



NRP1302G/P

User Manual

Copyright © 2025 New Rock Technologies, Inc. All Rights Reserved.

All or part of this document may not be excerpted, reproduced and transmitted in any form or by any means without prior written permission from the company.

Mail: global@newrocktech.com

Tel: 021-61202700 / 52217917

Document center: <https://en.newrocktech.com/DocumentCenter>

Catalogs

Catalogs.....	2
1 Figure.....	7
2 Table	9
3 Safety Instructions	10
4 Summarize.....	11
4.1 Summarize.....	11
4.2 Accessory Description.....	12
5 Installation wizard	13
5.1 PoE and the use of external power adapters	13
6 Appendice	16
6.1 Appendix I - Icons	16
6.2 Appendix II - Keyboard Character Lookup Table.....	17
7 User orientation	19
7.1 Key Description.....	19
7.1.1 Key Description.....	19
7.2 Use of handles, headphones, hands-free speakers	21
7.3 Standby mode	21
7.4 phone state.....	22
7.5 Web management	23
7.6 Network setup.....	23
7.7 Line Configuration	24
8 Basic Functions	26
8.1 Make a telephone call.....	26
8.2 Take a telegram	27
8.2.1 Call screen.....	27
8.2.2 Make/receive a second call.....	28
8.3 End of call.....	28
8.4 Reappoint.....	29
8.5 Dial-up search.....	29
8.6 Auto-answer	29
8.7 Unmute	29
8.7.1 In-call mute	30
8.7.2 Mute when ringing	30
8.8 Call Hold/Resume	30

8.9	Distraction-free.....	30
8.10	Call forward.....	31
8.11	Call transfer	32
8.11.1	Spineless	32
8.11.2	Half attended	32
8.11.3	Attending.....	32
8.12	Call Waiting.....	32
8.13	Sessions	33
8.13.1	Local meeting.....	33
8.13.2	Web conference.....	34
8.14	Call residing.....	34
8.15	Rush to pick up.....	35
8.16	Anonymous call.....	36
8.16.1	Anonymous call.....	36
8.16.2	Prohibition of anonymous calls.....	36
8.17	Hotline	37
8.18	Emergency call	37
9	Advanced Features	38
9.1	BLF.....	38
9.1.1	Configuring the BLF Function	38
9.1.2	Using the BLF Function	39
9.2	BLF List.....	39
9.3	Sound recordings	40
9.3.1	Network recording	40
9.3.2	SIP INFO recording	41
9.4	Act on behalf of sb. In a responsible position	41
9.5	Intercom.....	42
9.6	Multicast.....	42
9.7	SCA	43
9.8	Text.....	45
9.8.1	Text message.....	45
9.8.2	MWI	45
9.9	Point of special interest.....	46
10	Phone Settings	48
10.1	Set up.....	48
10.1.1	Multilingualism.....	48
10.1.2	Timing.....	48
10.1.3	Screen	50

10.1.3.1	Brightness and Backlighting	50
10.1.3.2	Screensavers.....	50
10.1.4	Ringtones.....	50
10.1.5	Loudness	51
10.1.6	Welcome speech	51
10.1.7	Reboot.....	51
10.2	Telephone book	51
10.2.1	Local contact	51
10.2.1.1	Add/edit/delete contacts.....	52
10.2.1.2	Add/Edit/Delete Groups	52
10.2.1.3	Browse/add/remove contacts in groups	52
10.2.2	Blocked Call List	52
10.2.3	Remote phone book.....	53
10.2.3.1	Configuring Cloud Phonebook.....	53
10.2.3.2	Download Cloud Phonebook.....	53
10.3	Call record	53
10.4	Earphones	54
10.4.1	Wired headset	54
10.4.2	EHS Headphones.....	54
10.5	Advanced setting	54
10.5.1	Wire Setting	54
10.5.2	Network setup	55
10.5.2.1	Network setup.....	55
10.5.2.2	Quality of service and virtual local area networks.....	55
10.5.2.3	VPN.....	56
10.5.2.4	Web server type	57
10.5.3	Setting up the secret key	57
10.5.4	Safeguard	57
10.5.5	Online Upgrade.....	60
10.5.6	Restore Factory Settings.....	62
11	Web Configuration	59
11.1	Authentication of web pages	59
11.2	Systems >> Information	59
11.3	System >> User Configuration.....	59
11.4	System >> System Configuration.....	59
11.5	System >> Upgrade	60
11.6	Systems >> Automated Deployment.....	60
11.7	System >> Accessibility	60

11.8	System >> Reboot Phone	60
12	Network >> Basic.....	61
12.1	Network >> Service Ports	61
12.2	Network >> VPN.....	61
12.3	Network >> Advanced	62
12.4	Line >> SIP	62
12.5	Line >> SIP Hot Spot	68
12.6	Lines >> Rules for receiving numbers	68
12.7	Line >> Basic Settings.....	70
12.8	Phone Settings >> Feature Settings	71
12.9	Phone Settings >> Media Settings.....	75
12.10	Phone Settings >> Multicast.....	76
12.11	Phone Settings >> Features	76
12.12	Phone Settings >> Time/Date.....	76
12.13	Phone Settings >> Time Management.....	78
12.13.1	Repeat cycle selection daily.....	79
12.13.2	Repeat cycle selection weekly.....	79
12.13.3	Time Management Lists	79
12.13.4	Removing.....	79
12.14	Phone Settings >> Tones	79
12.15	Phone Settings >> Advanced.....	80
12.16	Phonebook >> Contacts.....	80
12.17	Phonebook >> Cloud Phonebook	81
12.18	Phonebook >> Call List.....	82
12.19	Phonebook >> Web Dialing	82
12.20	Phonebook >> Advanced	82
12.21	Call record	83
12.22	Shortcuts >> Shortcuts	83
12.23	Shortcuts >> Softkey Settings	84
12.24	Shortcut >> Advanced.....	85
12.25	Applications >> Recording Management.....	85
12.26	Security >> Web Filtering	85
12.27	Security >> Certificate of Trust.....	86
12.28	Security >> Device Certificates.....	86
12.29	Security >> Network Firewall	87
12.30	Device Log >> Device Log	88
13	Fault Resolution	89
13.1	Getting phone system information.....	89

13.2 Restart A Telephone	89
13.3 Restore factory settings on the phone.....	89
13.4 Screenshot.....	89
13.5 Network Data Grabber.....	90
13.6 Getting log information	90
13.7 Common Failure Cases.....	91

1 Figure

Figure 1 - Desktop Phones Installation	14
Figure 2 - Desktop to Wall Mount Conversion	14
Figure 3 - Device Connections	15
Figure 4 - Key Description.....	19
Figure 5 - Screen Layout / Default Home Screen	21
Figure 6 - scroll icon.....	22
Figure 7 - Phone Handset Status.....	22
Figure 8 - Web Phone Status	23
Figure 9 - Web landing page	23
Figure 10 - Phone Line SIP Address and Account Information	25
Figure 11 - Web Line Registration	25
Figure 12 - Default Line	26
Figure 13 - Dialing before turning on the voice channel.....	26
Figure 14 - Dialing a number with the voice channel turned on	27
Figure 15 - Calling Numbers.....	27
Figure 16 - Answering Incoming Calls.....	27
Figure 17 - Call Mode Interface.....	27
Figure 18 - Second Incoming Call Screen.....	28
Figure 19 - Dual Talk.....	28
Figure 20 - Setting DND Options	31
Figure 21 - Enabling Do Not Disturb for a Line.....	31
Figure 22 - Setting Up Call Forwarding.....	32
Figure 23 - Setting up a local conference.....	33
Figure 24 - Web Conferencing	34
Figure 25 - Web Page Setup Call Dwell	35
Figure 26 - Web page robocall setup	36
Figure 27 - Web Anonymous Calling Enabled	36
Figure 28 - Web page settings to disable anonymous calls	37
Figure 29 - Setting the Emergency Call Number.....	37
Figure 30 - Web Page Configuration BLF Function Keys	38
Figure 31 - Configuring the BLF List Feature.....	40
Figure 32 - BLF List NumberDisplay	40
Figure 33 - Web Page Web Recording	40
Figure 34 - Web SIP info recording	41
Figure 35 - Web Registration BroadSoft.....	43
Figure 36 - Web Page Specifies BroadSoft Server.....	43

Figure 37 - Web page with SCA turned on	44
Figure 38 - Web Page Setup Private Hold Function Key	44
Figure 39 - Web Page Registration SIP	46
Figure 40 - Web page language settings	48
Figure 41 - Web Page Setting Time & Date	49
Figure 42 - Web Screen Settings	50
Figure 43 - Phone Screensaver	50
Figure 44 - Web Blocking Call List List	53
Figure 45 - Headset Feature Settings on Web Page	54
Figure 46 - Web Auto Deployment Setup	58
Figure 47 - Web Page Online Upgrade Page	61
Figure 48 - Service Port Settings Screen	61
Figure 49 - Receipt Rule Registration Settings	68
Figure 50 - Dialing Rule Customization Settings	69
Figure 51 - Dialing Rules Table	70
Figure 52 - Web Beeps	80
Figure 53 - Web Cloud Phonebook Setup	82
Figure 54 - Web Filter Settings	85
Figure 55 - Web Filter Table	85
Figure 56 - Certificate Settings	86
Figure 57 - Device Certificate Setup	86
Figure 58 - Network Firewall Settings	87
Figure 59 - Deleting a Firewall Rule	88
Figure 60 - Screenshot of the web page	89
Figure 61 - Web Page Capture Tool	90

2 Table

Table 1 - Key Screen Printing Instructions.....	16
Table 2 - Status Alert and Notification Icons.....	16
Table 3 - Character Lookup Table	17
Table 4 - Key Descriptions	20
Table 5 - Call Patterns	27
Table 6 - List of BLF Function Key Subtype Parameters	38
Table 7 - Proxy Models	41
Table 8 - Intercom Mode.....	42
Table 9 - Web Multicast Parameters.....	42
Table 10 - SIP Hotspot Parameters	46
Table 11 - Time Parameter Settings	49
Table 12 - Quality of Service and Virtual LANs.....	56
Table 13 - Automatic upgrade.....	58
Table 14 - Online Upgrade	61
Table 15 - Server Ports.....	61
Table 16 - Line Configuration on the Web	62
Table 17 - 7 Dialing Methods for Phones.....	68
Table 18 - Dialing Rules Configuration Table	69
Table 19 - Setting Line Global Configurations on Web Pages	71
Table 20 - Common phone feature settings on web pages	71
Table 21 - Voice Settings on the Web	75
Table 22 - Web Multicast Parameters	76
Table 23 - Time/Date Setting Parameters on the web page	77
Table 24 - Time Management.....	78
Table 25 - Shortcut Configuration	83
Table 26 - Softkey Configuration.....	84
Table 27 - Network Firewalls.....	87
Table 28 - Failure Cases	91

3 Safety Instructions

Please read the safety instructions carefully before installing and using the phone. This is the key to ensuring the safe and reliable operation of the equipment.

- Use the power adapter specified for the product. If you need to use a power adapter provided by other manufacturers for special circumstances, please make sure that the voltage and current of the provided adapter are in accordance with the specifications of this product, and it is also recommended to use products that have passed the safety certification, or it may cause fire or electric shock accidents. When using this product, do not damage the power cord, force it to twist, stretch pull and bundle, or press it under heavy objects or pinch it between objects, or it may cause fire or electric shock accidents due to broken power cord.
- Before using the product, please make sure that the temperature and humidity of the environment meet the working requirements of the product. (Moving the product from an air-conditioned room to a natural temperature may cause condensation on the surface or internal components of the product, so it is necessary to wait for the product to dry naturally before turning on the power to use it).
- Do not disassemble or repair the product by non-technical service personnel, improper repair may cause electric shock, fire and other accidents, and the warranty of your product will be invalidated.
- Do not put metal foreign objects such as large pins, wire, etc. into vents or crevices. Otherwise, it may cause electric shock and other injuries caused by the current passing through metal foreign objects. If foreign objects or similar metal objects fall into the product, it should be stopped in time.
- Do not dispose of or store the plastic bag used for packaging within the reach of young children, as this may cause blockage of the nose and mouth, which may lead to choking, by putting it over the head of the child.
- Please follow the instructions in this product manual to use this product correctly, long-term abnormal operation may lead to product damage and safety hazards.

4 Summarize

4.1 Summarize

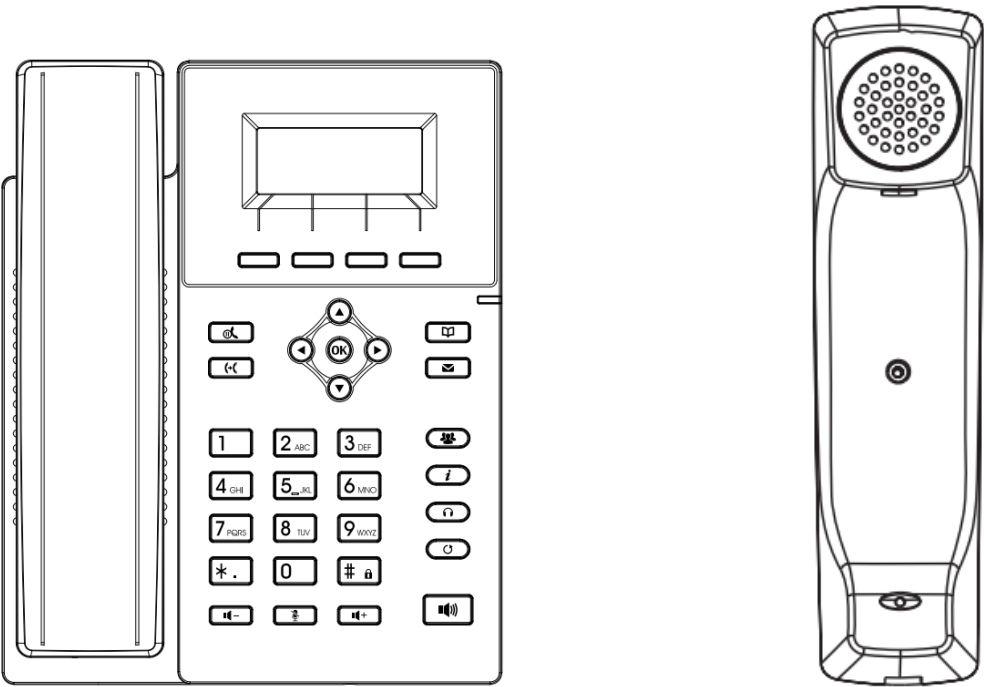
The NRP1302G/P is a VoIP phone for small and medium-sized businesses and homes designed for the Internet age. The device delivers a great user experience for home and office users with its simple design. The device is more than just a desk phone, it's a piece of jewelry for your living room or office.

This device is the latest generation of VoIP, inheriting many excellent features, such as high-definition voice, headset and high-performance echo-canceling full-duplex speakers, 100 Gigabit Ethernet, QoS, encrypted transmission, auto-configuration, etc.; brand new system, smooth operation, flat interface settings and many other advantages.

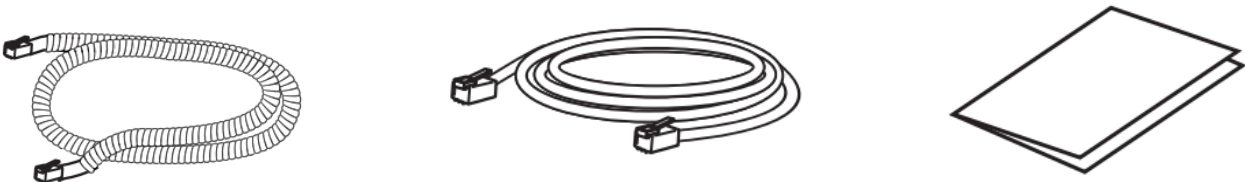
For business users, the device is a cost-effective office equipment that is environmentally friendly and provides convenient operation; for home users, the device is a very efficient communication equipment. It will be an ideal choice for business users and home users who pursue high quality and high performance.

In order to help some interested users to better understand the details of the product, this user manual can be used as a reference guide for the use of the equipment. This document may not apply to the latest version of the software, if you have any questions you can use the help screen that comes with the device or download and update your user manual from the manufacturer's official website.

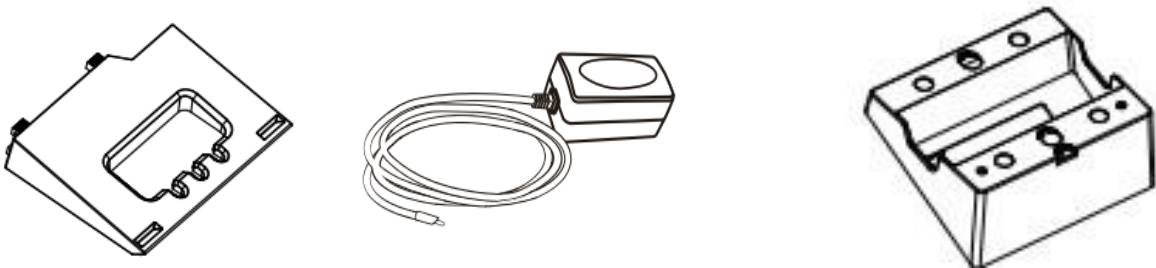
4.2 Accessory Description



Phone. Handset.



Handset Curve Netting Quick Installation Guide



Bracket Power adapter (optional) Wall bracket (sold separately)

5 Installation wizard

5.1 PoE and the use of external power adapters

The NRP1302G/P devices support two power supply methods, external power adapters and the switch power over Ethernet (PoE) mechanism.

The PoE power supply method saves space and the cost of additional power outlets. The phone is connected to the PoE switch via a network cable to provide power and data transfer. By connecting to the PoE switch of the UPS system, it can continue to work even if there is a power outage, just like a traditional PSTN phone powered through the phone line.

Users who do not have PoE devices can also use traditional power adapters. If the phone is connected to both the PoE switch and the power adapter, the PoE power supply is prioritized. If the PoE power supply fails it will be switched to the power adapter power supply.

To ensure proper operation of the device, use a manufacturer-specified power adapter and a PoE switch that meets the device standards.

Desktop and wall mounting method

The device supports two types of mounting, desktop and wall-mounted. If the phone is placed on a desktop for use, follow the instructions in the picture below to install the phone.

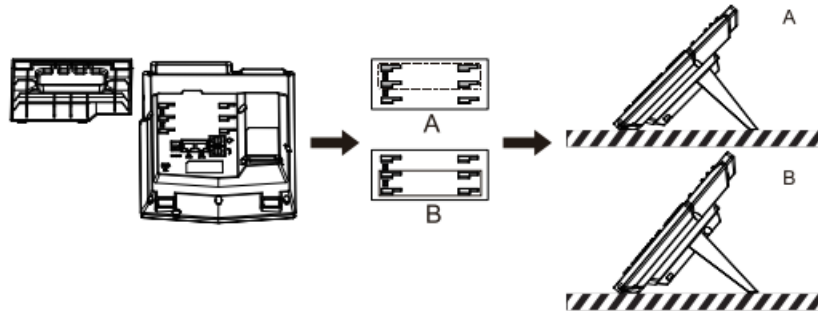


Figure 1 - Desktop Phones Installation

If you are mounting the phone on a wall for use, follow the instructions in the figure below.

Note: Wall mounting bracket needs to be purchased separately

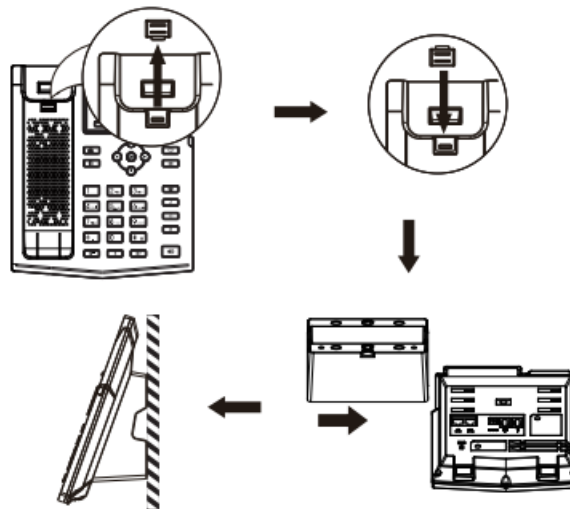


Figure 2 - Desktop to Wall Mount Conversion

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

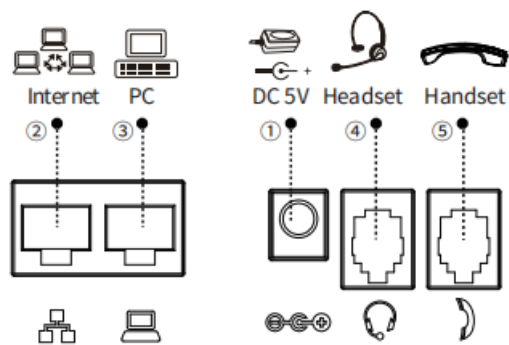


Figure 3 - Device Connections

Serial number	Connector	Clarification
①	Power connector	Connecting the power adapter
②	Network interface	Connecting to a LAN or the Internet
(iii)	PC port	Network interface to the computer
④	Headphone jack	Connecting Headphones
⑤	Earpiece connector	Connecting the microphone handset

6 Appendice

6.1 Appendix I - Icons

Table 1 - Key Screen Printing Instructions

	Redial
	Telephone book
	Hands-free calling
	Microphone mute
	Lower the volume
	Increase the volume
	Hold
	Earphones
	Voice message
	Tripartite meeting
	Call transfer

Table 2 - Status Alert and Notification Icons

	Call-out instruction
	Inbound caller ID
	Hold
	Network disconnection
	Enabling vlans
	Enable VPN
	Keyboard locked.
	Call forwarding record
	Outgoing records
	Inbound call log
	Missed call record
	Short message
	There's a new voice mail.
	Activate Do Not Disturb
	Call forwarding activated
	Turn on auto-answer
	Hands-free mode
	Headphone mode

	Joystick mode
	Microphone mute
	Voice quality of an intercom call
	Voice encryption for calls
HD	Talking voice hd
	Call recording
	Enable SIP hotspot

6.2 Appendix II - Keyboard Character Lookup Table

Table 3 - Character Lookup Table

Icon	Input method	Keystrokes	Character entry per single press
	numeric		1
			2
			3
			4
			5
			6
			7
			8
			9
			0
			*. +
			#
	lowercase		@:;()<>
			a b c
			d e f
			g h i
			j k l
			m n o
			p q r s
			t u v
			w x y z
			(space)
			. , */+ -: _ =

		#	# ^! &\$%
ABC	capital letters	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)
		*	. , */+ -: _ =
		#	# ^! &\$%
2aB	Numeric/alpha betic keypad	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 I 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
		*	. , */+ -: _ =
		#	# ^! &\$%
Abc	Upper/lower case keyboard	1	1
		2	A B C a b c
		3	D E F d e f
		4	G H I g h I
		5	J K L j k I
		6	M N O m n o
		7	P Q R S p q r s
		8	T U V t u v
		9	W Z Y X w z y x
		0	0
		*	. , */+ -: _ =
		#	# ^! &\$%

7 User orientation

7.1 Key Description

7.1.1 Key Description

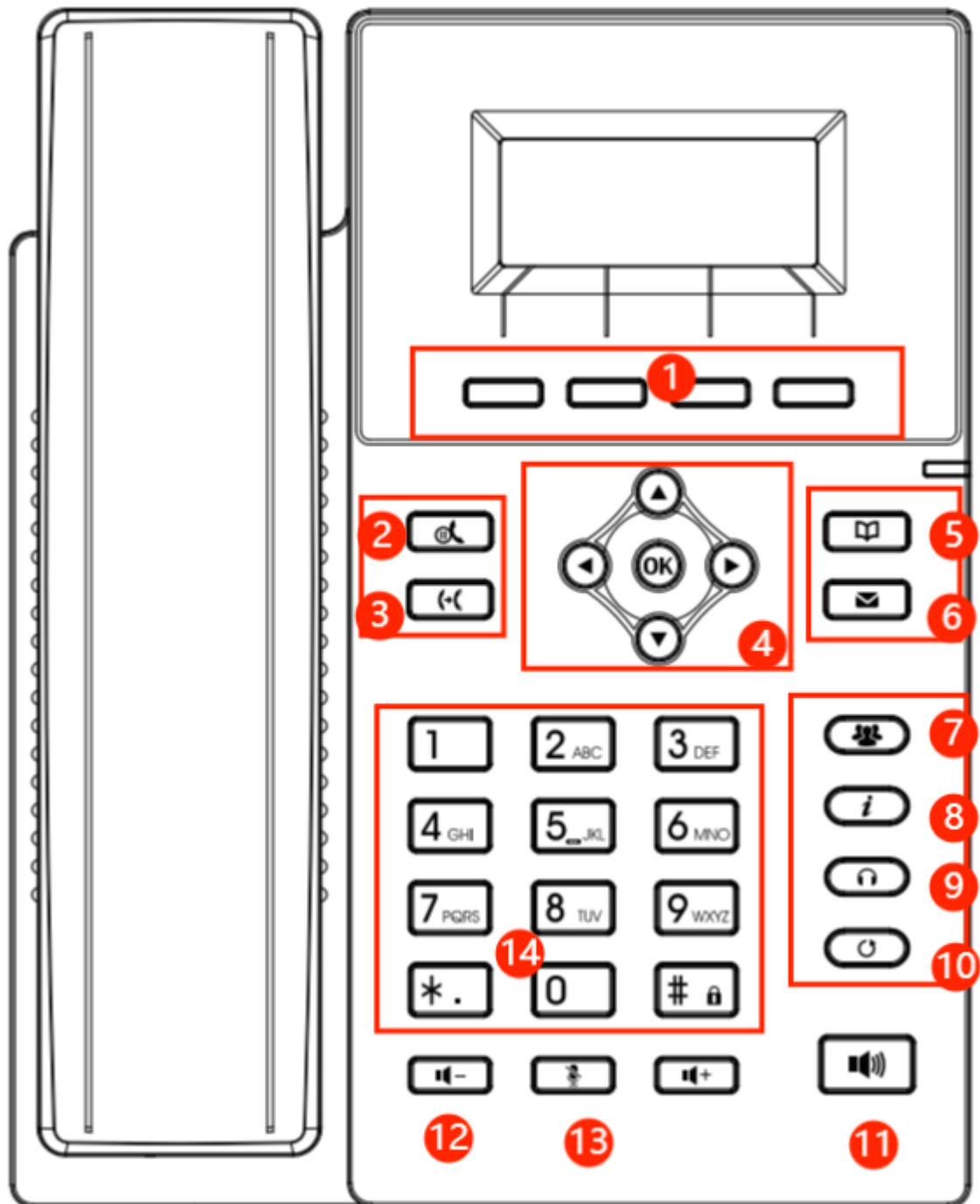


Figure 4 - Key Description

The image above shows the key layout of the phone. Each key provides its own specific function. The user can operate the handset by referring to the documentation describing the keys in the illustrations in this section.

Table 4 - Key Descriptions

Key Index	Key Name	Description
①	Function menu keys	These four buttons provide the menu functions displayed on the screen opposite to them
②	Maintenance	During a callPressing the "Hold" button allows the user to put the call on hold.Press again to release the hold and return to normal call status.
③	Transfer key	Press the "Transfer" buttonPress the "Transfer" button. The user can transfer the current call to another number.
④	Navigation key + OK key	The user can change the line or move the cursor in the screen list by pressing the up/down navigation keys. In some settings and text editing pages, users can press the left/right navigation keys to change options or move the cursor left/right in the screen list.
⑤	Contacts key	Pressing the "Contacts" button, users can enter the Contacts interface and select a contact to call.
⑥	Voice mail button	Press "Voice Mail"., and then press The user enters the SMS and voice message list interface.
⑦	Conference key	Pressing the "Conference" button during a two-way call allows the user to transfer the call to a conference, and then press it again to contact the conference and return to the normal call status.
⑧	Status key	Users can press this key to view the status information of the device
⑨	Headphone button	Users can press this button to open the headphone channel
⑩	Redial	Pressing the "Redial" button allows users to redial the last number dialed.
⑪	Speakerphone button	Users can press this button to turn on the hands-free speakerphone audio channel.
⑫	Volume up and down keys	In standby mode, ringer and ringer configuration interface, press this button to decrease/increase the ringer volume; in call or tone adjustment interface, press this button to decrease/increase the volume.
⑬	Mute button	During a call.The user can mute the microphone by pressing this button.
⑭	Standard keys on the	These 12 handset standard keys provide standard handset key functions. At the same time, certain keystrokes long

	Handset	Pressing can be triggered to provide special functions. Key #- Press and hold this key to lock the unit (turns on keypad lock configuration).
--	---------	--

7.2 Use of handles, headphones, hands-free speakers

■ Use of the handle

Regarding the use of the handle, the user can pick up the handle to dial the number or dial the number first and then pick up the handle and the number will be dialed. When the speaker or headset channel is on, the user can switch the microphone audio channel by picking up or putting down the handle.

■ Use of hands-free speakerphone

Regarding the use of the speakerphone, the user can either dial the number by pressing the speakerphone key first, or dial the number first and then press the speakerphone key. When the voice channel of the joystick is open, the user can switch the microphone audio channel by pressing the hands-free speaker button.

■ Use of headphones

Regarding the use of the headset, by default, the user can open the headset channel with a customized headset button. As with the joystick and hands-free speakerphone modes, users can turn on the headset before or after selecting a number to dial.

7.3 Standby mode



Figure 5 - Screen Layout / Default Home Screen

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

The upper half of the main screen area shows the status of the device, information and data that can be edited and manipulated (e.g., voice messages, missed calls, auto-answer, do not disturb, lock status, network connection status, etc.).

The lower half of the area is the function menu buttons, which are also the first level of the function menu buttons, through which the user can operate the handset.

The user can restore the phone to the default standby screen interface by picking up and putting down the handle.

Icons are described in [6.1 Appendix I](#) are described.

On some screens, the screen cannot be fully displayed due to too many items or too long text. They will be combined into a list that matches the scroll icon. If the user sees a scroll icon, they can use the up/down navigation keys to scroll through the list.

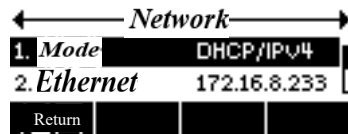


Figure 6 - scroll icon

7.4 phone state

The phone status includes the following information about the phone:

- Network Status:
 - VLAN ID
 - IPv4 or IPv6 status
 - IP address
 - network paradigm
- Device Information:
 - Mac address
 - Model
 - Hardware version number
 - software version number
 - Phone Memory
 - System runtime
- SIP account information:
 - SIP users
 - SIP Account Status (Registered / Unsubmitted / Trying / Timeout)
- TR069 Connection Information (displayed only in the state of the phone interface)
 - Users can check the status of the phone through both the phone interface and the web interface.
- Phone Interface: When the phone is on standby, press [Menu] >> [Status] and select the option to view the corresponding information, as shown:

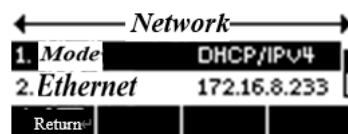


Figure 7 - Phone Handset Status

- Web interface: refer to [7.5 Web Management](#). Log in to the phone web page, enter the [System] >> [Information] page, check the status of the phone, as shown in the figure:

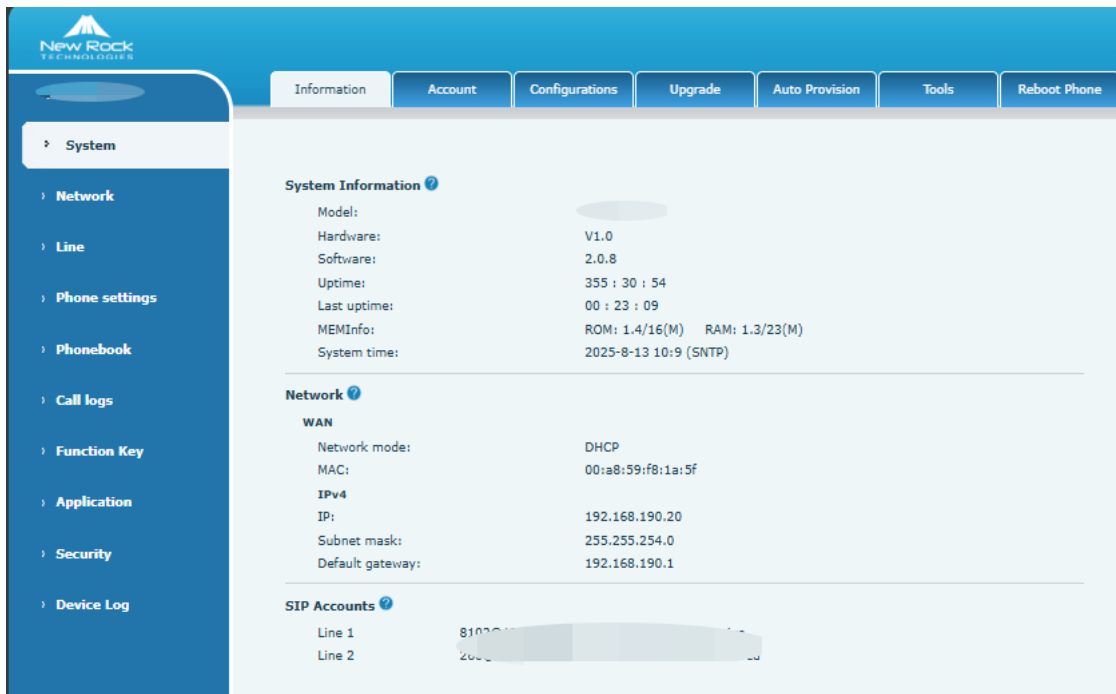


Figure 8 - Web Phone Status

7.5 Web management

Users can manage and operate the phone using the phone's web dialog. The user first needs to open the phone's web page by typing the phone's IP address into a browser. Users can check the IP address of the phone by pressing [Menu] >> [Status].

Open the browser, enter the IP address of the phone. The first thing you will see is the login page of the phone. The first thing you see is the login page of the phone.

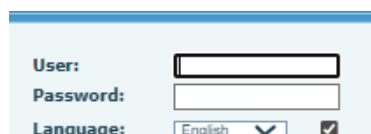


Figure 9 - Web landing page

Users must enter their username and password correctly to log in to the web page. **The default username and password are both "admin"**. For specific details on operating the web pages, please refer to [11 Web Page Configuration](#)

7.6 Network setup

The phone supports network connection: wired network connection, users need to choose the appropriate connection according to their own situation. The phone utilizes an IP network connection to provide the service. Unlike traditional phones based on line-circuit technology, IP phones exchange packets and data based on the phone's IP address connected to each other over a network.

To enable the phone, the network configuration must first be properly configured. To configure the

network the user needs to find the phone function menu button [Menu] >> [Advanced] >> [Network] >> [Network Settings].

The default password to access Advanced is "123".

Notes! If the user sees the  "Network not connected" icon at the top of the screen, it means that the network cable is not connected to the network port of the phone. Check that the network cable connects the phone to the network switch, router or modem.

The phone supports three network types, IPv4/IPv6/IPv4&IPv6.

There are three common types of IP configuration for IPv4

- DHCP - This is the mode that automatically gets the network configuration from the server. The user does not need to manually configure any Parameters. Suitable for most users.
- Static IP Configuration - This option allows the user to manually configure each IP parameter, including IP address, mask, gateway and primary and alternate DNS servers. This usually applies to some professional network user environments.
- PPPoE - This option is usually available for users who connect to the network through a broadband service account. To establish a PPPoE connection, the user should provide a username and password provided by the operator.
- The default configuration of the phone is auto-configured network mode

There are two common types of IP configuration for IPv6

- DHCP - This is the mode that automatically gets the network configuration from the server. The user does not need to manually configure any Parameters. Suitable for most users.
- Static IP Configuration - This option allows the user to manually configure each IP parameter, including IP address, mask, gateway and primary and secondary domain names. This usually applies to some professional network user environments.

For details on configuration and use, see [10.5.2.1 Network Settings](#)

7.7 Line Configuration

The phone must have at least one line properly configured to provide telephone service. The line configuration is similar in nature to the virtualization of the SIM card of a cell phone with a stored service provider and phone account authentication. When the phone applies these configurations, the phone automatically registers with existing information service providers, just as you can insert a SIM card into any cell phone, and the phone applies services based on the information in the SIM card rather than the phone itself.

Users can make line configurations on the phone interface or web interface by entering the corresponding information in the registered address, registered user name, registered password and SIP user, display name and registered port respectively, which are provided by the SIP server administrator.

- **Phone interface:** To manually configure a line, users can configure it by long-pressing the line key, or through the function menu button [Menu] >> [Advanced] >> [Account] >> [Line 1] / [Line 2] >> [Settings],

click to confirm and save the configuration.

Note! Users must be password-authenticated when accessing Advanced. The default PIN is 123

Please refer to the image below.

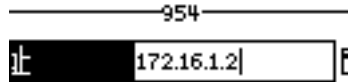


Figure 10 - Phone Line SIP Address and Account Information

- Web interface: After logging into the phone webpage, go to[Line] >>[SIP], select SIP1/SIP2 to configure, and click Submit to complete the registration after the configuration is completed, as shown below:

A screenshot of a web-based configuration interface for SIP settings. At the top, there's a 'Line' dropdown menu showing '8102@SI'. Below it is the 'Register Settings >>' section. This section includes 'Line Status' (Inactive), 'Activate' (checkbox), 'Username' (8102), 'Authentication User' (8102), 'Display name' (8102), 'Authentication Password' (masked with dots), 'Realm' (empty), and 'Server Name' (empty). Each input field has a help icon. Below this are two columns for 'SIP Server 1' and 'SIP Server 2'. Each column has fields for 'Server Address', 'Server Port', 'Transport Protocol' (set to UDP), 'Registration Expiration' (800 seconds), 'Proxy Server Address', 'Proxy Server Port' (5060), 'Proxy User', and 'Proxy Password'. At the bottom, there are links for 'Basic Settings >>', 'Codecs Settings >>', 'Advanced Settings >>', and 'SIP Global Settings >>'. An 'Apply' button is at the very bottom.

Figure 11 - Web Line Registration

8 Basic Functions

8.1 Make a telephone call

■ Default Line

The phone provides 2SIP lineservices. If both 2 lines are successfully configured, the user can make or receive calls on either line. If the user sets a default line, the current defaultnumber or nameusedwill be displayed in the upper left corner of the screen interface. To enable or disable the default line function, users can complete the setting through [Menu] >> [Features] >> [General] or on the webpage ([Webpage] >> [Phone Settings] >> [Feature Settings] >> [Basic Settings]).

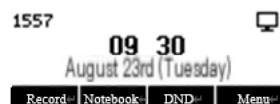


Figure 12 - Default Line

■ dialing method

Users can dial a number in several ways:

- direct dial
- Select a contact number to dial from the phonebook (see [10.2 Phonebook](#))
- Select a contact number to dial from the cloud phonebook (see [10.2.3 Remote Phonebook](#))
- Select a contact number from the call log to dial (see [10.3 Call Log](#))
- Redial the last number dialed

■ Open the voice channel after dialing the number

When the call is outgoing, the user can dial the number by one of the above methods. After confirming that the number is correct, the user can press the [Dial Out] button on the Function menu, or press the speakerphone button (to turn on the speaker or headset), or pick up the microphone.



Figure 13 - Dialing before turning on the voice channel

■ Dialing a number after opening the voice channel

The other method is the traditional method, first open the voice channel of the phone (pick up the handle or turn on the hands-free speaker or headset mode or press the line key) and then dialing, after the dialing is completed, the user can dial out by pressing the [Dial Out] button on the Function menu or the [OK] button. Phones can also send numbers overtime.



Figure 14 - Dialing a number with the voice channel turned on

■ Cancel Call

When calling the number, the user can press the [End] button or close the voice channel by putting back the handle/pressing the handsfree button.

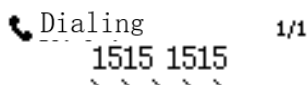


Figure 15 - Calling Numbers

8.2 Take a telegram

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

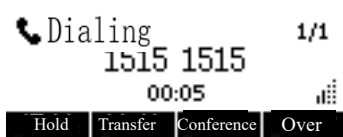


Figure 16 - Answering Incoming Calls

Users can open the headset/handsfree channel to answer calls by picking up the handle or pressing the [Answer] button or pressing the handsfree button. To reject a call, the user should press the [Reject] button.

8.2.1 Call screen

When the call is established, the user will see the Call Mode screen as shown below.

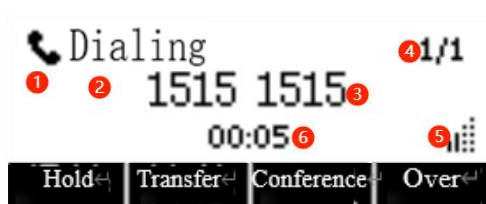


Figure 17 - Call Mode Interface

Table 5 - Call Patterns

Serial Number	Name	Description
①	voice channel	The icon shows the voice channel mode being used.
②	Call Status	Call status of the current call - in call
③	end-to-end communication	The name (front) and number (back) of the other party to the call.
④	airtime	The length of the call that lasts after the call is established.
⑤	How many ways to	Displays how many calls currently exist on the device.

	talk?	
⑥	voice quality	Displays the current call voice quality.

8.2.2 Make/receive a second call

The phone can support multiple calls. When there is an established call, the user can still answer another incoming call on either line or make a second call on either line.

■ Second caller.

When there is another incoming call during the call, the incoming call will be waiting for the user to answer. The user will see an interface similar to the one shown in Figure 23. The phone will not ring, but the phone will play a wait tone on the currently calling voice channel while the LED indicator will flash red. The user can answer or reject the call as if it were a normal call. If a new call is answered, the first call will be automatically put on hold.

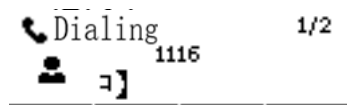


Figure 18 - Second Incoming Call Screen

■ Second outgoing

When the user dials the second call on the default line, the user can press the [Transfer] / [Conference] button to dial the second call or press the line button to specify the line to dial the second call. Another way to dial a second line call is to dial the BLF number set in the second screen or a number selected by the speed dial feature. When a user dials a second call using the above method, the first call can be held manually or automatically when the second call is dialed.

■ Switching between two calls

When a two-way call is established, the user will see the Dual Call screen screen as shown below.

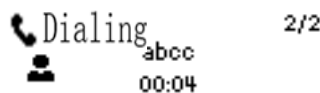


Figure 19 - Dual Talk

The user can switch between the interface by using the up/down navigation keys, the By pressing[Resume]. button to switch calls.

■ end a call

The user can hang up the current call by closing the voice channel or pressing the [End] button. The phone will return to the hold state in single call mode. Users can also recover the data by pressing [Resume] key to resume the current call.

8.3 End of call

When the user has finished talking, the user can put the handle back into the handset, press the

speakerphone or SoftkeyEnd] keyClose the voice channel to end the call.

Note! When the phone is on hold, you must press the Restore button to return to the call status before ending the call.

8.4 Reappoint

- Redial the last outgoing number:
While the phone is in standby, press the Redial key and the phone will call out the last dialed number.
- Use the redial key to call out any number:
The phone goes off-hook and enters the number, presses the redial key, and the phone calls out the number on the dial pad.
- Press the Redial key to enter the call log:
Log in to the phone webpage, go to [Phone Settings]>> [Feature Settings]>> [Redial Settings], check Redial to enter the call record, press the Redial button in standby to enter the call record page, and then press it again to call out the currently located number.

8.5 Dial-up search

The phone is enabled by default dialing query function, take off the phone to dial, enter two or more than two numbers, the dialing interface will automatically match the call log, contacts in the list of numbers, use the navigation key up and down keys can select the number, press the call out key or timeout call out.

8.6 Auto-answer

Users can enable the auto-answer feature on the handset, so that the handset will be able to answer calls automatically when they come in. Auto-answer can be enabled by distinguishing between lines.

Users can activate the auto-answer function from the handset interface or the web interface.

- **Phone Interface:**
Press the [Menu] >> [Function] >> [Auto Answer] button;
Press the key to select the line, use the left/right navigation keys to turn on/off the auto-answer option, set the auto-answer time, the default is 5 seconds
Press the [Confirm] button to save when finished;
The  icon in the upper right corner of the screen indicates that auto-answer is turned on.
- **Web Interface:**
Log in to the phone webpage, go to [Line] >> [SIP], select [SIP Route] >> select [Basic Settings], start Auto Answer, set the Auto Answer time and click Submit.

8.7 Unmute

You can turn on silent mode during a call to turn off the microphone of the phone so that the other party

cannot hear the local voice. Under normal circumstances, the silent mode is automatically turned off with the end of a call. You can also turn on the hold mute feature from any screen (e.g. the idle screen) to automatically mute the ringer on incoming calls.

Silent mode can be turned on in all call modes (joystick, headset or hands-free).

8.7.1 In-call mute

- Press the mute button on the handset during a call: 
The call screen displays a red mute icon as shown:
- To unmute a call: press  on the handset again to unmute the call. The mute icon is no longer displayed on the call screen.

8.7.2 Mute when ringing

- To turn on ringer mute: press the mute button while the phone is in standby: 
The ringer mute icon is displayed in the upper right corner of the handset , and the handset displays the Caller ID screen when there is an incoming call but does not ring.
- Cancel ringer mute: In standby or incoming call screen, press the Mute button  or volume up button  again to cancel ringer mute, after canceling, the mute icon is no longer shown in the upper right corner .

8.8 Call Hold/Resume

The user can press the [Hold] button to hold the current call, and at the same time this button will change to [Resume] button, the user can press [Resume] button to resume the call.

8.9 Distraction-free

Users can enable the Do Not Disturb (DND) feature on the phone to reject incoming calls (including call waiting). Do Not Disturb can be set on/off separately by SIP line.

Enable/disable the handset's Do Not Disturb for all lines as follows:

- Phone interface: default standby state.
 - 1) Standby Press the [Do Not Disturb] button to turn on the handset do not disturb function.
 - 2) Press the [Do Not Disturb] button again to turn off Do Not Disturb.

If the user wishes to enable/disable the do-not-disturb feature on a specific line, the user can set the do-not-disturb feature on the Configure Line page.

- 1) Press [Menu] >> [Function] >> [Do Not Disturb] button to enter the edit page of [Do Not Disturb].
- 2) Adjust the no-disturbance mode and status by selecting the line with the left/right navigation keys.
Press the [Confirm] button to save when finished.
- 3) The user will see the DND icon in the status bar and the SIP line is enabled for the Do Not Disturb mode.

Users can also use the no-disturb timer, which is set so that the no-disturb feature is automatically turned on during the time range.

- Web interface: Go to [Phone Settings] >> [Feature Settings] >> [DND Settings] to set the type of Do Not

Disturb (Off, Phone, Line), and the DND Timing function.

Figure 20 - Setting DND Options

Users turn on DND for a specific line on the webpage: Go to [Line] >> [SIP], select a certain [Line] >> [Basic Settings], and enable Do Not Disturb.

Figure 21 - Enabling Do Not Disturb for a Line

8.10 Call forward

Call Forwarding is a feature that forwards inbound calls to other numbers. Users can set up call forwarding for each line separately.

Call Forward There are three types of

- **Unconditional Forwarding** - Forward all inbound calls to the configured number.
- **Busy Line Forwarding** - When the user is busy, incoming calls are forwarded to the configured number.
- **No Answer Forwarding** - When a user times out with no answer, inbound calls are forwarded to the configured number..
- The phone interface: With the phone in the default standby state, the
 - 1) Press [Menu] >> [Function] >> [Call Forward] button, select the line by up/down navigation key, and press [Confirm] button to set Call Forward.
 - 2) Select the type of call forwarding by using the up/down navigation buttons. Click the [Confirm] button to configure the callforwardand delay time.
 - 3) Select on/off with the left/right navigation buttons.
 - 4) Navigate through the set Parameters with the up/down navigation keys and enter the required information. When finished, press[Confirm].button to save the changes made.
- Web interface: Enter [Line] >> [SIP], select a certain [Line] >> [Basic Settings], set the forwarding type and number and time.

Line: 8102@SI

Register Settings >>

Basic Settings >>

Enable Auto Answering:	☐	Auto Answering Delay:	5 (0~120)second(s)
Call Forward Unconditional:	☐	Call Forward Number for Unconditional:	
Call Forward on Busy:	☐	Call Forward Number for Busy:	
Call Forward on No Answer:	☐	Call Forward Number for No Answer:	
Call Forward Delay for No Answer:	20 (0~120)second(s)	Transfer Timeout:	0 second(s)
Conference Type:	Local	Server Conference Number:	

Figure 22 - Setting Up Call Forwarding

8.11 Call transfer

When a user is on a call with the other party and wants to transfer the call to another number, here are three ways to transfer the call, Blind Transfer, Attendance Transfer and Semi-Attendance Transfer.

- Blind transfer: Transfer the call directly to the other party without consulting them.
- Semi-Attended Transfer: transfers the call to the other party when the ringback tone is heard.
- Attend Transfer: Transfer the call to the other party when they answer the call.

Note! For more transfer settings, please refer to [12.6 Lines >> Receiving Number Rules](#) Please refer to [Line 6 >> Rules for receiving numbers](#).

8.11.1 Spineless

During the call, the user presses the function menu key [Transfer] button or the transfer button “” on the handset, inputs the number to be transferred or selects the number by pressing the Contacts button or the History button, presses the transfer button or “” again to blindly transfer to the third party, and after the third party rings, the handset displays a successful transfer and hangs up.

8.11.2 Half attended

During a call, the user presses the function menu key [Transfer] button or the transfer button “” on the handset, enters the number to be transferred or selects the number by pressing the Contacts button or the History button, presses the Call button, and when the third party does not answer, presses Transfer in the call interface to carry out a half-attendance transfer or presses End to cancel the half-attendance transfer.

8.11.3 Attending

Attendance transfer, also known as "courtesy mode," is a call transfer that occurs by dialing the other party and waiting for them to answer the call.

The call follows the same steps. In dual call mode, press the [Transfer] button to transfer the first call to the second call.

8.12 Call Waiting

- Enable Call Waiting: new incoming calls can be accepted during a call.

- Turn off Call Waiting: new incoming calls will be automatically rejected with a busy tone.
- Enable Call Waiting Tone: When you receive a new incoming call while you are on a call, the phone will sound a beep-beep tone.

Users can enable/disable the call waiting feature in both the handset interface and the web interface.

- Phone interface: Press [Menu] >> [Function] >> [Call Waiting] >> [Call Waiting Settings], and the navigation key left/right to turn on/off Call Waiting and Call Waiting Tone.
- Web interface: Go to [Phone Settings] >> [Feature Settings] >> [Basic Settings] to turn on/off call waiting; Go to [Phone Settings] >> [Feature Settings] >> [Tone Settings] to turn on/off call waiting tone.

8.13 Sessions

8.13.1 Local meeting

To conduct a local conference, users need to log in to the webpage, enter [Line] >> [SIP] >> [Basic Settings], set the conference mode as local (the default is local mode), as shown in the figure:

The screenshot shows the 'Basic Settings' web interface. On the left, there are checkboxes for 'Enable Auto Answering', 'Call Forward Unconditional', 'Call Forward on Busy', and 'Call Forward on No Answer', each with a help icon. Below these is a text input for 'Call Forward Delay for No Answer' set to '20' with a range '(0~120)second(s)' and a help icon. At the bottom left, 'Conference Type' is set to 'Local' with a dropdown arrow and a help icon. On the right, there is a text input for 'Auto Answering Delay' set to '5' with a range '(0~120)second(s)' and a help icon. Below this are three text inputs for 'Call Forward Number for Unconditional', 'Call Forward Number for Busy', and 'Call Forward Number for No Answer', each with a help icon. At the bottom right, there is a text input for 'Transfer Timeout' set to '0' with a range 'second(s)' and a help icon, and a text input for 'Server Conference Number' with a help icon.

Figure 23 - Setting up a local conference

Two ways to create a local meeting:

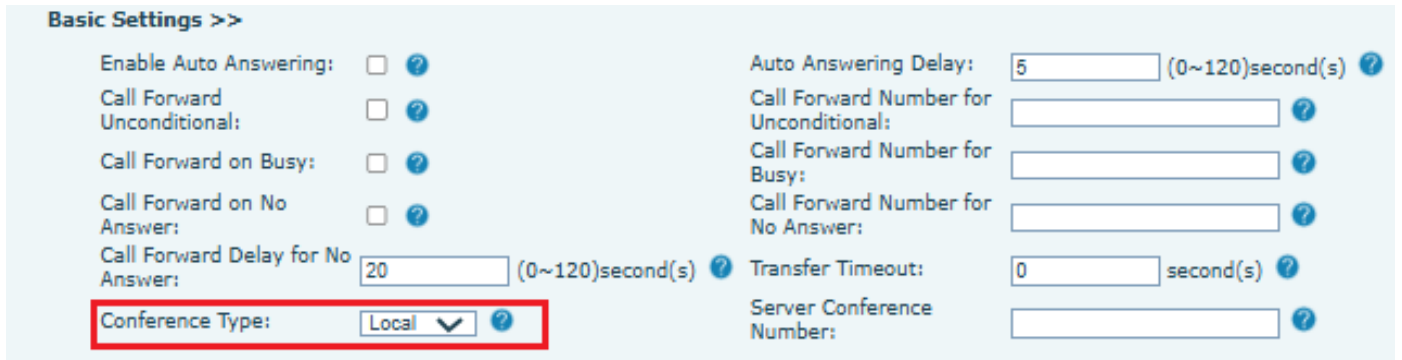
- 1) The device has two-way calls, press the conference button on the call interface, select the other way number that already exists when selecting the conference number, press the confirmation button, then you can establish a local three-way conference. When the device is in a three-way conference, then come back to one-way call, answer and press conference, then you can join the 4-way conference. The same could be done for 5-party meetings, 6-party meetings.
- 2) The device exists all the way to the call, press the conference key of the call interface, enter the number you need to join the conference, press the call; the other end of the answer, press the conference key again, you can establish a local three-party conference, the same reason to join the four-party conference, the same reason to join the five-party conference, six-party conference. As pictured:

Notes: During a conference, press the Separate key to separate the conference and press the End key to end the call.

8.13.2 Web conference

Users need server support for web conferencing.

Login to the webpage, go to [Line] >> [SIP] >> [Basic Settings], set the conference mode as Server Mode (default is Local Mode), and set the server room conference number (please consult your system administrator), as shown in the figure:



Basic Settings >>

Enable Auto Answering:	☐	?	Auto Answering Delay:	5	(0~120)second(s)	?
Call Forward Unconditional:	☐	?	Call Forward Number for Unconditional:			?
Call Forward on Busy:	☐	?	Call Forward Number for Busy:			?
Call Forward on No Answer:	☐	?	Call Forward Number for No Answer:			?
Call Forward Delay for No Answer:	20	(0~120)second(s)	Transfer Timeout:	0	second(s)	?
Conference Type:	Local	?	Server Conference Number:			?

Figure 24 - Web Conferencing

Ways to join a server meeting:

- Multi-party call server conference room number, enter the password to access the conference room all.
- The two phones have established an ordinary call, press the conference key to invite a new member into the conference, and follow the voice prompts.

Note: The maximum number of participants in a server meeting varies from server to server.

8.14 Call residing

Call Residency requires server support, please consult your system administrator for support.

When you are on a call, if it is inconvenient to answer the call at this time, you can press the configured park button to park the call; after successful parking, you can press the configured park button to resume the call in other devices.

Set the call park button.

Phone interface: Enter the function key setting interface through [Menu] >> [Settings] >> [Keyboard Settings] >> [Function Key Settings], set the function key type as Memory Key, the subtype as Call Resident, the value as the call resident number of the server, and set the corresponding SIP line.

Web Interface: Login to the web page of the phone, go to [Shortcut Key] >> [Softkey Setting] >> [Soft DSS Key Setting] page, select a soft key, set the function key type as Memory Key, the subtype as Call Resident, the value as the server's Call Resident Number, and set the corresponding SIP line.

SoftKey Settings

Softkey Mode:

Softkey Exit Style:

Screen:

Unselected Softkeys

- Call Back
- Join
- Voice Mail
- Local Contacts
- Pickup
- Dsskey1
- Dsskey2
- Dsskey3
- Dsskey4
- Dsskey5
- Dsskey6
- Dsskey7

Selected Softkeys

- Dial
- 2aB
- delete
- exit

Soft DSS Key Settings

Key	Type	Name	Value	Subtype	Line	PickUp Number
DSS Key 1	Key Event			Phonebook	AUTO	
DSS Key 2	None			None	AUTO	
DSS Key 3	None			None	AUTO	
DSS Key 4	None			None	AUTO	
DSS Key 5	None			None	AUTO	
DSS Key 6	None			None	AUTO	
DSS Key 7	None			None	AUTO	
DSS Key 8	None			None	AUTO	
DSS Key 9	None			None	AUTO	
DSS Key 10	None			None	AUTO	

Figure 25 - Web Page Setup Call Dwell

8.15 Rush to pick up

Robocall requires server support, please consult your system administrator for support.

You can use the robocall feature to answer incoming calls from other users. The phone is configured to snatch incoming calls by configuring the feature key as BLF and setting a snatch code.

Phone interface: Press [Menu] >> [Settings] >> [Keyboard Settings] >> [Function Key Settings] and select the function key to be set.

- Setting up the line, with a function key type of Memory Key and a subtype of BLF/NEW CALL, setting up the subscription number, and grabbing the pickup code
- The other phone is calling the subscription number and the other end is ringing.
- The phone will pick up the call and talk to it

Web Interface: Login to the phone web page, enter [Shortcut Key] >> [Softkey Setting] >> [Soft DSS Key Setting] page, select a certain soft key, set the memory key type as Memory Key, subtype as BLF/NEW CALL, and set the corresponding SIP line. and robocode

Key	Type	Name	Value	Subtype	Line	PickUp Number
DSS Key 1	Line			None	8102@SIP1	
DSS Key 2	Line			None	266@SIP2	

Figure 26 - Web page robocall setup

8.16 Anonymous call

8.16.1 Anonymous call

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

- You can see the anonymity under the phone [Menu] >> [Advanced] >> [Account] >> [SIP Line] >> [Advanced].
- The default is None, which is the off state, and there are two optional Parameters, RFC3323 and RFC3325.
- Select any one to enable anonymous calling.
- You can also enable anonymous calling on the webpage [Line] >> [SIP] >> [Advanced Settings].
- The setting to turn on anonymous calling also corresponds to the SIP line. That is, the setting under the SIP1 web page can only take effect on the SIP1 line.

Figure 27 - Web Anonymous Calling Enabled

The following is a call log of the phone receiving an anonymous call.

8.16.2 Prohibition of anonymous calls

The phone can be set to prohibit anonymous calls, that is, the other end to open the anonymous call to the local number will be rejected directly.

- In the phone [Menu] >> [Features] >> [Disable Anonymous Calls] click to enter will show all SIP lines list.
- Clicking the Softkey [Toggle] key or the [<] [>] key turns on disabling anonymous calls.

- Anonymous calls can also be blocked on the webpage [Line] >> [SIP] >> [Advanced Settings].
- Setting up a ban on anonymous calls also corresponds to the SIP line. That is, the setting under the SIP1 web page can only take effect on the SIP1 line.

The screenshot shows the 'Advanced Settings' for a SIP line. On the left, there are checkboxes for 'Enable Session Timer', 'Enable BLF List', 'Response Single Codec', 'Keep Authentication', and 'RTP Encrvption(SRTP)'. On the right, there are input fields for 'Session Timeout', 'BLF List Number', 'BLF Server', and 'Keep Alive Interval'. The 'Blocking Anonymous Call' checkbox is located at the bottom right and is currently unchecked.

Figure 28 - Web page settings to disable anonymous calls

8.17 Hotline

Support hotline dialing, after setting up the hotline dialing, directly pick up the handset, hands-free, headset, etc., the phone will automatically call according to the hotline delay time.

- In the phone [Menu] >> [Function] >> [Advanced] >> [Hotline] click into will show a list of all SIP lines.
- Then set the hotline according to each SIP line correspondingly. the default is off.
- Turn on the hotline, set the hotline number, and set the hotline delay time.
- Hotline can also be set on the webpage [Line] >> [SIP] >> [Basic Settings].
- Setting up the hot line also corresponds to the SIP line. That is, hotlines set up under the SIP1 web page can only take effect on SIP1 lines.

8.18 Emergency call

The Emergency Call feature is used when the keypad lock is enabled, and you can set the appropriate emergency call number on the handset. You can also call emergency services when the phone is locked.

1) Configure the emergency call number: Login to the phone webpage, go to [Phone Settings] >> [Feature Settings] >> [Basic Settings] webpage, set the emergency call code, if you need to set more than one emergency call code, please use "," to separate.

The screenshot shows the 'Basic Settings' for a SIP line. On the left, there are checkboxes for 'Allow IP Call', 'Enable Pre-Dial', and 'Block XML When Call'. On the right, there are input fields for 'P2P IP Prefix', 'Emergency Call Number', 'LDAP Search', 'Push XML Server', 'Contact As White List Type', 'SIP Notify', and 'Auto Resume Current'. The 'Emergency Call Number' field is set to '110'.

Figure 29 - Setting the Emergency Call Number

2) When the phone is set with keypad lock, you can call the emergency call number without unlocking it, as shown in the figure

9 Advanced Features

9.1 BLF

9.1.1 Configuring the BLF Function

- Web interface: Login to the phone web page, enter [Shortcut Key] >> [Softkey Setting] >> [Soft DSS Key Setting] page, select a soft key, set the function key type as memory key, and select BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF for the subtype.

The BLF/DTMF value is the subscribed number that sets the corresponding SIP line. The robocall number needs to be provided by the server, please refer to [8.16 Robocall](#) for more details, it can be left out.

Key	Type	Name	Value	Subtype	Line	PickUp Number
DSS Key 1	Key Event			Phonebook	AUTO	
DSS Key 2	None			None	AUTO	
DSS Key 3	None			None	AUTO	
DSS Key 4	None			None	AUTO	
DSS Key 5	None			None	AUTO	
DSS Key 6	None			None	AUTO	
DSS Key 7	None			None	AUTO	
DSS Key 8	None			None	AUTO	
DSS Key 9	None			None	AUTO	
DSS Key 10	None			None	AUTO	

Apply

Figure 30 - Web Page Configuration BLF Function Keys

- Phone Interface: Enter the function key setting interface through [Menu] >> [Setting] >> [Keyboard Setting] >> [Function Key Setting], set the function key type as Memory Key, the subtype as BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, BLF/DTMF, the value as the subscribed number, and set up the corresponding SIP line.

Table 6 - List of BLF Function Key Subtype Parameters

Subtype	Description under standby	In-call description
BLF/NEW CALL	Pressing this BLF key in standby will dial the number of this subscription.	When this BLF key is pressed during a call with another user, you create a new call all the way with the subscribed number.
BLF/BXFER	Pressing this BLF key in standby will dial the number of this subscription.	When this BLF key is pressed during a call with another user, you will blindly transfer the call to the subscribed number.
BLF/AXFER	Pressing this BLF key in standby	When this BLF key is pressed during a call

	will dial the number of this subscription.	with another user, you will transfer the call attendance to the subscribed number.
BLF/Conference	Pressing this BLF key in standby will dial the number of this subscription.	When this BLF key is pressed during a call with another user, you invite the subscribed number to join the conference.
BLF/DTMF	Pressing this BLF key in standby will dial the number of this subscription.	When this BLF key is pressed during a call with another user, the phone automatically sends the DTMF corresponding to the BLF key number.

9.1.2 Using the BLF Function

BLF, also known as "Busy Light Field", is used to alert the user to the status of the subscribed objects, and is used with the server to grab phone calls. BLF helps you to monitor the status of the other party (idle, ringing, in call, not on line).

BLF features are:

- Monitor the status of subscribed phones
 - Calling Subscribed Phones
 - Transferring Calls/Calls to Subscribed Numbers
 - Rip off calls from subscribed phones
- 1) Monitor the status of subscribed phones
 - 2) Calling Subscribed Phones

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

- 3) transferring calls/incoming calls to the subscribed number.

Referring to [Table 9 - BLF Function Key Subtype Parameter List](#), the BLF key can be used to perform Blind, Attendance and Semi-Attendance transfers for the current call, as well as to invite the subscribed number to join the call, send a DTMF, etc.

- 4) Rip off calls from subscribed phones

Configure the robocall number when configuring the BLF function key.

9.2 BLF List

BLF List Key is to put the numbers to be subscribed into a group on the server side, and the phone uses the URL of this group to subscribe uniformly, and according to the notify sent by the server, it parses out the specific information of each number, the number, the name, and the status. Then set the unoccupied MemoryKey to the BLF List Key. In the future, if the state of the subscribed object changes, change the corresponding shortcut led state.

Configure BLF List function: Login to the phone webpage, enter [Line] >> [SIP] >> [Advanced Settings] page, turn on BLF List, and configure BLF List number.

Code: _____

Enable Session Timer: ☐ ? Session Timeout: 0 second(s) ?

Enable BLF List: ☐ ? BLF List Number: _____

Response Single Codec: ☐ ? BLF Server: _____

Keep Alive Type: SIP Optic ? Keep Alive Interval: 60 second(s) ?

Keep Authentication: ☐ ? Blocking Anonymous Call: ☐ ?

RTP Encryption(SRTP): Disabled ?

Figure 31 - Configuring the BLF List Feature

Using the BLF List feature: After the configuration is complete, the phone will automatically subscribe to the contents of the BLF List group. Users can monitor, call and transfer the corresponding numbers by pressing the BLF List key.

Key	Type	Name	Value	Subtype	Line	PickUp Number
DSS Key 1	Key Event			Phonebook	AUTO	
DSS Key 2	BLF List Key			None	AUTO	
DSS Key 3	None			None	AUTO	
DSS Key 4	None			None	AUTO	
DSS Key 5	None			None	AUTO	
DSS Key 6	None			None	AUTO	
DSS Key 7	None			None	AUTO	
DSS Key 8	None			None	AUTO	
DSS Key 9	None			None	AUTO	
DSS Key 10	None			None	AUTO	

Apply

Figure 32 - BLF List NumberDisplay

9.3 Sound recordings

The phone supports recording operation during a call.

9.3.1 Network recording

When you use the web server to record, you have to turn on the recording in the phone webpage [Application] >> [Recording Management], select the type as network, fill in the recording server address and port as well as select the voice code. The page is below:

Record Setting

Enable Record: ☒

Record Type: Network

Voice Codec: G729

Server Address: 0.0.0.0 Server Port: 10000

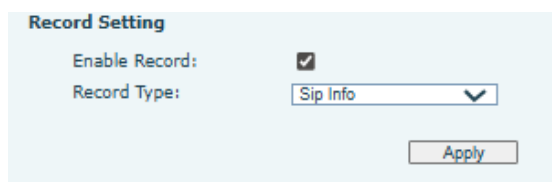
Apply

Figure 33 - Web Page Web Recording

Note: To be used with recording software

9.3.2 SIP INFO recording

The phone is registered to a server that supports SIP INFO recording. After registering an account, in the webpage [Application] >> [Recording Management] of the recording module check the box to open the recording, and select the recording type as SIP INFO.



Record Setting

Enable Record: ☒

Record Type: Sip Info

Apply

Figure 34 - Web SIP info recording

9.4 Act on behalf of sb. In a responsible position

The Agent (i.e., proxy function) of the phone can be realized: when more than one person uses a device for Agent's service at different times, he/she can quickly register his/her SIP account on the same server. The Agent function of the phone is divided into two types: Normal and Hotel Guest. Hotel Guest mode requires server support.

Normal mode:

Configure agent function: Go to [Menu] >> [Function] >> [Agent] to enter agent page. You need to configure the SIP server before configuring the account.

Table 7 - Proxy Models

Parameters	Description
Normal mode	
Numbers	Set the proxy account number.
Subscribers	Set the proxy account number to verify the username.
Cryptographic	Set the proxy account number to verify the password.
Bus route	Select the SIP line.
Tourist mode	
Numbers	Set the proxy account number.
Subscribers	Set the proxy account number to verify the username.
Cryptographic	Set the proxy account number to verify the password.
Bus route	Select the SIP line.
State of affairs	The user can select the status of the number. The available statuses are: logged in, logged out, invalid, valid, and newsletter.

Use the proxy function:

- 1) The phone has been configured with a SIP server, fill in the correct number and user name and password, click login, the phone can be registered to the SIP server;
- 2) After registering, click Logout and the phone can delete the user name and password to log out of

the SIP account.

- 3) Click Cancel Registration, the phone retains the user name and password and logs out of the SIP account.

9.5 Intercom

The phone can automatically answer intercom calls when intercom is enabled.

Table 8 - Intercom Mode

Parameters	Description
Intercom mode	When the intercom system is enabled, the device will accept the SIP header Call-Info command of the incoming call request to answer the call automatically.
Intercom mode mute	Enable mute feature during intercom mode calls
Intercom mode tone	With intercom mode incoming calls will hear an alert tone
Second intercom pickup	Automatically answer calls in intercom mode during a call and refuse to answer a new intercom mode if the current call is in intercom mode

9.6 Multicast

Using the multicast function, you can simply and conveniently send voice broadcasts to all members of a multicast group. By setting the multicast key on the phone, you can send multicast RTP streams to a pre-configured multicast address. By configuring the listen multicast address on the phone, listen and play RTP streams for that multicast address.

Table 9 - Web Multicast Parameters

Parameters	Description
Normal Call Priority	Defines the priority in the current call. The optional meanings are as follows: 1-10: defines the priority of normal calls, with 1 being the highest and 10 being the lowest
Enable page priority	Page Priority determines how the microphone handles newly received multicast RTP streams when it is currently engaged in a multicast session. When the page priority switch is enabled, the microphone automatically ignores low-priority multicast RTP streams, receives high-priority multicast RTP streams, and places the current multicast session in a hold state; if it is not enabled, the microphone automatically ignores all received multicast RTP streams.
Name	Name of the server listening to the multicast
Group:End number	Server address to listen to multicast: port number

Multicast Calling:

- On the webpage [Shortcut Keys] >> [Softkey Settings] >> [Soft DSS Key Settings] select the type as multicast, set the multicast address, and select the encoding
- Once the settings are complete, click Submit
- On the webpage of the phone that accepts multicast [Phone Settings] >> [Multicast], set the name, host and port that accepts multicast
- Press the set multicast key
- The receiver receives the multicast call and automatically plays the multicast

9.7 SCA

Using an SCA account requires server-side support. For details, please refer to the document **Broadsoft's SCA Server and Terminal Configuration User's Guide**.

1) Configure the SCA account on the phone:

- Phones are registered to the BroadSoft server and can register the number created above on multiple endpoints.

Register Settings >>

Line Status: Inactive

Username: 8102

Display name: 8102

Realm:

Activate: ☐

Authentication User: 8102

Authentication Password: *****

Server Name:

SIP Server 1:

Server Address: 192.168.191.213

Server Port: 5081

Transport Protocol: UDP

Registration Expiration: 800 second(s)

Proxy Server Address:

Proxy Server Port: 5080

Proxy User:

Proxy Password:

SIP Server 2:

Server Address:

Server Port: 5080

Transport Protocol: UDP

Registration Expiration: 800 second(s)

Backup Proxy Server Address:

Backup Proxy Server Port: 5080

Figure 35 - Web Registration BroadSoft

- After the phone is registered to BroadSoft server, you have to set the server type: Login to the phone web page, go to [Line] >> [SIP] >> [Advanced Settings], specify the server type as BroadSoft, as shown in the figure:

User Agent:

SIP Version: RFC3261

Local Port: 5080

Enable user=phone: ☐

Auto TCP: ☐

Enable Rport: ☒

Specific Server Type: COMMO

Anonymous Call Standard: None

Ring Type: Default

Use Tel Call: ☐

Enable PRACK: ☐

Figure 36 - Web Page Specifies BroadSoft Server

- The phone uses SCA function, you need to turn on SCA function: Login to the phone web page web

page, go to [Line] >> [SIP] >> [Advanced Settings], turn on SCA (if SCA is not turned on, the registered line is Private Line), as shown in the figure:

Figure 37 - Web page with SCA turned on

After configuring the account and registering successfully, in order to easily check the status of the calls in the group, you can go to the Shortcut Keys page to configure the shortcut keys as Shared Call Appearance-enabled Line, each Line key can represent a Call Appearance, you can refer to [6.3 Appendix III Line Key Lights](#) to know the status of the calls.

To facilitate private hold, you can go to the Shortcuts page to configure the private hold key. Note that the public hold key is Softkey-hold in the call.

Key	Type	Name	Value	Subtype	Line	PickUp Number
DSS Key 1	Line			None	8102@SIP1	
DSS Key 2	Line			None	266@SIP2	

Figure 38 - Web Page Setup Private Hold Function Key

Note that each phone registered to the BroadSoft server should be configured as above, and then you can use the SCA function.

2) Shared Call Appearance (SCA) Scenario Example

To make it easier for users to understand, a few scenario examples are presented here.

In the scenario, the manager and the secretary have registered the same SCA account and configured it as per the above steps.

Scenario 1: When there is an incoming call to this account, both the manager's and secretary's phones receive the call and ring. If the manager is busy, he can simply reject the call and stop ringing, while the secretary's phone will still ring until she rejects/answers the call or the call times out.

Scenario 2: When there is an incoming call to this account, if the secretary picks up the call and needs the manager to answer it, he can press the public hold key to hold the call and notify the manager, who can answer the call by pressing the Line key corresponding to the same SCA.

Scenario 3: The current manager has an important call with a customer and needs to leave for a while, if he does not want someone else to retrieve the call, he can press the private hold function key.

Scenario 4: The current manager establishes a one-way call with a customer that requires the secretary

to participate in the recording, and the secretary can press the corresponding SCA line key to barge-in the call.

9.8 Text

9.8.1 Text message

If the service of this line supports the function of SMS, when the other party sends an SMS to this number, the user will receive a notification of the SMS and an icon with a new SMS will be displayed on the standby screen interface.

Send a text message:

- Select [Message] under [Menu]
- Go to [SMS Service] under [Messaging] to create a new message, select the line and the number to send it to.
- Once you're done editing, just click send

View short messages:

- Right-click on the navigation key to select the standby icon [Messages]
- Once selected, press the navigation key [OK] key to enter the SMS Inbox interface
- Select the unviewed message and press [OK] key to read the unread message

Reply to text message:

- Right-click on the navigation key to select the standby icon [Messages]
- Once selected, press the navigation key [OK] key to enter the SMS Inbox interface
- Select the message you want to reply, choose Softkey's [Reply], edit it, and then click Send.

9.8.2 MWI

If the server of the line supports the function of voice message, the caller can keep the voice message to the server when the user does not answer. The user will be notified of the server's voicemail messages and an icon with a new voicemail message will be displayed on the standby screen interface.

 Voice mail icon

To listen to voice messages, the user must first configure a voice mailbox number. With the voicemail number configured, users can retrieve voicemail messages from the default line.

When the phone is in the default standby state, the

- The sub-screen is pre-programmed with a shortcut key for voice messages - the [MWI] button.
- Press [MWI] to open the Voice Message Configuration interface, and select the line you want to configure by pressing the Up/Down navigation buttons.
- Press the [Edit] button to edit the number of the voice message, when finished, press the [Confirm] button to save the configuration.
- SIP1 (1/1) line "1" before the parentheses represents unread voice messages and "1" represents the

total number of voice messages.

9.9 Point of special interest

SIP hotspots are a simple and useful feature. It is easy to configure, can realize the function of group vibration, and can expand the number of SIP accounts.

Use one phone A as a SIP hotspot and the other phones (B and C) as SIP hotspot clients. When someone calls to talkphone A, talkphones A, B, and C will ring and answer with any one of them, and all other talkphones will stop ringing and cannot answer at the same time. When Phones B or C call out, they all call out with the SIP number registered to Phone A.

In order to set up a SIP hotspot, it is necessary to register at least one SIP account

The screenshot shows a web interface for SIP registration. At the top, there's a 'Line' dropdown menu set to '8102@SI'. Below it, the 'Register Settings >>' section is active. It includes fields for 'Line Status' (Inactive), 'Username' (8102), 'Display name' (8102), 'Realm' (empty), 'Authentication User' (8102), 'Authentication Password' (masked with dots), 'Server Name' (empty), 'SIP Server 1' (Server Address: 192.168.191.213, Server Port: 5061, Transport Protocol: UDP, Registration Expiration: 600 seconds), 'SIP Server 2' (Server Address: empty, Server Port: 5060, Transport Protocol: UDP, Registration Expiration: 600 seconds), 'Proxy Server Address' (empty), 'Proxy Server Port' (5060), 'Proxy User' (empty), 'Proxy Password' (empty), 'Backup Proxy Server Address' (empty), and 'Backup Proxy Server Port' (5060). There are also links for 'Basic Settings >>', 'Codecs Settings >>', 'Advanced Settings >>', and 'SIP Global Settings >>'. An 'Apply' button is at the bottom.

Figure 39 - Web Page Registration SIP

Table 10 - SIP Hotspot Parameters

Parameters	Description
Turn on the hotspot	Set the "Enable Hotspot" option in the SIP Hotspot Configuration item to Enabled.
Model	Selected as "Hotspot" to indicate that the phone exists as a SIP hotspot Select "Client" to use this phone as a client.
Type of monitoring	Choose either broadcast or multicast for the monitoring type. If you want to limit broadcast packets in the network, you can select multicast. Server-side and client-side monitoring types must be the same, for example, when the client-side machine is selected as multicast, as a SIP hotspot server-side machine must also be set to

	multicast
Monitoring address	The multicast communication address used by the client and server when the monitoring type is multicast. If you are using broadcast, you do not need to configure this address, the system will use the broadcast address of the IP of the wan port of the phone to communicate by default
Local port	Fill in the customized hotspot communication port. Server and client ports need to be consistent
Surname	Fill in the name of the SIP hotspot, this configuration is used to distinguish between different hotspots under the network, to avoid connection conflicts
Wire setting	Set whether to enable the SIP hotspot function on the corresponding SIP line association.

Client Settings:

As a SIP hotspot client, there is no need to set up a SIP account, which will be automatically obtained and automatically configured when the phone is enabled. You only need to change the mode to "Client", and the other options are the same as the hotspot.

When the phone is used as a hotspot server, the default extension number is 0. When the phone is used as a client, the extension number increases from 1 (you can check the extension number through the webpage [SIP Hotspot] page).

Call the internal extension number:

- Hotspot servers and clients can previously dial each other through extension numbers
- If extension 1 calls extension 0

10 Phone Settings

10.1 Set up

10.1.1 Multilingualism

Users can set the language of the phone through the phone interface and the web interface.

- Phone interface: After restoring factory settings and rebooting, users need to set the language; to set the language in standby, go to [Menu] >> [Settings] >> [Language Settings], as shown in the figure.
- Web Interface: Login to the phone webpage and set the language in the drop-down box at the top right corner of the page, as shown in the figure:

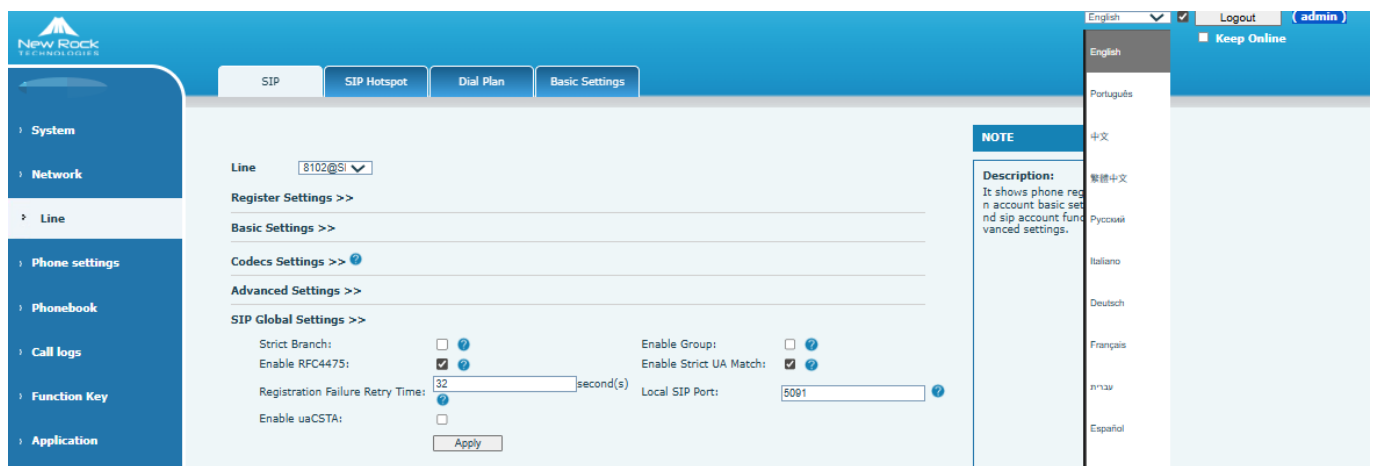


Figure 40 - Web page language settings

- The right checkbox of the web interface language setting box is "Synchronize language to phone"; if it is checked, the language of the phone will be synchronized with the web language, if it is unchecked, it will not be synchronized.

10.1.2 Timing

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

- Phone interface: the phone is in the default standby statePress the function menu button [Menu] >> [Setup] button. >> [Time & Date] button, use the up/down navigation button to edit/time Parameters, press [Confirm] to save when finished, as shown:
- Web interface: Login to the phone webpage, enter [Phone Settings] >> [Time/Date], as shown in the picture:

Network Time Server Settings

Time Synchronized via SNTP ☒

Time Synchronized via DHCP ☐

Time Synchronized via DHCPv6 ☐

Primary Time Server

Secondary Time Server

Time zone

Resync Period second(s)

Time/Date Format

12-hour clock ☐

Time/Date Format

Daylight Saving Time Settings

Location

DST Set Type

Manual Time Settings

Figure 41 - Web Page Setting Time & Date

Table 11 - Time Parameter Settings

Parameters	Description
Paradigm	Choose to synchronize the server time or set it manually
Network time server	SNTP Service Address
Time zones	Select Time Zone
Specification	Select the time format from the following formats: ■ 1 JAN, MON ■ JAN 1, MON ■ MON, 1 JAN ■ MON, JAN 1 ■ DD-MM-YY ■ DD-MM-YYYYYY ■ MM-DD-YY ■ MM-DD-YYYYYY ■ YY-MM-DD ■ YYYYYY-MM-DD
Delimiter	Select the separator between the year, month and day
12-hour system	Displays 12-hour clock
Daylight saving time	Daylight Saving Time setting, off/auto/manual (web setting)

10.1.3 Screen

Users can set the phone screen Parameters through the phone interface and web interface.

- Phone interface: the phone is in the default standby state Press the function menu button [Menu] >> [Settings] >> [Screen Settings] to edit the screen Parameters, click [Confirm] to save after editing, as shown in the figure:
- Web Interface: Go to [Phone Settings] >> [Advanced], edit the screen Parameters and click Submit to save.

10.1.3.1 Brightness and Backlighting

- Set the brightness level from 1 to 16 in the usage state, and[<] or [>] toggles the brightness level.
- Set the backlight brightness level from 0 to 16, and[<] or [>] toggles the brightness level.
- Sets the backlight timeout period.
- Customized backlight idle wait time, can be customized to set the time, the range is (1-54000S).
- Web Interface: Go to[Phone Settings]>>[Advanced], edit the screen Parameters and click Submit to save.

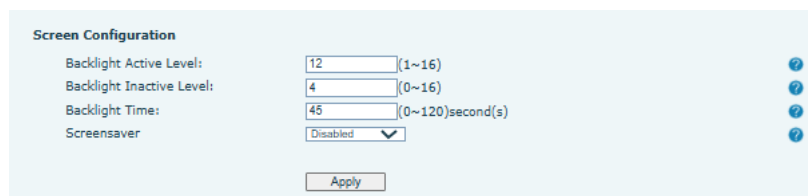


Figure 42 - Web Screen Settings

10.1.3.2 Screensavers

- Press [Screen Settings] to find the [Screen Saver Protection] button, press the [Left] / [Right] button to turn on / off the screen saver, set the timeout period, the default is 2h, and press the [Confirm] button to save.
- Save and go back to standby, and after 2h enter the screen saver as follows:



Figure 43 - Phone Screensaver

10.1.4 Ringtones

When the phone is in the default standby state, and the phone is in default standby.

- Press the function menu button [Menu] to find the [Settings] button.
- Press the [Settings] button to find the [Ringtone Settings] button.
- Press the [Ringtone Settings] button to find the [Headset] and [Handsfree] buttons, use the left and right navigation buttons to edit the ringtone volume, and then press [Confirm] to save it when finished.
- Press [Ringtone Type] button, use the left and right navigation buttons to edit the ringtone type, and

press [Confirm] to save it when finished.

10.1.5 Loudness

When the phone is in the default standby state, and the phone is in default standby.

- Press the function menu button [Menu] to find the [Settings] button.
- Press the [Settings] button to find the [Voice Volume] button.
- Press the [Voice Volume] button to find the [Handset Volume] and [Handsfree Volume] and [Headset Volume] buttons.
- Press the [Handset Volume] or [Handsfree Volume] option, and use the left and right navigation buttons to edit the audio volume.
- Press [Confirm] to save when finished.

10.1.6 Welcome speech

The phone is in the default standby state

- Press the function menu button [Menu] to find the [Settings] button.
- Press the [Settings] button to find the [Welcome Message] button
- Press [Confirm] to enter the setting interface
- Press [Confirm] to save when finished

Note: The welcome message can be displayed in the upper left corner of the standby only if the default line selection feature is disabled

10.1.7 Reboot

With the phone in the default standby stateThe

- Press [Menu] to find the [Settings] button.
- Press [Settings] and find [Reboot System] Press the button.
- Press [Confirm] and a message "Restart Now" will prompt the user.
- Press [Confirm] to restart the handset when finished or press [Cancel] to exit.

The phone is in standby.

- The [OK] key can be configured as a restart key, pressing [OK] will prompt the user with the message "Restart Now".
- Press [Confirm] to restart the handset or press [Cancel] to exit.

10.2 Telephone book

10.2.1 Local contact

Users can save a contact's information in the phonebook and can dial that contact's phone number directly from the phonebook. Users can open the phonebook by pressing the function menu button [Contacts] or the preset button [Phonebook] on the handset under the default main interface.

By default the phonebook is empty, users can manually add or add contacts from the call log (or cloud phonebook) to the phonebook.

Notes! The phone's user account can store up to 1,000 contacts' information.

The contact's existing records will be sorted in alphabetical order. Users can navigate by pressing the

up/down navigation keys. The current record indicator tells the user the current location of the specifically located contact. The user can view the contact's information by pressing the [OK] key.

10.2.1.1 Add/edit/delete contacts

To add a contact, users need to press the [Add] button first to open the interface of adding a contact, and then add the following contact information.

- name (of a person or thing)
- office number
- mobile telephone number
- (sth. or sb) else
- bus route
- ringtones
- cluster
- avatar

Users can edit the contact by pressing the [Edit] button.

To delete a contact, the user should first move the indicator to the location where they want to delete it, Press [Configuration] >> [Delete] button and press [Confirm] to confirm.

10.2.1.2 Add/Edit/Delete Groups

By default the list of groups is empty. A user can create his/her own group, edit the group name, add or remove contacts from the group, delete the group.

- To add a group, press the [Add Group] button.
- To delete a group, press the [Configuration] >> [Delete] button and confirm by pressing [Confirm].
- To edit the group, press the [Edit] button.

The total number of records for the subgroup is indicated in parentheses.

10.2.1.3 Browse/add/remove contacts in groups

Open the group list, users can browse the contacts in the group by pressing the [Confirm] button.

When users browse the contacts in the group, they can add contacts to the group by Press [Add] button to enter into the add contact interface Then press [Confirm] to save the contact, and the contact will be synchronized to the local phonebook. It can also be deleted by [Configuration] >> [Delete Delete contacts in a group].

10.2.2 Blocked Call List

The device supports blocking incoming call list, if you add a number to the blocking incoming call list, the incoming call from this number will be rejected directly to the other end, and the local phone will show that there is a missed call. (preventing numbers in the caller list from being able to make normal outbound calls)

- There are several ways to add a phone to the blocked call list, you can add it directly within [Menu] >> [Contacts] >> [Blocked Call List].

- Select any number within the phone book (both local and network) to configure the addition.
- Select any number within the call log to configure the addition.
- There are several ways to add web pages to the block call list, you can add them in [Phonebook] >> [Call List] >> [Restricted Caller Numbers].
- Select any number within the phone book (both local and network) to configure the addition.
- Select any number within the call log to configure the addition.

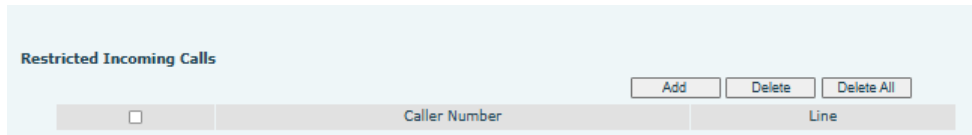


Figure 44 - Web Blocking Call List List

10.2.3 Remote phone book

10.2.3.1 Configuring Cloud Phonebook

Cloud Phonebook allows users to download the phonebook from a cloud server to the phone. This is very convenient for office users in terms of using the phonebook, which can be downloaded from the cloud phonebook server with one click, and is very convenient in creating and maintaining contact lists.

Notes! The cloud phonebook only temporarily downloads its contacts to the phone each time it is opened to ensure that the content is the latest version. But the download may take a few seconds depending on the quality of the network connection when the phone is in use. Therefore, in order to save time waiting for the download, it is highly recommended that users save their important contact information locally from the cloud phonebook.

To open the list of cloud phonebooks, press [Menu] >> [Contacts] >> [Cloud Phonebooks].

Note: The first time configuration needs to be configured in [Phonebook] >> [Cloud Phonebook] on the webpage, and after setting it on the webpage, you can add or edit it on the phone.

10.2.3.2 Download Cloud Phonebook

In the Cloud Phonebook screen interface, users can open Cloud Phonebook by pressing the [OK] button. The phone starts loading the cloud phonebook. If the download fails, a warning message will be prompted.

After downloading the cloud phonebook, users can browse the contact list to dial numbers, the same way as using the local phonebook.

10.3 Call record

The phone can store up to 600 call records. Users can press [Call Records] to open the call records, and can check the records of all accessed, outgoing and missed calls.

In the Call History screen interface, users can navigate through the call history by pressing the Up/Down

navigation keys.

Each call record displays "Call Type", "Caller Number/Name" and "Call Time". Users can further check the details of the call log by pressing the [OK] button and dialing by pressing the [Dial Out] button, the Or by pressing [Configuration] >>Add to Contacts] to add the number in the call log to the phonebook.

Users can press[] button to delete the call history or by pressingDelete] button to delete the call log, or in theConfigurationmenu and select [Delete All] to delete all the call records.

Users can also filter call records for specific call types to narrow down the search for records by selecting a call record type using the left and right navigation keys.

-  - Unanswered Calls
-  - Inbound call log
-  - Outgoing call log
-  - Forward Call Records

10.4 Earphones

10.4.1 Wired headset

The device supports wired headset with RJ9 interface, which realizes headset playback of incoming call alert tone, headset call, and so on.

On the web page [Phone Settings] >> [Feature Settings] you can set up the headset answer function and headset ringing for incoming calls.

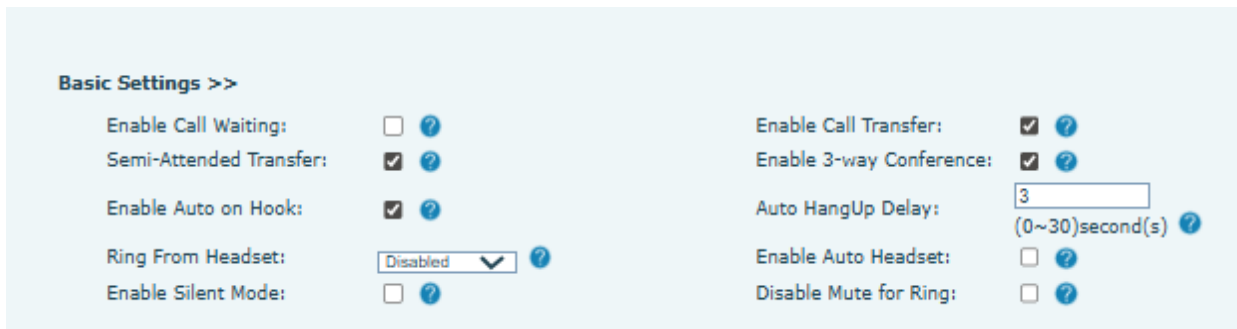


Figure 45 - Headset Feature Settings on Web Page

10.4.2 EHS Headphones

The phone goes to [Menu] >> [Function] >> [Advanced] and selects [EHS Headset] to turn on the EHS headset (the default is to turn off the EHS headset).

10.5 Advanced setting

10.5.1 Wire Setting

So far, all the basic configurations have been completed, save the configuration and the phone can be used normally.

If more settings are needed, the user can change them on the web page or in the Advanced section of each line.

10.5.2 Network setup

10.5.2.1 Network setup

■ IP Mode

The three modes of the phone IP Mode can be selected from IPv4, IPv6, IPv4&IPv6.

Press[<] or [>] to switch the mode, and finally press the Softkey [Confirm] key to save the effect.

■ IPv4

The network type has three modes available DHCP, PPPoE, and Static IP.

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

- Use Dynamic Domain Name Service: Enabled by default. Enabled for domain name resolution.
- Use Dynamic Time: Off by default, controls the time when the DHCP server issues network IPs.

When the network is set to PPPoE, the phone is issued a network IP address by the PPPoE server.

- User: Fill in the user name of the PPPoE server.
- Password: Fill in the password of the PPPoE server.

When the network is set to static IP, the phone has its IP address set by you actively.

- IP Address: Fill in the IP address you want to set.
- Mask: The subnet mask is filled in.
- Gateway: used to realize network interconnection, can be filled according to their own needs.
- Primary domain name: IP address of the primary DNS server. The default is 8.8.8.8 provided by Google for free.
- Subdomain: IP address of the alternate DNS server.

■ IPv6

There are three modes of network type optional DHCP, static IP.

- DHCP network settings are the same as for IPv4.
- The static IP network setup passes through IPv4, just need to fill in the prefix in IPv6 Prefix.
- IPv6 Prefix: IPv6 prefix bits, the prefix represents the network bits, similar to the IPv4 subnet mask.

10.5.2.2 Quality of service and virtual local area networks

■ LLDP

The Link Layer Discovery Protocol (LLDP) is a vendor-independent Layer 2 protocol that allows network devices to advertise their device identity and performance on the local subnet.

LLDP is first used to discover and find information about neighboring devices and save it. The learning function is then used to learn and apply the discovered device information to the phone.

■ CDP

It is a private Layer 2 networking protocol introduced by Cisco that runs on top of most Cisco devices. By running the CDP protocol, Cisco devices are able to share information about operating system software versions, as well as IP addresses, hardware platforms, and other relevant information among the devices directly connected to them.

Table 12 - Quality of Service and Virtual LANs

Parameters	Description
LLDP Settings	
Reporting	Enable LLDP
Intervals	LLDP transmit detection cycle
Do	Learning discovered device information to the phone
Quality of Service qos	
Paradigm	Voice quality assurance (off by default)
Wan port virtual wan	
Wan port virtual wan	WAN Port Settings
LAN port virtual LAN	
LAN port virtual LAN	LAN port settings
Cdp setup	
Cdp setup	CDP on setting, on interval setting

10.5.2.3 VPN

A virtual private network (VPN) is a technology that allows a phone to create a mechanism by which it can connect to a server and become part of the server's network. The network transmission of the phone can be connected through the VPN server routing function.

For some users, especially corporate users, it may be necessary to establish a VPN connection before activating line registration. The device supports two VPN modes, Layer 2 (L2TP) and OpenVPN protocols.

Users must enable (or disable) the configuration of the VPN by logging into the web page.

■ L2TP

Notes. The phone supports only basic unencrypted authentication and data transfer. If you need data encryption, please use OpenVPN function instead. Please check it out:

Users must log in to the phone's web page, open the web page [Network] >> [VPN] to establish an L2TP connection. In VPN mode, check the "Enable VPN" option, select "L2TP", then fill in the L2TP server address, L2TP authentication user name and authentication password. Click "Apply" and the phone will try to connect to the L2TP server.

When a VPN connection is established, the IP address of the VPN is displayed in the VPN status. There may be a delay in establishing a connection. Users need to refresh the page to update the status in time.

The VPN is successfully configured and the phone will automatically try to connect to the VPN every time before the user disables it. Sometimes, if the VPN connection is not established in time, users can try to reboot the device and check whether the VPN is successfully established after reboot.

■ OpenVPN

To establish an OpenVPN connection, the user gets the authentication and profile name from the OpenVPN service provider as follows.

OpenVPN Configuration file.	client.ovpn
CA Root Certification.	ca.crt
Client Certification.	client.crt
Client Key.	client.key

The user then uploads these files to the phone's web page [Network] >> [VPN], selecting OpenVPN Files. Users need to select the "Enable VPN" option, select "OpenVPN" in the VPN mode, and finally check the "Apply" box to enable the OpenVPN function.

Same as the L2TP connection method, the connection is established every time the system reboots before the user manually shuts it down.

10.5.2.4 Web server type

Configure the type of web server HTTP/HTTPS, after reboot, you can use HTTP/HTTPS to log into the phone web page

10.5.3 Setting up the secret key

With the phone in the default standby stateThe

- Click[Menu] to find [Advanced] and click Softkey [Confirm] or [OK] to enter.
- The phone prompts for a password (the default password is 123), find [Setup Secret Key] to enter will see the following interface.

The menu password is used to set the password for the phone to access the advanced interface,.

- Current Password]is the password you set if not set then the default password 123.
- [New Password]is the new menu password you want to reset.
- The password takes effect immediately after setting and is not displayed in plain text after it is entered.

The keypad lock password is used to unlock the keypad after it has been locked.

The keypad lock password setting for the phone interface is limited to on or off, the

- Select[Keyboard Lock Password] and click Softkey [Confirm] or [OK] key to enter, a pop-up box will prompt you to enter the password, this password is the menu password you have set up (if you have not set up the menu password, it is the default password 123).
- If the password is correct, you will be taken to the keypad lock status screen. The keypad lock status is off by default, and the timeout period will take effect after selecting On.
- If no timeout is set (default is 0), you need to press and hold[#]in the standby interface, the phone will lock the keypad, and there will be a lock icon on the top of the phone, at this time, pressing any key prompts for a password.

10.5.4 Safeguard

Web Interface: Login to the phone webpage and enter [System] >> [Auto Deploy] page.

Basic Settings

CPE Serial Number: 00A859F81A5F

Authentication Name:

Authentication Password:

Configuration File Encryption Key:

General Configuration File Encryption Key:

Download Fail Check Times: 5

Update Contact Interval: 720 (0, >= 5) minute(s)

Save Auto Provision Information: ☒

Download CommonConfig enabled: ☒

Enable Server Digest: ☐

DHCP Option >>

DHCPv6 Option >>

SIP Plug and Play (PnP) >>

Static Provisioning Server >>

Autoprovision Now >>

TR069 >>

Figure 46 - Web Auto Deployment Setup

Phone Interface: Go to [Menu] >> [Advanced] >> [Maintenance] >> [Auto Deploy].

Vendor terminals support SIP Plug and Play, DHCP Selected Parameters, Static Deployment Server, and TR069 to get automatic deployment of application Parameters. If all four methods are configured at the same time, the terminal will get the auto deployment application Parameters according to the following process/priority when it starts:

PNP>DHCP>TR069> Static Provisioning

Transport Protocol Support: FTP, TFTP, HTTP, HTTPS.

This article only briefly describes Auto Provision, please refer to the document **Auto Provision**.

Table 13 - Automatic upgrade

Parameters	Description
Basic settings	
Terminal equipment serial number	Display device serial number
Verify User Name	Configure the user name of the auto-deployment server; the TFTP protocol does not need to be configured.
Verify Password	Configure the password corresponding to the user of the auto-deployment server.
Configuration file encryption key	If the configuration file to be upgraded is an encrypted configuration file, then the encryption password for that configuration needs to be entered here.

Generic profile encryption key	If the generic profile to be upgraded is an encrypted profile, then the encryption password for that profile needs to be entered here.
Number of failed downloads	The number of times the download failed and was re-downloaded.
Phonebook download cycle.	At preset intervals, the phone will automatically download the phonebook and update it.
Saving automatic deployment information	Holds HTTP/HTTPS/ftp protocol username and password authentication information and input ID information. As long as the URL configured on the server does not change, the information saved to the phone does not change.
Enable download of generic profiles	Whether or not to download a generic configuration file when performing an automatic upgrade.
Calculate digest from server before opening download	If the terminal is matching the configuration file content by Digest checksum, then whenever the configuration on the server is modified, or the configuration on the terminal does not match the configuration on the server, then the terminal will also update and download.
DHCP option settings	
Selected parameter values	Configure DHCP option, DHCP option supports DHCP custom option, DHCP option 66, and DHCP option 43 to get auto deployment application Parameters. When obtaining the auto-deployment application Parameters through DHCP, the user can choose any one of them, and the default of the terminal is DHCP option 66.
Customized selection of parameter values	Valid values for the customization options are from 128 to 254. The custom option type must be the same as that defined by the DHCP server.
Enable dhcp option 120	Set the SIP server address by DHCP option 120.
SIP Plug and Play	
Enable SIP Plug and Play	Configure whether to enable PnP, if the terminal is enabled to enable PnP mode, it will send SIP SUBSCRIBE messages periodically in multicast form after the terminal starts. Any SIP server compatible with supporting this particular message will respond with a SIP NOTIFY message back containing the path to the autoconfiguration/deployment server, through which the terminal is able to obtain the configuration file to be downloaded.
Server address	Configure the PnP server.
Server port	Configure the PnP port.
Transportation protocol	Configure the PnP transport protocol.
Update cycle	Configure the PnP timeout time in hours.

Static deployment of servers	
server address	Configure the address of the server you are setting up. The address of the server can be in the form of an IP or a domain name.
Configuration file name	Configure the name of the configuration file to be upgraded; When this item is configured as empty, the device will use the generic configuration file name and its own MAC address as the file name to get the file from the server; The filename can also be \$mac, \$input or other customized filenames, and the file extension can be CFG/TXT/XML.
Protocol type	Select the server type, there are four types: FTP, TFTP, HTTP and HTTPS
update cycle	Configure the time in hours between upgrades.
update mode	Type of automatic upgrade 1.not be upgraded 2.Upgrade after reboot 3.Interval upgrades, i.e. how many times between upgrades
TR069	
Enable TR069	Check the box to turn on TR069.
ACS server type	Select whether the ACS server type is Common or CTC.
ACS Server Address	Configure the ACS server address.
ACS Login Account	Configure the ACS user name, up to a maximum of 59 characters.
ACS Login Password	Configure the ACS password, up to a maximum of 59 characters.
Turn on the TR069 tone	There is a beep when making a TR069 connection after checking the box.
TLS version	TLS version (TLS 1.0, TLS 1.1, TLS 1.2)
Notification delivery cycle	Configure the cycle interval in seconds, valid value range:1~9999.

10.5.5 Online Upgrade

- Web interface: Login to the phone webpage and go to [System] >> [Upgrade] page.

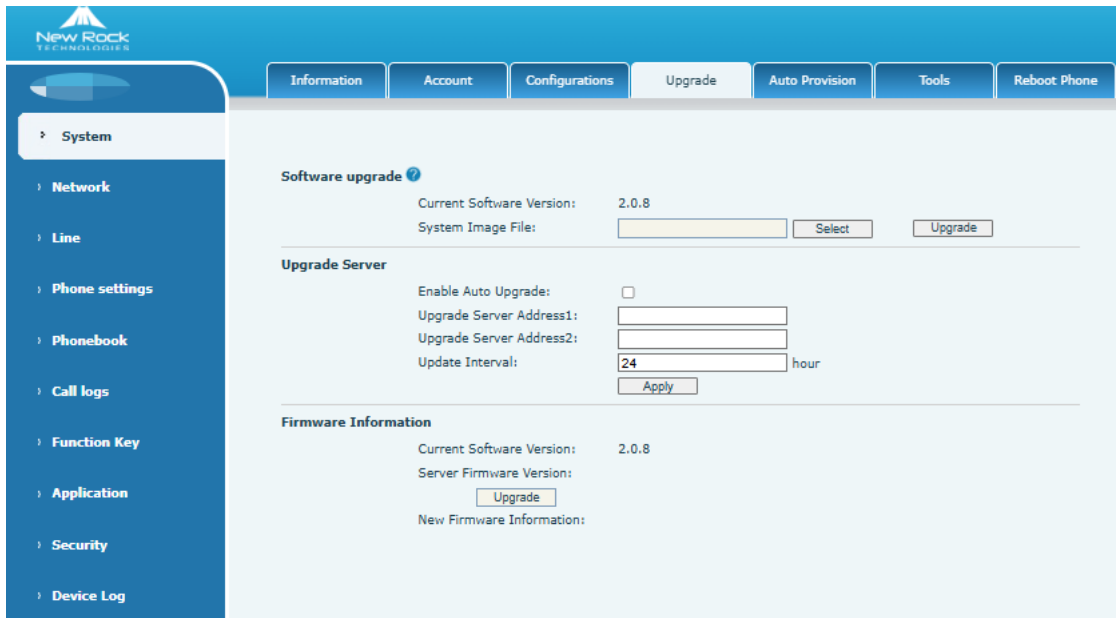


Figure 47 - Web Page Online Upgrade Page

- Phone interface: Go to [Menu] >> [Advanced] >> [Software Upgrade].

Table 14 - Online Upgrade

Parameters	Description
Upgrade Server	
Allow automatic upgrades	Allow Automatic Upgrade: Check Allow Automatic Upgrade, the phone can detect the TXT version information and available versions in the HTTP server.
Upgrade server address 1	Fill in the address of the available primary upgrade server (HTTP server).
Upgrade server address 2	Fill in the address of the available backup upgrade server (HTTP server) and request the backup server when the primary server is not available.
update cycle	Update cycle: Web page to enable automatic detection of upgrades, configure the interval time, if the server has a new version, the phone to the interval time will be prompted to upgrade.
Software version information	
Current software version	Displays the current phone software version number.
Server software version	Displays the server software version number.
[Upgrade] button	When the corresponding TXT file and version are available on the server side, the [Upgrade] button changes from grayed out to available, and you can choose whether to upgrade or not by clicking [Upgrade].
New version	When there is a corresponding TXT file on the server side when and version, the

description information	and version information in the TXT is displayed under the new version description information.
-------------------------	--

- The file that the phone requests from the server is a TXT file called vendor_model_hw10.TXT. hw is followed by the hardware version number, if you do not distinguish between hardware are written as hw10. All spaces in filenames have been changed to underscores.
- The URL of the phone request is HTTP://server address/ , and both the new version and the requested file need to be placed in the download directory of the HTTP server
- TXT file format must be UTF-8
- The vendor_model_hw10.TXT file format is referenced below:
Version=0.0.15 #Software version
Firmware=xxx/xxx.z #URL, supports relative paths, but also absolute paths, differentiated by the presence or absence of protocol headers
BuildTime=2022.08.29 15:33
Info=TXT|XML
Xxxxx
Xxxxx
Xxxxx
Xxxxx
- After the update cycle interval is reached, if the server has available files and versions, the phone prompts for a software upgrade, click View to view the version information and upgrade.

10.5.6 Restore Factory Settings

1) With the phone in the default standby state The

- Press the [Function] button to find the [Advanced] button, and then press the [OK] button.
- Press [Advanced] and enter the password (default password is 123) to enter the interface.
- Press the [Restore Factory Settings] button, and select the file you want to erase.
- Press [Confirm] to clear when finished. When you select Clear Profile and Clear All, the phone will restart automatically after clearing.

2) Press and hold the OK button for 6S in standby to reset.

11 Web Configuration

11.1 Authentication of web pages

Users can log in to the phone's web page to manage information about the user's phone and to operate the phone. Users must provide the correct username and password to log in.

11.2 Systems >> Information

This page allows the user to access the system information of the phone, including:

- model number
- hardware version
- software version
- running time

This page also contains information about the status of the network, including:

- networking method
- MAC address
- IP address
- subnet mask
- routing gateway

This page also contains information about the SIP account, including:

- SIP users
- SIP Account Status (Registered / Unsubmitted / Trying / Timeout)

11.3 System >> User Configuration

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

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

11.4 System >> System Configuration

Users with administrator privileges can view, export, or import the phone configuration on this page, as well as restore the phone to factory settings.

■ Clear Configuration

Select the module in the configuration file to clear.

SIP: account-related configuration

AUTOPROVISION: Automatically upgrading related configurations

TR069:TR069 Related Configurations

MMI:MMI module, including authenticated user information, web access protocols, etc.

Shortcut Keys: Soft Key Configuration

- **Clearing the database**

Select the local data tables to be cleared, the default is All.

- **Restore Factory Settings**

The phone data is completely emptied, including configuration and database tables.

11.5 System >> Upgrade

Upgrade the phone software version, customize ringtones, background images, etc. You can also delete the upgrade file. Ringtones support .wav format.

11.6 Systems >> Automated Deployment

Automated deployment helps IT managers or service providers to deploy and manage phones in bulk. For more information about Auto Provisioning, read the Auto Provision Description.

11.7 System >> Accessibility

This page provides tools for users to solve problems. Please refer to [13 Troubleshooting](#) for details.

11.8 System >> Reboot Phone

This page restarts the phone.

12 Network >> Basic

This page allows users to configure the network connection type and Parameters.

12.1 Network >> Service Ports

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

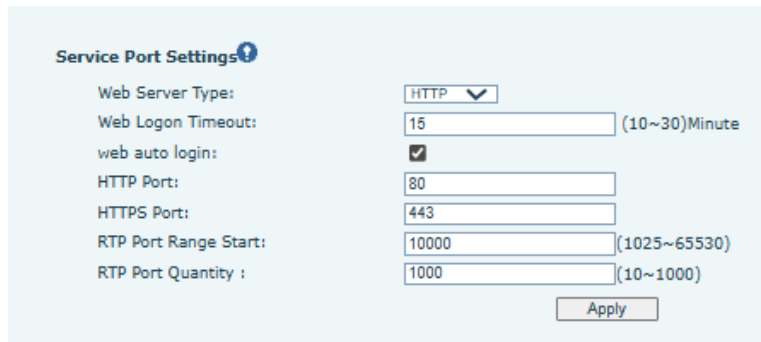


Figure 48 - Service Port Settings Screen

Table 15 - Server Ports

Parameters	Description
Web server type	Set it and reboot to take effect. Optional web login for HTTP/HTTPS
Web Login Timeout	Default 15 minutes, timeout time will automatically exit the login page, need to log in again
Automatic Web Login	After the timeout period, you don't need to enter your username and password, you will be automatically logged into the webpage.
HTTP port	Default is 80, if you want the system to be safe, you can set the port other than 80 e.g. 8080, web login: HTTP://ip:8080
HTTPS port	Default is 443, same as HTTP port usage
RTP port start range	The range of values is 1025-65535, and the value of the rtp port starts from the initial value set, and the value of the voice and video ports is increased by 2 for each call made.
Number of RTP ports	Number of calls

12.2 Network >> VPN

This page allows users to configure the VPN connection. Please refer to [10.6.2.3 VPN](#) for more details.

12.3 Network >> Advanced

Network advanced settings are usually configured by IT administrators to improve the quality of phone service. For configuration, please inquire [10.6 Advanced](#).

12.4 Line >> SIP

Configure the service configuration of the line on this page.

Table 16 - Line Configuration on the Web

Parameters	Description
Registration Settings	
Line state	Displays the status of the current line on this screen. To get the latest online status, users must manually refresh the page.
Start using	Service on this line is enabled
User ID	Enter the user name of the service account.
User authentication	Enter the authentication name for the service account.
Display name	Enter the name that is displayed when the call request is sent.
Verify password	Enter the authentication password for the service account.
Domain (taxonomy)	Enter the SIP domain provided by the service provider.
Server name	Enter the server name
Sip server 1	
Server address	Enter the SIP server address
Server port	Input SIP server port
Transportation protocol	Setting SIP Transport Lines to Use TCP or UDP or TLS
Registration validity period	Setting the SIP expiration date
Sip server 2	
Server address	Enter the SIP server address
Server port	Input SIP server port
Transportation protocol	Setting SIP Transport Lines to Use TCP or UDP or TLS
Registration validity period	Setting the SIP expiration date
SIP proxy server address	Enter the IP of the SIP proxy server.
Proxy server port	Enter the port of the SIP proxy server, the default is 5060.
Proxy user	Input proxy userName

Proxy Password	Enter the proxy password
Backup proxy server address	Enter the backup proxy server address.
Backup proxy server port	Enter the backup proxy server port, which defaults to 5060.
Basic settings	
Enable auto-attendant	Enable Auto Answer, incoming calls will be answered automatically after the delay time is exceeded.
Auto Answer Waiting Time	Set the waiting time for the system to answer automatically.
unconditional switching	Enable unconditional forwarding and all incoming calls will be forwarded to the set forwarding number.
Unconditional number forwarding	Set the number for unconditional forwarding.
Busy line transfer	Enable busy line forwarding so that all incoming calls will be forwarded to the set forwarding number when the phone is occupied.
Busy line transfer number	Setting the number for busy line forwarding
No Answer Transfer	Enable no-answer forwarding, incoming calls are not answered beyond the configured delay time and incoming calls will be forwarded to the set forwarding number.
No Answer Transfer Number (NAT)	Set the no-answer call forwarding number.
No Answer Transfer Waiting Time	Set the no-answer call forwarding wait time.
Call Transfer Timeout Setting	Set the timeout for the call forwarding process.
Conference room model	Mode of setting up a conference call room, local = a conference call is set up on the phone itself, supporting up to two calls, server = a call conference is set up by dialing into a conference on the server.
Server room number	Set the conference room number for the server conference type.
Enable voice mail	Enable Voice Message Waiting Notification, if enabled, the phone will receive a notification from the server if a voice message is left on the server.
Extract voice mail number	Setting up the number for receiving voice messages
Voice mail check cycle	Set the time interval for the voice message checking cycle.

Activation of the hotline	Enable Hotline Configuration, turn on the voice channel (remove the handle, turn on the speakerphone or headset channel) the phone will dial the set number.
Hotline delay time	Setting the delay time for outgoing calls to hotline numbers
hotline number	Set the hotline dialing number.
Allow unregistered callouts	is enough to allow unregistered callouts
Enable missed call log	Configure whether to turn on missed call logging, if you turn on missed call logging, there will be a missed call log in the history record
DTMF type	Sets the DTMF type of the line.
DTMF SIP INFO mode	Set SIP INFO mode to send '*' and '#' or '10' and '11'.
Whether the URI carries port information	Whether the URI carries port information
Turn on Do Not Disturb	With Do Not Disturb enabled, any incoming calls on the line will be automatically rejected.
Using a VPN	Set up the line to use a VPN network.
Using STUN	Ensure that the NAT penetration setup line uses STUN.
Enable Failback	Whether to switch to the primary server when it is available
Failback Interval	Interval for periodically probing the availability of the master Proxy using Register messages
Signal Failback	Whether to allow invite/register requests to also perform failbacks in case of multiple proxies.
Signal Retry Counts	Number of attempts for SIP Request to consider proxy unavailable in case of multiple proxies
Codec Setting	Set codec priority and availability by adding or removing them from the list.
Advanced Settings	
Open Service Code	If this setting is enabled, the features in this section will not be controlled by the phone itself, but by the server. To control the phone, the phone will send a feature code to the server via the number specified in the Dial Code field.
Do Not Disturb	Dial the feature code to the server.
Deactivate Do Not Disturb	Dial the feature code to the server.
Enable unconditional switching	Dial the feature code to the server.
Deactivate unconditional	Dial the feature code to the server.

switching	
Enable Busy Line Transit	Dial the feature code to the server.
Disable busy line transfer	Dial the feature code to the server.
Enable no-answer forwarding	Dial the feature code to the server.
Deactivate no-answer transfer	Dial the feature code to the server.
Enable do-not-call anonymity	Dial the feature code to the server.
Deactivate do-not-call anonymous calls	Dial the feature code to the server.
Call waiting open code	Dial the feature code to the server.
Call Waiting Shutdown Code	Dial the feature code to the server.
Send an anonymous call opening code	Dial the feature code to the server.
Send an anonymous call closure code	Dial the feature code to the server.
SIP encryption	Enable SIP encryption, SIP transmission will be encrypted.
RTP encryption	Turn on RTP encryption and the RTP transfer will be encrypted.
Enable call timing	Enable the call timing feature, if the call conference time is not received before the timeout, the call will end.
Call timeout setting	Set the call timeout period.
Open BLF List	Configure whether to enable the BLF List function
BLF List number	BLF List is able to listen to the status of multiple accounts. Supports multiple BLF lists
Responding to calls using a single decoder	If the setting is enabled, the phone will use a single codec to respond to incoming requests.
BLF server	Normal BLF application is that the phone sends subscription packets to the registered server, if your server does not support subscription packets, please enter the BLF server, which separates the registered server from the subscription server
Maintain on-line type	Set the line to use UDP or SIP OPTION packets to ensure that NAT is

	turned on.
Maintaining the online interval	Sets the keep-alive packet delivery interval.
Maintaining certification	Maintains previously validated validation Parameters.
Rejection of anonymous calls	Reject any incoming calls without caller ID.
user agent	Setting up the user agent to conform to the software version by default
Specify the server type	Working with specific server types
SIP version	Setting the SIP Version
Standards for anonymous calls	Setting criteria for anonymous calls
local port	Setting the local port
Ringtone type	Setting the ringer type for a line
Enable user=phone	There is a user=phone field value in the invite SIP message.
Calling by phone	Configure whether to enable the use of phone calls
Automatic use of TCP transmission	Configure automatic transmission using TCP protocol when the message body exceeds 1500 bytes; guarantee transmission availability.
Enable Rport	Set the line to add the Rport SIP header.
Enable PRACK	Set the line to support PRACK SIP messages.
DNS mode	Select DNS mode, A, SRV, NAPTR
Long contact field	Configure the Contact field to carry more Parameters; work with SEM servers
Configuring Compatible Specialty Servers	Compatible with special servers (return messages with the source address of the other party, no longer using the address in the via field)
URI conversion	Configuration is all about turning on URI conversion
Use quotes for display names	Whether to add quoted display names
Open GRUU	Configure to turn on GRUU
walkie-talkie time	Synchronization with server time
Enable Inactive Hold	After enabling the call hold packet capture you can see (in the INVITE packet) that the SDP is inactive
Caller ID header fields	Setting the Caller ID Header Field
Setting the Call Waiting Response	Setting Up the Handset for Use 182 Interjecting Responses

Code (182)	
Enable Feature Sync	Enable/disable Feature Sync
Open SCA	Enable/disable SCA (shared calls)
CallPark number	Setting the CallPark number
Use the server's timeout	Setting the timeout for using the server
TLS version	Select TLS Version
uaCSTA number	Setting the uaCSTA number
Enable Click To Talk	Use with special servers, click to call out directly when enabled
Enable port updates	Whether to enable port updates
Flash mode	Flash mode selection, normal and SIP info
Flash Info Content-Type	Setting the SIP info content type
Flash Info Content-Body	Setting the SIP info body
snatch number	Setting the robocall number when robocall is turned on
JoinCall Number	Setting the JoinCall number
intercom number	Setting the intercom number
initiate a logout	Whether to enable the logout function
Enable MAC Header	Whether to enable SIP packets and user agent with or without MAC during registration.
Enable Register MAC Header	Whether to enable registration is user agent with or without MAC
BLF Session Exact Match	Whether to enable BLF session exact matching
global setting	
Strictly matches Branch fields	Setting Strict Match Branch Fields
Open Groups	Setting Enable Groups
Open RFC4475	Setting to enable RFC4475
Enable Strict UA Matching	Enable Strict UA Matching
Registration Failure Retry Time: 1 hour	Setting the registration failure retry time
SIP port of the phone	Modify the phone's SIP port
Enable uaCSTA	Setting Enable uaCSTA Function

12.5 Line >> SIP Hot Spot

See [9.9 Hot Spots](#) for details

12.6 Lines >> Rules for receiving numbers

Basic Settings

- ☒ Press # to invoke dialing
- ☐ Dial Fixed Length to Send
- ☒ Send after second(s) (3~30)
- ☐ Press # to Do Blind Transfer
- ☐ Blind Transfer on Onhook
- ☐ Attended Transfer on Onhook
- ☐ Attended Transfer on Conference Onhook
- ☐ Enable E.164

Figure 49 - Receipt Rule Registration Settings

Table 17 - 7 Dialing Methods for Phones

Parameters	Description
"#" dialer	The user dials the other party's number followed by the # sign to dial out.;
Dialing in fixed length	Automatic dialing of user-entered numbers after a fixed length has been reached
timeout dialing	The system automatically dials the number after the timeout
blind carbon copy	After entering the number to be transferred, users can press the "#" key to transfer the current call to a third party.
blind switch	After entering the number, the user hangs up the handle or turns off the speakerphone to be able to transfer the current call to a third party
Hang up and attend the turn	Hang up the joystick or press the hands-free button to hang up to realize the Attendance Transfer function, which is able to transfer the current call to a third party
Conference attendance hang-ups	During a three-way call, hang up the handle and the remaining two parties stay on the line
Enable E.164	Please refer to E.164 Standard Specification

Add a dialing rule:

Dial Plan Add

Digit Map:

Apply to Call: Outgoing Call ?

Match to Send: No ?

Line: SIP DIALPEER ?

Destination: Port:

Alias(Optional): No Alias ?

Phone Number: Length:

Suffix:

Figure 50 - Dialing Rule Customization Settings

Table 18 - Dialing Rules Configuration Table

Parameters	Description
Dialing Rules	In order to add the outgoing number, the outgoing number setting can be divided into two types: one is exact match, after configured as exact match, if this number and the user dialed the called number if exactly the same, the phone will use this number mapping IP address or configuration for calling; one is prefix match (equivalent to the area code prefix function of PSTN), if this number and the user dialed the called number before the first N digits (the length of prefix number) is the same, then the phone will use this number mapping IP address or configuration for calling.) is the same, then the phone will use the IP address or configuration of this number mapping to make the call. Configuring prefix matching requires adding a T to the prefix number to differentiate it from the exact match number; up to 30 digits are supported;
Note: Use two different special characters. x -- is a match for any of the digits [] -- is a range of specified digits; can be a range, separated by commas, or a list of digits;	
destination (location)	Configure the destination address, and if configured as a point-to-point call, write the peer IP address directly.
ports	Configure the signaling port of the other protocol, this is an optional configuration item, the default is 5060.
prefix	Configure the alias, which is optional: the replacement number to be used when the other party's number has a prefix.
Note: There are four types of aliases, which must be set in conjunction with the replacement length: 1) Add: xxx with xxx in front of the number. This helps the user to save the dialing length; 2) All: xxx, numbers are all replaced by xxx; speed dialing can be realized 3) Delete: Delete the first n digits of the number, n is set by the replacement length; 4) Substitution: xxx, the first n digits of the number are replaced by xxx, n is set by the substitution length.	
Suffix	Configure the suffix, which is optional: i.e. add this suffix after the dialed number
Lengths	Configure the replacement/deletion length to replace/delete the user-entered number by

	this length; this is optional; for example, if you set the deletion to 3, the first three digits of the number will be deleted.
--	---

Examples of alias applications

This feature allows users to create rules to simplify dialing. There are several different options available for dialing rules. The following example will show how it works.

Example 1. Replace all

If a user dials the other party directly through IP point-to-point mode, and the other party's IP is 172.168.2.208, configure a rule similar to the one shown below, and you can call the user with IP 172.168.2.208 by dialing 123.

User-defined Dial Plan Table ⓘ

Index	Digit Map	Call	Match to Send	Line	Alias Type: Number(length)	Suffix
1	"123"	Out	No	SIP DIALPEER(172.16.2.208:5060)		

Figure 51 - Dialing Rules Table

Example 2. Partial replacement

For example, to make a PSTN call to Beijing, we can set up the following dialing rule, all the phone numbers starting with 1 will be initiated by this rule, such as want to dial 010-62213123 this number, we only need to dial 162213123 that is.

User-defined Dial Plan Table ⓘ

Index	Digit Map	Call	Match to Send	Line	Alias Type: Number(length)	Suffix
-------	-----------	------	---------------	------	----------------------------	--------

Table of rules

Example 3. increase

Two examples are cited. Option 1, assuming the user dials any 11-digit number beginning with 131, will be sent by automatically adding 0 to the number when sending; Option 2, assuming the user dials a number beginning with 135 to 139, will be sent by adding 0 to the number when sending after the 11-digit number has been collected.

Use two different special characters:

x is matching any bit;

[] is a range of specified digits; it can be a range, separated by commas, or a list of digits.

Through these rules, users can easily set up their own dialing rules, as well as create dialing rules suitable for use with their own servers, greatly providing the convenience and practical performance of the phone.

12.7 Line >> Basic Settings

Setting up the Registration Global Configuration

Table 19 - Setting Line Global Configurations on Web Pages

Parameters	Description
STUN Setting	
server address	Set the STUN server address.
server port	Set the STUN server port to 3478 by default.
Binding cycle	Set the STUN binding period to ensure that NAT penetration is on.
SIP Waiting Time	Sets the timeout for STUN binding before transmitting SIP messages. ,
Documents for certificates	
TLS Authentication File	Upload or delete TLS authentication files used to encrypt SIP transmissions.

12.8 Phone Settings >> Feature Settings

Configuring phone features

Common features

Table 20 - Common phone feature settings on web pages

Parameters	Description
Basic settings	
Turn on call waiting	Enabled by default. On allows the user to answer a second call while on hold.
Enable call forwarding	Call forwarding is allowed.
Semi-attendance is permitted	Semi-attendance transfers are allowed.
Allows three-way calling	Three-way calling is allowed.
Automatic return to standby at the end of a call	Configure whether or not to enable automatic hangup to automatically return to standby at the end of a call
Automatic hang time	Configure the auto handdown time so that the phone automatically returns to standby after the auto handdown time is exceeded if it is in hands-free mode, or automatically puts dial tone after the auto handdown time is exceeded if it is in joystick mode.
Headset ringer	Configure ringing through the headset.
Automatic answer selection	Turn on this feature, if the user has a headset plugged into the handset, press the Answer key or Line key to answer the call using the headset, and turn on the Auto Answer feature as well .
Allow silent mode	The phone is in mute state after turning on, mute not ring when there is an incoming call, you can use the volume key and mute key to release this state.

disable mute	The mute button on the handset does not take effect when turned on.
Open Default Line	If enabled, the phone is assigned a default line instead of SIP1.
Enable automatic line switching	If turned on, the phone automatically selects an available line as the default line.
Default call to outside line	Select the default line used for callouts.
Do not call out	Disable outgoing calls, when enabled, the off-hook dialing will send busy tone and prompt to hang up.
Hide DTMF	Configure the form of the hidden DTMF.
Save call records	Select whether or not to save the call log.
Enabling restricted caller lists	Whether or not to enable restricted caller lists
Enabling the Allowed Caller List	Whether or not to enable the Allowed Caller List
Enabling restricted dial-out lists	Whether to enable restricted dial-out lists
Enabling country codes	Enable or disable country code
country code	Fill in the country code
area code	Fill in the area code
Enable number privacy	Whether to enable number privacy
Matching direction	Match direction, with both right-to-left and left-to-right rules
starting position	Start hidden location with number privacy turned on
Hidden Bits	Hiding the digits of a number when number privacy is turned on
Allow IP calls	If enabled, the phone allows direct IP calls, otherwise it does not.
P2P Ip Prefix	Sets the prefix for point-to-point IP calls.
Caller Name Display Priority	Selects the priority for the display of the caller's name.
emergency call number	Setting the emergency call number
Contact search range	Select the search scope.
Idap Search	Required if LDAP is selected for the search scope.
emergency call number	Set the emergency call number, and the phone can be called out even when it is locked.
Restrict Active URI Source IP	Sets the phone to accept valid URI commands from a specific IP address. Notes! This feature is typically used for device management, the
XML Push Server Address	Configure the XML server so that when the phone receives a request, it will decide whether or not to display the corresponding content sent by the server to the specified phone.

Enable pre-dialing	Disabling this feature will automatically open the audio channel when the user enters a number. Enable function, user enters the number, no audio channel is turned on.
Turn on Multi Line	If enabled, up to 10 simultaneous calls can exist on the phone, and if disabled, up to 2 calls exist on the phone.
Customized line formats	Customized line format with SIPn/SIPn: xxx/xxx@SIPn
Local Contacts as a Type of Allowed Caller List	Contacts allow caller type with NONE/BOTH/DND White List/FWD White List
Disable XML Push in Calls	Whether to enable in-call disable XML push
SIP notify	When enabled, the message will be displayed when the phone receives a notify containing the relevant content.
Tone setting	
Call hold tone	When turned on, a tone plays while the call is on hold.
Turn on call waiting tone	If this feature is turned off, no beep will be heard while the call is on hold.
Playback of DTMF tones during a dialed call	There is a DTMF tone when the user presses the number buttons while dialing. The phone is turned on by default.
Playing an in-call DTMF tone	When the user presses the handset numeric buttons during a call there will be a DTMF tone. The phone is turned on by default.
Call restriction setting	
Call restriction options	Choose whether you want the line or the phone to be on or off
Turn on call barring at regular intervals	Whether or not to turn on the call restriction on a regular basis, after turning on the call restriction, the DND will be turned on automatically from the start time to the closing time.
Call restriction start time	Setting the start time
Call restriction off time	Setting the closing time
Intercom setup	
Intercom mode	When the intercom system is enabled, the device will accept the SIP header Alert-Info command of the incoming call request to answer the call automatically.
Intercom mode mute	Enable mute feature during intercom mode calls
Intercom mode tone	With intercom mode incoming calls will hear an alert tone
Second intercom pickup	Automatically answer calls in intercom mode during a call and refuse to answer a new intercom mode if the current call is in intercom mode
Response code setting	
Do Not Disturb response code	Set the Do Not Disturb SIP response code.

Busy response code	Sets the SIP response code when the phone is busy.
Reject response code	Sets the SIP response code when a call is rejected.
Password dialing settings	
Encrypted dialing	Enable password dialing when entering the number of starting password prefixes, the following number of N* prefixes will be hidden after the password, N represents the value you entered in the Password Length field. For example: if you set a password with a prefix of 3 and enter a password length of 2, then you enter the number 34567, which will be displayed on the phone 3* * 67
Encryption number length	Configure the length of the hidden number
Password dialing prefix	Configure the prefix for password dialing numbers
Led power lights	
General lamps	Power light status under standby, off when off, on is red always on. Close by default
Short message/voice message	The status of the power lamp when there is unread SMS/voice message, including off/on/slow flash/fast flash, default slow flash.
Unanswered call	There is the status of the power light when there is a missed call, including off/on/slow flash/fast flash, the default is slow flash.
Call/dial	Power lamp status during call/dialing state, off is off, on is red and always on, default off
Ringtones	Power lamp status when there is an incoming call ringing, including off/on/slow flash/fast flash, default fast flash
Unmute	Power light status during mute, including off/on/slow flash/fast flash, default off
Maintenance/retention	Power lamp status on hold/retention, including off/on/slow flash/fast flash, default off
Pop-up notification window	
Show missed call popup	There is a missed call pop-up alert when it is turned on, and no pop-up alert when it is turned off, and it is turned on by default.
Show voice message pop-up window	There is a pop-up alert for unanswered voicemail when it is turned on, and no pop-up alert when it is turned off; it is turned on by default.
Show external device popup	With Bluetooth adapter on, WIFI adapter connected with a pop-up prompt, no pop-up prompt when off, default on.
Show short message popup	There is a pop-up alert for unread text messages when it is turned on, no pop-up alert when it is turned off, and it is turned on by default.
Show other pop-ups	There is a pop-up prompt when the handle is not hung back, registration failure, IP acquisition failure, Tr069 connection failure and other exceptions when it is turned on, and when it is turned off, there is no prompt for exceptions, and it is

	turned on by default.
--	-----------------------

12.9 Phone Settings >> Media Settings

Changing voice settings

Table 21 - Voice Settings on the Web

Parameters	Description
Code Setting	Select the voice code to enable or disable: g.711a/u, g.722, g.729, g726, ilbc, opus
Audio Settings	
Joystick Volume Settings	Set the volume of the microphone, the volume is 1~9
Preset ringtone types	Configure the preset ringtones. If no special ringtone is set for the caller's number, the default ringtone will be used.
Handsfree volume setting	Set the hands-free volume, volume 1~9
Headset ringer volume setting	Set the volume of the headset ringer, volume 1~9
Headphone Volume Settings	Set the headphone volume, volume 1~9
Handsfree ringer volume setting	Set the hands-free ringer volume from 1 to 9
G.723.1 Bit rates	Selectable 5.3kb/s or 6.3kb/s
DTMF Load Type	Enter DTMF Load Type, value must be 96~127.
AMR Load Type	Sets the AMR load type.Range 96~127
Headphone Mic Gain	Setting the radio volume gain of the headset in order to adapt to different models of headset calls
Opus payload type	Set Opus load type, range 96~127
OPUS Sample Rate	Setting the Opus sampling rate with OPUS-NB (8KHz) and OPUS-WB (16KHz)
ILBC Load Type	Set ILBC load type, range 96~127
ILBC Load Length	Selecting the ILBC Load Length
Turn on voice mail dial tone	When a new voice mail message is received, the phone will activate a special dial tone.
Enable Voice Activity Detection	Whether to enable voice activity detection
Fork Spring Reaction Time	Configure the minimum response time, default is 200ms

EHS Headphones	Enable to talk on EHS headset
RTP Control Protocol (RTCP) Settings	
CNAME user	Setting up the CNAME user
CNAME Hosting	Setting up CNAME hosts
RTP Settings	
RTP keep alive	Stay on the line, send a packet once in 30s after you turn it on
Incoming Call Designated Ring Type Setting (alert-info)	
Value of notification message 1 to notification message 10	Sets the value of the ringer type specified by the caller
Ringtone type	Type1-Type9

12.10 Phone Settings >> Multicast

Using the multicast function, you can simply and conveniently send announcements to each member of the multicast. By setting the multicast key on the phone, you can send a multicast RTP stream to a pre-configured multicast address. By configuring the listen multicast address on the phone, listen and play RTP streams sent from this multicast address.

Table 22 - Web Multicast Parameters

Parameters	Description
Prioritization of common calls	Defines the priority in the current call, with 1 being the highest priority and 10 being the lowest.
Enable page priority	No matter which of the two multicasts calls in first, the phone will answer the multicast with the higher priority first.
Name	Name of the server listening to the multicast
Group:End number	Server address to listen to multicast: port number

12.11 Phone Settings >> Features

Action URL

Notes! The operation URL is used by the IPPBX system to submit a phone event. Refer to the Action URL for details.

12.12 Phone Settings >> Time/Date

Users can configure the time settings of the phone on this page.

Table 23 - Time/Date Setting Parameters on the web page

Parameters	Description
Network Time Server Settings	
Time synchronization using SNTP	Enable time synchronization using the SNTP protocol.
Time synchronization using DHCP	Enable time synchronization using the DHCP protocol.
Primary time server address	Setting the master time server address
Secondary time server address	Set the address of the backup time server. When the primary server is unavailable, the phone will try to connect to the backup time server to get time synchronization.
Time zones	Select Time Zone
Time synchronization cycle	Resynchronize the time with the time server.
12-hour system	Set the 12-hour mode time display.
Date format	Select the date/time display format.
Daylight saving time (SST)	
Spot	Select your location
DST type setting	Setting the DST type
Fixed type	Is the Daylight Saving Time rule based on a specific date or is it converted for use relative to the date of the rule. Read-only display in auto mode.
Revised value	Corresponding acceleration/reconciliation times after the opening/closing of daylight saving time
Month start	DST Starting month
Day of the week.	DST Start of week
Workday start	DST Start of working day
Hour start	DST start hour
Minutes to go.	DST Starting minute
Month end	DST End month
Week end	DST End of week
Working day Ended	DST End of working day
Hour end	DST end of hour
Minutes. Over.	DST End Minute
Manual time setting	Manually set the current time

12.13 Phone Settings >> Time Management

Time Plan (time management) settings can be set up to set the time point, you can also set the time period, the time point is to perform a certain action at a certain time, the time period for a certain period of time to perform a certain action

Table 24 - Time Management

Types of time management	1: Timed reboot 2: Timed Upgrade 3: Timed forward rotation 4: Timed configuration changes	Type, point in time/time period of action performed
Repetition cycle	0: not repeated 1: Daily 2: Weekly 3: Monthly	Repeat type
Work in weeks	0-6: Sunday-Saturday, supports multiple separations by ";" 1-31: 1-31	Value is null when repeat type is daily/non-repeat
Day-based	xx:xx-xx:xx	Start time - end time period

When Time Plan Type is selected as Timed Forward, the web page prompts for the Forward Number and Forward Line, as shown.

Forwarding Number: Configure the forwarding number to which the number will be forwarded during the set time period.

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

1. Timed forward turn rule:

- When there is forwarding under line, the forwarding number under line is used; when there is no forwarding number under SIP line, when there is an incoming call within the time period set by Timed Forwarding, the phone is forwarded to the number specified by Timed Forwarding; when it is outside the time period, it is not forwarded. That is, priority Line>Time Plan.
- The timed forward types are all unconditional forward.

Setting	Value
Select Your Tone:	United States
Dial Tone:	350+440/0
Ring Back Tone:	440+480/2000,0/4000
Busy Tone:	480+820/500,0/500
Congestion Tone:	
Call waiting Tone:	440/300,0/10000,440/300,0/10000,0/0
Holding Tone:	
Error Tone:	
Stutter Tone:	
Information Tone:	
Dial Recall Tone:	350+440/100,0/100,350+440/100,0/100,350+440/100,0/100,350+440/0
Message Tone:	
Howler Tone:	
Number Unobtainable Tone:	400/500,0/6000
Warning Tone:	1400/500,0/0
Record Tone:	440/500,0/5000
Auto Answer Tone:	

Apply

Figure 52 - Web Beeps

12.15 Phone Settings >> Advanced

Users can talk to the dialog for advanced here.

- screen configuration
 - Turn on energy saving mode
 - Backlight Timeout
- Setting the menu password

The default password is 123.

- Setting the key code
- welcome speech

When the phone is in standby, the welcome message is displayed in the upper left corner of the screen. Up to 16 characters can be entered. Default is "VoIP Phone".

12.16 Phonebook >> Contacts

This page allows users to add, delete or edit contacts in the phonebook. Users can browse the phonebook by filtering by name, phone number, and group.

To add a new contact, click on Add contact user needs to enter contact information and then press "Confirm" button to add it.

To edit a contact, click the checkbox in front of the contact, the contact information will be copied to the contact edit box, after finishing editing, press the "Modify" button.

To delete one or more contacts, select the checkboxes in front of the contacts you want to delete, click the "Delete" button, and click "Delete All" to empty the phonebook.

Users can also add multiple contacts to a group via the "Add to Group" button. At the bottom of the contact list, select the checkbox and click Add to Group to add the selected contacts to the selected group.

Similarly, users can select multiple users and add them to the blocked call list by clicking on the "Add to blocked call list" button.

12.17 Phonebook >> Cloud Phonebook

cloud phonebook

Users can configure up to 8 cloud phonebooks. Each cloud phonebook needs to be set up with a URL that points to the XML phonebook, which can be transmitted using HTTP, HTTPS, or FTP. If the server requires authentication, the user needs to configure a username and password.

To configure Cloud Phonebook, the following information is required:

- Phone Book Name (required)
- Phonebook URL (required)
- Username (optional)
- Password (optional)

LDAP Configuration

Cloud Phonebook allows users to retrieve a list of contacts from an LDAP server via the LDAP protocol.

In order to use it on the device, the user must configure LDAP server information and Search Base. If the LDAP server requests authentication, the user should also provide a username and password.

To configure the LDAP phonebook, the following information is required:

- Show title (required)
- LDAP server address (required)
- LDAP server port (required)
- Query Base (required)
- Username (optional)
- Password (optional)

Note! When creating an LDAP phonebook and phonebook server, refer to the LDAP technical documentation.

Page Preview

Phone web page support for previewing VoIP phonebook directory and contacts

- After setting up the XML web phonebook or LDAP, the
- Select [Phonebook] >> [Cloud Phonebook] >> [Cloud Phonebook] Select Type
- Click on XML/LDAP in the settings to download contacts for browsing.

Figure 53 - Web Cloud Phonebook Setup

12.18 Phonebook >> Call List

■ Restricted Caller ID.

He does the same thing as a blocked call list. Add the number to the blocked call list and the user will not receive calls from the number until the user removes the number from the list.

Users can add specific numbers to the blocked call list or add a specific prefix to the blocked call list to block calls from all numbers with this prefix.

■ Allowed caller ID.

When DND is turned on, allow incoming numbers to still call in.

■ Restrictions on dialing out numbers:

Adding a number that restricts outgoing calls prevents the number from being called until the number is removed from the table.

12.19 Phonebook >> Web Dialing

Call, answer and hang up operations using web pages.

12.20 Phonebook >> Advanced

Users can export the local phonebook in XML, CSV, VCF format files and save it on the local computer.

Users can also import contacts into the phone book of the phone in XML, CSV, VCF format files.

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

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

12.21 Call record

Users can browse the complete call logs on this page. Call logs can be sorted by time, caller ID, contact name or line used, and call logs can also be filtered by call log type (inbound, outbound, missed call).

The user can also save the numbers in the call log to his/her phonebook or add them to the block call list/allow call list.

Users can also web dial by clicking on the number in the call log.

12.22 Shortcuts >> Shortcuts

One-touch transfer settings: Create new call, Blind transfer, Attend transfer, One-touch 3-way, Play DTMF.

The device provides 2 user-defined shortcuts and the user can configure each shortcut on the web page.

Table 25 - Shortcut Configuration

Parameters	Description
Memory key	BLF . Enables the user to see the status of the subscribed extension number and to snatch the call when the subscribed extension number is ringing. The status of the subscription number is: idle, ringing, in call Note: If a user needs to grab a subscribed extension number, he or she must configure a robocode Presence . As opposed to BLF, Presence can also see if a user is online or not. Speed Dial : Users can dial the set number directly. This feature makes it easy for customers to call frequent numbers Walkie Talkies . This feature allows operators or secretaries to quickly connect to the phone for a wide range of office environments.
Line key	It can be configured as a line key. Users are able to make and receive calls by pressing line keys
Function key	The user can select a function key as a shortcut to trigger an event Example. MWI / DND / Release / Headset / Hold / etc.
Dual-tone, multi-frequency	Allows users to dial or edit numbers more easily
Url	Open a specific URL directly
Multicast	Configure the multicast address and voice encoding. . The user presses this key to initiate a multicast
Action URL	Users can use specific URL dialogs for basic call operations
XML browser	Users can set up shortcuts to specific URLs for downloading, etc.

12.23 Shortcuts >> Softkey Settings

The user sets the mode and display style, and the display page.

Table 26 - Softkey Configuration

Parameters	Description
Softkey mode	
Softkey mode	Deactivated and More, default is Deactivated
Softkey Display Style	
Softkey Display Style	Left display and right display
Show page	
Telephone dialing	Redial/2aB/Delete/Exit/Return/Dial/Dial/Join 3-Party/MWI/Local Contacts/Rush/Call Log/Missed/Clear/Ln/Dial Out/Pause/Next line/Prev line/Headset/Audio/Video/Remote XML/Shortcuts
Sessions	Hold/Split/End/Release/Mic Mute//Shortcut/Headset
Desktop	Call Log/Menu/Local Contacts/Call Restriction/Previous Account/Next Account/Block Caller List/Callback/Call Forwarding/Lockout/Memo/Missed/MWI/Dial-Out/Reboot/Redial/Remote XML/SMS Status/Headset/Network//Shortcuts/In
Forward motion	Redial/2aB/Delete/Exit/Forward/Local Contacts/Call History/Clear/Missed/Dial Out/Headset/Video/Audio/Remote XML//Shortcuts
Close	Redial/End/Headset/Release// Shortcuts
Pre-dial	Dialing/2aB/Delete/Exit/Callback/Local Contacts/Redial/Resuscitation/MWI/Join 3-Party/Call Log/Clear/Missed/Pause/Dial Out/Headset/Video/Audio/Remote XML//Shortcuts/In/Next line/Prev line
Bell	Answer/Forward/Reject/Microphone Mute/Release/Headset/Video/Audio//Shortcuts
Hold a conversation	Hold/Transfer/Conference/End/Microphone Mute/Release/New Call/Local Contacts/Listen/Call Log/Next call/Prev call/Private/Headset/Video/Audio//Shortcuts
Transfer alerts	End/Transfer/Headset/Release// Shortcut Keys
Redirect dialing	Redial/Delete/Exit/2aB/Dialing/Local Contacts/Transfer/Call History/Clear/Missed/Dialed Out/Pause/Headset/Video/Audio/Remote XML//Shortcuts
Attempts	End/Release/Headset// Shortcut Keys
Wait for	Hold/Transfer/Conference/End/Answer/Forward/Microphone Mute/Next call/New call/Prev call/Reject/Release/Headset/Listen/video/audio/Video/Audio///Shortcut Keys

12.24 Shortcut >> Advanced

One-touch transfer: e.g. set up memory key 4370, press the memory key when talking to 4374 to decide whether to start a new call to 4370 or to transfer (blind/attendance transfer) 4374 to 4370.

■ Programmable key settings

Please refer to [Table 27 Softkey Settings](#).

12.25 Applications >> Recording Management

See [9.3 Recordings](#) for details on the use of recordings

12.26 Security >> Web Filtering

Users can set up to allow only machines with a certain network segment IP to access the Configuration Management Phones.

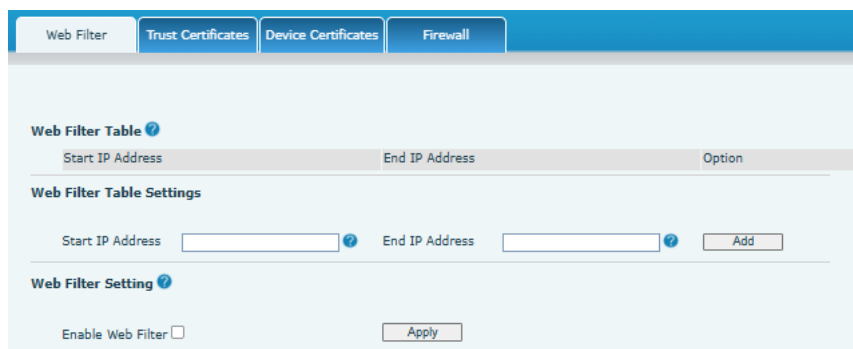


Figure 54 - Web Filter Settings

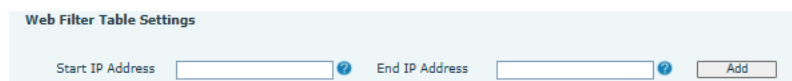


Figure 55 - Web Filter Table

Add and delete IP segments that allow access; configure the starting IP address within the start IP and the ending IP address within the end IP, and then click **[Add]** to submit to take effect. You can set up a large network segment or divide it into several segments to add. When deleting, select the starting IP of the segment to be deleted in the drop-down menu, and then click **[Delete]** to take effect.

Enable Web Filtering: Configure to enable/disable web access filtering; click the **[Submit]** button to take effect.

Note: If the device from which you are accessing the phone is on the same network segment as the phone, do not configure the web filtering segment to be outside of your own segment, or you will not be able to log in to the web page.

12.27 Security >> Certificate of Trust

Set whether to enable license certificate and regular name verification, and select the certificate module.
Uploaded certificates can be uploaded and deleted

Permission Certificate

Permission Certificate ?

Common Name Validation ?

Certificate mode ?

Import Certificates ?

Load Server File

Certificates List ?

Index	File Name	Issued To	Issued By	Expiration	File Size
					Delete

Figure 56 - Certificate Settings

12.28 Security >> Device Certificates

Select the device certificates as Default Certificate and Custom Certificate.
You can upload and delete uploaded certificates.

Device Certificates ?

Device Certificates (existence)

Import Certificates ?

Load Server File

Certification File ?

File Name	Issued To	Issued By	Expiration	File Size
				Delete

Figure 57 - Device Certificate Setup

12.29 Security >> Network Firewall

Figure 58 - Network Firewall Settings

Through this page, you can set whether to enable the input and output firewalls, and you can also set the input and output rules of the firewalls, which can be utilized to prevent some malicious network access or restrict the internal users from accessing some resources in the external network to improve the security.

Firewall Rule Setup is a simple module for firewalls. This feature supports two types of rules: input rules and output rules. Each rule will be assigned a serial number, with a maximum of 10 rules of each type allowed to be set.

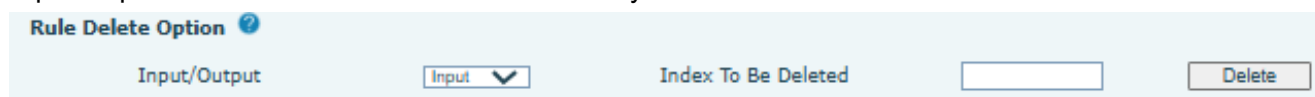
Considering the complexity of firewall settings, the following is an example:

Table 27 - Network Firewalls

Parameters	Description
Enable input rules	Indicates that input rule application is enabled
Enable output rules	Indicates that output rule application is enabled
Input/output	To select whether the currently added rule is an input or output rule
Prohibited/allowed	To select whether the current rule configuration is disabled or allowed;
Protocol type	There are four protocol types for filtering: TCP, UDP, ICMP, and IP.
Filter port range	Filtered port range
Source address	is the source address. The source address can be a host address, a network address, or the full address 0.0.0.0; it can also be something like *.*.*.0 network address, such as 192.168.1.0.
Destination address	is the destination address, which can be a specific IP address or the full address 0.0.0.0; it can also be something like *.*.*.0 network address, such as 192.168.1.0.

Source subnet mask	is the source address mask, when configured as 255.255.255.255 that means it is a specific host, when set to a subnet mask of type 255.255.255.0, it means that the filtering is for a network segment;
Destination subnet mask	is the destination address mask, when configured as 255.255.255.255 that means it is a specific host, when set to 255.255.255.0 type of subnet mask, it means that the filtering is a network segment;

When the settings are set and clicked [Add], a new item will be added to the firewall output rules. Then select and click the button [Submit]. This way, when the device runs: ping 192.168.1.118, it will not be able to send packets to 192.168.1.118 because the output rule is disabled. However, pinging other IPs in the 192.168.1.0 segment can still receive response packets from the destination host normally.



The screenshot shows a light blue interface titled "Rule Delete Option" with a help icon. Below the title, there is a label "Input/Output" followed by a dropdown menu currently showing "Input". To the right of the dropdown is the label "Index To Be Deleted" followed by an empty text input field. Further right is a grey button labeled "Delete".

Figure 59 - Deleting a Firewall Rule

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

12.30 Device Log >> Device Log

You can capture device logs and send them to technicians to locate the problem when you encounter abnormal problems. For details, see [13.4.1 Getting log information](#).

13 Fault Resolution

When the phone is not working properly, users can try the following methods to restore the phone to normal operation or collect relevant information to send a problem report to the vendor's technical support e-mail.

13.1 Getting phone system information

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

Network information, device information (model, hardware and software version), etc.

13.2 Restart A Telephone

Users can reboot the phone by using the function buttons.[Menu] >> [Settings] >> [Reboot System] and press [Yes].Or just unplug the power to restart the phone.

13.3 Restore factory settings on the phone

Restoring factory settings deletes all configurations, preferences, databases, and profiles on the phone, and the phone is restored to its factory default state.

User Restore Factory Reset Press [Menu] >> [Advanced], then enter the password to enter the interface, select [Restore Factory Settings] and press [Enter]. The phone will be restored to its factory default state.

13.4 Screenshot

If there is a problem with the phone, screen shots can help the technician locate the function, as well as clarify the problem phenomenon. In order to get a screen shot, log in to the phone webpage [System] >> [Auxiliary Tools], and under Screen Shot, you can capture the picture of the main screen and the sub-screen (which can be intercepted at the interface where the problem occurs).

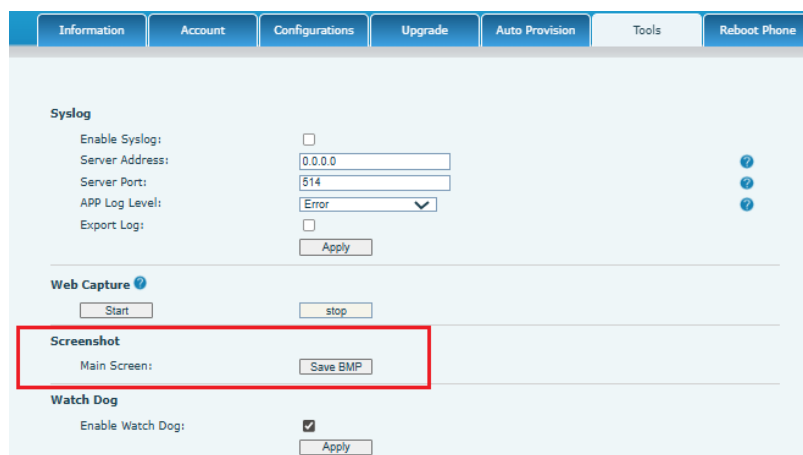
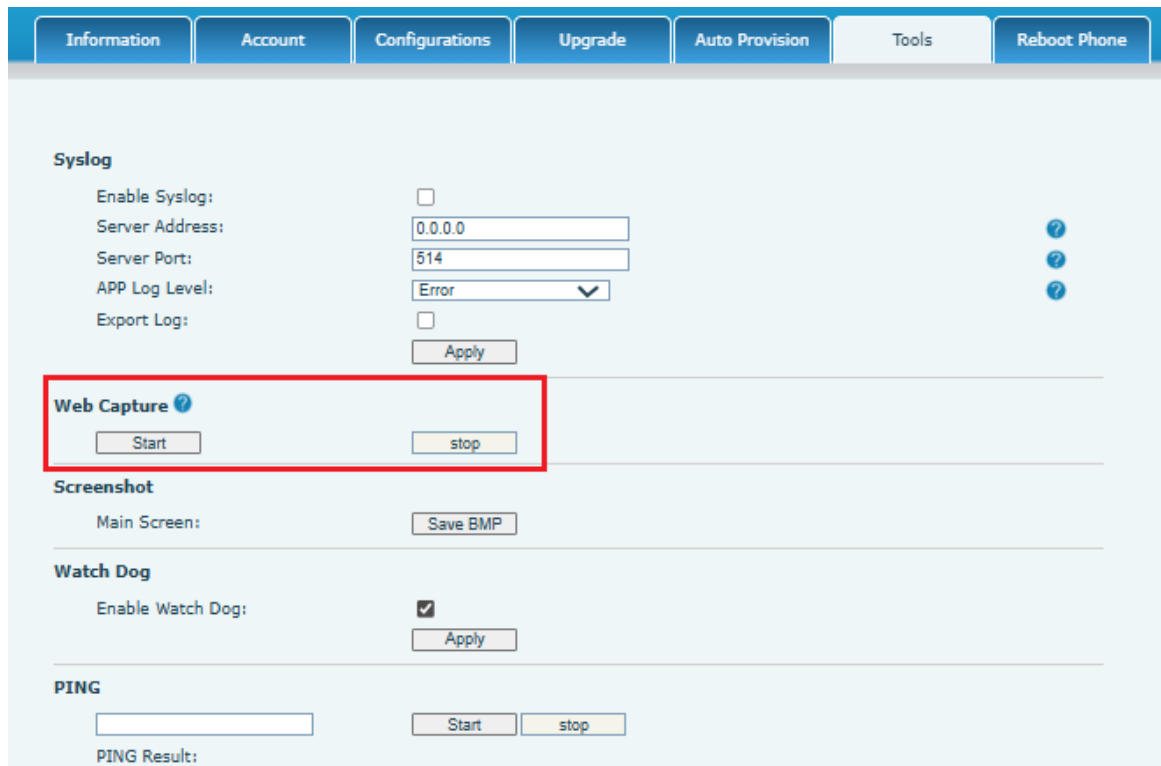


Figure 60 - Screenshot of the web page

13.5 Network Data Grabber

Sometimes network packets for phone problems are helpful. In order to get the packets from the phone, the user needs to log in to the phone's web page, open the web page [System] >> [Auxiliary Tools], and then click the [Start] start option in "Network Packet Capture". A message will pop up asking the user to save the capture to a file. At this point the user can perform related operations, such as activating/deactivating the line or making a call, and then click the [Stop] button in the web page when finished. Network packets for the period of the phone are saved in the file.



The screenshot shows a web interface with a top navigation bar containing tabs: Information, Account, Configurations, Upgrade, Auto Provision, Tools, and Reboot Phone. The 'Tools' tab is selected. Below the navigation bar, there are several sections: 'Syslog' with fields for 'Enable Syslog' (checkbox), 'Server Address' (text input '0.0.0.0'), 'Server Port' (text input '514'), 'APP Log Level' (dropdown menu 'Error'), and 'Export Log' (checkbox), followed by an 'Apply' button; 'Web Capture' which is highlighted with a red rectangular box and contains 'Start' and 'stop' buttons; 'Screenshot' with a 'Main Screen' label and a 'Save BMP' button; 'Watch Dog' with 'Enable Watch Dog' (checkbox) and an 'Apply' button; and 'PING' with a text input field, 'Start' and 'stop' buttons, and a 'PING Result:' label.

Figure 61 - Web Page Capture Tool

Users can analyze packets or send them to the vendor's technical support mailbox.

13.6 Getting log information

Log messages are helpful when unusual problems are encountered. In order to obtain the log information of the phone, users can log in to the phone's webpage, open the webpage [Device Log], click on the "Start" button, follow the steps until the problem appears, and then click on the "Finish" button to "Save" the log to the local area for analysis or send it to the technician to locate the problem. "Save" the log information locally for analysis or send it to a technician to locate the problem.

13.7 Common Failure Cases

Table 28 - Failure Cases

Fault case	prescription
The phone won't start.	1. The phone is powered through the power adapter or the external power supply of the PoE switch. Use a power adapter or a PoE switch that meets standard specifications and check that the phone is connected to a power source. 2. If you see the phone enter "POST mode", it means that the phone's system is damaged. Contact the manufacturer's technical support staff to help you restore the phone system.
Phone cannot be registered to service provider	1. Please check if the phone is connected to the network. The network cable must be connected to the  [Network] port instead of the  [Computer] port. If the network cable is not connected to this network icon  [Network Connection Disconnected] will flash at the top right of the handset screen. 2. Please check if the phone has an IP address. Check the system information, if the IP address is Negotiating..., it means that the phone has not acquired an IP address. Please check if the network configuration is correct. 3. If the network connection is good, check your line configuration again. If all configurations are correct, contact your service provider for support or follow the "13.4 Network Data Capture Packets" for instructions to obtain registered network data packets and send them to the vendor's technical support email address to help analyze the issue.
Phones with no or poor audio	1. Check that the handle is properly connected to the handset () port and not the headset () port. 2. Network bandwidth and latency may not be suitable for audio calls at this time.
Poor or low audio on headphones	1. Two types of headphone cables are currently on the market. Please use the headset supplied by the manufacturer, if you want to use a third party headset, you need to ask the manufacturer for the headset cable order. 2. Network bandwidth and latency may not be suitable for audio calls at this time.

Audio Plunge in Hands-Free Speaker Mode	This is usually due to a loud speaker volume feedback channel microphone. Please turn down the speaker volume a little bit and the phenomenon will disappear.
---	---