FEDERAL SIGNAL
Safety and Security Systems

IP Phone for Intelligent Systems

Indoor Controller
Model: I-IPTEL1

Description, Specifications, Installation, and User Manual

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Limited Warranty

This product is subject to and covered by a limited warranty, a copy of which can be found at www.fedsig.com/SSG-Warranty. A copy of this limited warranty can also be obtained by written request to Federal Signal Corporation, 2645 Federal Signal Drive, University Park, IL 60484, email to info@fedsig.com or call +1 708-534-3400.

This limited warranty is in lieu of all other warranties, express or implied, contractual or statutory, including, but not limited to the warranty of merchantability, warranty of fitness for a particular purpose and any warranty against failure of its essential purpose.

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⚠ WARNING

It is important to follow all instructions shipped with this product. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

Listed below are important safety instructions and precautions you should follow:

Important Notice

Federal Signal reserves the right to make changes to devices and specifications detailed in the manual at any time in order to improve reliability, function, or design. The information in this manual has been carefully checked and is believed to be accurate; however, no responsibility is assumed for any inaccuracies.

- FIRE HAZARD: Do not put the device on carpets, cushions, in direct sunlight, or any ill-ventilated place because the phone could overheat enough to burn you or cause a fire.
- SHOCK HAZARD: Do not damage the power cord. If the power cord or plug is impaired, do not use it because it may cause fire or electric shock.
- SHOCK HAZARD: When lightning, do not touch the power plug, it may cause an electric shock.
- SHOCK HAZARD: Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or it may cause fire, or electric shock.
- Use the external power supply that is included in the package. Other power supplies may cause damage to the phone and affect the behavior or induce noise.
- Before using the external power supply in the package, check the power voltage. Inaccurate power voltage may cause fire and damage the phone.
- Do not drop, knock, or shake the phone. Rough handling can break internal circuit boards.
- Avoid exposing the phone to high temperatures, temperatures below 0°C, or high humidity. It may damage the phone.
- Avoid wetting the unit with any liquid. It may damage the phone.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. It may damage the phone. Wipe it with a soft cloth slightly dampened in a mild soap and water solution.

After installation, test the system to confirm that it is operating properly. Test the system regularly to confirm that it will be operational in an emergency.

Each Informer device has its own Installation Manual. See fedsig.com for online manuals.

Read and understand the information contained in this manual.

General Description

Introduction

This comprehensive user manual describes the features on the IP Phone and how to configure using the phone and web interface. You can use the IP Phone with the Federal Signal line of IP-based speakers. The Federal Signal Informer series IP products incorporate industry-standard Web, SIP phone, IP Multicast, and Modbus® PLC interfaces to simplify integration with existing systems. Each Informer device has its own Installation Manual. See fedsig.com for online manuals.

The IP Phone can be used with the Federal Signal model UVRI-B Alarm Control and Fire Alarm interface and the following list of Federal Signal Informer series IP products:

- Informer-IP Desk Mount (I-IP-IO)
- Informer-IP Wall Mount (I-IPW)
- Informer15 Speaker (I-IP15)
- Informer100 Speaker (I-IP100AC and I-IP100DC)
- Informer-PA for Public Address Interface (I-IP2)
- Informer Sensor Interface Unit (I-IPSIU)

Federal Signal Informers can be set up in a wide variety of networks and configurations. This manual provides a standard setup and programming process for IP Phones.

I-IPTEL1 Overview

The IP Phone for Intelligent Systems (I-IPTEL1) can multicast live PA directly to a multicast group of speakers and then multicast DTMF control commands to activate pre-recorded voice messages or siren tones stored within the speakers. Include the IP phone on an existing SIP phone system or LAN, or deploy the phones and speakers on a separate LAN/VLAN dedicated for emergency or frequent VoIP use. The IP phone supports RTP multicast and DTMF activation of pre-recorded siren tones/pre-recorded voice messages on the Federal Signal IP series of devices.

It is easy to configure the phone with preset buttons to provide one-touch dialing and pre-configured speaker activations. When used with the Federal Signal line of IP-based speakers, you can configure a low-cost alerting system for industrial and campus environments. An IP phone can call individual speakers for live Public Addresses and then use a DTMF keypad to activate pre-recorded voice or siren tones.

The phone can multicast live PA directly to a multicast group of speakers and then multicast DTMF control commands to speakers to activate pre-recorded voice messages or siren tones to an unlimited number of speakers simultaneously. This allows the creation of zones for wide areas or specific notification groups. You can use multiple phones to create multi-location control and activation points. The phones can also be configured to allow direct SIP calls between phones and speakers without a SIP server.

Power the phone via PoE or from an (included) external power supply. The IP phone has three color displays for ease of use. You can configure each button for name and color. The phone has high-definition audio on the handset and speaker phone. The phone is configurable for 19 different languages.

The Federal Signal line of Informer IP Speakers includes an internal SD card and amplifier to deliver tone warnings and intelligible voice messages from locally stored memory, enabling autonomous operation from contact closures and single button controls from the I-IPTEL1 phones. The Informer IP Speaker also has remote volume control for optimizing sound levels across your alerting area. The remote volume control also includes an ambient-noise-monitoring capability to adjust volume automatically depending on external noise levels.

Informer IP Speaker allows connecting up to four external switches to activate predefined alert events. You can program and configure the Informer IP Speaker as a standalone device to only use the inputs for activation. This may be useful if the location has no network connectivity but where voice and tone alerts from locally activated inputs are required. The Informer IP Speaker can be networked later for additional functionality.

Features

The I-IPTEL1 has the following features.

- Over 60 programmable buttons
- Phone-to-phone dialing without SIP Server
- Phone-to-speaker for live PA
- Direct SIP calling and multicast paging
- Ability to activate pre-recorded voice messages or siren tones
- Fully functional with Informer15 and Informer100 Speakers
- Static IP or DHCP
- Supports LLDP, CDP, DHCP VLAN, VLAN Custom Options, QoS, WAN VLAN, and 8021x
- Configurable from Web User Interface
- Multicast DTMF to control Federal Signal's IP products
- Desktop stand
- Ability to synchronize with Network Time Protocol (NTP) clock

Applications

The I-IPTEL1 has the following applications.

- Indoor and outdoor locations
- Muster or call stations
- Campuses or school alerting
- Use with video security systems
- Plant wide live-PA announcements
- Loud industrial facilities
- Weather alerting

- Food processing locations
- Plant wide alerting system

Ordering Information

Table 1 Ordering Information

Model	Description
I-IPTEL1	IP Phone with 60 programmable buttons
I-IP15	15 W PoE+ indoor/outdoor speaker
I-IP100	100 W indoor/outdoor speaker
I-IPIO*	1/2 W desktop speaker
I-IPW*	1/2 W wall mount speaker
I-IP2*	Two-channel 1/2 W audio outputs module
I-IPSIU*	I/O module with 16 contact inputs, 8 contact outputs, 1/2 W audio output

*Requires Series C Model.

See www.fedsig.com for additional product information.

Unpacking the Device

Ensure that the parts listed are included in the package. If you are missing any parts, contact Customer Support.

The following is a list of the contents:

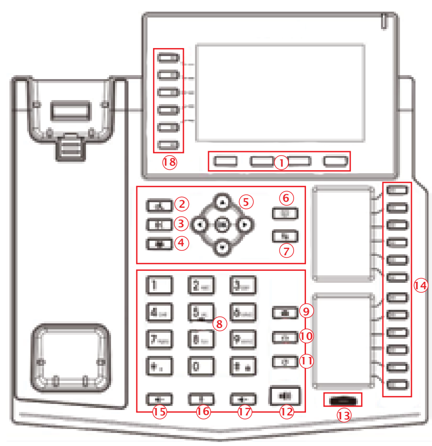
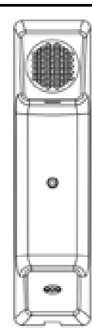
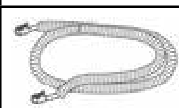
- IP Phone
- Handset
- Handset Cord
- Ethernet Cable
- Stand
- Power Adapter

Specifications

Table 2 Specifications

Physical LAN Port:	Physical: 10/100/1000 Mbps Ethernet, dual bridged port for PC bypass
IP Configuration	Static IP / DHCP
Power Consumption (PoE)	10.46 W maximum
Power Consumption (AC Adapter)	9.13 W maximum
DC Power Input	5 V / 2 A max
Operating temperature range	32°F to 113°F (0°C to 45°C)
Working Humidity	Up to 95%, non-condensing
Installation	Desktop Stand
Handset/Hands-free-/Headset mode	
Device Dimensions	Desktop Stand (Angles 1): 9.4 x 7.8 x 7.3 inches (239 x 199.3 x 185.8 mm) Desktop Stand (Angles 2): 9.4 x 7.4 x 7.7 inches (239 x 188.5 x1 95.6 mm)

Table 3 Contents list

Description	**Picture**
IP Phone	
Handset	
Handset Cord	
Ethernet Cable	

Description	Picture
Stand	
Power Adapter	
IP Phone for Intelligent Systems Description, Installation, and Configuration Manual	Manual number 25500823

Installation and Setup

Ways to Provide Power to Your IP Phone

You can provide power to your phone in the following ways:

- If your network supports Power over Ethernet (PoE), you can plug your phone into the network. Plug an Ethernet cable into the Ethernet phone port and into the network.
- Use the power adapter that comes with your phone.

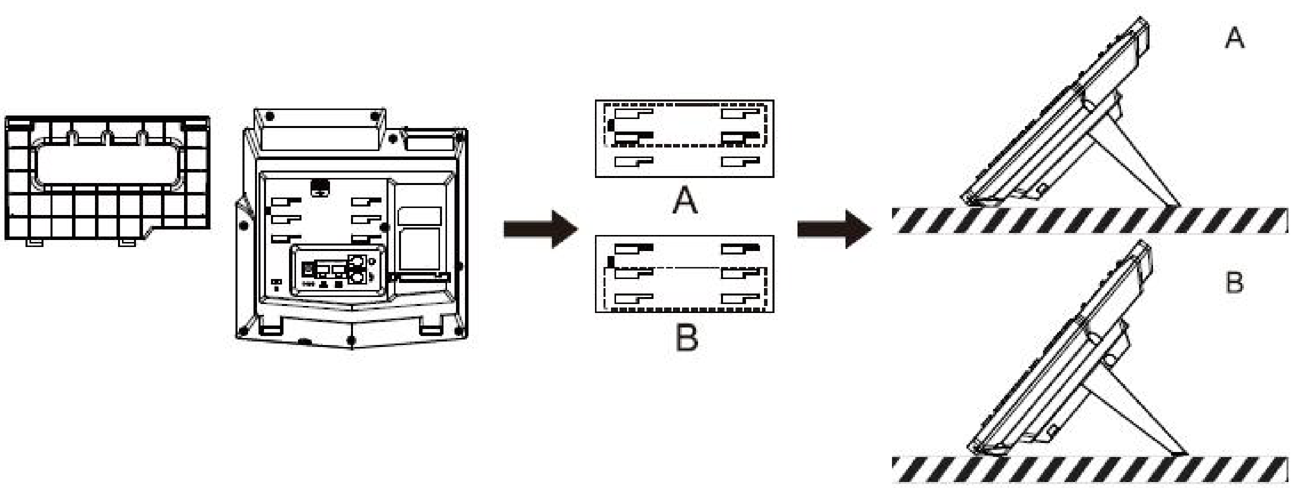
If the phone is connected to a PoE switch and a power adapter simultaneously, the power adapter has priority; if it fails, the phone switches to PoE power.

If you are not sure whether your network supports PoE, check with your administrator.

Installing IP Phone on the Desktop or Wall

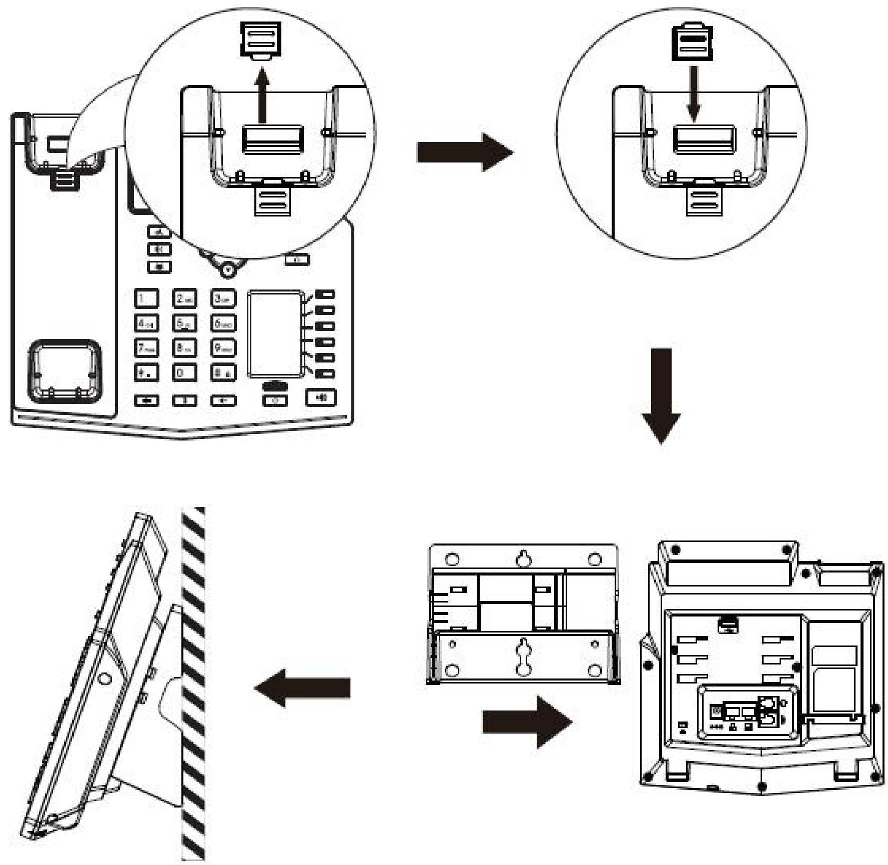
You can place the phone on the desktop or mount on the wall. The IP Phone includes an adjustable footstand. When placing the phone on a desktop surface, you can adjust the tilt height to different angles.

Figure 1 Desktop Installation



Mounting the IP Phone on the wall requires some tools and equipment that are not provided. The tools and parts required for a typical IP Phone wall installation are a hanging bracket, screwdriver, and screws to secure the phone to the wall.

Figure 2 Wall-mount Installation



Connecting the IP Phone

This section illustrates and describes the connectors on your phone.

Connect the power adapter, network, PC, handset, and headset to the appropriate ports as shown in the picture below.

Figure 3 Connecting to the Phone

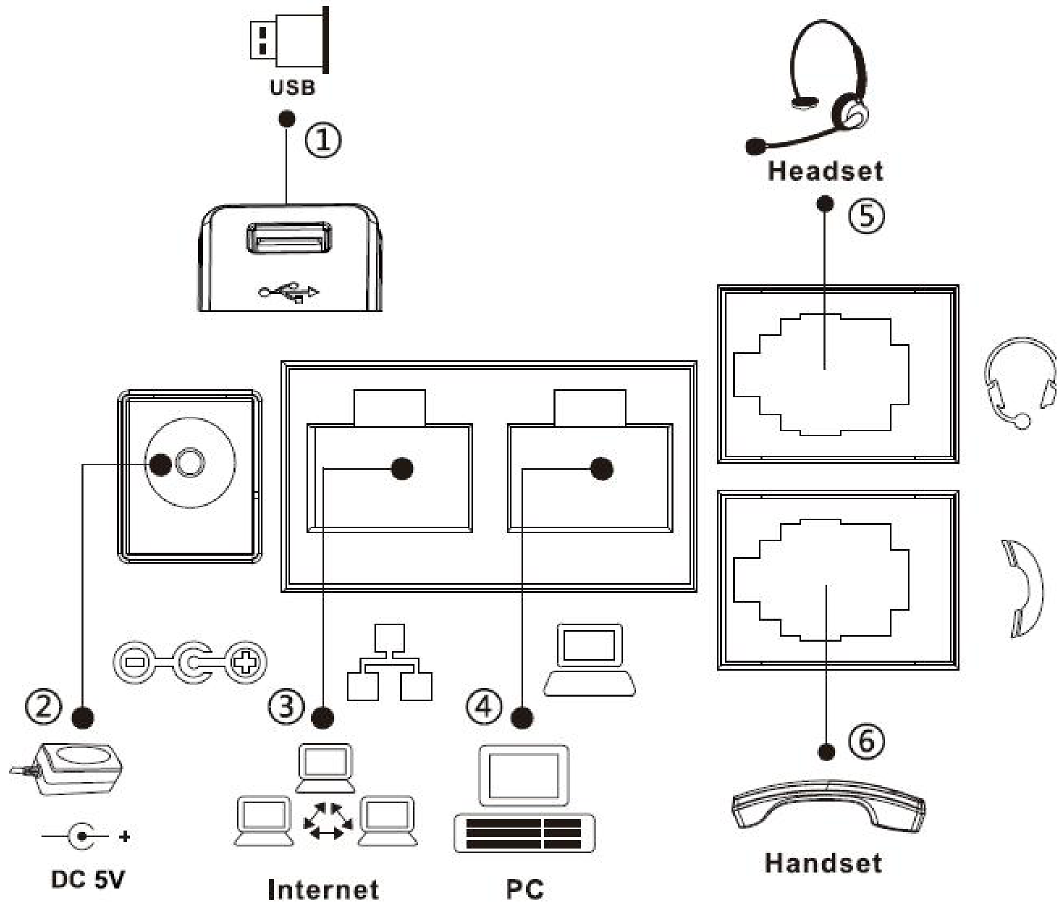
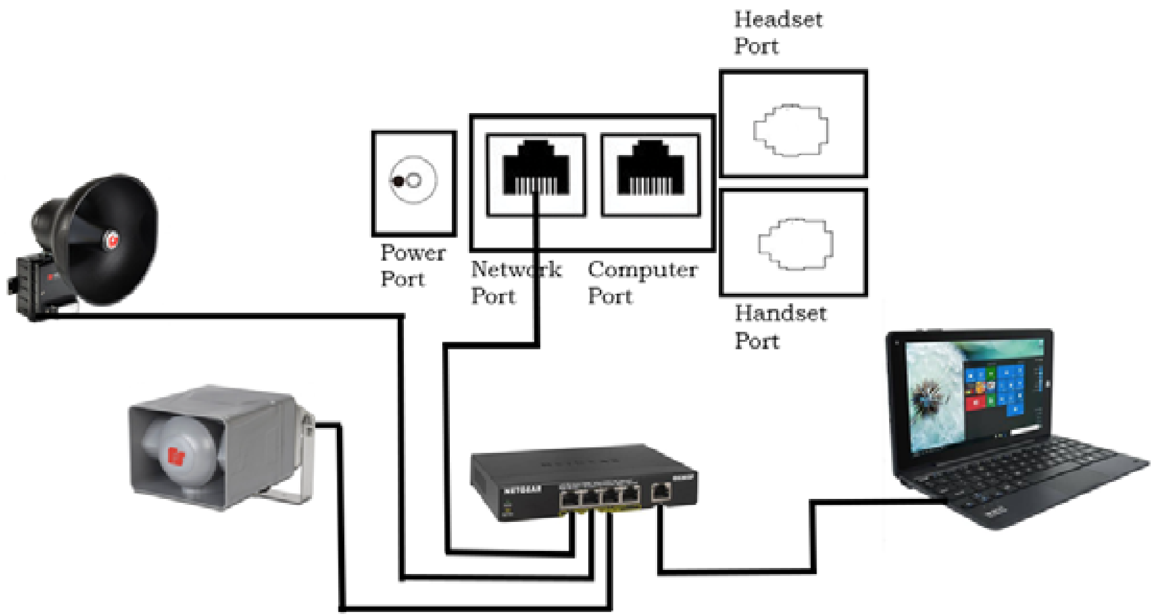


Table 4 Hardware Interface Description

No.	Interface	Description
1	USB Port	USB device connection
2	Power Port	Power adapter connection
3	Network Port	LAN or Internet connection
4	PC Port	Network port for computer connection
5	Headset Port	Headset connection
6	Handset Port	Handset connection

Figure 4 Typical IP Phone Setup



Defining Phone Buttons

Introduction of the Keypad

The following figure illustrates the front view and the keypad of the IP Phone, and the table describes the keys.

Figure 5 Keypad of IP Phone

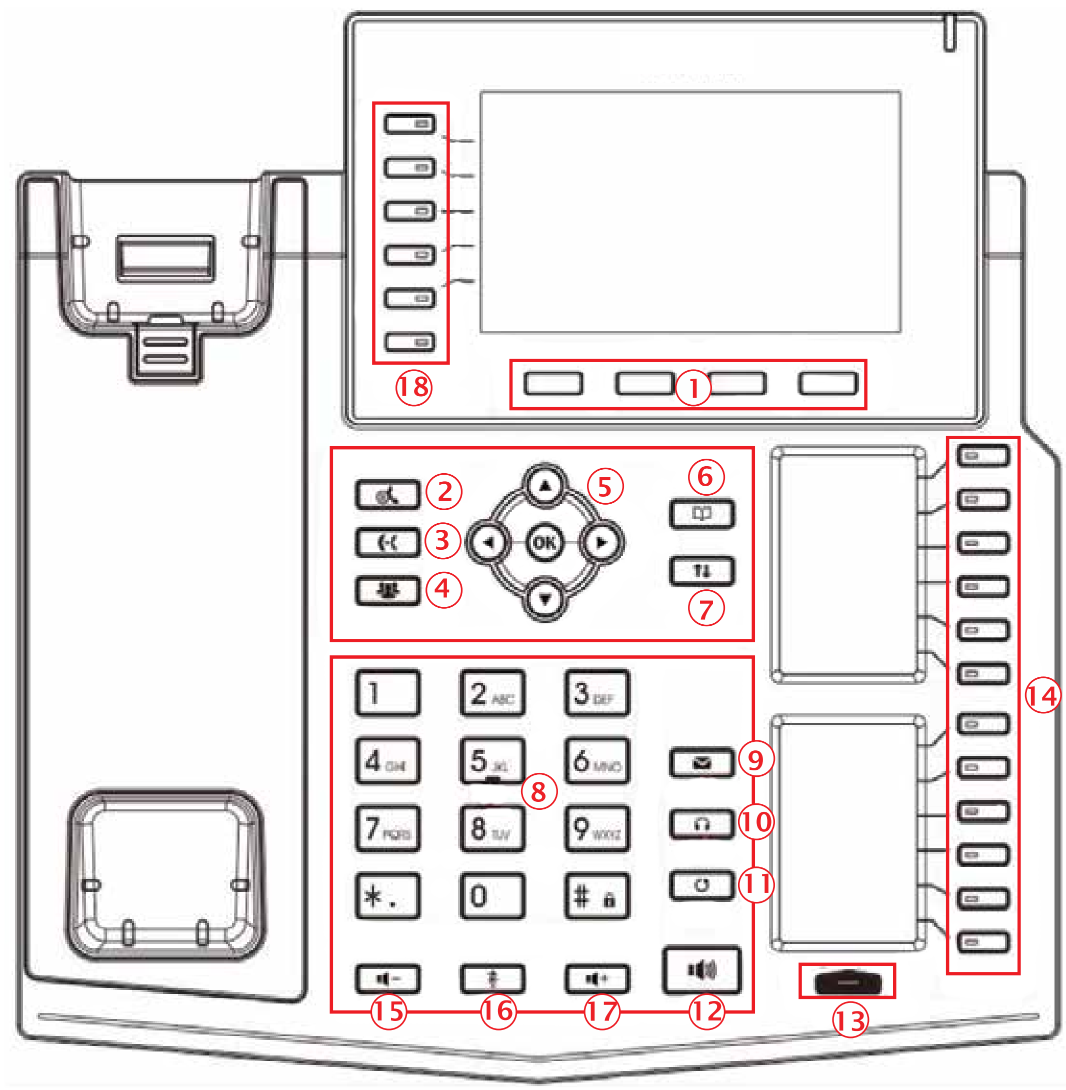
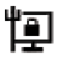
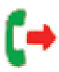
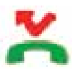
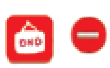
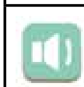
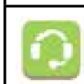
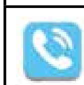
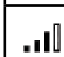


Table 5 Keypad Description

No.	Interface	Description
1	Soft-menu Buttons	These four buttons provide different functions corresponding to the soft menu displayed on the screen.
2	Hold Key	Press the Hold key during the call to hold the call, and press it again to cancel the hold.
3	Transfer Key	Press the Transfer button to transfer the current call to other numbers.

No.	Interface	Description
4	Conference	Press the Conference button to initiate a three-party conference.
5	Navigate/OK Keys	Press the up/down navigation key to change the line or move the cursor in the screen list. On some Settings and text editing pages, you can press the left/right navigation key to change options or move the cursor in the screen list to the left/right. OK key: Default is equivalent to soft button confirmation; you can customize.
6	Contact Key	Press the Contact key to enter the address book interface and select the contact person to call.
7	Call Log Key	Press the Call Log key to see the incoming/outgoing calls.
8	Standard Telephone Keys	The 12 standard telephone keys provide the same function as standard telephones; some keys also provide a special function by long pressing the key. (To long press, press and hold the button for more than 2 seconds.)
9	Voice Mail	Press the Voice Mail key to enter the interface of SMS and voice mail list.
10	Headset Key	Press the Headset key to open the headset channel
11	Redial	Press the Redial key to redial the last number dialed
12	Hands-free Key	Press the Hands-free key to open the audio channel of the speakerphone.
13	Next Page Key	Scrolls through multiple screens.
14	DSS KEY	Short press a DSS key to select a function. Long press a DSS key to display the settings interface. (To long press, press and hold the button for more than 2 seconds.)
15	Volume Down Key	In the standby state (on-hook), press this button to reduce the ring volume and, when on a call, this button lowers the volume on the call.
16	Mute Key	During a call, press this key to mute the microphone.
17	Volume Up Key	In the standby state (on-hook), press this button to increase the ring volume and, when on a call, this button increases the volume on the call.
18	Side Key	Long press the side key to enter the settings interface to set the required functions.

Table 6 Status Prompt and Notification Icons

Screen Icon	Description
> > > >	Call out
	Call in
	Call Hold
	Network Disconnected
	Open VLAN
	Open VPN
	Keypad Locked
	Call forward calls
	Outgoing calls
	Incoming calls
	Missed calls
	SMS
	New voice message waiting
	Do Not Disturb inactivated on phone
	Call forward activated
	Auto-answering activated
	Hands-free (HF) Mode
	Headphone (HP) Mode
	Handset (HS) Mode
	Mute Microphone
	The Voice quality of calling

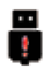
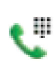
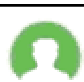
Screen Icon	Description
	The Voice encryption of calling
	Speech High Definition
	Record
	SIP Hotspot
	Bluetooth®
	Wi-Fi
	USB Insert
	USB overload

Table 7 DSSKEY Icon

DSSKEY Icon	Side key Icon	Description
		BLF/New call
		BLF/XFER
		BLF/AXFER
		BLF/Conference
		BLF/DTMF
		Presence
		Voice Message
		Speed Dial
		Intercom

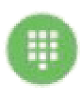
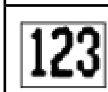
DSSKEY Icon	Side key Icon	Description
		Call Park
		Call Forward
		Keyevent
		URI
		BLF List
		MCAST Paging
		None for Memory Key
		None for DSSKEY
		Line Key
		DTMF

Table 8 Keyboard character

Mode Icon	Text Mode	Key Button	Characters of Each Press
	Numeric	1	1
		2	2
		3	3
		4	4
		5	5
		6	6
		7	7
		8	8
		9	9
		0	0
		*	* . +
		#	#

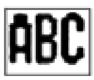
Mode Icon	**Text Mode**	**Key Button**	**Characters of Each Press**
	Lower Case Alphabets	1	@ : ; ( ) < >
		2	a b c
		3	d e f
		4	g h i
		5	j k l
		6	m n o
		7	p q r s
		8	t u v
		9	w x y z
		0	(space)
		*	. , * / + - : _ =
		#	# ^ ! & $ %
	Upper Case Alphabets	1	@ : ; ( ) < >
		2	A B C
		3	D E F
		4	G H I
		5	J K L
		6	M N O
		7	P Q R S
		8	T U V
		9	W Z Y X
		0	(space)
		*	. , * / + - : _ =
		#	# ^ ! & $ %
	Mixed type input	1	1
		2	2 a b c A B C
		3	3 d e f D E F
		4	4 g h I G H I
		5	5 j k l J K L
		6	6 m n o M N O
		7	7 p q r s P Q R S
		8	8 t u v T U V
		9	9 w z y x W Z Y X
		0	0
		*	. , * / + - : _ =
		#	# ^ ! & $ %

Table 9 DSS Key LED Definition

Type	LED Light	State
Line Key	Off	Line inactive
	Green On	Line ready (Registered)
	Green Blinking	Ringing
	Red Blinking	Line is trying to register
	Red Blinking	Line error (Registration failure)
	Red On	Dialing/Line in use (Talking)
	Yellow Blinking	Call holding
BLF	Green On	Subscription number is idle
	Red On	Subscription number is busy
	Red On	Subscription number is dialing
	Off	Subscription number is unavailable
Presence	Green On	Subscription number is idle
	Red On	Subscription number is busy
	Red On	Subscription number is dialing
	Off	Subscription number is unavailable
DND	Red On	Enable DND
	Off	Disable DND
MWI (Message Waiting Indicator)	Green Blinking	New voice message waiting
	Off	No new voice message

Basics of Using the Phone to Place a Call

Using the Handset

To talk over the handset, pick up the handset off the device and dial the number, or dial the number first, then pick up the handset, and the number is dialed. You can switch the audio channel to the handset by lifting the handset when the audio channel is turned on in the speaker or headphones.

Using the Hands-free Speaker

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

Using the Headphone

To use headphones, use the headset button defined by the DSS keys to turn on the headphones. Same as the handset and hands-free speaker, you can dial the number before or after the headphone is turned on.

Using Line Keys (Defined by DSS Key)

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

Standby Screen (On-Hook)

The following is a picture of the default standby screen (on-hook).

Figure 6 Phone Home Screen

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

The home screen's lower half is the function menu keys, which are the first layer of function menu keys through which you can operate the phone.

You can restore the phone to the default standby screen interface by lifting the handset and hanging up. You can customize the side keys. See "Table 5 Keypad Description" on page 22.

For some screens, many items or long text may not fit. These are arranged in a list or as multiple lines with a scroll bar. If you see a scroll bar, use the up/down navigator buttons to scroll the list. By long pressed the navigator keys, you can scroll the list or items faster.

Phone Status

You can view the phone status through the phone interface and the web interface.

NOTICE

NETWORK INSTALLATION PRECAUTIONS: Review network settings with your Network Administrator before connecting to an existing network.

Phone Interface

To view through the phone status through the phone interface:

1. Press the Menu Soft-menu button. The Menu screen appears.
2. Select the Status icon. Press the OK button. The Network screen appears.

   NOTE: Here you can view the IP address of the phone.

Web Interface

To view the phone status through the web interface:

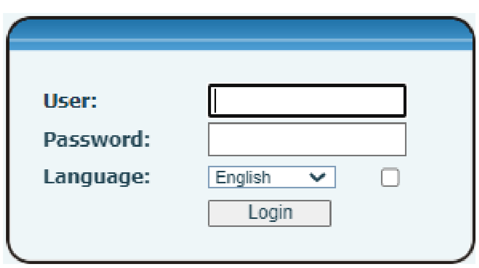
1. Change your Local Area Connection (Ethernet) adapter address to 10.10.10.“*your number*”. As shown below:

   ```
   IP Address: 10.10.10.1
   Subnet Mask: 255.255.0.0
   Default Gateway: 10.10.10.1
   ```

   When connecting directly to the phone, your computer’s IP Address and Default Gateway must match. Changing your IP Address, Subnet Mask, and Gateway allows you to access the IP Phone at its default IP address (10.10.10.100).

2. Enter the default 10.10.10.100 IP address or the preconfigured static address for the IP Phone into your web browser to view the web page of the device.

3. The login window appears.



4. Enter the Username:

   `admin`

5. Enter the Password:

   `fedsig`

   NOTE: The password is case sensitive.

6. Click Login.

   The System > Information screen appears.

The phone status includes the following information about the phone: System Information, Network, VQ status, and SIP Accounts.

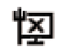
Network Configurations

The IP Phone relies on an IP network connection to provide service. IP devices are connected to each other over the network and exchange data in packets based on the devices' IP addresses.

To enable this phone, configure the network configuration.

To configure the network, press Menu > Advanced Settings > Network > Network.

NOTE: You need a password to use Advanced Settings.

IMPORTANT: if you see the WAN Disconnected- icon  flashing in the middle of the screen, it means the network cable was not correctly connected to the device's network port. Check if the cable is connected correctly to the device and to the network switch, router, or modem.

The device supports three types of networks: IPv4, IPv6, IPv4 and IPv6.

IPv4

There are three common IP configuration modes for IPv4:

- Dynamic Host Configuration Protocol (DHCP): This is the automatic configuration mode. The system gets the network configurations from a DHCP server and applies it to the device. Recommended for the most users.
- Static IP Configuration: This option allows you to configure each IP parameter manually, including IP Address, Subnet Mask, Default Gateway, and DNS servers. Usually used in a technical environment of network users.
- PPPoE: This option is often used by users who connect the device to a broadband modem or router. To establish a PPPoE connection, username and password is provided by the service provider.

The default configuration for the phone is in DHCP mode.

IPv6

There are two common IP configuration modes for IPv6:

- Dynamic Host Configuration Protocol (DHCP): This is the automatic configuration mode. The system gets the network configurations from a DHCP server and applies them to the device. Recommended for most users.
- Static IP configuration: This option allows users to manually configure each IP parameter, including IP address, mask, gateway, and primary and secondary domains. This usually applies to some professional network user environments.

See "Network Settings" on page 86 for information on how to configure.

SIP Configurations

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

You can conduct line configuration on the phone or web interface, and input the corresponding information at the registered address, registered user name, registered password, and SIP user and registered port, respectively, which the SIP server administrator provides.

Phone Interface

To configure the SIP Line through the phone interface:

- Long press a Side Key. The DssKey screen appears for that SIP line selected.

  NOTE: To long press, press and hold the button for more than 2 seconds.

- Press Menu > Advanced Settings > Accounts > Line *n*.

  n = the number of the SIP line.

NOTE: You need a password to use Advanced Settings.

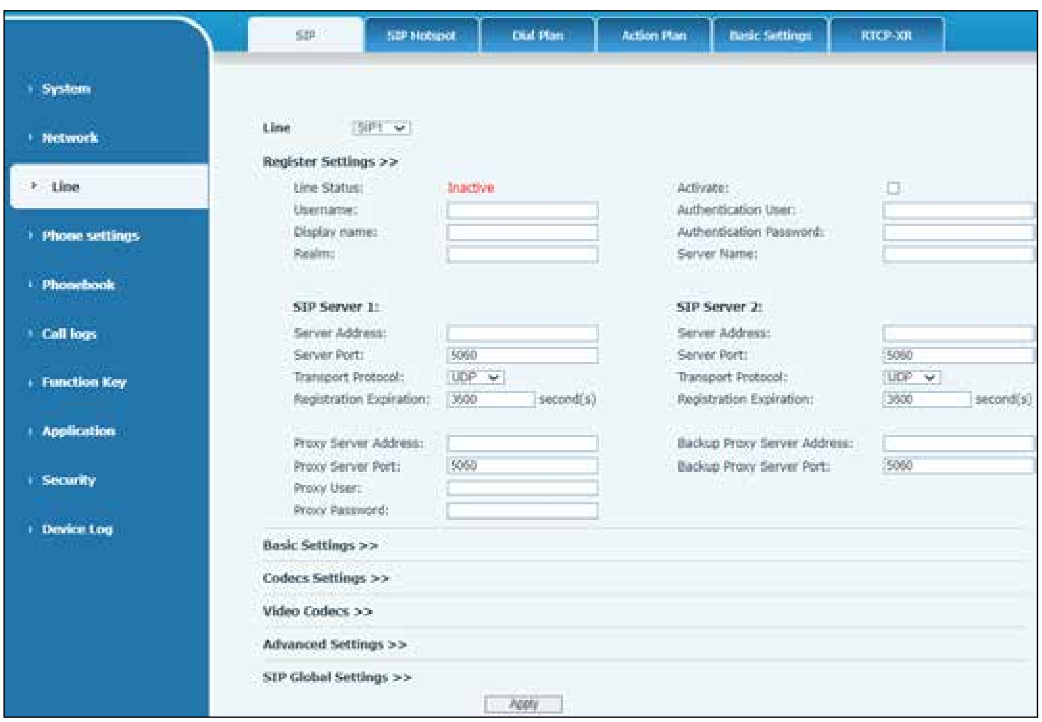
Web Interface

To configure the SIP Line through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Line > SIP screen. The Line SIP screen appears.



3. Enter Username, Display name, Realm, Server Address.
4. Click Apply.

Basic Functions

Making Phone Calls

Your phone works just like a regular phone.

Default Line

The phone has twenty line services. If both lines are configured, you can make or receive phone calls on either line. If you configure a default line, then there is a default line for making outgoing call which is indicated on the top left corner of the phone screen.

To change the default line, use the left and right navigator buttons to switch between lines.

Phone Interface

To configure the default line through the phone interface:

1. Select Menu > Features > General > Default Line.
2. Select Enabled or Disable by using the navigator buttons.

Dialing Methods

You can dial a phone number by the following options:

- Enter the number directly using the keypad
- Select a phone number from phonebook contacts (See “Local Contact” on page 79.)
- Select a phone number from Cloud phonebook contacts (See “Cloud Contacts” on page 81.)
- Select a phone number from call logs (See “Call Log” on page 82.)
- Redial the last most recently dialed phone number

Dialing the Number and then Opening Audio

To make a phone call:

Dial the phone number and then do one of the following.

- Press the Dial soft-menu button
- Press the hand-free button  to turn on the speaker or headphones
- Lift the handset to call out with the current line
- Press the line key (Configured by DSS Keys) to call out with a specified line

Opening Audio and then Dialing the Number

To make a phone call:

1. Lift the handset.
2. Press the hand-free button  to turn on the speaker or headphones.
3. Dial the number.
4. Press the Dial or OK button to make a call, or the number can be dialed out automatically after timeout.

Canceling a Call

To cancel a call while calling the number, hang up or press the hands-free button .

Answering Phone Calls

When your phone rings, the Call in icon displays .

To answer the phone call:

- Press the flashing line button to answer the call.
- Lift the handset, press the hands-free button , or the Answer button using the soft-menu buttons. You can also use with a standard headset.

To divert the incoming call, press the Divert soft-menu button.

To reject the incoming call, press the Reject soft-menu button.

Talking Mode Screen

When the call is connected, you will see a talking mode screen.

Table 10 Talking Mode

Number	Name	Description
1	Default line	The line currently used by the phone.
2	Voice channel	The icon shows the voice channel mode being used.
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	Numbers of line	Shows how many calls are present on the current device
6	Speech quality	Displays the current voice quality of the call.
7	HD audio	Call using G.722 voice coding calls when displayed HD voice icon.

Making and Receiving a Second Call

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

Second Incoming Call

When there is another incoming call during an active call, you will see the call message in the middle of the current screen. You know a call is waiting when you hear the call waiting tone and see the line button flashing green. You may accept or reject the call the same as a regular incoming call. When call waiting is answered, the first call is placed on hold.

Second Outgoing Call

To make a second call:

1. Press the Xfer or Conference soft-menu button or the line key to make a new call on that line.
2. Dial your second number.

Another way to make a second call:

- Press the DSS keys, or
- Dial out from the configured keys (BLF/Speed Dial)

When you are making a second call, you can hold the first call manually or it will be put on hold automatically.

Switching between Two Calls

When there are two calls established, you will see both calls on the screen.

To switch between two calls:

1. Press the arrow keys to select the number.
2. Press the Resume button.

Ending One Call

To end the current call, hang up or press the End soft menu button. The phone returns to the single call mode in a holding state.

Ending a Call

To end a call, hang up or press the End soft menu button.

NOTE: When the phone is in the reserved state, press the Resume button to return to the call state, or put the receiver back and press the hands-free button to end the call.

Redial

You can call the most recently dialed phone number. To redial the last outgoing phone number, press the redial button .

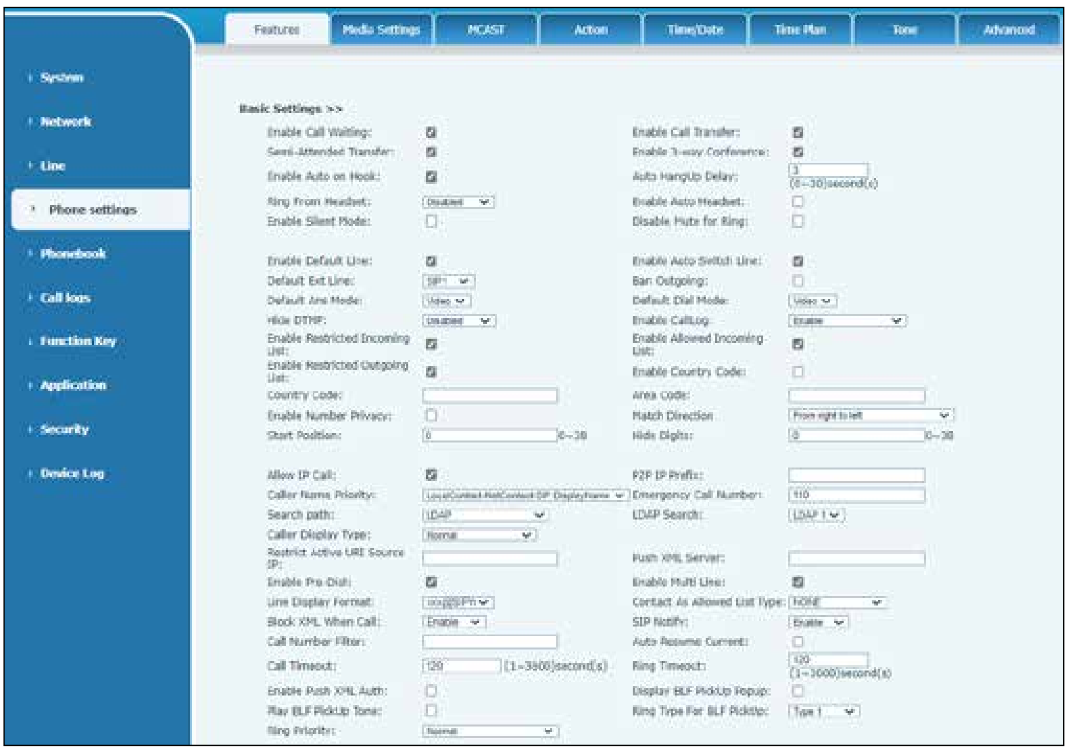
Web Interface

To redial the last outgoing phone number through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Phone settings > Features. The Phone settings Features screen appears.



3. Select the Redial Settings.

4. Click the Redial Enter CallLog check box.
5. Click Apply.

Dial-up Query

The dial-up inquiry is on by default. The dial interface automatically matches call records contacts in the number list. Use the navigate arrow keys to select the number and press the call-out key or wait for a time out.

Auto-Answering

The auto-answering mode allows calls to be automatically answered (not including call waiting). The auto-answering can be enabled on line basis.

Phone Interface

To turn on the auto-answering mode through the phone interface:

1. Press Menu > Features > Auto Answer.
2. Select the SIP line and press the OK button.
3. Use the navigate arrow keys to Disable or Enable the Auto Answer.
4. Use the telephone keys to set the Auto Answer Delay. The Auto Answer Delay is set to 5 seconds by default.
5. Press OK to save. The icon in the upper right corner of the screen indicates that auto answer is enabled.

Web Interface

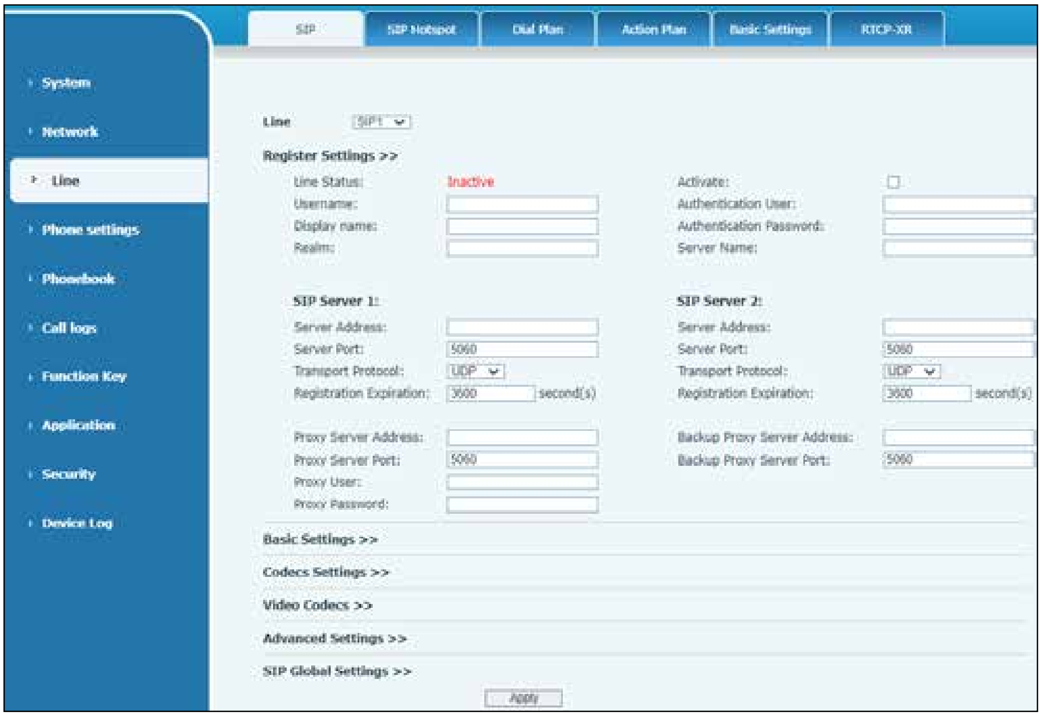
To turn on the auto-answering mode through the web interface:

1. Login to the web interface.

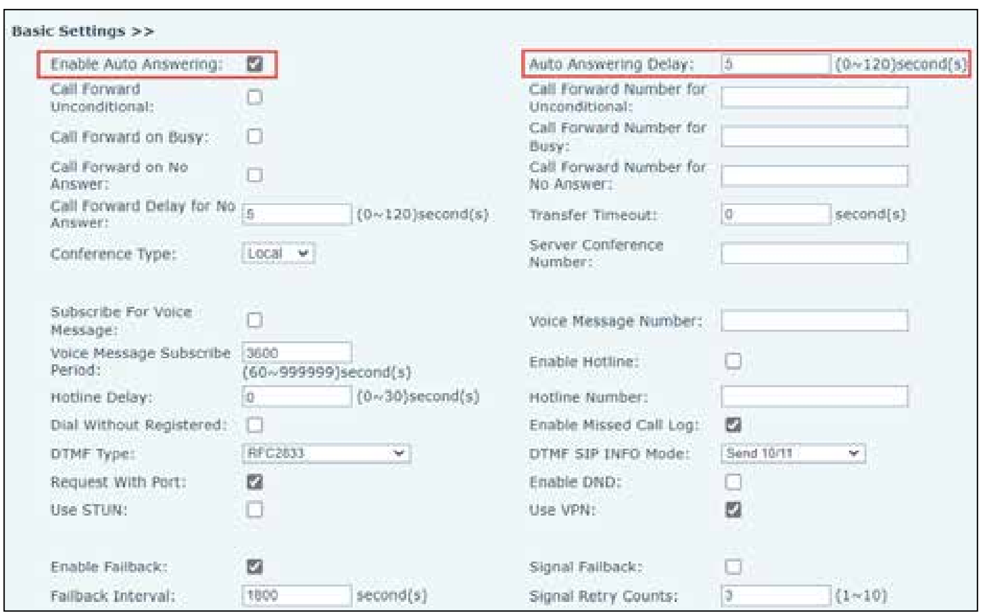
   The System > Information screen appears.

2. Select Line > SIP screen. The Line SIP screen appears.

3. Select the SIP Line.



4. Select Basic Settings. The Basic Settings screen appears.



5. Click the Enable Auto Answering. Set the to Answering Delay to 5 seconds.

6. Click Apply to save.

Callback

The callback option allows you to dial the phone number of the last outgoing call. If there is no call history, the phone displays can't process.

Phone Interface

To turn on callback option through the phone interface:

1. Press Menu > Basic > Keyboard > DSS Key Settings.
2. Use the arrow keys to select the Type as Key Event.
3. Use the arrow keys to select the Key as Call Back.
4. Select OK to save.

Web Interface

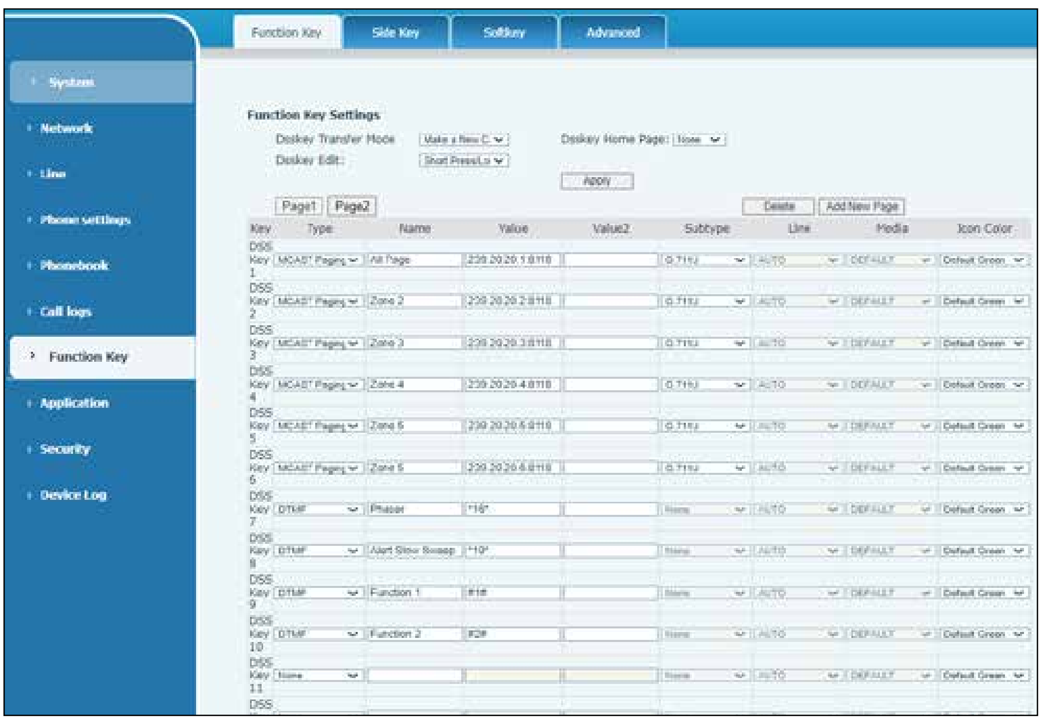
To configure the callback option through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Function Key > Function Key screen. The Function Key Settings screen appears.

   NOTE: You can also use the Function Key > Side Key screen.



3. Select the function key. Click the Type arrow and select Key Event.
4. Click the Subtype arrow and select Call Back.
5. Click Apply to save.

Mute Your Call

While on a call, you can mute the audio, so that you can hear the other person, but he cannot hear you. You can use the mute in all call modes (handles, headphones, or hands-free).

To mute the audio, press the press the mute button . The mute icon appears on the phone display. Press the mute button again to turn mute off.

To mute the ringtone on an incoming call, press the mute button . The mute icon appears on the phone display. Press the mute button again to turn mute off.

Hold Your Call

You can put an active call on hold and then resume the call when you are ready.

To hold your call, press the press the hold button . The hold button because the resume button. Press the resume button to restore the call.

Turn On Do Not Disturb (DND)

The Do Not Disturb (DND) soft menu button reject incoming calls (including call waiting). It affects all lines on your phone. When you turn on DND, your incoming calls are forwarded to another number, such as your voicemail, if it is set up. However, you will always receive intercom and emergency calls, even when DND is turned on. All calls you receive while DND is enabled are logged in your Recent Calls list.

Phone Interface

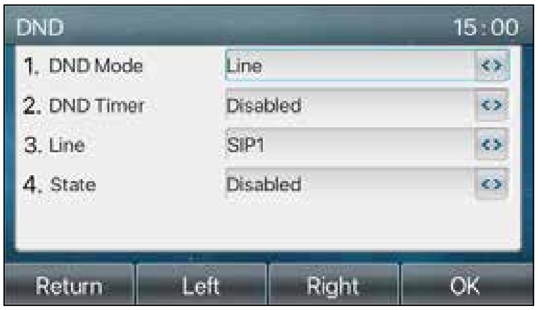
To turn on the DND feature through the phone interface:

1. Press the DND long soft menu button. The Do Not Disturb icons appear on the phone display.
2. Press the DND long soft menu button again to turn the DND feature off.

Configuring the DND on a Specific Line

To configure the DND feature on a specific line:

1. Press Menu > Features > DND option.
2. Use the left and right navigation buttons to select the line to adjust the DND Mode. The DND setting interface appears.

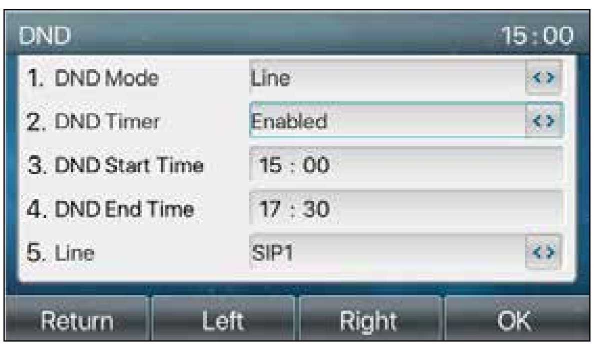


3. Make changes
4. Press OK to save.

Configuring the DND Timer

To configure the DND timer.

1. From the DND screen, select the DND Timer option.
2. Use the arrow keys to select Enable.



3. Set your start and end times.
4. Press OK to save. You will see the DND icon turn red, and the DND mode for the selected sip line is turned on.

Web Interface

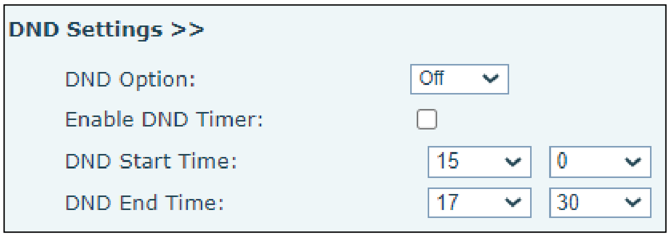
To configure the DND option through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Phone settings > Features screen. The Features screen appears.

3. Select the DND Settings.

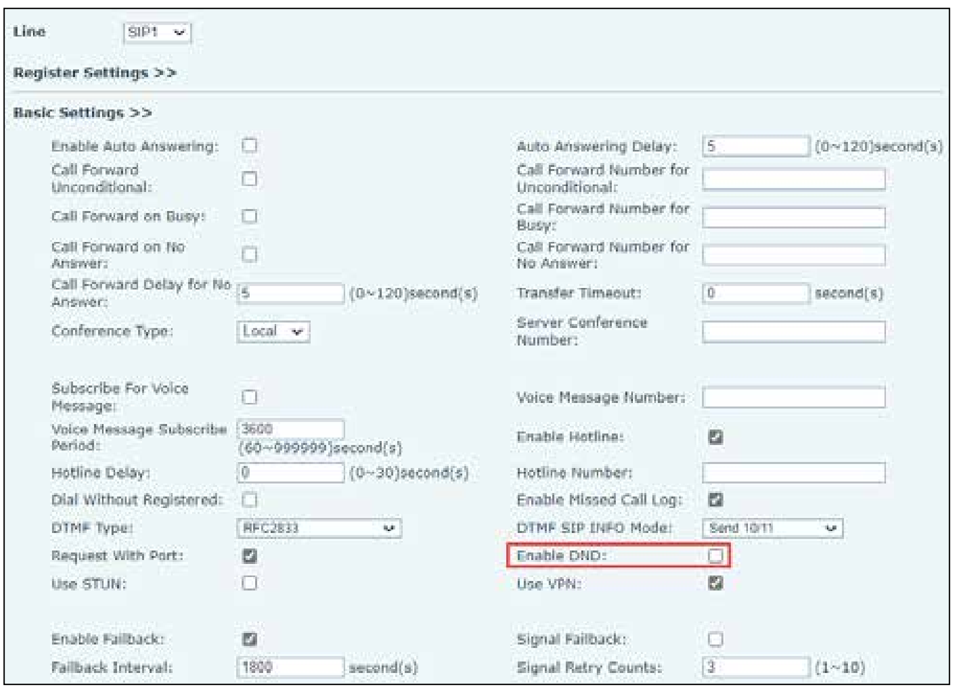


4. Under the DND option, click the down arrow to select either Off, Phone, or Line.
5. Set your start and end times.
6. Click Apply to save.

Configuring the DNS for a Specific Route

To configure the DND for a specific route:

1. Select Line > SIP. The SIP screen appears.
2. Click the Line down arrow to select a SIP Line.
3. Select Basic Settings. The Basic Settings screen appears.



4. Select the Enable DND check box.
5. Click Apply to save.

Call Forward

You can forward calls from any line on your phone to another number for each SIP line. You can configure the call forward settings of each line.

There are three types of call forward:

- Unconditional Call Forward: Forward any incoming call to the configured number.
- Call Forward on Busy: When the phone is busy, the incoming call is forwarded to the configured number.
- Call Forward on No Answer: When there is no answer, the incoming call is forwarded to the configured number.

Phone Interface

To configure Call Forward through the phone interface:

1. Press Menu > Features > Call Forward option.
2. Select the SIP line to configure. Press OK.
3. Select the Call Forward type.
4. Configure call forwarding.

5. Use the arrow keys to select Enabled.
6. Enter the call forward target number exactly as you would dial it from your phone, or select an entry from your list of recent calls.
7. Press OK to save.

Web Interface

To configure Call Forward through the web interface:

1. Login to the web interface.

   The System > Information screen appears.
2. Select Line > SIP. The SIP screen appears.
3. Click the Line down arrow to select a SIP Line.

4. Select Basic Settings. The Basic Settings screen appears.

5. Set the type and enter the call forward target number exactly as you would dial it from your phone.
6. Click Apply to save.

Call Transfer

You Can Transfer an active call to another person.

There are three ways to Call Transfer.

- Blind transfer: Transfer the call with out talking to the person that you're transferring the call to.
- 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 other party.

NOTE: For more transfer settings, see "Line > Dial Plan" on page 113.

Blind Transfer

To Blind Transfer your call:

1. During your call, press the transfer button .
2. Enter the number to transfer or press the contact button or the history button to select the number.
3. Press the transfer button again. After the third party rings, the phone shows that the transfer is successful.
4. Hang up.

Semi-Attended Transfer

To Semi-Attend Transfer your call:

1. During your call, press the transfer button .
2. Enter the number to transfer or press the contact button or the history button to select the number.
3. Press the transfer button again. Before the third party answers, press the transfer on the call interface to make the semi-attendance transfer or press the end button to cancel the semi-attendance transfer.

4. Hang up.

Attended Transfer

To Attend Transfer your call:

1. During your call, press the transfer button .
2. Enter the number to transfer or press the contact button or the history button to select the number.
3. Press the transfer button again. Wait for the third party to answer.
4. Press the transfer button to transfer the first call to the second call.
5. Hang up.

Call Waiting

When you are on an active call, you know that a call is waiting when you hear a single beep and see the line button flash.

- Enable call waiting: new calls can be accepted during a call.
- Disable call waiting: new calls are automatically rejected, and a busy tone is prompted.
- Enable call waiting tone: when you receive a new call on the line, a tone beeps.

You can enable or disable the Call Waiting function through the phone interface or the web interface.

Phone Interface

To configure Call Waiting through the phone interface:

1. Press Menu > Features Call Waiting.
2. Select the Call Waiting Settings.

3. Use the arrow keys to select Enabled.
4. Press OK to save.

Web Interface

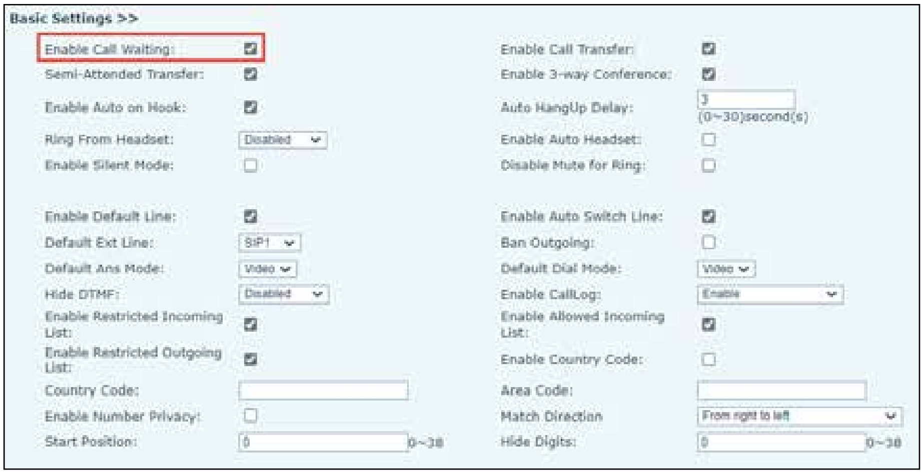
To configure Call Waiting through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Phone settings > Features screen. The Features screen appears.

3. Select the Enable Call Waiting check box.



4. Under Tone Settings, select the Enable Call Waiting Tone check box.

5. Click Apply to save.

Conference Calls

You can talk with several people in a single call. You can dial another person and add them to the call. If you have multiple phone lines, you can join two calls across two lines.

When you add more than one person to a conference call, wait a few seconds between adding participants.

As the conference host, you can remove individual participants from the conference. The conference ends when all participants hang up.

The Conference Type is set to Local by default.

Phone Interface for Local Conference Settings

To conference in another person to a call:

1. During your call, press the conference button.

2. Enter the phone number and press the conference button.

There are two ways to create a local conference:

The phone has two channels of communication.

1. Press the conference button .
2. Enter the conference number.
3. Select New Call and the other number that already exists.
4. If the device has a call all the way, press the soft-menu conference key.
5. Enter the number to join the meeting and press the call.
6. After the call is answered, press the conference button again to set up the local conference between three calls.

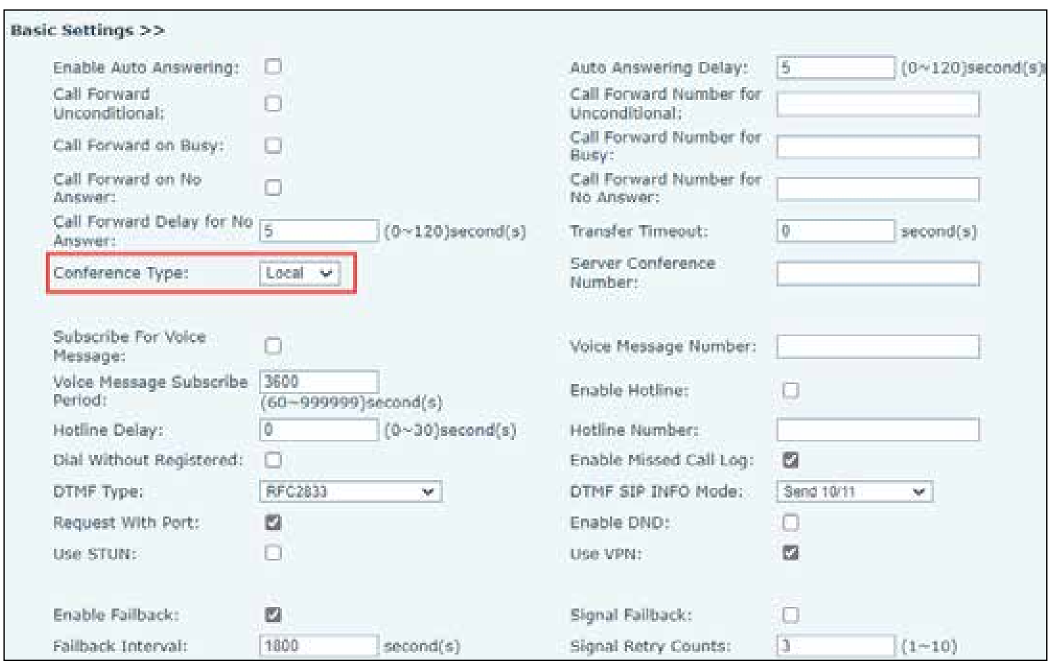
NOTE: During the conference call, press the split button to split the conference and press the end button to end the call.

Web Interface for Local Conference Settings

To conference in another person to a call through the web interface:

1. Login to the web interface.

   The System > Information screen appears.
2. Select Line > SIP. The SIP screen appears.
3. Select Basic Settings. The Basic Settings screen appears.



4. Select Local.

Web Interface for Network Conference Settings

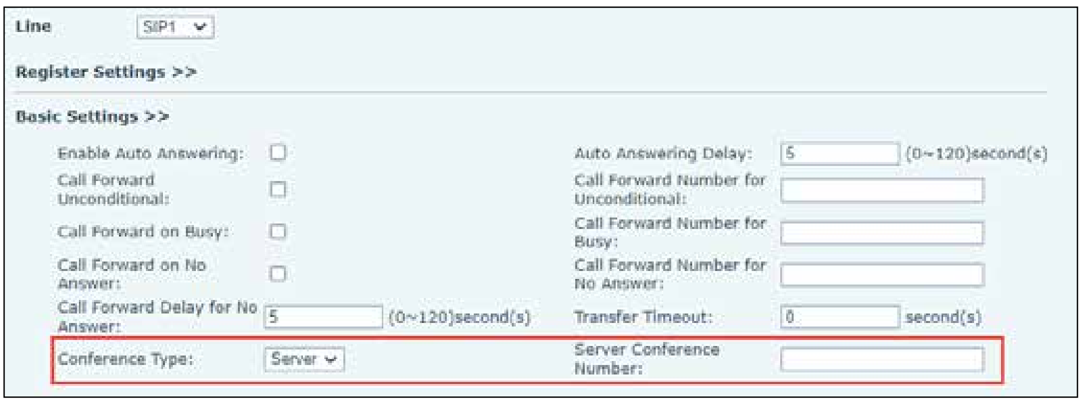
You need server support for network conference.

To conference in another person to a call through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Line > SIP. The SIP screen appears.
3. Select Basic Settings. The Basic Settings screen appears.
4. Under Conference Type, click the arrow and select Server.
5. Under Server Conference Number, enter the server conference room number. (Consult your System Administrator.)



Method to join a network conference:

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

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

Call Park

The Call Park feature allows you to put a call on hold on one phone and continue the conversation from another phone.

Call Park requires server support. Consult your System Administrator for support. You must configure the park button.

To put a call on hold and continue the conversation from another phone:

1. During your call, press the configured park button to hold the call.
2. Go to another phone to resume the call by pressing the configured park button.

Phone Interface

To set the Call Park button through the phone interface:

1. Long press a side key or press Menu > Basic > Keyboard > DSS Key Settings.
2. Set the Type as Memory Key.
3. Set the Subtype to Call Park.
4. Select OK to save.

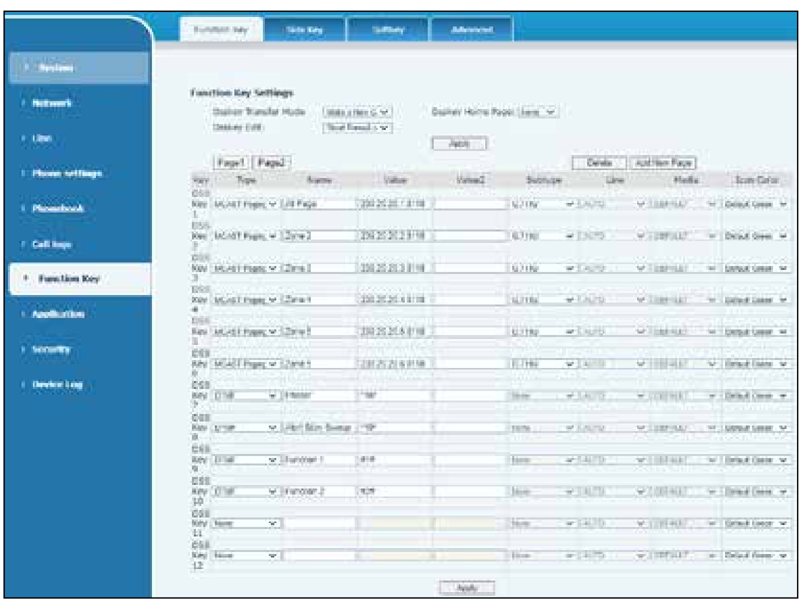
Web Interface

To set the Call Park button through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Function Key > Function Key. The Features screen appears.



3. Select a DSS Key. Set the function key type as a Memory Key, the Subtype as Call Park, and the value as the call park number of the server, and set the corresponding SIP line.
4. Click Apply to save.

Answer Someone Else's Phone (Pick Up)

The Pick Up feature allows you to answer someone else's phone call by configuring the DSS Key for BLF and setting the Pick Up code.

The Pick Up feature requires server support. Consult your System Administrator for support.

Phone Interface

To configure the Pick Up feature through the phone interface:

1. Press Menu > Basic > Keyboard > DSS Key Settings.
2. Set the line, function key type as Memory Key, subtype as BLF/NEW CALL, set subscription number, and pick up code.
3. Other phones call the subscription number, and the opposite end is in the incoming ring.
4. Press the DSS key to pick up the phone.
5. The caller picks up the call.

Web Interface

To set the Pick Up feature through the web interface:

1. Login to the web interface.

   The System > Information screen appears.
2. Select Function Key > Function Key. The Features screen appears.

3. Select a DSS Key. Set the function key type as Memory Key and the subtype as BLF/NEW CALL. In the Value box, type the subscription number and pick up code.
4. Click Apply to save.

Hide Caller ID (Anonymous Call)

You can set up the phone to hide the call number and name. Your outgoing name and number will display anonymous.

Phone Interface

To configure the Anonymous Call feature through the phone interface:

1. Press Menu > Advanced.
2. Type the password. (The password is fedsig.)
3. Select Accounts > Advanced.
4. The default is none, which is off, and RFC3323 and RFC3325 are optional. Select any one to open the anonymous call.

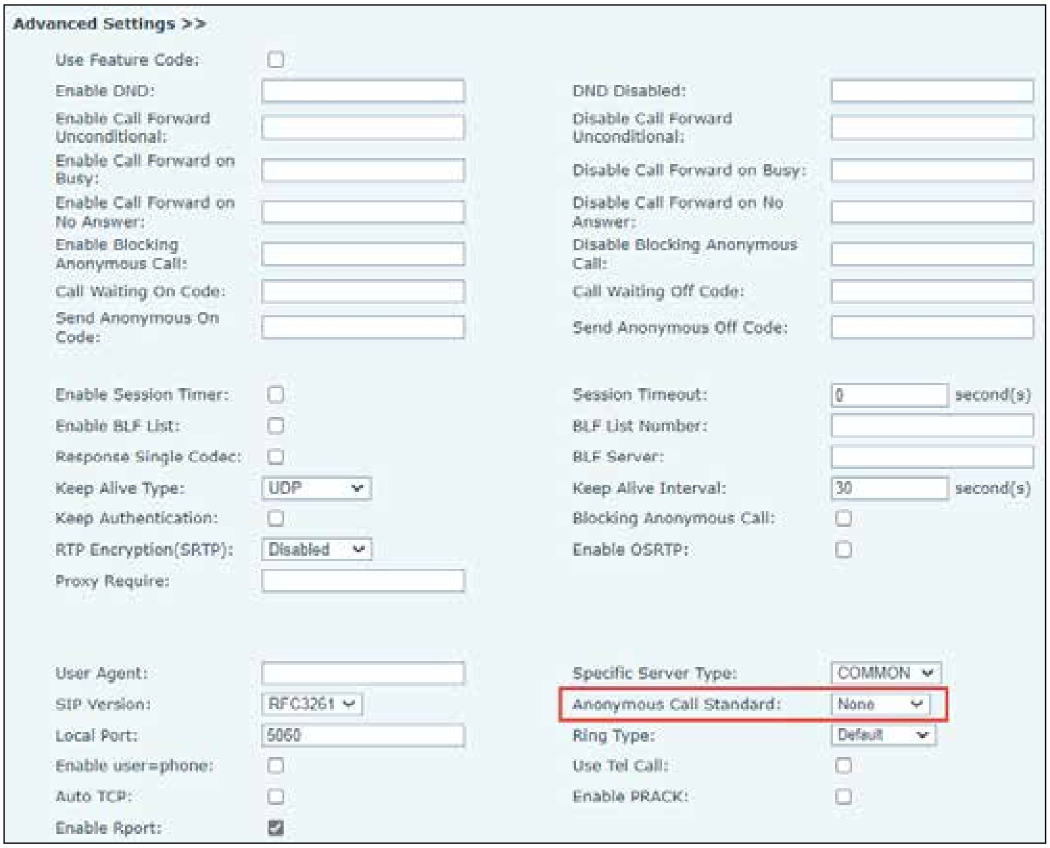
Web Interface

To configure the Anonymous Call feature through the web interface:

1. Login to the web interface.

   The System > Information screen appears.
2. Select Line > SIP. The SIP screen appears.
3. Click the Line arrow to select the SIP line. These changes only take effect on the selected SIP line.
4. Select Advanced Settings. The Advanced Settings screen appears.

5. Click the Anonymous Call Standard to select none, RFC3323, or RFC3325.



6. Click Apply to save.

Block Anonymous Calls (Ban Anonymous Calls)

You can configure the phone to block anonymous calls.

Phone Interface

To Ban (block) Anonymous Calls through the phone interface:

1. Press Menu > Features > Ban Anonymous Call.
2. Use the arrow keys to select the SIP line.
3. Use the arrow keys to change State to Enabled.
4. Select OK.

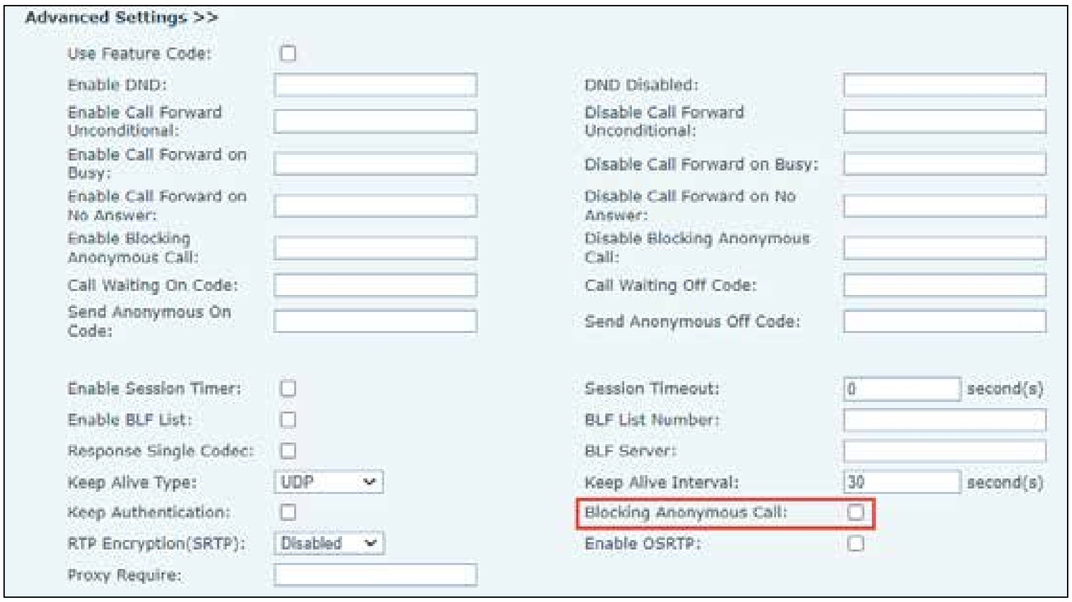
Web Interface

To Ban (block) Anonymous Calls through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Line > SIP Key. The SIP screen appears.
3. Click the Line arrow to select the SIP line. These changes only take effect on the selected SIP line.
4. Select Advanced Settings. The Advanced Settings screen appears.



5. Select the Blocking Anonymous Call check box.
6. Click Apply to save.

Hotline

The phone supports Hotline dialing. After setting up Hotline dialing, the phone automatically calls a designated number when you lift the receiver.

Phone Interface

To configure the phone with Hotline dialing through the phone interface:

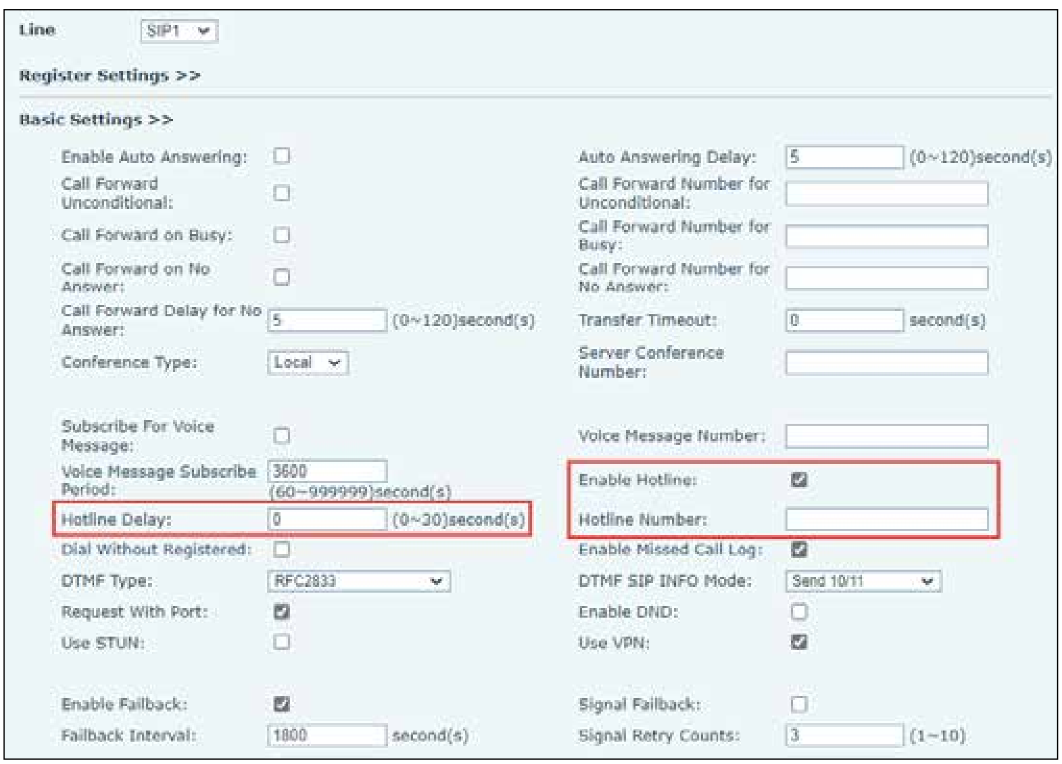
1. Press Menu > Features > Advanced > Hot Line.
2. Select a SIP line.
3. Use the arrow keys to select Enabled.
4. Enter the Hotline.
5. Select OK.

Web Interface

To configure the phone with Hotline dialing through the web interface:

1. Login to the web interface.

   The System > Information screen appears.
2. Select Line > SIP Key. The SIP screen appears.
3. Click the Line arrow to select the SIP line. These changes only take effect on the selected SIP line.
4. Select Basic Settings. The Basic Settings screen appears.



5. Set the Hotline Delay.
6. Check the Enable Hotline check box.

7. Type in the Hotline Number.
8. Click Apply to save.

Emergency Call

You can place an Emergency Call without unlocking the phone. You can configure the designated emergency number.

Phone Interface

To configure the Emergency Call number through the phone interface:

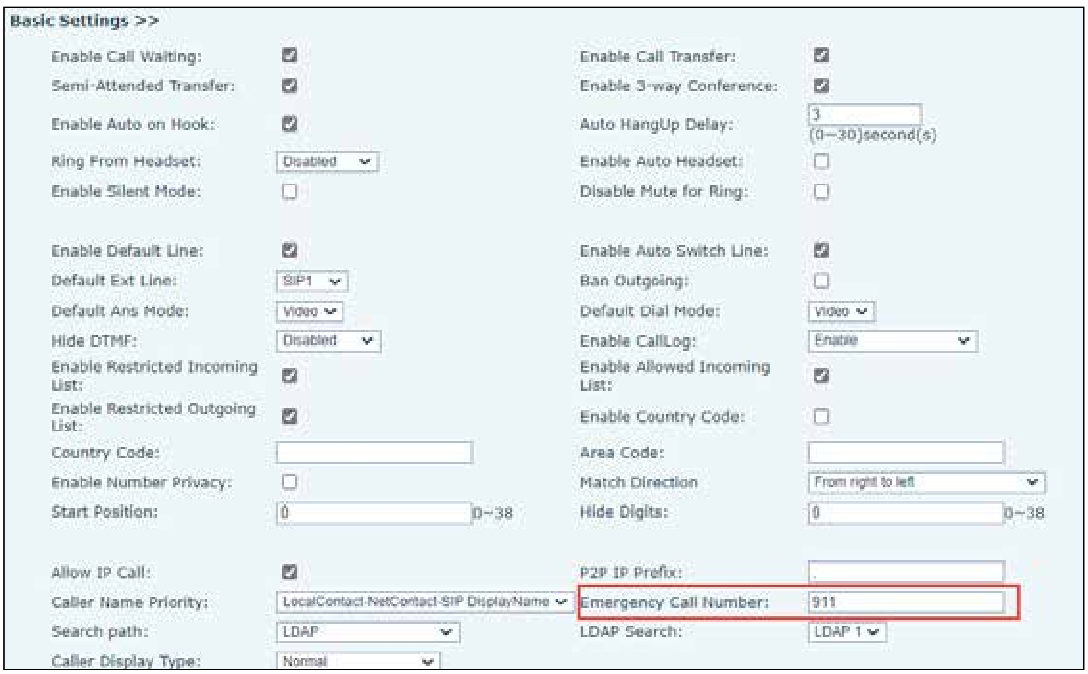
Web Interface

To configure the designated emergency number through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Phone settings > Features. The Features screen appears.
3. Select Basic Settings.



4. In the Emergency Call Number box, type in an emergency number.
5. Click Apply to save.

Advance Functions

Busy Lamp Field (BLF)

BLF (Busy Lamp Field) shows the status of other users' phones (busy, ringing, available) connected to your phone system. You need to have this user on one of your BLF list.

Phone Interface

To configure the BLF through the phone interface:

1. Long press a side key or press Menu > Basic > Keyboard > Soft DSS Key Settings. Select Soft function key to set the settings interface.
2. Set the Type as Memory Key.
3. Set the Subtype to one of the following: BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, BLF/DTMF.
4. The values are the subscription number and set up corresponding SIP lines.
5. Select OK to save.

Table 11 BLF Function Key Subtype Parameter List

Subtype	Standby Description	Calling Description
BLF/NEW CALL	Pressing the BLF key while on 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 on 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 on standby to dial the subscriber number.	When you press this BLF key while talking to another user, you attend transfer the call to the subscribed number.
BLF/Conference	Pressing the BLF key while on 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 on 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.

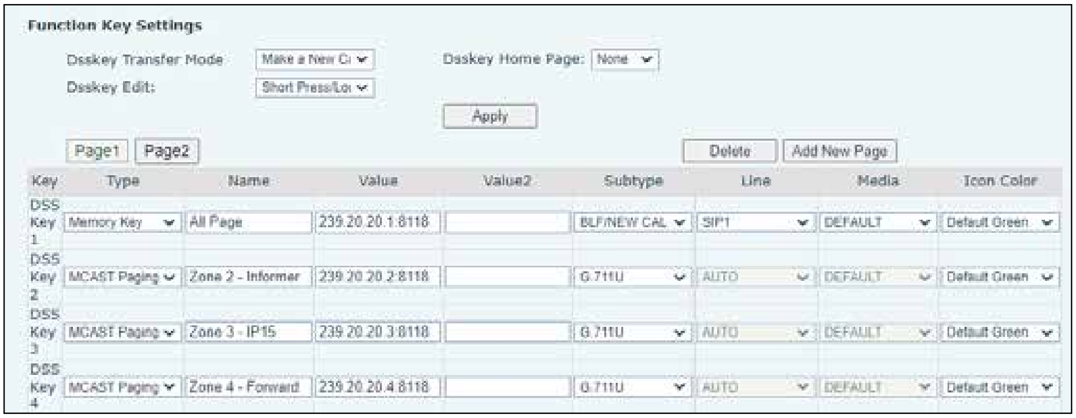
Web Interface

To configure the BLF through the web interface:

1. Login to the web interface.

The System > Information screen appears.

2. Select Function Key > Function Key. The Features screen appears.



3. Select a DSS Key. Set the function key type as Memory Key.

4. Select the Subtype as one of the following: BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, set BLF/DTMF value as the number to be subscribed.

5. Set the corresponding SIP line. The pickup number is provided by the server. See “Answer Someone Else’s Phone (Pick Up)” on page 51.

6. Click Apply to save.

Using the BLF Function

BLF (Busy Lamp Field) shows the status of other users’ phones (idle, busy, ringing, available) connected to your phone system. You need to have this user on one of your BLF’s list.

BLF function:

- Monitor the status of subscribed phones.
- Call the subscribed number.
- Transfer calls/calls to the subscribed number.
- Pickup incoming calls from the subscribed number.

Monitors the status of subscribed phones.

Configuring BLF function keys, when the subscription of the number of the state (idle, ringing, talking) is changed, the LED lights of function key will have a corresponding change, see “Table 9 DSS Key LED Definition” on page 28.

Call the subscribed number.

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

Transfer calls to the subscribed number.

See “Table 11 BLF Function Key Subtype Parameter List” on page 57. The BLF key can be used for blind rotation, attention-rotation, and semi-attention-rotation of the current call and also can invite the subscribed number to join the call and send DTMF, etc.

Pick up incoming calls from subscribed phones.

When configuring the BLF function key, configure the pickup number.

When the subscription number telephone rings, the LED turns red. At this point, press the BLF button to answer the incoming call from the subscribed number. See “Table 9 DSS Key LED Definition” on page 28.

BLF List

Use the BLF list to put numbers in a group on the server side, and the phone uses the URL of this group to make a unified subscription. The specific information, number, name, and status of each number will auto-populate based on the notification sent from the server. The unoccupied Memory Key is then set as the BLF List Key. If the state of the subscription object changes, the corresponding LED light state changes.

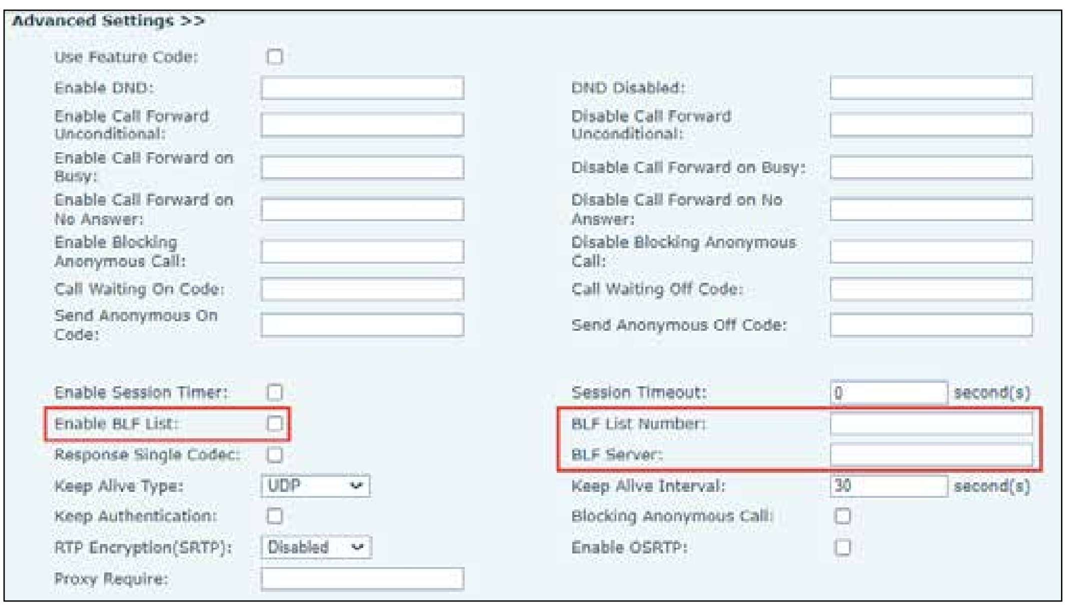
Web Interface

To configure the BLF List through the web interface:

1. Login to the web interface.

The System > Information screen appears.

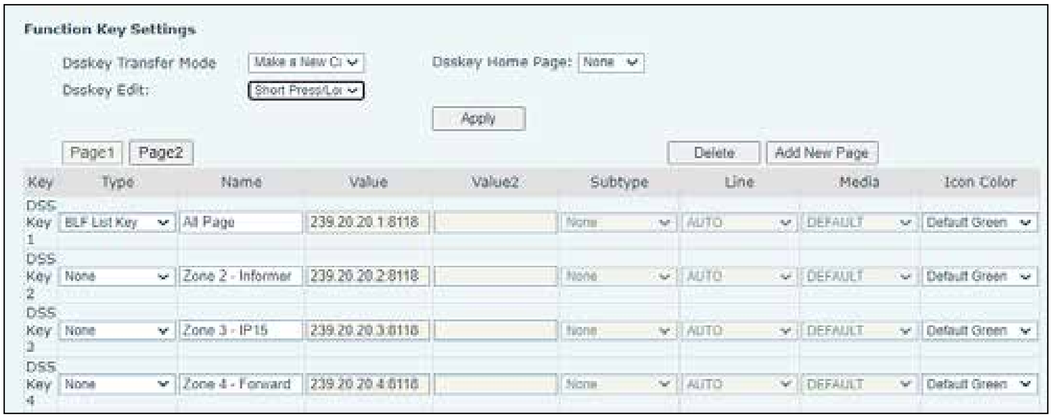
2. Select List > SIP > Advanced Settings. The Advanced Settings screen appears.



Using the BLF List Function

When you complete the configuration, the phone automatically subscribes to the contents of the BLF List group. You can monitor, call, and transfer the corresponding number by pressing the BLF List key.

Select Function Key > Function Key.



Recording a Call

You can record a call. This phone supports a USB flash drive.

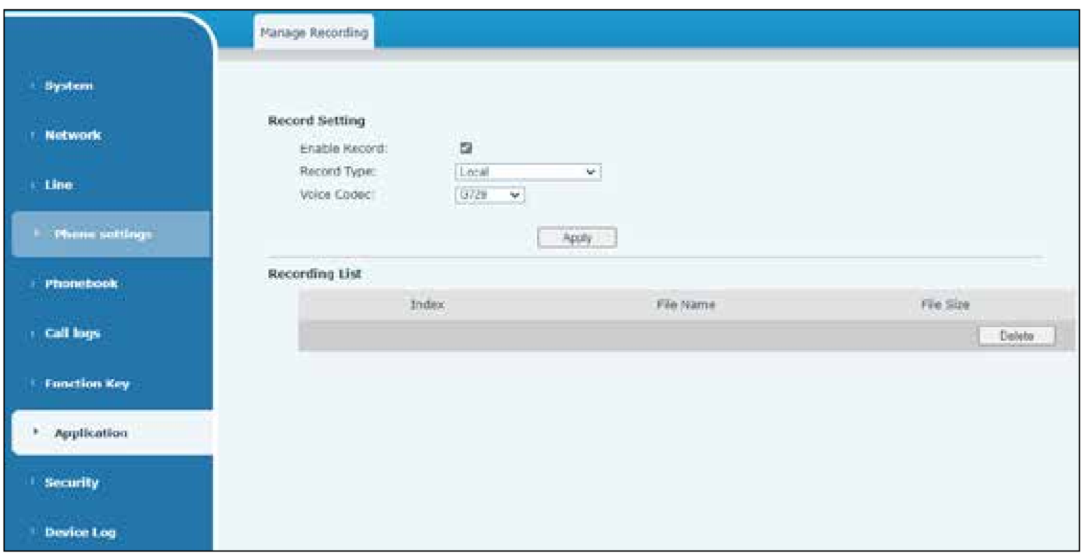
Web Interface

To record a call locally through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Application > Manage Recording. The Manage Recording screen appears.



3. Select the local type and set the voice coding.

Recording Locally

To record a call locally:

1. Plug the U disk (also called USB hard disc drive) into the USB port of the phone, open the recording on the web page, and set the recording type as local recording.
2. Set the DSSkey type as key event and type as record in the phone/web interface.
3. Set up one line call and press the recording key (set DSSkey).
4. End the recording.
5. End the call.

Viewing the Local Recording through the Phone Interface

To view the local recording through the phone interface:

1. Select Menu > Application > USB.
2. Enter USB to view the recording file.

Viewing the Local Recording through the Web Interface

To view the local recording through the web interface:

1. Login to the web interface.

   The System > Information screen appears.
2. Select Application > Manage Recording. The Manage Recording screen appears.
3. View the recorded file.

Listening the Local Recording through the Phone Interface

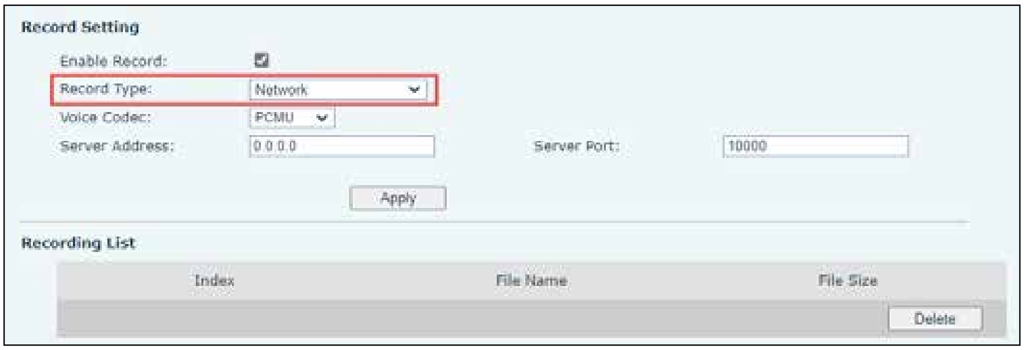
To listen to the local recording through the phone interface:

1. Select Menu > Application > USB.
2. Enter USB to view the recording file.
3. Select the recording file that you want to listen to, and click the play button of Soft key to listen to the recording.

Recording with Network Server

When using the network server to record:

1. Open the recording in the phone web page. Select Application > Manage Recording.
2. Go to Record Type and select Network. The Server Address and Server Port of the recording server are populated and the voice coding is selected. The web screen is as follows.



SIP INFO Record

The phone is registered with a server that supports SIP INFO recording. After registering the account, check the recording module.

To check the recording module:

1. Open the recording in the phone web page Application > Manage Recording.
2. Go to Record Type and select Sip Info. The web screen is as follows.

Agent

Phone Agent means a channel of the Service which allows the Agent to connect to the Contact Center with a simple telephone to manage the Calls (without accessing a computer desktop). The Agent will be able to access some features of the Service with the touch-tones of the telephone.

Normal Mode

To configure the Agent function:

- Through the web interface: set a DSSkey as an Agent, click the Function Key.
- Trough the phone interface, press Menu > Features > Agent to enter the agent page.

IMPORTANT: The SIP server needs to be configured before the account can be configured.

Parameter	Description
Normal mode	
Number	Set the proxy account number.
User	Set the proxy account number to verify the username.
Password	Set the proxy account number to verify the password.
Line	Select the SIP line.
CallLog	Choose to save all types or delete.
Hotel Guest mode	
Number	Set the proxy account number.
Password	Set the proxy account number to verify the password.
Line	Select the SIP line.
CallLog	Choose to save all types or delete.

Using Agent Functions

To using agent functions:

1. After the phone has been configured on the SIP server, fill in the correct number and username password, click login, and then the phone can be registered to the SIP server.
2. After registration, click logout, and the phone can delete the username and password, and log out of the SIP account.
3. Click Unregister, and the phone retains the username and password and logs out of the SIP account.

Intercom Settings

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

Web Interface

To enable the Intercom Settings through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Phone settings > Features > Intercom Settings. The Phone settings Features screen appears.

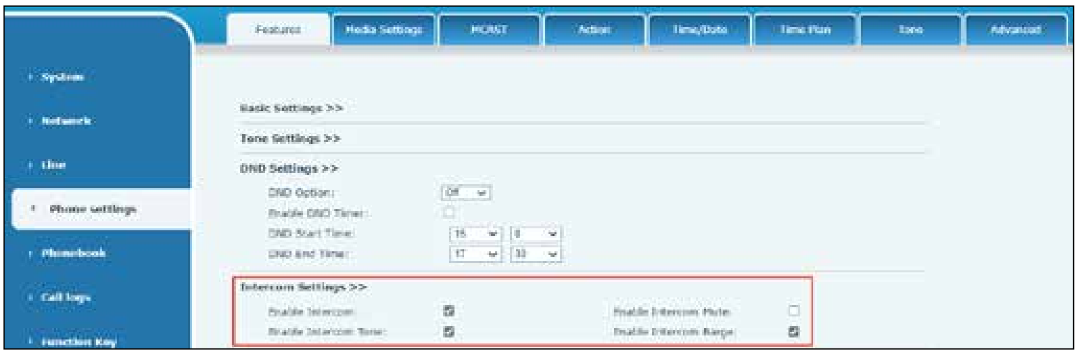


Table 12 Phone settings > Features > Intercom Settings Parameters

Parameter	Description
Enable Intercom	The phone accepts the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after a specific delay.
Enable Intercom Mute	Enable mute mode during the intercom call.
Enable Intercom Tone	If the incoming call is an intercom call, the phone plays the intercom tone.
Enable Intercom Barge	The phone auto answers the intercom call during a call. If the current call is an intercom call, the phone rejects the second intercom call.

MCAST

MCAST allows you to make broadcast calls to people in a multicast group. You can configure a multicast DSS Key on the phone, which allows you 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 a pre-configured multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.

Web Interface

To enable the MCAST through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Select Phone settings > MCAST. The MCAST settings Features screen appears.

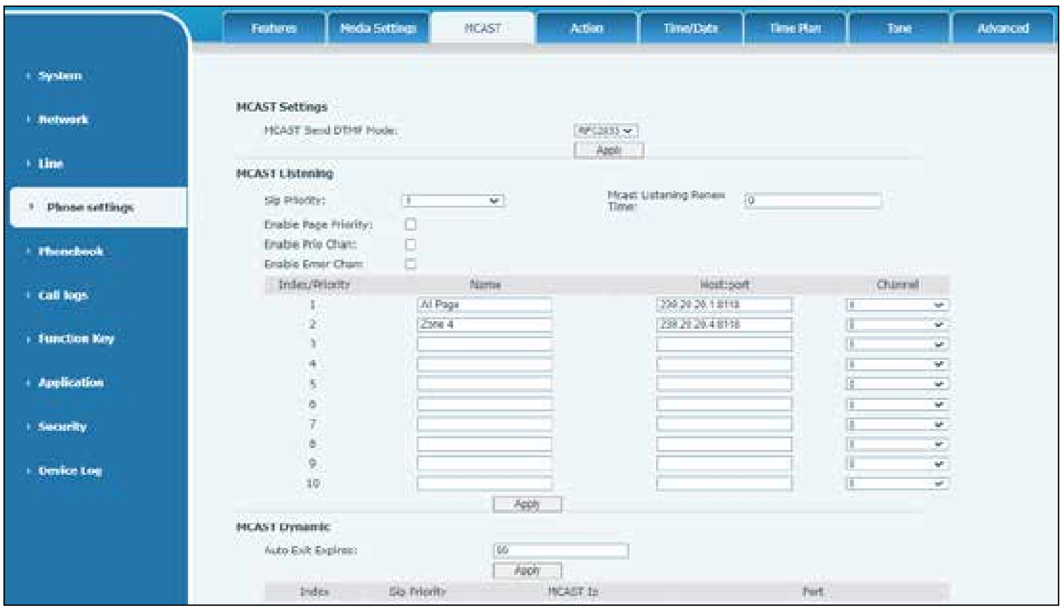


Table 13 Phone settings > MCAST Parameters

Parameter	Description
Sip 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.
Name	Type the multicast server name.
Host:port	Type the multicast server's multicast IP address and port.

Multicast

To enable the multicast:

1. Go to the web page of Function Key > Function Key, type the address in the value field, and select the codec in the subtype field.
2. Click Apply.
3. Long press the side key on the phone. Set up the name, host and port of the receiving multicast.
4. Press the DSSKY of Multicast Key which you set.
5. Receive end receives multicast call and plays multicast automatically.

SCA (Shared Call Appearance)

Users need the support of server end to use SCA (Shared Call Appearance) function.

Configuring on a Phone

To configure:

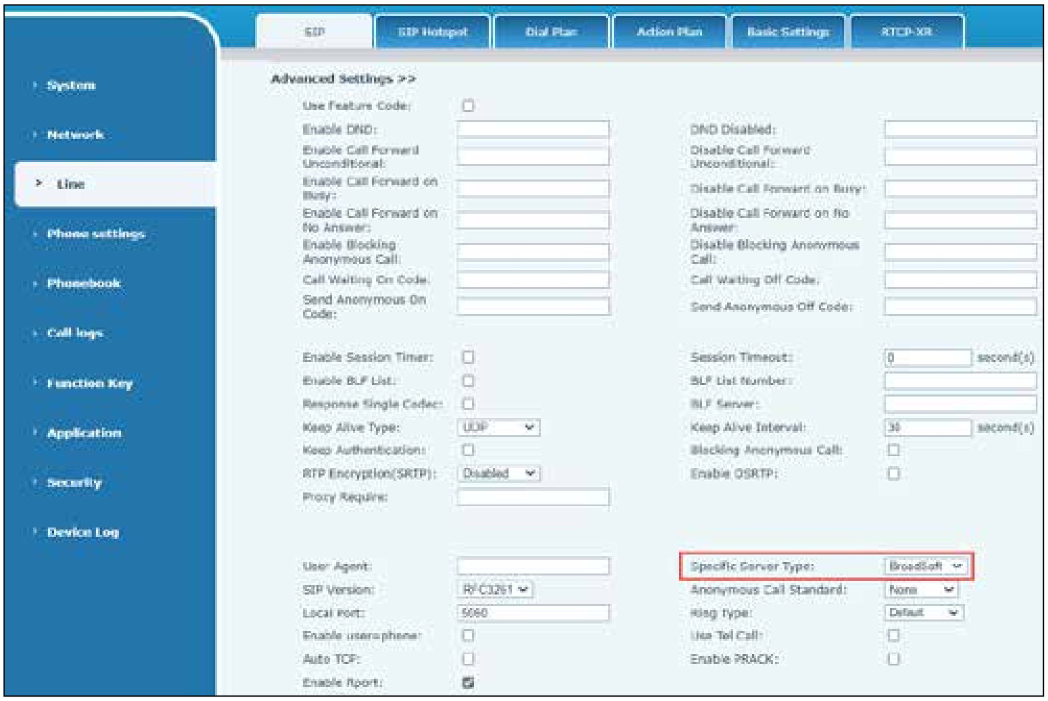
1. When registering with the BroadSoft server, your phone can register the account created previously on multiple terminals.
2. Go to the web page Line > SIP.

Table 14 Line > SIP Parameters

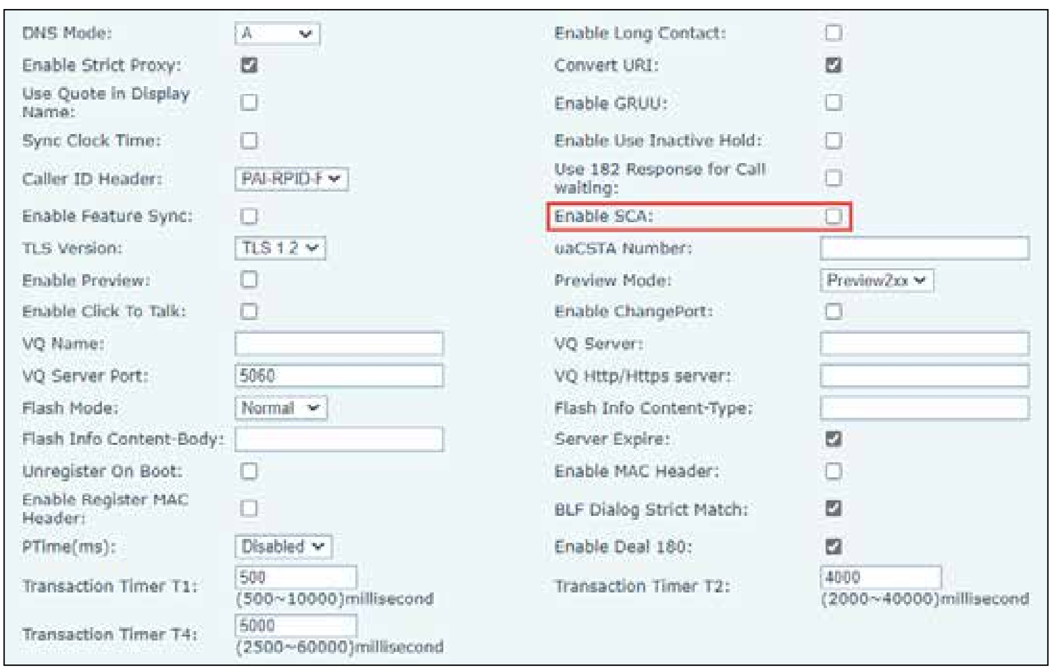
Parameter	Description
Line Status	Should be Registered.
Username	The created SCA accounts.
Authentication User/ Password	The username and password of the primary account created.
Server Address	BroadSoft Server address.

3. After the phone set registers with the BroadSoft server, set a server type. Go to the web page of Line > SIP > Advanced Settings.

4. Set Specific Server Type to BroadSoft.

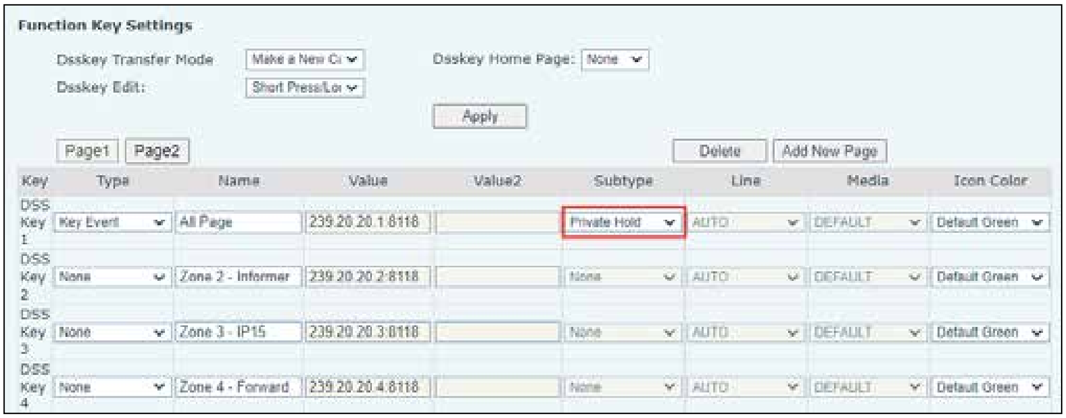


5. If the phone needs to enable the SCA function. Go to the web page of the phone, select Line > SIP > Advanced Settings. Click Enable SCA. If SCA is not enabled, the registered line is the private line.



6. After an account is configured and successfully registered, you can configure lines whose DSS Key is Shared Call Appearance on the Function Key page to facilitate viewing the call status of the group. Each line key represents a call appearance. See "Table 9 DSS Key LED Definition" on page 28.

7. To enable a private hold, configure keys whose DSS Key is Private Hold on the Function Key page. Pay attention that the public hold key is the softkey-hold key during a call. Go to the web page Function Key > Function Key.



Configure each phone registered with the BroadSoft server, and then you can use the SCA function.

LED Status

To enable viewing the call status of a group, configure the DSS Key as SCA. The following table describes the LEDs of lines in different states.

Table 15 LED Status of SCA

State and Direction	Local	Remote
Idle	Off	Off
Seized	Steady green	Steady red
Progressing (outgoing call)	Steady green	Steady red
Alerting (incoming call)	Fast blinking green	Fast blinking green
Active	Steady green	Steady red
Public Held (hold)	Slow blinking green	Slow blinking red
Held-private (private hold)	Slow blinking yellow	Steady red
Bridge-active (Barge-in)	Steady green	Steady red
Bridge-held	Steady green	Steady red

Shared Call Appearance (SCA)

The following describes four scenarios. In the following scenarios, the manager and assistant register the same SCA account, configure the account based on the following steps.

Scenario 1

When this account receives an incoming call, the phone sets of both the manager and the assistant receives the call and ring. If the manager is busy, the manager can reject the call, and the manager's phone stops ringing, but the assistant's phone keeps ringing until the assistant rejects or answers the call or the call times out.

Scenario 2

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

Scenario 3

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

Scenario 4

The manager is on a call with a customer and requires the assistant to join the call to make records. The assistant can press the corresponding SCA line key to join this call.

Messaging

SMS with the Web Interface

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

Send Messages

To send messages through the phone interface:

1. Press Menu > Message > SMS.
2. Create new messages, select lines and send numbers.
3. After editing is completed, press Send.

View SMS

To view SMS through the phone interface:

1. Use the navigation keys to select the standby icon (message).
2. After selecting, press the OK button to enter the SMS inbox interface.
3. Select the unread message and press the OK button to read the unread message.

Reply to SMS

To reply to SMS through the phone interface:

1. Use the navigation keys to select the standby icon (message).
2. After selecting, press the OK button to enter the SMS inbox interface.
3. Select the message you want to reply to, select Softkey's (Reply), edit it, and click Send.

MWI (Message Waiting Indicator)

If the service of the lines supports the voice message feature, when the phone call is not answered, the caller can leave a voice message. You will receive a voice message notification from the server, and the phone will display a voice message waiting icon on the standby screen.

Enabling the Voice Message Number

To listen to a voice message, you must first enable the voicemail number. After the voicemail number is enabled, you can retrieve the voicemail of the default line.

To send voicemail through the phone interface:

1. Press Menu > Message > Voice Message.
2. Select the line to configure.
3. Select Enabled by using the navigator buttons.
4. Press OK to save.

SIP Hotspot

A SIP hotspot is a useful function. Configure the SIP hotspot function to use group ringing. You can expand SIP accounts.

Set functions as a SIP hotspot, and other phones (B and C) function as SIP hotspot clients. When somebody calls phone A, phones A, B, and C all ring simultaneously. 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 up a SIP hotspot, register at least one SIP account.

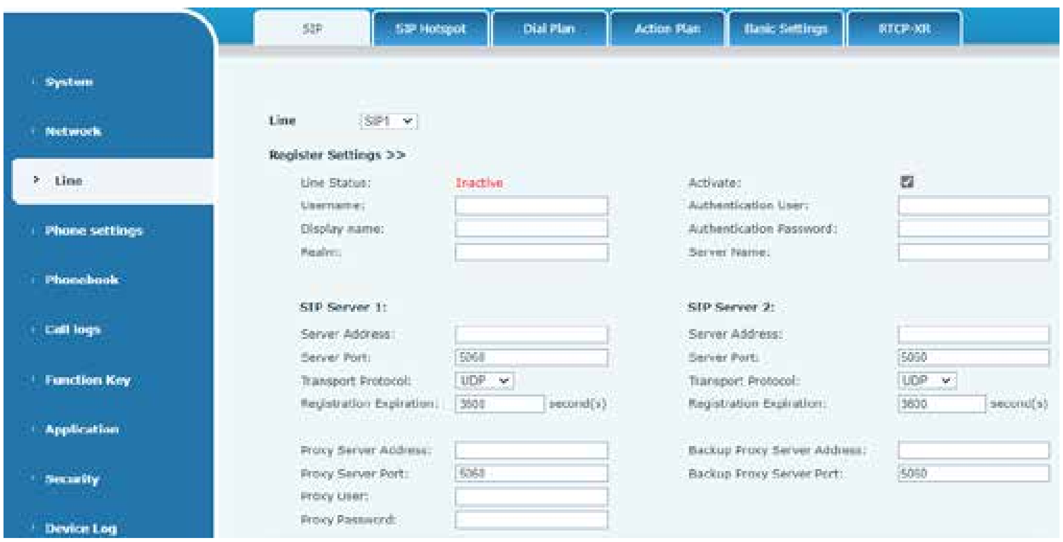
Web Interface

To enable the SIP hotspot through the web interface:

1. Login to the web interface.

The System > Information screen appears.

2. Go to the web page Line > SIP.



Configuring a SIP Hotspot Server

To configure the SIP hotspot server through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Line > SIP Hotspot.
3. Set Enable Hotspot to Enabled.
4. Set Mode to Hotspot.
5. Click Apply to save.

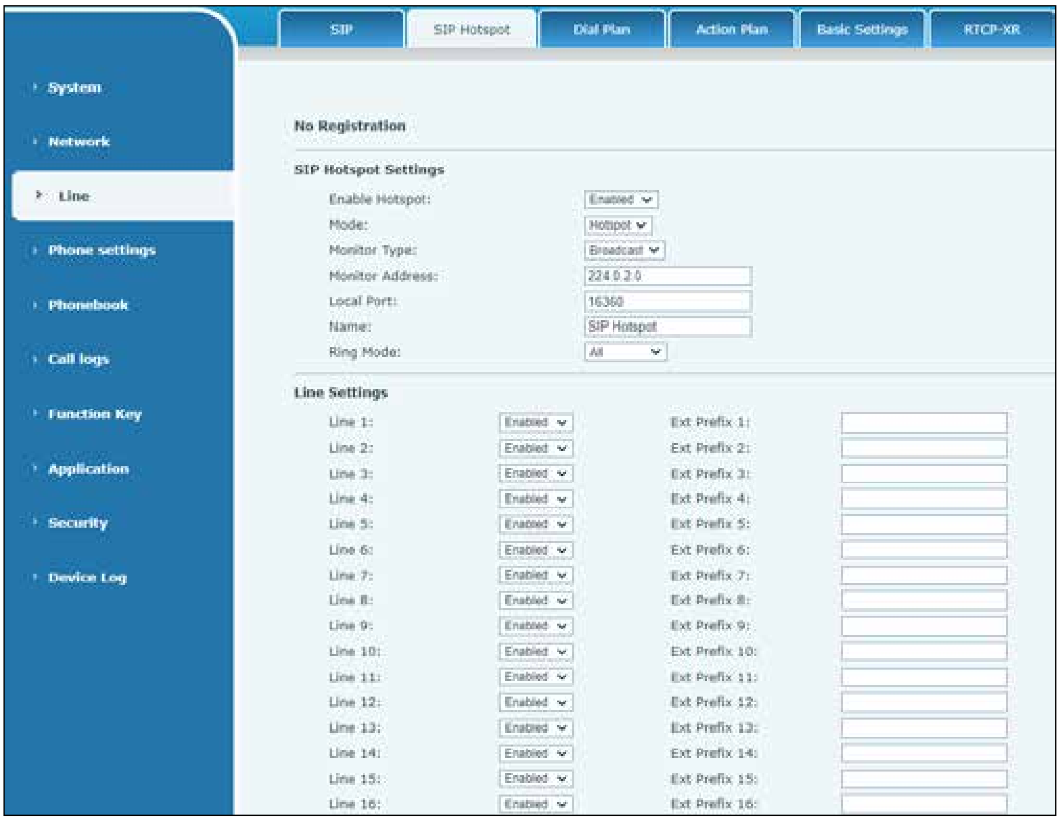


Table 16 Line >SIP Hotspot Parameters

Parameter	Description
Device Table	If your phone is set to a SIP hotspot server, Device Table will display as Client Device Table, which is connected to your phone. If your phone is set to a SIP hotspot client, Device Table will display as Server Device Table, which you can connect to.
SIP Hotspot Settings	
Enable Hotspot	Set to Enable to enable the feature.
Mode	Select Hotspot for the phone to be a SIP hotspot server. Select Client for the phone to be a SIP hotspot Client.
Monitor Type	Select either Multicast or Broadcast. If you want to limit the broadcast packets, use broadcast. If you select broadcast, the SIP hotspot phone must be broadcast.

Parameter	Description
Monitor Address	The address of broadcast. The hotspot server and hotspot client must be the same.
Local Port	Type the local port number.

Configuring a SIP Hotspot Client

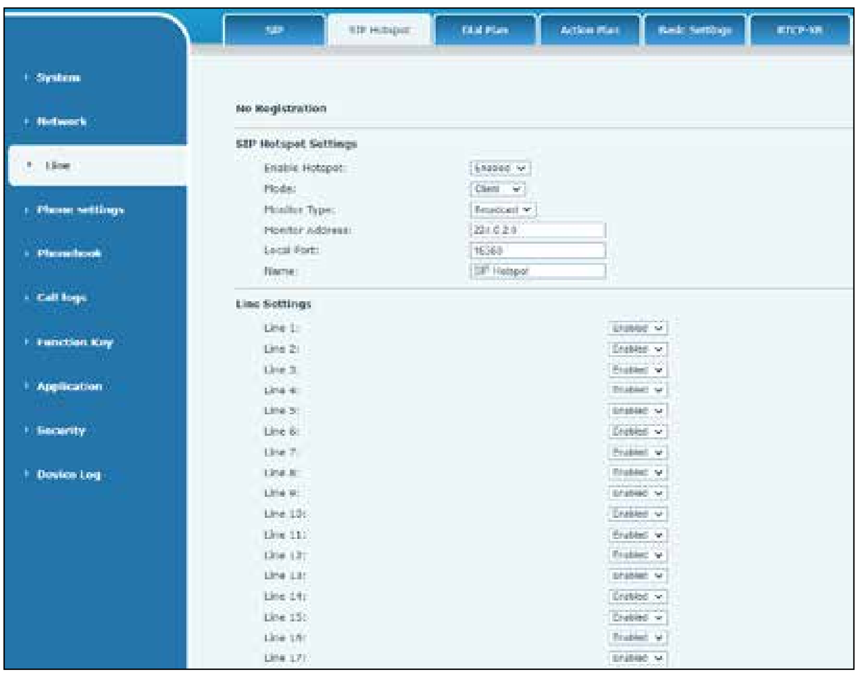
To set as a SIP hotspot client, you do not need to set up a SIP account. The Phone set will automatically obtain and configure a SIP account.

To configure the SIP hotspot client through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Line > SIP Hotspot.
3. Set Enable Hotspot to Enabled.
4. Set Mode to Client.
5. Click Apply to save.



As the hotspot server, the default extension number is zero (0). When the phone is used as the client, the extension number is increased from one (1). View the extension number through the SIP Hotspot screen.

Call extension number:

- The hotspot server and the client can dial each other through the extension number; for example, extension one (1) dials extension zero (0).

Phone Settings

Basic Settings

Language

Your phone can display text in many languages.

Phone Interface

After resetting the factory settings, you need to set the language.

To set the language through the phone interface:

1. Press the Menu key > Basic > Language.
2. Select the desired language and press the OK soft key.

Web Interface

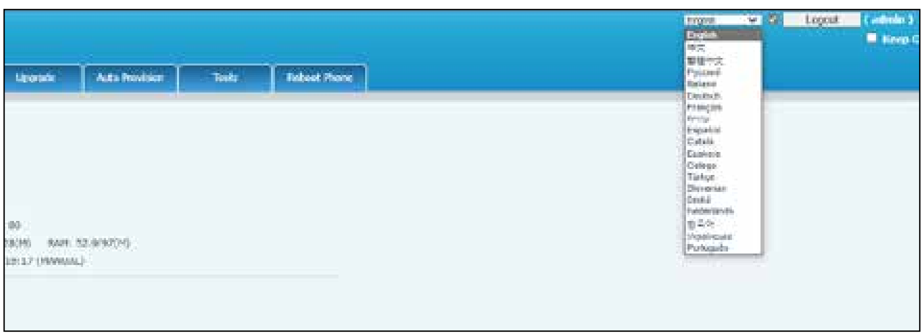
To set the language through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. In the top right corner, click the arrow to select the language.

   The function box on the right side of the language drop-down box is the synchronize language to phone check box. If selected, the phone language is synchronized with the web page language. If it is not selected, it is not synchronized.



Time and Date

You can set the phone's time and date.

Phone Interface

To set the phone's time and date through the phone interface:

1. Press the Menu key > Basic > Time & Date.
2. Press the up/down navigation key to change the line.
3. Press OK to save.

Web Interface

To set the phone's time and date through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Phone settings > Time/Date.

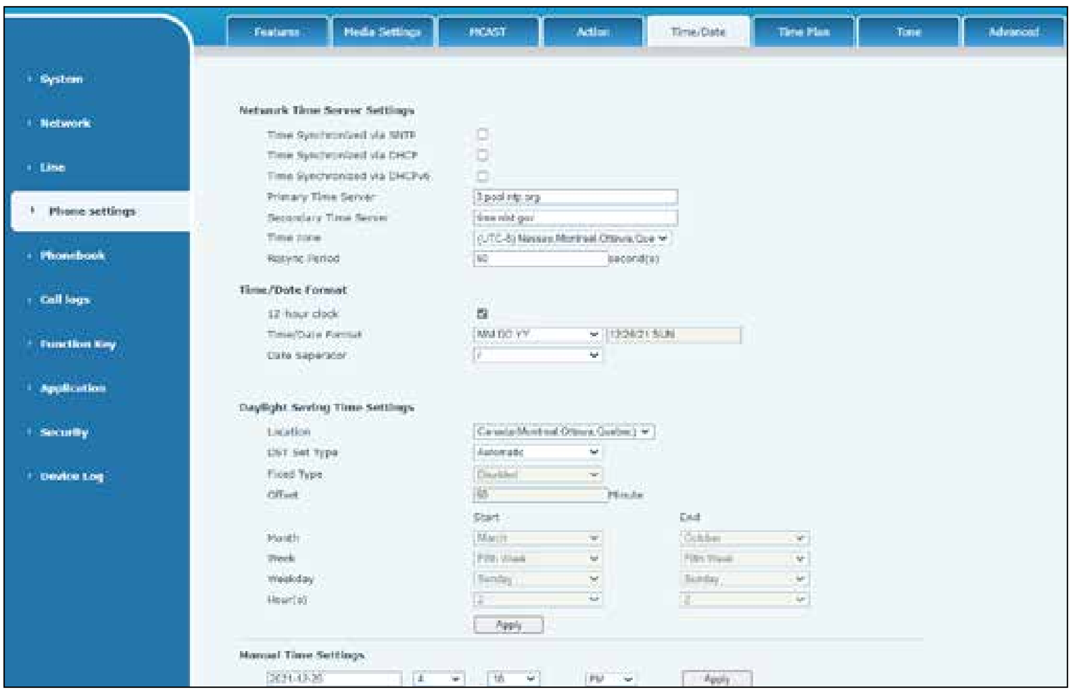


Table 17 Phone settings > Time/Date Parameters

Parameter	Description
Network Time Server Settings	When checked, enables network time synchronization. The default is enabled. When unchecked, you can modify data manually.
Primary Time Server	If the options are unchecked, enter the server address.
Secondary Time Server (Optional)	If the options are unchecked, enter the secondary server address.
Time zone	Click the down arrow to select the time zone.
Resync Period	Type the resync period in second(s).
Time/Date Format	

Parameter	Description
12-Hour Clock	If checked, displays the clock in a 12-hour format.
Time/Date Format	Click the down arrow to select the time format. An example of the format appears in the text box on the right.
Date Separator	Click the down arrow to select the separator between year, month, and day.
Manual Time Settings	Select the date using the date picker. Click the down arrow to select the hour, minute, and AM or PM.
Apply	Click to save.

Screen

You can set the phone's screen parameters.

Phone Interface

To set the phone's screen parameters through the phone interface:

1. Press the Menu key > Basic > Screen.
2. Press the up/down navigation key to change the line.
3. Press OK to save.

Web Interface

To set the phone's screen parameters through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Phone settings > Advanced.

Table 18 Phone settings > Advanced Parameters

Parameter	Description
Backlight Active Level	Set the brightness level in use from 1 to 16.
Backlight Inactive Level	Set the brightness level in the energy-saving mode from 0 to 16.
Backlight Time	Set the backlight time in seconds.
Screensaver	Click the down arrow to Enable or Disable the screensaver.
Timeout to Screensaver	Set the screensaver in seconds.
Apply	Click to save.

Ring

You can change the sound that your phone uses for incoming calls.

Phone Interface

To set the phone's ring tone through the phone interface:

1. Press the Menu key > Basic > Ring.
2. Press the right/left navigation key to change the line.
3. Press OK to save.

Voice Volume

You can change the voice volume on the phone.

Phone Interface

To set the voice volume through the phone interface:

1. Press the Menu key > Basic > Voice Volume.
2. Press the right/left navigation key to change the line.
3. Press OK to save.

Greeting Words

You can set Voicemail greeting on the phone.

Phone Interface

To set the Voicemail greeting through the phone interface:

1. Press the Menu key > Basic > Greeting Words.
2. Press the right/left navigation key to change the line.
3. Press OK to save.

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

Reboot

You can reboot the phone.

Phone Interface

To reboot the phone through the phone interface:

1. Press the Menu key > Basic > Reboot System.
2. Press OK to reboot or Cancel to exit.

Phone Book

Local Contact

Use the phone book to store contact information of coworkers.

To set up the phone book through the phone interface:

1. Press Contact soft key or Menu > PhoneBook > Local Contacts.
2. Select All Contacts.
3. To dial an entry, press the Dial soft key or pickup handset or press or .

Adding a new contact entry

To add a new contact entry:

1. Press Contact soft key, press All Contacts or other group, and press Add.
2. Enter the name and number and press the OK soft key.

When there are contact records in the phone book, the records are arranged in alphabetical order. You may browse the contacts with up and down navigator keys. The record indicator tells you which contact is selected. You may review the contact's information by pressing the OK button.

Editing a contact entry

To edit a contact entry:

- Press Option > Edit button.

To delete a contact:

1. Select the contact to be deleted, and press Option > Delete.
2. Press OK to confirm.

Working with group lists

Set up a group list:

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

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

Browsing contact lists

To brows a contact list:

You can browse contacts in a group by opening the group in group list with the OK button.

When you are browsing contacts of a group, you can add contacts in that group:

1. Press the Add button to enter the group contacts management interface.
2. Press the OK button to save the contact.

   NOTE: The contact is also added in local phonebook. You can delete the contact from the group by pressing Option > Delete.

Blocked Numbers

Phone Interface

To block a phone number through the phone interface:

1. Press Contact soft key or Menu > PhoneBook > Blocked List.
2. Select Blocked List.
3. Press the Add button.
4. Enter the Number, Line, and Number/Prefix.
5. Press OK to confirm.

Web Interface

To block a phone number through the web interface:

1. Login to the web interface.

   The System > Information screen appears.
2. Go to the web page Phonebook > Call List.
3. Use the Add, Delete, and Delete All Buttons to add, delete, and delete all to edit the contact list.

Cloud Contacts

Cloud phonebook allows you to configure the phone by downloading a phonebook from a cloud server. It is convenient for you to use the phonebook from a single source to save the effort of creating and maintaining the contact list individually.

NOTE: The Cloud phonebook is only temporarily downloaded to the device to ensure the you get the latest phonebook. However, the downloading may take a few seconds, depending on the network condition. Therefore, it is highly recommended that you save important contacts from the cloud to local phonebooks to save download time.

Phone Interface

To configure the Cloud phonebook:

- Press Menu > PhoneBook > Cloud Contacts.

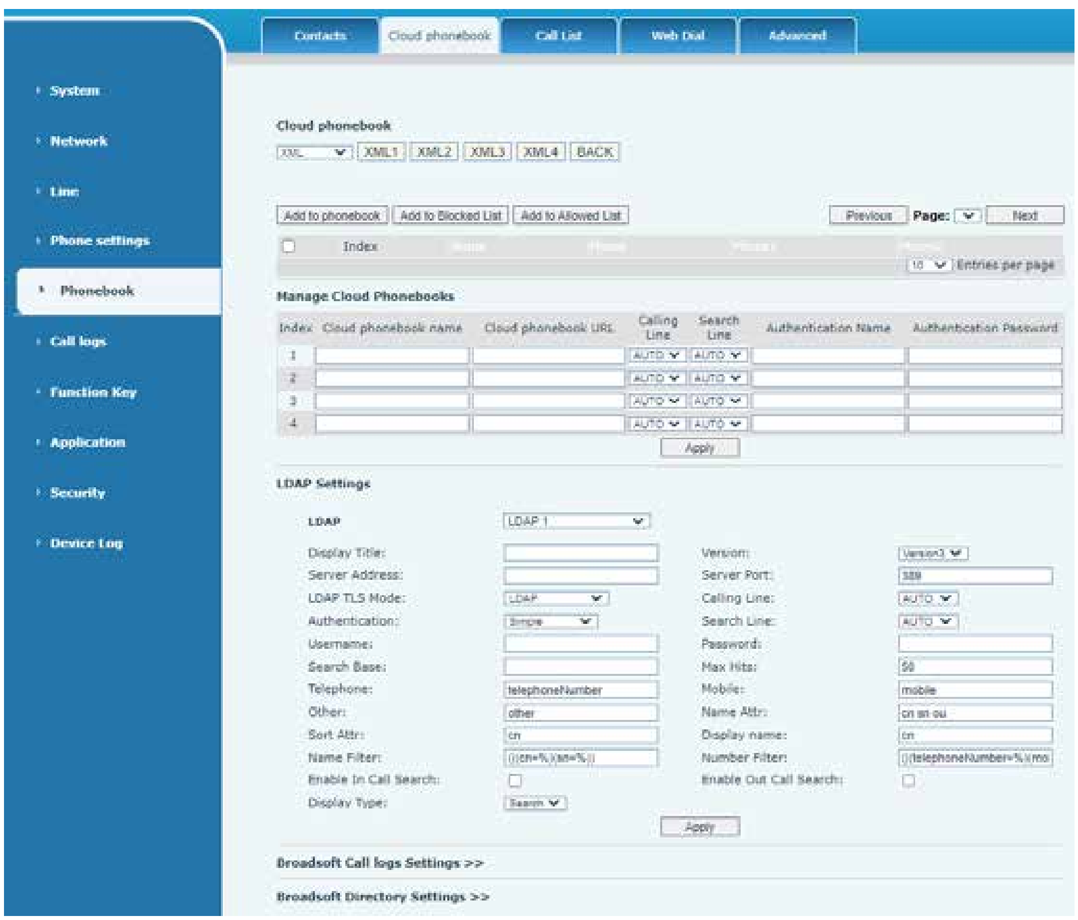
Web Interface

Complete the first configuration on the cloud through the web interface:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Phonebook > Cloud phonebook.



Downloading the Cloud Phonebook

To download the Cloud phonebook:

- Press either OK or Enter.

The phone will start downloading the phonebook. You will be prompted with a warning message if the download fails. Once the Cloud phone book is downloaded completely, you can browse the contact list and dial the contact number the same as in the local phonebook.

Call Log

Call logs are a record of when phone calls are made and who they were with.

To view you call log:

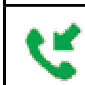
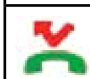
1. Press CallLog soft key or Menu > CallLog.

2. Browse the call logs with up and down navigator keys.

Each call log record is presented with a call type and a call party number or name.

- Check further call log detail by pressing the OK button.
- Dial the number by pressing the Dial button.
- Add the call log number to phonebook by pressing Option > Add to Contact.
- Delete a call log by pressing the Delete button.
- Clear all call logs by pressing the Delete All button.

You can filter the call records of specific call types to narrow the scope of search records and select a call record type with the left and right navigation keys. For a complete list, see “Table 6 Status Prompt and Notification Icons” on page 24.

	Call forward calls
	Outgoing calls
	Incoming calls
	Missed calls

Function Key

Phone Interface

You can use the page switch key to switch DSS display pages quickly. Also, you can long press each DSS key to modify the corresponding key Settings.

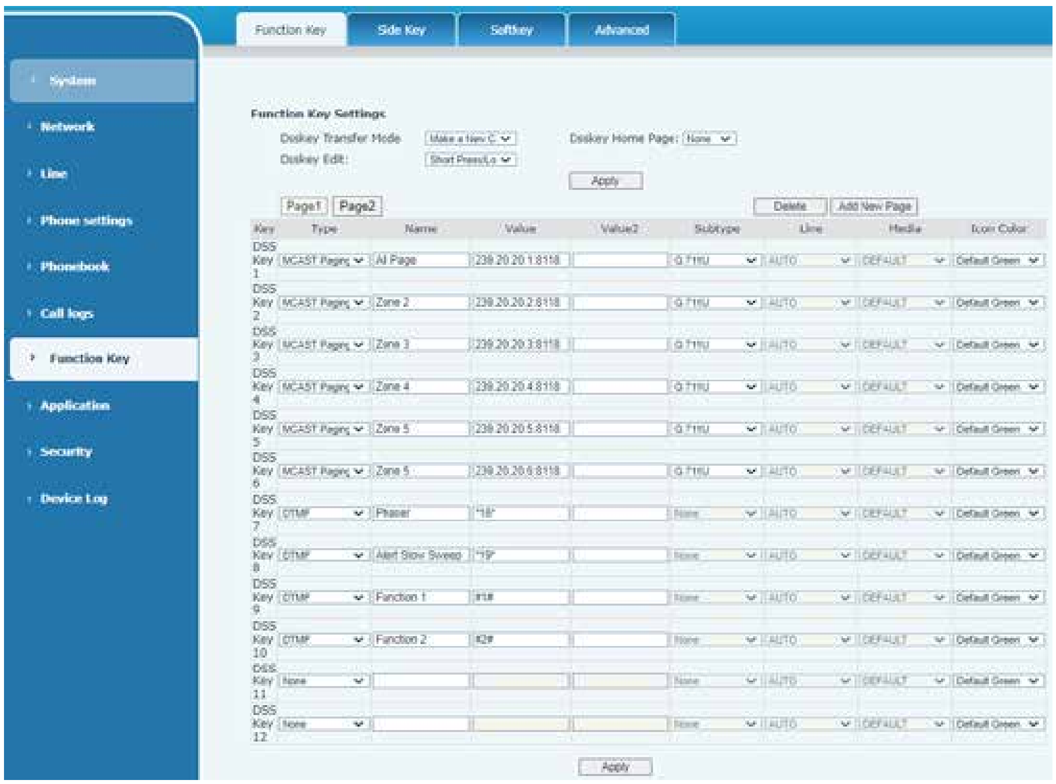
Web Interface

To modify function keys:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Function Key > Function Key.



You can add the user-defined title for the DSS Keys, which is configured as Memory Key/ Line/URL/Multicast/Prefix. For more information, see "Function Key > Function Key" on page 140 and "Table 9 DSS Key LED Definition" on page 28.

Wi-Fi

The phone supports wireless Internet access and requires a USB Wi-Fi adapter specified by location; therefore, the phone needs to support a U disk.

Phone Interface

To enable the WLAN options through the phone interface:

1. Press Menu > Basic > WLAN.
2. Select the wireless network and use the left and right keys to activate it. Enable the device to search the current wireless network automatically.
3. Select the available network.
4. Enter the user name and password to connect successfully.

   NOTE: if no wireless USB dongle is inserted, the prompt "wireless adapter has been removed" appears. If a USB dongle is plugged in, the wireless network will be the priority network even if the network cable is plugged in.

Headset

Wired Headset

The phone supports wired earphones with an RJ9 interface, which can play incoming call sounds and talk with earphones.

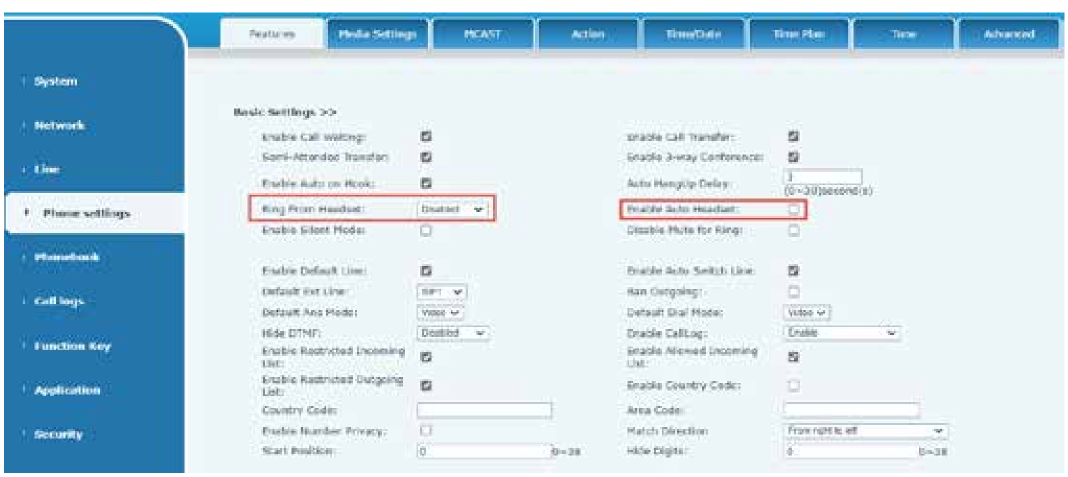
After the phone is connected to the headset, the default DSS key is the green LED, which indicates that the headset can be used normally.

Web Interface

To modify the headset:

1. Login to the web interface.

   The System > Information screen appears.
2. Go to the web page Phone settings > Features.
3. Set the headset answering function and the ring tone.



EHS Headset

Phone Interface

To enable the EHS headset through the phone interface:

1. Press Menu > Features > Advanced.
2. Select EHS Headset. The default is Disable.

Bluetooth Headset

The phone supports Bluetooth® applications and can be compatible with Bluetooth® headsets with the CSR 4.0 chip. It needs to use a USB Bluetooth® adapter specified by location. Therefore, the device needs to support a U disk.

Phone Interface

To enable Bluetooth® through the phone interface:

1. Press Menu > Basic > Bluetooth.
2. Select Bluetooth and use the left and right keys to enable Bluetooth. Select Paired Device. If no paired is displayed, press the Scan key to search and select the scanned device to connect.

Using the Bluetooth headset.

To use the Bluetooth headset:

- Call answering: When the Bluetooth headset is connected to the phone, answer the incoming by pressing the Bluetooth answer button.
- Hang up: When talking with a Bluetooth headset, hang up the phone by pressing the button on the 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.
- Bluetooth 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 the double-click redial function. Whether this function is supported or not, 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.

Advanced

Line Configurations

Save the adjustment by pressing the OK button when done.

If you want to configure more options, use the web management portal to modify or Advanced Settings in accounts on the individual line to configure those options.

Network Settings

Network Settings

IP Mode

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

You can select the available mode via “<” or “>”. The selected IP mode is activated after pressing the OK button.

IPv4

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

DHCP

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

- Use DHCP DNS: It is enabled as default, which means the phone gets the DNS address from DHCP server. Disable means it does not.
- Use DHCP time: It is disabled as default. Select Enable to manage the time of get DNS address from the DHCP server. Disable means it does not.

PPPoE

When using PPPoE, the phone gets the IP address from PPPoE server.

- Username: PPPoE username.
- Password: PPPoE password.

Static IP mode

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

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

IPv6

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

- DHCP configuration refers to IPv4.
- Static IP configuration is almost the same as IPv4's, except for the IPv6 Prefix.
- IPv6 Prefix: IPv6 prefix, it is similar with mask of IPv4.

QoS and VLAN

Link Layer Discovery Protocol (LLDP)

LLDP is a vendor-independent link layer protocol used by network devices for advertising their identity and capabilities to neighbors on a LAN segment.

The phone could use LLDP to find the VLAN switch or other VLAN devices and use LLDP learn feature to apply the VLAN ID from the VLAN switch to the phone itself.

Cisco Discovery Protocol (CDP)

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

Table 19 QoS and VLAN

Parameters	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

Virtual Private Network (VPN)

VPN is a technology that allows devices to create a tunneling connection to a server and become part of the server's network. The network transmission of the device may be routed through the VPN server.

You may need to establish a VPN connection before activating a line registration. The device supports two VPN modes: Layer 2 Transportation Protocol (L2TP) and OpenVPN.

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

Layer 2 Transportation Protocol (L2TP)

IMPORTANT The device only supports non-encrypted basic authentication and non-encrypted data tunneling. If you need data encryption, use OpenVPN instead.

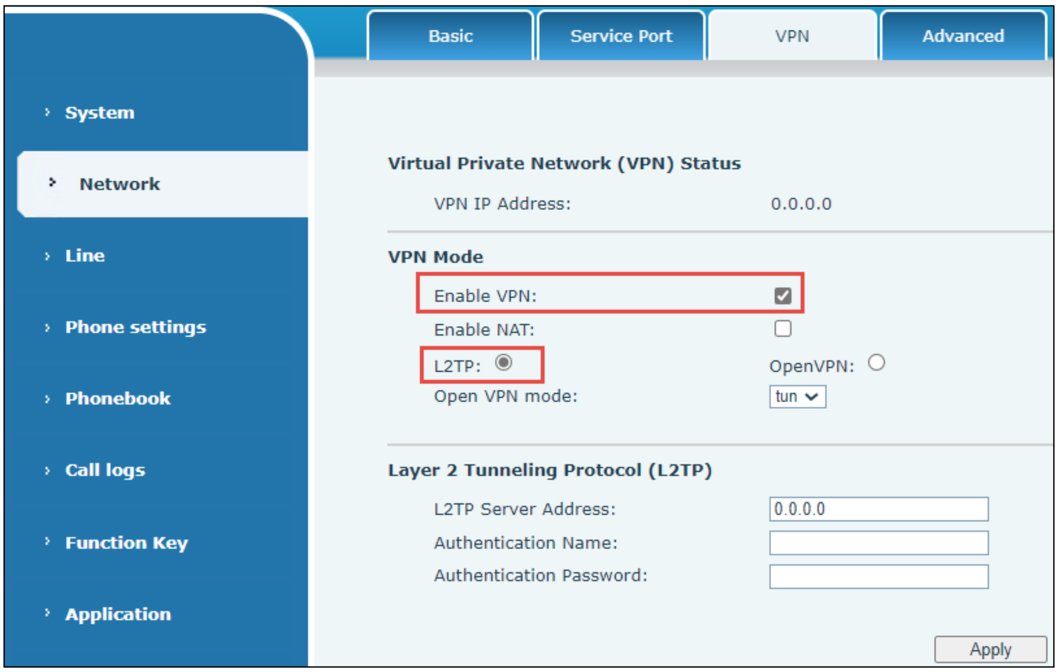
Web Interface

To establish an L2TP connection:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Network > VPN.



3. Select the Enable VPN check box.
4. Click L2TP.
5. In the L2TP Server Address, type in the server address.
6. In the Authentication Name, type in the username.
7. In the Authentication Password, type in the password.
8. Click Apply for the device to connect to the L2TP server.

When the VPN connection is established, the VPN IP Address displays in the VPN status window. There may be a delay in the connection. You may need to refresh the page to update the status. Once the VPN is configured, the device tries to connect with the VPN automatically when the device boots up every time until you disable it. Sometimes, if the VPN connection does not establish immediately, you may try to reboot the device and check if the VPN connection is established after reboot.

OpenVPN

To establish an OpenVPN connection:

1. Get the following authentication and configuration files from the OpenVPN hosting provider and name them as the following:
    - OpenVPN Configuration file: client.ovpn
    - CA Root Certification: ca.crt
    - Client Certification: client.crt
    - Client Key: client.key
2. Upload these files to the device.
3. Go to the web page Network > VPN.

4. Click Enable VPN.
5. Click OpenVPN Files.
6. Click Apply to enable an OpenVPN connection.

Like L2TP connection, the connection is established whenever the system reboots until you manually disable it.

Web Server Type

Configure the Web Server mode to be HTTP or HTTPS, which is activated after the reboot. Then you could use HTTP/HTTPS protocol to access a pone web page.

Set The Secret Key

Phone Interface

When the device is in the default standby mode:

- Press Menu > Advanced setting. Select OK.
- The Advance setting password is fedsig.

To change the current password:

1. Enter your current password.
2. Enter a new password and confirm the new password.

    NOTE: After configuring the menu password, it works immediately.

A keyboard password is used to unlock the phone when locked.

You can only set to enable or disable the keyboard password through the phone interface.

- Enter the keyboard password and select OK. The default is fedsig.

  NOTE: The keyboard password is disabled by default. When it is enabled, the keyboard is locked after timeout.

  If you do not configure the keyboard lock time, zero (0) is the default. Long pressing the pound (# key) locks the phone.

There will be a lock icon in the top of LCD. Phone will display “Enter Password” after pressing any keys.

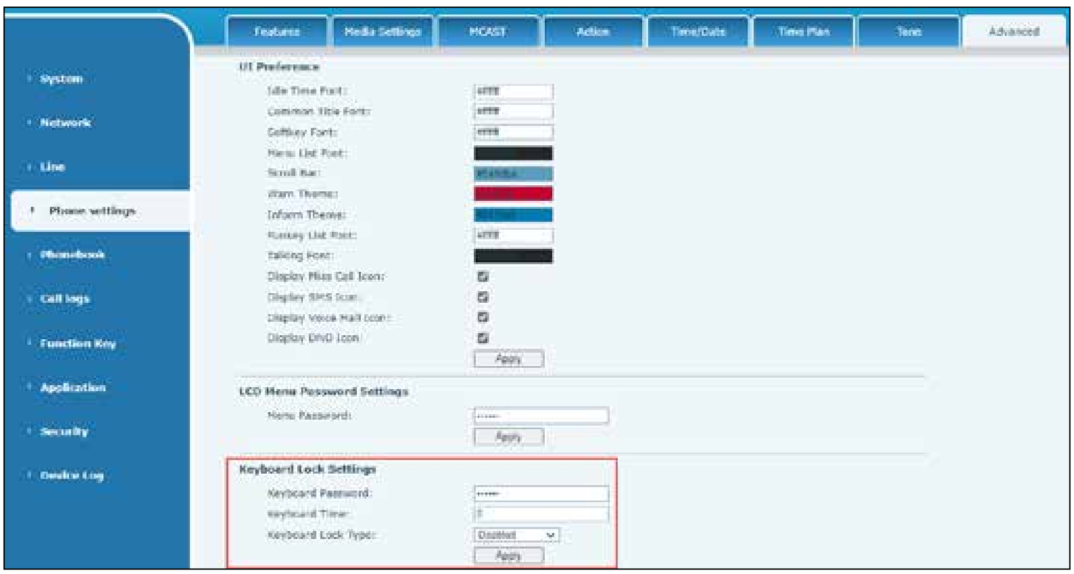
Web Interface

To unlock keyboard:

1. Login to the web interface.

The System > Information screen appears.

2. Go to the web page Phone settings > Advanced.



3. Enter your Keyboard Password.

4. Click Apply.

Maintenance

Phone Interface

To change Auto Provision:

- Press Menu > Advanced setting > Maintenance > Auto Provision.

The phone supports SIP PnP, DHCP options, Static Provision, TR069. If all of the four methods are enabled, the priority from high to low is PNP > DHCP > TR069 > Static Provisioning. Transferring protocol: FTP, TFTP, HTTP, HTTPS.

Web Interface

To change Auto Provision:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page System > Auto Provision.

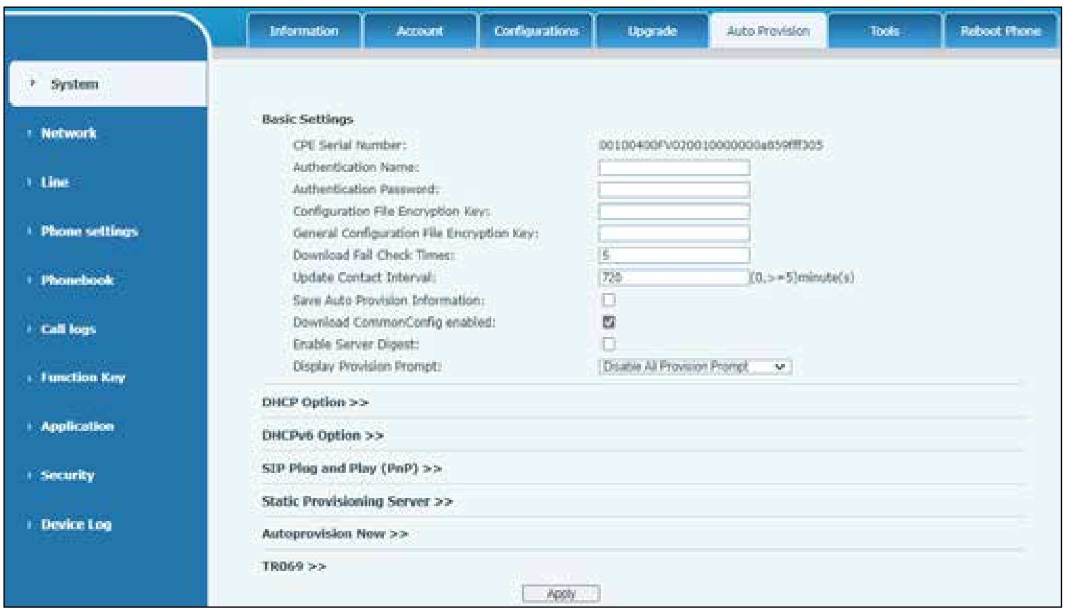


Table 20 System >Auto Provision Parameters

Parameters	Description
Basic settings	
CPE Serial Number	Displays the device Serial Number.
Authentication Name	The username of the provision server.
Authentication Password	The password of the provision server.
Configuration File Encryption Key	If the device configuration file is encrypted, add the encryption key here
General Configuration File Encryption Key	If the common configuration file is encrypted, add the encryption key here.
Download Fail Check Times	If download failed, the phone retries with the configured times.
Update Contact Interval	Phone updates the phonebook with the configured interval time. If it is 0, the feature is disabled.

Parameters	Description
Save Auto Provision Information	Save the HTTP/HTTPS/FTP username and password. If the provision URL is kept, the information is kept.
Download Common Config enabled	If checked, phone downloads the common configuration file.
Enable Server Digest	If checked, if the configuration of server is changed, the phone downloads and updates.
DHCP Option	
Option Value	Configures the DHCP option. The DHCP option supports DHCP Option 66, Option 43, Custom Option, or Disabled. The default is Option 66.
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be the same as the server define.
Enable DHCP Option 120	If checked, gets the SIP server address from DHCP server.
DHCPv6	
Option Value	Configures the DHCP option. The DHCP option supports DHCP Option 66, Option 43, Custom Option, or Disabled. The default is Option 66.
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be the same as the server define.
SIP Plug and Play (PnP)	
Enable SIP PnP	If checked, the phone sends a SIP SUBSCRIBE message with the broadcast method. Any server that can support this feature response and sends a notification with the URL to the phone. The phone could get the configuration file with the URL.
Server Address	Broadcast address. The default is 224.0.0.0.
Server Port	PnP port
Transport Protocol	PnP protocol. The options are UDP or TCP.
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 the field is empty, the phone requests the common file and the device file, which is named as its MAC address. The filename 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. The default is one (1), which means the phone checks the update every 1 hour.
Update Mode	Provision Mode. The options are Disabled, Update After Reboot, and Update at Time Interval.
Autoprovision Now	Click to begin the auto provisioning process.
TR069	
Enable TR069	If checked, enables TR069.

Parameters	Description
ACS Server Type	There are the following options: China Telecom, Common, China, and eSight.
ACS Server URL	ACS server address.
ACS User	ACS server username (up to is 59 characters).
ACS Password	ACS server password (up to is 59 characters).
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.
TLS Version	TLS version. The options are TLS 1.0, TLS 1.1, TLS 1.2.
INFORM Sending Period	INFORM signal interval time. It ranges from 1 to 999999 seconds.
STUN Server Address	Configure STUN server address.
STUN Enable	If checked, enables STUN server for TR069
Apply	Click to save.

Firmware Upgrade

Phone Interface

To change upgrade the firmware:

- Press Menu > Advanced setting > Firmware Upgrade.

Web Interface

To change upgrade the firmware:

1. Login to the web interface.

The System > Information screen appears.

2. Go to the web page System > Upgrade.

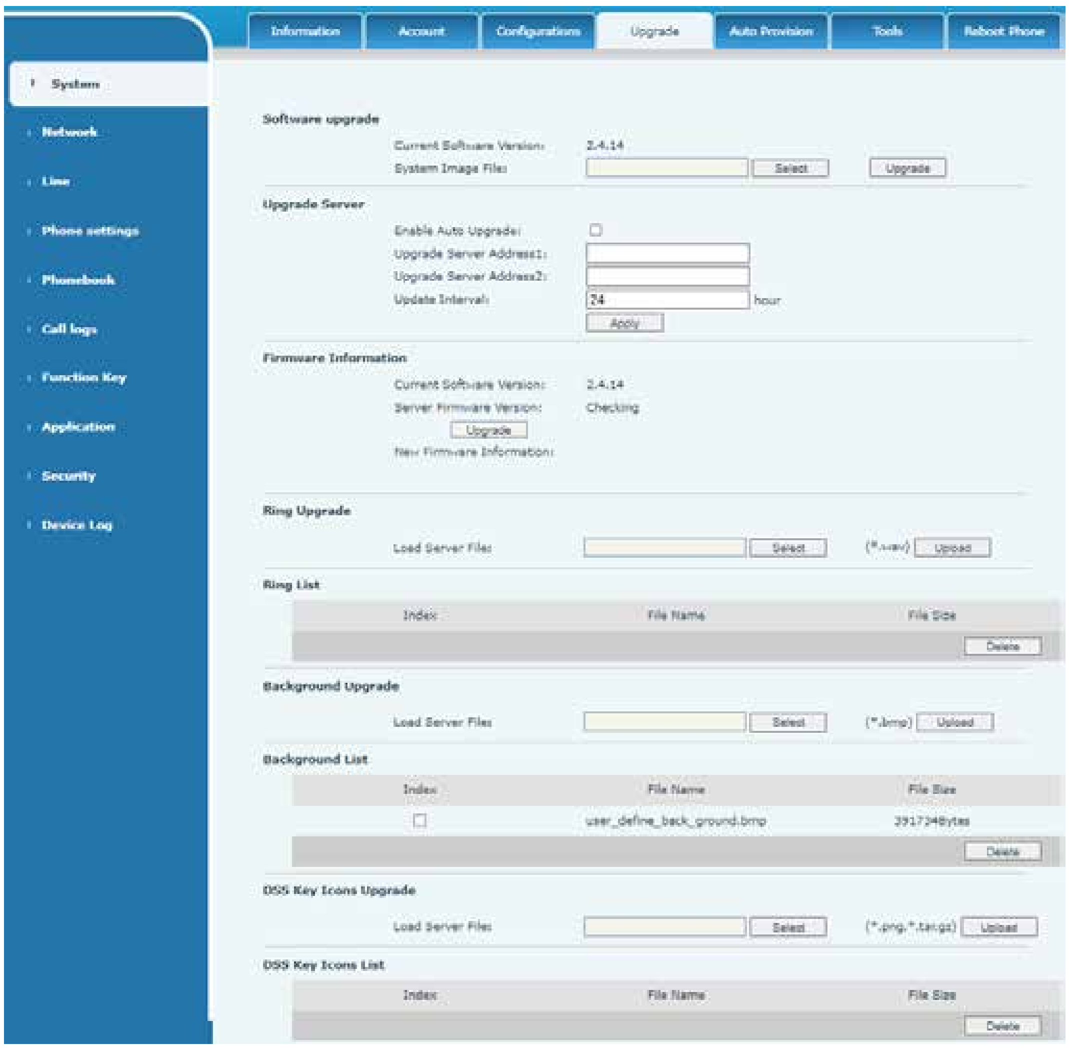


Table 21 System > Upgrade Parameters

Parameters	Description
Software upgrade	
Current Software Version	Displays the current software version.
System Image File	Click the Select button to select the system image file.
Upgrade	Click to upgrade the software.

Parameters	Description
Upgrade server	
Enable Auto Upgrade	If checked, the phone automatically upgrades the software version. After the update, the phone displays an upgrade message.
Upgrade Server Address1	Type the upgrade server address.
Upgrade Server Address2	Type the upgrade server address.
Update Interval	Type in the Update Interval in hours.
Apply	Click to save.
Firmware Information	
Current Software Version	Displays the Current Software Version.
Server Firmware Version	Displays the Server Firmware Version.
Upgrade	If there is a new version txt and software firmware on the server, the screen displays version information, and the upgrade button becomes available. Click the Upgrade button to upgrade to the new firmware.
New Firmware Information	When there is a corresponding TXT file and version on the server side, the TXT and version information is displayed under the new version description information.
Ring Upgrade	Select the file. Click Upload to upgrade.
Load Server File	Select the file. Click Upload to upgrade.
Ring List	Displays the Index, File Name, and File Size. Click the Delete button to delete the selected file.
Background Upgrade	Select the file. Click Upload to upgrade.
Background List	Displays the Index, File Name, and File Size. Click the Delete button to delete the selected file.
DSS Key Icons Upgrade	Select this file. Click the Upgrade button to upgrade the selected file.
DSS Key Icons List	Displays the Index, File Name, and File Size. Click the Delete button to delete the selected file.

The file requested from the server is a TXT file called vendor_model_hw10.txt where hw10 is the hardware version number. All Spaces in the filename are replaced by underline.

The URL requested by the phone is HTTP:// server address/vendor_Model_hw10.txt. Place the new version and the requested file in the download directory of the HTTP server.

TXT file format must be UTF-8

vendor_model_hw10.TXT

where the file format is 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 the update cycle arrives, the phone prompts if the server has available files and versions. Click view to check the version information and upgrade.

Factory Reset

Phone Interface

To perform a factory reset on the phone:

1. Press Menu > Advanced setting.
2. Enter the password. (The default password is fedsig.)
3. Press Restore factory Settings to select the file to be cleared.
4. Press OK to clear. When you select clear configuration file and clear all, the phone restarts automatically after clearing.

Web Pages

Web Page Authentication

You can log in to the web page of the phone to manage phone information and operation of the phone. You must provide the correct username and password to log in.

System Web Pages

System > Information

You can get the following system information:

- Model
- Hardware Version
- Software Version
- Uptime
- Summary of the network status
- Network information including Network mode, MAC Address, IP, Subnet mask, and Default gateway
- SIP account status including SIP User, and SIP account status (Registered/Unapplied/Trying/Timeout)

System > Account

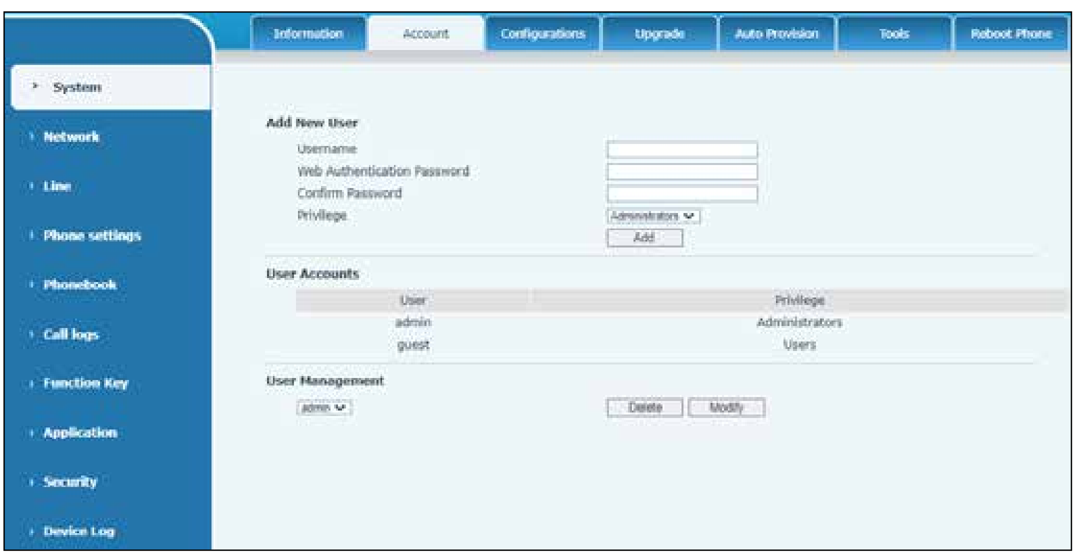
Web Interface

To configure new users:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page System > Account.



On this screen, you can change the password for a user.

If you have administrator's rights, you can add or delete users, manage users, and set permissions and passwords for new users.

System > Configurations

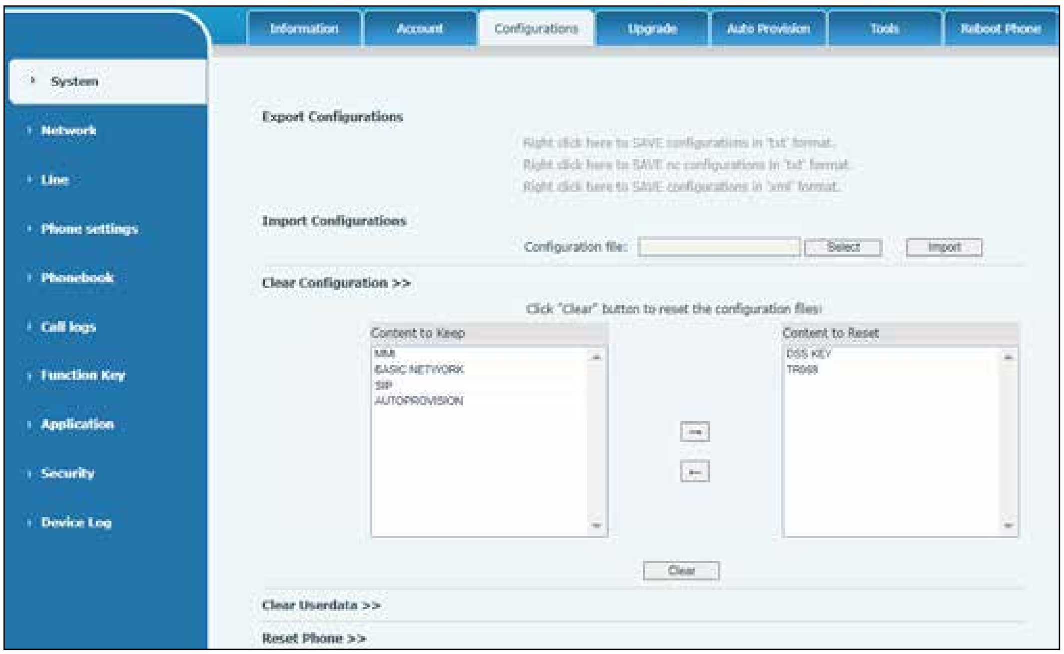
Web Interface

To add new users:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page System > Configurations.



If you have administrator's rights, you can view, export, or import the phone configuration or restore the phone to factory Settings.

Table 22 System > Configurations Parameters

Parameters	Description
Clear Configurations	Select the module in the configuration file to clear. SIP: account configuration. • AUTOPROVISION: automatically upgrades the configuration TR069:TR069 related configuration • MMI: MMI module, including authentication user information, web access protocol, etc. • DSS Key: DSS Key configuration
Clear Userdata	Select the local data table clear, all selected by default.
Reset Phone	Click the Reset button to clear the phone date, including configuration and database tables.

System > Upgrade

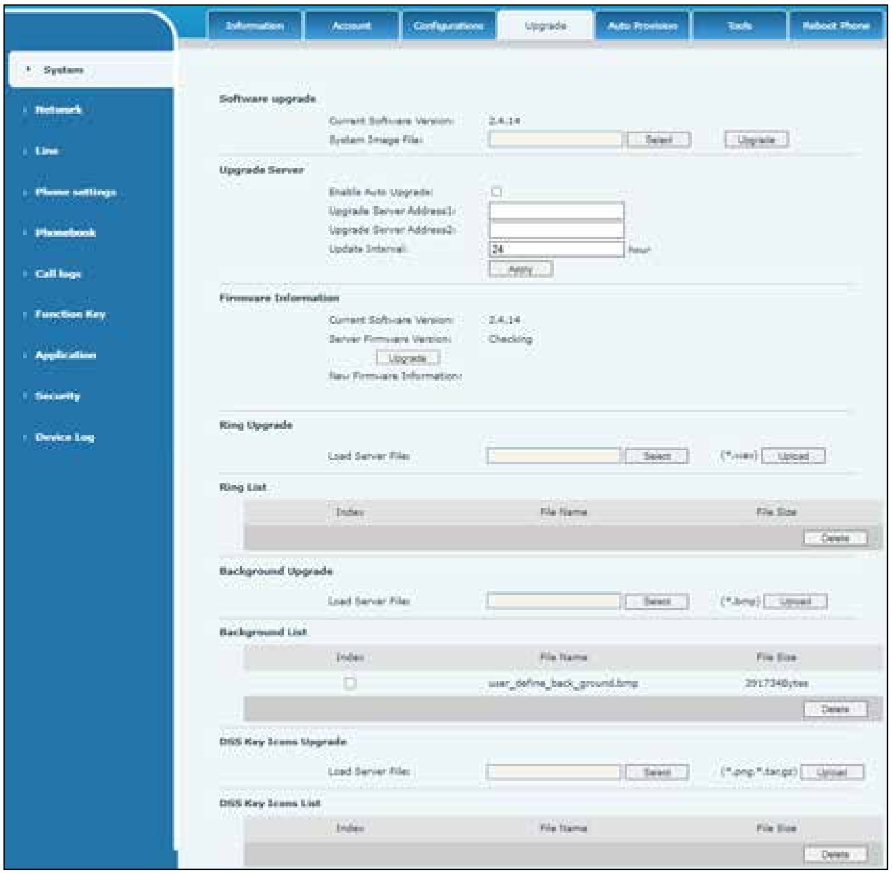
Web Interface

To upgrade the phone system:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page System > Upgrade.



Here you can upgrade the phone's software version, customize the ringtone, and upgrade the background, and DSS Key icon. Ring tone support .WAV format.

System > Auto Provision

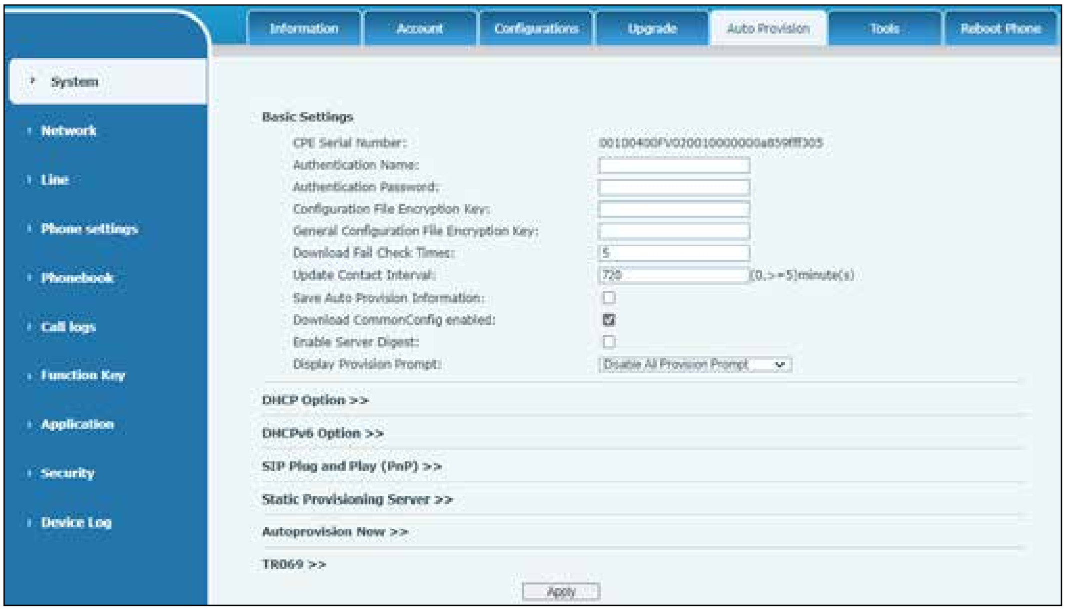
Web Interface

To deploy and manage the phone system in mass volume:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page System > Auto Provision.



System > Tools

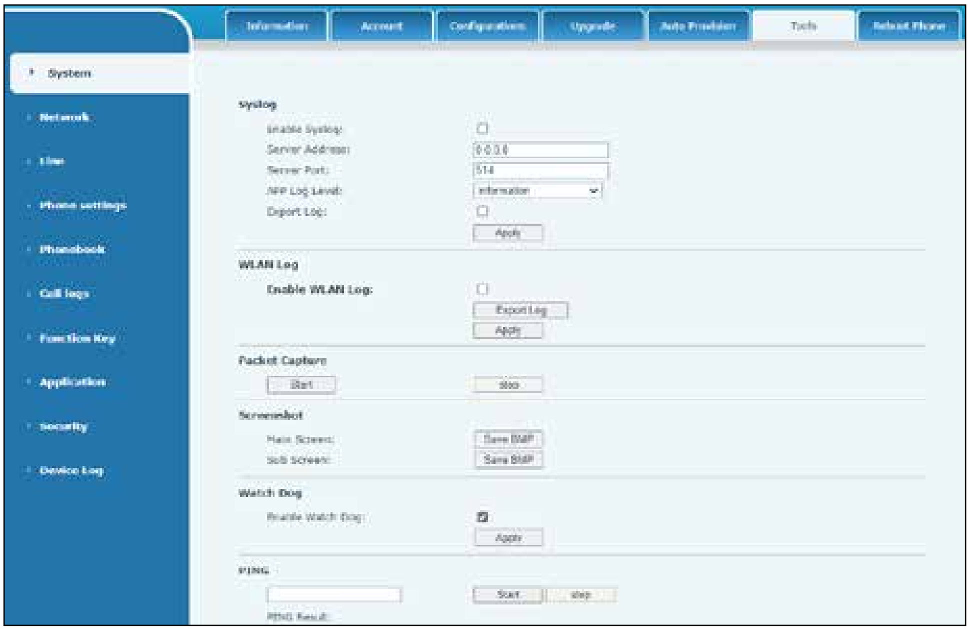
Web Interface

To troubleshoot the phone system:

1. Login to the web interface.

The System > Information screen appears.

2. Go to the web page System > Tools.



See “Troubleshooting” on page 151 for more detail.

System > Reboot Phone

Web Interface

To reboot the phone system:

1. Login to the web interface.

The System > Information screen appears.

2. Go to the web page System > Reboot Phone.

This page can restart the phone.

Network Web Pages

Network > Basic

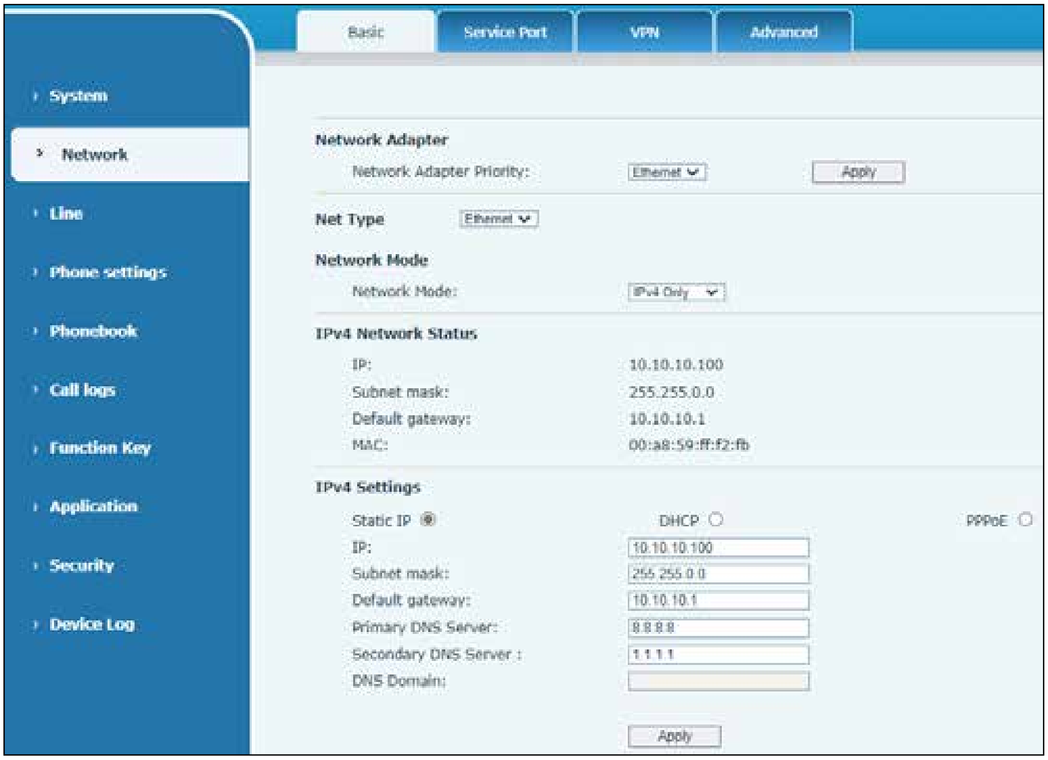
Web Interface

To configure network connection types and parameters.

1. Login to the web interface.

The System > Information screen appears.

2. Go to the web page Network > Basic.



Network > Service Port

Web Interface

To configure settings for Web page login protocol, protocol port settings, and RTP port:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Network > Service Port.

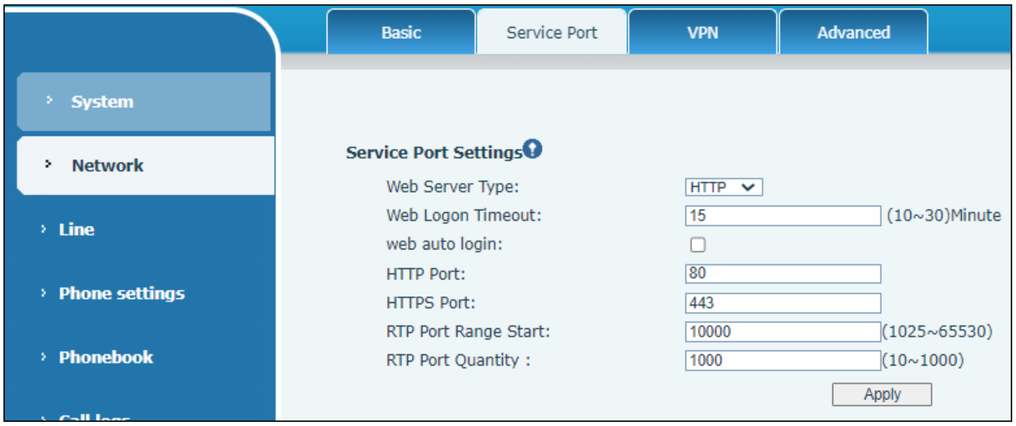


Table 23 Network > Service Port Parameters

Parameters	Description
Web Server Type	Reboot to take effect after settings. Optionally, the web page login is HTTP/HTTPS.
Web Logon Timeout	Automatically exits the login page when a user does not perform any action. The default is 15 minutes.
Web auto login	Automatically login to the web page after a timeout. The user does not need to enter a username and password.
HTTP Port	The default is 80. If you want system security, set ports other than 80, for example 8080, webpage login: HTTP://ip:8080.
HTTPS Port	The default is 443, the same as the HTTP port.
RTP Port Range Start	The value range is 1025 to 65535. The value of RTP port starts from the initial value set. For each call, the value of voice and video port is added 2.
RTP Port Quantity	Number of calls.

Network > VPN

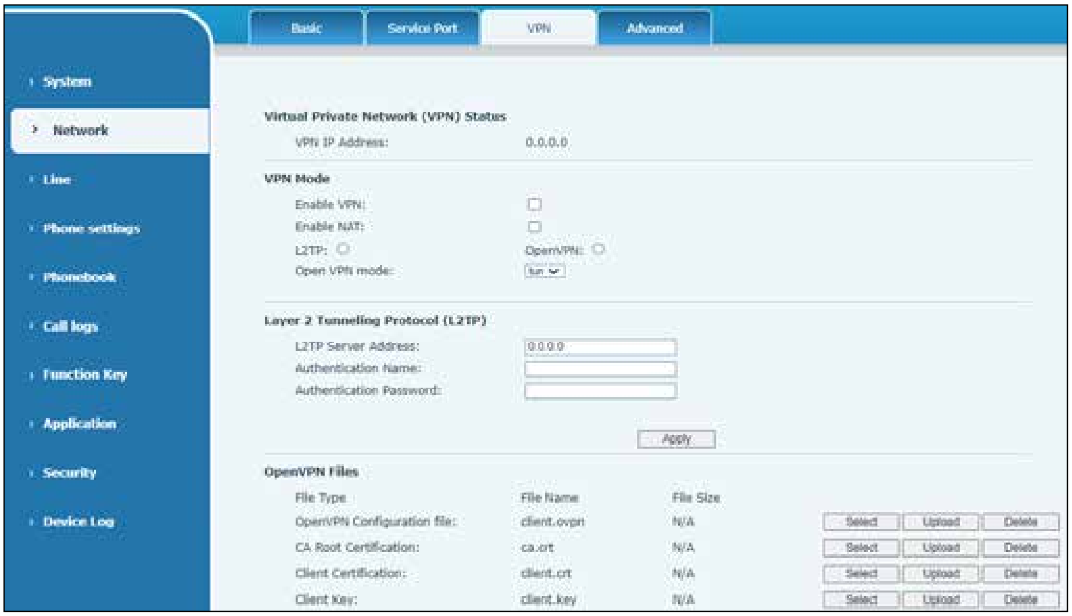
Web Interface

To configure a VPN connection:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Network > VPN.



See “Virtual Private Network (VPN)” on page 87 for more details.

Network > Advanced

Advanced network settings are typically configured by the IT Administrator to improve the quality of the phone service.

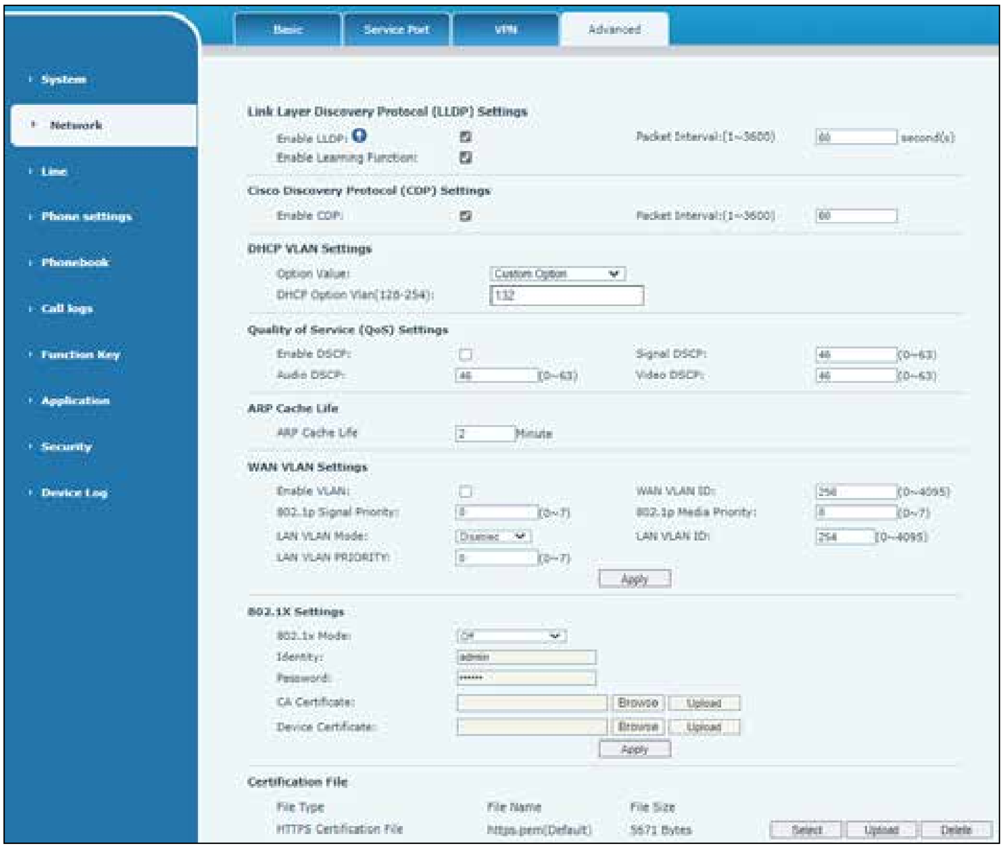
Web Interface

To configure the advanced settings: .

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Network > Advanced.



For configuration, see “Advanced” on page 86.

Line Web Pages

Line > SIP

Web Interface

To configure SIP settings:

1. Login to the web interface.

   The System > Information screen appears.

2. Go to the web page Line > SIP.

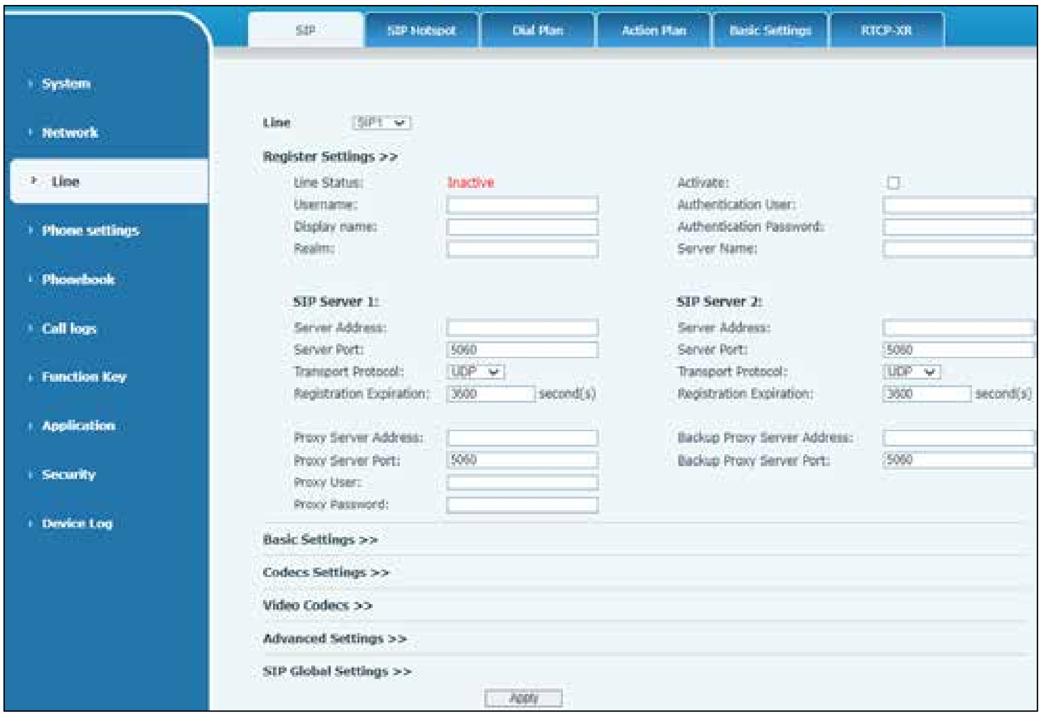


Table 24 Line > SIP > Register Settings Parameters

Parameters	Description
Line	Click the down arrow to select the SIP line.
Register Settings	
Line Status	Displays the current line status at page loading. To get the up-to-date line status, refresh the page.
Activate	Select the check box when 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	Enter server name.
SIP Server 1	
Server Address	Enter the IP or FQDN address of the SIP server.
Server Port	Enter the SIP server port. The default is 5060.

Parameters	Description
Transport Protocol	Set up the SIP transport line using UDP, TCP, or TLS.
Registration Expiration	Set SIP expiration date.
SIP Server 2	
Server Address	Enter the IP or FQDN address of the SIP server.
Server Port	Enter the SIP server port. The default is 5060.
Transport Protocol	Set up the SIP transport line using UDP, TCP, or TLS.
Registration Expiration	Set SIP expiration date.
Proxy Server Address	Enter the IP or FQDN address of the SIP proxy server.
Proxy Server Port	Enter the SIP proxy server port. The 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. The default is 5060.

Table 25 Line > SIP > Basic Settings Parameters

Parameters	Description
Basic Settings	
Enable Auto Answering	If checked, incoming calls are answered automatically after the delay time.
Auto Answering Delay	Enter the number in seconds of the delay for incoming call before the system automatically answers it.
Call Forward Unconditional	If checked, all incoming calls are forwarded to the number specified in the next field.
Call Forward Number for Unconditional	Redirects all calls to the number you specify here.
Call Forward on Busy	If checked, enable call forward on busy. When the phone is busy, any incoming call is forwarded to the number specified in the next field.
Call Forward Number for Busy	Redirects busy calls to the number you specify here.
Call Forward on No Answer	If checked, when an incoming call is not answered within the specified amount, the call is forwarded to the number specified in the next field.
Call Forward Number for No Answer	Redirects the call that was not answer to the number here.
Call Forward Delay for No Answer	Enter the number in seconds of the delay time if call is not answered before being forwarded.
Transfer Timeout	Enter the number in seconds that the call waits for a response. If the call is not answered, the transfer is canceled.
Conference Type	Click the arrow to select the type of call conference: • Local: sets up call conference by the device itself, maximum supports two remote parties. • Server: sets up call conference by dialing to a conference room on the server.
Server Conference Number	Enter the conference room number when conference type is set to be the server.

Parameters	Description
Subscribe For Voice Message	If checked, the device receives notification from the server if there is voice message waiting on the server.
Voice Message Number	Enter the number for retrieving voice message.
Voice Message Subscribe Period	Enter the number in seconds of the voice message notification subscription.
Enable Hotline	If checked, the phone dials to the specific number immediately at audio channel opened by off-hook handset or turns on the hands-free speaker or headphone.
Hotline Delay	Enter the amount in seconds the delay for hotline before the system automatically dials it.
Hotline Number	Enter the hotline dialing number.
Dial Without Registered	If checked, activates the call out by proxy without registration.
Enable Missed Call Log	If checked, the phone saves missed calls into the call history record.
DTMF Type	Click the down arrow to select the DTMF type to be used for the line: In-band, RFC2833, SIP_INFO, AUTO, and RFC2833+SIP_INFO.
DTMF SIP INFO Mode	Click the down arrow to set the SIP INFO mode to send: 10, 11, * or #.
Request With Port	If checked, requests the defined DTMF type.
Enable DND	If checked, any incoming call to this line are rejected automatically.
Use STUN	If checked, the line uses STUN for NAT traversal.
Use VPN	If checked, the line uses VPN restrict route.
Enable Failback	If checked, the line switches to the primary server when it is available.
Signal Failback	If checked, in multiple proxy cases, whether to allow the invite/register request to also execute failback.
Failback Interval	Enter the amount in seconds a register message is used to periodically detect the time interval for the availability of the main proxy.
Signal Retry Counts	Enter the number of attempts that the SIP request considers proxy unavailable under multiple proxy scenarios.

Table 26 Line > SIP > Codecs Settings Parameters

Parameters	Description
Codecs Settings	Set the priority and availability of the codecs by adding or remove them from the list.

Table 27 Line > SIP > Video Codecs Parameters

Parameters	Description
Video Codecs	Select video code to preview video.

Table 28 Line > SIP > Advanced Settings Parameters

Parameters	Description
Advanced Settings	
Use Feature Code	If checked, the features in this section are handled by the server and not the phone. To control the enabling of the features, the phone sends a feature code to the server by dialing the number specified in each feature code field.
Enable DND	Enter the feature code to dial to the server.
DND Disabled	Enter the feature code to dial to the server.
Enable Call Forward Unconditional	Enter the feature code to dial to the server.
Disable Call Forward Unconditional	Enter the feature code to dial to the server.
Enable Call Forward on Busy	Enter the feature code to dial to the server.
Disable Call Forward on Busy	Enter the feature code to dial to the server.
Enable Call Forward on No Answer	Enter the feature code to dial to the server.
Disable Call Forward on No Answer	Enter the feature code to dial to the server.
Enable Blocking Anonymous Call	Enter the feature code to dial to the server.
Disable Blocking Anonymous Call	Enter the feature code to dial to the server.
Call Waiting On Code	Enter the feature code to dial to the server.
Call Waiting Off Code	Enter the feature code to dial to the server.
Send Anonymous On Code	Enter the feature code to dial to the server.
Send Anonymous Off Code	Enter the feature code to dial to the server.
Enable Session Timer	If checked, enables call ending by session timer refreshment. The call session is ended if there is not a new session timer event update received after the timeout period.
Session Timeout	Enter in seconds the session timer timeout period.
Enable BLF List	If checked, enables the BLF List.
BLF List Number	Enter the number that allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Response Single Codec	If checked, the device uses a single codec in response to an incoming call request.
BLF Server	Enter the registered server that receives the subscription package from ordinary application of BLF phone. Enter the BLF server, if the sever does not support subscription package, the registered server and subscription server are separated.
Keep Alive Type	Click the down arrow to select either Disabled, UDP, or SIP Option packet to keep NAT pinhole opened.
Keep Alive Interval	Enter in seconds the keep alive packet transmitting interval.
Keep Authentication	If checked, keeps the authentication parameters from previous authentication.

Parameters	Description
Blocking Anonymous Call	If checked, rejects any incoming call without presenting caller ID.
RTP Encryption (SRTP)	Click the down arrow to enable RTP encryption such that RTP transmission will be encrypted.
Enable OSRTP	If checked, enables OSRTP.
Proxy Require	Type the proxy information.
User Agent	Type the user agent information. The default is the model with the Software Version.
Specific Server Type	Click the down arrow to collaborate with specific server type.
SIP Version	Click the down arrow to select the SIP version.
Anonymous Call Standard	Click the down arrow to select the standard to be used for anonymous.
Local Port	Enter the local port.
Ring Type	Click the down arrow to select the ring tone type for the line.
Enable user=phone	If checked, sets user=phone in SIP messages.
Use Tel Call	If checked, sets use tel call.
Auto TCP	If checked, using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes.
Enable PRACK	If checked, the line supports PRACK SIP messages.
Enable Rport	If checked, the line adds rport in SIP headers.
DNS Mode	Click the down arrow to select DNS mode: A, SRV, or NAPTR.
Enable Long Contact	If checked, allows more parameters in contact field per RFC 3840.
Enable Strict Proxy	If checked, it enables the use of strict routing. When the phone receives packets from the server, it uses the source IP address, not the address in via field.
Convert URI	If checked, converts not digit and alphabet characters to %hh hex code.
Use Quote in Display Name	If checked, adds a quote in display name.
Enable GRUU	If checked, supports Globally Routable User-Agent URI (GRUU).
Sync Clock Time	If checked, Time Sync with server.
Enable Inactive Hold	If checked, with the post-call hold capture package enabled, you can see that in the INVITE package, SDP is inactive.
Caller ID Header	Click the down arrow to set the Caller ID Header.
Use 182 Response for Call waiting	If checked, sets the device to use 182 response code at call waiting response.
Enable Feature Sync	If checked, Feature Sync with the server.
Enable SCA	If checked, enables SCA (Shared Call Appearance).

Parameters	Description
TLS Version	Click the down arrow to select the TLS version.
uaCSTA Number	Enter the uaCSTA number.
Enable Preview	If checked, enables preview.
Preview Mode	Click the down arrow to select either Preview18x or Preview2xx.
Enable Click To Talk	If checked, with the use of special server, calls out directly after enabling.
Enable Changeport	If checked, allows changeport.
VQ Name	Enter the server parameters.
VQ Server	Enter the server parameters.
VQ Server Port	Enter the server parameters.
VQ Http/Https server	Enter the server parameters
Flash mode	Click the down arrow to select either Normal or SIP Info.
Flash Info Content-Type	Enter the SIP info content type.
Flash Info Content-Body	Enter the SIP info content body.
Server Expire	If checked, sets the timeout to use the server.
Unregister On Boot	If checked, enables the logout function.
Enable MAC Header	If checked, when opening the registration, are IP package and user agent with MAC.
BLF Dialog Strict Match	If checked, enables accurate matching of BLF sessions.
PTime(ms)	Click the down arrow to select whether to bring ptime field. The default is Disabled.
Enable Deal 180	If checked, allows Deal 180.
Transaction Timer T1	Enter in milliseconds the transaction time.
Transaction Timer T2	Enter in milliseconds the transaction time.
Transaction Timer T4	Enter in milliseconds the transaction time.
CallPark Number	Enter the CallPark number.
PickUp Number	Enter the scramble number when the Pickup is enabled.
JoinCall Number	Enter the JoinCall Number.
Intercom Number	Enter the Intercom Number.
Retrieve Type	Click the down arrow to select either Auto, Invite Replace, or Park Number.
Retrieve Number	Enter the retrieve number.

Table 29 Line> SIP > SIP Global Settings Parameters

Parameters	Description
SIP Global Settings	
Strict Branch	If checked, strictly matches the Branch field.
Enable Group	If checked, sets to open group.
Enable RFC4475	If checked, enables RFC4475.
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Enter in seconds the registration failure retry time.
Local SIP Port	Modify the phone SIP port.
Enable uaCSTA	If checked, enables uaCSTA.
Apply	Click to save.

Line > SIP Hotspot

See “SIP Hotspot” on page 71.

Line > Dial Plan

Go to the web page Line > Dial Plan.

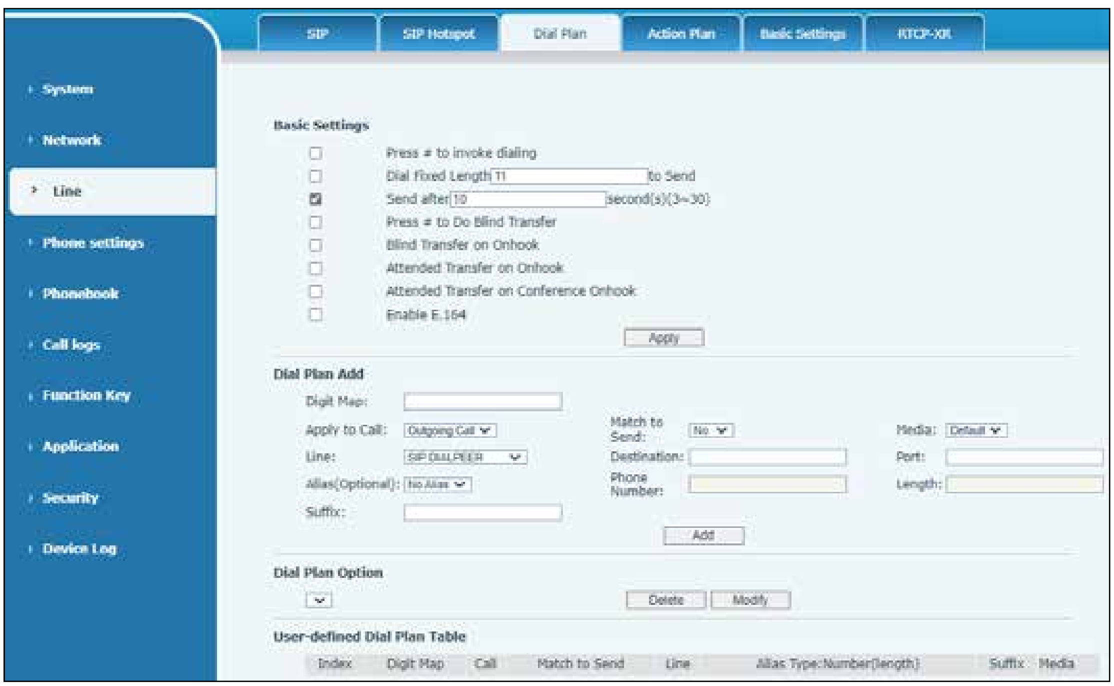


Table 30 Line > Dial Plan > Basic Settings Parameters

Parameters	Description
Press # to invoke dialing	If checked, the user dials the other party's number and then adds the # number to dial out.
Dial Fixed Length	Enter the number that is automatically dialed out when it reaches a fixed length.
Send after	Enter the number in seconds to delay.
Press # to Do Blind Transfer	If checked, the user enters the number to be transferred and then presses the "#" key to transfer the current call to a third party.

Parameters	Description
Blind Transfer on Onhook	If checked, after the user enters the number, he hangs up or turns off the hands-free function to transfer the current call to a third party.
Attended Transfer on Onhook	If checked, hang up 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 Onhook	If checked, during a three-way call, hang up and the remaining two parties remain on the call.
Enable E.164	See E.164 standard specification.
Apply	Click to save.

Dial Plan Add

There are two types of matching: Full Matching and Prefix Matching. In Full Matching, the phone number is entered and mapped per the Dial Peer rules. In Prefix Matching, only part of the number is entered, followed by T. The mapping takes 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.

Table 31 Line > Dial Plan > Dial Plan Add Parameters

Parameters	Description
Destination	Type the destination address. This is for IP direct.
Port	Type the signal port. The default is 5060 for SIP.
Alias (Optional)	Click the down arrow to set the Alias. Select either No Alias, All, Add, Delete, or Replace. This is the text to be added, deleted, or replaced. It is an optional item. • All: xxx-xxx replaces the phone number. • Add: xxx–xxx is dialed before any phone number. • Delete: The characters is deleted from the phone number. • Replace: xxx–xxx is substituted for the specified characters.
Length	Set the number of characters to be deleted. For example, if this is set to 3, the phone deletes the first 3 digits of the phone number. It is an optional item.
Suffix	Characters to be added at the end of the phone number. It is an optional item.

User-defined Dial Plan Examples

The User-defined Dial Plan Table option allows you to create rules to make dialing easier. There are several different options for dial rules.

Example 1

All Substitution — Assume that it can make a direct IP call to IP address 172.168.2.208. Using this feature, 123 can be substituted for 172.168.2.208.

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.

Example 3

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

- x — Matches any single digit that is dialed.
- [] — Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.

Line > Action Plan

When an IP phone calls a phone, the bound IP camera synchronously transmits video to the other phone (video is supported).

When SIP calls, multicast calls, or intercom calls are made, the device converts calls that conform to the number rules into group calls.

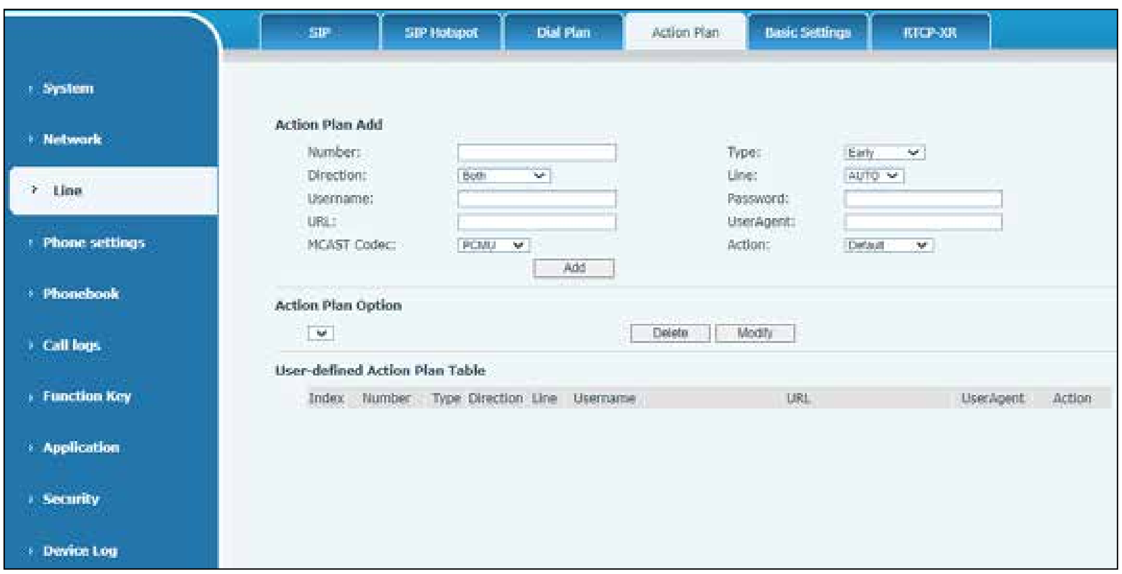


Table 32 Line > Action Plan > Action Plan Add Parameters

Parameters	Description
Number	Enter the auxiliary phone number (supports video).
Type	Click the down arrow to select either Early or Connected. Supports video display on call.
Direction	Click the down arrow to select Both, Outgoing Call, or Incoming Call. For call mode, incoming or outgoing calls display video.
Line	Click this down arrow to set up outgoing lines.
Username	Bind the username of the IP camera.
Password	Bind IP camera password.
URL	Video streaming information;Mcast Address (mcast://IP:port)

Parameters	Description
User Agent	Enter the user agent information

Line > Basic Settings

Set up the register global configuration.

Table 33 Line > Basic Settings Parameters

Parameters	Description
STUN Settings	
Server Address	Enter the STUN server address.
Server Port	Enter the STUN server port. The default is 3478.
Binding Period	Enter the STUN binding period which can be used to keep the NAT pinhole opened.
SIP Waiting Time	Enter the timeout of STUN binding before sending SIP messages.
The TLS authentication	
TLS Certification File	Upload or delete the TLS certification file used for encrypted SIP transmission.

Line > RTCP-XR

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

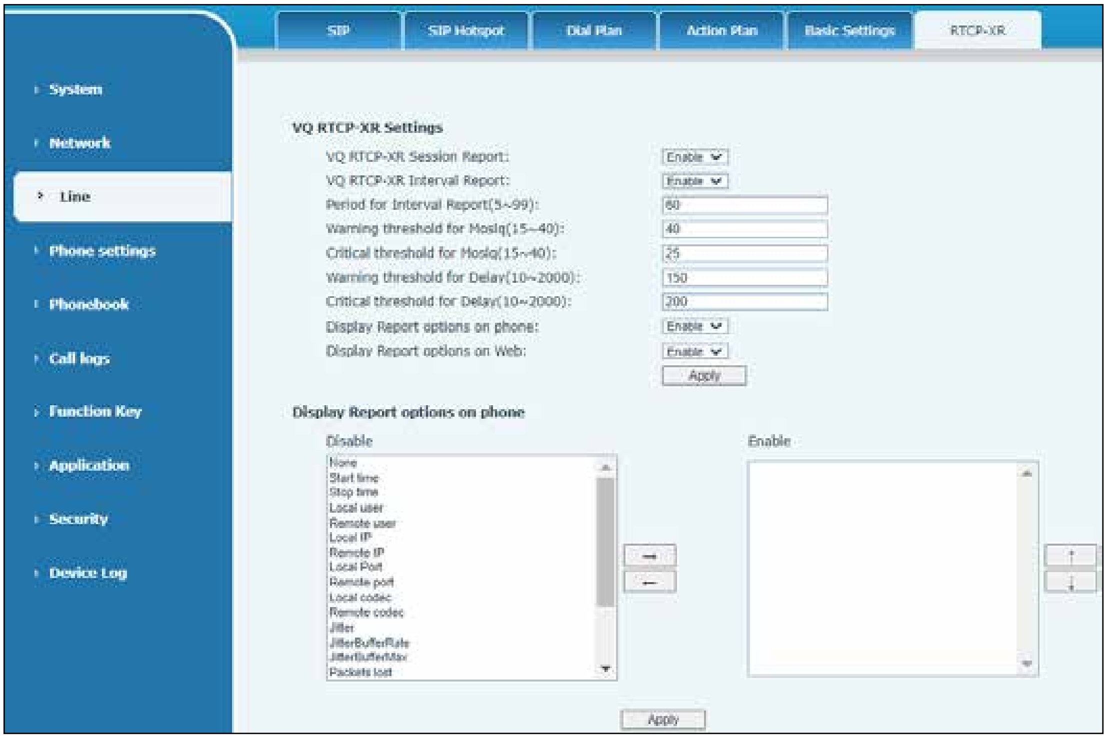
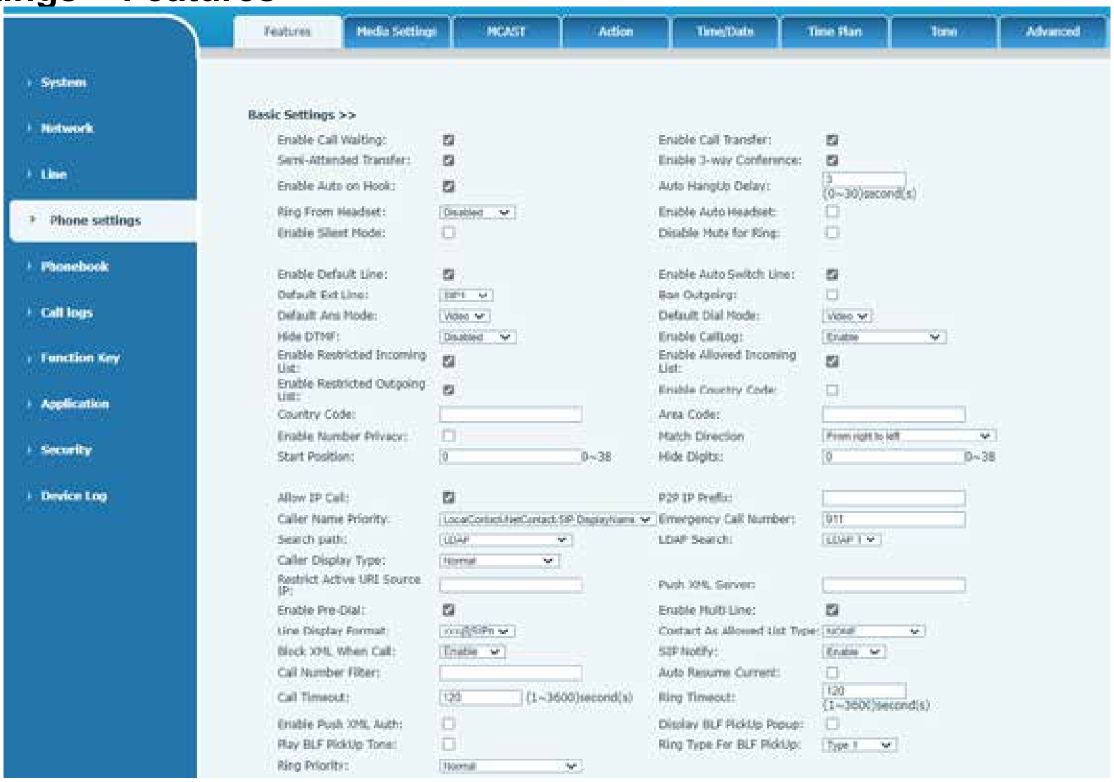


Table 34 Line > RTCP-XR Parameters

Parameters	Description
VQ RTCP-XR Settings	
VQ RTCP-XR Session Report	Click the down arrow to select either Disable or Enable the VQ report.
VQ RTCP-XR Interval Report	Click the down arrow to select either Disable or Enable the Interval mode.
Period for Interval Report (5~99)	Enter the time interval that VQ reports are sent.
Warning threshold for Moslq (15~40)	When the phone calculated the Moslq value x10 below the set threshold, a warning was issued.
Critical threshold for Moslq (15~40)	When the phone calculates the Moslq value x10 below the set threshold, the critical report is issued.
Warning Threshold for Delay (10~2000)	When the one-way delay of the phone is greater than the set threshold, a warning is issued.
Critical Threshold for Delay (10~2000)	When the phone computes that the one-way delay is greater than the set threshold, the critical report is issued.
Display Report options on Web	Click the down arrow to select either Disable or Enable the VQ report data for the last call through the web page.

Phone Settings Web Pages

Phone settings > Features



Basic Settings

Table 35 Phone settings > Features > Basic Settings Parameters

Parameters	**Description**
Basic Settings	
Enable Call Waiting	If checked, allows you to take second incoming call during an established call. Default enabled.
Enable Call Transfer	If checked, enables you to transfer a call.
Semi-Attended Transfer	If checked, enables you to semi attend a transfer call.
Enable 3-way Conference	If checked, enables three-way conference call.
Enable Auto on Hook	If checked, the phone hangs up and returns to the idle automatically at hands-free mode.
Auto HangUp Delay	Enter the Auto Onhook time. The phone hangs up and returns to the idle automatically after Auto Hand down time at hands-free mode, and play dial tone Auto Onhook time at handset mode.
Ring From Headset	Click the down arrow to select Disabled, Enable, or Group Ring. When Enable is selected, the phone plays ring tone from the handset.
Enable Auto Headset	If checked, when headsets are plugged in the phone, you press the answer key or line key to answer a call with the headset automatically.
Enable Silent Mode	If checked, the phone is muted, the phone does not ring. Use the volume keys and mute key to unmute.

Parameters	Description
Disable Mute for Ring	If checked, you cannot mute the phone.
Enable Default Line	If checked, you can assign default SIP line for dialing out rather than SIP1.
Enable Auto Switch Line	If checked, enables phone to select an available SIP line as default automatically.
Default Ext Line	Click the down arrow to select the default line to use for outgoing calls.
Ban Outgoing	If checked, you cannot dial out any number.
Default Ans Mode	Click the down arrow to select either Audio or Video.
Default Dial Mode	Click the down arrow to select either Audio or Video.
Hide DTMF	Click the down arrow to select Disabled, All, Delay, Last Show, or Display Null.
Enable CallLog	Click the down arrow to select Disabled, Enable, Save Updated CallLog.
Enable Restricted Incoming List	If checked, restricts incoming call list.
Enable Allowed Incoming List	If checked, allows the call list.
Enable Restricted Outgoing List	If checked, restricts outgoing list.
Enable Country Code	If checked, the country code is enabled.
Country Code	Enter the country code.
Area Code	Enter the area code.
Enable Number Privacy	If checked, enables number privacy.
Match Direction	Click the down arrow to select either From left to right or From right to left.
Start Position	Enter the 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 checked, the user can dial out with the IP address.
P2P IP Prefix	Enter the prefix a point-to-point IP call.
Caller Name Priority	Click the down arrow to change caller ID display priority.
Emergency Call Number	Enter the emergency call number. The default is 911.
Search path	Click the down arrow to select the search path.
LDAP Search	Click the down arrow to select from one of the LDAP.
Caller Display Type	Click the down arrow to select either Display Name Priority, Display Number Only, Display Blank, or Normal.
Restrict Active URI Source IP	Enter the device to accept the active URI command from a specific IP address.
Push XML Server	Enter the Push XML Server. When the phone receives a request, it determines whether to display the corresponding content on the phone sent by the specified server or not.
Enable Pre-Dial	If checked, the user enter the number without opening an audio channel.
Enable Multi Line	If checked, up to 10 simultaneous calls can exist on the phone. If unchecked, up to two simultaneous calls can exist on the phone.

Parameters	Description
Line Display Format	Click the down arrow to select the custom line format: SIPn, SIPn: xxx, or xxx@SIPn.
Contact As Allowed List Type	Click the down arrow to select NONE, BOTH, DND Allowed Llist, or FWD Allowed List.
Block XML When Call	Click the down arrow to Disable or Enable XML push on call.
SIP notify	Click the down arrow to Disable or Enable. When enabled, the phone displays the information when it receives the relevant notification content.
Call Number Filter	Enter the number the phone will look for.
Auto Resume Current	If checked, if the SIP line drops, the phone will auto resume.
Call Timeout	Enter in seconds the amount of time the call will auto resume.
Ring Timeout	Enter in seconds the amount of time you want the phone to ring before it stops.
Enable Push XML Auth	If checked, it allows XML pushes.
Display BLF PickUp Popup	If checked, it allows BLF PickUp Popup to be displayed.
Play BLF PickUp Tone	If checked, the phone will display and play a tone.
Ring Type For BLF PickUp	Click the down arrow to select the ring type for your BLF pickup.
Ring Priority	Click the down arrow to select either Normal, Priority, or Auto Answer When Dialing.

Tone Settings

Table 36 Phone settings > Features > Tone Settings Parameters

Parameters	Description
Enable Holding Tone	If checked, a tone plays when the call is held.
Enable Call Waiting Tone	If checked, a tone plays when a call is waiting.
Play Dialing DTMF Tone	If checked, the phone plays a DTMF tone when the user presses a phone digit at dialing. The default is enabled.
Play Talking DTMF Tone	If checked, the phone plays a DTMF tone when the user presses a phone digit during taking. The default is enabled.

DND Settings

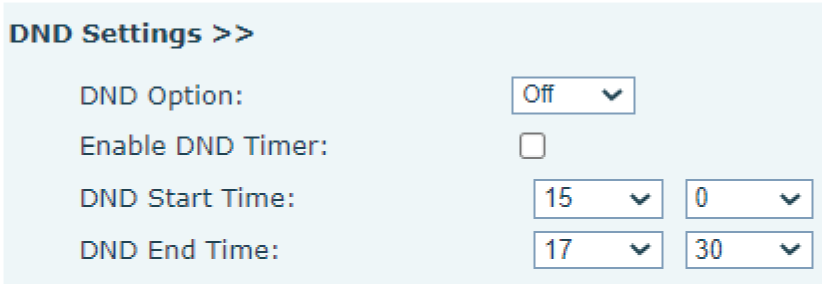


Table 37 Phone settings > Features > DND Settings Parameters

Parameters	Description
DND Option	Click the down arrow to select Off, Phone, or Line.
Enable DND Timer	If checked, the DND is automatically turned on from the start time to the off time.
DND Start Time	Click the down arrow to set the DND Start Time.
DND End Time	Click the down arrow to set the DND End Time.

Intercom Settings

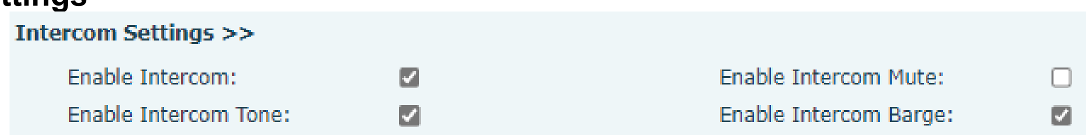


Table 38 Phone settings > Features > Intercom Settings Parameters

Parameters	Description
Enable Intercom	If checked, the phone accepts the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after the specific delay.
Enable Intercom Mute	If checked, enables the mute mode during the intercom call.
Enable Intercom Tone	If checked, the phone plays the intercom tone.
Enable Intercom Barge	If checked, the phone auto answers the intercom call during a call. If the current call is a intercom call, the phone rejects the second intercom call.

Redial Settings

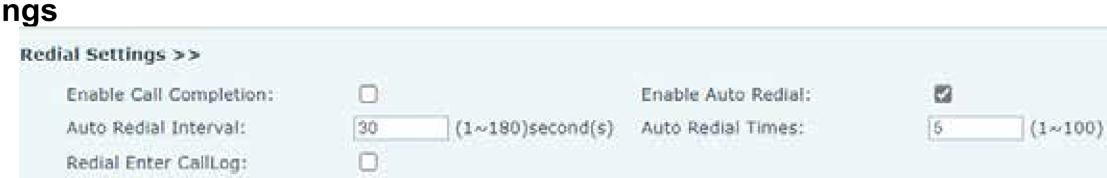


Table 39 Phone settings > Features > Redial Settings Parameters

Parameters	Description
Enable Call Completion	If checked, it gives a notification when a line is no longer busy.
Enable Auto Redial	If checked, the line keeps redialing.
Auto Redial Interval	Enter the amount of time in seconds that the call waits to redial.
Auto Redial Times	Enter the amount of time that the call will redial.

Parameters	Description
Redial Enter CallLog	If checked, the phone automatically dials a specific number.

Response Code Settings

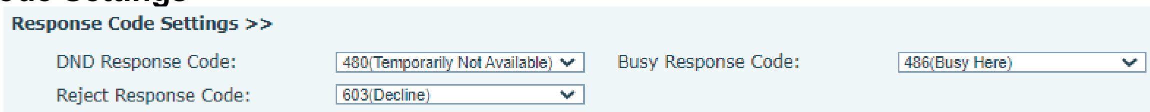


Table 40 Phone settings > Features > Code Settings Parameters

Parameters	Description
DND Response Code	Click the down arrow to set the SIP response code on a call rejection on a DND.
Busy Response Code	Click the down arrow to set the SIP response code on a busy line.
Reject Response Code	Click the down arrow to set the SIP response code on a call rejection.

Password Dial Settings

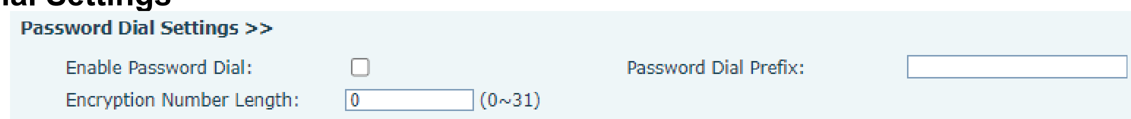


Table 41 Phone settings > Features > Dial Settings Parameters

Parameters	Description
Enable Password Dial	If checked, when the number entered begins with the password prefix, the following N numbers after the password prefix are hidden as *, where N stands for the value that you enter in the Password Length field. For example, if you set the password prefix to 3, enter the Password Length as 2, then enter the number 34567, it displays 3**67 on the phone.
Encryption Number Length	Enter the Encryption Number length.
Password Dial Prefix	Enter the prefix of the password call number.

Bluetooth Settings

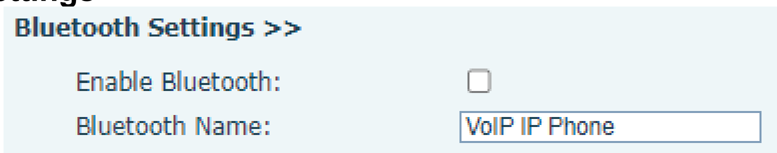


Table 42 Phone settings > Features > Bluetooth Settings Parameters

Parameters	**Description**
Enable Bluetooth	If checked, Bluetooth is on.
Bluetooth Name	Displays the Bluetooth name.

Power LED

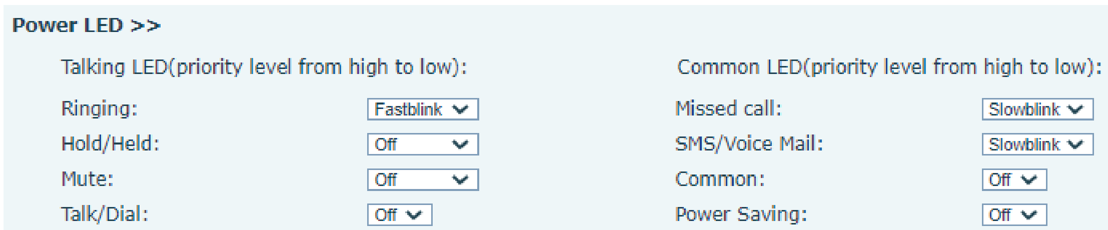


Table 43 Phone settings > Features > Power LED Parameters

Parameters	**Description**
Talking LED (priority level from high to low)	
Ringing	Click the down arrow to set the light when there is an incoming call set to off, on, slowblink, or fastblink.
Hold/Held	Click the down arrow to set the light to off, on, slowblink, or fastblink.
Mute	Click the down arrow to set the light in mute mode from off, on, slowblink, or fastblink.
Talk/Dial	Click the down arrow to set the light to either off or on. On is red bright, the default is off.
Common LED (priority level from high to low)	
Missed call	Click the down arrow to set the light when there is a missed call to off, on, fast blink, or slow blink.
SMS/Voice Mail	Click the down arrow to set the light when there is a unread voicemail transcription or voice mail. Select from off, on, fast blink, or slow blink.
SMS/MWI	The status of power lamp when there is unread short message/voice.
Common	Select from either on or off.
Power Saving	Select from either on or off.

DssKey Settings

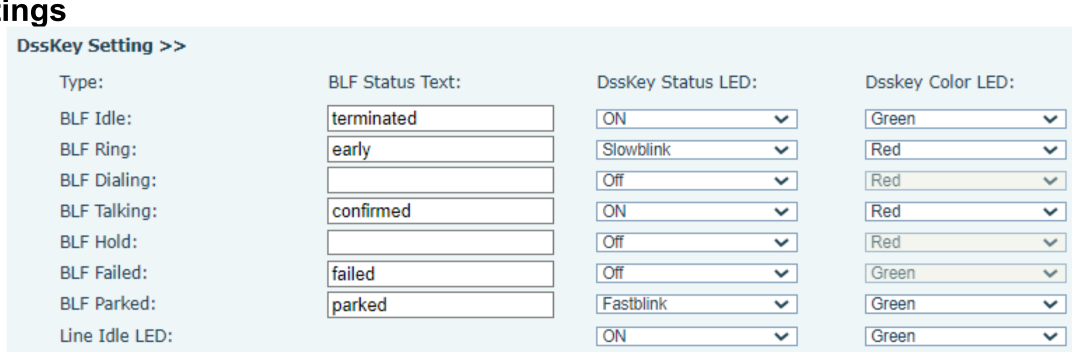


Table 44 Phone settings > Features > DssKey Settings Parameters

Parameters	Description
BLF Idle	Enter the BLF status text. Click the down arrow to set the light to either off, on, slow blink, or fast blink. Click the down arrow to set the color of the light to either Green, Red or Orange.
BLF Ring	Enter the BLF status text. Click the down arrow to set the light to either off, on, slow blink, or fast blink. Click the down arrow to set the color of the light to either Green, Red or Orange
BLF Dialing	Enter the BLF status text. Click the down arrow to set the light to either off, on, slow blink, or fast blink. Click the down arrow to set the color of the light to either Green, Red or Orange
BLF Talking	Enter the BLF status text. Click the down arrow to set the light to either off, on, slow blink, or fast blink. Click the down arrow to set the color of the light to either Green, Red or Orange
BLF Hold	Enter the BLF status text. Click the down arrow to set the light to either off, on, slow blink, or fast blink. Click the down arrow to set the color of the light to either Green, Red or Orange
BLF Failed	Enter the BLF status text. Click the down arrow to set the light to either off, on, slow blink, or fast blink. Click the down arrow to set the color of the light to either Green, Red or Orange
BLF Parked	Enter the BLF status text. Click the down arrow to set the light to either off, on, slow blink, or fast blink. Click the down arrow to set the color of the light to either Green, Red or Orange
Line Idle LED	Click the down arrow to set the light to either off, on, slow blink, or fast blink. Click the down arrow to set the color of the light to either Green, Red or Orange

Notification Popups

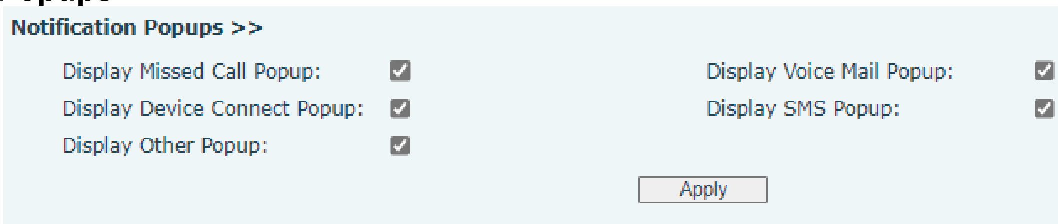
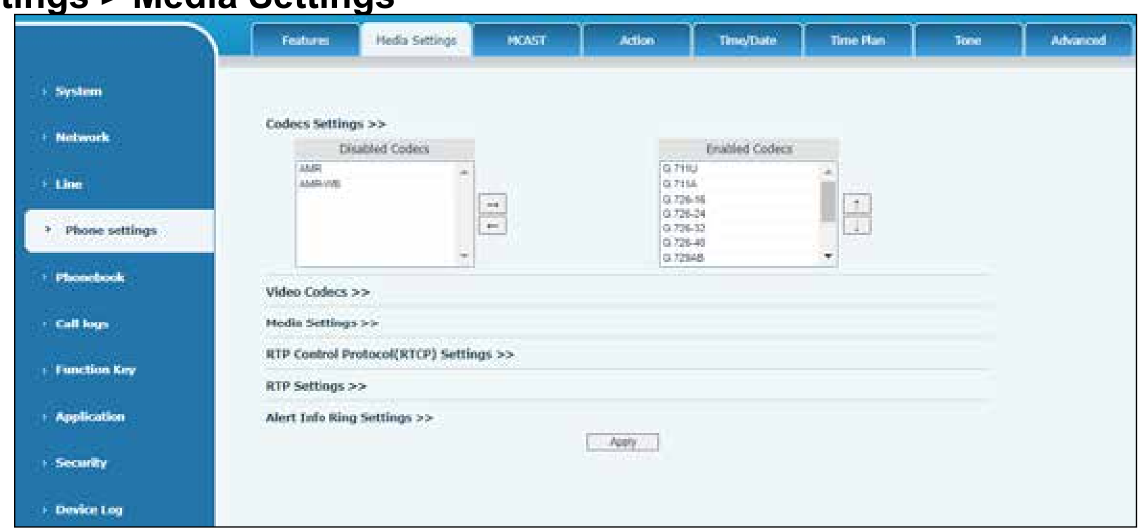


Table 45 Phone settings > Features > Notification Popups Parameters

Parameters	Description
Display Missed Call Popup	If checked, no incoming call popup prompt after opening, no popup prompt when closing, open by default.
Display Voice Mail Popup	If checked, a popup appears on the phone.
Display Device Connect Popup	If checked, the voice message popup prompt is not answered after opening, and it is opened by default if there is no popup prompt when closing.
Display SMS Popup	If checked, there is popup prompt for unread messages after opening, and there is no popup prompt when closing. It is opened by default.
Display Other Popup	If checked, when the handle is not hung back after opening, registration fails, IP acquisition fails, Tr069 connection fails, and other abnormalities, there is a popup prompt when it is opened; otherwise, there is no prompt when it is closed, and it will be opened by default.

Phone settings > Media Settings



Codes Settings

Table 46 Phone settings > Media Settings > Codecs Settings Parameters

Parameters	Description
Codecs Settings	Click the down arrow to select enable or disable voice encoding: G.711A/U, G.722, G.729, G.726-16, G726-24, G726-32, G.726-40, ILBC, Opus

Video Codecs

Table 47 Phone settings > Media Settings > Video Codecs Parameters

Parameters	Description
Video Codecs	Click the down arrow to select either None or H264.

Media Settings

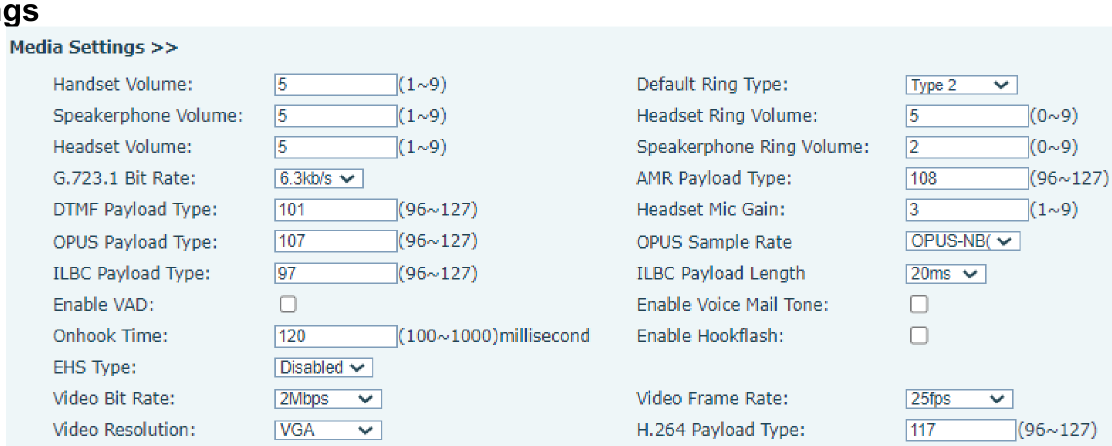


Table 48 Phone settings > Media Settings > Media Settings Parameters

Parameters	Description
Handset Volume	Enter the Handset volume from 1 to 9.
Default Ring Type	Click the down arrow to select the default ringtone. If a ringtone is not selected, the default ringtone is used.
Speakerphone Volume	Enter the hands-free volume from 1 to 9.
Headset Ring Volume	Enter the volume of the earphone ringtone from 1 to 9.
Headset Volume	Enter the volume of the headset from 1 to 9.
Speakerphone Ring Volume	Enter the volume of hands-free ringtone from 1 to 9.
G.723.1 Bit Rate	Click the down arrow to select either 5.3 kb/s or 6.3 kb/s.
AMR Payload Type	Enter the AMR load type. The value is 96 to 127.
DTMF Payload Type	Enter the DTMF payload type. The value is 96 to 127.
Headset Mic Gain	Enter the earphone's radio volume gain to fit different models of earphones.
OPUS playload type	Enter Opus load type. The value is 96 to 127.
OPUS Sample Rate	Click the down arrow to set the opus sampling rate, including opus-nb (8 KHz) and opus-wb (16 KHz).
ILBC Payload Type	Enter the ILBC Payload Type. The value is 96 to 127.
ILBC Payload Length	Click the down arrow to set the ILBC Payload Length: 20 ms or 30 ms.
Enable VAD	If checked, voice activity detection is enabled.
Enable Voice Mail Tone	If checked, when there is a new voice message message, the phone will start a special dial tone.
Onhook Time	Enter a minimum response time. The default is 200 ms.

Parameters	Description
Enable Hookflash	If checked, hook flash is enabled.
EHS Type	Click the down arrow to diable or enable the EHS headset.
Video Bit Rate	Click the down arrow to select the video bit rate.
Video Frame Rate	Click the down arrow to select the video frame rate.
Video Resolution	Click the down arrow to select the video resolution.
H264 Payload Type	Enter the payload type. The value is 96 to 127.

RTP Control Protocol (RTCP) Settings

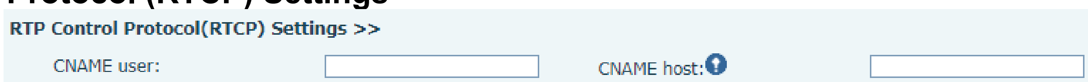


Table 49 Phone settings > Media Settings > RTP Control Protocol (RTCP) Settings

Parameters	Description
CNAME user	Enter the CNAME user.
CNAME host	Enter the CNAME host.

RTP Settings

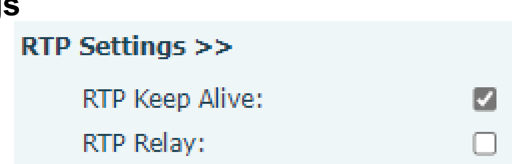


Table 50 Phone settings > Media Settings > RTP Settings Parameters

Parameters	Description
RTP keep alive	If checked, the phone holds the call and sends the packet after 30 seconds.
RTP relay	If checked, enables the RTP relay

Alert Info Ring Settings

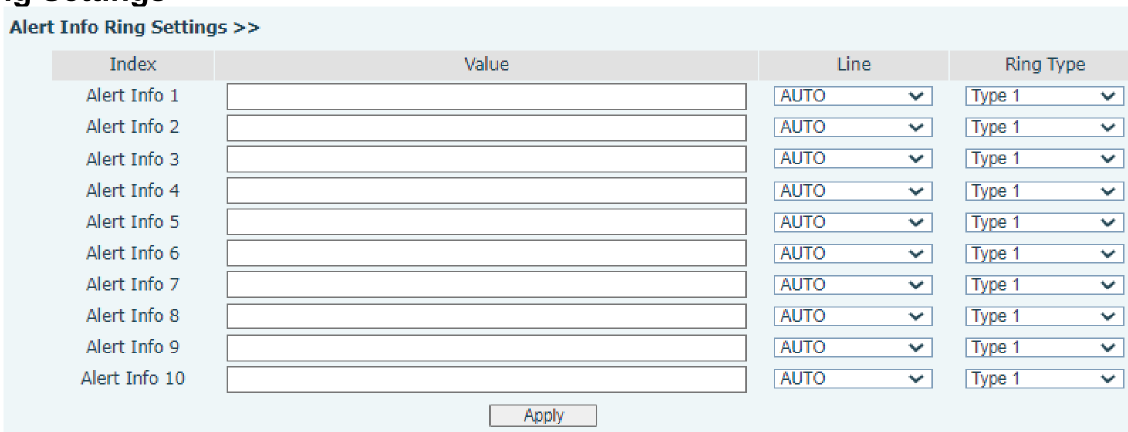


Table 51 Phone settings > Media Settings > Alert Infor Ring Settings Parameters

Parameters	**Description**
Value	Enter the value to specify the ring type.
Line	Click the down arrow to select the line.
Ring Type	Click the down arrow to select from Type1 to Type20, Mobile, or Auto.

Phone settings > MCAST

MCAST allows you to make a broadcast call to people who are in multicast groups. You can configure a multicast DSS Key on the phone, which allows you 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 a pre-configured multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.

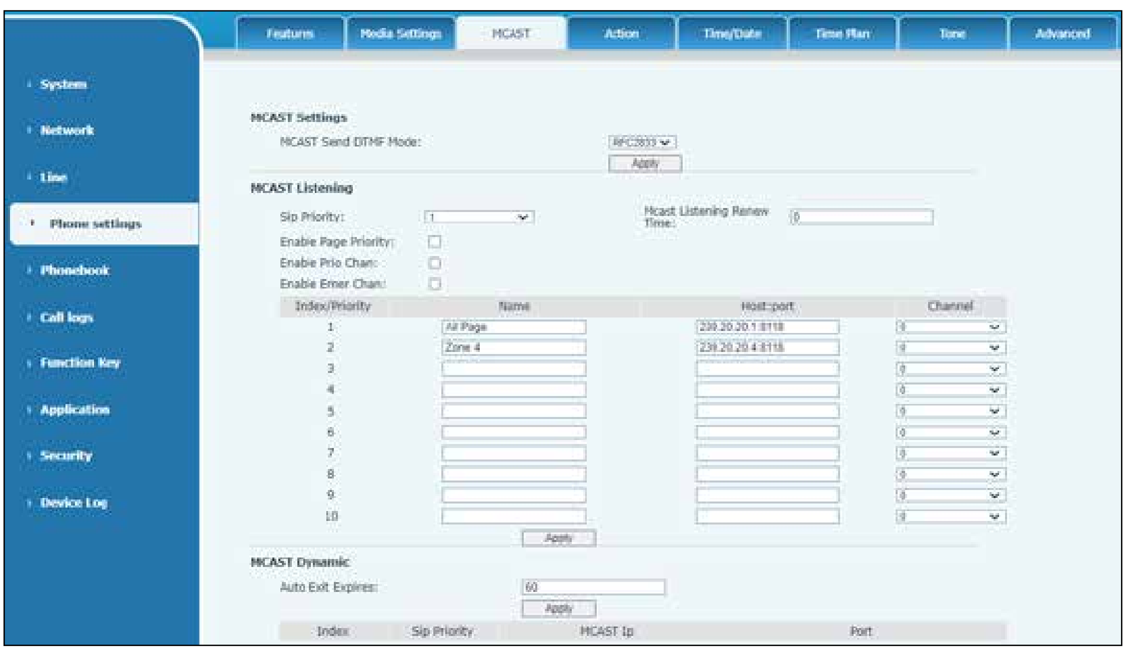
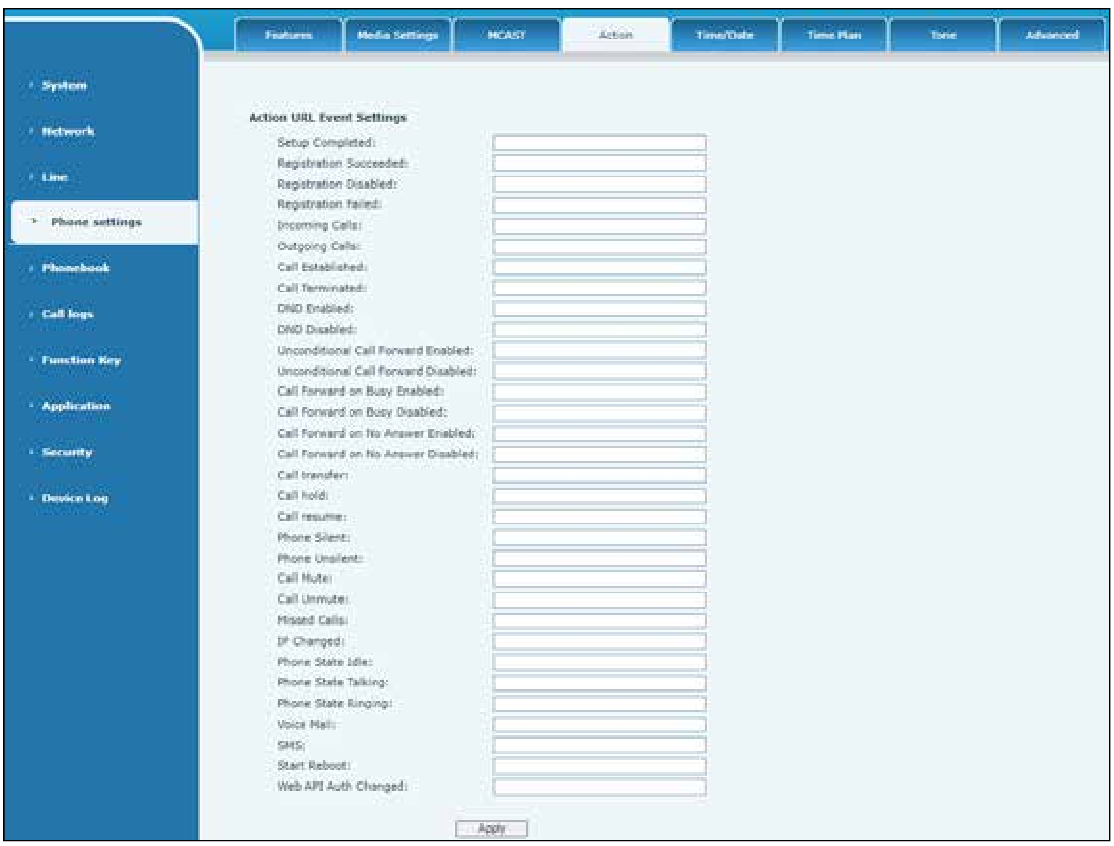


Table 52 Phone settings > MCAST Parameters

Parameters	Description
MCAST Settings	
MCAST Send DTMF Mode	Click the down arrow to select either RFC2833 or in-band.
MCAST Listening	
SIP Priority	Click the down arrow to select the SIP priority of the active call; 1 is the highest priority, 10 is the lowest.
MCast Listening Renew Time	Enter in the renew time amount in seconds.
Enable Page Priority	If checked, the voice call in progress takes precedence over all incoming paging calls.
Enable Prio Chan	If checked, enables the priority channel.
Enable Emer Chan	If checked, enables the emergency channel.
Name	Enter the multicast name.
Host: port	Enter the multicast IP address and port.
Channel	Click the down arrow to select the channel.
MCAST Dynamic	
Auto Exit Expires	Enter the amount when the auto exit expries.
Apply	Click the Apply button to save.

Phone settings > Action

Specify the Action URL that records the operation of the phone, and send the corresponding information to the server.



Phone settings > Time/Date

Use the Phone settings > Time/Date page to configure the time settings of the phone.

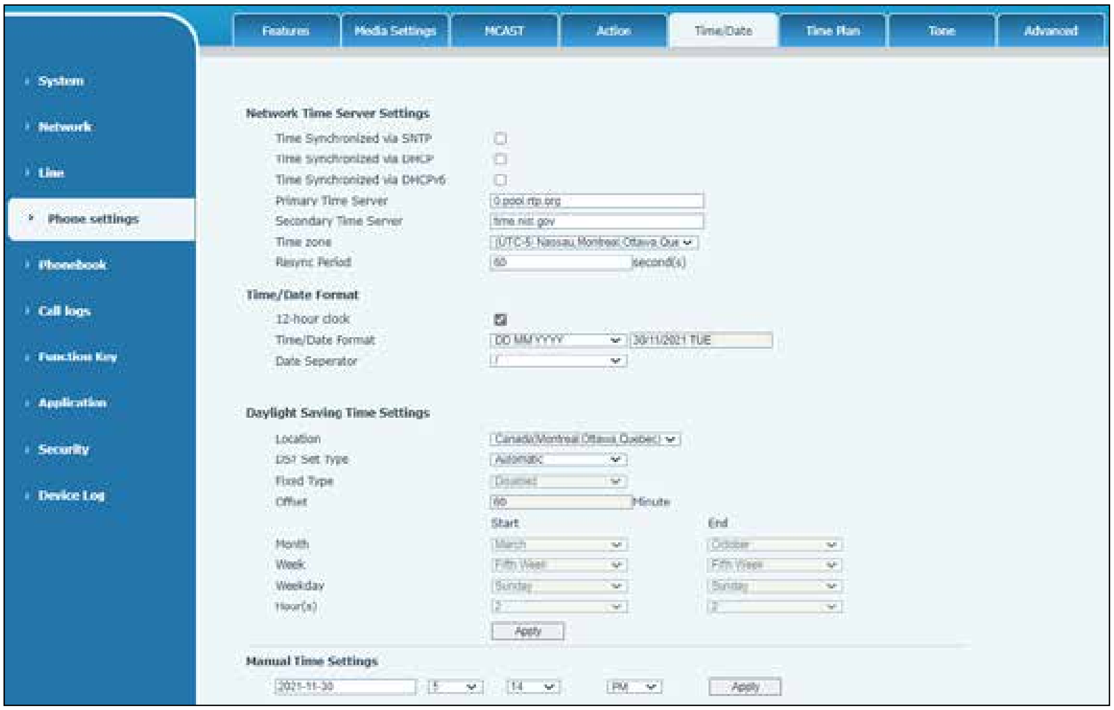


Table 53 Phone settings > Time/Data Parameters

Parameters	**Description**
Network Time Server Settings	
Time Synchronized via SNTP	If checked, enables time-sync through SNTP protocol.
Time Synchronized via DHCP	If checked, enables time-sync through DHCP protocol.
Time Synchronized via DHCPv6	If checked, enables time-sync through DHCPv6 protocol.
Primary Time Server	Enter the primary time server address.
Secondary Time Server	Enter the secondary time server address, when the primary server is not reachable, the device tries to connect to the secondary time server to get time synchronization.
Time Zone	Click the down arrow to select the time zone.
Resync Period	Enter the time of re-synchronization with the time server.
Time/Date Format	
12-Hour clock	If checked, sets the time display in 12-hour mode.
Time/Date Format	Click the down arrow to select the time and date format. The example displays at the right.
Data Separator	Click the down arrow to select the data separator.
Daylight Saving Time Settings	
Location	Click the down arrow to select your phone's location. The phone sets daylight saving time automatically based on the location.
DST Set Type	Click the down arrow to select the DST Set Type. If you select, Manual, you need to set the start and end times.

Parameters	Description
Fixed Type	Click the down arrow to set By Week, By Date, or Disabled. The daylight saving time rules are based on specific dates or relative rule dates for conversion. Unavailable if Automatic is specified in the DST Set Type field.
Offset	Enter the offset minutes when DST started. Unavailable if Automatic is specified in the DST Set Type field.
Month Start	Enter the DST start month. Unavailable if Automatic is specified in the DST Set Type field.
Week Start	Enter the DST start week. Unavailable if Automatic is specified in the DST Set Type field.
Weekday Start	Enter the DST start weekday. Unavailable if Automatic is specified in the DST Set Type field.
Hour Start	Enter the DST start hour. Unavailable if Automatic is specified in the DST Set Type field.
Month End	Enter the DST end month. Unavailable if Automatic is specified in the DST Set Type field.
Week End	Enter the DST end week. Unavailable if Automatic is specified in the DST Set Type field.
Weekday End	Enter the DST end weekday. Unavailable if Automatic is specified in the DST Set Type field.
Hour End	Enter the DST end hour. Unavailable if Automatic is specified in the DST Set Type field.
Manual Time Settings	Click the down arrow to set your time manually. Unavailable if Automatic is specified in the DST Set Type field.

Phone settings > Time Plan

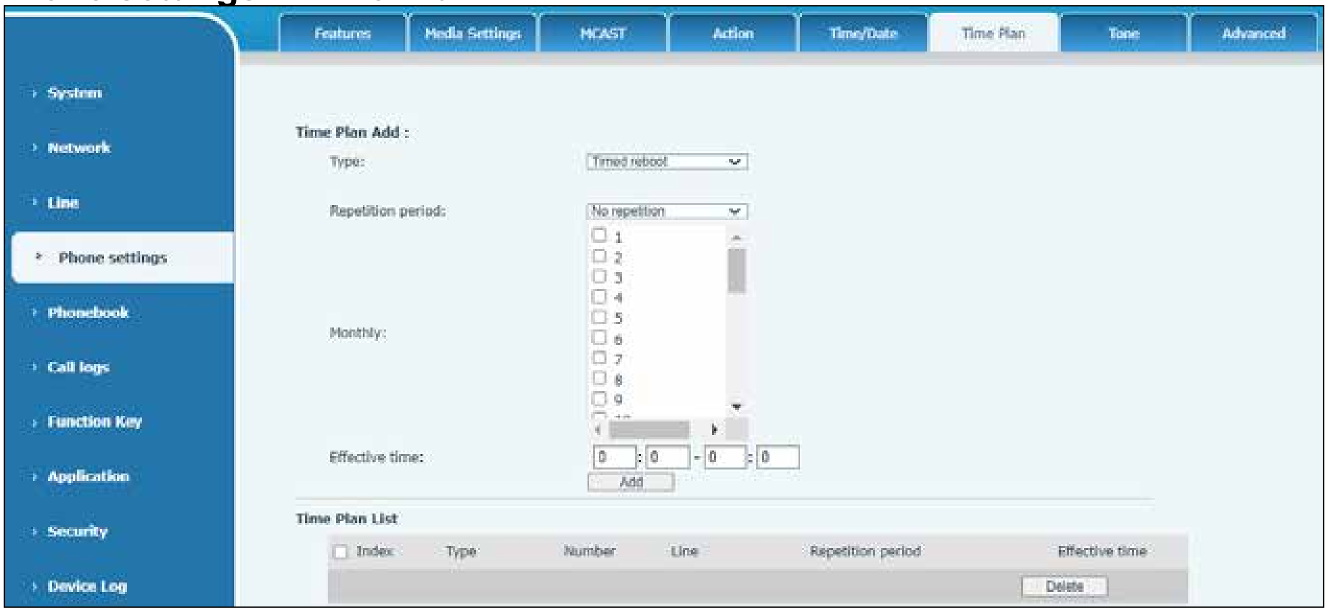


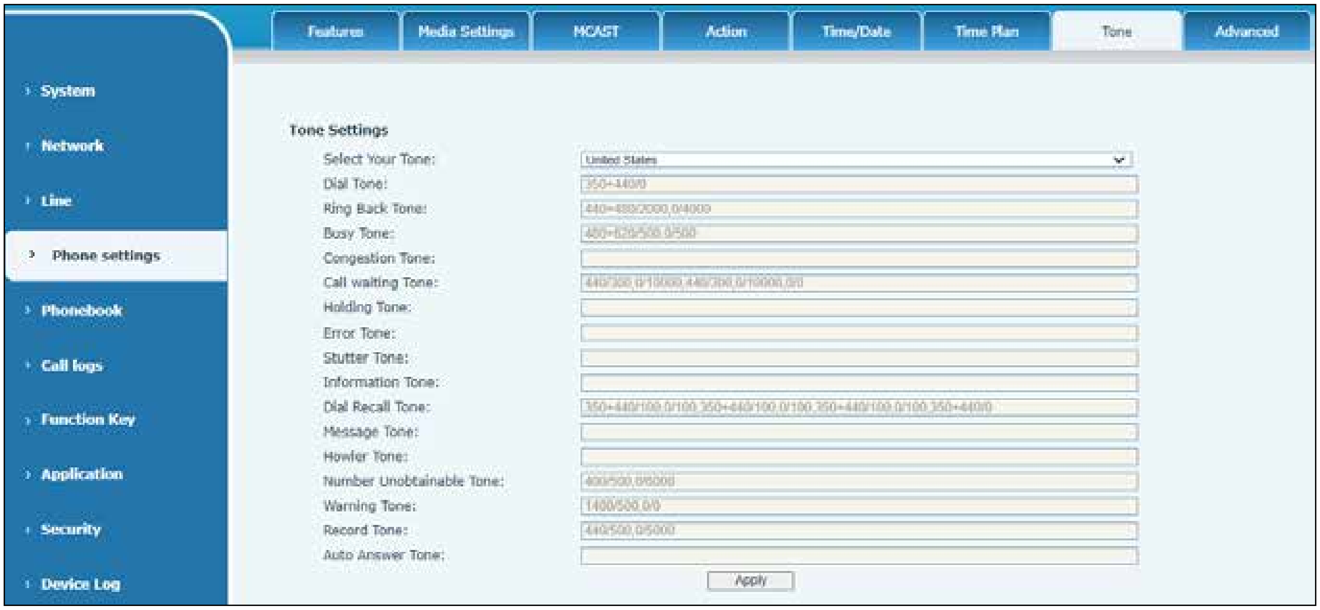
Table 54 Phone settings > Time Plan Parameters

Parameters	Description
Time Plan Add	
Type	Click the down arrow to select Timed reboot, Timed upgrade, or Timed forward.
Repetition period	Click the down arrow to select No repetition, Daily, Weekly, or Monthly.
Effective time	Enter in the time.
Add	Click the Add button to add the Time Plan to the Time Plan List.
Delete	Click the Delete button to delete the time plan from the time Plan list.

Phone settings > Tone

Phone settings > Tone page allows you to configure a phone prompt.

You can either select the country area or customize the area. If the area is selected, it populates automatically. If you choose to customize the area, you can modify the button tone, call back tone, and other information.

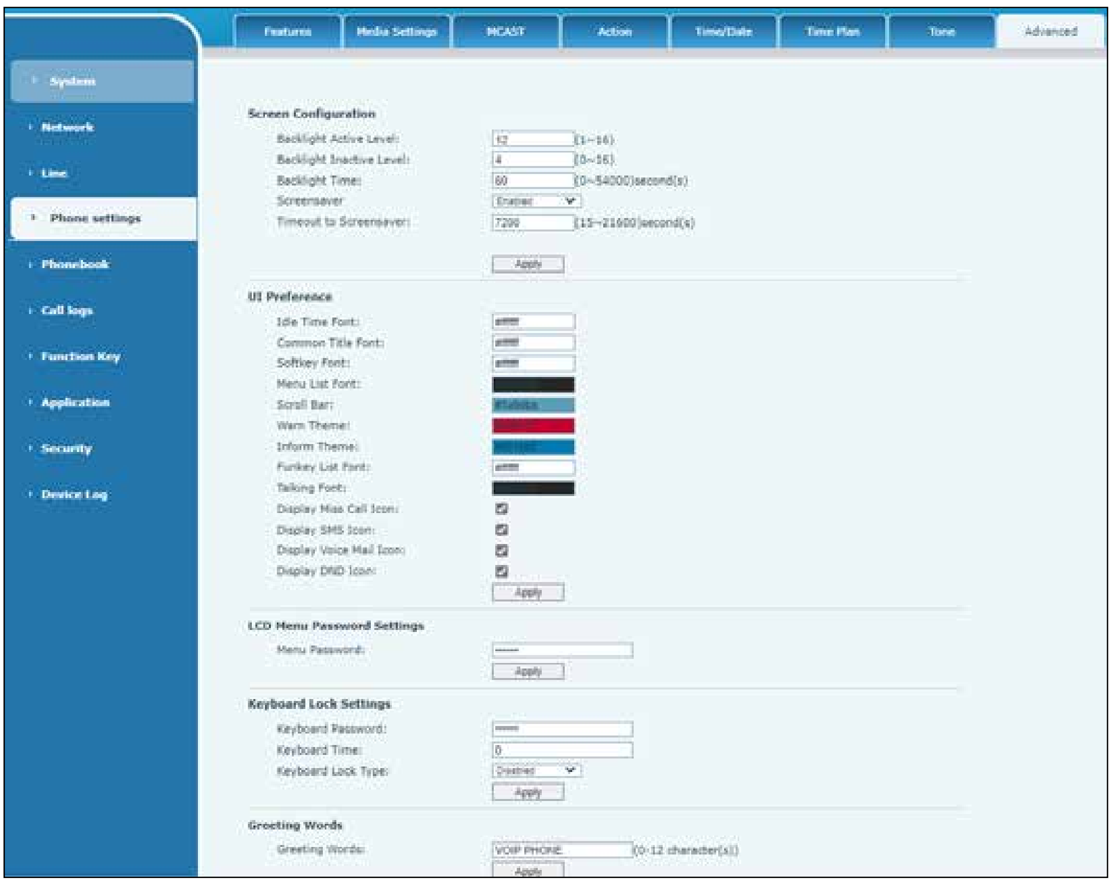


Phone settings > Advanced

Use the Phone settings > Advanced page to configure the advanced configuration settings.

- Screen Configuration.
- Enable Energy Saving
- Backlight Time
- LCD Menu Password Settings. The password is 123 by default.
- Keyboard Lock Settings.
- Configure Greeting Words

The greeting message will display on the top left corner of the LCD when the device is idle. The greeting message is limited to 16 characters. The default characters are VOIP PHONE.



Phonebook Web Pages

Phonebook > Contact

You can add, delete, or edit contacts in the phonebook on this page. You can browse the phonebook and sort it by name, phones, or filter them out by group.

Adding a New Contact

To add a new contact:

1. Click the Add new contact button. The following dialog box appears.

2. Enter the contact’s information and click the OK button.

Editing a Contact

To edit a contact, click the check box in front of the contact. The contact information is copied to the contact edit boxes. Press the Modify button to save the changes.

Deleting a Contact

To delete one or multiple contacts, click the check box in front of the contact(s) and click the Delete button or the Clear button.

Adding Multiple Contacts into a Group

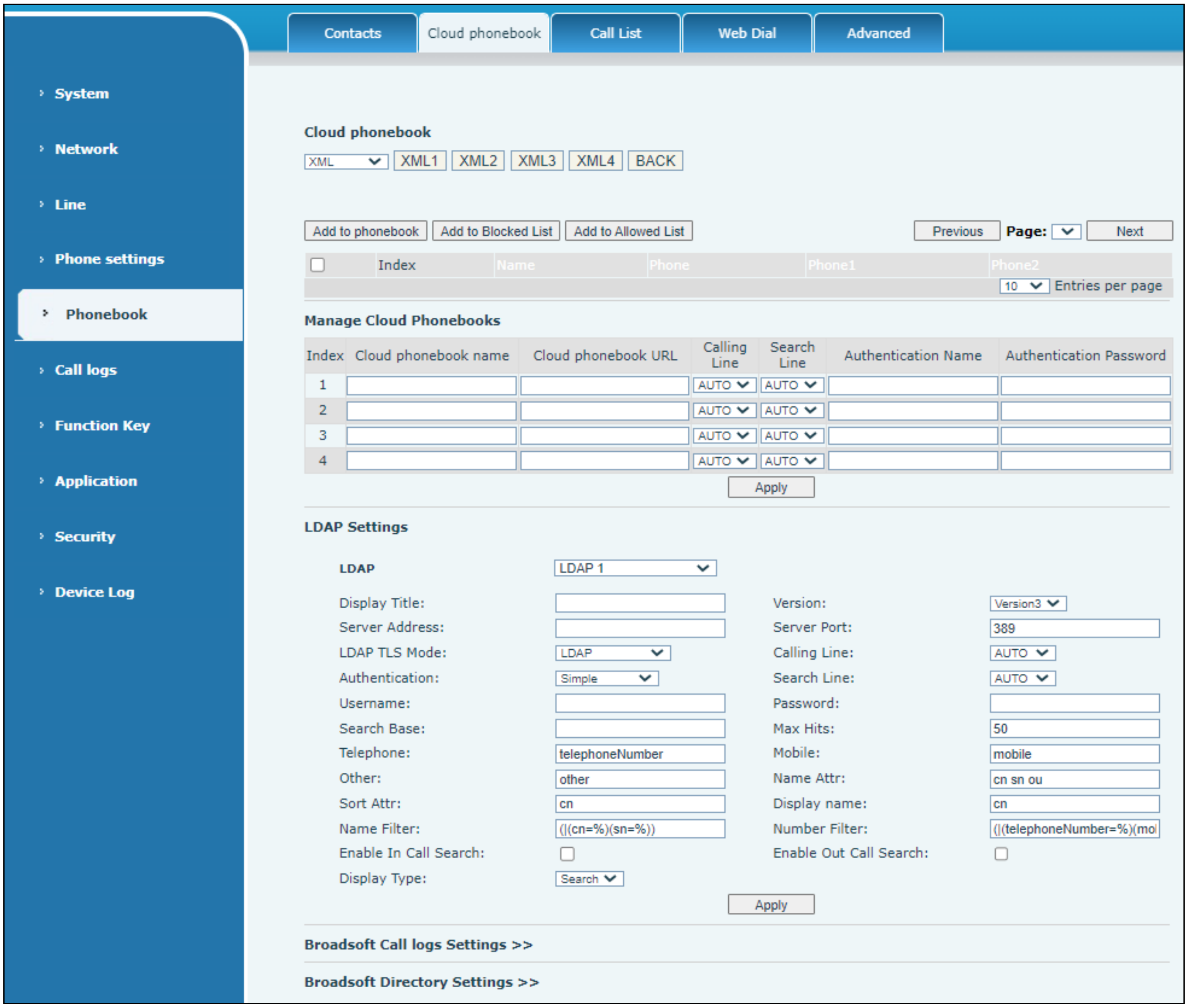
To add multiple contacts into a group, select the group in the drop-down option in front of the Add to Group button. Click the check box to select a contact. Click the Add to Group button.

Adding Multiple Contacts to the Blocked List

To add multiple contacts into a blocked list, click the check box in front of the contact(s) and click the Add to Blocked List button.

Phonebook > Cloud Phonebook

You can configure up to eight Cloud phonebooks. You must configure each Cloud phonebook with a 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, you must configure the username and password.



Manage Cloud Phonebooks

To configure a Cloud phonebook, enter the following information:

- Cloud phonebook name
- Cloud phonebook URL
- Authentication name and Authentication password (optional)

LDAP Settings

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

To configure an LDAP phonebook, enter the following information:

- Display Title
- Server Address
- Server Port
- Search Base
- Username name and Password (optional)

Web page preview

Phone page supports a preview of the Internet phone directory and contacts

1. After setting up the XML VoIP directory or LDAP, select Phonebook > Cloud phonebook > Cloud phonebook and select the type.
2. Click the set XML/LDAP to download the contact for browsing.

Phonebook > Call List

Use the Phonebook > Call list page to blocked specified phone numbers, allow incoming calls, and restrict outgoing calls.

Restricted Incoming Calls

Restricted Incoming Calls means that the phone number is blocked. Add the phone number to the list, and you will no longer receive calls from the stored number until you remove it from the list.

You can add specific numbers to the list or add specific prefixes to the list to block calls with all numbers with this prefix.

Allowed Incoming Calls

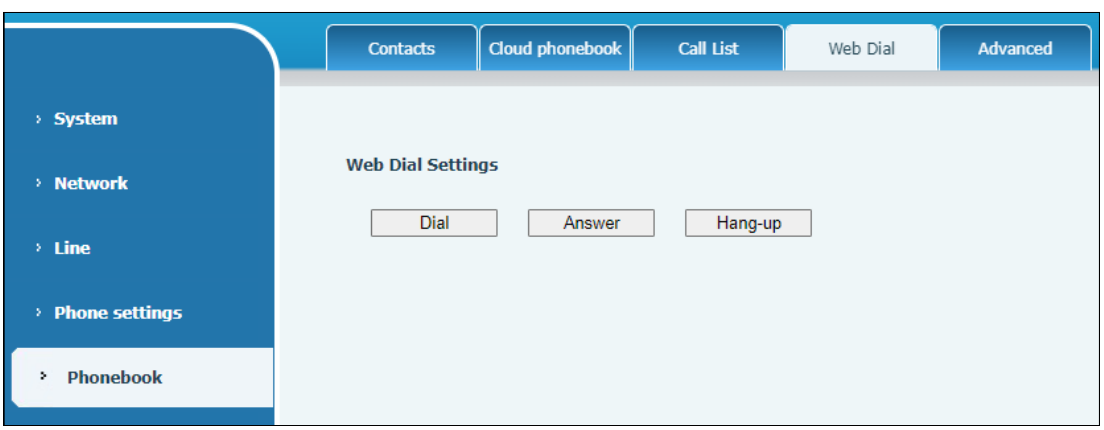
When DND is enabled, the incoming call number can still be called.

Restricted Outgoing Calls

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

Phonebook > Web Dial

Use the Phonebook > Web Dial page for call, reply, and hang up operations.

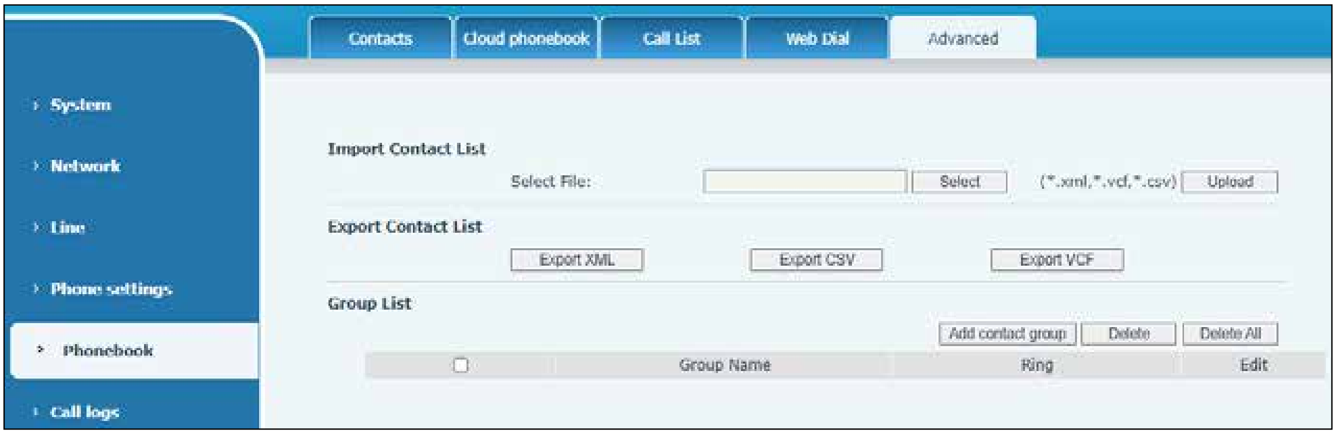


Phonebook > Advanced

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

IMPORTANT: If you repeatedly import the same phone book, the same contact will be ignored. If the name is the same, but the number is different, the contact is created again.

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

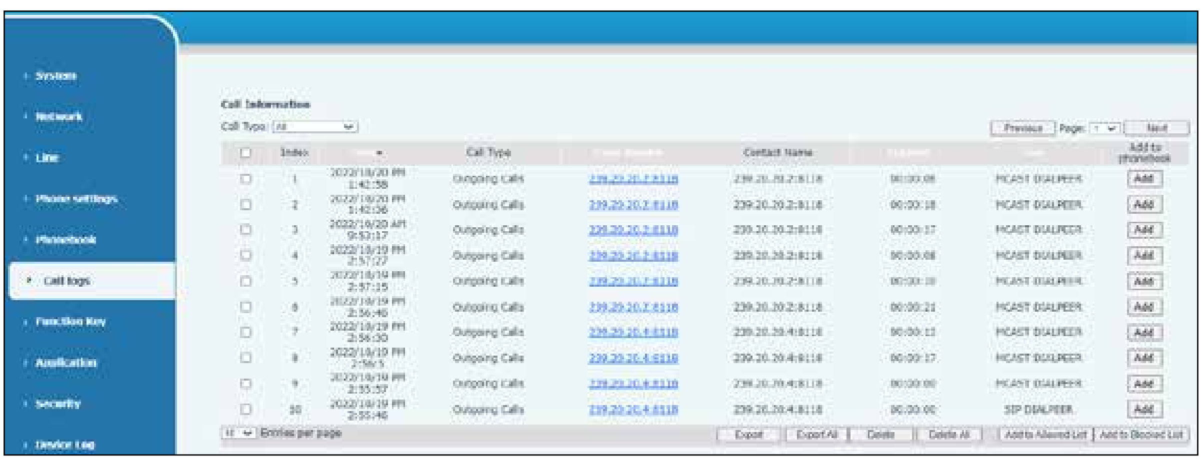


Call logs Web Page

On the Call logs page, you can browse the complete call record. You can sort the call record by time, call number, contact name, or line. You can screen the call record by call record type (incoming calls, outgoing calls, missed calls, or forward calls).

You can save the number in the call record to your phone book or add it to the Allowed List and Blocked List. You can dial the web page by clicking on the number in the call log.

You can download call records conditionally and save them locally.



Function Key Web Pages

Function Key > Function Key

Dsskey Transfer Mode options: make a new call, blind transfer, attended transfer, conference call, or Play DTMF.

The device provides two user-defined shortcuts that you can configure on a web page.

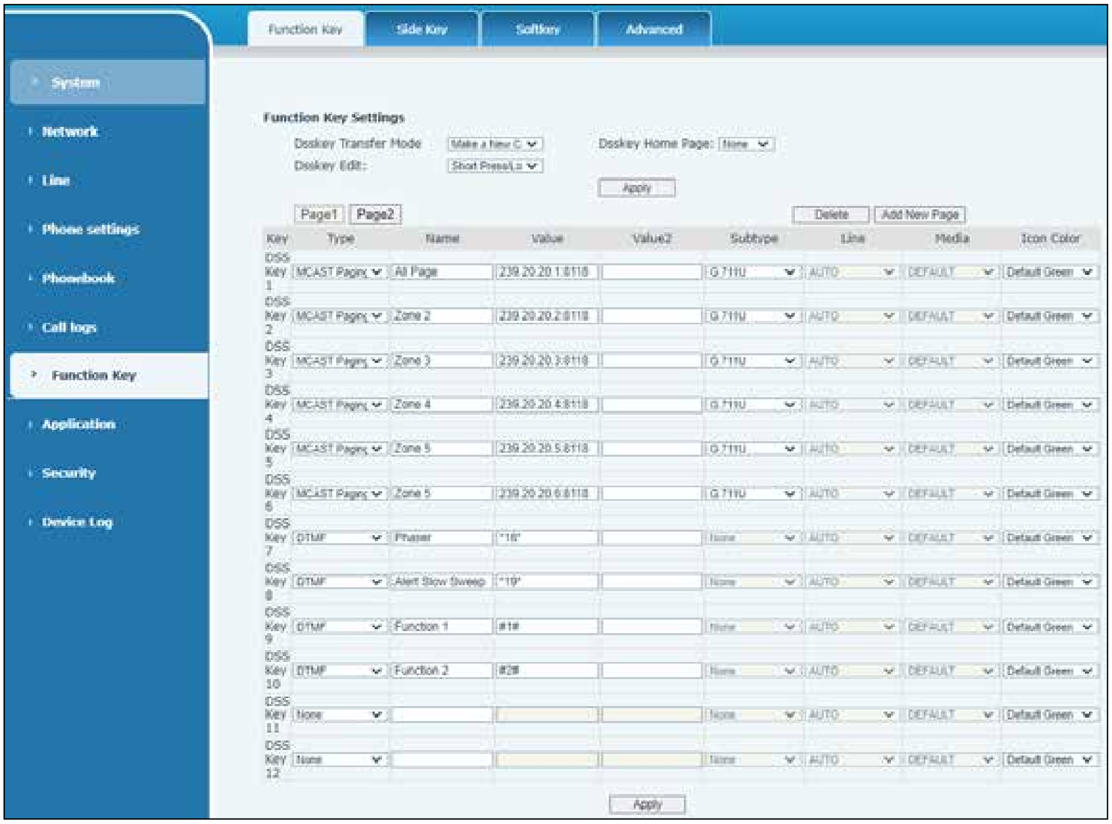


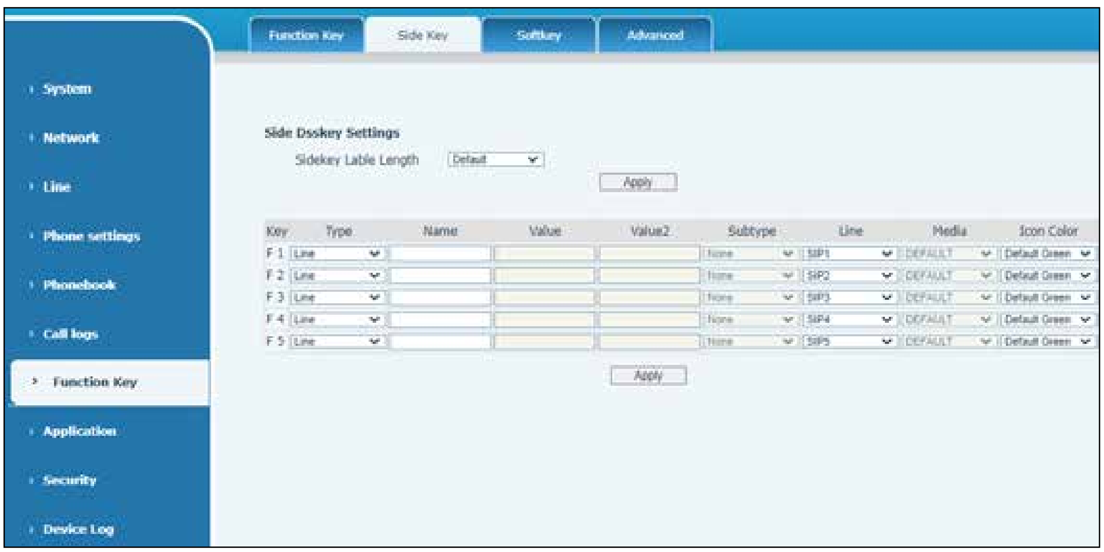
Table 55 Function Key > Function Key Parameters

Parameters	Description
Memory Key	• BLF (New Call/BXFE /AXFER): Used to prompt users on the state of the subscribe extension, and it can also pick up the subscribed number, which helps users to monitor the state of subscribe extension (that is, idle, ringing, or a call). There are three types of one-touch BLF transfer methods. **NOTE**: You enter the pick-up number for the specific BLF key to fulfill the pick-up operation. • Presence: Used to view whether the user is online. **NOTE**: You cannot subscribe to the same number for BLF and Presence at the same time • Speed Dial: Used to call the number directly. You must enter the number. This function is useful for users who dial certain numbers regularly. • Intercom: Allows the operator to connect the phone quickly.
Line	Configure as a Line Key. You can make a call by pressing the Line Key.

Parameters	Description
Key Event	You can select a key event as a shortcut to trigger. For example, MWI / DND / Release / Headset / Hold / etc.
DTMF	Allows you to dial or edit dial numbers easily.
URL	Opens the specific URL directly.
Multicast	Configure the multicast address and audio codec. You press the key to initiate the multicast.
Action URL	Use a specific URL to make basic calls to the phone.
XML browser	Use can set the DSS Key for specific URL downloads and other operations.

Function Key > Side Key

A Side Key is a key on both sides of the screen that functions as a shortcut key. The default configuration is a line key that you can customize on the webpage. For the Side Key function and settings, see “Function Key > Function Key” on page 140.



Function Key > Softkey

Softkeys provide a way to customize the phone's user interface depending on the context and your personal preferences.

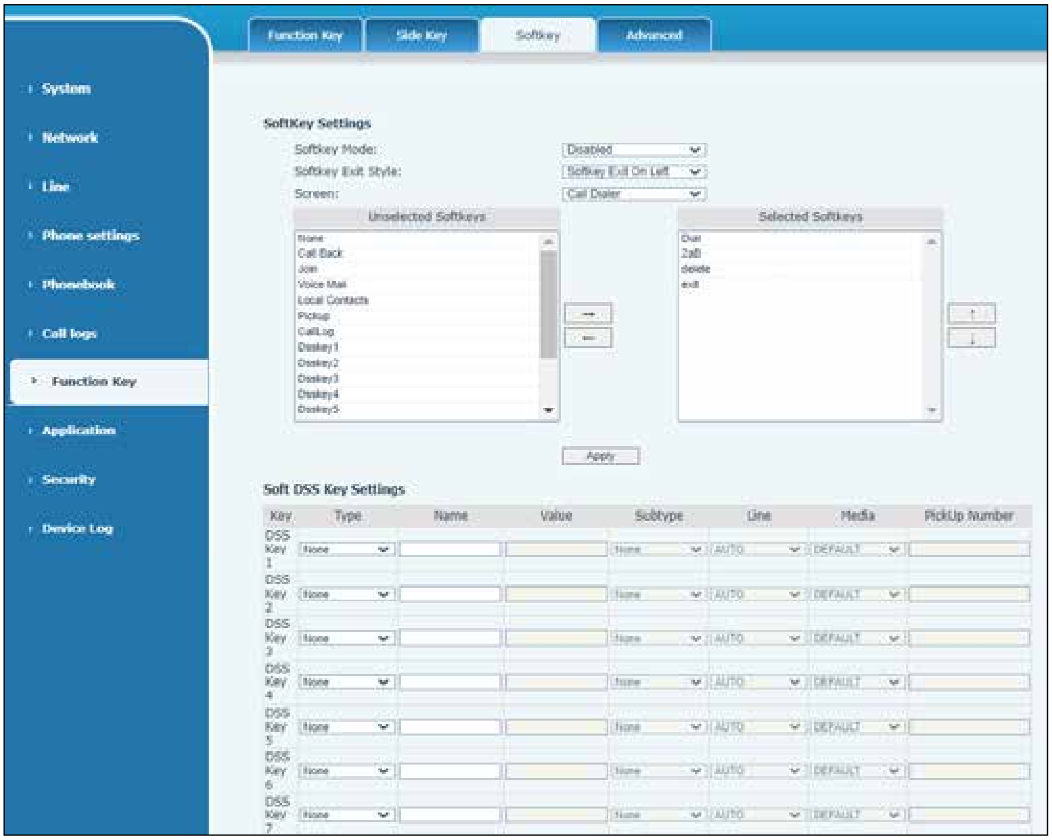


Table 56 Function Key > Softkey Parameters

Parameters	**Description**
Softkey Settings	
Softkey Mode	Select from Disabled and More. The default is Disabled.
Softkey Exit Style	Select from Softkey Exit On Right or Softkey Exit On Left.
Screen	
Call Dialer	Select from Call Dialer, Conference (Conf), Desktop, Divert Dialer, Ending, Predictive Dialer, Ringing, Talking, Transfer Alerting, Transfer Dialer, Trying Waiting.
Call Dialer	Select from None, Call Back, Join, Voice Mail, Local Contacts, Pickup, CallLog DssKey1-10, Clear, In, Missed, Next Line (Next), Dialed, Pause, Prev Line(Prev.), Remote XML (R-XML), Headset, Video, Audio.
Conference	Select from None, Mute, Release, DssKey1-10, Headset.
Desktop	Select from None, Pre Account, Next Account, Blocked List, Call Back, Call Forward, In, Lock, Memo, Missed, Voice Mail, Dialed, Reboot, Redial, Remote XML, SMS, Status, Headset, Network, DssKey1-10.
Divert Dialed	Select from None, Forward, Local Contacts, Call Log, DssKey1-10, Clear, In, Missed, Dialed, Remote XML, Headset.

Parameters	Description
Ending	Select from None, DssKey1-10, Headset, Release.
Predictive Dialer	Select from None, Call Back, Local Contacts, Voice Mail, Pickup, Join, Call Log, DssKey1-10, Clear, In, Missed, Next line, Dialed, Pause, Prev line, Remote XML, 2aB, Headset, Video, Audio.
Ringing	Select from None, Mute, Release, DssKey1-10, Headset, Video, Audio.
Talking	Select from None, Mute, New Call, Release, Local Contacts, Listen, RTP, Call Log, DssKey1-10, Next call, Prev call, Private Hold, Headset, Video, Audio.
Transfer Alerting	Select from None, DssKey1-10, Headset, Release.
Transfer Dialer	Select from None, Local Contacts, Call Log, DssKey1-10, Clear, In, Missed, Dialed, Pause, Remote XML, 2aB, Headset, Video, Audio.
Trying	Select from None, DssKey1-10, Headset.
Waiting	Select from None, Conference, Forward, Hold, Mute, New call, Next call, Prev call, Release, Headset, Listen, DssKey1-10, Video, Audio.

Function Key > Advanced

One key transfer example: set the memory key to 4370. Press the memory key when talking with 4374 to decide whether to call 4370 or transfer 4374 to 4370.

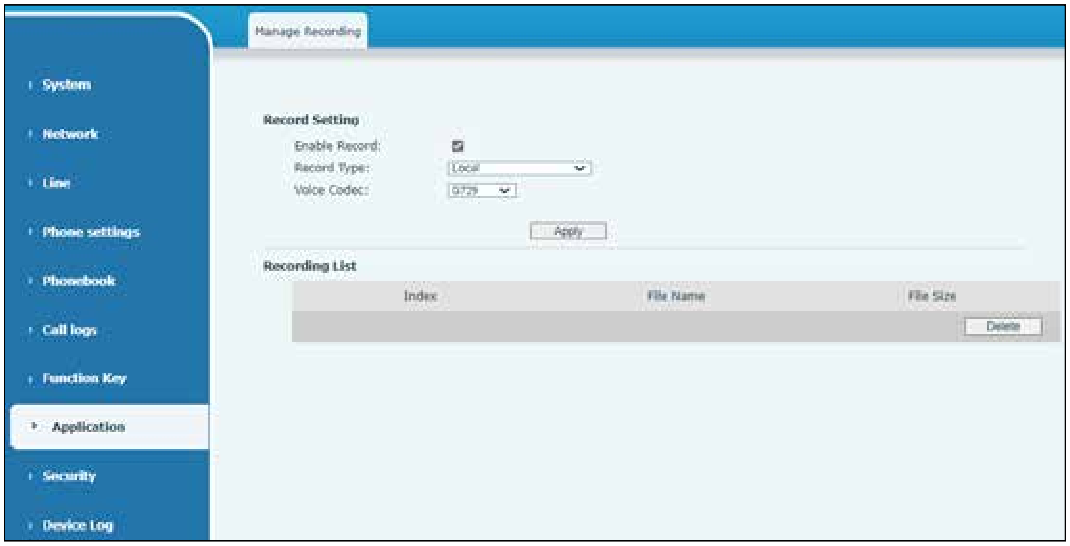
Select memory key function example: the phone sets the memory key value to 4370. When 4370 calls, press this key to hold the call or hang up.

See "Table 31 Line > Dial Plan > Dial Plan Add Parameters" on page 114.

Application Web Page

Application > Manage Recording

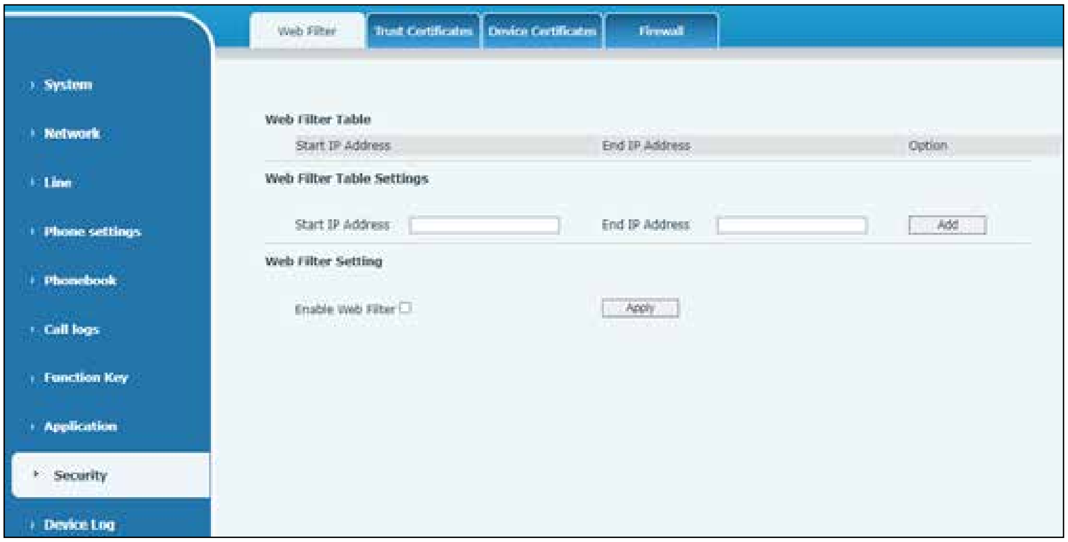
See “Recording a Call” on page 60 for details.



Security Web Pages

Security > Web Filter

You can set up a configuration management phone that allows only machines with a specific network segment IP access.



Adding and Removing IP Segments

To add and remove IP segments:

1. Go to the Security > Web Filter page.
2. Enter the starting IP address in the Start IP Address field.
3. Enter the ending IP address in the End IP Address field.
4. Click the Add button to submit changes.

NOTE: You can divide a large network segment into several smaller network segments to add.

To delete, select the initial IP of the network segment from the drop-down menu, and click Delete.

NOTE: If the device you are accessing is in the same network segment as the phone, 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.

Table 57 Security > Web Filter Parameters

Parameters	Description
Web Filter Table Settings	
Start IP Address	Enter the start of the IP Address.
End IP Address	Enter the end of the IP Address.
Add	Click to save changes.
Web Filter Settings	
Enable Web Filter	Click to enable web page filtering.
Apply	Click to save changes.

Security > Trust Certificates

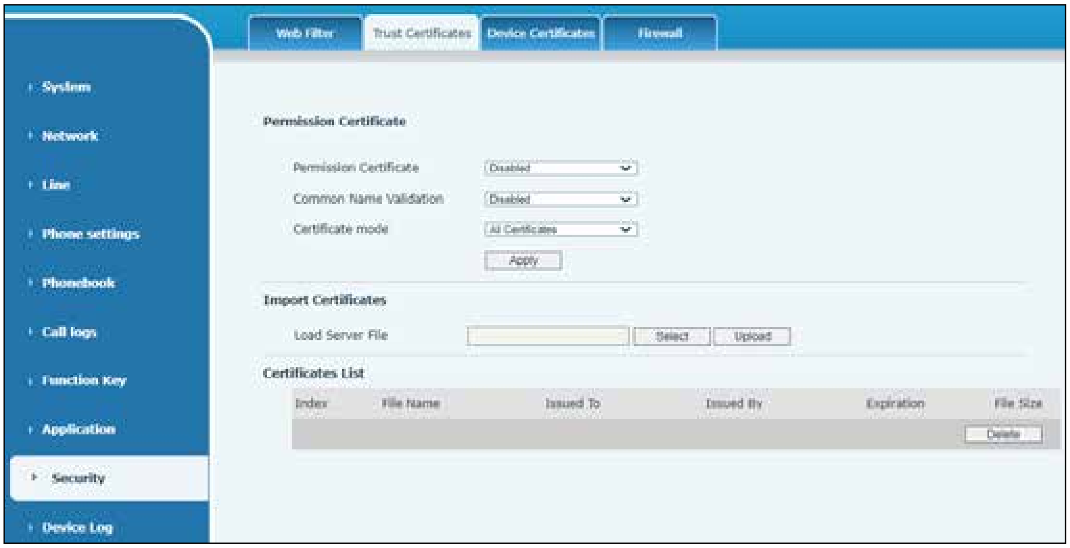


Table 58 Security > Trust Certificates Parameters

Parameters	Description
Permissions Certificate	
Permissions Certificate	Click the arrow to select Disable or Enabled.
Common Name Validation	Click the arrow to select Disable or Enabled.
Certificate mode	Click the arrow to select All Certificates, Default Certificates, or, Custom Certificate.
Apply	Click to apply changes.
Import Certificates	
Load Server File	Click the Select button to select a file and then click the Upload button.
Certificate List	
Delete	Click the selected certificate and click delete.

Security > Device Certificates

Table 59 Security > Device Certificates Parameters

Parameters	**Description**
Device Certificates	
Device Certificates	Click the arrow to select Disable or Enabled.
Apply	Click to apply changes.
Import Certificates	
Load Server File	Click the Select button to select a file and then click the Upload button.
Certificate File	
Delete	Click the selected certificate and click delete.

Security > Firewall

You can set whether to enable the input through this page and set the firewall input and output rules. Using these settings can prevent malicious network access or restrict internal users' access to external network resources, which can improve security.

A 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.

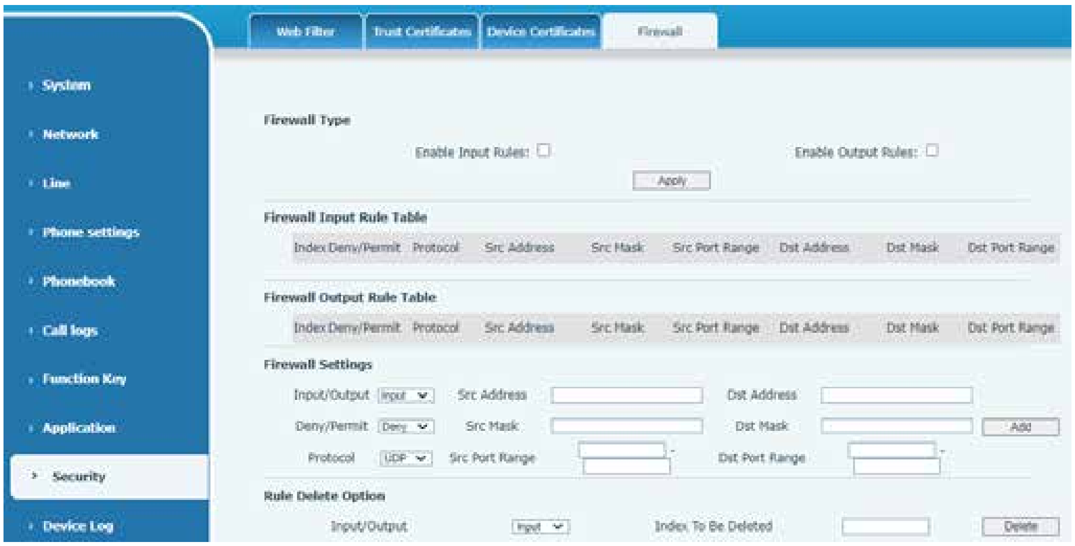


Table 60 Security > Firewall Parameters

Parameters	Description
Enable Input Rules	Check to enable input rules.
Enable Output Rules	Check to enable output rules.
Firewall Settings	
Input/Output	Click the down arrow to select whether the currently added rule is an input or output rule.
Src Address	Enter the source address. It can be the host, network, or all addresses, 0.0.0.0. Also, it can be a network address similar to *.*.*.0, such as 192.168.1.0.
Dst Address	Enter the destination address. It can be the specific IP address or the full address 0.0.0.0. Also, it can be a network address like *.*.*.0, such as 192.168.1.0.
Deny/Permit	Click the down arrow to select whether the current rule configuration is disabled or allowed.
Src Mask	Enter 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.
Protocol	Click the down arrow to select the types of filtering protocols: TCP, UDP, or ICMP.
Src Port Range	Enter the source port range.
Dst Port Range	Enter the destination port range.

Parameters	Description
Apply	Click the Apply button. 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, the other IP of the ping 192.168.1.0 network segment can still receive the response packet from the destination host normally.
Add	After setting, click to add a new item in the firewall input rule.
Delete	Click to delete the selected list.

Device Log Web Page

Device Log > Device Log

You can review the device log, and when you encounter a problem, you can send the log to the technician to locate the problem. See “Get Log Information” on page 151.

Troubleshooting

When the phone is not in regular use, you can try the following methods to restore the phone’s regular operation or collect relevant information and send a problem report to Federal Signal technical support mailbox.

Getting the Phone’s System Information

You can get information by pressing the Menu > Status option on the phone. The following information is provided:

- Network
- Phone (which includes MAC, model, hardware, software, and RAM)
- Account

Rebooting the Phone

You can reboot the phone from the soft menu by pressing Menu > Basic > Reboot System. Reboot by pressing OK or remove the power supply and restore it again.

Reseting the Phone to Factory Default

Resetting Device to Factory Default erases all the phone’s configuration, preferences, database, and profiles and restores the phone to factory default.

To perform a factory default reset:

1. Press Menu > Advanced and enter the password.
2. Select Factory Reset and press Enter.
3. Confirm the action by pressing OK.

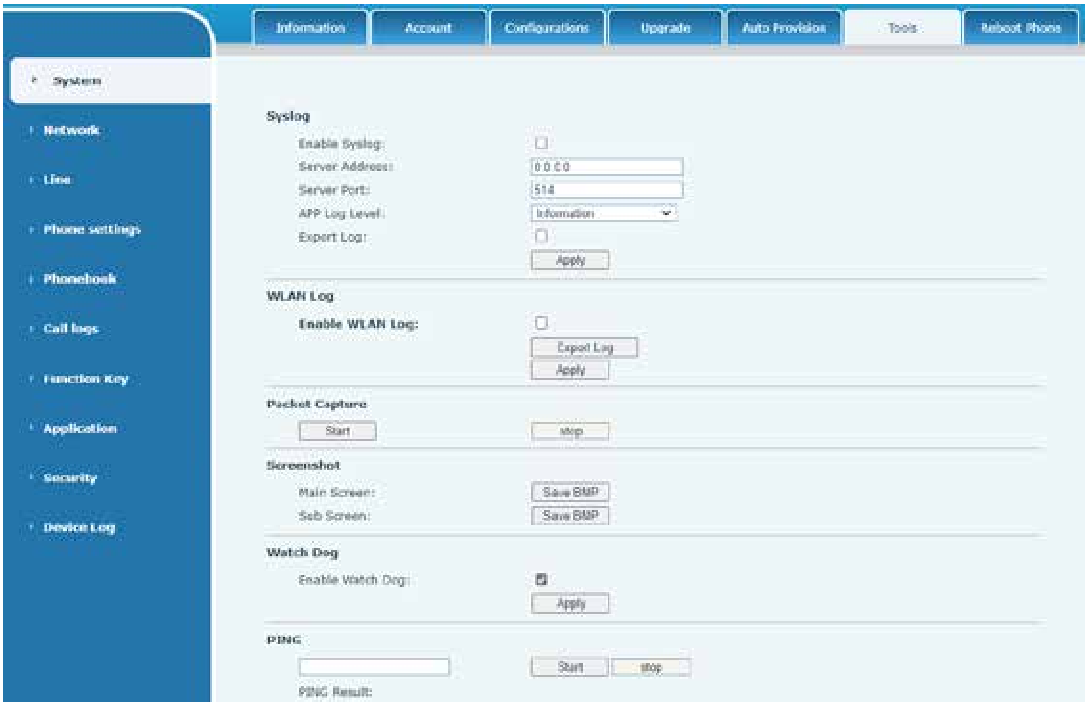
The device reboots to a clean factory default state.

Screenshot

If there is a problem with the phone, a screenshot can help the technician locate the function and identify the problem.

To take screen shots:

1. Log in the phone webpage System > Tools.
2. Under Screenshot, click the Main Screen and Sub Screen buttons.



Network Packets Capture

Sometimes it is necessary to analyze packet traffic between systems to troubleshoot difficult application or networking problems.

To get the packets dump of the phone:

1. Log in the web portal, go to System > Tools.
2. Click Start under the Packet Capture section. A pop-up message prompts you to save the capture file.
3. Perform the relevant operations such as activating or deactivating line or making phone calls.
4. Click the Stop button when operation finished. The network packets of the device during the period have been dumped to the saved file.

You may examine the packets with a packet analyzer or send it to Federal Signal Technical Support.

Get Log Information

Log information is helpful when encountering problems that cannot be handled locally

To get the log information off the phone:

1. Log on the phone web page and open the Device log.
2. Click the Start button.
3. Follow the steps of the problem until the problem appears, and then click the End button.
4. Save to local analysis or send the log to the technician to locate the problem.

Troubleshooting

Table 61 Troubleshooting

Problem	Action
Phone could not boot up	• An external power supply powers the device via a power adapter or PoE switch. Use the standard power adapter provided by the manufacturer or PoE switch that meets the specification requirements and check if the device is well connected to the power source. • You see POST MODE on the phone's screen. The phone's system image has been damaged. Contact technical support to help you restore the phone system.
Device could not register to a service provider	• Check if the phone is well connected to the network. The network Ethernet cable should be connected to the Network port and 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. • 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. Check if the network configurations are correct. • If the network connection is fine, recheck your line configurations. If all configurations are correct, contact your service provider to get support, or follow the instructions in "Network Packets Capture" on page 150 to get the network packet capture of the registration process and send it to manufacturer support to analyze.
No Audio or Poor Audio on Handset	• Check if Handset is connected to the correct Handset ( ) port and not the Headphone ( ) port. • The network bandwidth and delay may not be suitable for an audio call at the moment.
Poor Audio or Low Volume in Headphone	• There are two Headphone wire sequences in the market. Use the Headphone provided by the manufacturer or consult the manufacturer for the wire sequence if you wish to use a third-party headphone. • The network bandwidth and delay may not be suitable for an audio call.
Audio is chopping at far-end in Hands-free speaker mode	Usually due to loud volume feedback from speaker to microphone. Lower the speaker volume slightly, and the chopping will be gone.

Getting Service

If you are experiencing any difficulties, contact Federal Signal Customer Support at 800-548-7229 or 708-534-3400 extension 7511 or Technical Support at 800-524-3021 or 708-534-3400 extension 7329 or email at techsupport@fedsig.com. For instruction manuals and information on related products, visit http://www.fedsig.com.