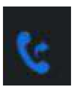


Figure 93 - Call Log

Table 17 - Call Log

Index	Function
①	Allows filtering call records by specific call types to narrow down the search scope.
②	Displays information about the call, such as the callee's name, number, call time, and call mode. Click to quickly call the number; if there are related recordings, screenshots, or snapshots, the corresponding icons will be displayed synchronously.
③	Click to view all call records, call recordings, snapshots, and screenshots with the contact.
④	Supports fuzzy matching search; enter a partial segment of the number to filter out relevant call records.
⑤	Click to select and delete the selected call records.

Table 18 - Four Call Types

	Missed Call Log		Incoming Call Log
	Outgoing call Log		Forward call Log

In the Call Log interface, click Icon ③ to view the detailed call records with the corresponding contact, as shown in the figure below:

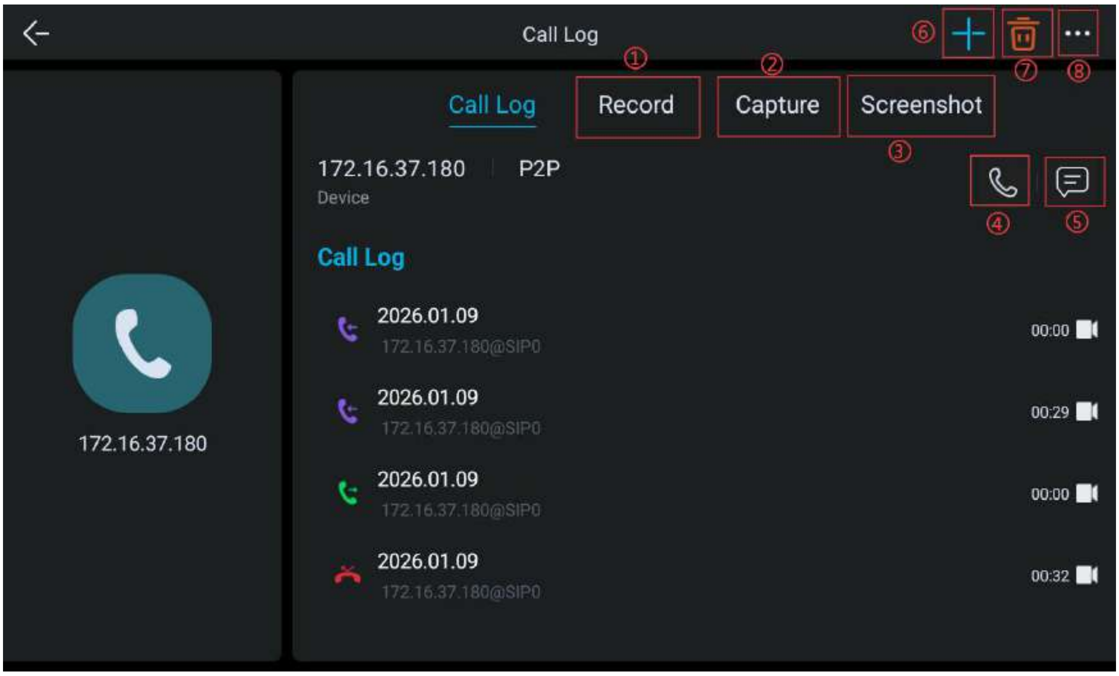


Figure 94 - Call log Details

Table 19 - Call Log Details

Index	Function
①	View the call recordings with this number.
②	View the snapshot photos taken during calls with this number.
③	View the screenshot photos captured during calls with this number.
④	Click to directly initiate a call to the number corresponding to the current call record.
⑤	Click to send a short message to this number.
⑥	Click to add this number to the phonebook.
⑦	Click to select and delete all call records with this contact.
⑧	Click to add this number to the Allowed Incoming Calls List / Blocked Incoming Calls List / Allowed Outgoing Calls List.

When there are missed calls, a quantity icon will display a reminder on the standby interface.

Figure 95 - Missed Call Prompt

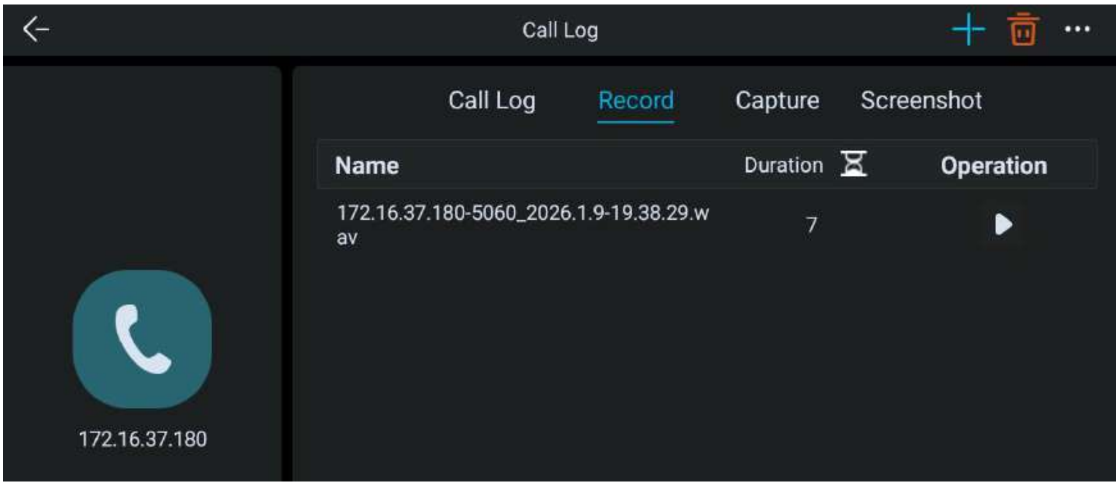


Figure 96 - Recording of the Call With Corresponding Number

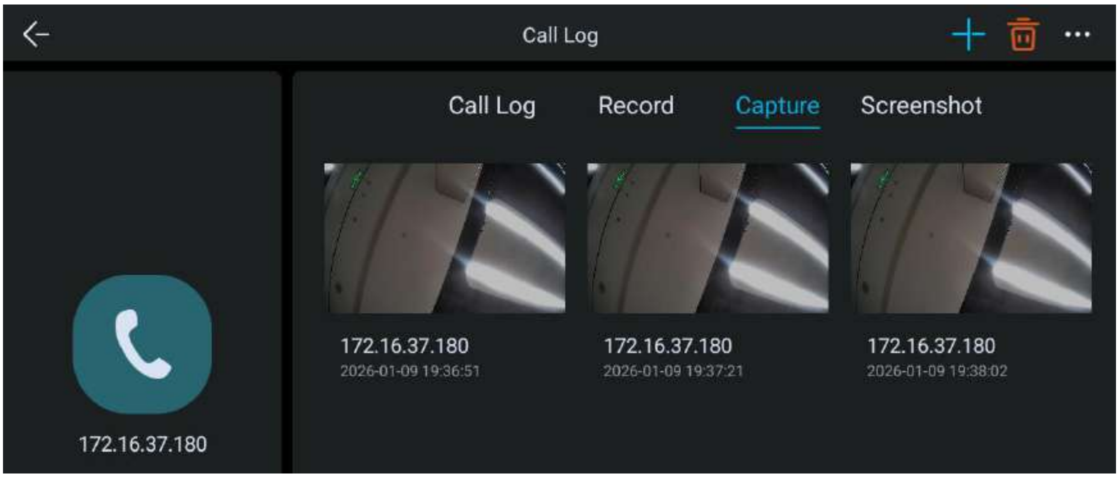


Figure 97 - Capture Record With Corresponding Number

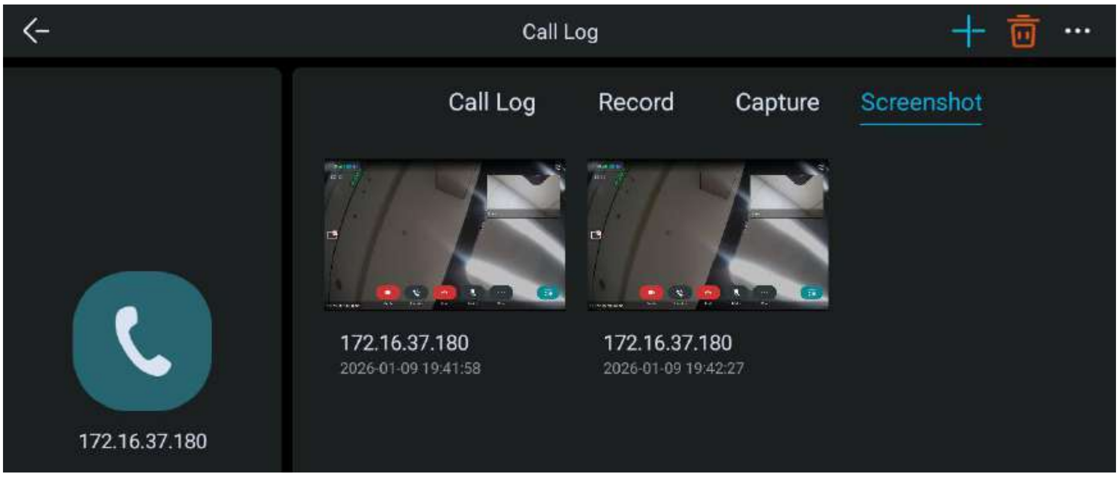


Figure 98 - Screenshot Record With Corresponding Number

10.4 Call Record

In the [**Call**] >> [**Call Record**] interface, you can view the recordings of calls with all numbers. Click the icon ② on the right side of a recording to play the audio of that recording. Click the icon ③ in the upper right corner to select and delete recording audio files.

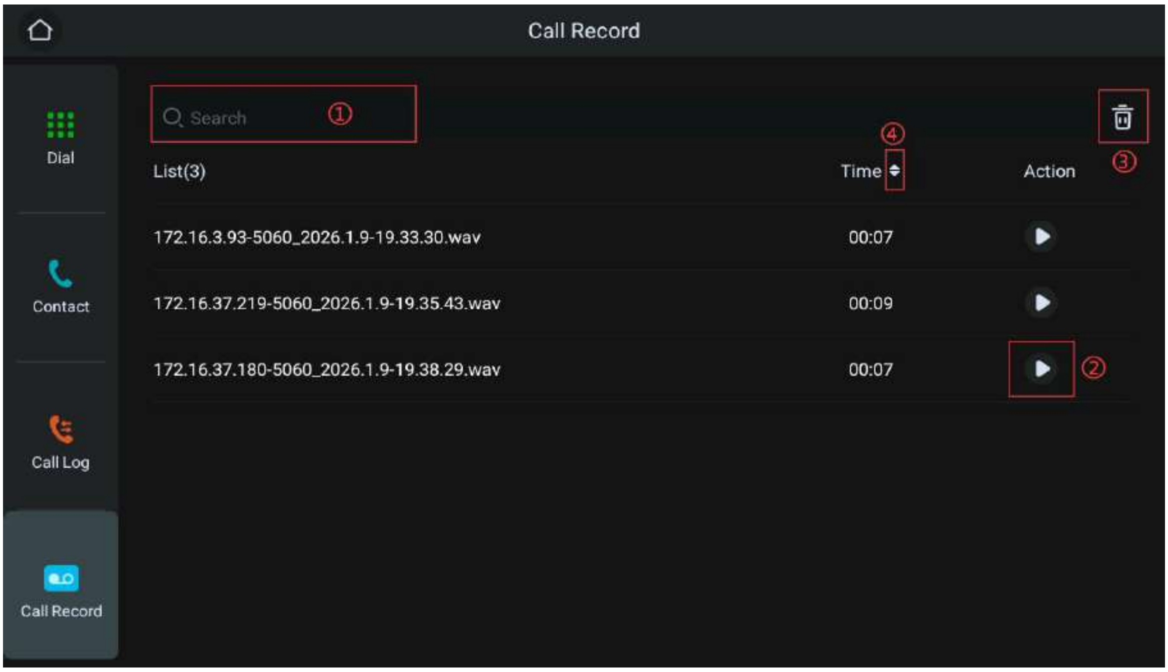


Figure 99 - Call Record

Table 20 - Call Record

Index	Function
①	Enter keywords to quickly search for the target call recordings.
②	Click to preview the call recording file of the corresponding entry.
③	Click to select multiple recording files and perform the delete operation.
④	Click to sort the recording list by recording duration (ascending/descending order).

11 Console

The console displays the total number of extensions, the number of offline extensions, and the operating status of each extension. After selecting extensions, you can quickly perform corresponding operations on them by clicking the function buttons on the right, such as Call, Conference, Intercom, Paging, V-Monitor, and Monitor.

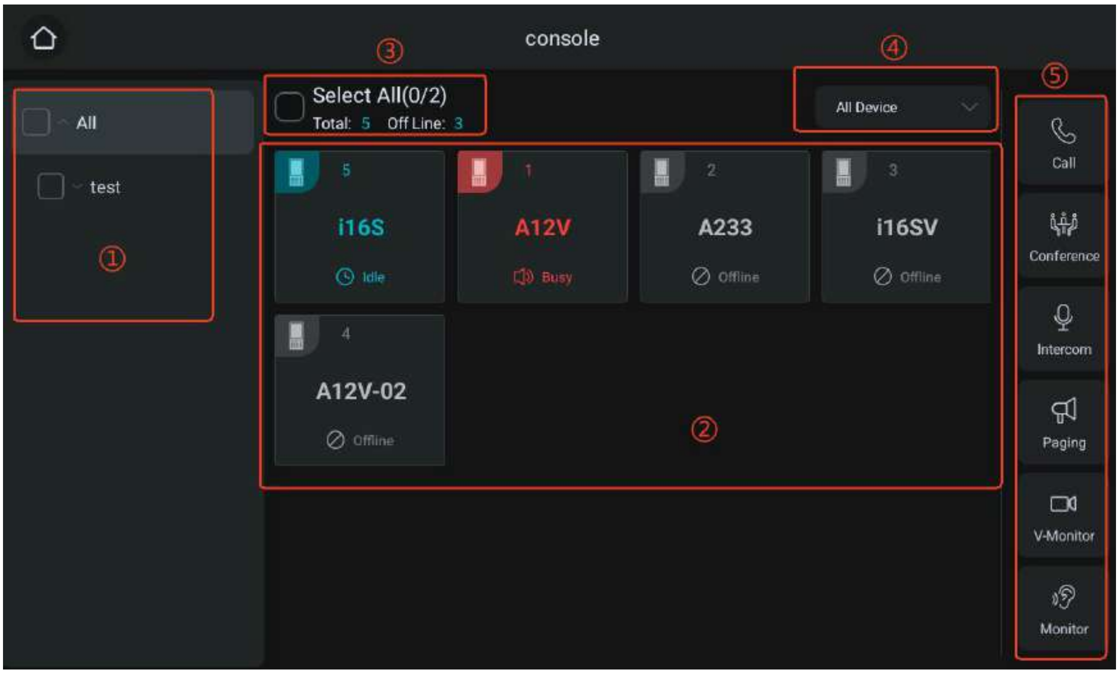


Figure 100 - Console

Table 21 - Console

Index	Function
①	Displays device groups; all devices in a group can be selected at once.
②	Displays connected devices and their statuses; click a device card to select it.
③	Allows filtering by the online status of devices.
④	Displays the number of online/offline devices; all online devices can be selected at once.
⑤	Click to perform the corresponding operation on the selected extensions.

11.1 Call

Click this button to initiate a call to the selected single extension.

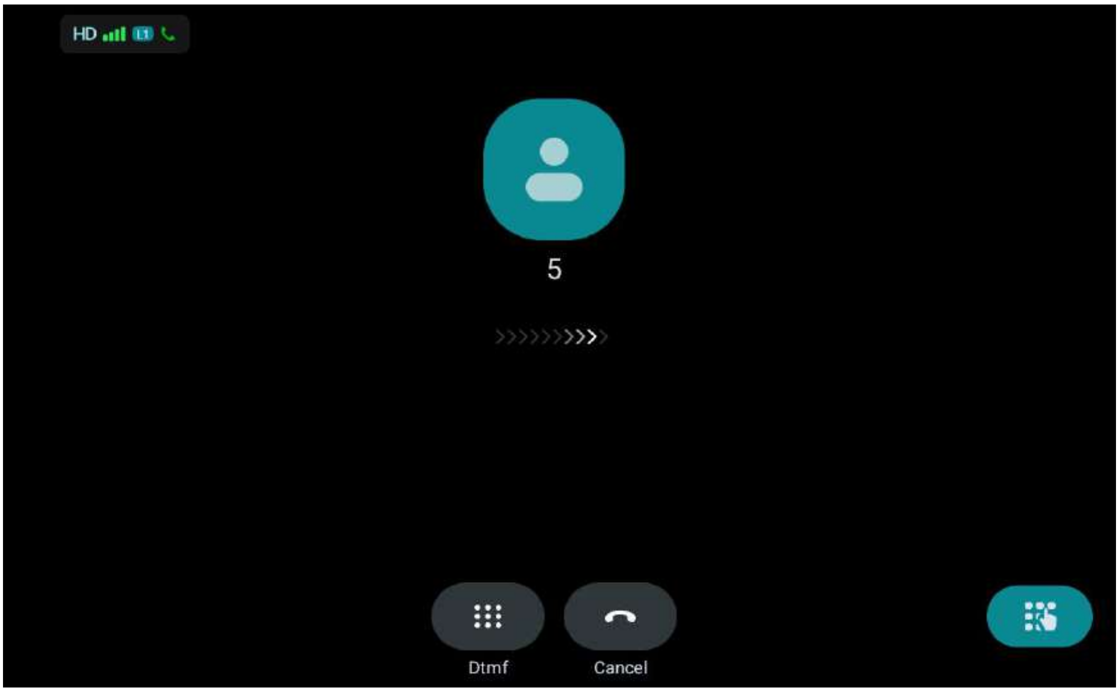


Figure 101 - Call Extension

11.2 Conference

After selecting multiple extensions, click this button to initiate calls to the selected extensions simultaneously and establish a conference. The extensions will automatically join the conference after answering the calls.

Figure 102 - One-Click Conference

11.3 Intercom

Click this button, and the host will initiate an intercom call to the selected single extension, which will automatically answer the intercom call.

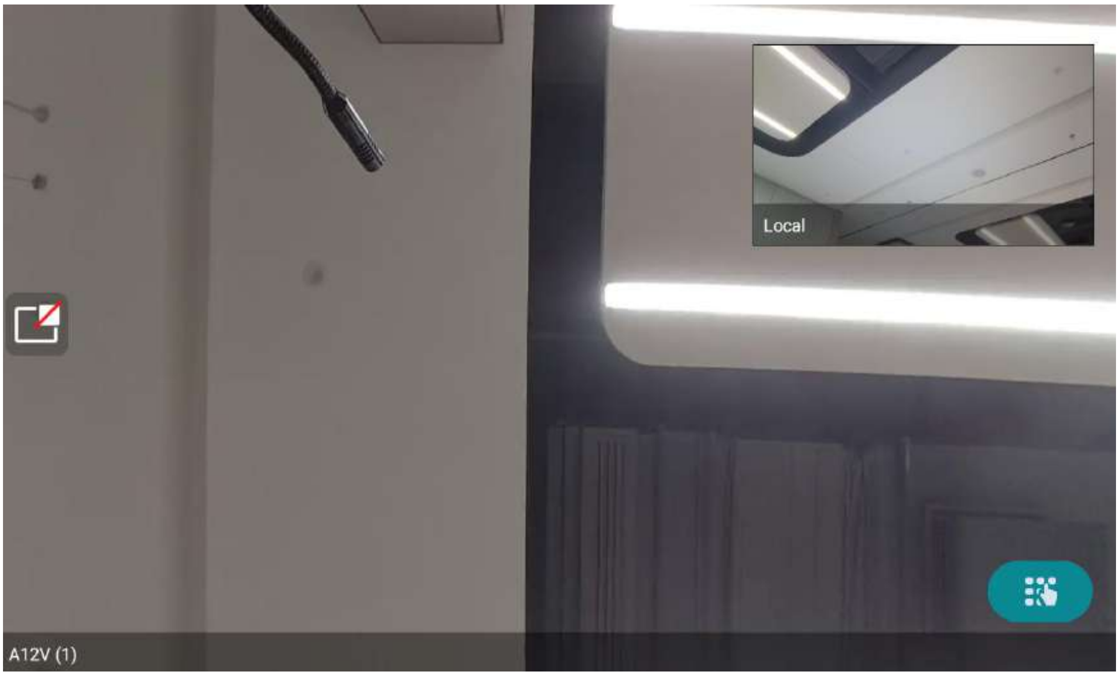


Figure 103 - Initiate Intercom

11.4 Paging

Click this button to start one-way intercom broadcast: the user can speak through a handset or gooseneck microphone, and the selected extensions will play the broadcast content simultaneously.

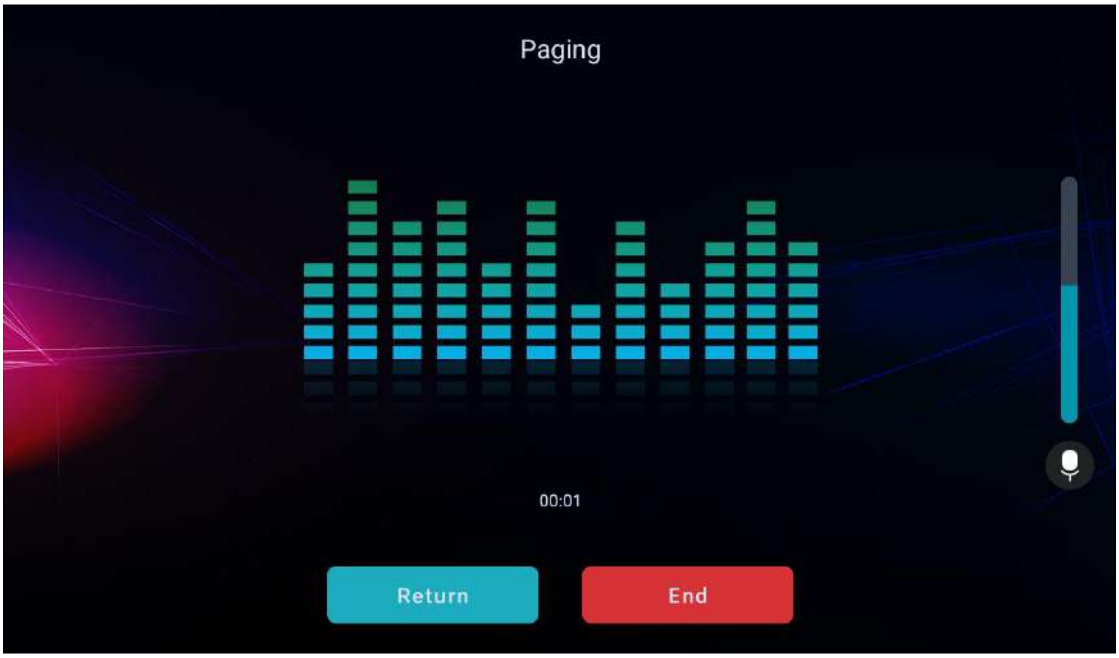


Figure 104 - Paging

11.5 V-Monitor

Use this function to monitor the live video of the selected extensions. If more than four extensions are selected, the monitoring interface will automatically display in video tour mode, and the tour interval can be customized in [**Settings**] >> [**Monitoring**].

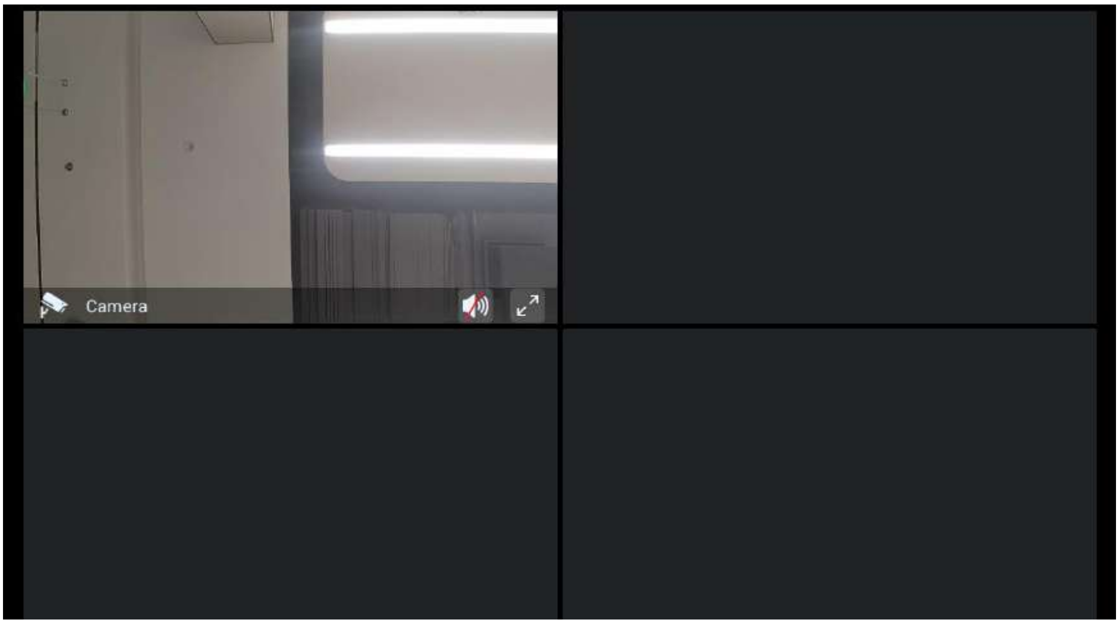


Figure 105 - Video Monitor

11.6 Monitor

This function enables real-time audio monitoring of the environment around selected extensions, allowing operators to listen in without establishing a voice call.

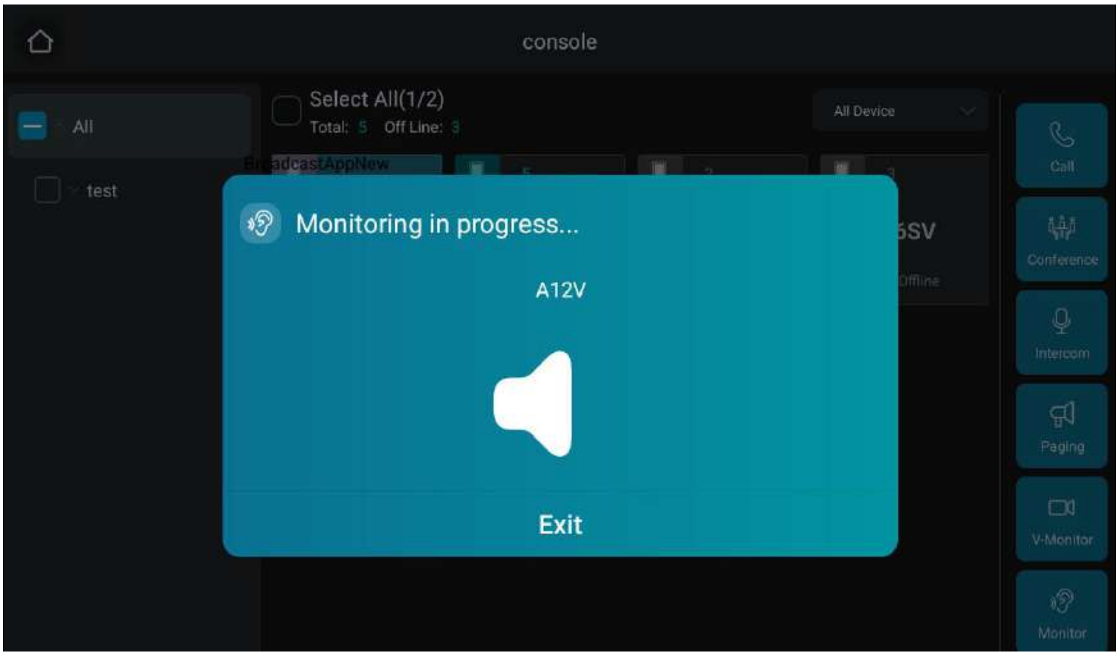


Figure 106 - Monitoring

12 Task

12.1 Real-time Task

Real-time tasks support the creation of three types of tasks: Mic, Music Broadcast, and One-Click Conference. After a task is created, it can be manually triggered for execution at any time. Enter the [**Tasks**] >> [**Real-Time Task**] interface to configure relevant real-time tasks.

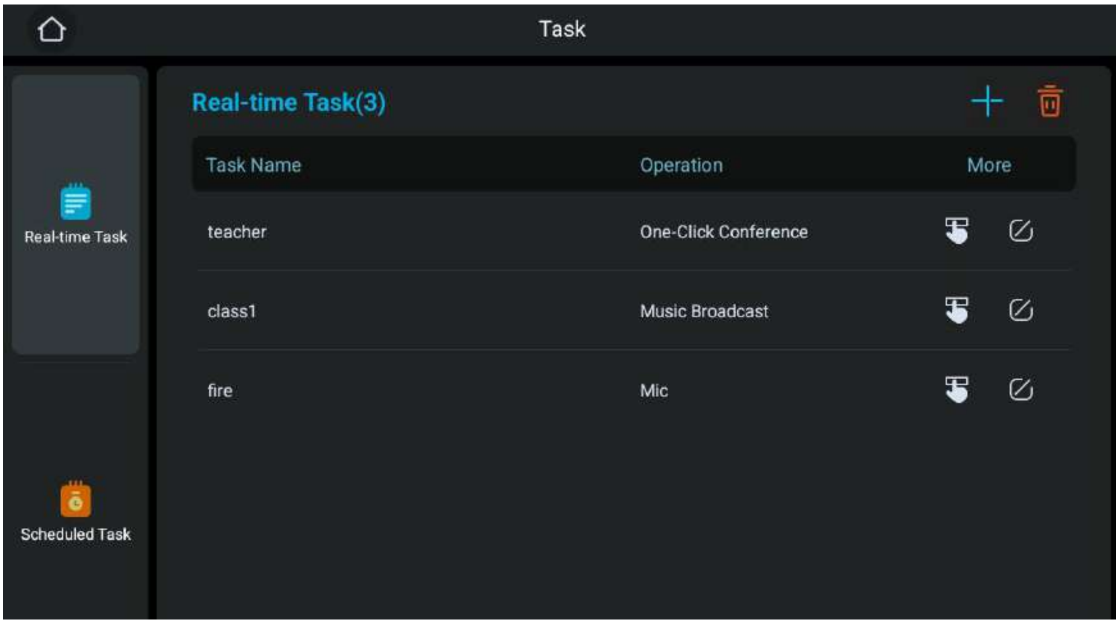


Figure 107 - Real-Time Task

Click the button in the upper right corner to create a real-time task, fill in the task name, then click "**Operation**" to select the real-time task type. After completing the specific configuration and saving it, the task will be displayed in the list. Press the button on the right side of the task to execute it.

Click the button in the upper right corner to select and delete the created tasks.

12.1.1 Mic

Initiate a real-time intercom broadcast to the selected extensions for fast and efficient voice message delivery.

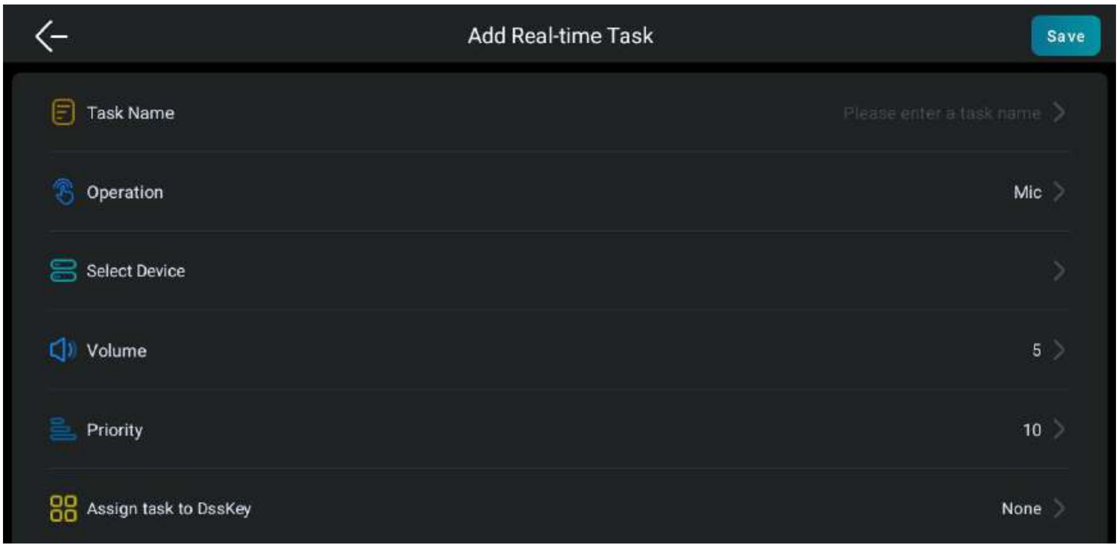


Figure 108 - Real-Time Task - Mic

Table 22 - Real-Time Task - Mic

Parameter	Description
Select Device	Select the extensions that need to receive the broadcast.
Volume	Set the volume level for the broadcast.
Priority	Set the task execution priority. The larger the value, the higher the priority, ensuring that important announcements are executed first.
Assign task to Dsskey	You can assign the task to a dsskey. Afterwards, you can quickly execute the task by pressing the corresponding dsskey.

12.1.2 Music Broadcast

Initiate a music broadcast via the selected extensions, supporting local audio file playback or audio input from external devices via Line-in.

1. Before initiating a music broadcast, you first need to confirm the audio source of the broadcast.

Two audio sources are supported for music broadcast. Select the appropriate one as needed and configure the corresponding parameters:

（1）**Local Audio** (supports audio files in mp3 and wav formats)

- **Import from Media Library:** In the [**Settings**] >> [**Media Library**] >> [**Audio**] interface, import audio files via a computer or U-disk. Please refer to 14.4.1 Audio.
- **Use Audio Files in SD Card or USB drive:** Create a "**Music**" folder in the root directory of the SD card or U-disk, and place the audio files in this folder. When the SD card or U-disk is connected to the device, the audio files in the "**Music**" folder will be automatically displayed in the file list.

（2）**Line-in**

After connecting an external audio device (such as a player or a microphone) to the host via the Line-in interface, you can directly select Line-in as the audio source to initiate a real-time music broadcast to the target extensions.

2. Add a music broadcast task, and set the extensions to be broadcasted, audio source, playback mode, etc. The configurations and their descriptions are as follows:

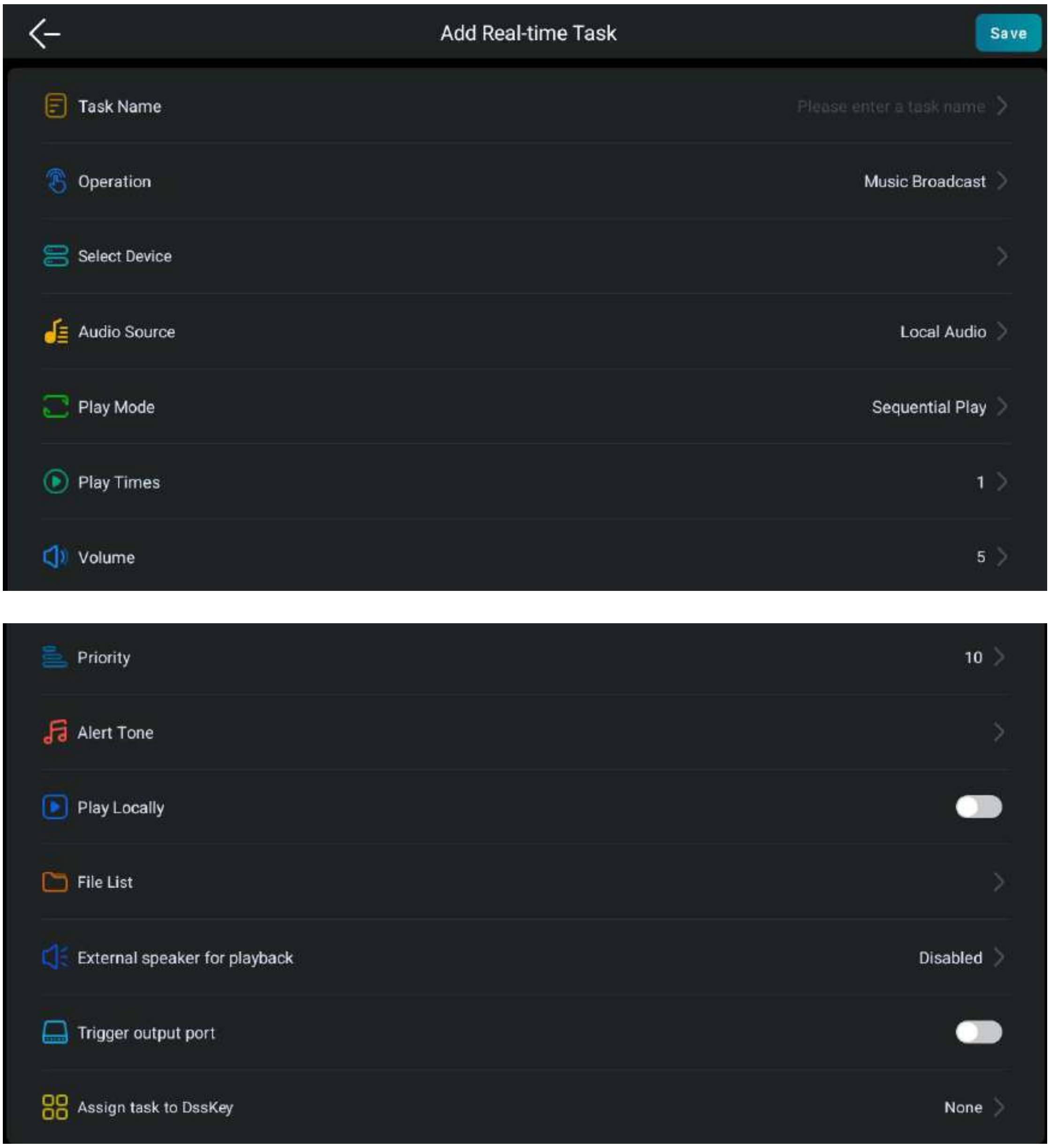


Figure 109 - Real-Time Task - Music Broadcast

Table 23 - Real-Time Task - Music Broadcast

Parameter	Description
Select Device	Select the extensions that need to receive the music broadcast.
Audio Source	Optional: Local Audio or Linein.

Play Mode	Optional: Sequential Play or Random Play.
Play Times	Set the number of playback times for the audio file. Single playback or multiple loop playback is supported.
Volume	Adjust the volume of the music broadcast to meet the auditory needs of different scenarios.
Priority	Set the task execution priority. The larger the value, the higher the priority.
Alert Tone	Configure the alert tone before the broadcast to remind the extension side that the broadcast is about to start.
Play Locally	When enabled, the host and extensions will play the audio simultaneously, making it easy for the host side to monitor the broadcast content.
File List	Only required for Local Audio. Select the uploaded/imported local audio files (supports audio files in the media library and the "Music" folder of the USB flash drive/SD card).
External speaker for playback	Disabled: When local playback is enabled, the device audio is output through the hands-free speaker.
	ON: When local playback is enabled, the device audio is output through the external speaker connected to the Line-out interface.
	speaker & Lineout: Output through both the device hands-free speaker and the external speaker simultaneously.
Trigger output port	When this configuration is enabled, the output port will be triggered synchronously during task execution, which can be used to control external devices such as electric locks and alarms.
Assign task to Dsskey	You can assign the task to a dsskey. Afterwards, you can quickly execute the task by pressing the corresponding dsskey.

3. After completing the configuration, click "**Save**" in the upper right corner, and the task will be displayed in the task list.

After adding a task, you can click the button on the right to start the music broadcast. During task execution, you can adjust the broadcast volume of the extensions in real time through the volume slider in the task window.

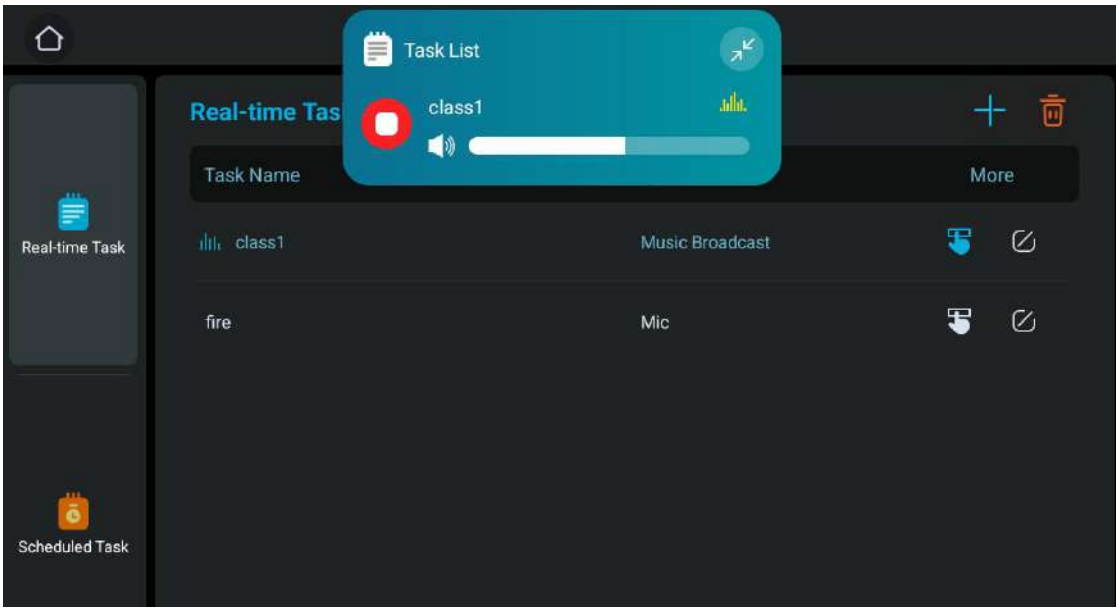


Figure 110 - Music Broadcast Interface

12.1.3 One-Click Conference

The One-Click Conference function supports quickly initiating a voice conference with multiple extensions participating, enabling real-time multi-party audio interaction between the host and extensions, as well as between extensions themselves.

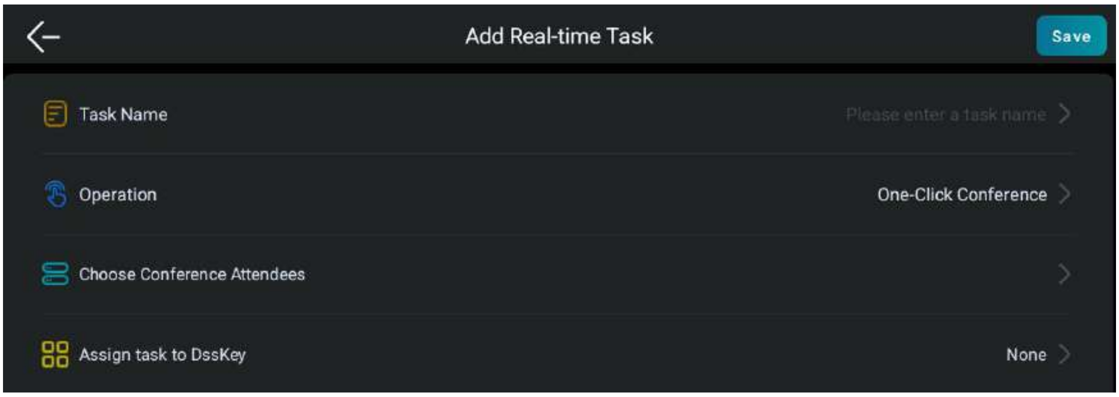


Figure 111 - Real-Time Task - One-Click Conference

Table 24 - Real-Time Task - One-Click Conference

Parameter	Description
Choose Conference Attendees	Select the extensions that need to participate in the conference.
Assign task to Dsskey	You can assign the task to a shortcut key. Afterwards, you can quickly execute the task by pressing the corresponding shortcut key.

12.2 Scheduled Task

12.2.1 Add Plan

A ”Plan“ is a logical unit used to centrally manage scheduled tasks within the Scheduled Task feature.

It allows multiple tasks serving the same use case or operational scenario to be grouped together, enabling unified control over their execution

Addition Steps:

1. Enter the [**Task**] >> [**Scheduled Tasks**] interface and click the "+" button in the upper right corner to enter the "Add Plan" interface.

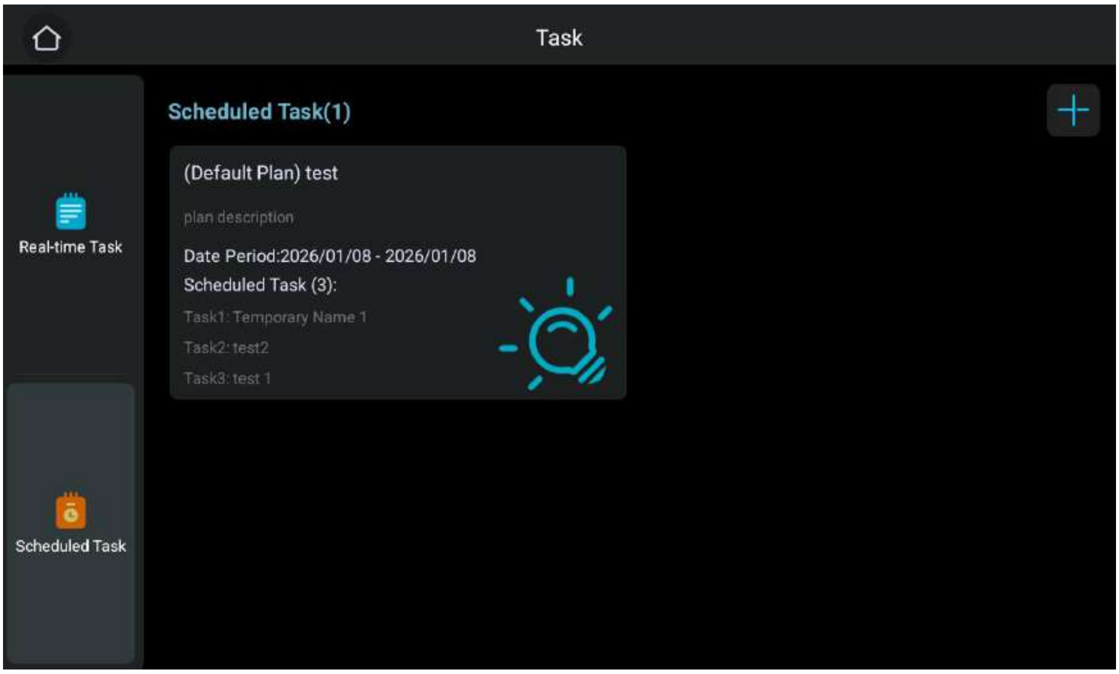


Figure 112 - Scheduled Task

2. Fill in the following configurations in sequence and click **[Save]** in the upper right corner to add a new plan.

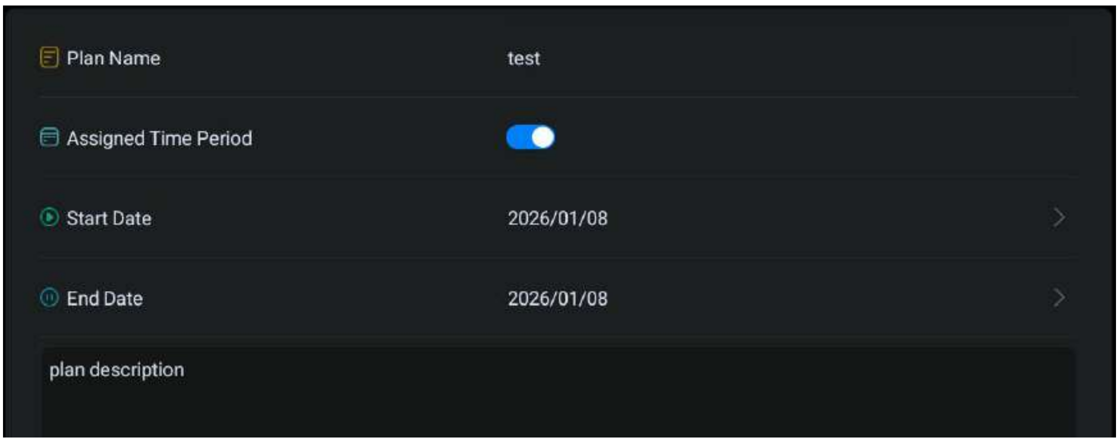


Figure 113 - Add Plan

Table 25 - Add Plan

Parameter	Description
Plan Name	Required field. Used to identify the plan.
Assigned Time Period	When enabled, you can set the effective time range of the plan. Only within this time period can the tasks under the plan be triggered and executed; when disabled, the tasks have no time limit.
Start Date	When Specified Time Period is enabled, it is used to set the start date of the plan's validity.
End Date	When Specified Time Period is enabled, it is used to set the end date of the plan's validity.
Plan Description	Can be used to supplement the description of the plan's purpose and applicable scenarios.

3. Click on the created plan to enter the **[Details of the Plan**] interface. Press the light bulb icon in the upper right corner (see icon ② in Figure 114) to activate the plan. Only after activation can the tasks under the plan be triggered and executed.

4. In the [**Details of the Plan**] interface, click the button  on the right side of the task list (see icon ④ in Figure 114) to add a task. A task mainly consists of two configurations: Trigger and Operation. The "**Trigger**" is used to set the trigger conditions for task execution; when the set conditions are met, the task will be triggered and executed. The "**Operation**" is used to set the specific actions to be executed after the task is triggered.

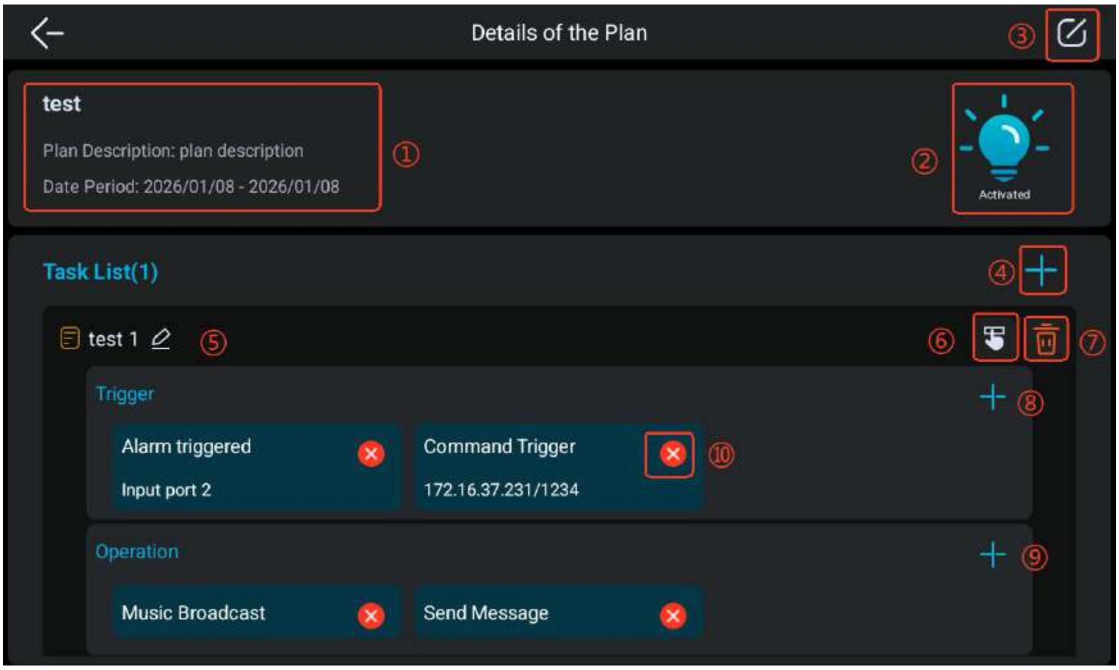


Figure 114 - Details of the Plan

Table 26 - Details of the Plan

Index	Function
①	Displays basic plan information: name, description, and time period.
②	Controls the activation/deactivation status of the current plan: When enabled, tasks under the plan can be triggered and executed normally; when disabled, all associated tasks are suspended.
③	Click to edit the plan configuration.
④	Click to add a new scheduled task under the current plan.
⑤	Click to edit the task name.
⑥	Click to manually trigger the execution of the current task.
⑦	Click to delete the currently selected task. After the operation, the task will be removed from the plan and cannot be recovered.
⑧	Click to configure trigger conditions for the current task and control the task execution timing.
⑨	Click to configure execution operations for the current task and control the operations to be performed after the task is triggered.
⑩	Delete the corresponding trigger or operation individually.

12.2.2 Add Task

12.2.2.1 Trigger

Press the button on the right of "Trigger" to configure the task trigger conditions. Four trigger modes are supported: Timed Trigger, DTMF Trigger, Command Trigger, and Alarm Trigger.

- **Timed Trigger**

You can set the task to be automatically triggered and executed within a specified time period to meet the requirement of scheduled execution for periodic tasks. Three types of task types are available for selection as needed:

- **Everyday:** You need to configure the start date, end date, start time, and end time of the broadcast. After configuration, the task will be automatically triggered daily within the set time interval during the period from the start date to the end date. triggered at a set time interval every day from the set start date to the end date.
- **Weekly:** First, check the days of the week on which the broadcast needs to be executed (e.g., only Tuesday, or Tuesday and Thursday), then configure the start date, end date, start time, and end time of the broadcast task. After configuration, the task will be automatically executed within the set time interval on the corresponding days of the week within the specified time period.
- **One Day:** Select a specific date, then configure the start time and end time of the broadcast. After configuration, the task will be automatically triggered within the set time interval on the selected

date.

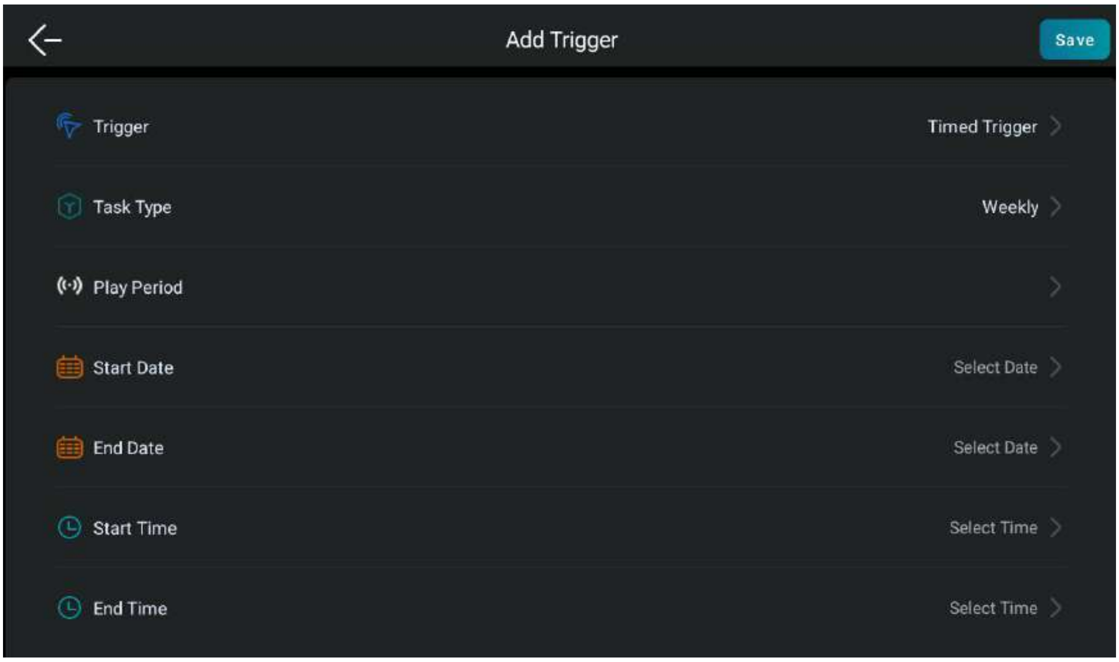


Figure 115 - Timed Trigger

- **DTMF Trigger**

When in a call with the corresponding number, the remote party can press the configured DTMF keys to trigger the task. This is applicable to remote command triggering scenarios.

1. Fill in the specific configurations, and enter the number and the corresponding DTMF password.

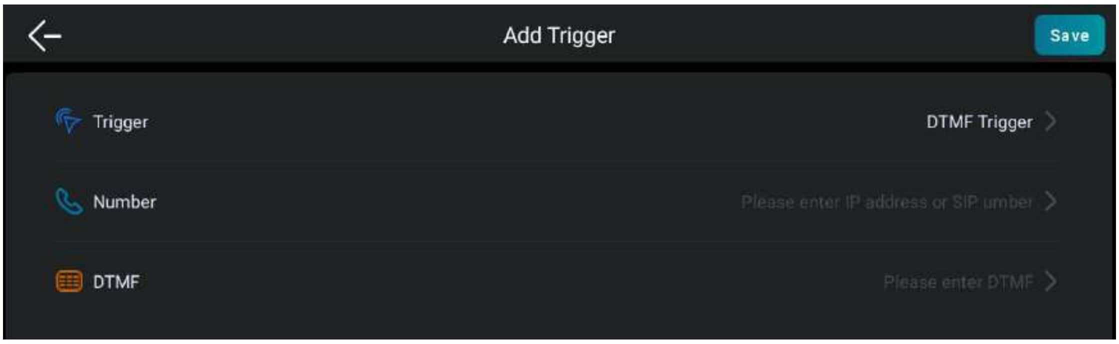


Figure 116 - DTMF Trigger

Table 27 - DTMF Trigger

Parameter	Description
Number	Enter the IP or SIP number of the corresponding trigger device. This field can be left blank.
DTMF	Set the DTMF code used to trigger the task.

2. After configuration, when the phone establishes a call with the local device using an IP or SIP number, pressing the preset DTMF code + “#” will trigger the corresponding task.

Example: If the DTMF trigger configuration number is 803201 and the DTMF code is 1234, then when the other phone with the number 803201 calls the local device and establishes a call, the task will be triggered after entering the DTMF code 1234#.

Note! If the "Number" field is left blank, you can use any number to make a call to the local device and press the preset DTMF password to trigger the corresponding task in the same way.

- **Command trigger**

You can trigger the corresponding task by sending a specific HTTP URL command via the corresponding IP address, which is applicable to remote command triggering scenarios.

1. Fill in the number and command, and press “Save” in the upper right corner after completing the configuration.

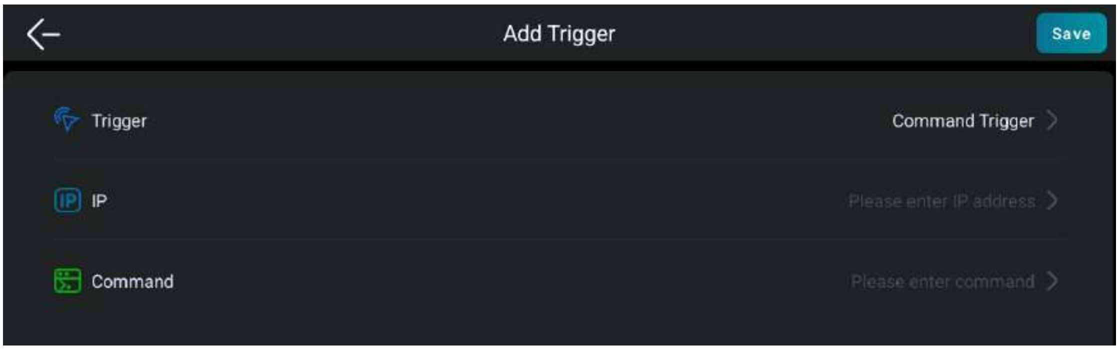


Figure 117 - Command Trigger

Table 28 - Command Trigger

Parameter	Description
IP	Enter the IP address of the device that needs to send the trigger command. This field can be left blank.
Command	Enter the specific command used to trigger the task.

2. After completing the configuration, you can send a URL command via the configured IP address: http://[Username]:[Password]@[Local IP Address]:8080/cgi-bin/bcast?code=[Command] to trigger the execution of the corresponding task. The username and password are the same as those for web page login, with the default being admin/admin. You can view and add users in 17.3 System >> Account.

Example: The phone IP address is 172.16.37.219, and the computer IP address is 172.16.37.231. Configure the command-triggering IP as the computer IP address and the trigger command as 123; enter http://admin:admin@172.16.37.219:8080/cgi-bin/bcast?code=123 in the computer browser and press Enter to trigger the task.

Note!

1. ***When the IP field is left blank, the task can be triggered by sending a URL command from any IP address.***
2. ***When using a URL command to trigger a music broadcast, if the audio is still playing, triggering the URL command again will stop the music broadcast.***

- **Alarm Trigger**

Triggering the corresponding alarm input port will activate the device's alarm function and trigger the associated scheduled task simultaneously. This is applicable to security anomaly early warning scenarios. For details about the input port alarm function, please refer to 14.3.3 Alarm Logs.

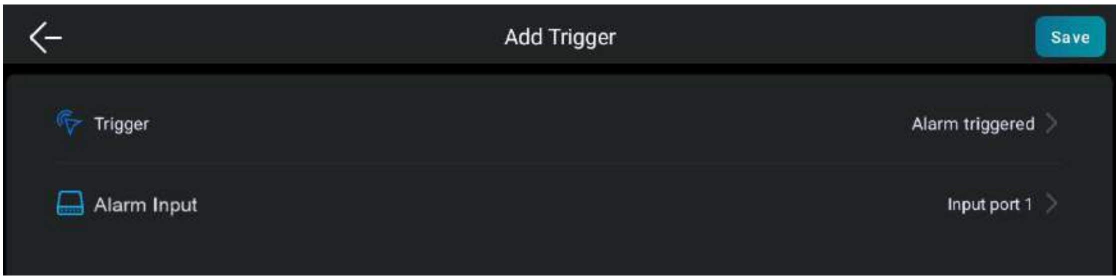


Figure 118 - Alarm Trigger

Note! When an music broadcast is triggered via an input port, the music broadcast will stop after the alarm is cleared by entering the reset code (default: 1234).

12.2.2.2 Operation

Press the button on the right of "**Operation**" to set the specific operation to be executed when the task is triggered. Two action types are supported: Music Broadcast and Send Message.

- **Music Broadcast**

When the operation is set to Music Broadcast, the selected extensions will play the specified audio after the task is triggered. In scheduled tasks, Music Broadcast only supports local music broadcast. For the configuration method, please refer to section 12.1.2 Music Broadcast.

When configuring music broadcasting in the scheduled task settings, the user can additionally set the "Play Interval" parameter to control the time duration between two consecutive audio playbacks, which is suitable for scenarios such as loop playback.

Figure 119 - Scheduled Tasks - Music Broadcast

- **Send Message**

When the "**Operation**" is set to Send Message, the device will send a short message to the configured number via the corresponding line after the task is triggered. After filling in the short message receiving number,short message content and selecting the line, press “**Save**” in the upper right corner to complete the setup.

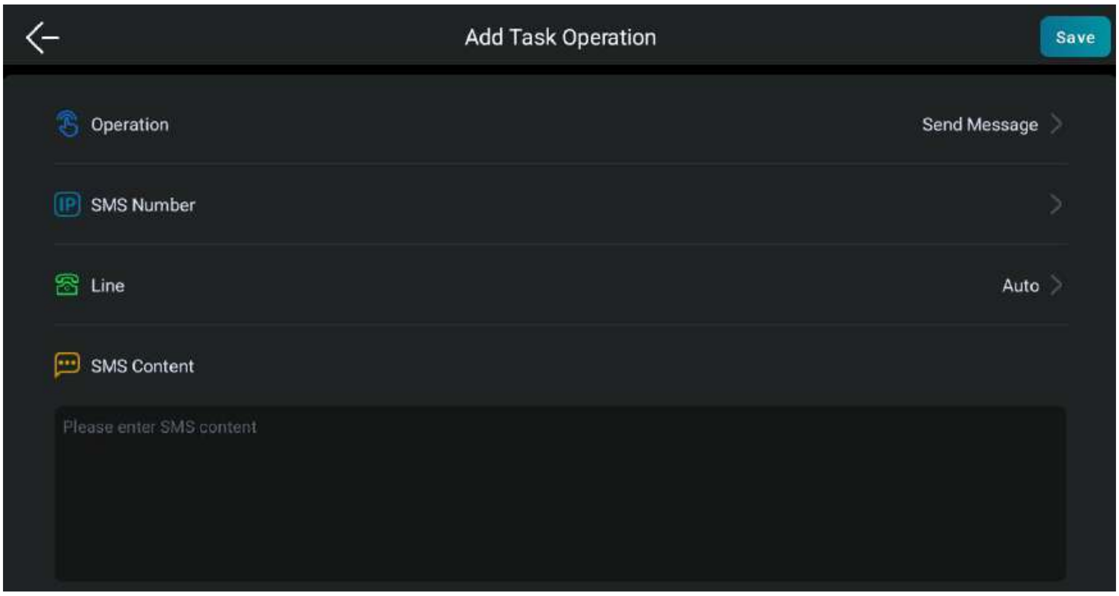


Figure 120 - Scheduled Task - Send Message

Table 29 - Scheduled Task - Send Message

Parameter	Description
SMS Number	Used to set the target number for receiving short messages.

Line	The communication line used for sending short messages. You can select the default line \ P2P \ specific line.
SMS Content	Used to fill in the specific content of the short message.

After configuring the triggers and operations, the preset tasks will be executed automatically when the corresponding trigger conditions are met. You can also manually execute them by pressing the icon  on the right side of the task (see icon ⑥ in Figure 114).

13 Monitor

13.1 Monitor

13.1.1 Monitoring Device Management

It can automatically scan and deploy video extensions (such as video intercoms and video doorphones) in the network segment, and also supports manual addition of monitoring devices.

In the [**Settings**] >> [**Monitoring**] interface, you can add, edit, and delete monitoring devices. It also supports setting the enablement status and rotation time for the rotation display of the monitoring interface.

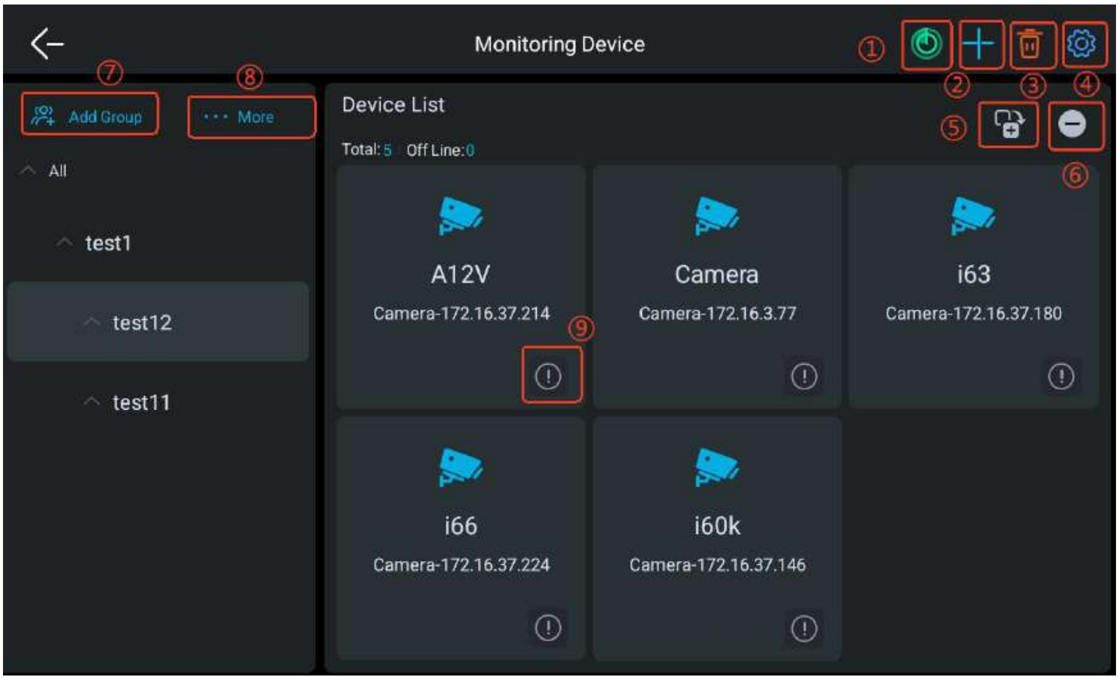


Figure 121 - Monitoring Device Management

Table 30 - Monitoring Device Management

Index	Function
①	Click to automatically scan for monitoring devices in the current network segment.
②	Manually add monitoring devices, applicable to cross-segment devices.
③	Click to remove obsolete, offline, invalid, or incorrectly added devices from the management list.
④	Click to set the enablement status and rotation time for the automatic rotation display of the monitoring interface.
⑤	Click this button to add the monitoring device to the current group. If the device is not included in the parent group, it will also be displayed in the parent group synchronously.
⑥	Click to remove the selected device from the group.

⑦	Click to add a lower-level subgroup to the current group.
⑧	Click to edit or delete the current group.
⑨	Click to display the detailed information of the device. You can click the edit button to modify the device information.

13.1.1.1 Adding Monitoring Devices via Scanning

The system will automatically scan for devices in the network segment and add them to the monitoring device list. Press icon ① (see Figure 121) to view the scanned devices.

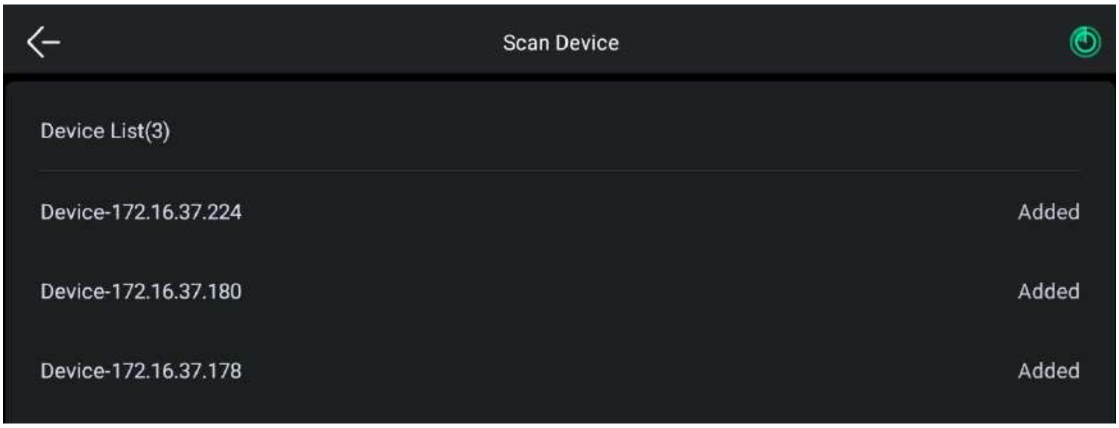


Figure 122 - Scanning Device interface

13.1.1.2 Manually Add Monitoring Device

Press icon ② (see Figure 121) to enter the Manual Device Addition page. After filling in all the required configurations as instructed (refer to Table 31 - Manually Add Monitoring Device for configuration details), press the corresponding button in the upper right corner to complete the device addition successfully. Supported device manufacturers include FANVIL, TOPSEE, XM, HIKVISION, DAHUA, AXIS, and DIGITAL_WATCHDOG.

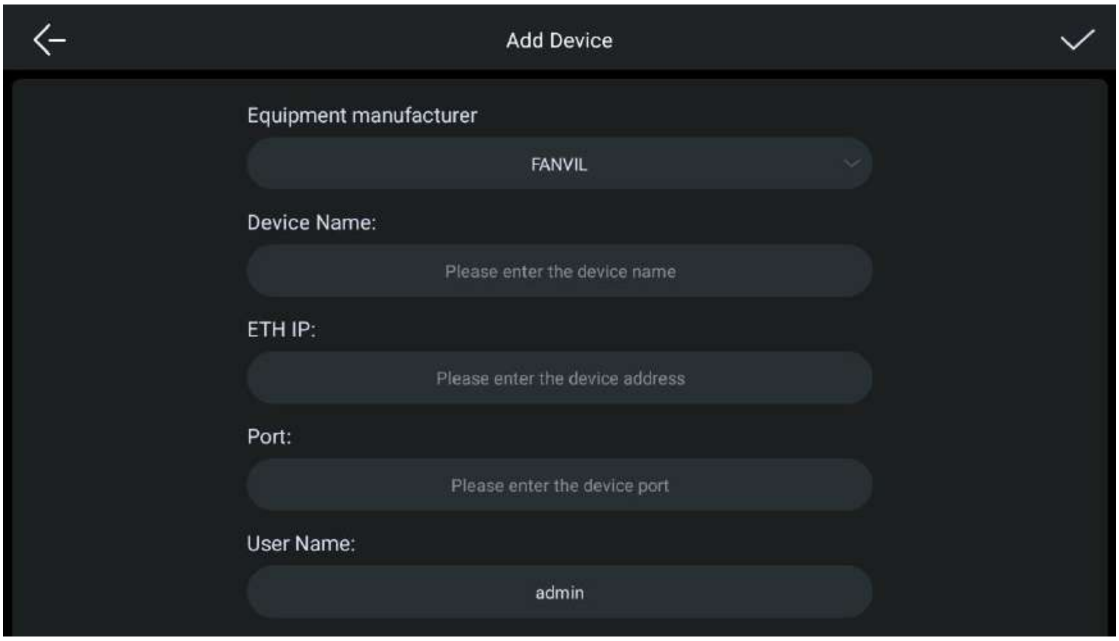


Figure 123 - Manually Add Monitoring Device

If the manufacturer of the monitoring device is not in the supported list, you can select "Customize" as the "**Equipment manufacturer**", and fill in the corresponding RTSP stream address, username, and password of the monitoring device to complete the configuration.

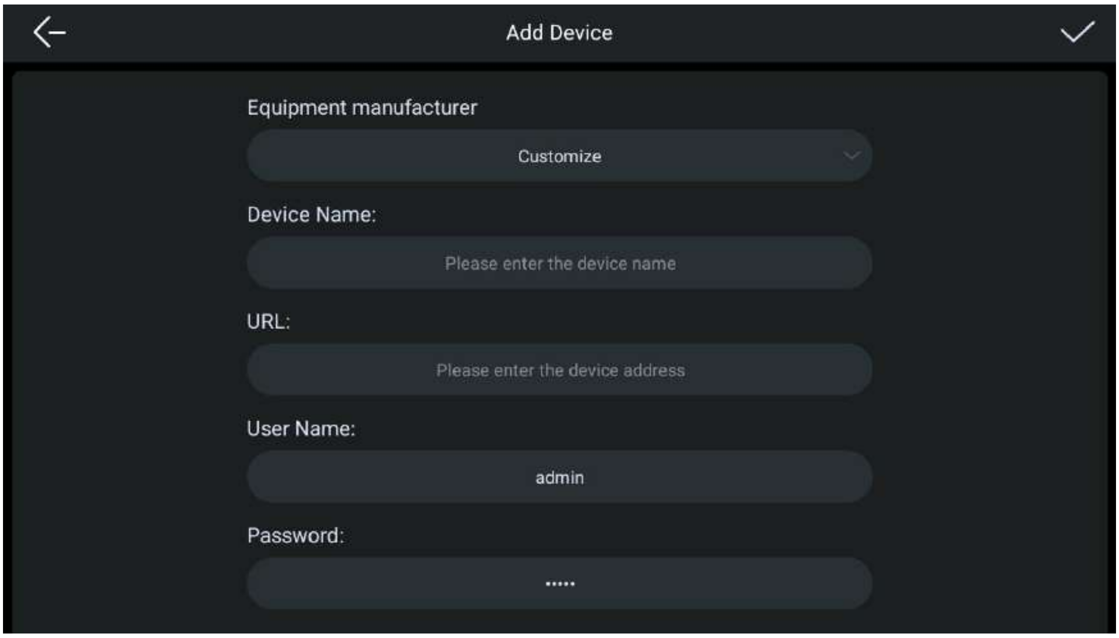


Figure 124 - Manually Add Monitoring Device-Customize

Table 31 - Manually Add Monitoring Device

Parameter	Description
Equipment manufacturer	Select the brand corresponding to the monitoring device. The default is "FANVIL".

Device name	Enter a custom name to identify the device.
URL	When "**Customize**" is selected as the device manufacturer, you can fill in the RTSP stream address of the monitoring device here.
ETH IP	Fill in the IP address of the monitoring device.
Port	Enter the port of the RTSP stream of the monitoring device.
User Name	Enter the RTSP authentication username of the monitoring device.
Password	Enter the RTSP authentication password of the monitoring device.

13.1.1.3 Monitor Rotation Configuration

Press the gear icon in the upper right corner (see icon ④ in Figure 121) to configure the enablement/disablement of automatic rotation display and the rotation time in the interface.

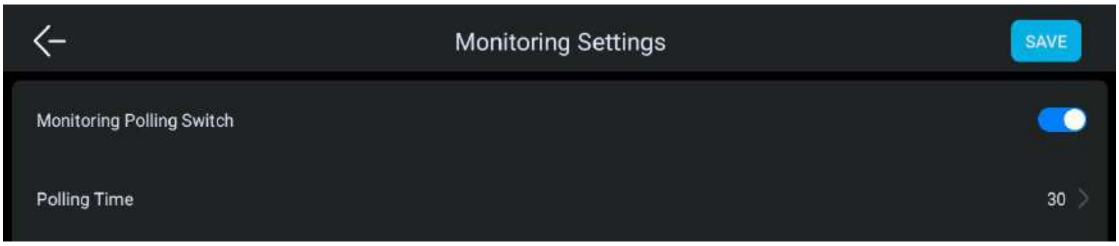


Figure 125 - Monitoring Rotation configuration

Table 32 - Monitoring Rotation Settings

Parameter	Description
Monitoring Polling Switch	Controls the enablement status of the "**Monitor Rotation**" function. When enabled, if the number of devices exceeds 4, the interface will automatically turn pages to display other monitoring interfaces according to the set rotation time. When disabled this function, the interface will display fixed content, and you need to manually turn pages to view the monitoring interfaces of other devices.
Polling Time	Sets the interval duration for automatic page turning of the interface.

13.1.2 Monitor

1. Access the [**Monitoring**] interface and select the device to be monitored.

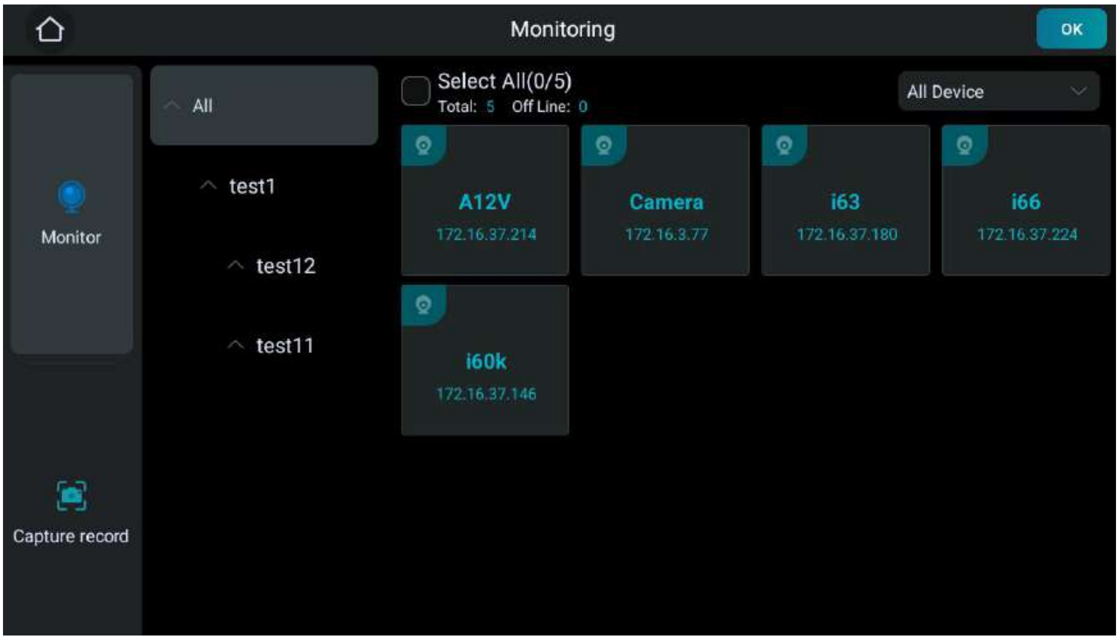


Figure 126 - Monitoring And Selecting Device

2. Press **[OK]** to enter the monitoring interface.

When the number of selected monitoring devices exceeds four, the monitoring interface will automatically switch to video tour mode (2x2 grid layout). The tour interval can be custom-configured via the [**Settings**] >> [**Monitoring**] menu (Please refer to 13.1.1.3 Monitoring Rotation Settings for details).

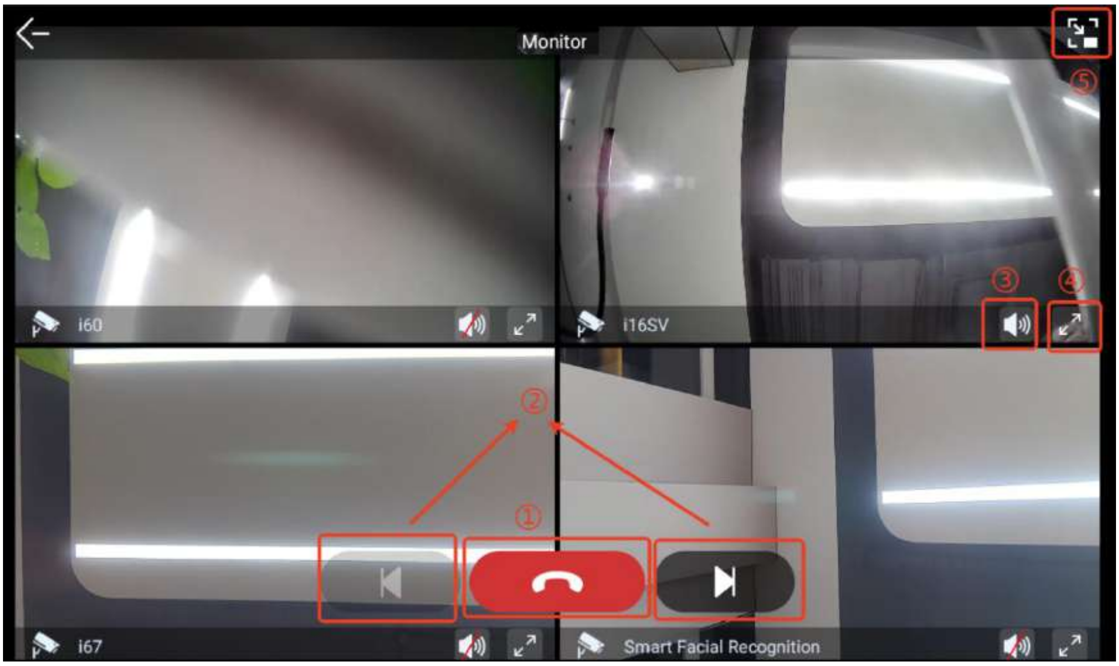


Figure 127 - Small-Screen Monitoring Interface

Table 33 - Small-Screen Monitoring Interface

Index	Function
①	Click to exit the monitoring preview interface and return to the device list.
②	Click this button to turn pages for viewing when more than four monitoring devices are selected.
③	Click to monitor the device and play the sound collected by the device.
④	Switch the current monitoring screen to full-screen display mode and enlarge the screen to view details.
⑤	Minimize the monitoring interface and shrink the current monitoring window to the background, facilitating the simultaneous operation of other functions.

3. Press the full-screen display button in the small-screen monitoring interface (see icon ④ in Figure 127) to enter the full-screen monitoring interface, as shown in the figure below.

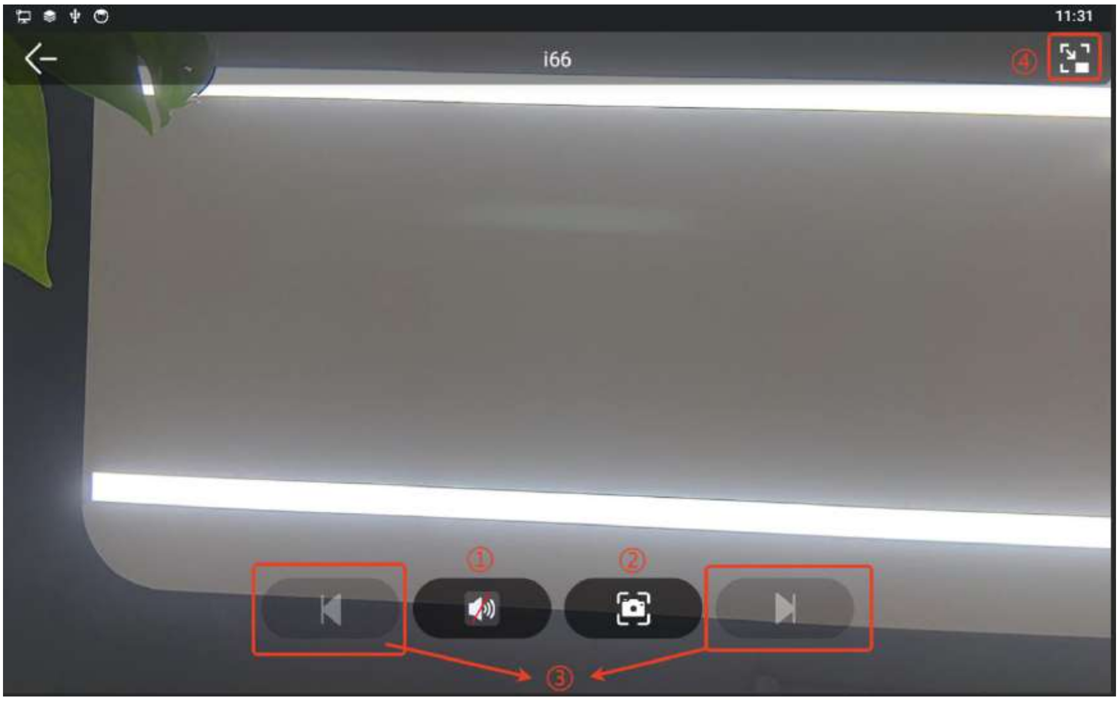


Figure 128 - Full-Screen Monitoring Interface

Table 34 - Full-Screen Monitoring Interface

Index	Function
①	Click to monitor the device and play the sound collected by the device.
②	Perform a capture operation to capture and save the current monitoring screen (viewable in capture records).
③	When multiple monitoring devices are selected, click the arrow to switch to the full-screen monitoring interface of other devices.
④	Minimize the monitoring interface and shrink the current monitoring window to the background, facilitating the simultaneous operation of other functions.

13.2 Capture Record

Images captured and saved by clicking the capture button in the monitoring interface will be stored in [**Monitoring**] >> [**Capture record**]. Users can access this interface to view the captured images. Press icon ① to perform a keyword search on the capture records. Press icon ② to delete the selected capture records.

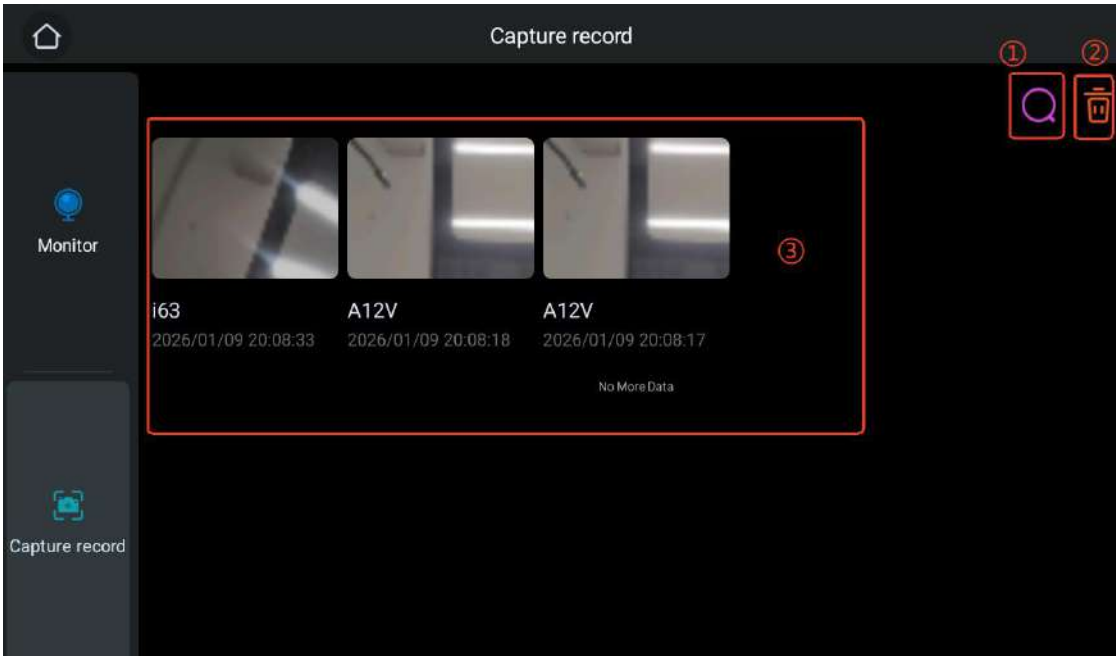


Figure 129 - Capture Record

Table 35 - Capture Record

Index	Function
①	Click to perform a keyword (e.g.device name, time, etc.) search on snapshot records.
②	Click to select and delete the selected capture records.
③	Display all snapshot images; click the corresponding photo to view it in full-screen view.

14 Settings

14.1 Status

The phone status includes the following information about the phone

- **Common**: MAC Address, IP Address, Version, Phone Model, Wi-Fi MAC Address.

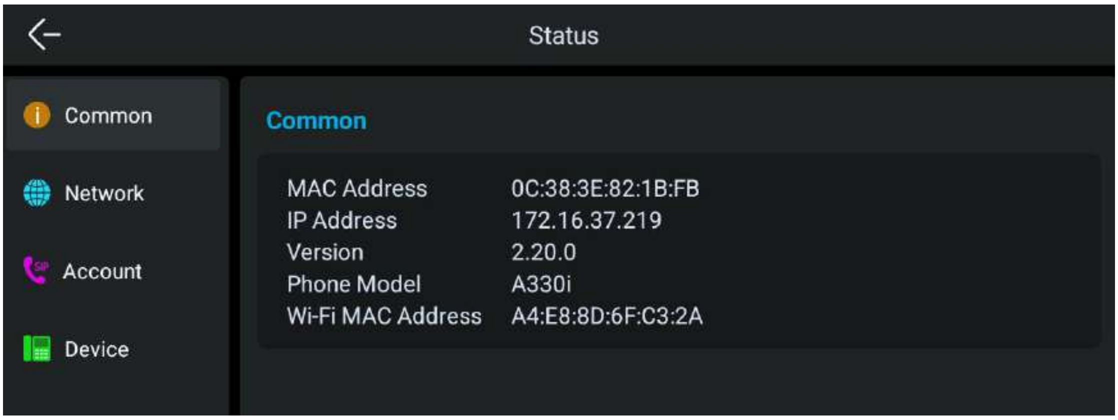


Figure 130 - Common Information

- **Network**: Network type, Connection Mode, IPv4 status, MAC Address.

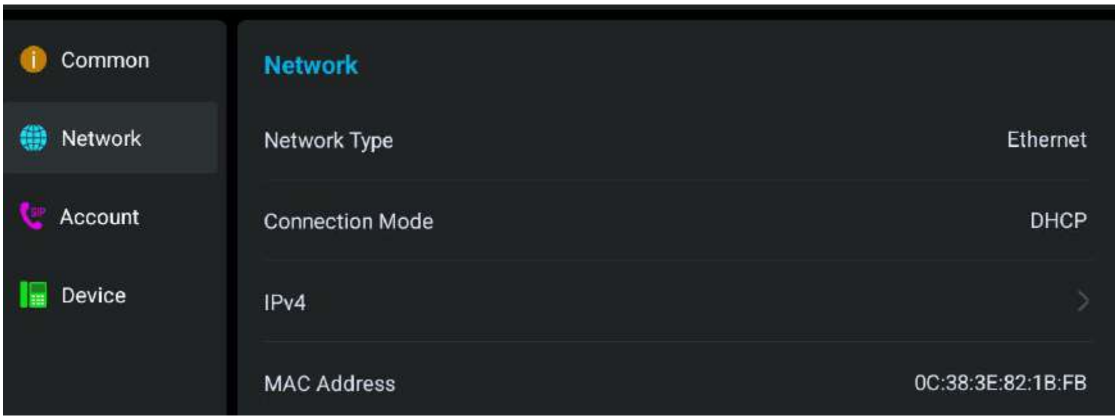


Figure 131 - Network Information

- **Account**: SIP User, SIP account status (Registered / Not Applied / Trying / Timeout / Failed)

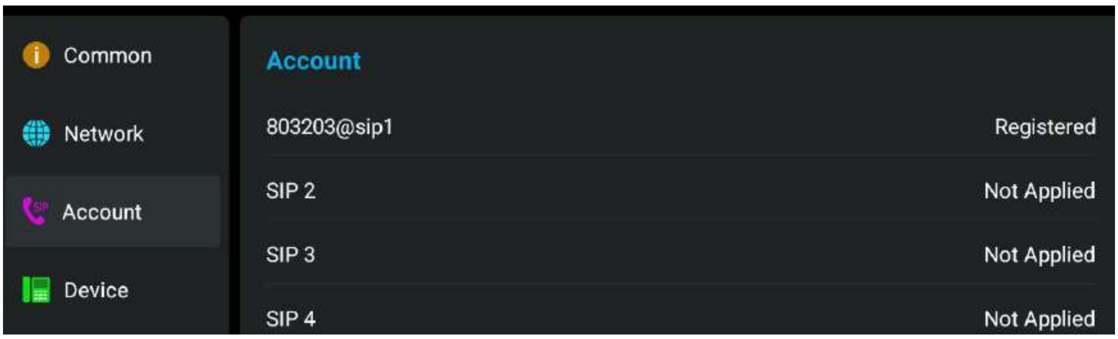


Figure 132 - Account Information

- **Device**: Device Model, Memory (RAM and ROM), Uptime, Software Version, Hardware Version, Android Version.

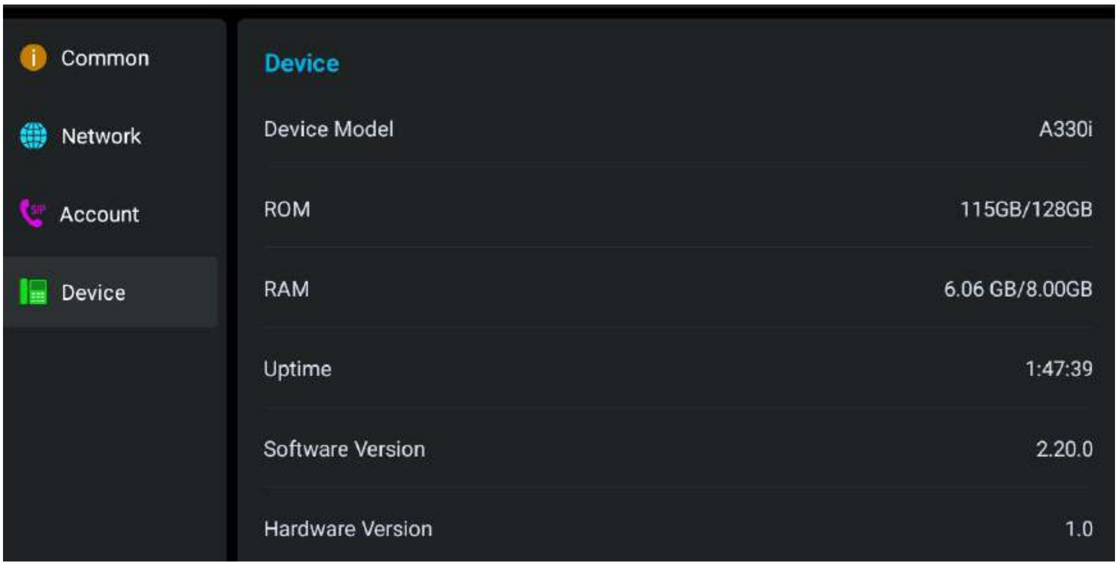


Figure 133 - Device Information

14.2 Broadcast Settings

14.2.1 Emergency Broadcast

Emergency broadcast is the highest-priority broadcast task—it can interrupt other running tasks. When an extension plays an emergency broadcast, the volume is automatically set to maximum.

1. In the broadcast intercom system, access the [**Settings**] >> [**Broadcast Settings**] >> [**Emergency Broadcast**] interface, then press the button + in the upper right corner to add an emergency task.

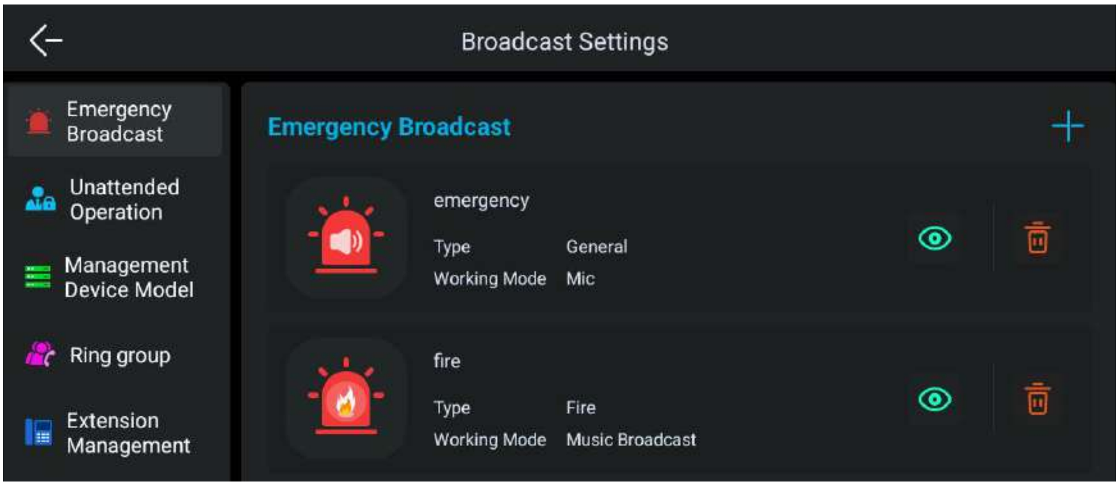


Figure 134 - Emergency Broadcast

2. Fill in the relevant configuration.

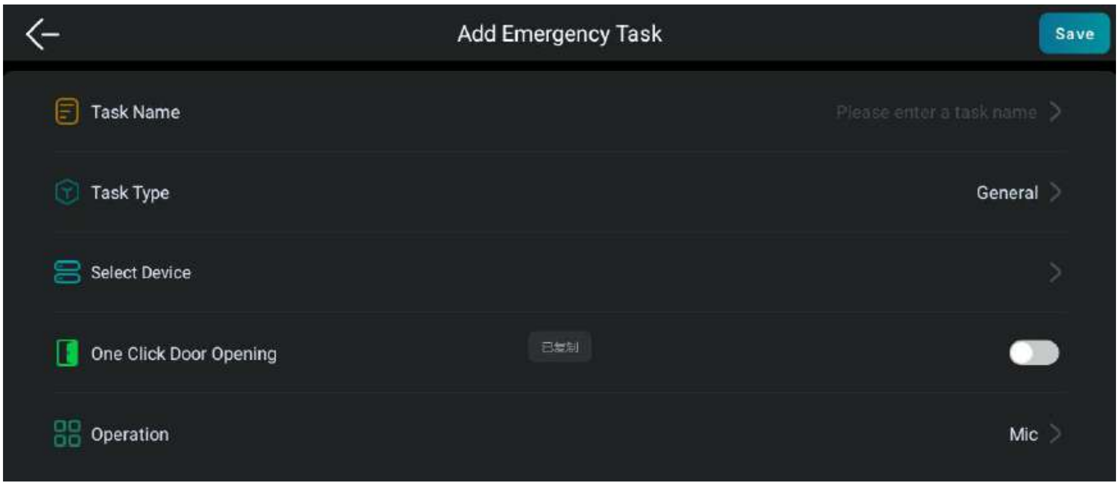


Figure 135 - Add Mic Emergency Task

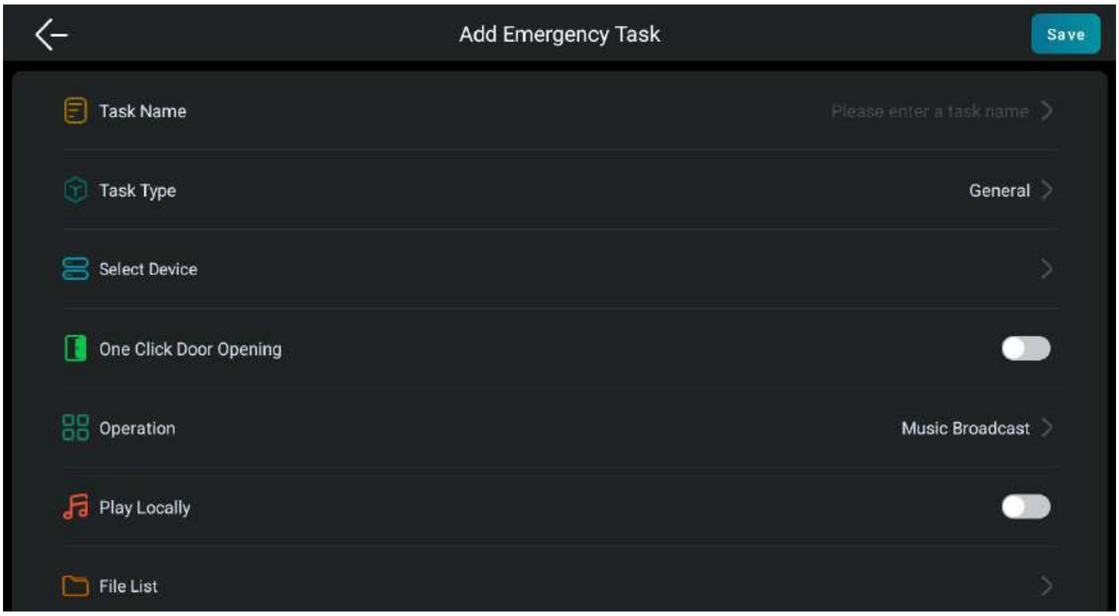


Figure 136 - Add Music Broadcast Emergency Task

Table 36 -Add Emergency Task

Parameter	Description
Task Name	Enter an identification name for the task to distinguish between different emergency tasks.
Task Type	Optional types: "General", "Fire", "Natural Disaster", "Intrusion", "Equipment Failure"—different types display different task icons.
Select Device	Select the extensions that need to receive the emergency broadcast.
One Click Door Opening	When enabled, initiating the task will trigger all doorphone to unlock (see 10.2.2 Doorphone for configuration details).
Operation	- Mic: Initiate a real-time voice emergency broadcast to

	extensions; - Music Broadcast: Initiate an audio emergency broadcast to extensions.
Play Locally	Choose whether to play the emergency audio synchronously on the host device.
File List	Select the audio file for the emergency broadcast (imported via web page/USB drive/SD card).

3. After completing the configuration, press "Save"—the task will be displayed in the Emergency Broadcast List. Press the icon  on the right side of the task to activate or deactivate the task (it is activated by default). When activated, the task icon will appear on the idle screen; otherwise, it will be hidden and cannot be triggered.

4. When there is an activated emergency task, an "**Emergency Broadcast**" icon (①) will be displayed on the home page. Press this icon to pop up the list of activated tasks (②), and press the corresponding task to trigger it immediately.

Figure 137 - Emergency Broadcast Icon

14.2.2 Unattended Operation

After enabling unattended mode, when another device calls the line number corresponding to the device, the call will be forwarded to the configured call forwarding number, realizing call transfer in unattended scenarios.

1. Access the [**Settings**] >> [**Broadcast Settings**] >> [**Unattended Operation**] interface, press "**Transfer Device**". In the pop-up "**The forwarding number**" interface, configure the call forwarding number for the corresponding line. After configuration, press "**Confirm**", then press the "**SAVE**" button in the upper right corner.

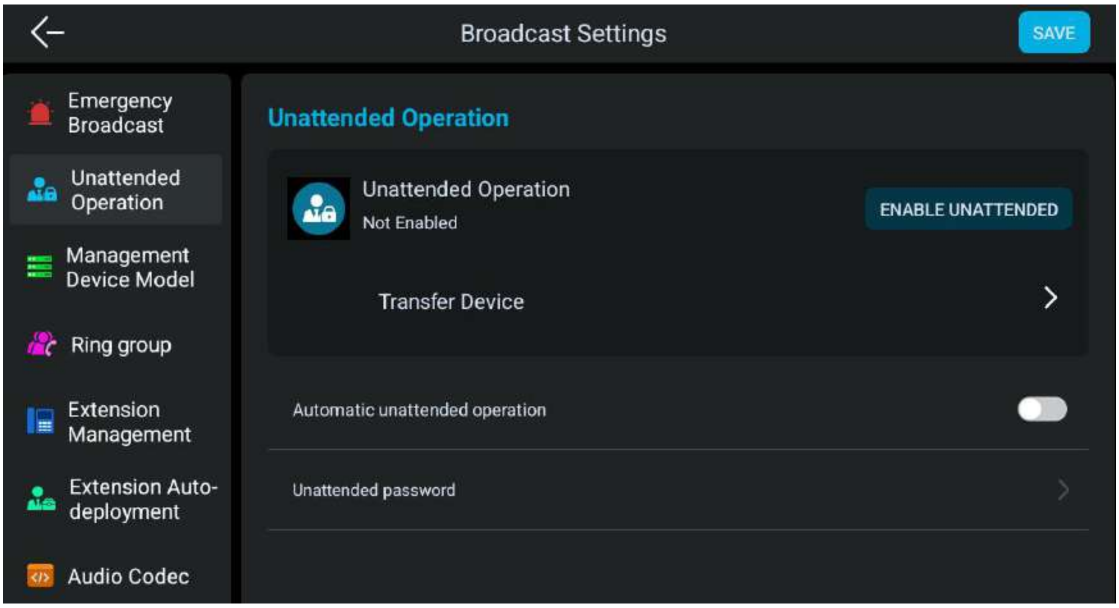


Figure 138 - Unattended Operation

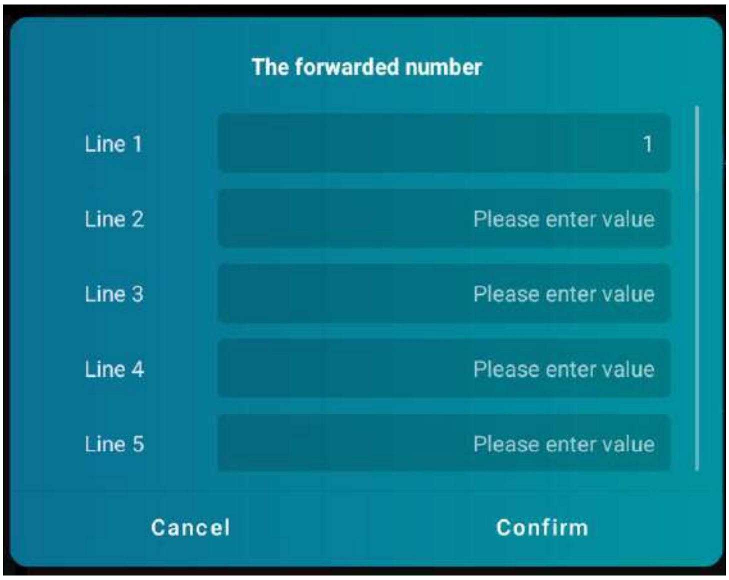


Figure 139 - Set the Forward Number

2. **Enable Unattended Mode**

- **Manual Activation:** Press the "**ENABLE UNATTENED**" button and enter the unattended password (default: 123456). The device will then enter unattended mode.
- **Automatic Activation:** Turn on the "**Automatic unattended operation**" toggle and set the time period. The device will automatically enter unattended mode during the specified time.

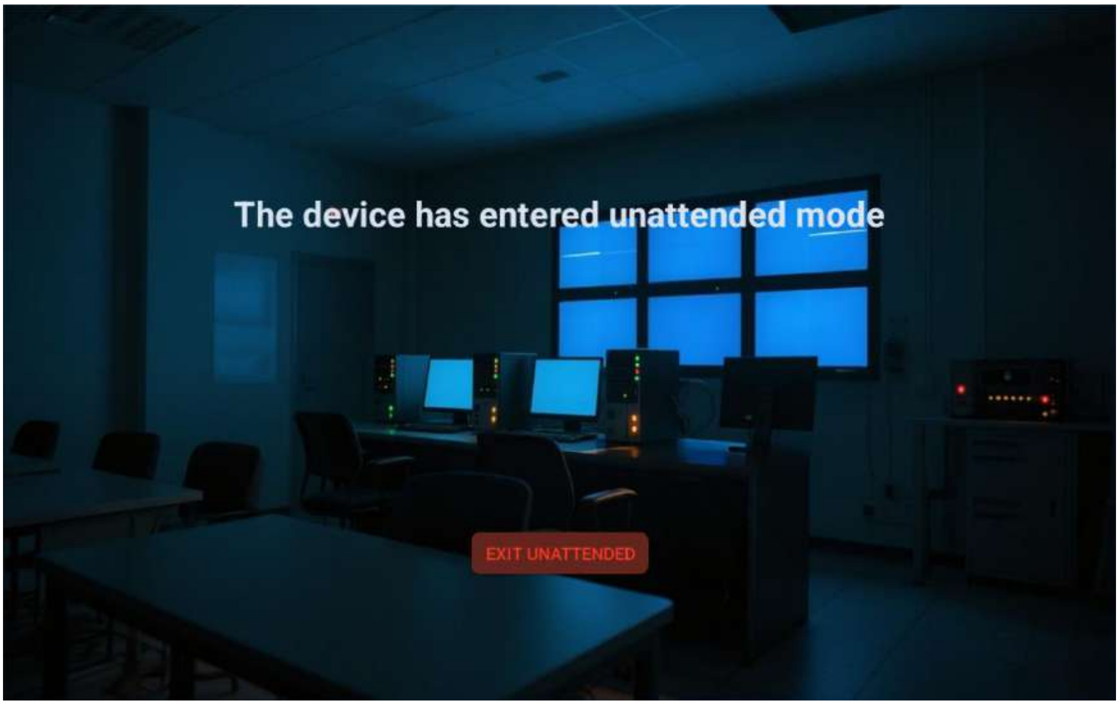


Figure 140 - Unattended Mode

3. When the device enters unattended mode, if another device calls this phone and a call forwarding number is configured for the line, the call will be forwarded to that number. To manually exit the mode, press "**EXIT UNATTENDED**" and enter the unattended password.

4. To modify the dedicated password for manually enabling/disabling unattended mode, follow these steps:

- First, access the Unattended Mode Settings page, press the "**Unattended password**" option.
- Enter the old password, new password, and confirm password accurately.
- Finally, press the "SAVE" button in the upper right corner of the page to complete the password modification.

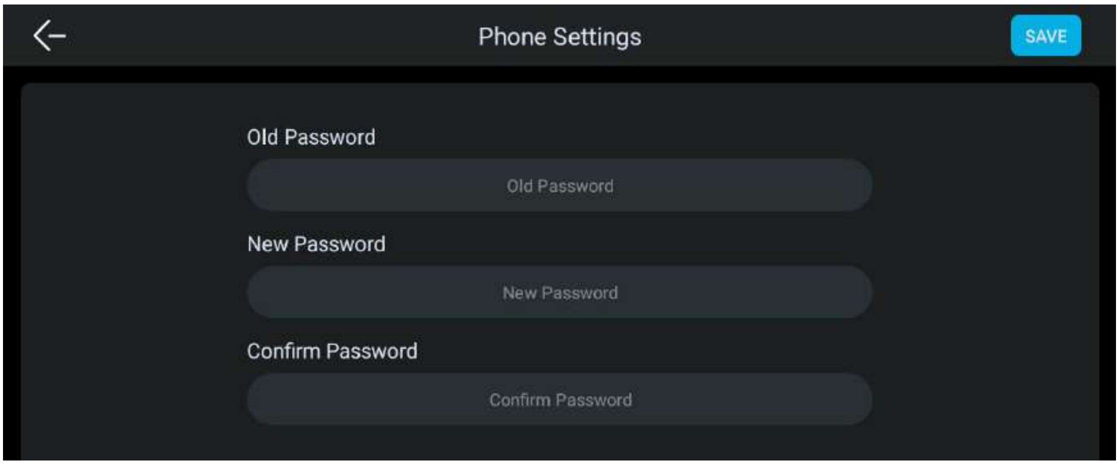


Figure 141 - Unattended Password Modification

14.2.3 Management Device Model

This interface allows you to select extension models that support automatic management.

- If the scanned extension model is in the selected list, the automatic deployment process will start directly to complete the extension access without manual intervention.
- If the scanned extension model is not in the selected list, the extension will be automatically classified into the **"Unmanaged Device List"**, and no automatic deployment will be performed. It can be brought under management after subsequent manual configuration.

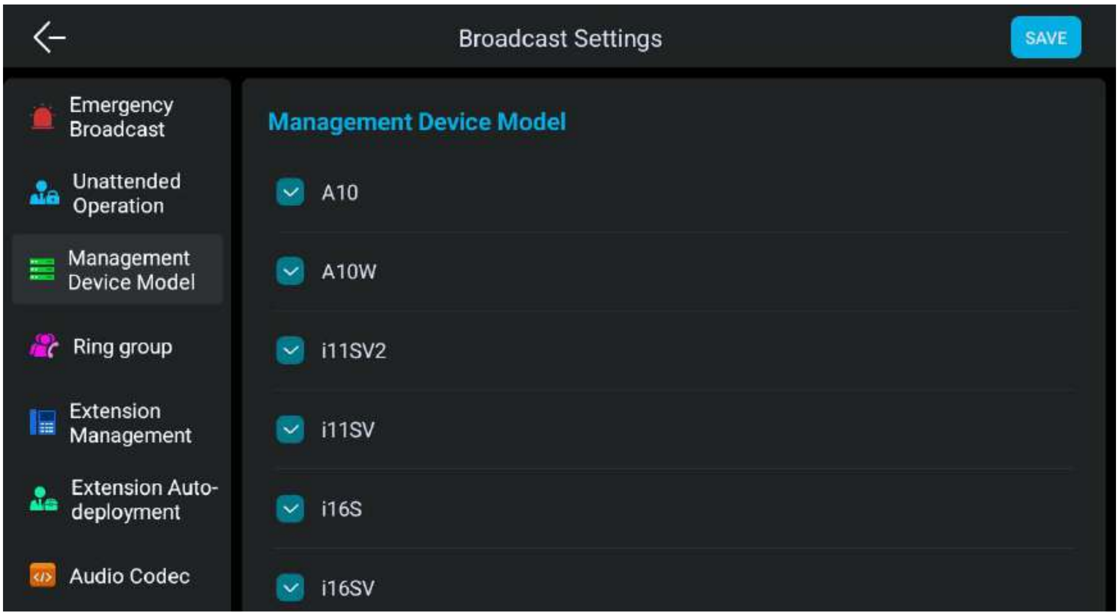


Figure 142 - Management Device Model

14.2.4 Ring Group

Users can customize ring groups and add extensions to the group (select the device for editing in the Device Management interface). When the phone calls the group number, all terminals in the simultaneous ring group will ring.

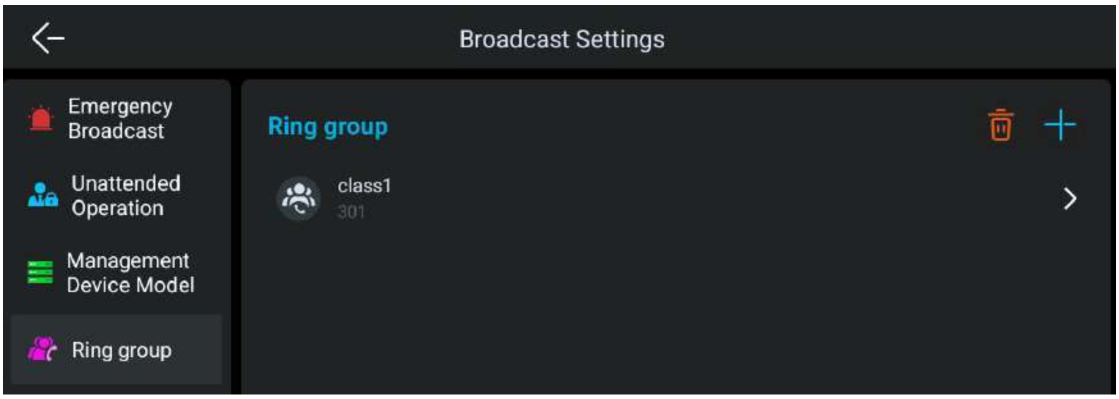


Figure 143 - Ring Group

1. In the [**Settings**] >> [**Broadcast Settings**] >> [**Ring group**] interface, press the button to add a ring group. Fill in the following configurations and press "**Save**":

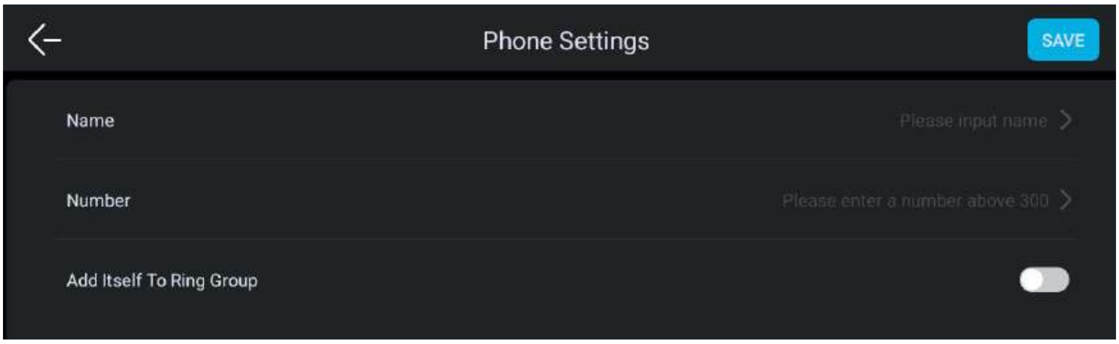


Figure 144 - Add Ring Group

Table 37 - Add Ring Group

Parameter	Description
Name	The identification name of the ring group.
Number	When the phone calls this number, all extensions in the simultaneous ring group will ring at the same time. (System extension numbers range from 0 to 300, which cannot be set as the ring group number.)
Add Itself To Ring Group	When this toggle is enabled, the local device can be added to this simultaneous ring group. When the group number is called, the local device will ring together with other extensions in the group.

2. After creating a ring group, you can edit the device in [**Broadcast Settings**] >> [**Extension Management**] to add the extension to the ring group (see 7.2 Configure extension information for details). You can also batch add devices in the web page [**Line**] >> [**Hotspot Managed Extension**] (see 9.4 SIP Hotspot for details).

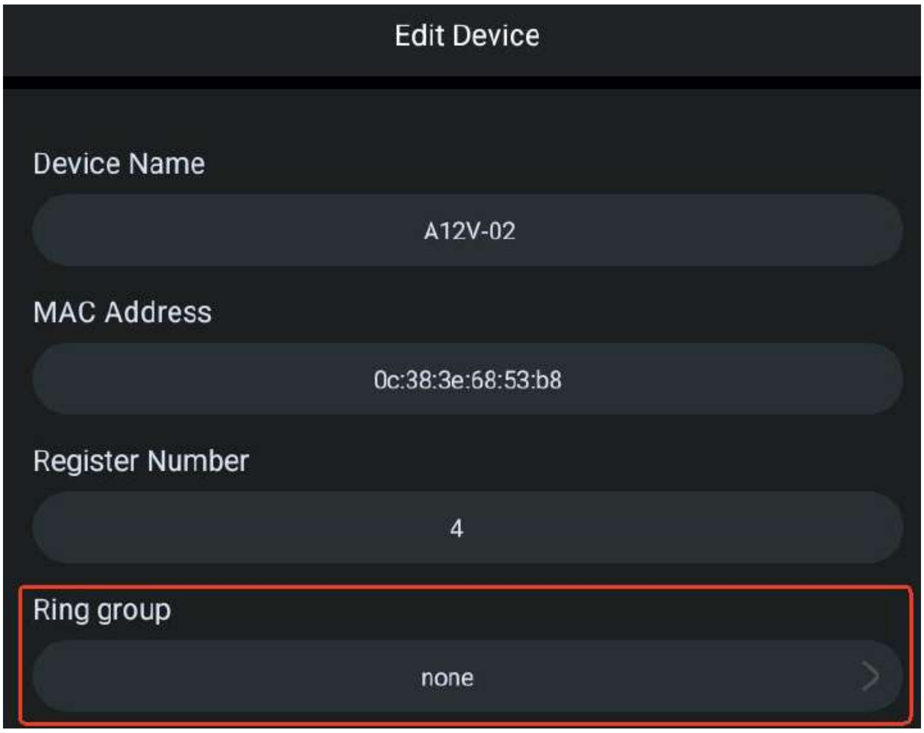


Figure 145 - Add Extension to Ring Group

3. After completing the configuration, use the local device or another extension to call the ring group number. All devices in the group will ring simultaneously, and if any device answers the call, the other devices will stop ringing.

14.2.5 Extension Management

In the Extension Management interface, you can view the status of deployed extensions. It supports operations such as deploying unmanaged devices, deleting, modifying, and grouping extensions. For the steps to deploy extensions, refer to 7 Broadcast Configuration process.

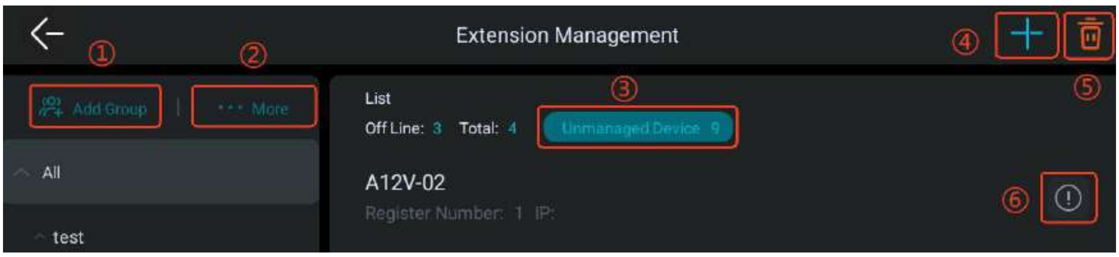


Figure 146 - Extension Management

Table 38 - Extension Management

Index	Function
①	Add a sub-group to the current group.
②	Click to edit or delete the current group.
③	Click to display the list of all scanned devices in the current network segment. Devices whose models are not included in the managed device model list will not be automatically deployed as managed devices by the system. You can select the device on this interface and click "**Move to Managed**" in the upper right corner to complete manual configuration.
④	Manually add new extension devices. Supports accessing by entering device information such as device name and MAC address.
⑤	Click to select and delete the selected extension(s).
⑥	Click to display detailed device information. You can click the edit button to modify the device information.

14.2.6 Extension Auto-deployment

In the [**Broadcast Settings**] >> [**Extension Auto-deployment**] interface, turn on the "**Enable automatic deployment of extensions**" toggle to activate the automatic scanning and deployment function for extensions.

Once enabled, the system will automatically identify and deploy accessible extensions, simplifying the deployment process and improving management efficiency. This configuration is enabled by default.

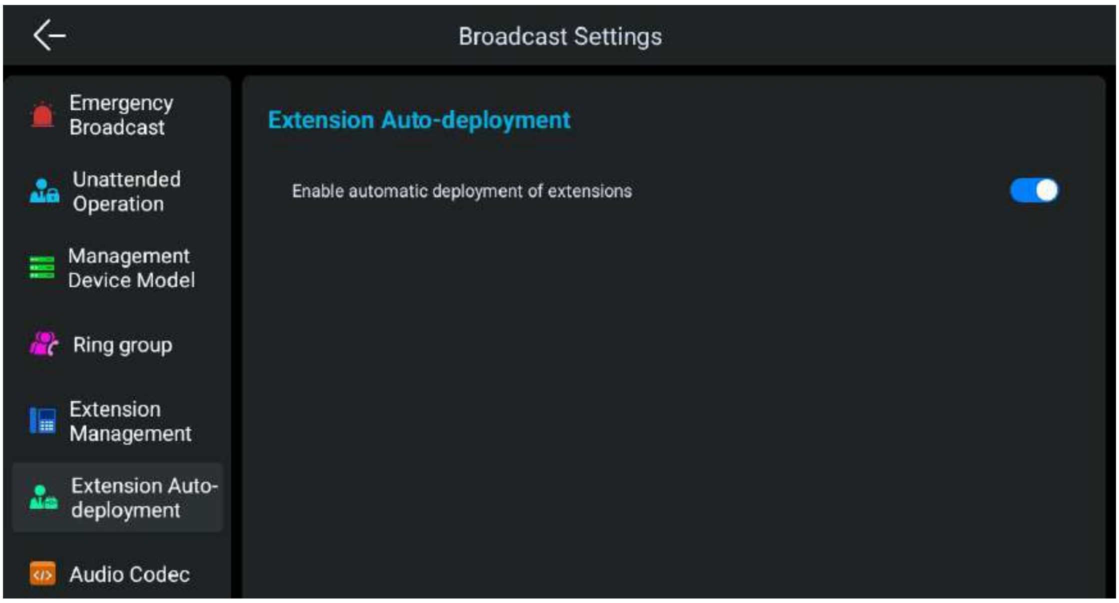


Figure 147 - Auto-deployment of Extension

14.2.7 Audio Codec

You can select the audio coding format for broadcasting. Two coding methods are supported: **G.722** and **MP3**, with **G.722** as the default.

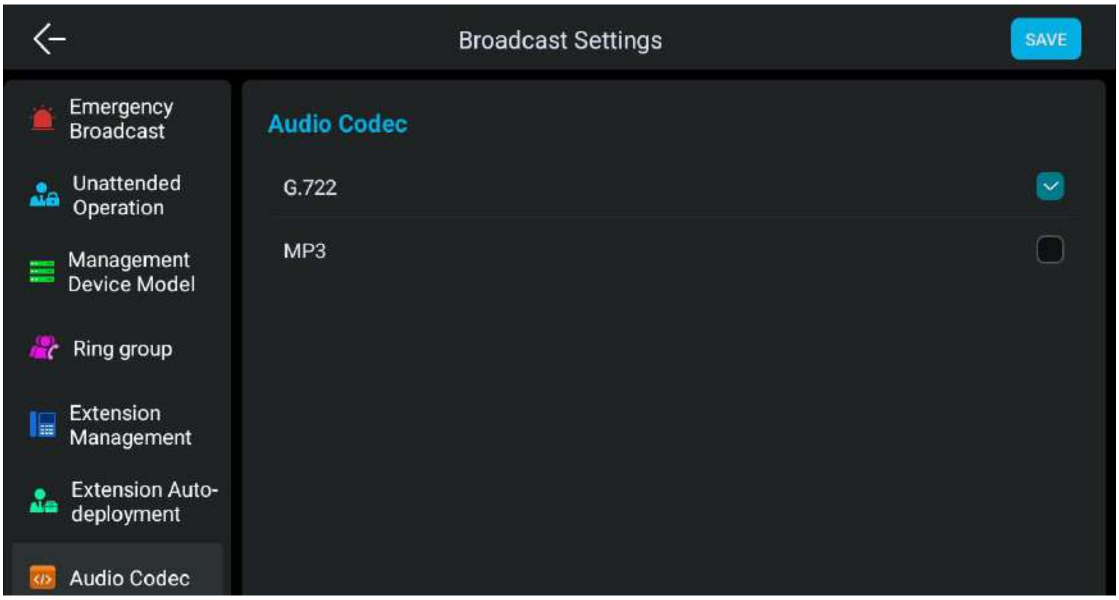


Figure 148 - Audio Codec

14.3 Messages

14.3.1 SMS

If the current line supports the SMS function, the user will receive an SMS notification when a message is sent to this number, and a new SMS icon will be displayed in the top status bar.

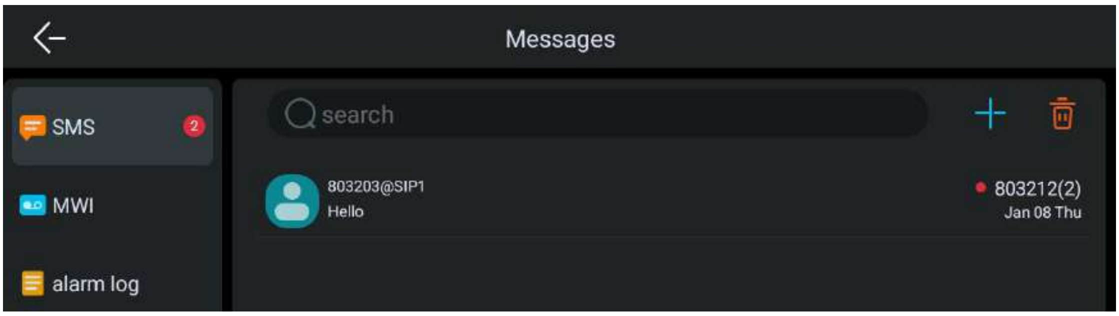


Figure 149 - SMS

Send messages:

- Enter the [Messages] >> [SMS] interface and press the button + in the upper right corner to create a new message.
- Select the line, enter the recipient's number, type the SMS content in the input box below, and press the send button in the lower right corner to send.

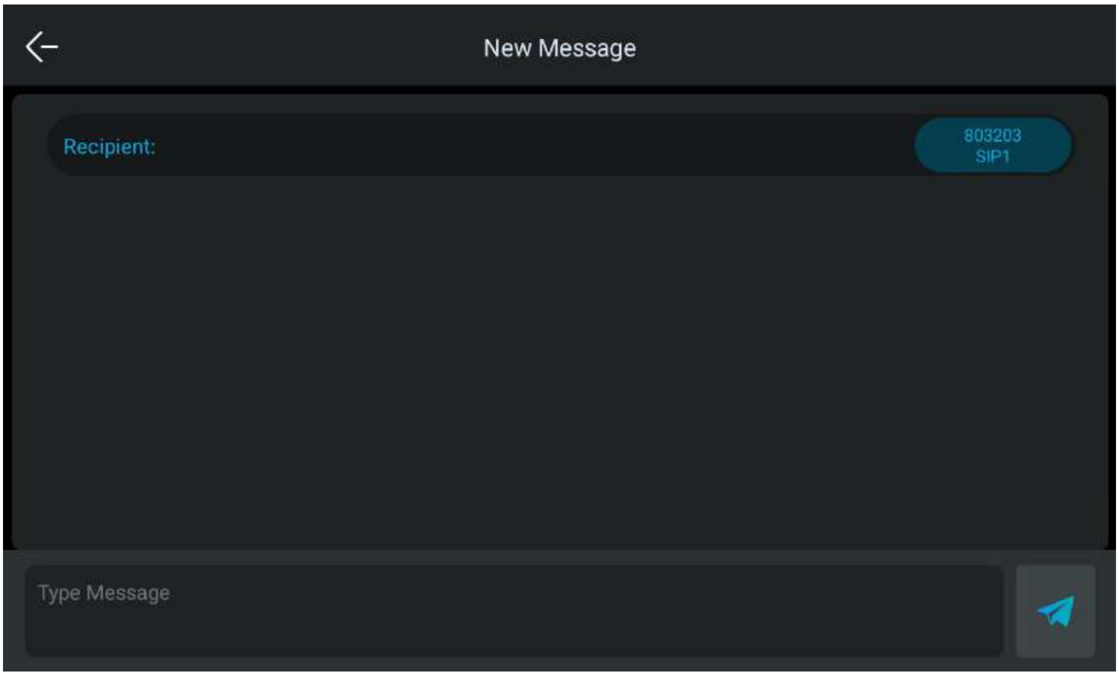


Figure 150 - Send Message

View messages:

- In the [**Messages**] >> [**SMS**] interface, if there are unread SMS messages, a red number will be displayed on the left navigation bar to indicate the number of unread messages (see Figure 149 SMS).
- Select a conversation to view the SMS history and reply with a new message.

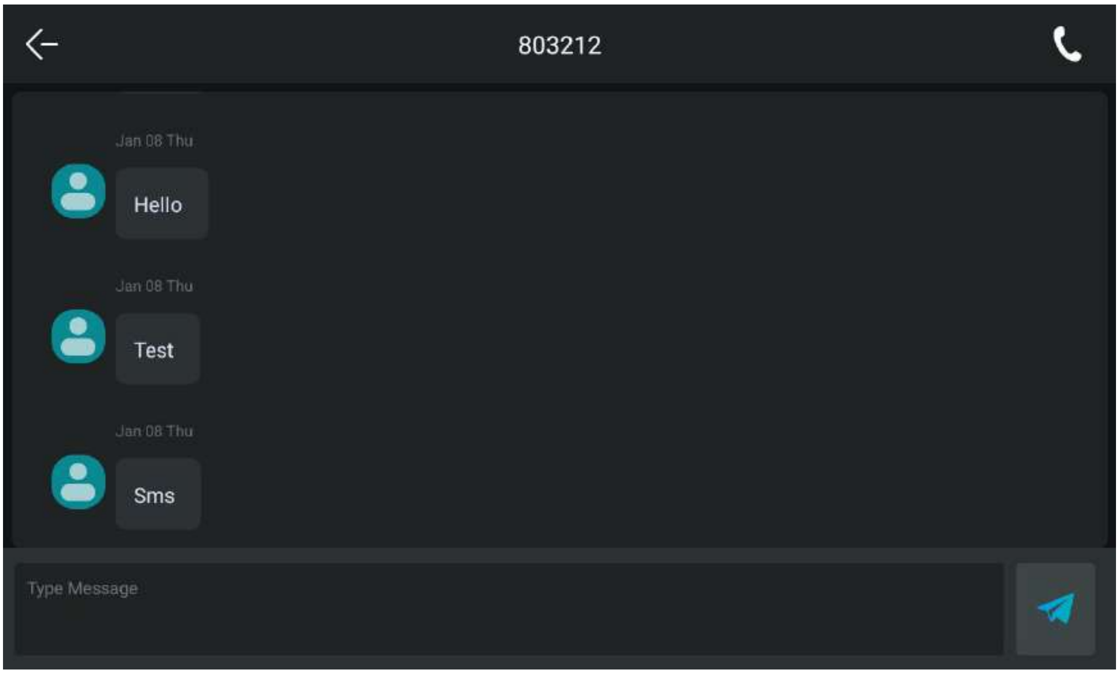


Figure 151 - View Messages

14.3.2 MWI（Message Waiting Indicator）

If the server of the phone's SIP account supports the voice message function, the caller can leave a voice message on the server when the user does not answer. The user will receive a voice message notification from the server, and a new voice message icon will be displayed in the top status bar.

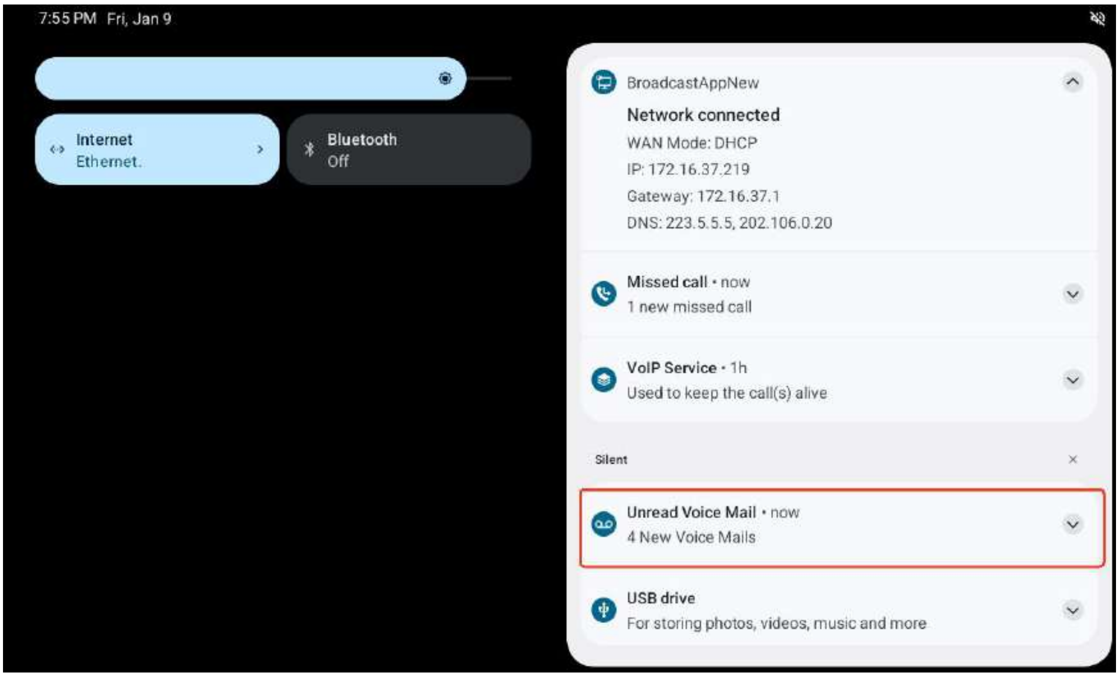


Figure 152 - New Voice Message Prompt

To listen to voice messages, the user must first configure a voicemail number. After the voicemail number is configured, the user can retrieve voice messages for the corresponding line.

When the phone is in the default standby state:

- In the [**Messages**] >> [**MWI**] Interface, if there are unread voice messages, a red number will be displayed on the left navigation bar to indicate the number of unread voice messages.
- Press the button  to configure a voice mail number. Press Save after configuration.
- Press the corresponding button  to call the voicemail number and listen to the messages.

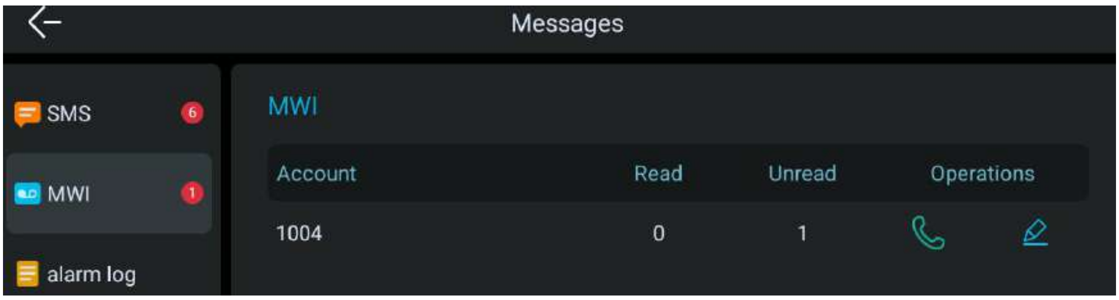


Figure 153 - MWI

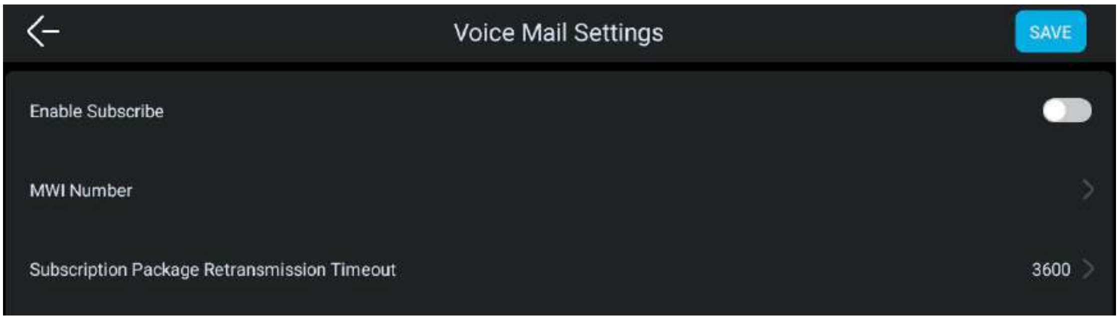


Figure 154 - Voice Mail Settings

14.3.3 Alarm Log

When an alarm input port of the device is triggered, the device will immediately pop up an alert window and play an alarm ringtone. Meanwhile, it can link and execute preset tasks (such as sending text messages, delivering music broadcasts, triggering output ports, etc.) or an Action URL to complete the alarm response. Alarm information (including the region and time) can be viewed in the Alarm Log interface.

14.3.3.1 Security Settings

You can set the trigger modes of the input and output ports on the [**Security Settings**] page of the web.

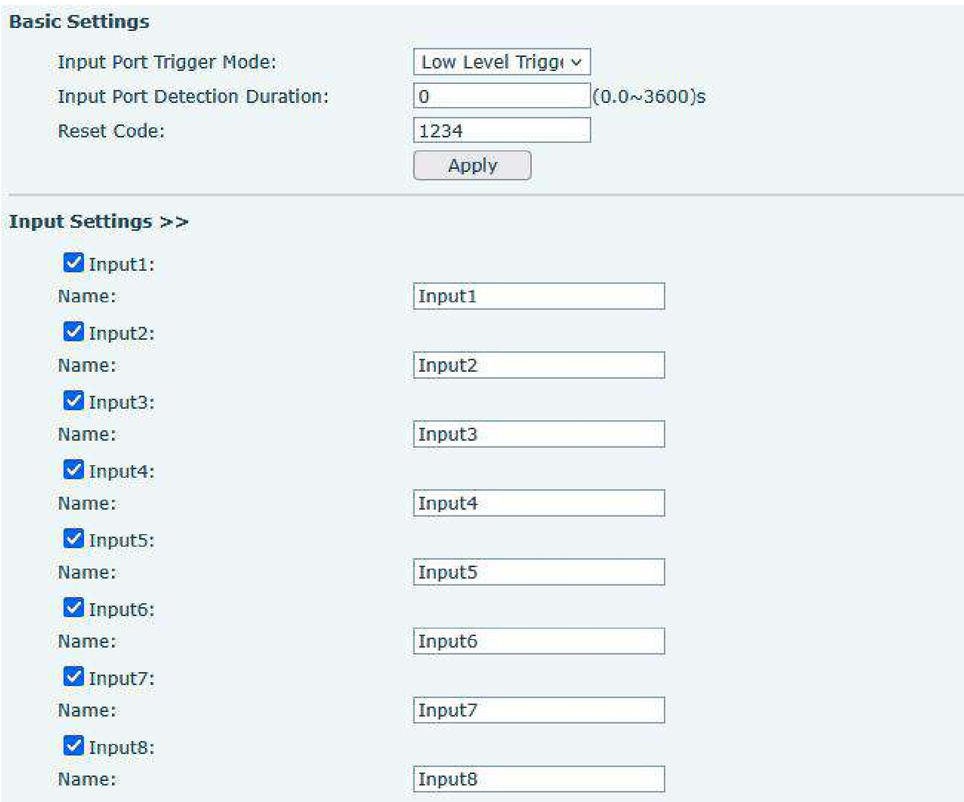


Figure 155 - Security Settings on Web

Table 39 - Security Settings

Parameter	Description
Basic Settings	
Input Port Trigger Mode	When choosing the low level trigger (closed trigger), detect the input port (low level) closed trigger.
	When choosing the high level trigger (disconnect trigger), detect the input port (high level) disconnected trigger.
Input Port Detection Duration	Set the duration (range: 0.0~3600s) that the input port trigger signal needs to persist. The corresponding function will be triggered only after the duration is met, which is used to prevent false triggers (current value: 0).
Reset Code	The code used to clear the alarm state (current value: 1234). Take effect by pressing "Apply" after configuration.
Input Settings	
Checkbox (Input1~Input8)	Checking the box enables the corresponding input port; unchecking disables it (all ports are enabled in the current interface).
Name	You can customize the label of the input port. When an alarm is triggered, the pop-up window will display the name of this input port simultaneously (current names are Input1~Input8 by default).

14.3.3.2 Alarm Function

When an alarm input port of the device is triggered, the device will immediately pop up an alert window and play an alarm ringtone. For the locations and descriptions of the input ports, refer to Picture 2 Connecting to the Device and Table 1 Hardware Interface Description.

The alert window will display the name of the triggered input port. The window and ringtone will not turn off automatically; you must press the "**Disarm the Alarm**" button and enter the reset code to stop the ringtone and close the window.

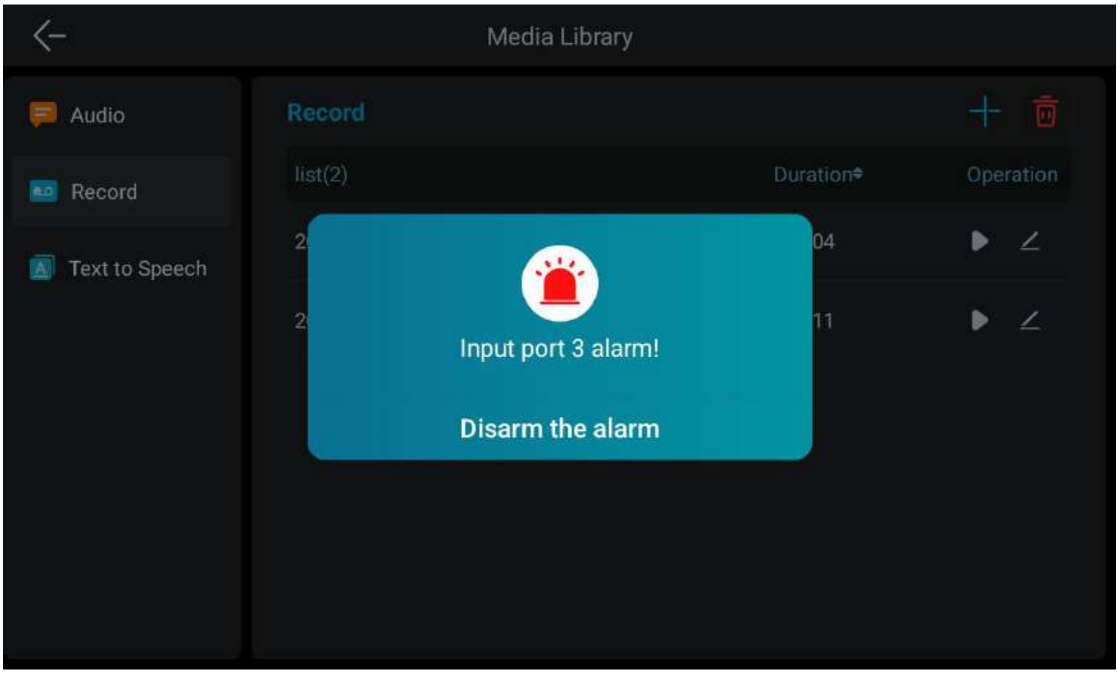


Figure 156 - Alarm Pop-up Window

Triggering an input port alarm can link and activate a scheduled task. For details, please refer to 12.2.2 Add Task.

14.3.3.3 Alarm Log

In [**Settings**]>>[**Messages**]>>[**alarm log**], the region where the alarm was triggered and the time when the alarm occurred are recorded, so that users can view the trigger source and time node of the historical alarm.

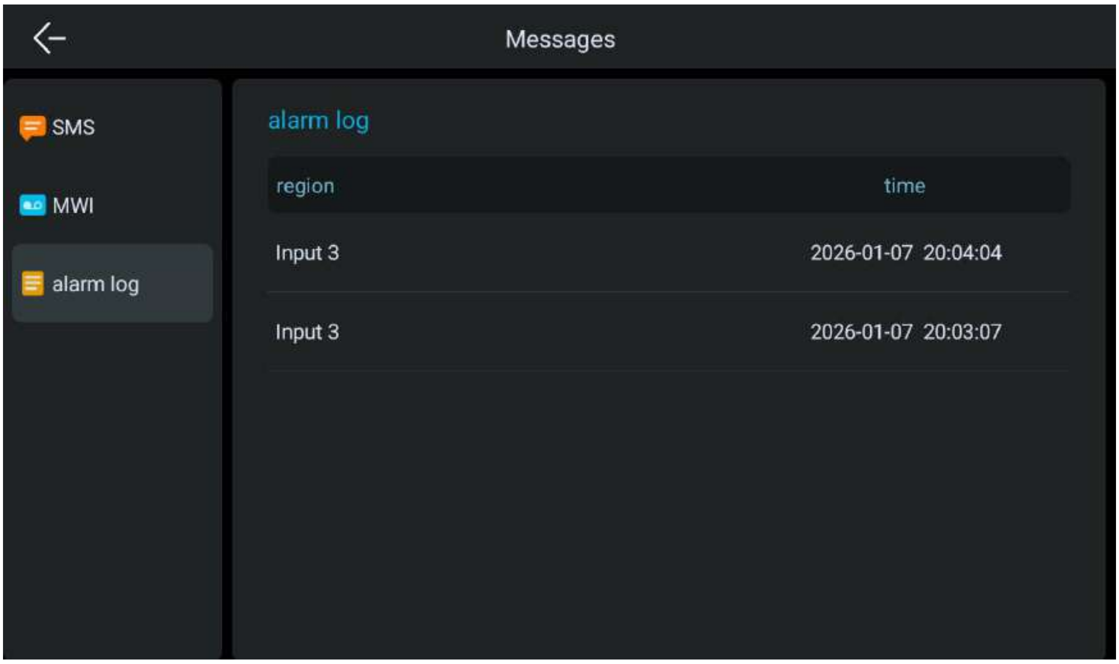


Figure 157 - Alarm Log

14.4 Media Library

14.4.1 Audio

Users can centrally manage local audio files in the audio interface. After importing audio files via a computer or USB drive, the audio files will be displayed in the list, and operations such as previewing and deleting can be performed on them.

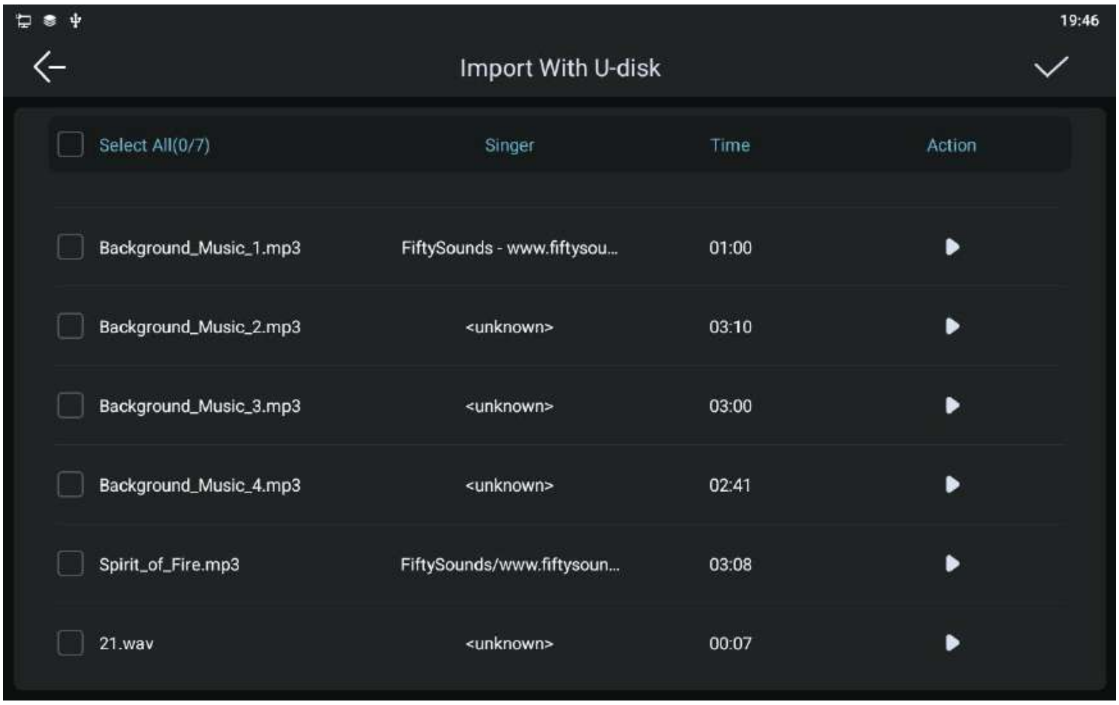


Figure 158 - Audio List

Audio Import via Computer:

- Enter [**Media Library**] >> [**Audio**] in the broadcast intercom system and press the button in the upper right corner; an "Add Audio" pop-up window will appear.
- Select "**Import with Computer**": the host IP address will be displayed. Users can log in to the host web page and upload audio files on the [Library] page.

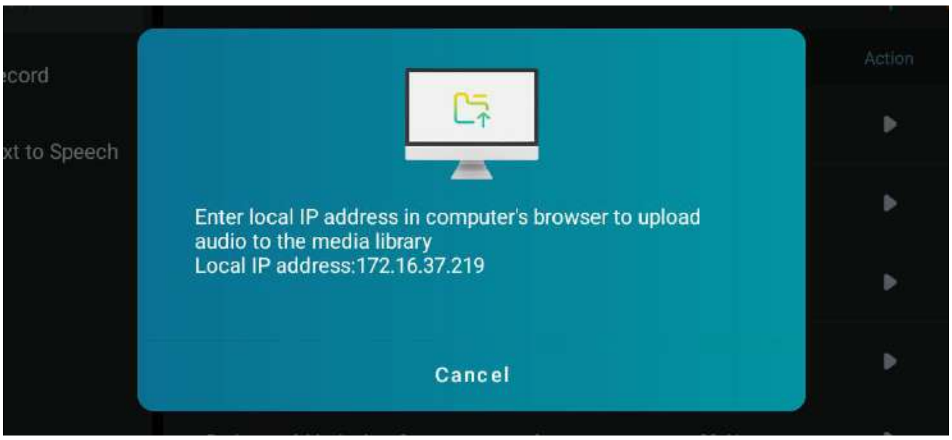


Figure 159 - Audio Import via Computer

Audio Import via U-disk :

- **Create a folder named "Music" in the USB flash drive**, and place the audio files to be imported (in MP3 or WAV format) in this folder.
- Select "**Import with U-disk**" in the add audio pop-up window; the host will scan the audio files in the Music folder. After users select the required files, press the button in the upper right corner to save them.

14.4.2 Record

1. Enter the [**Media Library**] >> [**Record**] interface and press the "+" button in the upper right corner to start recording.

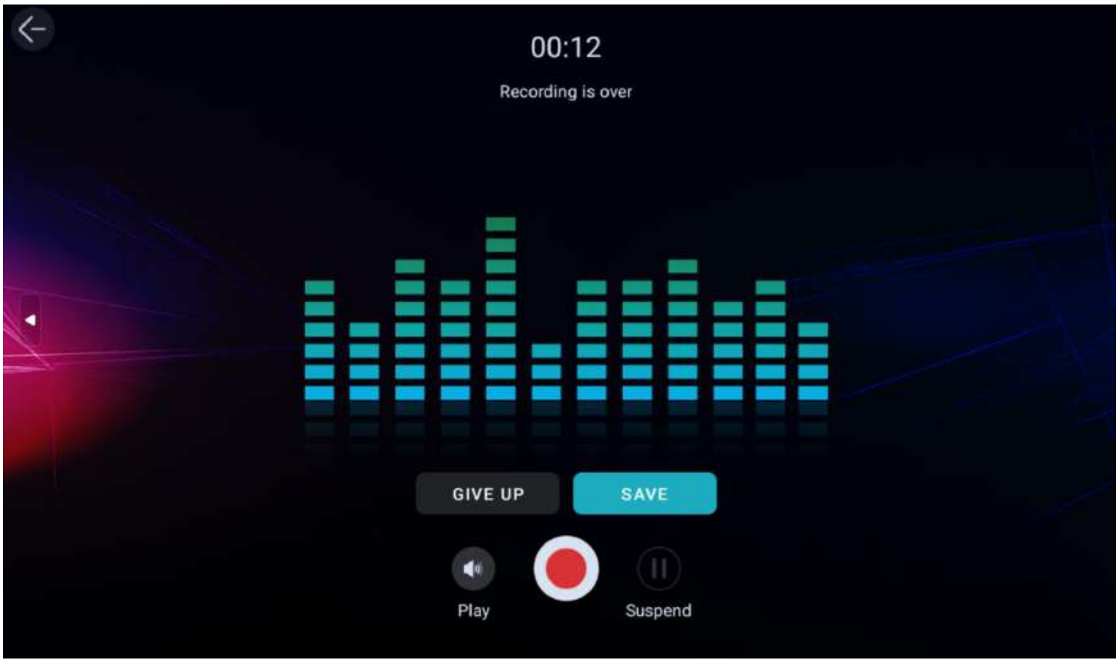


Figure 160 - Recording Interface

2. After recording is completed, you can play and preview the recorded audio, and then choose to discard or save the recording.

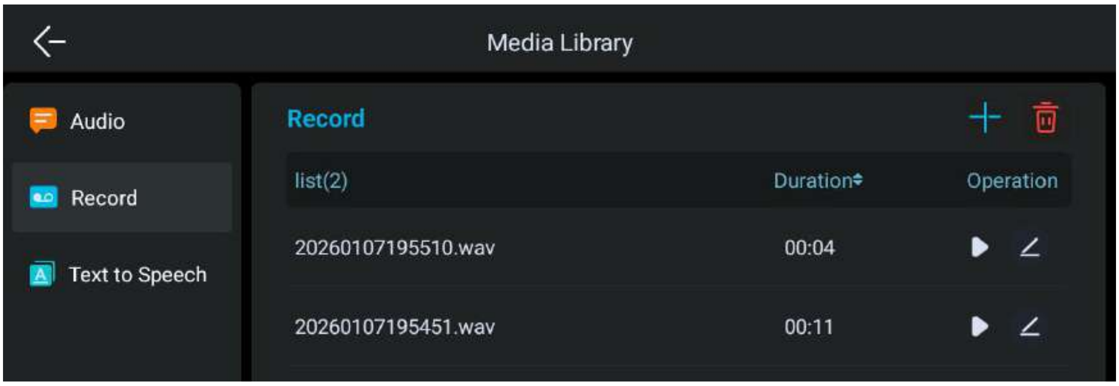


Figure 161 - Record Audio List

14.5 Monitoring

In the [**Settings**] >> [**Monitoring**] interface, you can manage monitoring devices and configure the enablement/disablement or duration of the auto-rotation display on the monitoring interface. For specific functions and configurations, refer to 13.1.1 Monitoring Device Management.

14.6 System Settings

Enter the [**Settings**]>>[**System Settings**] interface, you can use the following functions:

14.6.1 Screen Display

- **Verify Password:**

Enter the correct original password, new password, and confirm password to modify the password for accessing Advanced Settings.

- **Greetings:**

The greeting is displayed in the upper left corner of the screen when the phone is in standby mode. A maximum of 12 characters can be entered. The default value is "**VoIP Phone**".

Note! The greeting can only be displayed in the upper left corner of the standby screen when the default line selection function is disabled (in [Phone Settings] on the device web interface, uncheck "Enable Default Line").

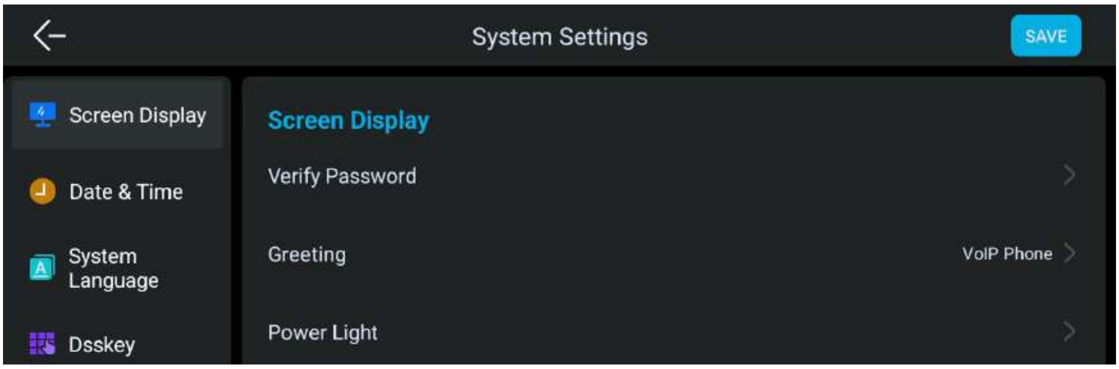


Figure 162 - Screen Display

- **Power Light**

This is used to configure the display method of the device's power indicator. You can set the display of the power indicator in different scenarios, helping users intuitively identify the device's current status through the power indicator.

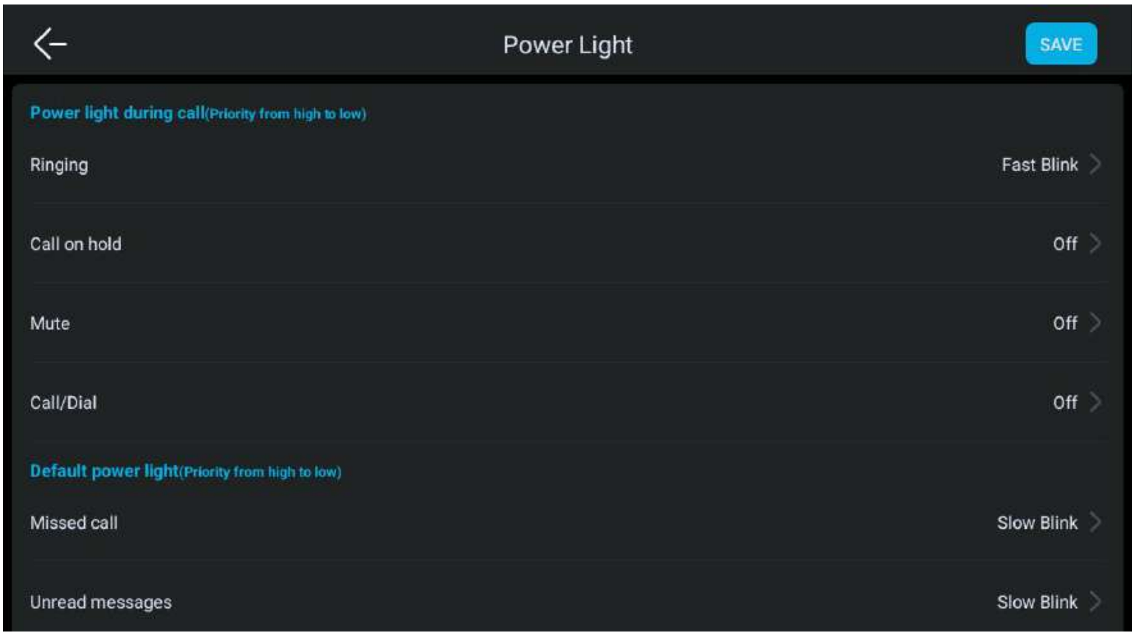


Figure 163 - Power Light

Table 40 - Power Light Settings

Parameter	Description
Ringing	The power indicator status when the phone is ringing.
Call on hold	The power indicator status when a call is on hold.
Mute	The power indicator status when the phone's microphone is muted during a call.
Call/Dial	The power indicator status when the phone is in a call or dialing state.
Missed call	The power indicator status when there is a missed call on the phone.
Unread messages	The power indicator status when there are unread voice messages or SMS on the phone.
Registration failed	The power indicator status when the SIP account registration fails.
Phone mute	The power indicator status when the phone's ringer is muted.
Default power light	The power indicator status when the phone is in standby mode.

14.6.2 Date & Time

When the device time is inaccurate, you can correct the date and time in this interface.

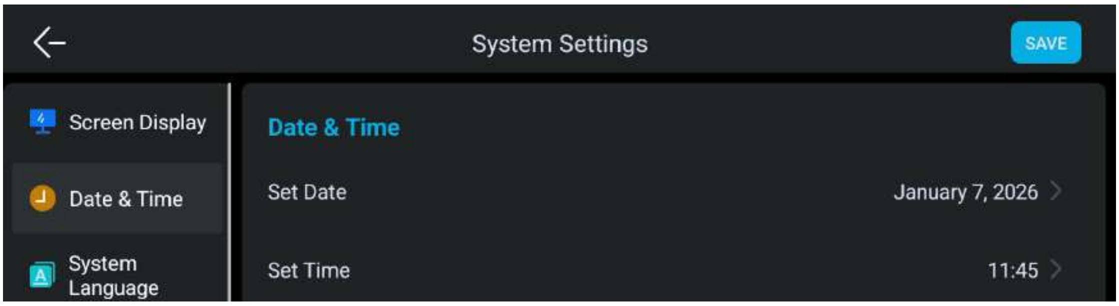


Figure 164 - Date & Time

14.6.3 System Language

You can select and switch the language of the system in this interface; the setting takes effect after being saved.

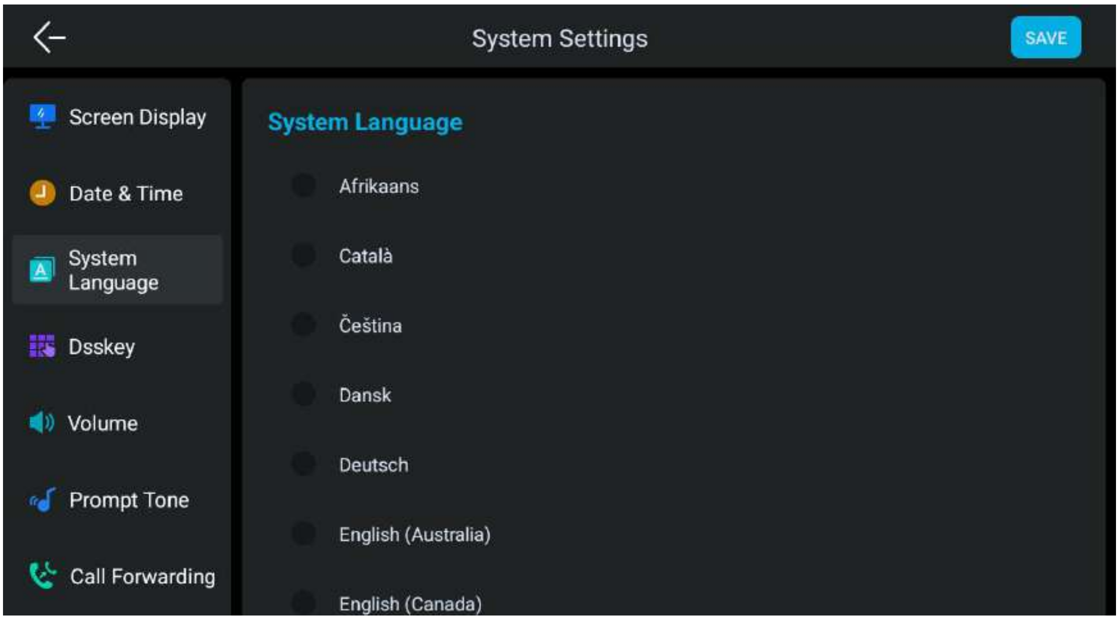


Figure 165 - System Language

14.6.4 Dsskey

Press the corresponding dsskey to configure a Soft dsskey. You can then add the Soft Dss Key to different interfaces (including dialing, calling, conference, and ringing) in the [Function Key] >> [Softkey] section of the web interface. The Soft Dsskey only supports the doorphone type. For specific configuration details, refer to 10.2.2.3 One-Click Door Opening and 17.35 Softkey.

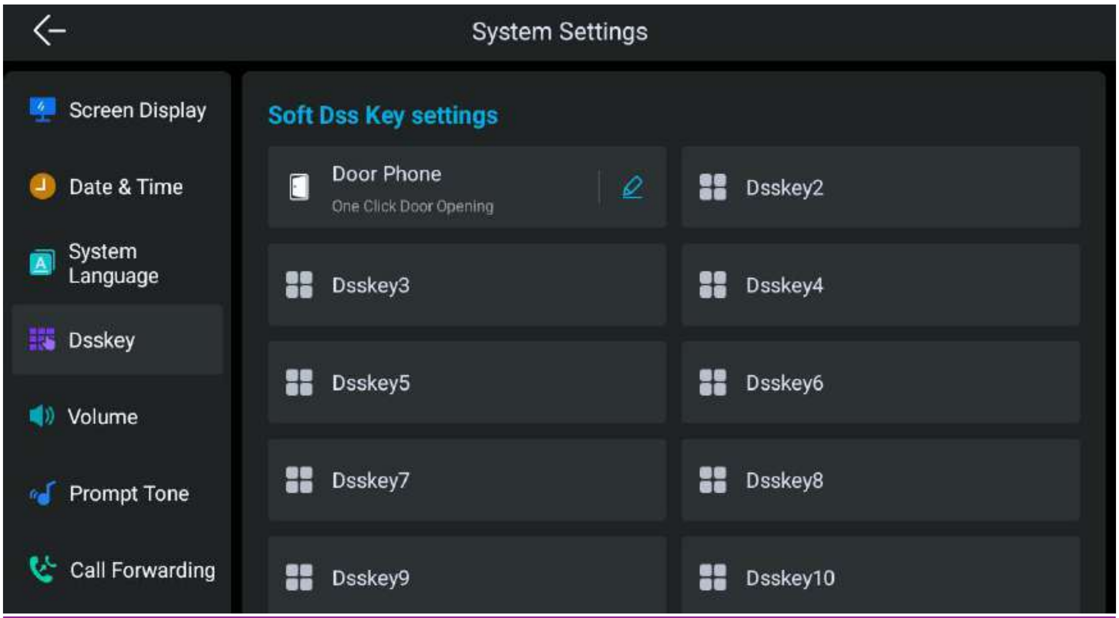


Figure 166 - Soft Dsskey Setting

14.6.5 Volume

In this interface, you can adjust the volume levels for media, calls, ringtones, notifications, and alarms. It also supports enabling the silent mode (once enabled, all ringtones will be muted).

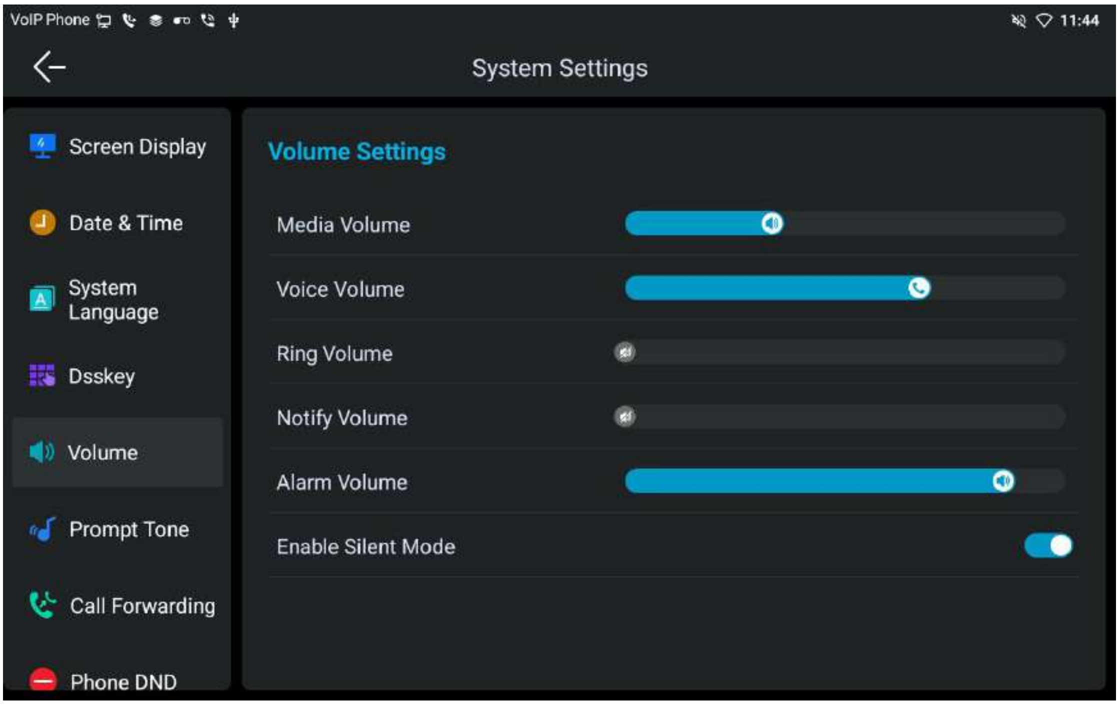


Figure 167 - Volume

14.6.6 Prompt Tone

This interface is used to configure all types of prompt tone-related settings on the device: you can select the ringtone styles for incoming calls, notifications and alerts, and also toggle the prompt tones for scenarios such as touch, dialing/call DTMF, call waiting, and screen lock.

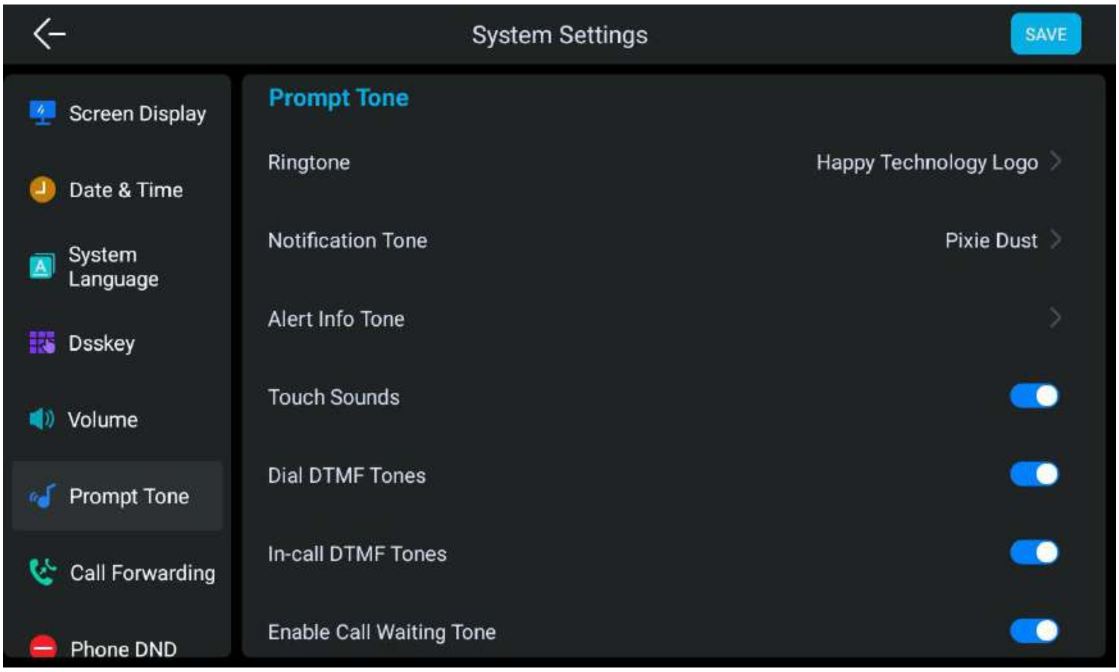


Figure 168 - Prompt Tone

14.6.7 Call Forwarding

This interface allows you to configure the rules for the device's call forwarding function.

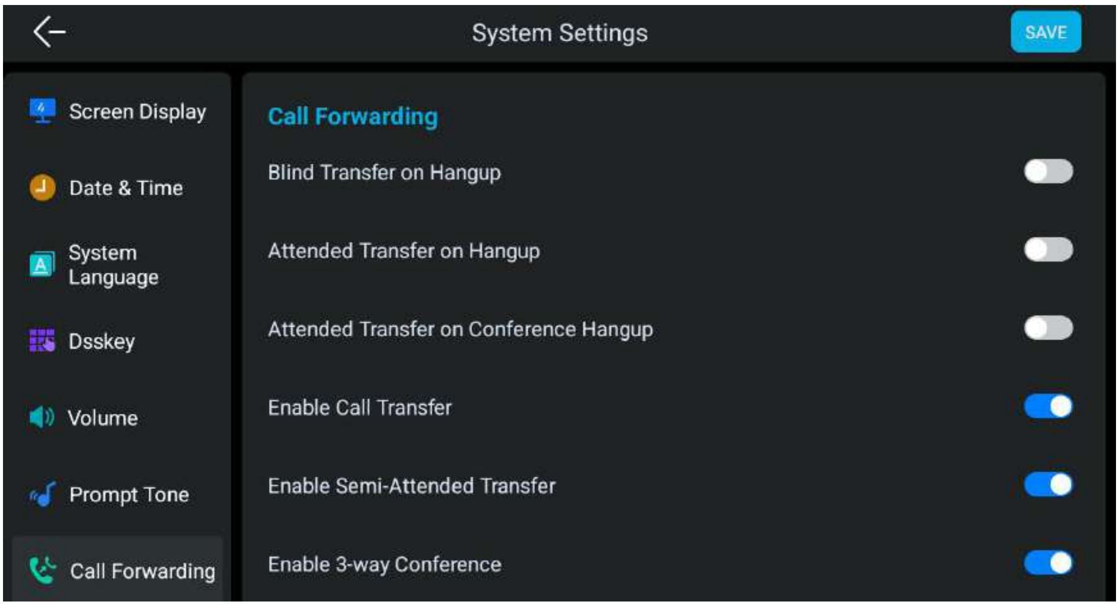


Figure 169 - Call Forwarding

Table 41 - Call Forwarding Settings

Parameter	Description
Blind Transfer on Hangup	Press the [Transfer] key first, and 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 Hangup	Hang up the handle or press the hands-free button to realize the function of attention -transfer, which can transfer the current call to a third party.
Attended Transfer on Conference Hangup	During a three-way call, hang up the handle and the remaining two parties remain on the call.
Enable Call Transfer	Enable Call Transfer.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it
Enable 3-Way Conference	Enable 3-way conference by selecting it

14.6.8 Phone DND

This interface allows you to enable Do Not Disturb (DND) for the entire phone or a specific SIP line. You can also set a valid time period for it to take effect automatically. For details, please refer to 8.11 DND.

14.6.9 Call Settings

This interface is used to configure the call-related functions of the device.

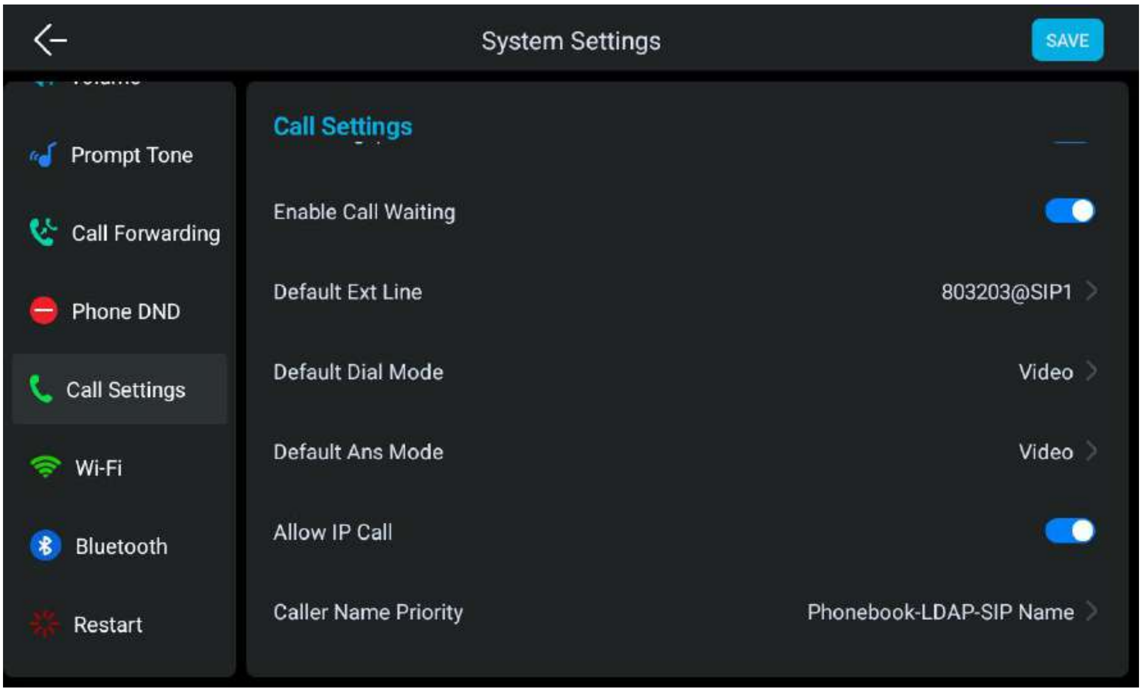


Figure 170 - Call Settings

Table 42 - Call Settings

Parameter	Description
Ban Outgoing	When enabled, the device is restricted from making outgoing calls.
Ban Hangup	When enabled, hanging up calls from the phone will be prohibited.
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
Enable Default Line	If enabled, user can assign default SIP line for dialing out rather than SIP1.
Default Dial Mode	The default call mode (voice or video) for the device to initiate outgoing calls
Default Ans Mode	The default call mode (voice or video) for the device to answer incoming calls.
Allow IP Call	If enabled, user can dial out with IP address
Caller Name Priority	Change caller ID display priority.

14.6.10 Wi-Fi

The device supports the wireless Internet access function, and comes with a built-in Wi-Fi module, eliminating the need for external devices.

When the phone is in the default standby mode, press the function menu key **[Settings]**, navigate to **[System Settings] >> [Wi-Fi]**, set Wi-Fi enable and select the wireless network you want to connect to, and enter the password as prompted to complete the connection.

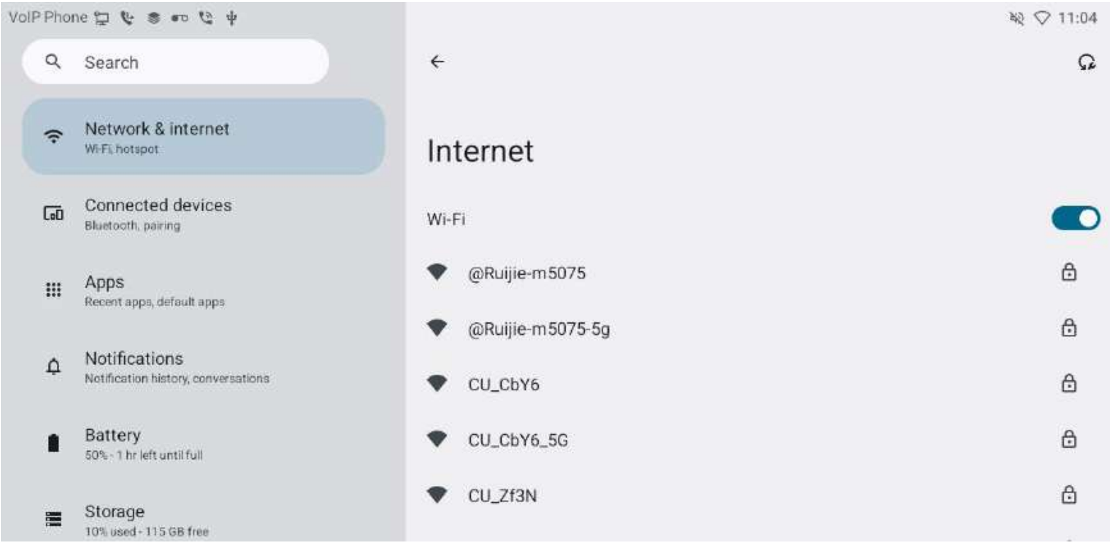
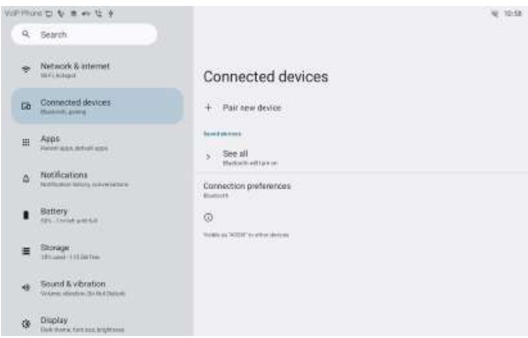


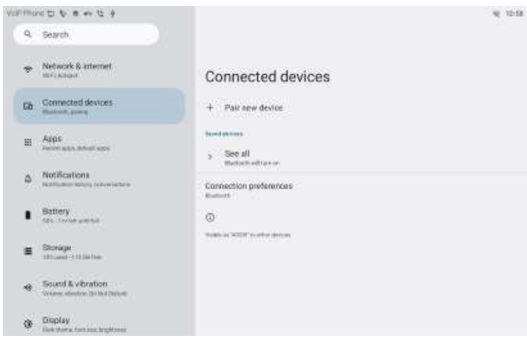
Figure 171 - Wi-Fi Settings

14.6.11 Bluetooth

When the device is in the default standby mode,

- Navigate to [**Settings**] >> [**System Settings**] >> [**Bluetooth**] to jump to the Bluetooth device connection and management interface.
- Tap [**Connection preferences**]>>[**Bluetooth**] and toggle the switch to enable Bluetooth; the phone will automatically search for available devices.
- Tap [**+ Pair new device**] and select a device to perform pairing and connection.





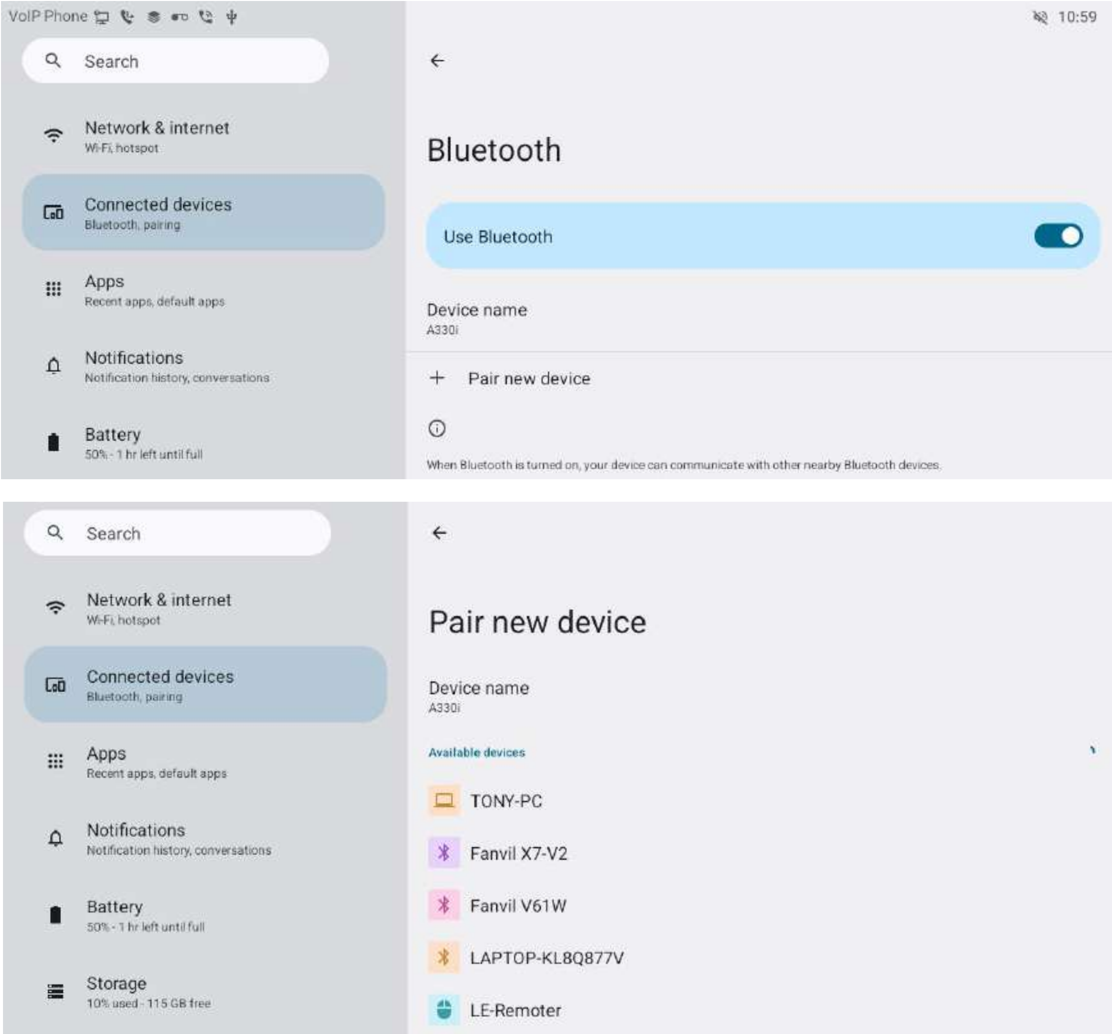


Figure 172 - Connect Bluetooth Device

14.6.12 Restart

Tap this option to restart the device.

- Navigate to [**Settings**] >> [**System Settings**] >> [**Restart**].
- Tap [**Restart**] to trigger a prompt asking whether to restart the phone.
- Tap [**Confirm**] to restart the phone; tap [**Cancel**] to exit the prompt and return to the configuration interface.

14.7 Advanced Settings

To access the [**Settings**] - [**Advanced Settings**] interface, you need to enter the verification password first, which is admin by default.

14.7.1 Ethernet

This interface supports the configuration of network modes. Only IPv4 is supported currently, and there are two optional network types: DHCP and Static IP. Navigate to the [**IPv4**] interface for configuration.

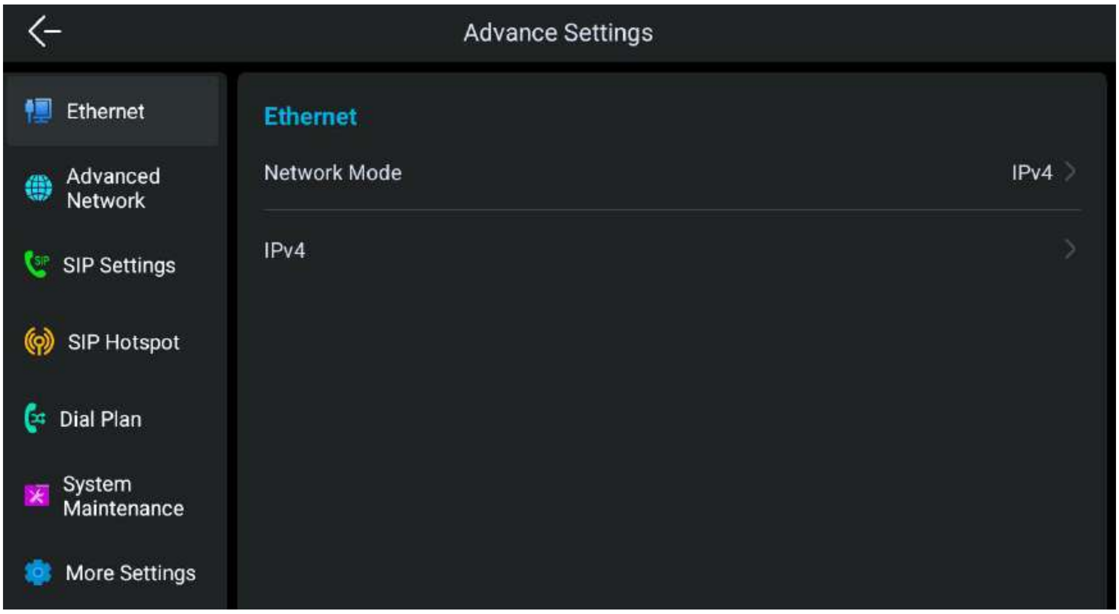


Figure 173 - Ethernet

When the network type is set to DHCP, the phone will be assigned a network IP address by the DHCP server (router).

Use Dynamic Domain Name Service: Enabled by default; it is used for domain name resolution when enabled.

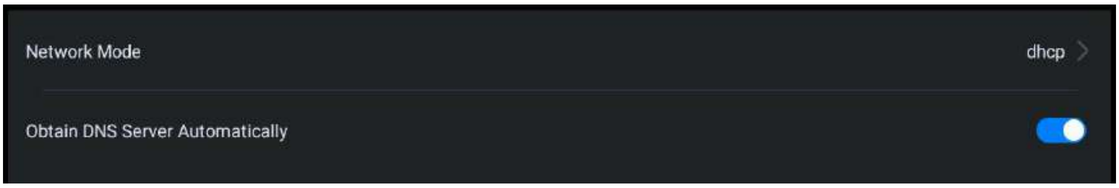


Figure 174 - DHCP Network Mode

When the network is set to Static IP, you need to manually assign an IP address to the phone.

- **IP Address**: Enter the IP address you want to set.
- **Subnet Mask**: Enter the subnet mask
- **Default gateway**: Used to achieve network interconnection, which can be filled in according to your own needs.
- **Primary DNS**: The IP address of the primary DNS server. The default is 8.8.8.8, which is provided free of charge by Google.
- **Backup DNS**: The IP address of the backup DNS server.

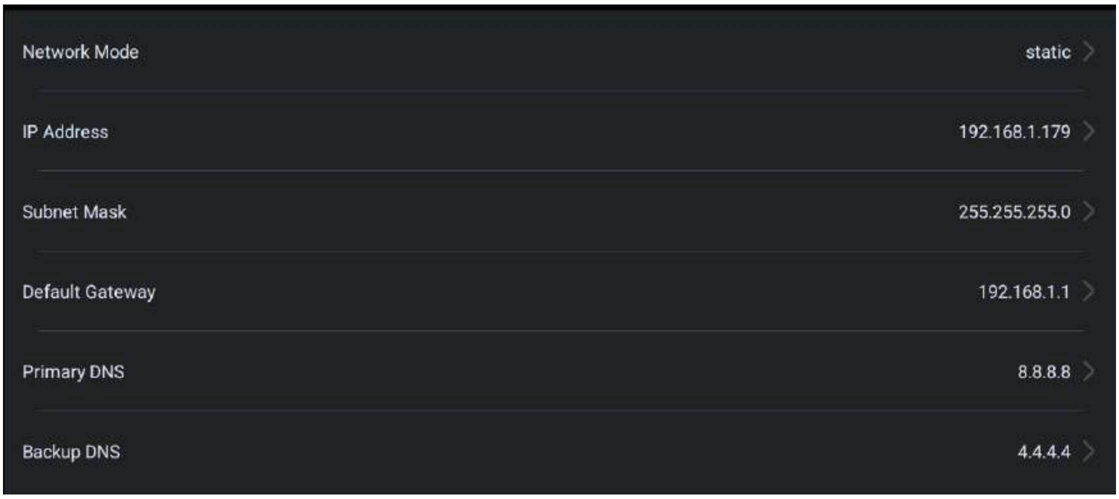


Figure 175 - Static IP Mode

14.7.2 Advanced Network

Advanced network settings are usually configured by IT administrators to improve the quality of phone services.

- **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 to learn feature to apply the VLAN ID from VLAN switch to phone its self.

- **CDP**

Cisco Discovery Protocol (CDP): A proprietary link-layer protocol for Cisco devices that shares device information (e.g., version, IP address, hardware version) with adjacent Cisco devices on the same LAN.

Table 43 - Advanced Network Settings

Parameter	Description
LLDP setting	
Report	Enable LLDP
Interval	LLDP requests interval time
Learning	apply the learned VLAN ID to the phone configuration
QoS	
QoS Mode	configure SIP 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

14.7.3 SIP Settings

This interface allows you to configure a SIP number for a line and modify the call configuration of the corresponding line. Select a SIP line on the phone, then configure the line in the [**Account Registration**] interface.

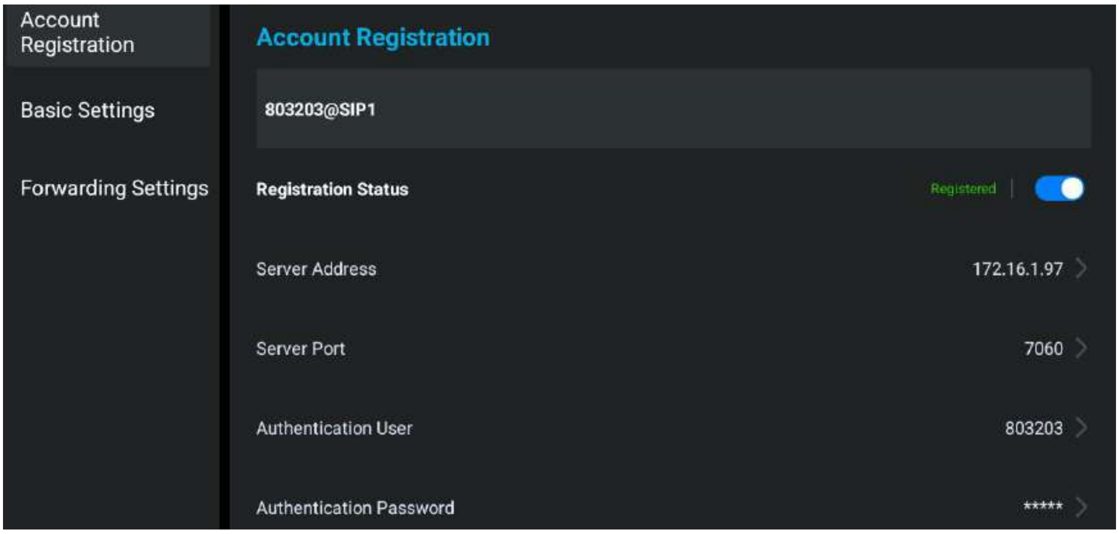


Figure 176 - Line Account Configuration Information

After selecting or filling in the corresponding configuration items, click Confirm to save the settings; the phone's SIP line will be available for normal use after confirmation. For additional configurations, you can log in via the web interface to modify them, or adjust them in the [**Basic Settings**] / [**Forwarding Settings**] interfaces.

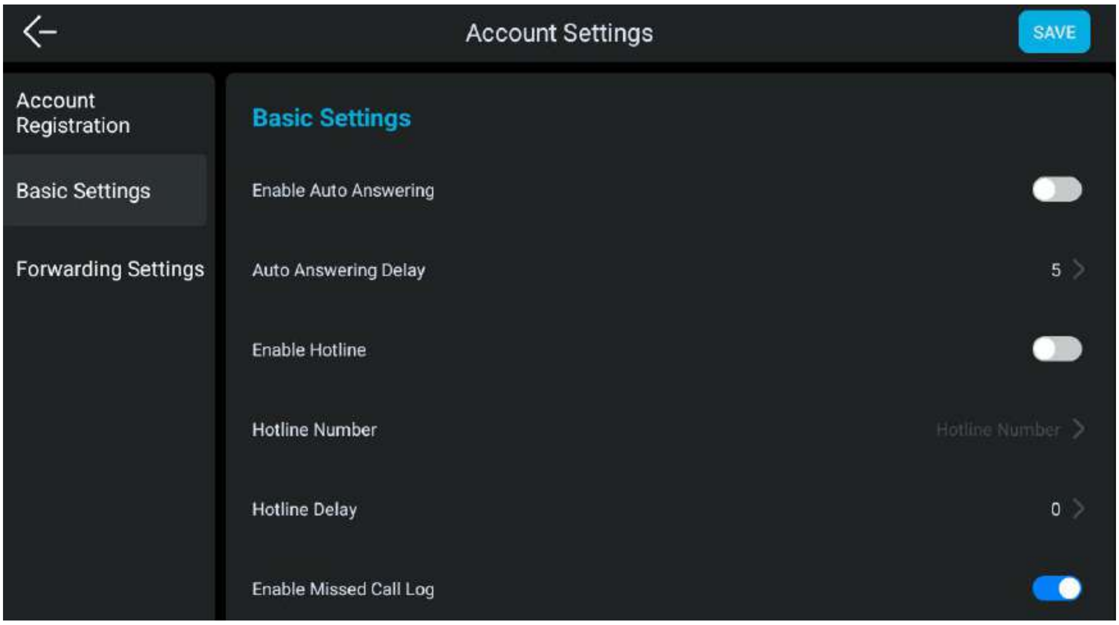


Figure 177 - Basic Line Settings

Table 44 - Basic Line Settings

Parameter	Description
Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
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 headset
Hotline Number	Set the hotline dialing number
Hotline Delay	Set the delay for hotline before the system automatically dialed it
Enable Missed Call Log	If enabled, the phone will save missed calls into the call log record.
Dial Without Registered	Set call out by proxy without registration
Use STUN	Set the line to use STUN for NAT traversal
DTMF Type	Set the DTMF type to be used for the line

About forwarding Settings, please refer to 8.12 Call Forward for detailed functions.

14.7.4 SIP Hotspot

This interface is used for the configuration of the Hotspot function. After the Hotspot function is enabled, the hotspot server will automatically connect to client devices and issue extension numbers.

For the configuration and usage of the Hotspot function, refer to 9.4 SIP Hotspot.

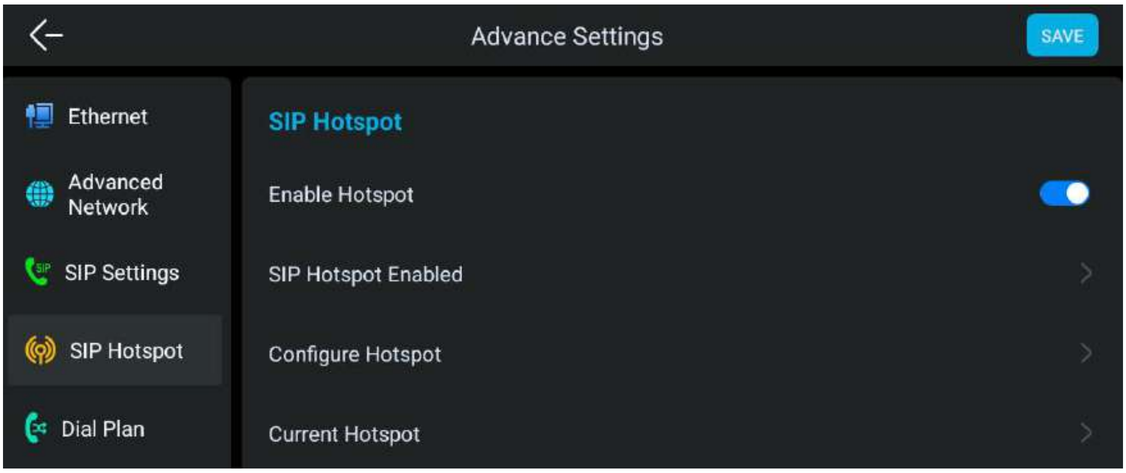


Figure 178 - SIP Hotspot Settings

14.7.5 Dial Plan

This interface is used to configure the dialing-related rules of the phone.

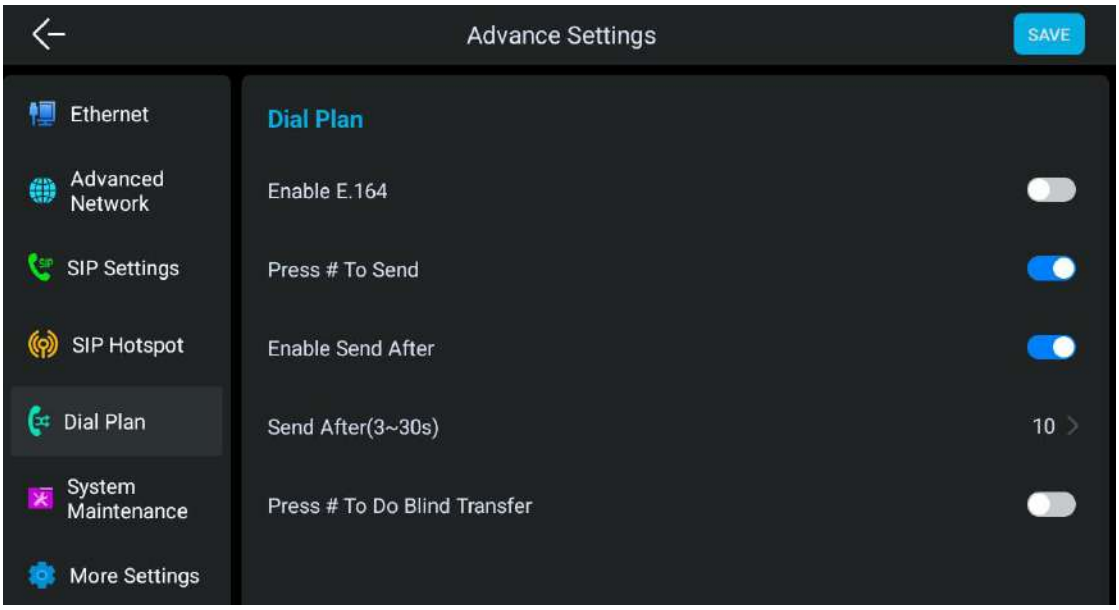


Figure 179 - Dial Plan

Table 45 - Dial Plan Settings

Parameter	Description
Enable E.164	Enable **E.164**. When dialing numbers with different leading digits, the call will be placed automatically once the configured length is reached.
Press # To Send	The user dials the other party's number and then adds the # number to dial out;
Enable Send After	The system will dial automatically after the timeout.

Send After(3~30s)	Set the timeout duration for automatic dialing.
Press # To Do Blind Transfer	Press the [Transfer] key first, and after the user enters the number to be transferred and then presses the "#" key to transfer the current call to a third party

14.7.6 System Maintenance

This interface allows you to perform factory reset, back up and restore device configurations, helping you manage the device system status, reset or migrate configurations.

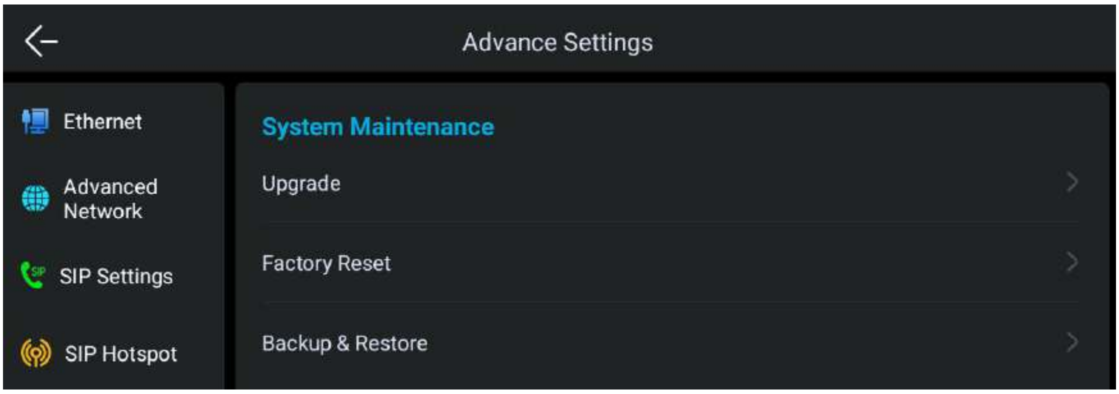


Figure 180 - System Maintenance

14.7.6.1 Upgrade

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

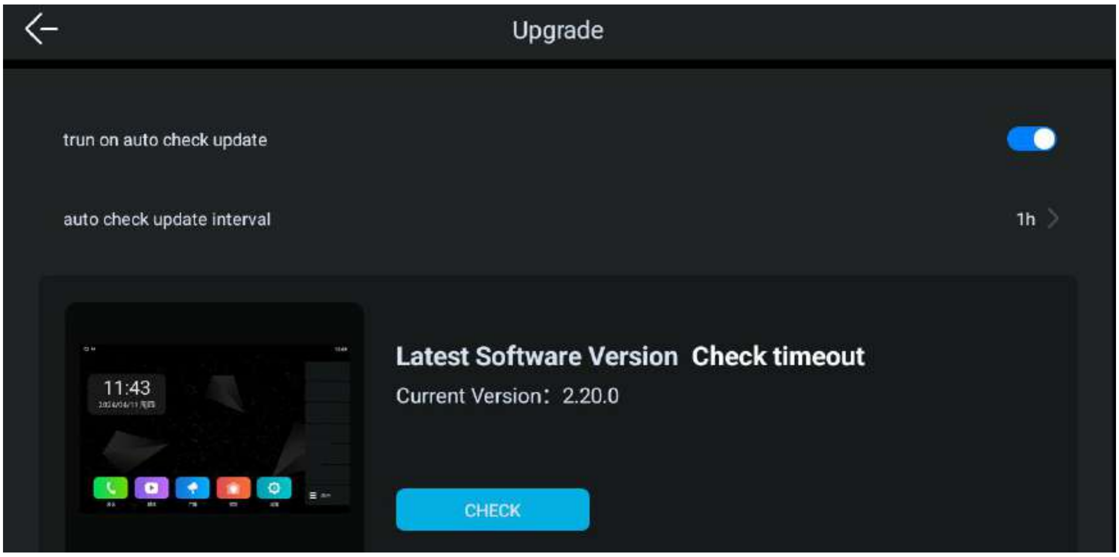


Figure 181 - Firmware Upgrade

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

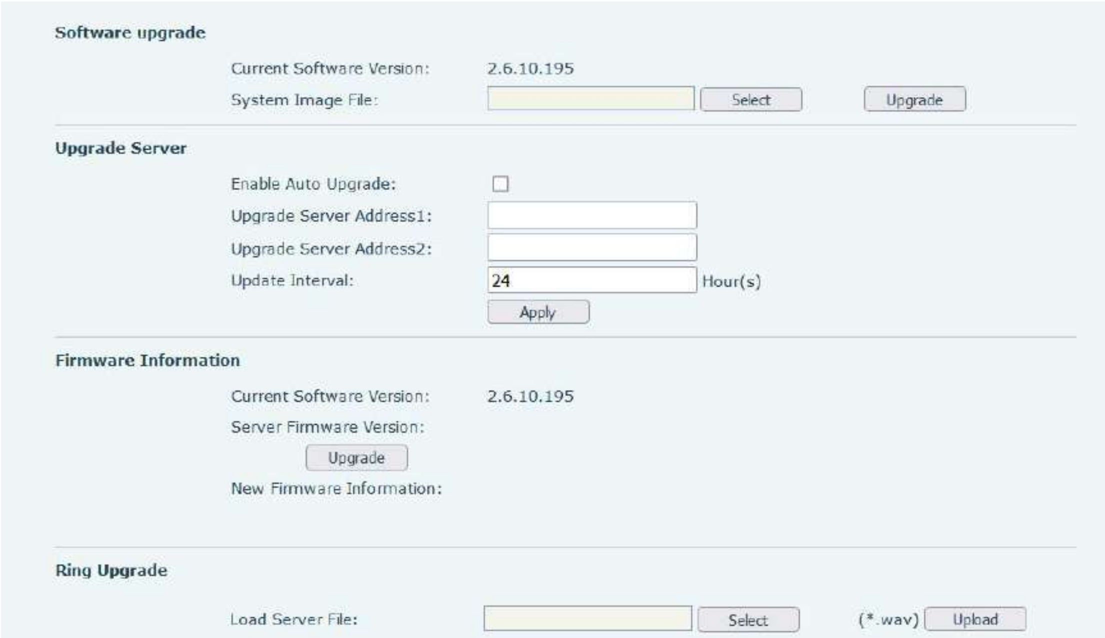


Figure 182 - Firmware Upgrade on Web

Table 46 - 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 version description 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 with underscores.

- 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=1.6.3 #Firmware

  Firmware=xxx/xxx.z　#URL,Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers.

  BuildTime=2018.09.11 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 will prompt as shown below. Click [**CHECK**] to check the version information and upgrade.

14.7.6.2 Factory Reset

- Tap [**Factory Reset]** and a confirmation pop-up will appear; tap "**Confirm**" to reset the device.

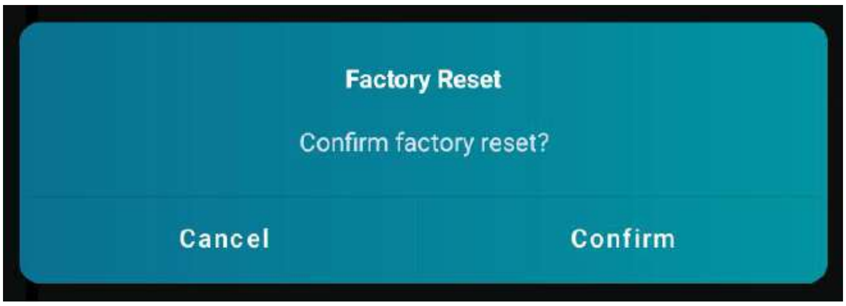


Figure 183 - Factory Reset

14.7.6.3 Backup & Restore

- In [**Backup & Restore**], you can save all current device configurations as a backup file in the system. Later, you can quickly restore the original settings by importing this file, which facilitates configuration migration or fault recovery.

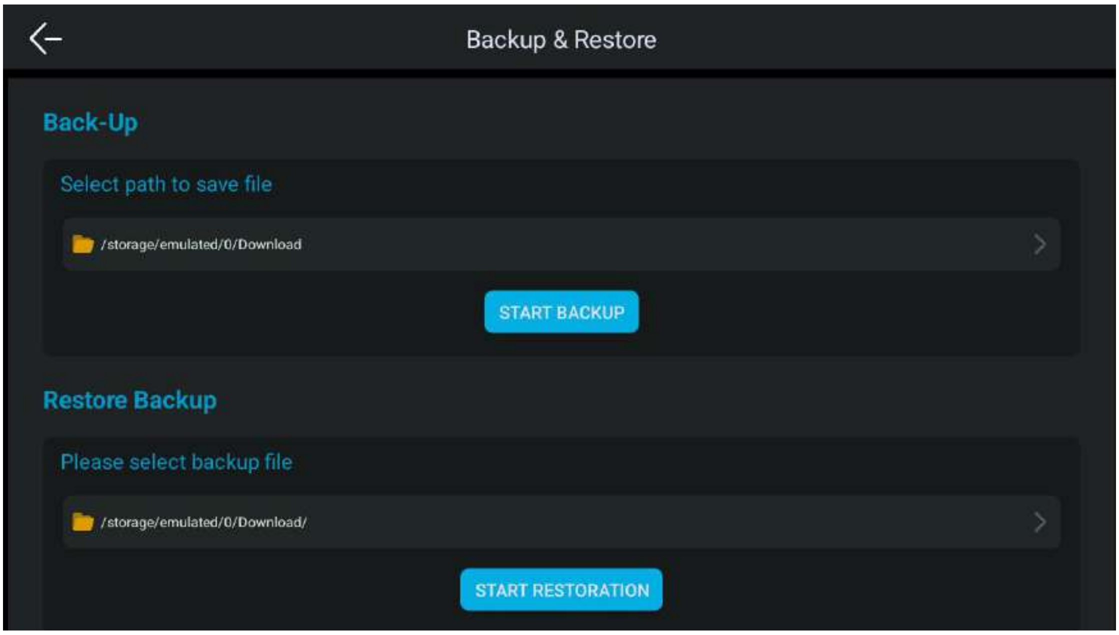


Figure 184 - Backup & Restore

14.7.7 More Settings

In [**More Settings**], you can enable the User Management function. This function is used to control the access rights of the intercom and broadcast system, and supports enabling login verification to ensure the security of system usage.

1. **Function Activation**: Navigate to the **[More Settings] >> [User Management]** module. After tapping the **"Enable User Authentication"** switch, enter the authentication password “**admin**” to activate this function.

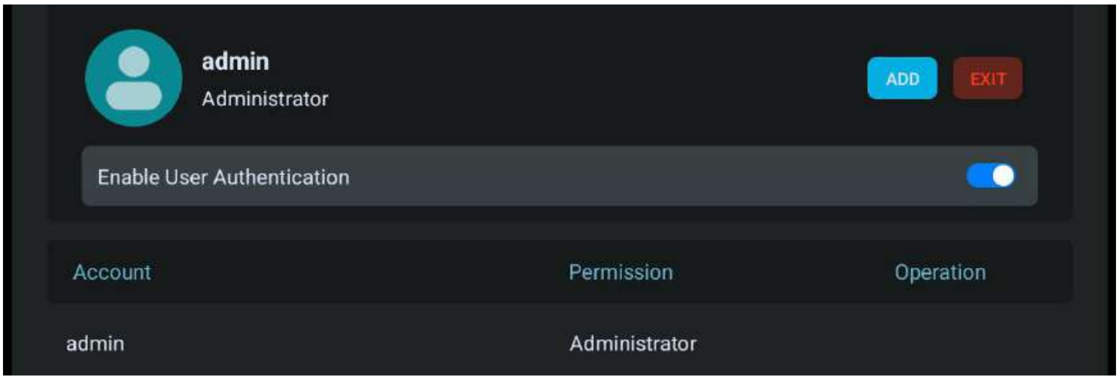


Figure 185 - User Management

2. **Login Rules**: After the User Authentication function is enabled, the login interface will pop up automatically every time the device starts up the system. You need to enter the correct username and password to access the system. The default administrator username and password are **admin/admin**.

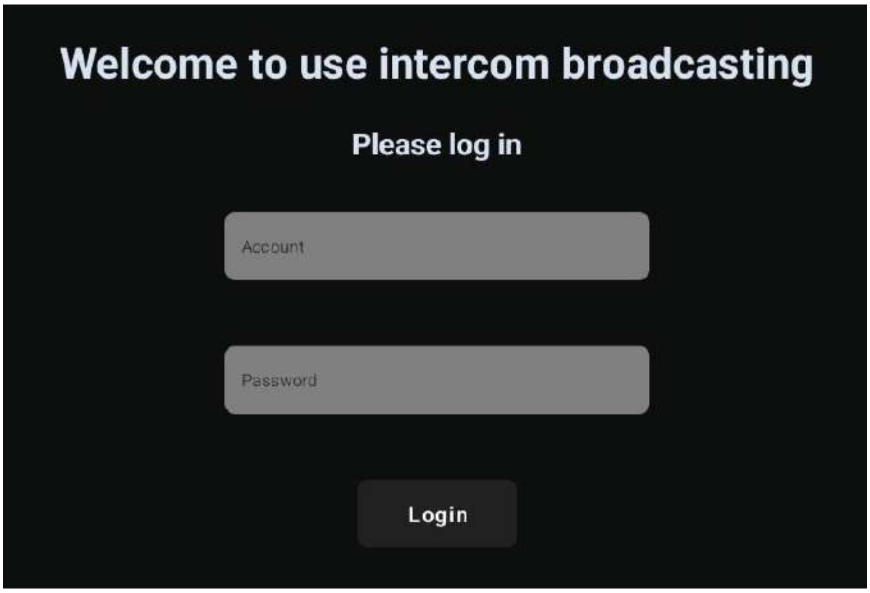


Figure 186 - Login Interface

3. After logging in with the default administrator account, you can click **"Add"** in the **"User Management"** module to create a new account.

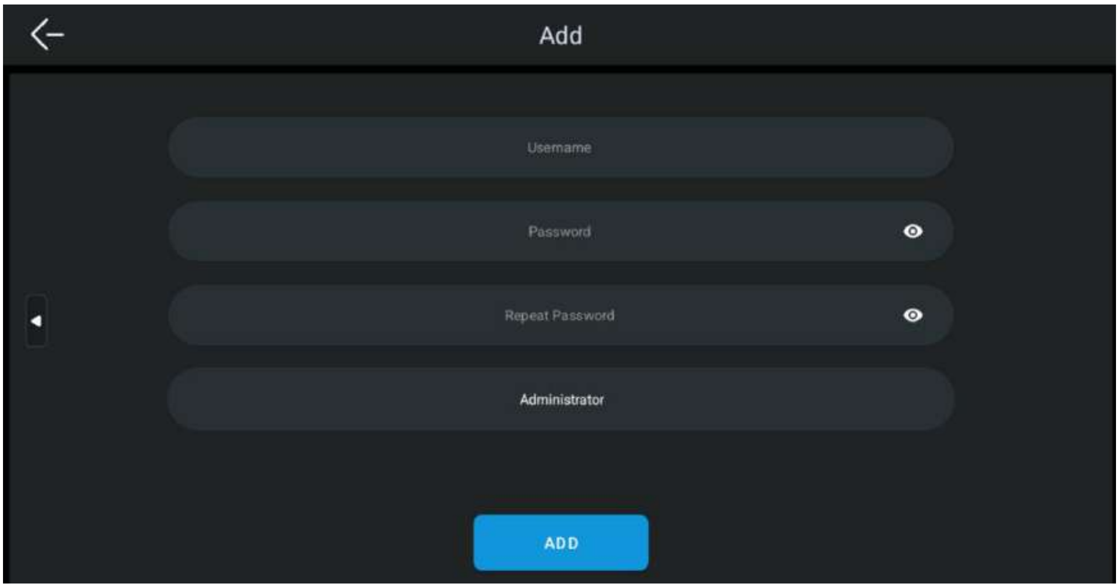


Figure 187 - Add User

15 Apps

Table 47 - Application Introduction

Chromium	Support access to various websites.
Notes	Notes and records -- convenient for users to note events, and electronic post-it notes can be viewed at any time.
Gallery	Support Bmp, Jpeg, Png image preview and save.
Files	are used to save all downloaded files of various types.
Calendar	Display and view dates, create activity reminders, etc.
Clock	Can configure alarm clock, time, stopwatch, countdown Time - supports global time zone selection.
Camera	Users can take and record images.
Video	Only supports MP4 format video playback.
Calculator	Scientific calculator - allows users to quickly process data.
Settings	This setting is the default setting of Android system
Music	Music player - can import recording and music play.
Explorer	View usb flash drive and system related files.

16 Phone Settings

16.1 Basic Settings

16.1.1 Language

Users can set the phone language via the phone interface and the web interface.

- **Phone Interface**: Refer to 14.6.3 System Language.
- **Web Interface**: Log in to the phone's web interface and set the language in the drop-down box at the top right corner of the page, as shown in the figure:

Figure 188 - Language Setting on Web

- The function of the selection box on the right side of the language setting box in the web interface is **"Synchronic Language to Device"**; if checked, the phone language will be synchronized with the web interface language; if not checked, synchronization will not be performed.

16.1.2 Date & Time

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

- **Phone Interface**: Please refer to 14.6.2 Date & Time.
- **Web Interface**: Log in to the phone's web interface and navigate to [Phone Settings] >> [Time/Date], as shown in the figure:

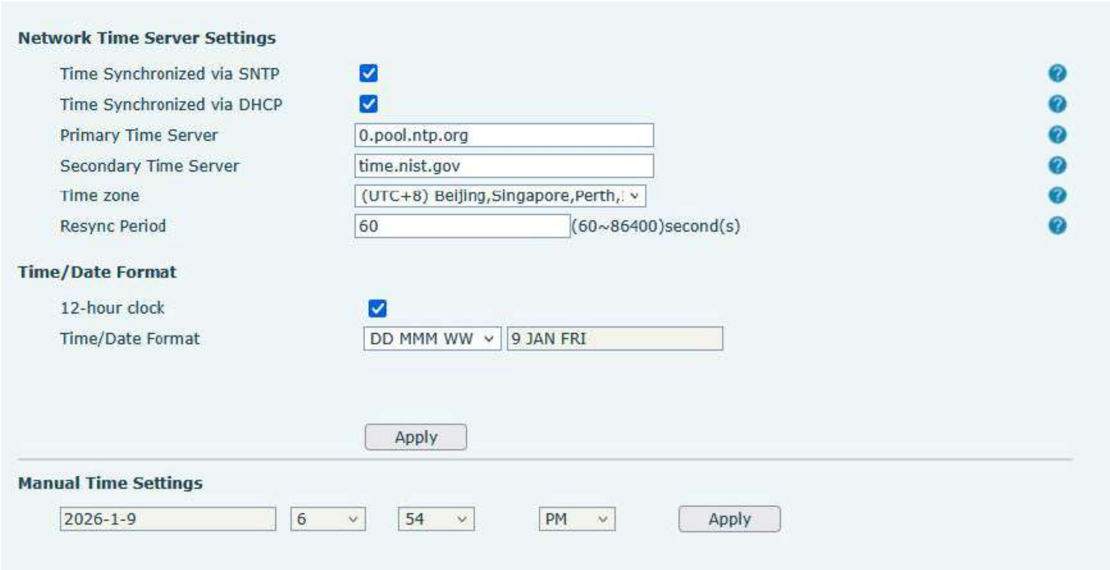


Figure 189 - Time &date Settings on Web

Table 48 - Time & Date Settings

Parameter	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
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
12-Hour Clock	Set the time display in 12-hour mode
Date Format	Select the time/date display format ● MMM DD ● YY MM DD ● YYYY MM DD
Separator	Select the separator between the year, month, and day.
Daylight Saving Time	Daylight Saving Time settings: Off/Auto/Manual (configured on the web interface).
Manual Time Settings	You can set your time manually

16.1.3 Screen

16.1.3.1 Brightness and Backlight

- **Phone interface:**

1) In standby mode, swipe down from the top of the screen to access the status bar; you can drag the slider to set the device brightness.
2) Navigate to the [Apps] >> [Settings] >> [Display] interface on the phone to adjust the brightness and change the appearance, among other operations.

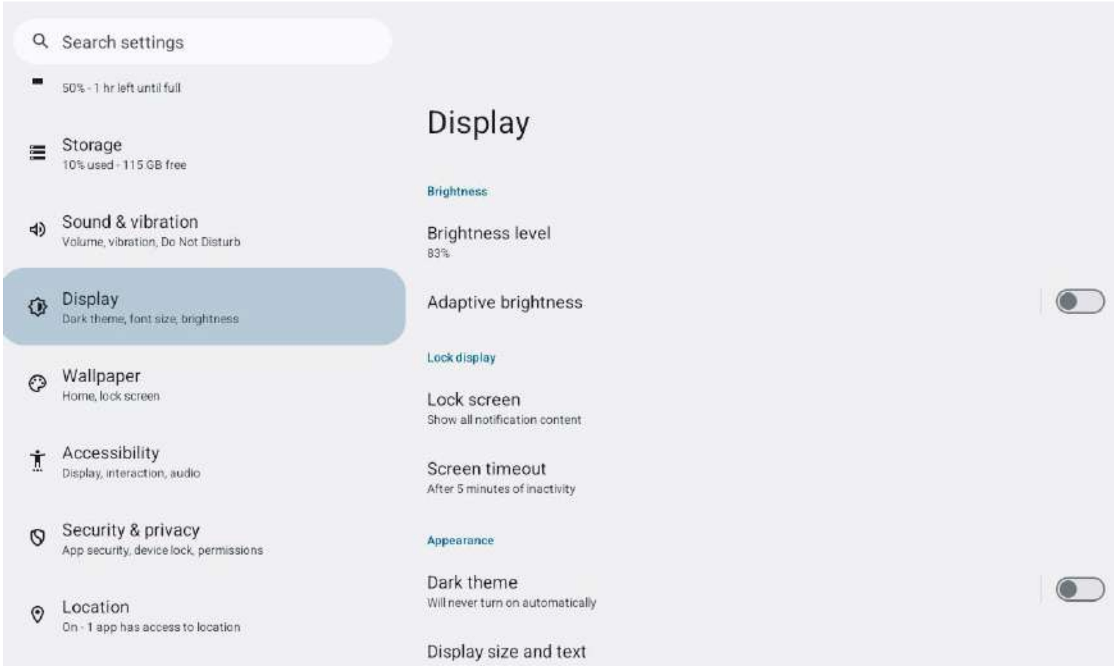


Figure 190 - Screen Setting

16.1.3.2 Screensaver

- **Phone Interface:**

When the phone is in the default standby mode, navigate to [Apps] >> [Settings] >> [Display] >> [Screen Saver] to enable "Use Screen Saver", as shown in the figure:

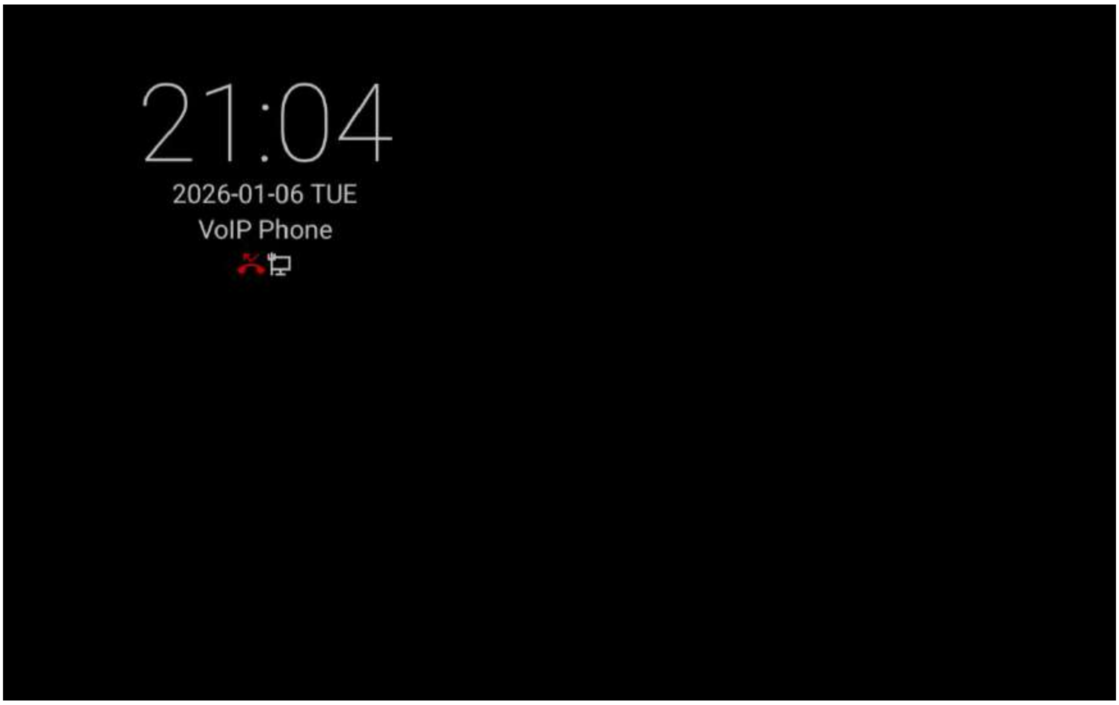


Figure 191 - Screensaver

16.1.4 Function Keys

- **Function Key Settings:**

There are six function keys displayed on the screen in standby mode, and each of them can be customized as a dsskey (Unfold keys are not supported). When expanded, there will be 29 Function dsskeys, totaling 4 pages. Users can customize each Function Key on every page.

Users can add/delete dsskey pages via the web interface, and switch between dsskey pages using the page switch key. In addition, users can also long-press each Function Key to modify the settings of the corresponding key.

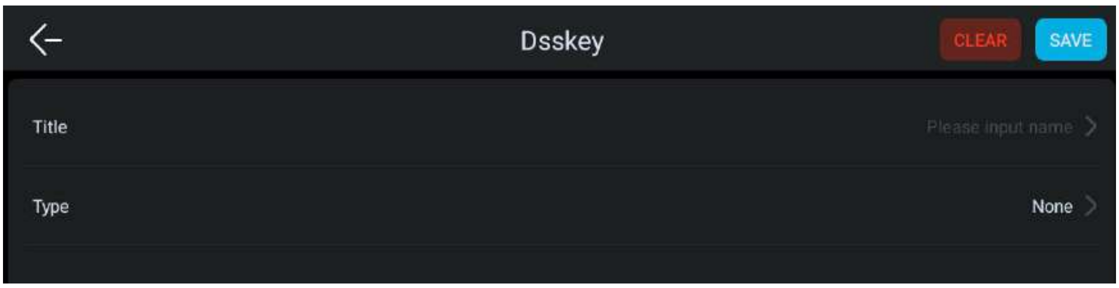


Figure 192 - Dsskey Setting

The shortcut keys have the following settings, for example,

- Memory key
  - Speed dial/intercom/BLF/attendance/MWI/callresident/call forwarding(to someone)/transfer, etc.
- line

- Function keys
  - MWI/DND/Call Hold/Call Transfer/Contacts/Redial/Pick Up/Call Log/Headset/Lock Screen/Record/SMS, etc.
- DTMF
- URL request
- Group broadcast
- Action URL
- Group broadcast monitoring
- Doorphone
- task

For shortcut keys of "speed dial key / line / URL" and other types, users can also customize the label of the keys.

For more details, please refer to 17.34 Function Key>> Function Key and 5 Appendix Table

16.1.5 Headset

16.1.5.1 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 dsskey of headset will be green light which indicates that the headset can be used normally.

On the web [**Phone settings**] >> [**Features**], you can set the headset answering function, and the ring tone for headset.

Figure 193 - Headset Function Settings on Web

16.1.5.2 Bluetooth Headset

For Bluetooth headset connection steps, please refer to 14.6.11 Bluetooth.

The use of Bluetooth headset can be divided into three types: Call Answering; Hang up; Redial.

- Call Answering

When the Bluetooth headset is connected to the phone, the incoming call can be answered by pressing the Bluetooth answer button.

- Hang up
  - When talking with Bluetooth headset, you can hang up the phone by pressing the button on Bluetooth headset.
  - When there is an incoming call, double-click the answer button to reject the call.
  - When the caller is in the ringing state, press the answer button of the headset to cancel the call.
- Redial

When the Bluetooth headset is connected, double-click the answer button to redial the number dialed last time.

Note! some models do not support double - click reject the call or redial function. Whether this function is supported or not, you can check the instruction of the headset, or connect the Bluetooth headset to the phone, and double-click the answer button to see whether it will redial.

16.2 Advanced Settings

16.2.1 Network Settings

For network settings, please refer to 14.7.1 Ethernet.

16.2.2 Web Server Type

Navigate to the [**Network**] >> [**Service Port**] page on the web interface, where you can configure the network service ports on the phone.

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.

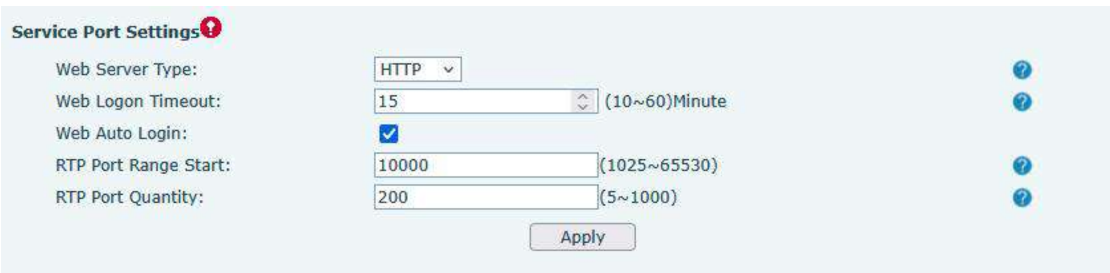


Figure 194 - Phone Configuration Web Server Type

16.2.3 Maintenance

Web interface: Login and go to [**System**] >> [**Auto provision**].

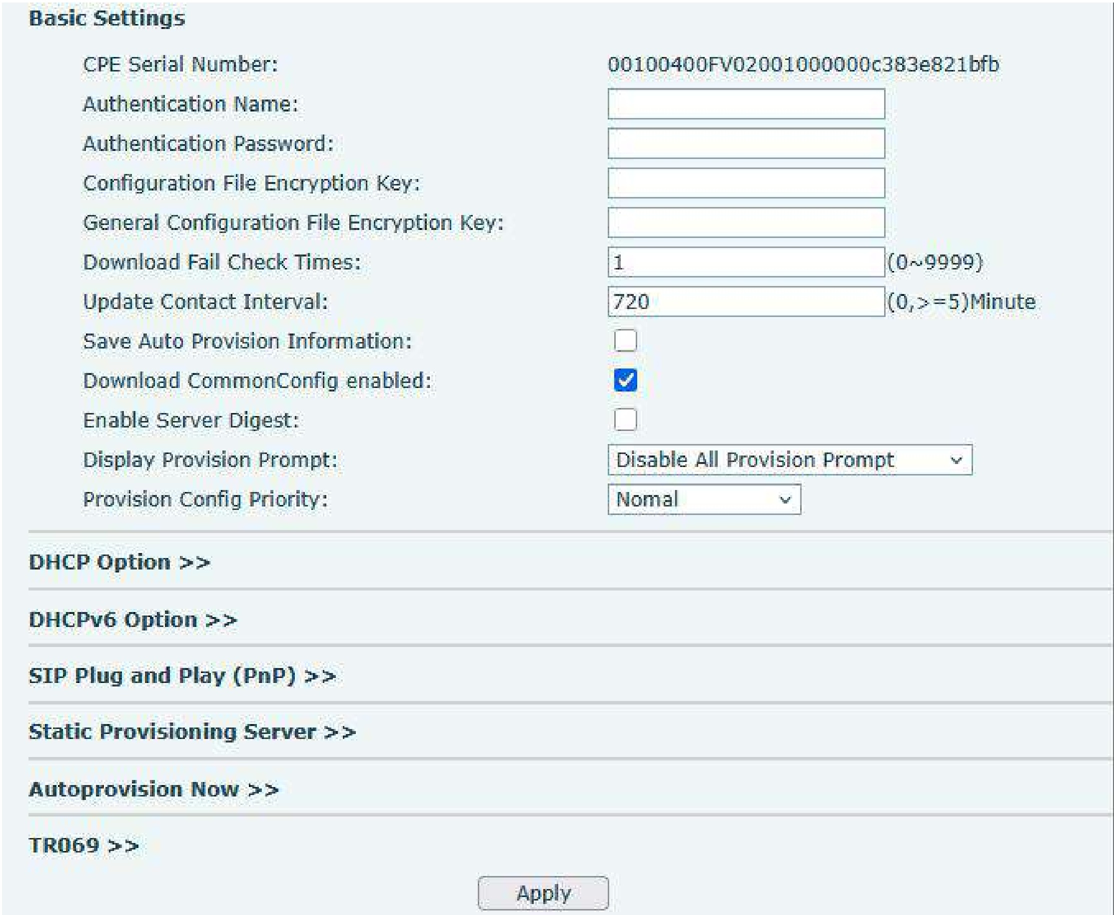


Figure 195 - Auto Provision Settings

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

Details refer to the document **Fanvil Auto Provision.**

Table 49 - Auto Provision Settings

Parameter	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	It configures the Display Provision Prompt.
Provision Config Priority	Provision Config Priority.
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.
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 and click Autoprovision Now. The phone will be provisioned immediately without the need to restart it.
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, TLS 1.3)
INFORM Sending Period	INFORM signal interval time. It ranges from 1s to 9999s
STUN Server Address	Configure STUN server address
STUN Enable	To enable STUN server for TR069

16.2.3.1 Network Settings

For network settings, please refer to 14.7.1 Ethernet.

16.2.4 Online Upgrade

16.2.4.1 Web Upgrade

Log in to the phone's web interface and navigate to the **[System] >> [Upgrade] >> [Software upgrade]** page. Select the upgrade file to start the upgrade.

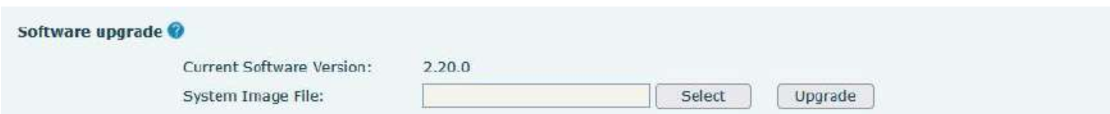


Figure 196 - Web Upgrade

16.2.4.2 Firmware Upgrade

Please refer to the section 14.7.6.1 Upgrade for details.

16.2.5 Local Upgrade

1. Place the upgrade file in a USB flash drive and connect the drive to the phone. Navigate to the **[Apps] >> [Settings] >> [About tablet] >> [Updater]** interface on the phone.

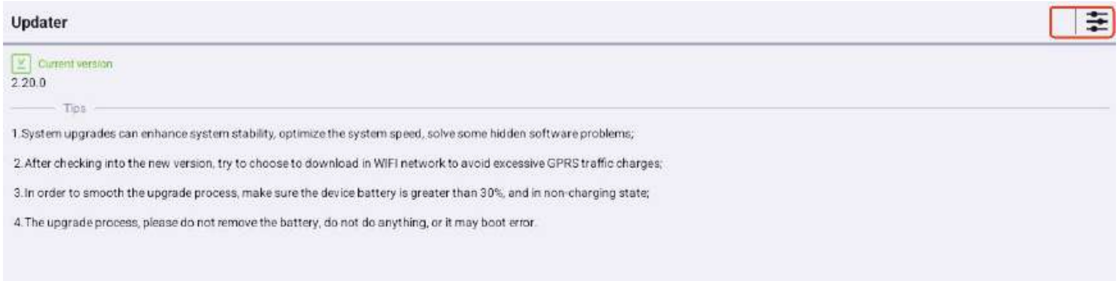


Figure 197 - System Upgrade

2. Tap the settings icon in the upper right corner to enter the System Update Settings interface, then tap Local Upgrade.

Figure 198 - Local Upgrade

3. Enter the USB flash drive folder, select the upgrade file, and tap “**INSTALL NOW**” to start the update.

17 Web Configurations

17.1 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 username and password to log in.

17.2 System >> Information

Users can get the system information of the device on this page including,

- Model
- Hardware Version
- Software Version
- Uptime

Summary of network status,

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

Besides, summary of SIP account status,

- SIP User
- SIP account status (Registered / Inactive / Trying / Timeout)

17.3 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.

17.4 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 Configurations**

Select the module in the configuration file to clear.

SIP: account configuration.

AUTOPROVISION: automatically upgrade the configuration

TR069:TR069 related configuration

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

DSS Key: Dsskey configuration

Note! The function of the Clear Configuration button is to permanently clear all basic configurations. Users can choose to retain the contents of the six sections in the options.

- **Clear Tables**

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

- **Reset Phone**

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

17.5 System >> Upgrade

This page is used to upgrade the device's software version, customized ringtone, background, Dsskey icon, etc., can also be upgraded to delete the file. Ringtones support [.wav] format.

17.6 System >> Auto Provision

The Auto Provision settings help IT managers or service providers to easily deploy and manage the devices in bulk. For the detail of Auto Provision, please refer to the document **Auto Provision Description**.

17.7 System >> Tools

Tools provided in this page help users identify issues during troubleshooting. Please refer to 18 TroubleShooting for more details.

17.8 System >> Reboot Phone

This page can restart the phone.

Figure 199 - Restart Phone

17.9 System >> System Backup

On this page, you can export or import system backup files to back up or restore system configurations.

Figure 200 - System Backup and Restore

17.10 Network >> Basic

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

17.11 Network >> Service Port

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

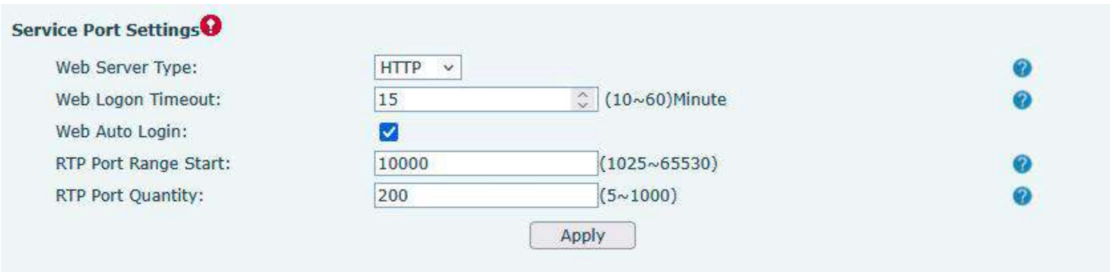


Figure 201 - Service Port Settings

Table 50 - Server Port

Parameter	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.
RTP Port Range Start	The value range is 1025 to 65535. The value of RTP port starts from the initial value set. For each call, the value of voice and video ports is added 2.
RTP Port Quantity	Number of calls.

17.12 Network >> Advanced

Please refer to 14.7.2 Advanced Network for details.

17.13 Line >> SIP

Configure the service configuration of the line on this page.

Table 51 - Line Configuration on Web

Parameter	**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 time.
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 time
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 delay time, the 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 headset
Hotline Delay	Set the delay for hotline before the system automatically dialed it
Hotline Number	Set the hotline dialing number
Dial Without Registered	Set call out by proxy without registration
Enable Missed Call Log	If enabled, the phone will save missed calls into the call log.
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 removing them from the list.
Video Codecs	Select video codecs for video preview.
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.
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 No Answer	Set the feature code to dial to the server
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 server 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. “123” vs 123
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 Chgport	Whether port updates are enabled.
VQ Name	Open the VQ name for VQ RTCP-XR.
VQ Server	Open VQ server address for VQ RTCP-XR.
VQ Port	Open VQ port for VQ RTCP-XR.
VQ HTTP/HTTPS Server	Enable VQ server selection for VQ RTCP-XR.
Flash mode	Choose 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 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	Set to enable the uaCSTA function.

17.14 Line >> SIP Hotspot

Please refer to 9.4 SIP Hotspot for details.

17.15 Line >> Dial Plan

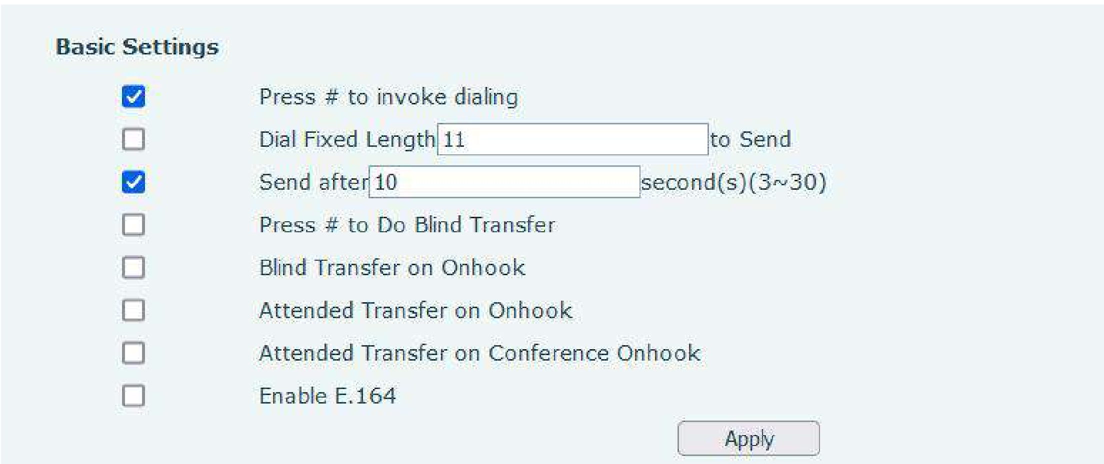


Figure 202 - Dial Plan Settings

Table 52 - Seven Dialing Methods

Parameter	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	Press the [Transfer] key first, and after 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	Press the [Transfer] key first, and 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 attended-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.

Add dialing rules:

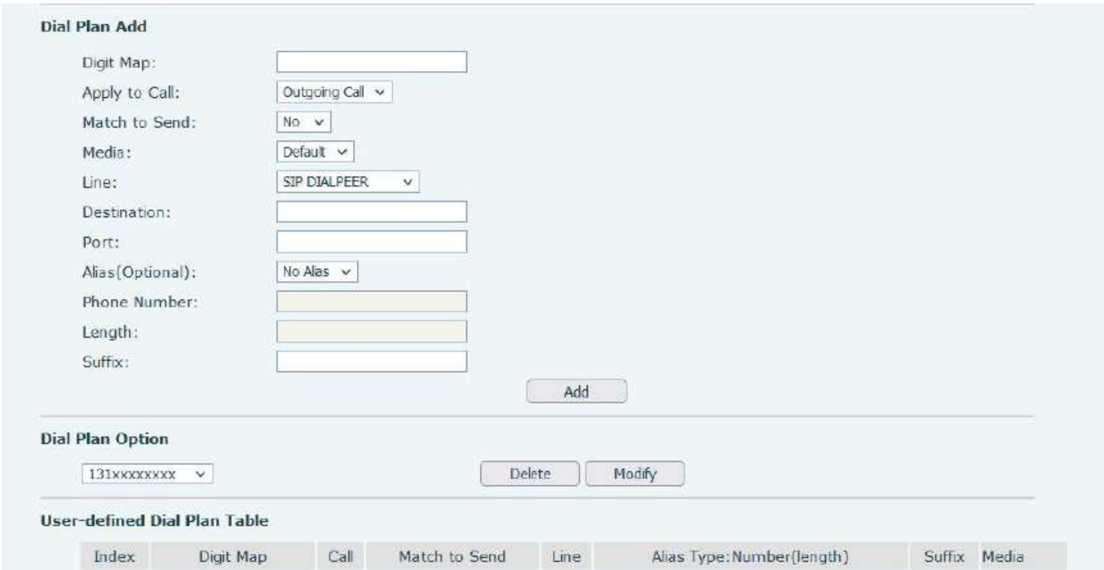


Figure 203 - Custom Setting of Dial Plan Rule

Table 53 - Dial Plan Rule Settings

Parameter	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.	
Destination	Set Destination address. This is for IP direct.

Port	Set the Signal port, and the default is 5060 for SIP.
Alias	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.	
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 dialing 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

Figure 204 - 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.

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media
1	"1T"	Out	No	Fanvil@SIP1	rep:010(1)		Default

Figure 205 - 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 131.

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix
1	"131xxxxxxxx"	Out	No	AUTO	add:0	

Figure 206 - Dial Rules Table (3)

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.

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix
1	"13[5-9]xxxxxxxx"	Out	No	AUTO	add:0	

Figure 207 - Dial Rules Table (4)

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.

17.16 Line >> Action Plan

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

Log in to the phone web, visit [**Line**] >[**Action plan**], and configure action plan rules.

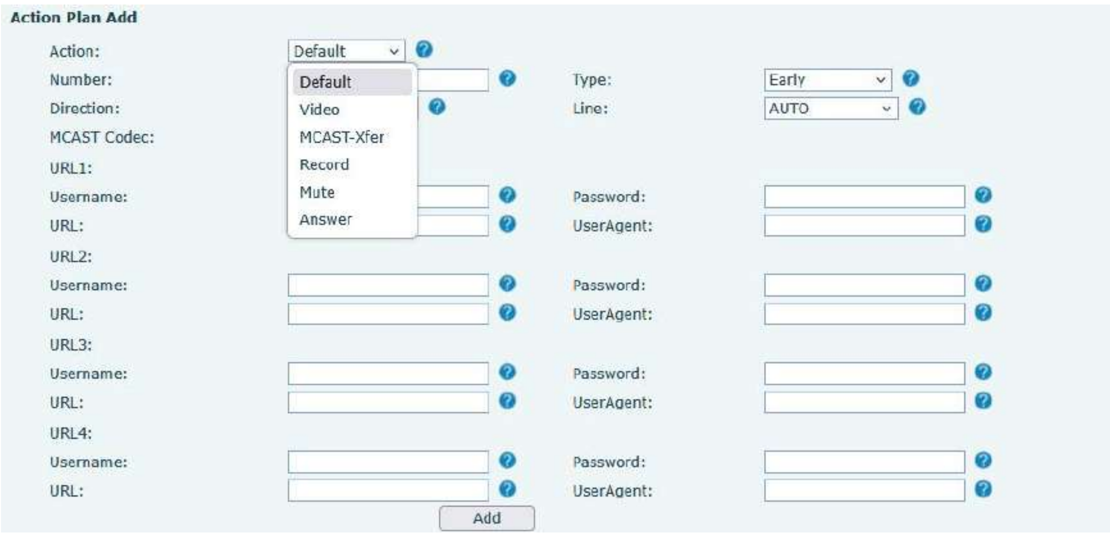


Figure 208 - Action Plan

Table 54 - Action Plan

Parameter	Description
Action	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. Video: when the rule is triggered, the phone accesses the RTSP URL configured by the URL to display the video. 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.
Number	Auxiliary phone number
Type	Early: trigger execution before call establishment. Connected: trigger execution after call establishment.
Direction	For call mode, incoming/outgoing call
Line	Set up outgoing lines.
Username	Bind the user name of the IP camera.
Password	Bind IP camera password.
URL	Video streaming information or MCAST IP address.
User Agent	Set user agent information

17.17 Line >> Basic Settings

Set up the register global configuration.

Table 55 - Set The Line Global Configuration on Web

Parameter	Description
STUN Settings	
STUN NAT Traversal	STUN NAT Traversal.
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	Enable auto answer. Incoming calls will be answered automatically after the delay time is exceeded.
Auto Answering Delay	Set the wait time for the system to answer calls automatically.
DTMF Type	Set the DTMF type for the line.
DTMF SIP INFO Mode	Set the SIP INFO mode to send * and #, or 10 and 11

Enable Preview	Enable line video preview. When there is an incoming video call, the remote party's video image will be displayed in the ringing state.
Preview Mode	Set the video preview mode.
Call-ID Format	Set the composition format of the Call-ID field in SIP messages.
Ring Type	Set the ringtone type for the line.

17.18 Line >> Hotspot Extension Management

You can manage hotspot extensions and hotspot groups, please refer to 9.4 SIP Hotspot for details.

17.19 Phone settings >> Features

Configure the phone features.

Table 56 - Features

Parameter	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 3-Way Conference	Enable 3-way conference by selecting it
Enable Auto Onhook	The phone will hang up and return to the idle automatically at hands-free mode
Auto Onhook Time	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 for 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 CallLog	Select whether to save the call log.
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	
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. More details please refer to this link
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, users 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.
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.
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

Redial Settings	
Enable Call Completion	When enabled, the device will display a call completion prompt when the dialed number is busy. After tapping confirmation, the device will automatically redial once the other party hangs up.
Enable Auto Redial	Set whether to enable automatic redial.
Auto Redial Interval	Set auto Redial Interval.
Auto Redial Times	Set auto Redial Times
Redial Enter CallLog	Set whether pressing the redial key enters the call record.
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 stands 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	
Common	Standby power lamp state, off when off, open is always bright red. Off by default.
SMS/MWI	The status of power lamp when there is unread short message/voice message, including off/on/slow flash/quick flash, default slow flash.
Missed	The state of the power lamp when there is a missed call, including off/on/slow flash/quick flash, the default slow flash.
Talk/Dial	In the talk/dial state, the power lamp state, off is off, on is always red bright, the default is off.
Ringing	Power lamp status when there is an incoming call, including off/on/slow flash/quick flash, default flash.

Mute	Power lamp status in mute mode, including off/on/slow flash/quick flash, off by default.
Hold/Held	The power lamp state, including off/on/slow flash/quick flash, is turned off by default when left/retained.
DssKey Setting	
Type	Type
BLF Status Text	Set BLF Status Text.
DssKey Status LED	Set DssKey Status LED.
Dsskey Color LED	Set Dsskey Color LED.
Notification Popups	
Display Handset Popup	When the handset is lifted and not replaced, the standby interface will display the prompt "Handset is not onhook"
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.

17.20 Phone Settings >> Media Settings

Change voice and video related settings

Table 57 - Media Settings

Parameter	Description
Codecs Settings	Select enable or disable voice encoding: G.711A/U,G.722,G.729, ILBC,opus,MPA
Video codec	
Video codec	Select to enable video encoding:H264
Media Setting	
Handset Volume	Set handset volume. Valid value:1-9.
Speakerphone Volume	Set speakerphone volume. Valid value:1-9.
Headset Volume	It configures the call volume of the headset.
Default Ring Type	it will use this ring type when you select defeault ring type.
Speakerphone Ring Volume	It configures the ringtone volume of the speaker.
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.
handset Mic Gain	It configures handset MIC gain.
Handfree Mic Gain	It configures handfree MIC gain.
Opus payload type	Set Opus payload 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	It enables or disables the VAD (Voice Activity Detection) feature on the IP phone.
Enable Voice Mail Tone	It enables or disables the IP phone to play voice message tone when dialing.
Onhook Time	Configure a minimum response time, which defaults to 200ms
Enable Hookflash	Enable the function of generating flash key when tapping hook
EHS Type	It configures EHS Type.
Video bit rate	Set the bit rate of video: 64kbps, 192kbps, 256kbps, 384kbps, 512kbps, 768kbps, 1Mbps, 1.6Mbps, 2Mbps, 3Mbps, 4Mbps
Video frame rate	Set the video frame rate: 5fps, 10fps, 15fps, 20fps, 25fps, 30fps
Video resolution	Set Video resolution: QVGA,CIF,VGA,4CIF,720P,1080P
H.264Payload Type	Set the H264 Payload Type, the value must be 96~127.
Video Iframe Interval	Set the I-frame interval, with a range of 0 to 20.
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 up RTP relay
Alert Info Ring Settings	
Value	Set the value to specify the ring type.
Ring Type	Type1-Type9

17.21 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 dsskey 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.See 9.3 Multicast for details

17.22 Phone Settings >> Action

You can set the Action URL Report type and the events to be reported.

Note! Action urls are used for IPPBX systems to submit phone events. Please refer to Action URL for details.

17.23 Phone Settings >> Time/Date

The user can configure the time Settings of the phone on this page, please refer to 16.1.2 Date & Time for details.

17.24 Phone Settings >> Time Plan

Users can set a specific time point or time period for the device to perform a certain action.

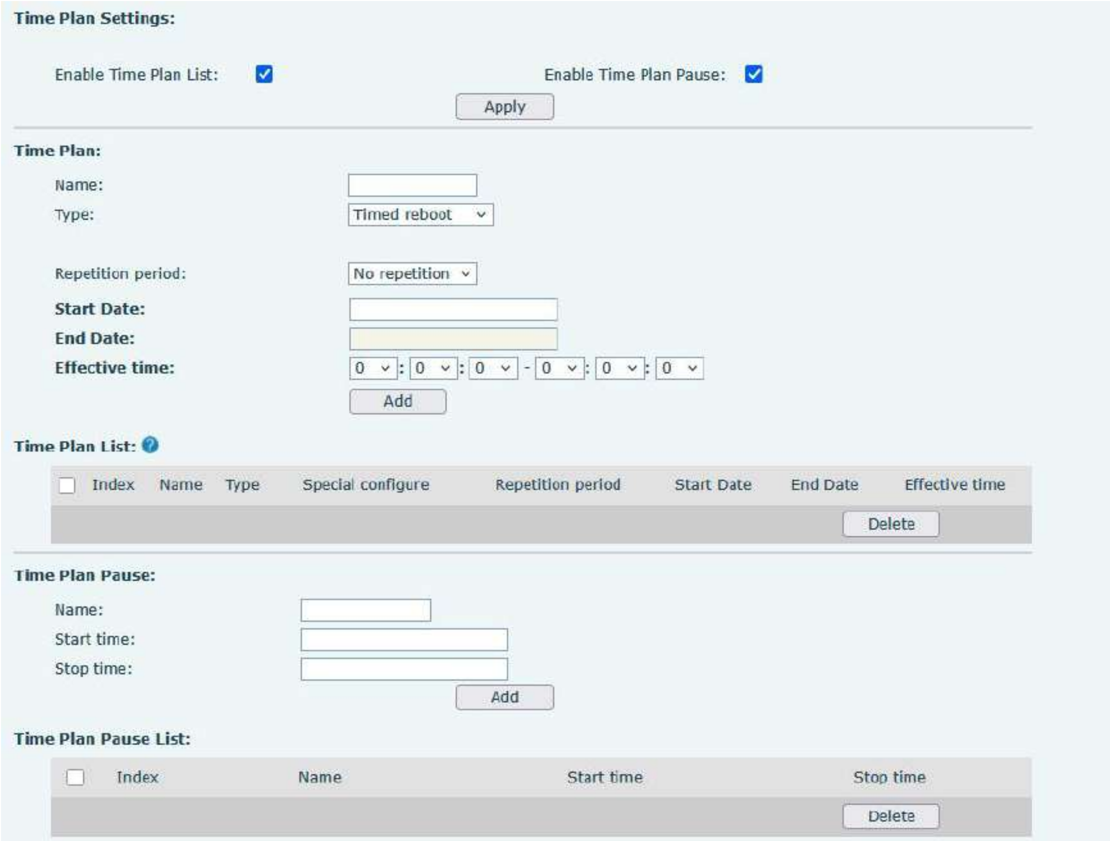


Figure 209 - Time Plan

Table 58 - Time Plan Settings

Parameter	Description
Enable Time Plan Pause	When checked, the time management tasks configured on the page will take effect.
Enable Time Plan Pause	When checked, the pause time configured on the page will take effect.
Type	Supports timed reboot\timed upgrade\timed forward\timed config
Repetition period	**No Repeat**: Execute once within the set time range **Daily**: Execute this operation at the same time range every day **Weekly**: Execute this operation at the specified time range on designated days of the week **Monthly**: Execute this operation at the specified time range on designated dates of the month
Effective time	Set the time period for execution.
Time Plan Pause	
Start/Stop time	During this time period, the time management tasks will not take effect.

17.25 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.

Figure 210 - Tone Settings

17.26 Phone Settings >> Advanced

User can configure the advanced configuration settings in this page.

- UI Preference

Set the font color for idle time, menu list, and function key list, as well as the dsskey transparency.

- LCD Menu Password Settings.

The password is admin by default.

- Configure Greeting Words

  The effect is the same as the greeting, see 14.6.1 Screen Display for details.

17.27 Library

In the Library page, you can upload audio files in [.mp3] and [.wav] formats for configuring the phone's music broadcast tasks. The uploaded audio files will be displayed in the music list; you can select a file and tap the **[Delete]** button to remove it from the local device.

Figure 211 - Library

17.28 Phonebook >> Contact

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 click **[Add new contact]** to enter contact's information and press **[OK]** button to add it.

To edit a contact, click on the checkbox in front of the contact, the contact information will be copied to the contact edit boxes, press **[OK]** button after finishing editing.

To delete one or multiple contacts, check 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 blacklist by click "Add to Blocked list" button..

17.29 Phonebook >> Cloud Phonebook

Cloud Phonebook
User can configure up to 8 cloud phonebook. 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,

- Phonebook name (required)
- Phonebook URL (required)

- Access username (optional)
- Access password (optional)

Web 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.

Figure 212 - Web Cloud Phonebook Settings

17.30 Phone book >> Call List

Table 59 - Call List Configurations

Parameter	Description
Enable Restricted Incoming List	Whether to enable the incoming call restriction list
Enable Allowed Incoming List	Whether to enable the incoming call allow list
Enable Restricted Outgoing List	Whether to enable the outgoing call restriction list
Enable Allowed Outgoing Calls	Whether to enable the outgoing call allow list

- Restricted Incoming Calls：

It is similar to a blacklist. Add the number to the blacklist, 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 blacklist or add specific prefixes to the blacklist to block calls with all Numbers with this prefix.

- Allowed Incoming Calls：

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.

- Allowed Outgoing Calls:

Add allowed numbers. When enabled, the phone will only allow outgoing calls to be made to the numbers in the list.

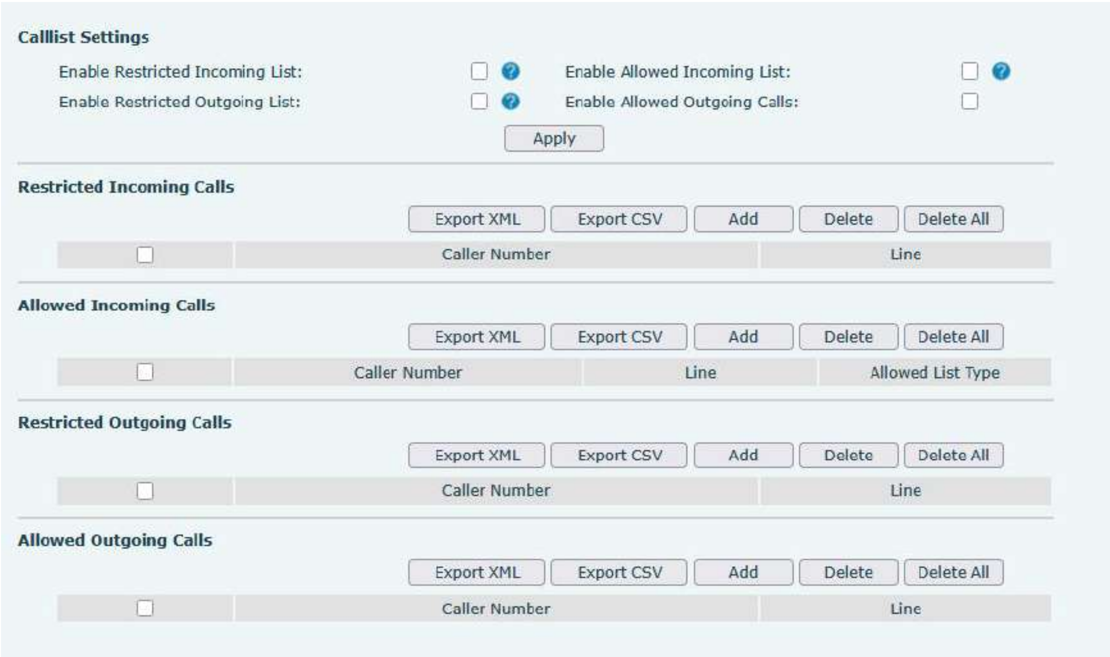


Figure 213 - Call List Settings on Web

17.31 Phonebook >> Web Dial

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

17.32 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 will not delete contacts in that group.

17.33 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 filtered 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 blacklist/whitelist.

Users can also initiate a call via the web interface by clicking on the number in the call log.
Users can also download call records conditionally and save them locally.

17.34 Function Key >> Function Key

- Function Key Configuration：

One-key transfer Settings: establish new call, blind transfer, attended transfer, one-key three-party, Play DTMF, end.
Dsskey home page: None/Page1/Page2/Page3/Page4
The device provides 116 user-defined shortcuts that users can configure on a web page.

Table 60 - Function Key Configuration

Parameter	Description
Memory Key	**BLF (NEW CALL/BXFE /AXFER)**: It is used to prompt the user of the subscribed extension's state, and it can also pick up the subscribed number, which help user monitor the state of subscribe extension (idle, ringing, a call). There are 3 types for one-touch BLF transfer method. p.s. User should enter the pick-up number for specific BLF key to fulfill the pick-up operation. **Presence**: Compared to BLF, the Presence is also able to view whether the user is online. Note: You cannot subscribe the same number for BLF and Presence at the same time **Speed Dial**: You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed. **Intercom**: This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments.
Line	It can be configured as a Line Key. User is able to make a call by pressing Line Key.
Key Event	User can select a key event as a shortcut to trigger. For example: MWI / DND / Release / Headset / Hold / etc.

DTMF	It allows user to dial or edit dial number easily.
URL	Open the specific URL directly.
Multicast	Configure the multicast address and audio codec. User presses the key to initiate the multicast.
Action URL	The user can use a specific URL to make basic calls to the phone.
Doorphone	Users can perform door open/close operations on all doorphones within a doorphone group.
Mission	Broadcast tasks can be assigned to dsskeys for quick access to the broadcast function.

17.35 Function Key >> Softkey

Table 61 - Softkey Configuration

Display page	Dsskey type
Call Dialer	Redial/Call Back/Dial/Join/Voice Mail/Local Contacts/ Pickup/Call Forward/Cancel/CallLog/Audio/Video/Dsskey
Conference	Conference(Conf)/Hold/Split/End/Release/Mute/Dsskey/Dialpad
Divert Dialed	Cancel/Forward/Local Contacts/CallLog/Dsskey
Ending	Redial/End/Complete/Auto Redial/Dsskey
Predictive Dialer	Audio/Video/Redial/Call Back/Dial/Local Contacts/Voice Mail/Call Forward/Pickup/Join/Cancel/CallLog/Dsskey
Ringing	Open/Forward/Audio/Video/Reject/Answer/Mute/Release/Dsskey
Talking Audio	Video/Transfer(XFER)/End/Mute/Hold/Open/New Call(New)/Conference(Conf)/Record/Dialpad/Screenshot/Release/Local Contacts/Listen/Dsskey
Talking Video	Video/Transfer(XFER)/End/Mute/Hold/Open/New Call(New)/Conference(Conf)/Record/Dialpad/Snapshot/Screenshot/ Release/Local Contacts/Listen/Dsskey
Transfer Dialer	Dial/Audio/Video/Transfer(XFER)/Local Contacts/CallLog/Cancel/Dsskey
Trying	Cancel/Release/Dialpad/Transfer(XFER)/Dsskey

The Soft Dsskey only supports the doorphone type. For detailed instructions, please refer to 10.2.2.3 One-Click Door Opening.

17.36 Function Key >> Advanced

- **Global key Settings**

The default configuration is empty, and the global memory key function can be configured.
The configured memory key has a call path. If the global configuration is maintained, pressing the memory key again will maintain the call path. If the same configuration hung up, press the memory key again will hang up this road call.

- **Programmable key Settings**

Please refer to the 17.35 Softkey.

- **IP Camera List**

Figure 214 - IP Camera List

17.37 Application >> Doorphone Settings

The user can add doorphone, see 10.2.2 Doorphone for details.

17.38 Application >> Manage Recording

See 8.16 Record for details of recording.

17.39 Security >> Web Filter

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

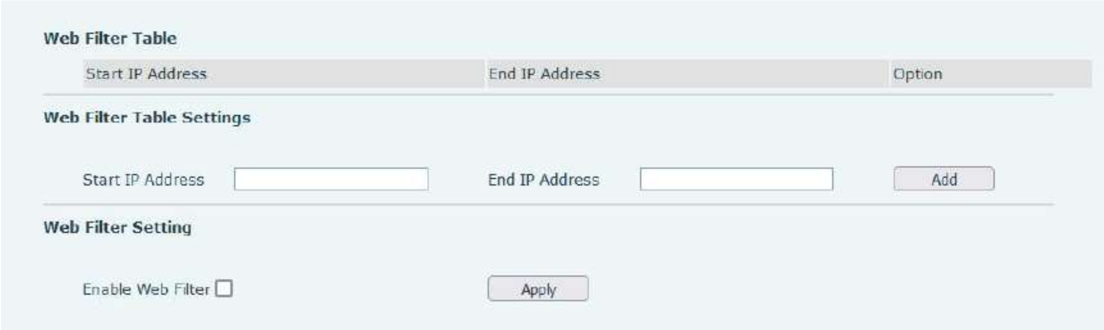


Figure 215 - Web Filter Settings

Figure 216 - 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 from the drop-down menu, 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 to the web page.

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17.40 Security >> Trust Certificates

Set whether to open license certificate and general name validation, select certificate module.
You can upload and delete uploaded certificates.

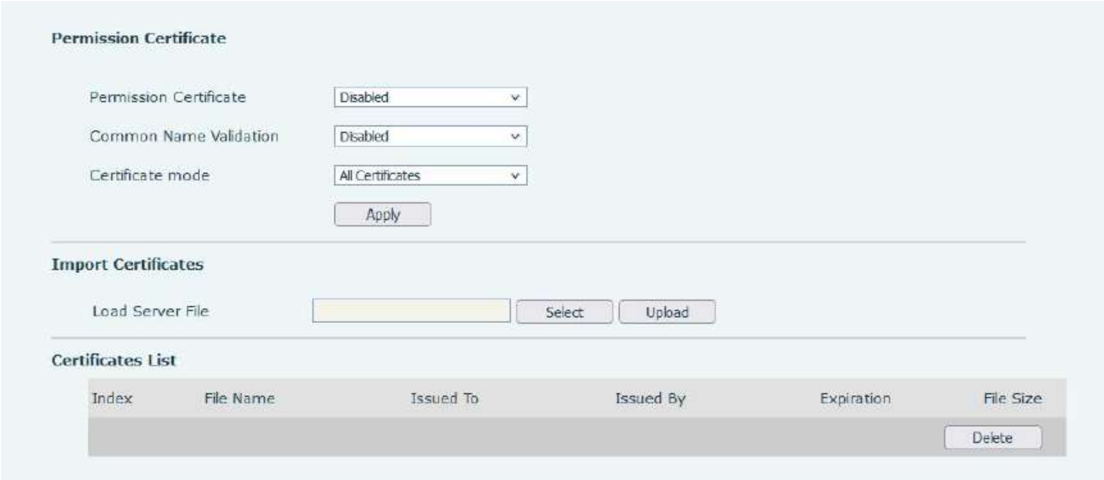


Figure 217 - Trust Certificate

17.41 Security >> Device Certificates

Select the device certificate as the default and custom certificate.

You can upload and delete uploaded certificates.

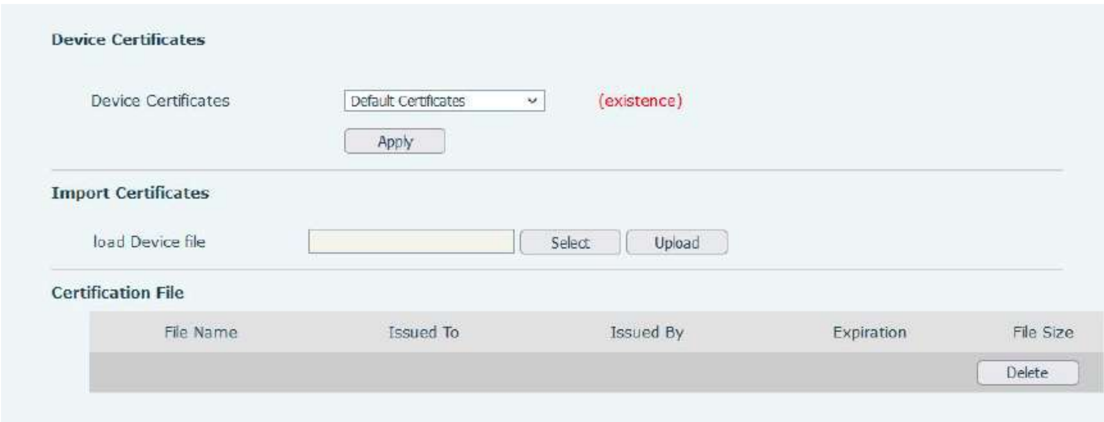


Figure 218 - Device Certificate Setting

17.42 Security >> Firewall

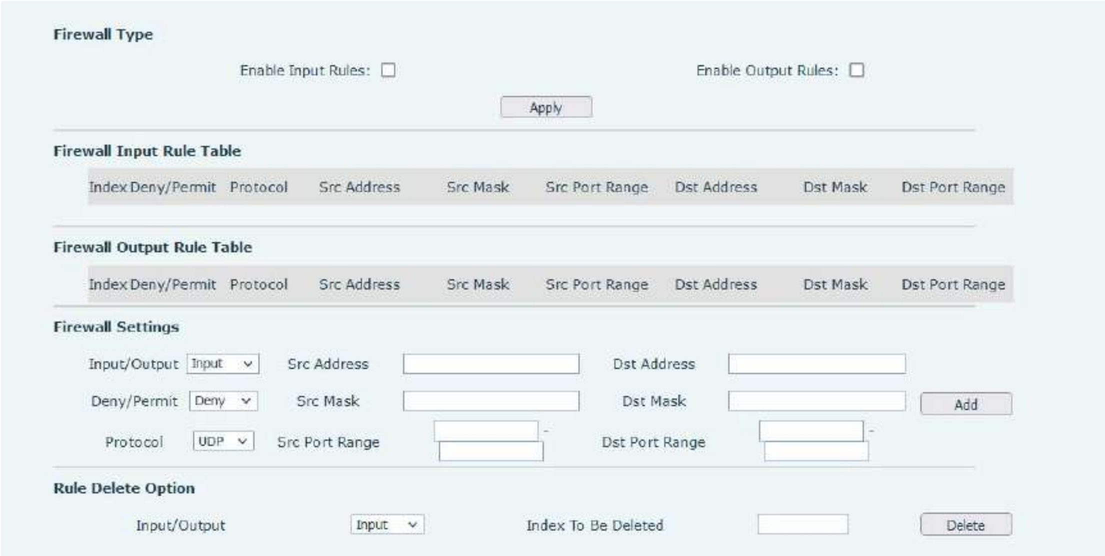


Figure 219 - Network Firewall Settings

This page allows you to enable inbound and outbound firewalls and configure their rules. These settings prevent malicious network access or restrict internal users from accessing external resources to 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.

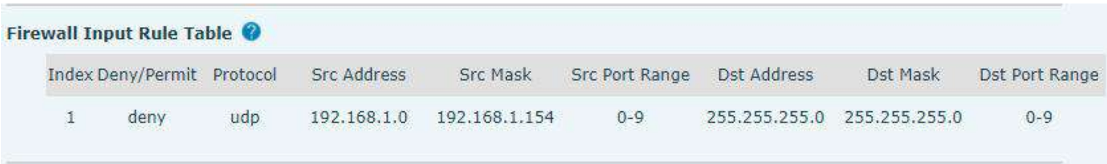
Considering the complexity of firewall Settings, the following is an example to illustrate:

Table 62 - 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 | IP.
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	udp	192.168.1.0	192.168.1.154	0-9	255.255.255.0	255.255.255.0	0-9

Figure 220 - 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.

Figure 221 - Rule Delete Option

In [**Rule Delete Option**], you can select the list you want to delete and click [**Delete**] to delete the selected list.

17.43 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 18.6 Get Log Information.

17.44 Security Settings

This interface is used to configure the input and output ports, see 14.3.3.1 Security Settings for details.

18 Troubleshooting

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.

18.1 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.

18.2 Reboot Device

Users can reboot the device from soft-menu, [**Menu**] >> [**Phone settings**] >> [**System**], and press [**Reboot**], Or, simply remove the power supply and restore it again.

18.3 Reset Device to Factory Default

Reset Device to Factory Default will erase all users' 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**] >> [**phone setting**]>> [**maintain**] , and then input the password to enter the interface. Then choose [**Phone Reset**] and press [**Reset**]. The device will be rebooted into a clean factory default state.

18.4 Screenshot

If there is a problem with the phone, the screenshot can help the technician locate the function and identify the problem. In order to obtain screen shots, log in the phone web [**System**] >> [**Tools**], and you can capture the pictures of the main screen (you can capture them in the interface with problems).

Figure 222 - Screenshot

18.5 Network Packet 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 "Network Packets 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.

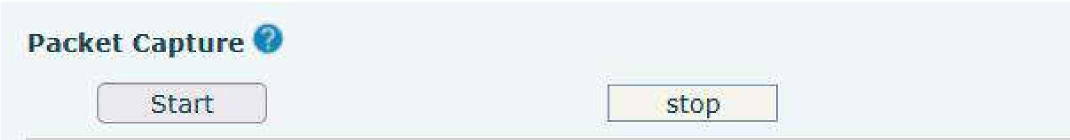


Figure 223 - Web Capture

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

18.6 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 **[System]**>>**[Tools]** >>**[Export Debug Data]**, **[Export]** to local analysis or send the log to the technician to locate the problem.

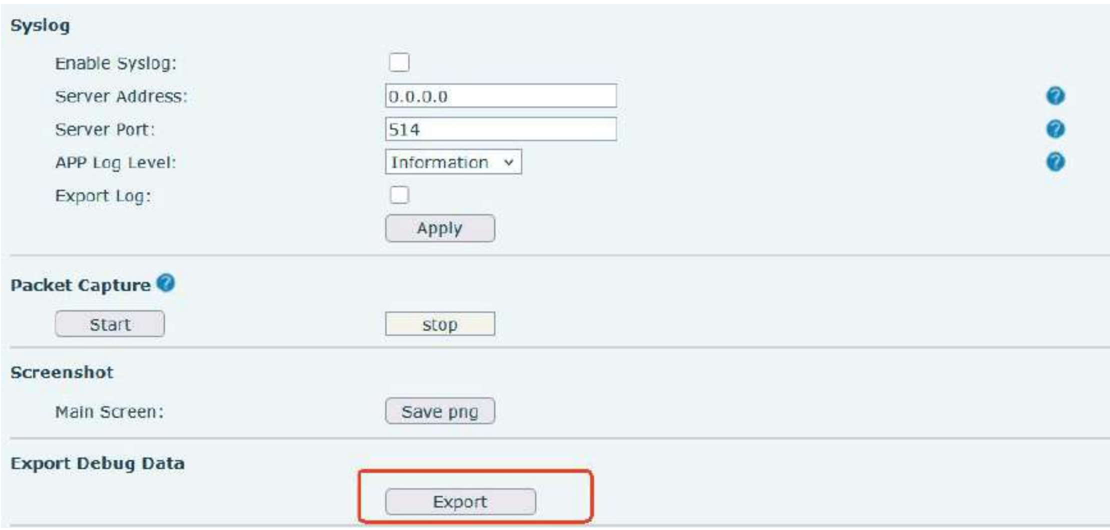


Figure 224 - One-Click Export

18.7 Common Trouble Cases

Table 63 - 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 the standard power adapter provided by the manufacturer or a PoE switch that meets

	specifications 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 “18.5 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 Headset ( ) port. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Poor Audio or Low Volume in headset	1. There are two headset wire sequence in the market. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Audio is choppy 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.