



i53W User Manual

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Directory

Directory.....	1
1 Picture.....	5
2 Table.....	8
3 Safety Instruction.....	9
4 Overview.....	10
4.1 Overview.....	10
4.2 Packing Contents.....	10
5 Install Guide.....	11
5.1 Use PoE or external Power Adapter.....	11
5.2 Desktop Installation.....	11
5.2.1 Peripheral connection.....	11
6 Appendix Table.....	13
6.1 Appendix I - Icon.....	13
6.2 Appendix II –Function key state definition.....	16
7 User Getting Started.....	17
7.1 Button description.....	17
7.2 Introduction to the User.....	18
7.2.1 Standby interface.....	18
7.2.2 Dial interface.....	18
7.2.3 Commonly used icons on the interface.....	20
7.3 Use of touch keyboard.....	20
7.4 Phone Status.....	21
7.5 Web Management.....	23
7.6 Network Configurations.....	24
7.7 SIP Configurations.....	25
8 Basic Function.....	27
8.1 Making Phone Calls.....	27
8.2 Answer a call.....	28
8.3 Call interface.....	28
8.4 Unlock.....	29
8.5 End of call.....	30
8.6 Video preview.....	30
8.7 Dial query.....	31
8.8 Auto Answer.....	31

8.9 Mute.....	33
8.9.1 Mute during a call.....	33
8.9.2 Mute when ringing.....	33
8.10 DND.....	34
8.11 Call Forward.....	36
9 Advance Function.....	38
9.1 Intercom.....	38
9.2 MCAST.....	38
9.3 SMS.....	40
9.3.1 SMS.....	40
9.3.2 MWI (Message Waiting Indicator)	41
9.4 SIP Hotspot.....	42
10 Phone Settings.....	45
10.1 Basic Settings.....	45
10.1.1 Language.....	45
10.1.2 Time & Date.....	46
10.1.3 Screen.....	48
10.1.3.1 Brightness and backlight.....	49
10.1.3.2 Screen Saver.....	49
10.1.4 Ring.....	50
10.1.5 Voice Volume.....	50
10.1.6 Greeting words.....	50
10.1.7 Reboot.....	51
10.2 Phonebook.....	51
10.2.1.1 Add / Edit / Delete Contact.....	52
10.2.1.2 Add / Edit / Delete Group.....	53
10.2.1.3 Add / Edit / Delete contact in Group.....	54
10.2.2 Black list.....	55
10.2.3 Cloud Phone Book.....	56
10.2.3.1 Configure Cloud Phone book.....	56
10.2.3.2 Downloading Cloud Phone book.....	57
10.3 Call Log.....	58
10.4 Function Key.....	59
10.5 Wi-Fi.....	61
10.5.1 Wireless network.....	61
10.5.2 AP setting.....	63
10.6 Advanced.....	64
10.6.1 Line Configurations.....	64

10.6.1.1 Network Settings.....	65
10.6.1.2 Network Settings.....	65
10.6.1.3 QoS & VLAN.....	68
10.6.1.4 VPN.....	69
10.6.1.5 Web Server Type.....	71
10.6.2 Set The Secret Key.....	71
10.6.3 Maintenance.....	74
10.6.4 Firmware Upgrade.....	77
10.6.5 Factory Reset.....	80
11 Web Configurations.....	81
11.1 Web Page Authentication.....	81
11.2 System >> Information.....	82
11.3 System >> Account.....	82
11.4 System >> Configurations.....	83
11.5 System >> Upgrade.....	83
11.6 System >> Auto Provision.....	83
11.7 System >> Tools.....	83
11.8 System >> Reboot Phone.....	84
11.9 Network >> Basic.....	84
11.10 Network >> Service Port.....	84
11.11 Network >> VPN.....	85
11.12 Line >> SIP.....	86
11.13 Line >> SIP Hotspot.....	92
11.14 Line >> Dial Plan.....	92
11.15 Line >> Action Plan.....	95
11.16 Line >> Basic Settings.....	95
11.17 Settings >> Features.....	96
11.18 Settings >> Media Settings.....	100
11.19 Settings >> MCAST.....	102
11.20 Settings >> Action.....	102
11.21 Settings >> Time/Date.....	102
11.22 Setting >> Time plan.....	103
11.23 Settings >> Tone.....	104
11.24 Settings >> Advanced.....	105
11.25 Phonebook >> Contact.....	105
11.26 Phonebook >> Cloud phonebook.....	106
11.27 Phonebook >> Call List.....	107
11.28 Phonebook >> Web Dial.....	108

11.29 Phonebook >> Advanced.....	108
11.30 Call Logs.....	108
11.31 Function Key >> Function Key.....	108
11.32 Function Key>> Side Key.....	109
11.33 Function Key >> Softkey.....	110
11.34 Function Key >> Advanced.....	111
11.35 Security >> Web Filter.....	111
11.36 Security >> Trust Certificates.....	112
11.37 Security >> Device Certificates.....	113
11.38 Device Log >> Device Log.....	113
11.39 Security Settings.....	114
12 Trouble Shooting.....	115
12.1 Get Device System Information.....	115
12.2 Reboot Device.....	115
12.3 Reset Device to Factory Default.....	115
12.4 Screenshot.....	115
12.5 Network Packets Capture.....	116
12.6 Get Log Information.....	117
12.7 Common Trouble Cases.....	117

1 Picture

Picture 1 - Peripheral connection diagram.....	12
Picture 2 - interface	12
Picture 3 - Key Description.....	17
Picture 4 - Standby interface.....	18
Picture 5 -Menu interface.....	19
Picture 6 - keyboard.....	20
Picture 7 - Keyboard numbers & characters.....	21
Picture 8 - keyboard.....	22
Picture 9 - phone status.....	23
Picture 10 - Login page.....	23
Picture 11 - Web Line Registration.....	26
Picture 12 - Dial interface.....	27
Picture 13 - Call interface	28
Picture 14 - Voice call interface.....	28
Picture 15 - Taking interface.....	29
Picture 16 - Web page configuration DTMF.....	30
Picture 17 - Call end interface.....	30
Picture 18 - Web page configuration hard button preview.....	30
Picture 19 - Webpage configuration standby interface preview.....	31
Picture 20 - Standby interface display after configuration.....	31
Picture 21 - Line 1 enables auto-answering.....	32
Picture 22 - Web page to start auto-answering.....	32
Picture 23 - Mute the call.....	33
Picture 24 - Ringing mute.....	34
Picture 25 - DND setting interface.....	35
Picture 26 - DND timer.....	35
Picture 27 - DND Settings.....	35
Picture 28 - Line DND.....	36
Picture 29 - Set call forward.....	37
Picture 30 - Set call forward.....	37
Picture 31 - Web Intercom configure.....	38
Picture 32 - Multicast Settings Page.....	39
Picture 33 - SMS icon.....	40
Picture 34 - New Voice Message Notification.....	41
Picture 35 - Voice message interface.....	42
Picture 36 - Register SIP account.....	42

Picture 37 - SIP hotspot server configuration.....	43
Picture 38 - SIP hotspot client configuration.....	44
Picture 39 - Set language.....	45
Picture 40 - web page language setting.....	46
Picture 41 - set time & date.....	47
Picture 42 - webpage set time &date.....	47
Picture 43 - set screen Parameters.....	48
Picture 44 - set screen on webpage	49
Picture 45 - Screensaver.....	50
Picture 46 - local contact.....	51
Picture 47 - add contact.....	52
Picture 48 - edit contact.....	53
Picture 49 - group.....	54
Picture 50 - browse the contact in group.....	54
Picture 51 - add blacklist.....	55
Picture 52 - Black list.....	56
Picture 53 - cloud contacts	57
Picture 54 -Browsing Contacts in Cloud Phone book.....	58
Picture 55 - call log.....	58
Picture 56 - Filter call record types.....	59
Picture 57 - Dss key settings.....	60
Picture 58 – WLAN.....	61
Picture 59 -wireless network.....	62
Picture 60 - webpage wireless connect.....	62
Picture 61 - AP info.....	63
Picture 62 -line configurations.....	64
Picture 63 - Configure Advanced Line Options.....	65
Picture 64 - IP Mode.....	66
Picture 65 - DHCP network mode.....	66
Picture 66 - PPPoE network mode.....	67
Picture 67 - Static IP network mode.....	67
Picture 68 - IPv6 Static IP network mode.....	68
Picture 69 - The phone configures the web server type.....	71
Picture 70 - Menu Password.....	72
Picture 71 - Menu password setting.....	72
Picture 72 - Menu password input.....	73
Picture 73 - keyboard password setting.....	73
Picture 74 - webpage keyboard password setting.....	74

Picture 75 - Page auto provision Settings.....	75
Picture 76 - Phone auto provision settings.....	75
Picture 77 - Web page firmware upgrade.....	78
Picture 78 - firmware upgrade.....	80
Picture 79 - factory reset.....	80
Picture 80 - web login.....	81
Picture 81 - default password prompt.....	82
Picture 82 - Network settings.....	84
Picture 83 - Service Port Settings.....	85
Picture 84 - Dial plan settings.....	92
Picture 85 - Custom setting of dial - up rules.....	93
Picture 86 - Dial rules table (1).....	94
Picture 87 - Dial rules table (2).....	95
Picture 88 - time plan.....	104
Picture 89 - Webpage Tone.....	105
Picture 90 - Web cloud phone book Settings.....	107
Picture 91 - IP Camera List.....	111
Picture 92 - Web Filter settings.....	112
Picture 93 - Web Filter Table.....	112
Picture 94 - Certificate of settings.....	113
Picture 95 - Device certificate setting.....	113
Picture 96 - Input and output settings.....	114
Picture 97 - Input and output settings.....	116
Picture 98 - Web capture.....	117

2 Table

Table 1 - interface	12
Table 2 - Keypad Icons.....	13
Table 3 - Status Prompt and Notification Icons.....	13
Table 4 - DSSkey Icons.....	14
Table 5 -Look-up Table of Characters	16
<i>Table 6 - Key Description.....</i>	17
Table 7 - Standby interface.....	18
Table 8 - dial interface.....	19
Table 9 - Commonly used icons.....	20
Table 10 - keyboard.....	20
Table 11 - Keyboard numbers & characters.....	21
Table 12 - Taking mode.....	29
Table 13 - Intercom configure.....	38
Table 14 - MCAST Parameters on Web.....	39
Table 15 - set time Parameters.....	47
Table 16 - QoS & VLAN.....	69
Table 17 - Auto Provision.....	75
Table 18 - firmware upgrade.....	78
Table 19 - Service port.....	85
Table 20 - Line configuration on the web page.....	86
Table 21 - Phone 7 dialing methods.....	92
Table 22 - Dial - up rule configuration table.....	93
Table 23 - action plan.....	95
Table 24 - Set the line global configuration on the web page.....	95
Table 25 - General function Settings	96
Table 26 - Voice settings.....	100
Table 27 - Multicast parameters.....	102
Table 28 – Time & Date settings.....	102
Table 29 - time plan.....	103
Table 30 - function key.....	109
Table 31 - Softkey configuration.....	110
Table 32 - Input and output parameters	114
Table 33 - Trouble Cases.....	117

3 Safety Instruction

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

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

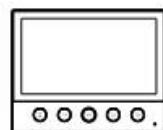
4 Overview

4.1 Overview

i53W is an indoor station with 7-inch color touch screen and rich interfaces. It is mainly used in residential area, villa, office building and other places for receiving calling and communicating through the door phone and achieving remote door-opening. It provides more reliable security assurance and the easier access control for the users, creating a safe and comfortable living environment.

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 i53W. This document may not apply to the latest version of the software. If you have any questions, you can use the help prompt interface that comes with the i53W device, or download and update your user manual from the official website.

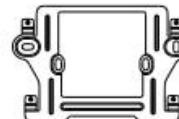
4.2 Packing Contents



Indoor Station



Quick Installation Guide



Wall-mount Bracket



10 pin Cable*1



4 pin Cable*1



2 pin Cable*1



TA4*30mm Screw*2



PM4*16mm Screw*2



Screw fixing seat*2

5 Install Guide

5.1 Use PoE or external Power Adapter

i53W supports two power supply modes, external power adapter and Ethernet (PoE) switch power supply mechanism

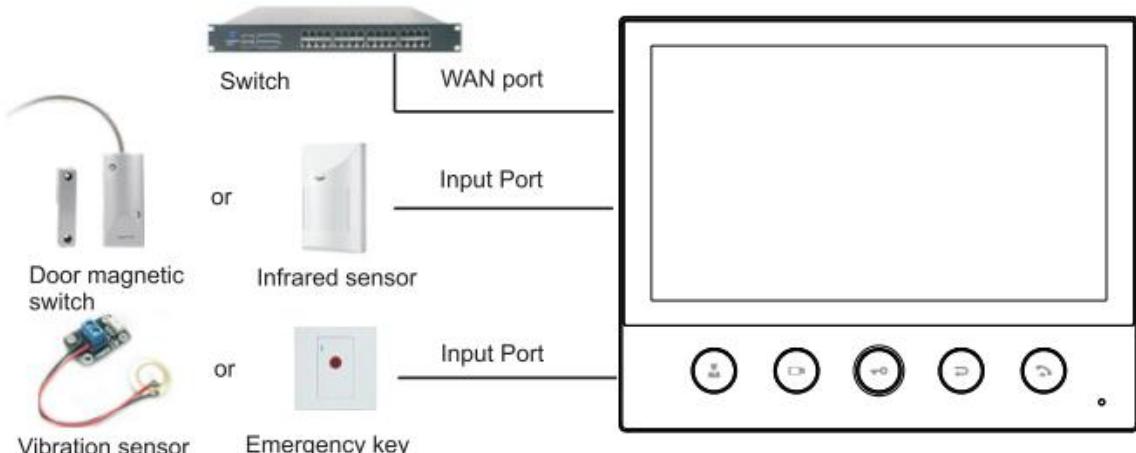
PoE power supply saves the space and cost of providing the device additional power outlet. With a PoE switch, the device can be powered through a single Ethernet cable which is also used for data transmission. By attaching UPS system to PoE switch, the device can keep working at power outage just like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

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

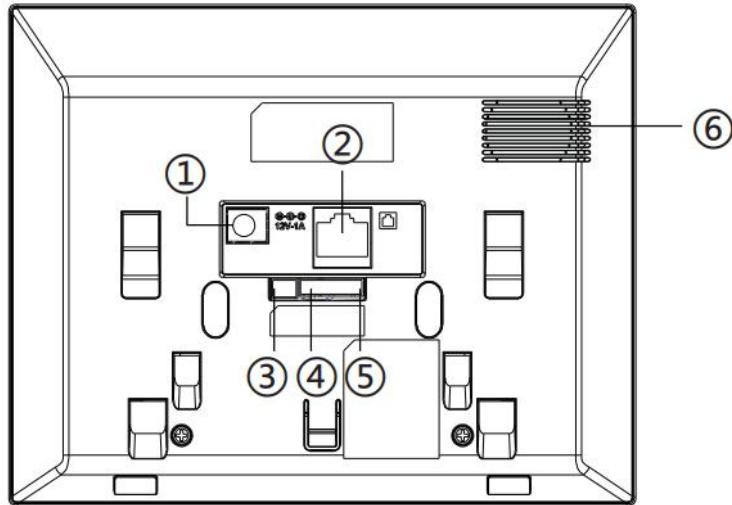
Please use the power adapter supplied by Fanvil and the PoE switch met the specifications to ensure the device work properly.

5.2 Desktop Installation

5.2.1 Peripheral connection



Picture 1 - Peripheral connection diagram



Picture 2 - interface

Table 1- interface

No.	Interface	Description
①		DC Power interface: 12V/1A input.
②		Ethernet interface: standard RJ45 interface, 10/100M adaptive, it is recommended to use CAT5 or CAT5E network cable.
③		Industrial power interface
④		2 sets of RS485 interfaces: can be connected to card reader, sensor etc.
⑤		8 sets of alarm input interfaces: input devices for connecting switches, infrared sensor, door sensor, vibration sensors etc.
⑥		Speaker

6 Appendix Table

6.1 Appendix I - Icon

Table 2 - Keypad Icons

	Control center
	Video Surveillance
	Unlock key
	Return key
	Answer key

Table 3- Status Prompt and Notification Icons

	Call out
	Call in
	Mute Microphone
	Call voice quality
	Call voice encryption
	Call Hold
	HD Audio
	Network Disconnected
	Enable VLAN
	Enable VPN
	Keyboard locked
	Call forward activated
	Auto-answering activated

	Connecting WIFI
	Wi-Fi network abnormal
	SIP Hotspot
	DND
	Missed call
	SMS
	Unread voice message
	Network storm

Table 4- DSSkey Icons

Icon	Translate
	BLF/NEW CALL
	BLF/BXFER
	BLF/AXFER
	BLF/CONF
	BLF/DTMF
	Presence
	MWI
	Speed Dial
	Intercom

	Call Park
	Call forward
	Key Event
	URL/Action URL
	BLF List
	Multicast
	Memory Key None
	None
	Line
	DTMF

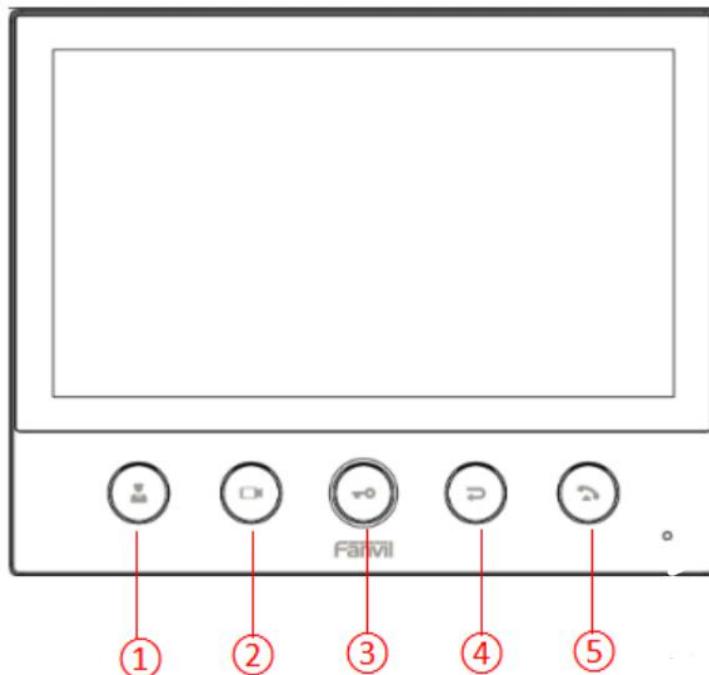
6.2 Appendix II –Function key state definition

Table 5-Look-up Table of Characters

Type	Icon	State	Description
Line Key		Gray	Line is not configured
		Green On	Line ready (Registered)
		Red Blinking	Line is trying to register
		Red Blinking	Line error (Registration failure)
BLF		Green On	Subscription number is idle.
		Red On	Subscription number is busy.
		Red On	Subscription number is dialing.
		Off	Subscription number is unavailable.
Presence		Green On	Subscription number is idle.
		Red On	Subscription number is busy.
		Red On	Subscription number is dialing.
		Off	Subscription number is unavailable.
DND		Red On	Enable DND
		Off	Disable DND
MWI		Green Blinking	New voice message waiting
		Off	No new voice message

7 User Getting Started

7.1 Button description



Picture 3 - Key Description

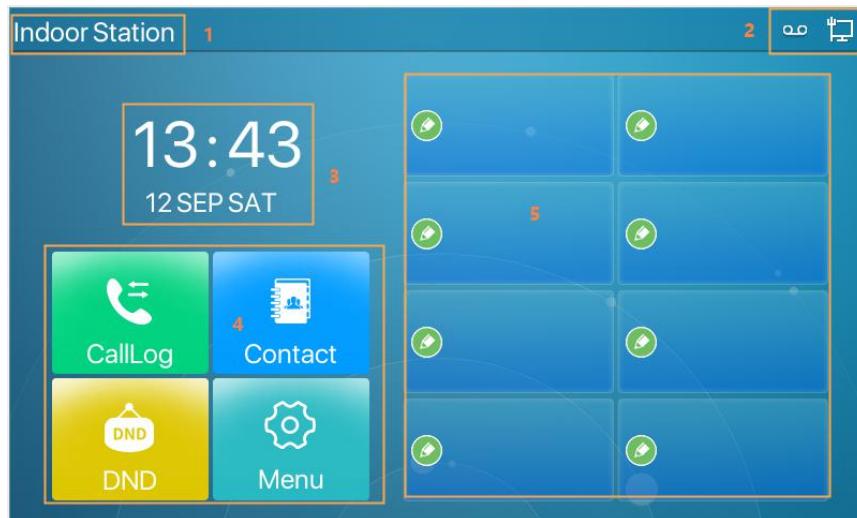
The picture above shows the button layout of the device. Each button provides its own specific function. The user can operate the device by referring to the description of the buttons in the illustrations in this section.

Table 6- Key Description

Number	The keypad names	Instruction
1	Control center	One key to call the set number; if there is no setting, press to enter the setting interface
2	Video Preview	One key to view the access control/outdoor situation; if there is no setting, press to enter the setting interface
3	Unlock key	One-key unlock during a call
4	Return key	Hang up during a call Menu, return to the upper level directory
5	Answer key	Incoming call, answer the call Open the dial pad in standby

7.2 Introduction to the User

7.2.1 Standby interface



Picture 4- Standby interface

The figure above 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 displays the welcome message, time and date, and status information (such as automatic answer, network connection status, etc.).

Table 7- Standby interface

Number	Description
1	Welcome word, number
2	Status icon
3	Time, Date
4	Common Functions
5	Custom function

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

7.2.2 Dial interface

The dial interface is mainly used to make calls, enter the contacts, call history interface



Picture 5-Menu interface

Table 8- dial interface

Number	Features	Description
1	Return key	Return to the previous menu, the main interface
2	Dialpad	Enter the dial interface
3	Contact	Enter the contact interface, view/edit contacts
4	Call records	Enter the call log interface, view the call log
5、6	Turn page	When the list supports multiple pages, page up and down
7	Home	Back to main interface
8	Volume Down Key	Volume Adjustment
9	Volume Up Key	Volume Adjustment
10	Dial key	After entering the number, press to call out
11	Numeric keypad	0-9, *, #
12	Input box	Enter number
13	Delete key	Delete entered number
14	End/input method/delete	End: Exit the current interface 123: Switch input method, 123/abc/ABC/Abc/2aB Delete: delete the entered number
15	Number matching record	After entering the number, the query record displays

7.2.3 Commonly used icons on the interface

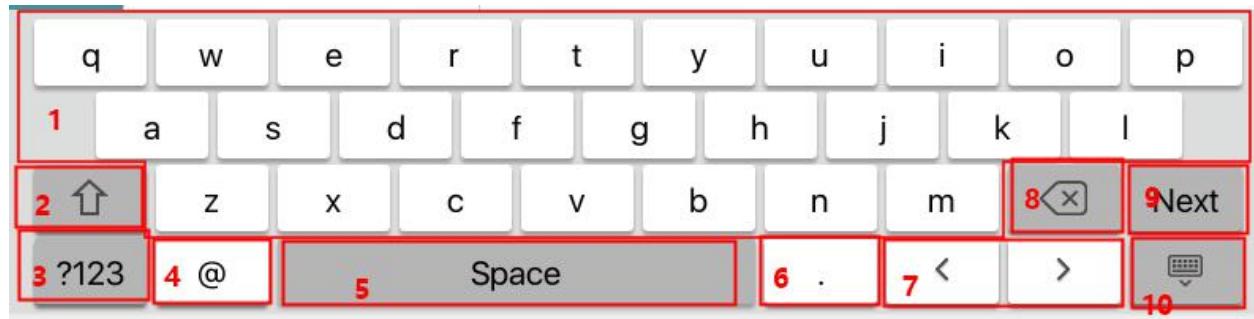
Introduction to icons commonly used by equipment.

Table 9- Commonly used icons

Icon	Description	Icon	Description
○	Back to main interface	^	Previous page
▽	Next page	◀	Return
🔍	Search for contacts	+	Add to
✓	Save		

7.3 Use of touch keyboard

The device supports using the touch keyboard to enter data.



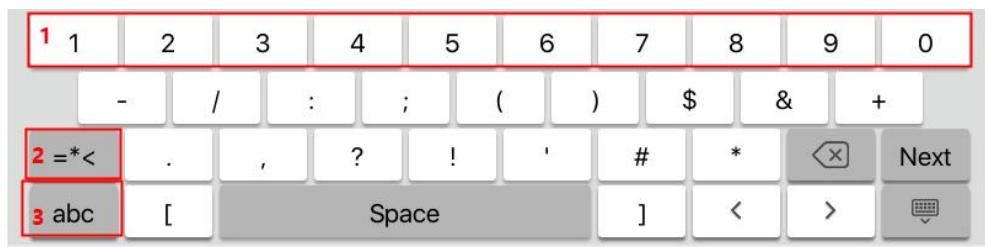
Picture 6- keyboard

Table 10 - keyboard

Number	Function keys	Description
1	26 English letters	Type letters
2	↑	Switch to uppercase letter input mode
3	?123	Switch to number, special character input mode
4、6	Special characters	@、.
5	Space Bar	Enter a Space
7	Switch	Switch input characters left and right

		right
8	Delete	Delete entered characters
9	Next	Switch to the next edit box
	Done	Save operation
10		Hide keyboard

Tap on the screen to switch **?123** to number and special character input mode.



Picture 7 - Keyboard numbers & characters

Table 11 - Keyboard numbers & characters

Number	Function keys	Description
1	Number Key	Type in data
2	=*<	Switch to special character input mode
3	abc	Switch lowercase English letter input mode

7.4 Phone Status

The phone status includes the following information about the phone:

- Network Status:
 - VLAN ID
 - IPv4 or IPv6 status
 - IP Address
 - Network Mode
- The Phone Device Information:
 - Mac Address
 - Phone Mode
 - Hardware Version number
 - Software Version number
 - Phone Storage (RAM and ROM)

System Running Time

- SIP Account Information:
 - SIP Account
 - SIP Account Status (registered / uncommitted / trying / time out)
- TR069 Connection Status (Displays only in the phone interface state)

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

Device interface: When the device is in standby, press [Menu] >> [Status], select options to view corresponding information, as shown in the figure:

Network		
Network	1. Vlan Id	None
Phone	2. Mode	DHCP/IPv4
Account	3. IPv4	172.16.12.207
TR069		
RTP		

Picture 8- keyboard

- WEB interface: Refer to 7.5Web management to log in the phone page, enter the [System] >> [Information] page, and check the phone status, as shown in the figure:

The screenshot shows the 'System' section of the phone's configuration interface. The left sidebar lists various system settings like Network, Line, Settings, Phonebook, Call logs, Function Key, Security, Device Log, and Security Settings. The main content area displays 'System Information' with details such as Model: i53W, Hardware: V1.0, Software: 0.1.2, Uptime: 00 : 18 : 43, MEMInfo: ROM: 29.3/128(M) RAM: 37.6/94(M), and System time: 2020-9-9 15:55 (SNTP). Below this is the 'Network' section, which includes 'WAN' and 'IPv4' settings. The 'WAN' section shows Network mode: DHCP, MAC: 0c:38:3e:46:1e:62. The 'IPv4' section shows IP: 172.16.12.207, Subnet mask: 255.255.255.0, and Default gateway: 172.16.12.1. The final section shown is 'SIP Accounts' with five entries: Line 1 (0305@172.16.1.2:5060, Registered), Line 2 (N/A, Inactive), Line 3 (N/A, Inactive), Line 4 (N/A, Inactive), and Line 5 (N/A, Inactive).

Picture 9- phone status

7.5 Web Management

Phone can be configured and managed on the web page of the phone. The user needs to enter the IP address of the phone in the browser at first and open the web page of the phone.

The user can check the IP address of the phone by pressing [Menu] >> [Status].



Picture 10- Login page

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

7.6 Network Configurations

The i53W device supports two network connection methods: wired network connection and wireless network connection. Users need to choose the corresponding connection method according to their own situation.

The device uses an IP network connection to provide services. Unlike traditional devices based on line circuit technology, IP devices are connected to each other through the network to exchange data packets and data based on the device's IP address.

To enable the device, the network configuration must first be properly configured. To configure the network, users need to find the device function menu button [**Menu**] >> [**Advanced Settings**] >> [**Network**] >> [**Network Settings**].

The default password for entering advanced settings is "123".



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

The device supports three network types, IPv4/IPv6/IPv4&IPv6

There are three common IP configuration types for IPv4

- DHCP – This is the mode that automatically obtains 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 users to manually configure each IP parameter, including IP address, mask, gateway and primary DNS server and backup DNS server. This usually applies to some professional network user environments.
- PPPoE – This option is usually suitable for users who connect to the network through a broadband service account. To establish a PPPoE connection, the user should provide the user name and password provided by the operator.
- The default configuration of the device is the network mode of automatic configuration
-
- There are two common IP configuration types for IPv6
- DHCP – This is the mode that automatically obtains 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 users 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 specific configuration and use, please refer to [10.6.2.1 Network Settings](#) and [10.5 WiFi](#)

7.7 SIP Configurations

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

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

Phone interface: To manually configure a line, the user can long press the line button or use the function menu button [**Phone Settings**] >> [**Account**] >> [**Line**] to configure each line, and click OK to save the configuration.

Line

Register Settings >>

Line Status:	Registered	Activate:	<input checked="" type="checkbox"/> ?
Username:	<input type="text" value="0305"/>	Authentication User:	<input type="text"/>
Display name:	<input type="text" value="0305"/>	Authentication Password:	<input type="text" value="*****"/>
Realm:	<input type="text"/>	Server Name:	<input type="text"/>

SIP Server 1:

Server Address:	<input type="text" value="172.16.1.2"/>	?
Server Port:	<input type="text" value="5060"/>	?
Transport Protocol:	<input type="text" value="UDP"/> ?	
Registration Expiration:	<input type="text" value="3600"/>	second(s) ?

SIP Server 2:

Server Address:	<input type="text"/>	?
Server Port:	<input type="text" value="5060"/>	?
Transport Protocol:	<input type="text" value="UDP"/> ?	
Registration Expiration:	<input type="text" value="3600"/>	second(s) ?

Proxy Server Address:

Proxy Server Port:

Proxy User:

Proxy Password:

Backup Proxy Server Address:

Backup Proxy Server Port:

Picture 11- Web Line Registration

8 Basic Function

8.1 Making Phone Calls

■ Default Line

The equipment provides 6 SIP line services. If all the 6 lines are configured successfully, the user can use any line to make or receive calls. If the user has set a default line, the number or name currently used by default will be displayed in the upper left corner of the screen interface. To enable or disable the default line function, the user can go through [Menu] >> [Function] >> [Basic Settings] >> [General] or complete the settings on the web page ([Web Page] >> [Settings] >> [Function Settings]) >> [Basic Settings]).

■ Dialing Methods

User can dial a number by:

- Entering the number directly
- Selecting a phone number from phonebook contacts (Refer to [10.2.1 Local contacts](#))
- Selecting a phone number from cloud phonebook contacts (Refer to [10.2.3 Cloud Phone Book](#))
- Selecting a phone number from call logs (Refer to [10.3 Call Log](#))
- Redialing the last dialed number
- Press the management center button to call



Picture 12- Dial interface

When calling a number, the user can press [End] or press the return button to

cancel the call



Picture 13- Call interface

8.2 Answer a call

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



Picture 14- Voice call interface

The user can answer the call by pressing the button or the interface. To reject an incoming call, the user presses or the reject button on the interface.

8.3 Call interface

When the call is established, the user will see the call mode screen as shown below:



Picture 15- Taking interface

Table 12 - Taking mode

Number	The keypad names	Instruction
1	Contact Name	The name of the other party
2	Contact Number	Call the other party's number
3	Call duration	Call duration
4	Mute icon	Icon indication after the call is muted
5	Voice quality, HD, voice encryption	Display the current call voice quality, voice call encryption and other icon indicators
6	End	Hang up
7、8	Volume addition and subtraction	Adjust call volume

8.4 Unlock

Web page settings [Function keys] >> [Function keys]>>[DSSKey3] is DTMF, the value is the access control caller's password to open the door, as shown below:

Press during a call , Can open the door with one key.

Function Key Settings							
Key	Type	Name	Value	Subtype	Line	Media	PickUp Number
DSS Key 1	DTMF		123	None	AUTO	DEFAULT	
DSS Key 2	None			None	AUTO	DEFAULT	

Picture 16- Web page configuration DTMF

8.5 End of call



Picture 17- Call end interface

When the user's call ends

- 1) You can press to end the call
- 2) You can press to end the call
- 3) You can press the end button on the call interface

8.6 Video preview

The user can bind the video stream of the camera of the door phone and view the situation outside the house with one click

Web page settings **[Function keys] >> [Function keys]>>[DSSKey2]** is the url, the value is the RTSP URL of the camera, and the setting is completed and submitted.

Press the video surveillance button in standby.

Function Key Settings								
Key	Type	Name	Value	Subtype	Line	Media	PickUp Number	
DSS Key 1	URL	1	rtsp://172.16.12.6	None	AUTO	DEFAULT		

Picture 18 - Web page configuration hard button preview

Web page settings **[Function keys] >> [Side key]**, select any key, the type is url, the value is the RTSP URL of the camera, and the setting is completed and submitted.

In standby, press the shortcut key set on the right side of the standby interface.

Side Dsskey Settings							
Key	Type	Name	Value	Subtype	Line	Media	PickUp Number
F 1	URL	door1	rtsp://172.16.12.1	None	0305@SIP1	DEFAULT	
F 2	URL	door2	rtsp://admin:admin@	None	0305@SIP1	DEFAULT	
F 3	Key Event			Phonebook	AUTO	DEFAULT	
F 4	Key Event			Call Logs	AUTO	DEFAULT	

Picture 19- Webpage configuration standby interface preview



Picture 20- Standby interface display after configuration

8.7 Dial query

The device defaults to enable the dial query function, open the dial pad to dial, enter one or more numbers, the dial interface will automatically match the call record, the number list in the contact, click to select the number and call out.

8.8 Auto Answer

The user can enable the automatic answering function on the device, and the device can automatically answer after a call comes in. Auto answer can be activated by distinguishing lines.

The user can start the automatic answer function on the device interface or the web interface.

- **Device interface:**

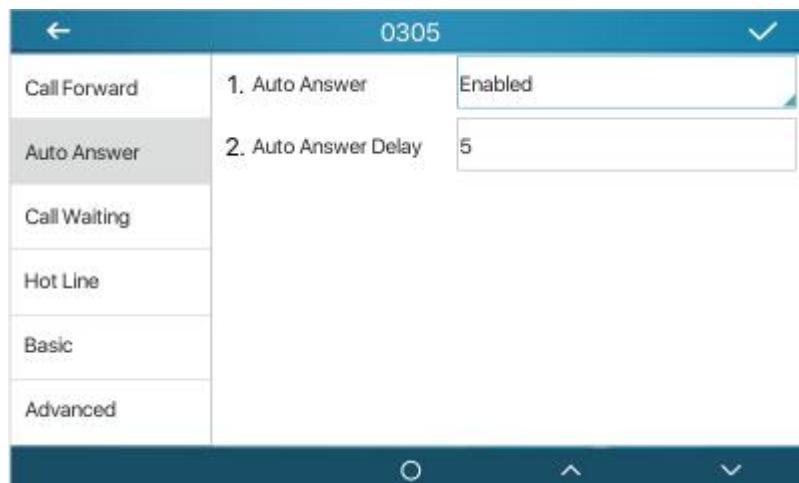
Press [Menu]>>[Function]>>[Auto Answer] button;

Press the button to select the line, use the left/right navigation key to turn

on/off the auto answer option, set the auto answer time, the default is 5 seconds

Press to save when finished

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



Picture 21 - Line 1 enables auto-answering

- **Web interface :**

Log into the device webpage, enter [Line]>>[SIP], Select [**Basic Settings**], enabled automatic answering, set the automatic answering time and click submit.

Line	0305@SIP1						
Register Settings >>							
Basic Settings >>							
Enable Auto Answering:	<input checked="" type="checkbox"/>		Auto Answering Delay:	5	(0~120)second(s)		
Call Forward Unconditional:	<input type="checkbox"/>		Call Forward Number for Unconditional:	<input type="text"/>			
Call Forward on Busy:	<input type="checkbox"/>		Call Forward Number for Busy:	<input type="text"/>			
Call Forward on No Answer:	<input type="checkbox"/>		Call Forward Number for No Answer:	<input type="text"/>			
Call Forward Delay for No Answer:	<input type="text"/> 5	(0~120)second(s)		Transfer Timeout:	<input type="text"/> 0	second(s)	
Subscribe For Voice Message:	<input type="checkbox"/>		Voice Message Number:	<input type="text"/>			
Voice Message Subscribe Period:	<input type="text"/> 3600	(60~999999)second(s)					
Dial Without Registered:	<input type="checkbox"/>		Enable Missed Call Log:	<input checked="" type="checkbox"/>			
DTMF Type:	<input type="text"/> AUTO		DTMF SIP INFO Mode:	<input type="text"/> Send 10/11			
Request With Port:	<input checked="" type="checkbox"/>		Enable DND:	<input type="checkbox"/>			
Use STUN:	<input type="checkbox"/>		Use VPN:	<input checked="" type="checkbox"/>			

Picture 22- Web page to start auto-answering

8.9 Mute

You can turn on the silent mode and turn off the microphone of the device during a call, so that the other party cannot hear the local voice. Under normal circumstances, the silent mode is automatically turned off as the call ends. You can also enable the keep mute function on any interface (such as the idle interface) to automatically mute the ringtone when a call comes in.

8.9.1 Mute during a call

Press the mute button  on the call interface during a call: The mute button on the device turns on the red light.

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



Picture 23- Mute the call

- Unmute the call: Press the unmute  on the device again. The mute icon is no longer displayed on the call interface. The red light of the device mute button is turned off.

8.9.2 Mute when ringing

- Turn on mute ringing: press the mute button on the incoming call interface when the device is ringing: 

The mute icon on the incoming call interface of the device  changes

to and there is no ringtone. After hanging up, the device will still ring the next time there is an incoming call.



Picture 24 - Ringing mute

8.10 DND

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

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

- Phone interface: Default standby mode,
 - 1) Press [**DND**] button to enter the DND setting interface, select line or phone to enable DND, the icon will become red. The phone status prompt bar will have a DND icon.

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

Press [**Menu**] >> [**Function**] >> [**Basic Settings**] >> [**DND**] button to enter the edit page of [**DND**].

Use the left/right navigation keys to select the line to adjust the DND mode and status. After finishing, press the [**OK**] button to save.

The user will see that the DND icon turns red, and the SIP line has enabled the DND mode.

DND		
Call Forward	1. DND Mode	Line
Auto Answer	2. DND Timer	Disabled
Call Waiting	3. Line	SIP1
Hot Line	4. State	Enabled
Basic		
Advanced		

Picture 25- DND setting interface

Users can also use the Do Not Disturb timer. After setting, within the time range, the Do Not Disturb function will be automatically turned on and the DND icon will turn red

DND		
Call Forward	1. DND Mode	Line
Auto Answer	2. DND Timer	Enabled
Call Waiting	3. DND Start Time	15 : 00
Hot Line	4. DND End Time	17 : 30
Basic	5. Line	SIP1
Advanced	6. State	Enabled

Picture 26- DND timer

Web interface: Go to [Settings] >> [Function Settings] >> [DND Settings], set the type of DND (off, phone, line), and DND timing function.

Basic Settings >>			
Tone Settings >>			
DND Settings >>			
DND Option:	Off		
Enable DND Timer:	<input checked="" type="checkbox"/>		
DND Start Time:	15	:	0
DND End Time:	17	:	30

Picture 27- DND Settings

The user opens the DND of a specific line on the webpage: enter [Line] >> [SIP] >> [Basic Settings], and enable DND.

Basic Settings >>

Enable Auto Answering:	<input type="checkbox"/>	?	Auto Answering Delay:	5	(0~120)second(s)	?	
Call Forward Unconditional:	<input type="checkbox"/>	?	Call Forward Number for Unconditional:	<input type="text"/>			
Call Forward on Busy:	<input type="checkbox"/>	?	Call Forward Number for Busy:	<input type="text"/>			
Call Forward on No Answer:	<input type="checkbox"/>	?	Call Forward Number for No Answer:	<input type="text"/>			
Call Forward Delay for No Answer:	<input type="text"/> 5	(0~120)second(s)	?	Transfer Timeout:	<input type="text"/> 0	second(s)	?
Subscribe For Voice Message:	<input type="checkbox"/>	?	Voice Message Number:	<input type="text"/>			
Voice Message	<input type="text"/> 3600		Subscribe Period:	(60~999999)second(s)			
Dial Without Registered:	<input type="checkbox"/>	?	Enable Missed Call Log:	<input checked="" type="checkbox"/>	?		
DTMF Type:	<input type="text"/> AUTO		DTMF SIP INFO Mode:	<input type="text"/> Send 10/11		?	
Request With Port:	<input checked="" type="checkbox"/>	?	Enable DND:	<input type="checkbox"/>	?	?	
Use STUN:	<input type="checkbox"/>	?	Use VPN:	<input checked="" type="checkbox"/>	?	?	

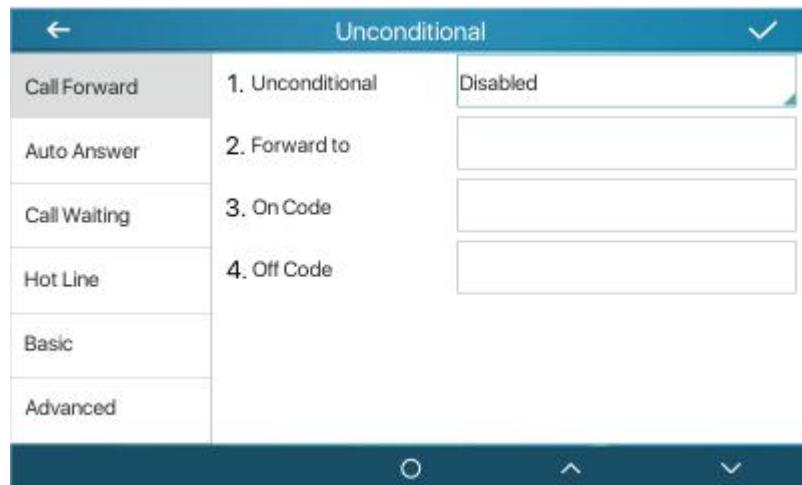
Picture 28 - Line DND

8.11 Call Forward

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

There are two types,

- **Unconditional Call Forward** – Forward any incoming call to the configured number.
 - **Call Forward on No Answer** – When user does not answer the incoming call after the configured delay time, the incoming call will be forwarded to the configured number.
- Phone interface: Default standby mode
- 1) Press **[Menu] >> [Function] >> [Call Forwarding]** to select the line
 - 2) Select the type of call forwarding. Turn on and set the number to be transferred, etc.
 - 3) Click the upper right corner to save the changes.



Picture 29- Set call forward

Web interface: Enter [Line] >> [SIP]>> [Basic Settings], and set the forward type, number, and time.

Line 0305@SIP1

Register Settings >>

Basic Settings >>

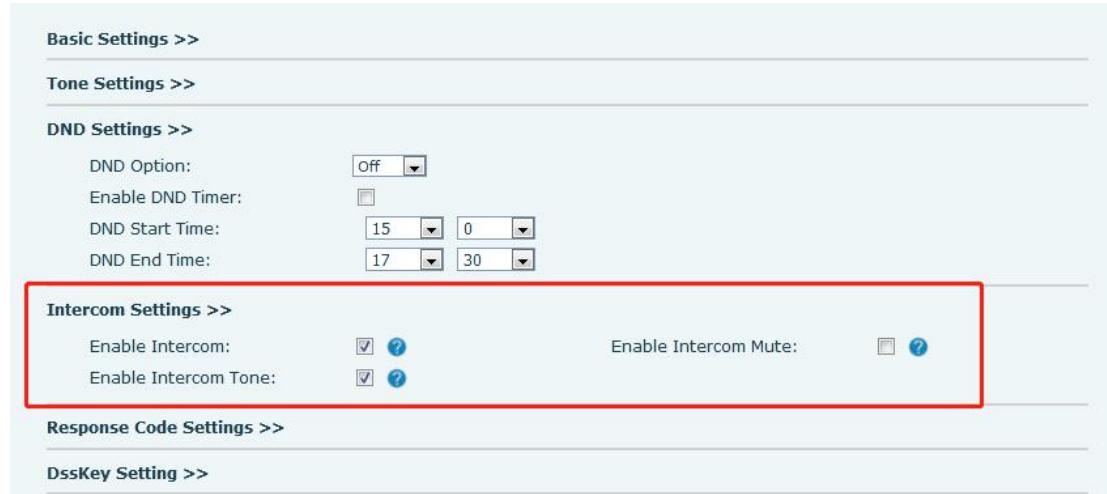
Enable Auto Answering:	<input type="checkbox"/>	(?)	Auto Answering Delay:	5	(0~120)second(s)	(?)
Call Forward:	<input type="checkbox"/>	(?)	Call Forward Number for Unconditional:			
Unconditional:	<input type="checkbox"/>	(?)	Call Forward Number for Busy:			
Call Forward on Busy:	<input type="checkbox"/>	(?)	Call Forward Number for No Answer:			
Call Forward on No Answer:	<input type="checkbox"/>	(?)	Call Forward Delay for No Answer:	5	(0~120)second(s)	(?)
Transfer Timeout:	0	second(s)	(?)			
Subscribe For Voice Message:	<input type="checkbox"/>	(?)	Voice Message Number:			
Voice Message:	3600			(?)		
Subscribe Period:	(60~999999)second(s)					
Dial Without Registered:	<input type="checkbox"/>	(?)	Enable Missed Call Log:	<input checked="" type="checkbox"/>	(?)	
DTMF Type:	AUTO	(?)	DTMF SIP INFO Mode:	Send 10/11	(?)	
Request With Port:	<input checked="" type="checkbox"/>	(?)	Enable DND:	<input type="checkbox"/>	(?)	
Use STUN:	<input type="checkbox"/>	(?)	Use VPN:	<input checked="" type="checkbox"/>	(?)	

Picture 30- Set call forward

9 Advance Function

9.1 Intercom

After the device enables intercom, it can automatically answer intercom calls.



Picture 31- Web Intercom configure

Table 13 - Intercom configure

Parameter	Description
Enable Intercom	When intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after specific delay.
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers the intercom call during a call. If the current call is intercom call, the phone will reject the second intercom call

9.2 MCAST

This feature allows user to make some kind of broadcast call to people who are in multicast group. User can configure a multicast DSS Key on the phone, which allows

user to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address without involving SIP signaling. You can also configure the phone to receive an RTP stream from pre-configured multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.

Picture 32- Multicast Settings Page

Table 14 - MCAST Parameters on Web

Parameters	Description
Normal Call Priority	Define the priority of the active call, 1 is the highest priority, 10 is the lowest.
Enable Page Priority	The voice call in progress shall take precedence over all incoming paging calls.
Name	Listened multicast server name
Host: port	Listened multicast server's multicast IP address and port.

Multicast:

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

- Receiver will receive multicast call and play multicast automatically.

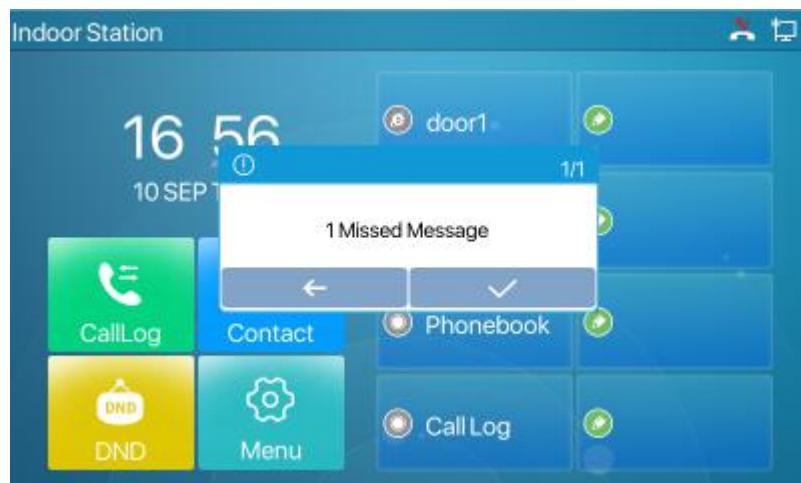
Dynamic multicast:

Function description: Send multicast configuration information through Sip Notify signaling. After receiving the information, the device configures it in the system for multicast monitoring or cancels multicast monitoring in the system

9.3 SMS

9.3.1 SMS

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



Picture 33- SMS icon

Send messages:

- Go to [Application] >> [SMS].
- Users can create new messages, select lines and send numbers.
- After editing is complete, click Send.

View SMS:

- Use the navigation keys to select the standby icon [message]
- After selecting, press the navigation key [OK] to enter the SMS inbox interface.
- Select the unread message and press [OK] to read the unread message.

9.3.2 MWI (Message Waiting Indicator)

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



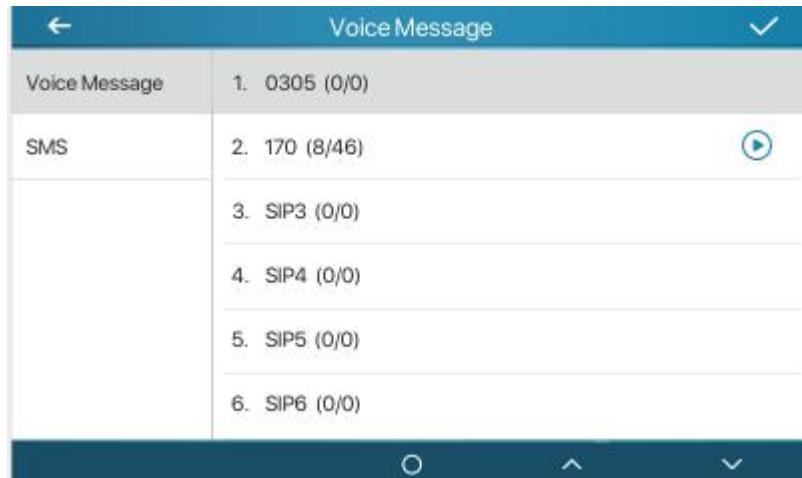
Picture 34- New Voice Message Notification

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

When the phone is in the default standby state,

- The voicemail icon displays the number of unread voicemail messages.
- Click the icon to view the total number of voicemail messages, or listen to the messages directly in the voicemail interface

- Select [**Message**] under [**Menu**]
- Enter [**Voice Message**] under [**Message**]
- The "8" in brackets on the SIP1 line represents unread voice messages, and "46" represents the total number of voice messages.
- Select the line to enter, enable the message and set the message number, press the upper right corner to save
- After setting the message number, press to listen to the message.



Picture 35 - Voice message interface

9.4 SIP Hotspot

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

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

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

SIP Hotspot Settings	
Enable Hotspot:	Disabled
Mode:	Client
Monitor Type:	Broadcast
Monitor Address:	224.0.2.0
Local Port:	16360
Name:	SIP Hotspot

Line Settings	
Line 1:	Enabled
Line 2:	Enabled
Line 3:	Enabled
Line 4:	Enabled
Line 5:	Enabled
Line 6:	Enabled

Picture 36- Register SIP account

Table 15- SIP hotspot Parameters

Parameters	Description
Device Table	If your phone is set to “SIP hotspot server”, Device Table will display as Client Device Table which connected to your phone. If your phone is set to “SIP hotspot client”, Device Table will display as Server Device Table which you can connect to.
SIP hotspot	
Enable hotspot	Set it to be Enable to enable the feature.
Mode	Choose hotspot, phone will be a “SIP hotspot server”; Choose Client, phone will be a “SIP hotspot Client”
Monitor Type	Either the Multicast or Broadcast is ok. If you want to limit the broadcast packets, you’d better use broadcast. But, if client choose broadcast, the SIP hotspot phone must be broadcast.
Monitor Address	The address of broadcast, hotspot server and hotspot client must be same.
Remote Port	Type the Remote port number.

Configure SIP hotspot server:

The screenshot shows the 'Client Table' section with one entry. The entry has IP 172.16.7.181, MAC 0c:38:3e:23:b5:9f, Alias 1, and Line 1. Below this is the 'SIP Hotspot Settings' section. It includes fields for 'Enable Hotspot' (set to Enabled), 'Mode' (set to Hotspot), 'Monitor Type' (set to Broadcast), 'Monitor Address' (set to 224.0.2.0), 'Local Port' (set to 16300), and 'Name' (set to SIP Hotspot). To the right of these settings are seven small circular icons, likely for cloning or deleting. At the bottom is the 'Line Settings' section with 'Line 1' and 'Line 2' both set to Enabled.

Picture 37- SIP hotspot server configuration

Configure SIP hotspot client:

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

Hotspot Table						
IP	Server name	Online Status	Connection Status	Alias	Line	
172.16.7.167	SIP Hotspot	OnLine	Connected	1	0	Disconnect

SIP Hotspot Settings

Enable Hotspot:	<input type="button" value="Enabled"/>	?
Mode:	<input type="button" value="Client"/>	?
Monitor Type:	<input type="button" value="Broadcast"/>	?
Monitor Address:	<input type="text" value="224.0.2.0"/>	?
Local Port:	<input type="text" value="16380"/>	?
Name:	<input type="text" value="SIP Hotspot"/>	?

Line Settings

Line 1:	<input type="button" value="Enabled"/>	?
Line 2:	<input type="button" value="Enabled"/>	?

Picture 38- SIP hotspot client configuration

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

Call extension number:

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

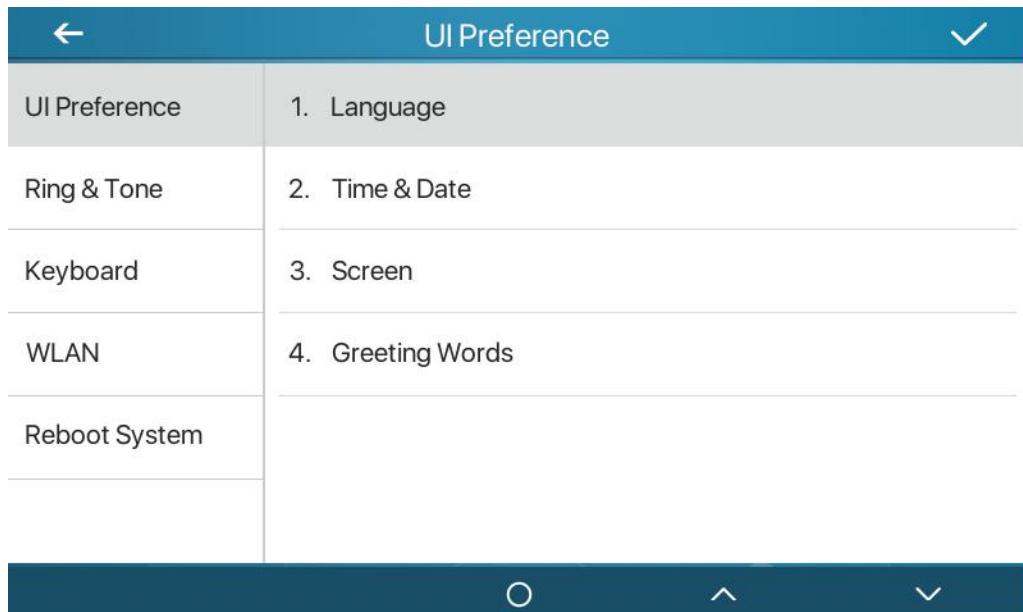
10 Phone Settings

10.1 Basic Settings

10.1.1 Language

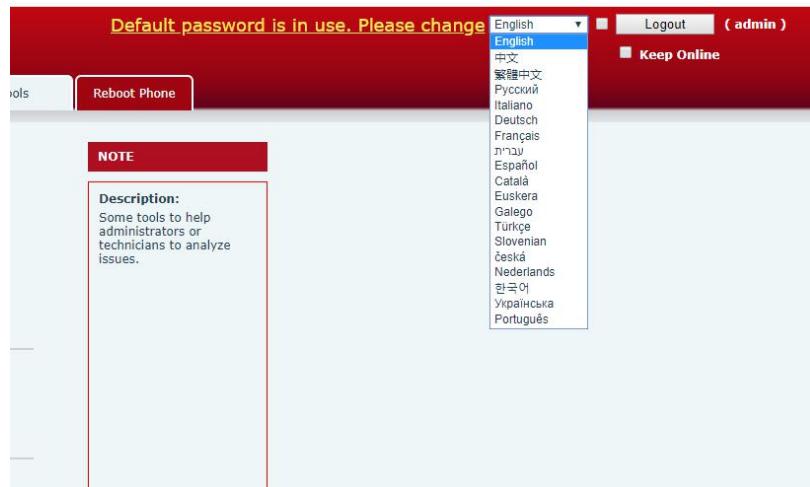
The user can set the phone language through the phone interface or web interface.

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



Picture 39 - Set language

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



Picture 40- web page language setting

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

10.1.2 Time & Date

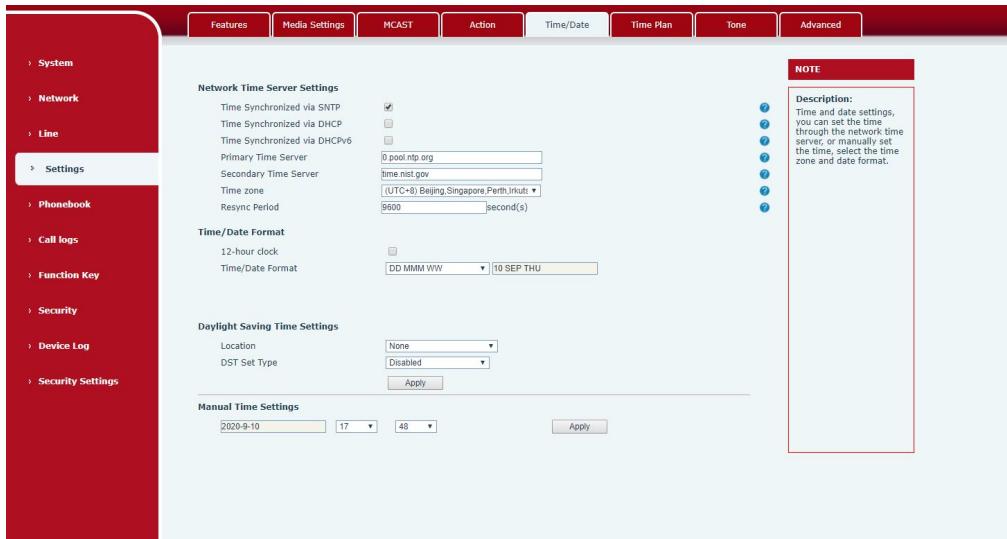
Users can set the phone time through the phone interface and web interface.

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

Time & Date	
UI Preference	1. Mode SNTP
Ring & Tone	2. SNTP Server 0.pool.ntp.org
Keyboard	3. Time Zone (UTC+8) Beijing, Singapore, Perth, I
WLAN	4. Format DD MMM WW
Reboot System	5. 12 Hours Clock Disabled
	6. Daylight Saving Time Disabled

Picture 41- set time & date

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



Picture 42- webpage set time &date

Table 15- set time Parameters

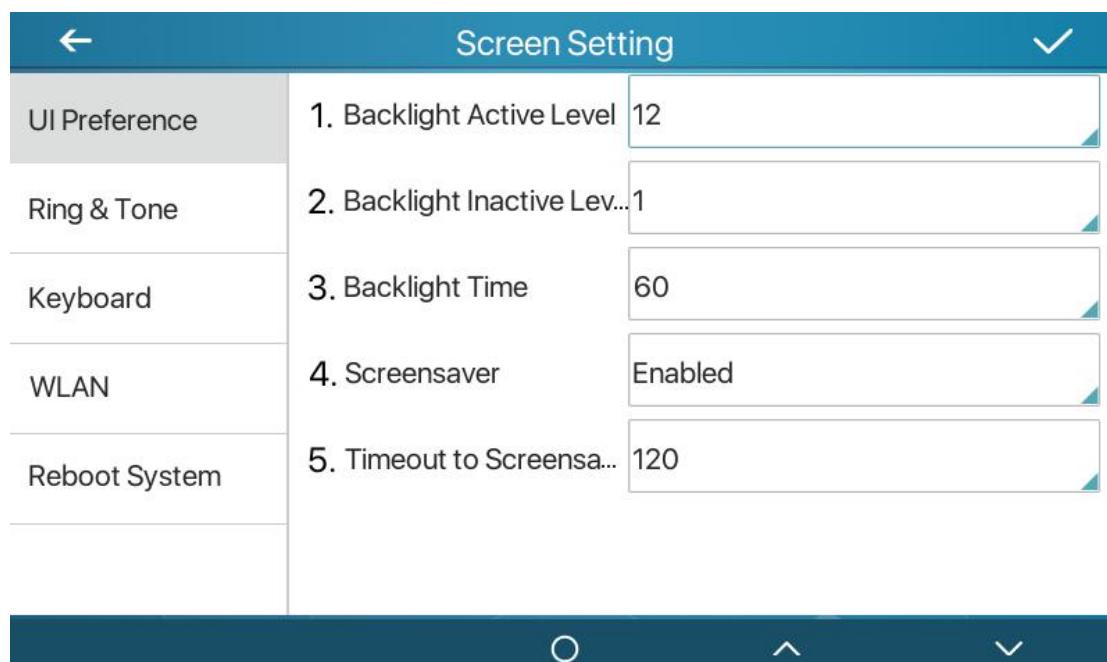
Parameters	Description
Mode	Auto/Manual Auto: Enable network time synchronization via SNTP protocol, default enabled. Manual: User can modify data manually.
SNTP Server	SNTP server address
Time zone	Select the time zone
Time format	Select time format from one of the followings: <ul style="list-style-type: none"> ■ 1 JAN, MON ■ 1 January, Monday ■ JAN 1, MON ■ January 1, Monday ■ MON, 1 JAN ■ Monday, 1 January ■ MON, JAN 1 ■ Monday, January 1 ■ DD-MM-YY ■ DD-MM-YYYY ■ MM-DD-YY

	<input type="checkbox"/> MM-DD-YYYY <input type="checkbox"/> YY-MM-DD <input type="checkbox"/> YYYY-MM-DD
Separator	Choose the separator between year and month and day
12-Hour Clock	Display the clock in 12-hour format
Daylight Saving Time	Enable or Disable the Daylight Saving Time

10.1.3 Screen

The user can adjust the brightness of phone screen in LCD in two ways.

- Slide down the outgoing status bar page in standby mode. Slide down again to adjust phone brightness conveniently.
- Enter the [Menu] >> [Basic Settings]>> [UI Preference]>>[Screen] and then adjust the brightness. Click  to save.



Picture 43- set screen Parameters

- Web interface: Enter [Settings] >> [Advanced], edit screen parameters, and click Submit to save

10.1.3.1 Brightness and backlight

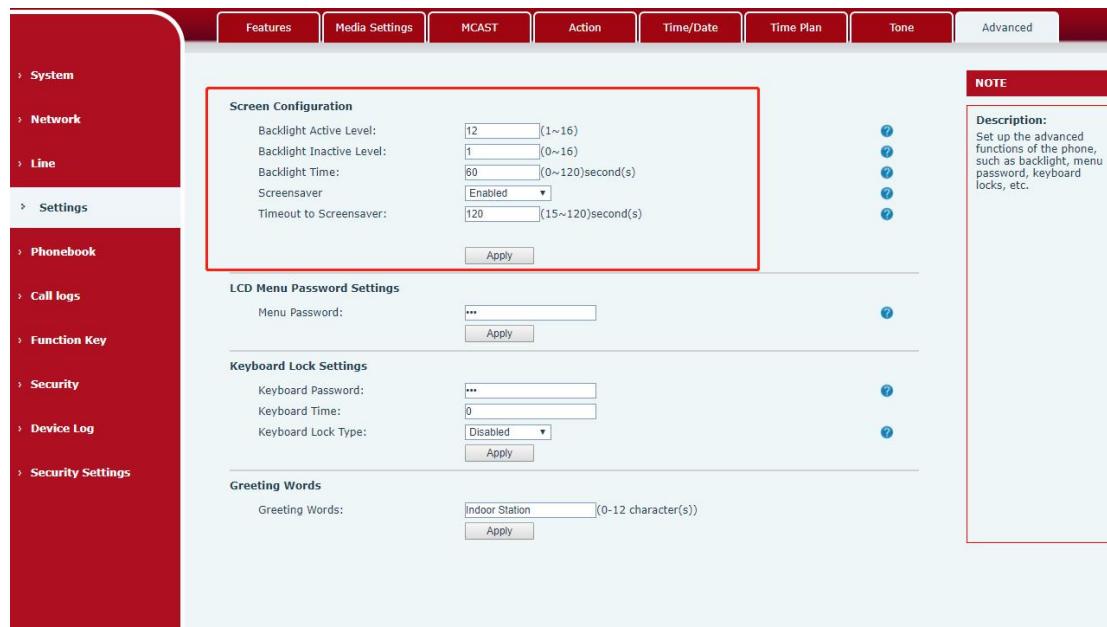
Set the brightness level in the use state from 1 to 16.

Set the brightness level in energy saving mode from 0 to 16 optional.

Set the backlight time, the default is 30 seconds, you can turn it off or choose 15 seconds/30 seconds/45 seconds/60 seconds/90 seconds/120 seconds.

The screen saver can be turned on or off, and it is turned on by default.

Web interface: Enter [**Settings**] >> [**Advanced**], edit the screen parameters, and click Submit to save.



Picture 44- set screen on webpage

10.1.3.2 Screen Saver

- Press [**Screen Settings**] to find the [**Screen Saver**] button, turn on/off the screen saver, set the timeout time, the default is 120S, press to save after finished.
- Return
- to standby after saving, screen saver will display after 120s, as follows



Picture 45- Screensaver

10.1.4 Ring

When the device is in the default standby mode,

- Enter [Menu] >> [Basic settings].
- Enter [Ring&Tone] >> [Ring]

Set ring type and save it by press 

10.1.5 Voice Volume

When the device is in the default standby mode

- Enter [Menu] >> [Basic settings].
- Enter [Ring&Tone] >> [Voice Volume]
- Set volume and save it by press .

10.1.6 Greeting words

The device is in the default standby state.

Press [Menu] to find the [Basic Settings] button.

Press the [UI Preference] button to find the [Welcome] button.

Enter the setting interface, press  to save after completion

Note: Only after the default line selection function is disabled, the welcome message can be displayed in the upper left corner of the standby

10.1.7 Reboot

When the device is in the default standby mode,

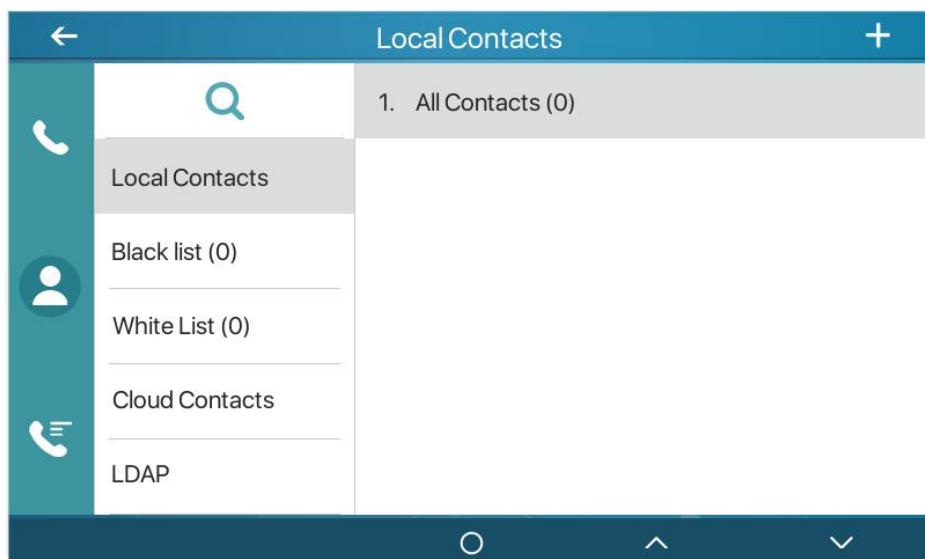
- Enter [Menu] >> [Basic setting] >> [Reboot] item.
- Click [Reboot] to indicate whether to restart the phone.
- Press  to restart the phone or press  to exit the prompt box to return to the configuration interface.

10.2 Phonebook

Users can save contact information in the phone book and dial the contact's phone number directly in the phone book. The user can open the phone book by pressing the function menu button "contact" or the preset button "phone book" on the phone in the default main interface.

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

NOTICE! The device can save up to total 2000 contact records.



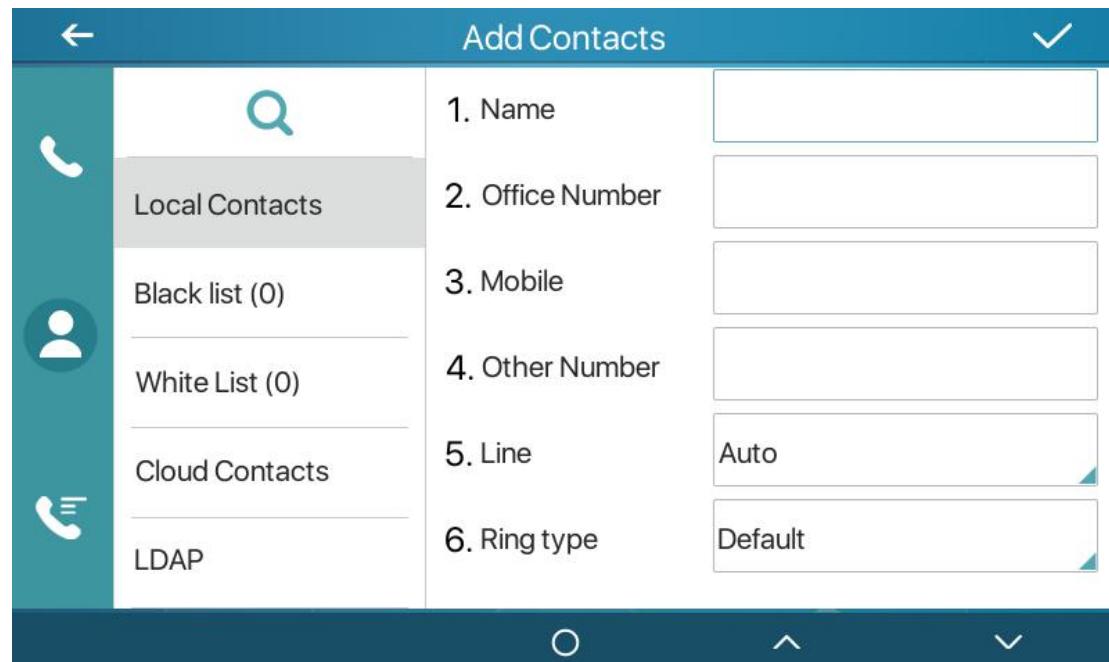
Picture 46- local contact

When there are contact records in the phone book, the contact records will be arranged in the alphabet order. User may browse the contacts with up/down navigator keys. The record indicator tells user which contact is currently focused. User may check the contact's information by pressing [OK] button.

10.2.1.1 Add / Edit / Delete Contact

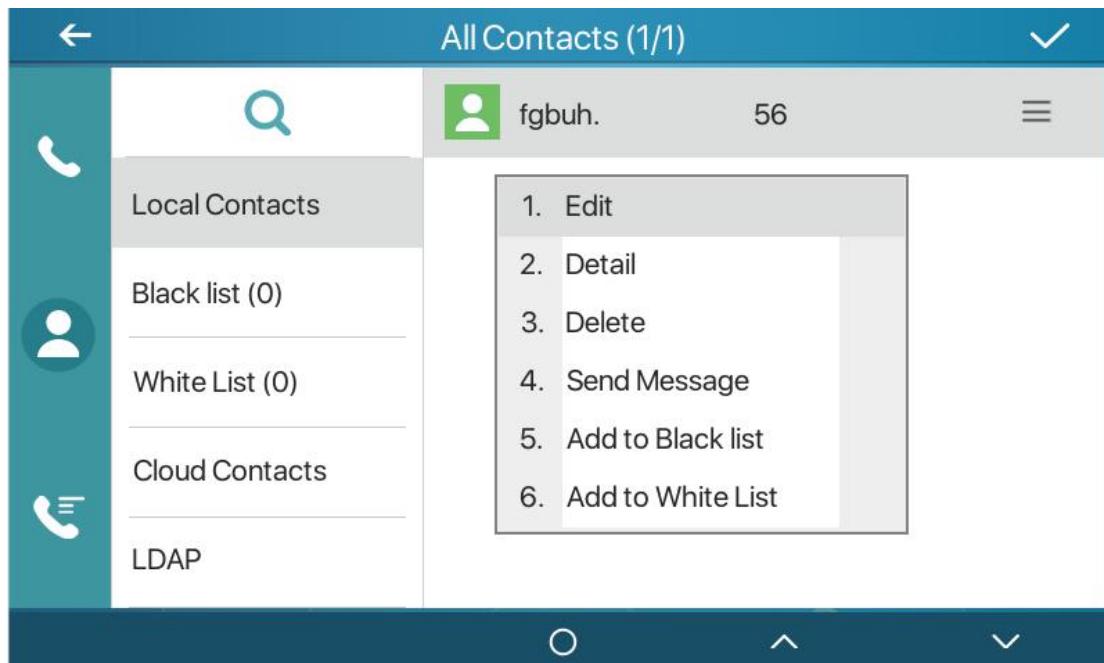
Add a contact, click to enter the contact interface, select the first icon (contact icon, selected by default) and add the following contact information.

- Contact Name
- Tel. Number
- Mobile Number
- Other Number
- Line
- Ring Tone
- Contact Group
- Photo



Picture 47- add contact

User can edit a contact by pressing button .



Picture 48- edit contact

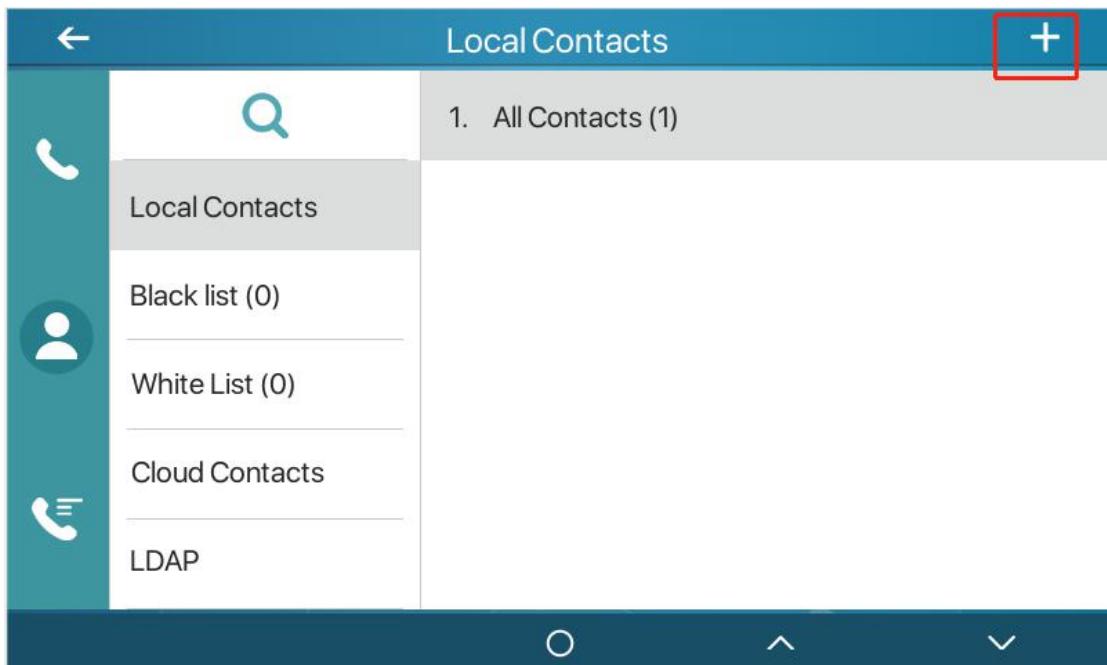
Delete the contact, press to delete the contact, when press it will prompt whether to delete, press to delete.

10.2.1.2 Add / Edit / Delete Group

By default, the group list is empty. Users can create their own group, edit group names, add or remove contacts from the group, and delete groups.

- Add group. Enter contact list interface, press to create groups.
- Delete groups, press to delete
- To edit the group, press to edit.

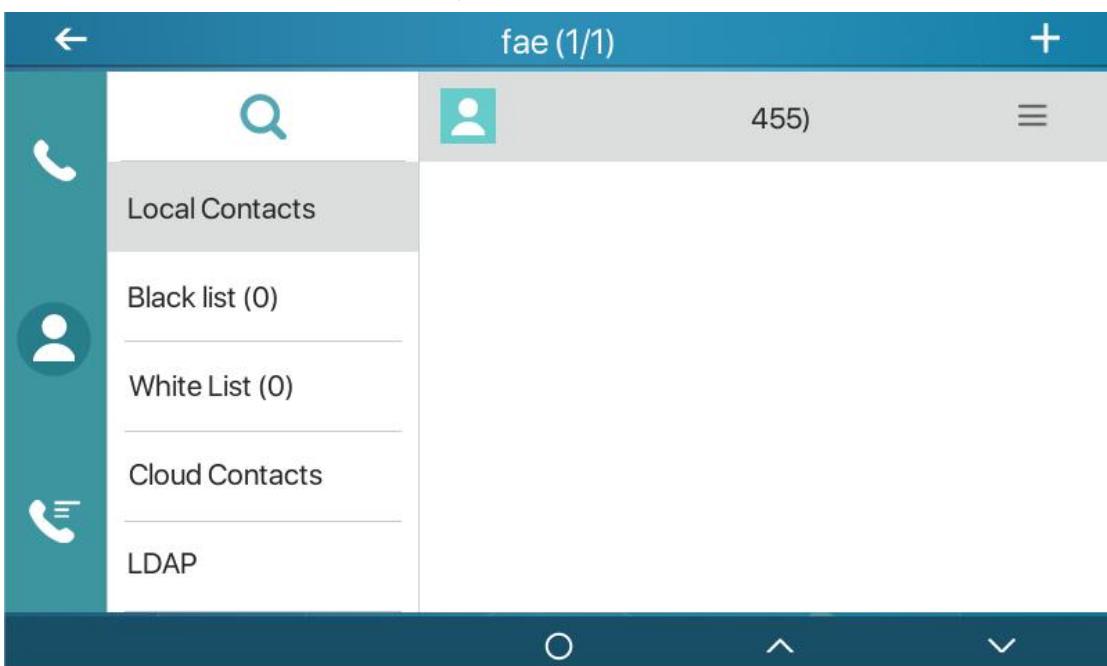
The brackets indicate the total number of records in the group.



Picture 49- group

10.2.1.3 Add / Edit / Delete contact in Group

User can browse the contact in group



Picture 50- browse the contact in group

When the user browses the contacts in the group, he can press to enter the add contact interface, and then press to save the contact, the contact will also be

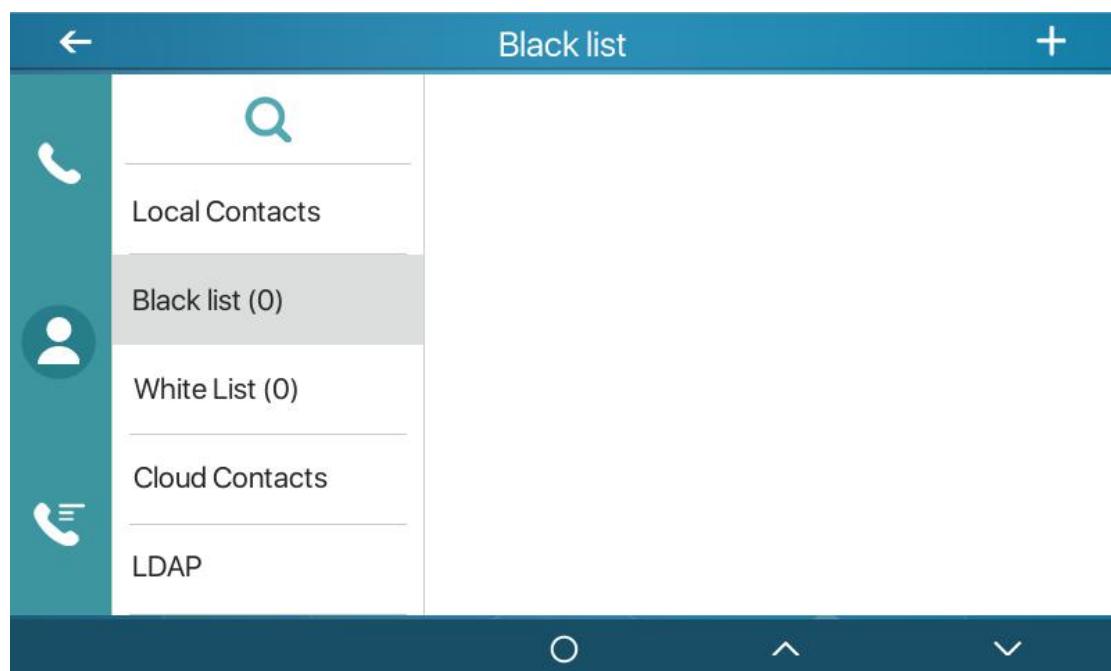
synchronized to the local phone book. You can also delete contacts in the group by press



10.2.2 Black list

The device supports blacklist, such as the number added to the blacklist, the number of calls directly refused to the end, the end of the phone shows no incoming calls. (Blacklisted Numbers can be called out normally)

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



Picture 51- add blacklist

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



Picture 52- Black list

10.2.3 Cloud Phone Book

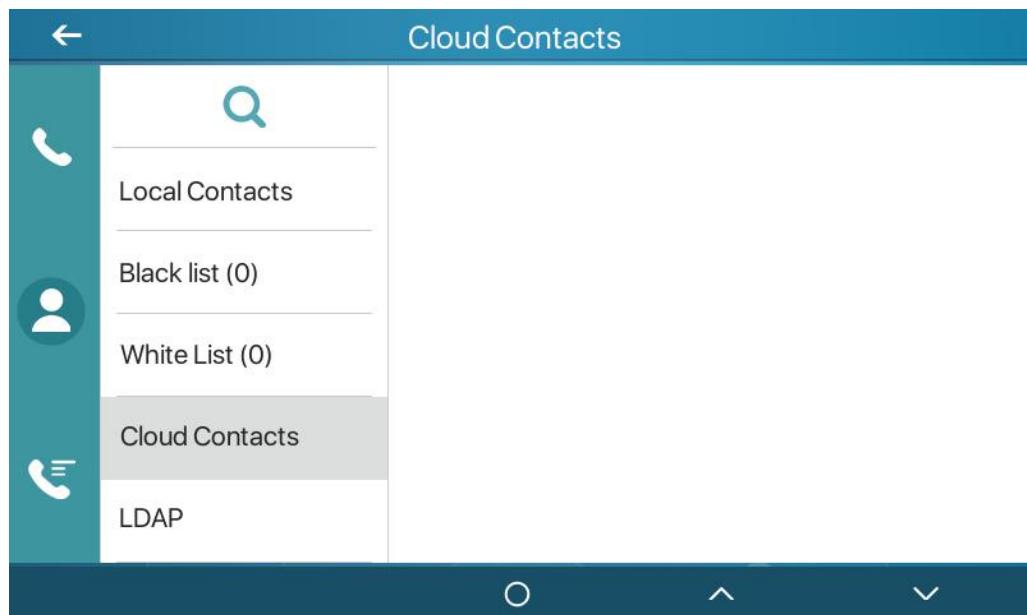
10.2.3.1 Configure Cloud Phone book

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

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

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

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



Picture 53 - cloud contacts

10.2.3.2 Downloading Cloud Phone book

In cloud phone book screen, user can open a cloud phone book by pressing the network phonebook. The device will start downloading the phone book. The user will be prompted with a warning message if the download fails,

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



Picture 54-Browsing Contacts in Cloud Phone book

10.3 Call Log

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

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

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

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

The screenshot shows the call log screen with a sidebar on the left containing icons for phone, person, and handset, and a search bar. The main area is titled 'All' and lists recent calls. Each call entry includes the call type (e.g., 123), date, time, and a more options icon. The sidebar also has buttons for 'All', 'In', 'Out', 'Miss', and 'Forward' to filter the call log.

All	123	10 Sep 16:23	
All	123	10 Sep 11:49	
In	123	10 Sep 11:07	
Out	123	09 Sep 20:57	
Miss	123	09 Sep 20:57	
Forward	123	09 Sep 20:42	

Picture 55- call log

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

- Missed Call Log

- Incoming Call Log

- Outgoing Call Log

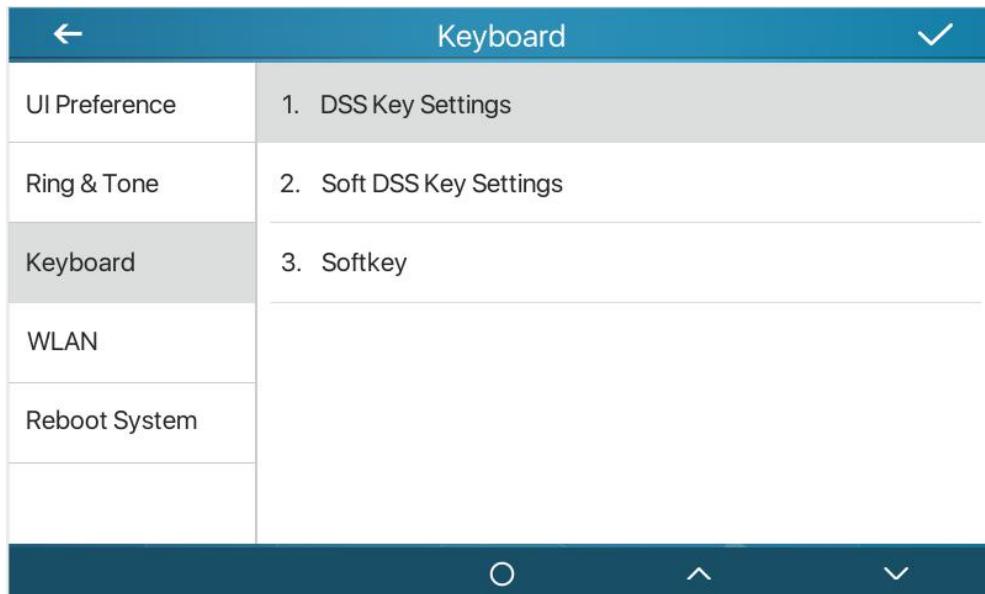
- Forward Call Log

		Out		
	All	123	09 Sep 20:39	
	In	6	09 Sep 20:35	
	Out	69	09 Sep 18:00	
	Miss	69	09 Sep 18:00	
	Forward	69	09 Sep 17:59	
		69	09 Sep 17:58	

Picture 56- Filter call record types

10.4 Function Key

It shows 8 DSSKEY keys in standby mode on Screen, each of which can be customized.



Picture 57- Dss key settings

The DSS Key could be configured as followings,

- ◆ Memory Key
 - Speed Dial/Intercom/BLF/Presence/Call Park/Call Forward (to someone)
- ◆ Line
- ◆ Key Event
 - MWI/DND/Hold/Transfer/Phonebook/Redial/Pickup/Call Forward (to specified line)/Headset/ SMS/Release
- ◆ DTMF
- ◆ Action URL
- ◆ BLF List Key
- ◆ MCAST Paging
- ◆ MCAST Listening
- ◆ Action URL
- ◆ XML Browser

Moreover, user also can add the user-defined title for the DSS Keys, which is configured as Memory Key / Line / URL / MCAST Paging / Prefix.

NOTICE! User-defined title is up to 10 characters.

More detailed information refers to [12.31 Function Key >> Function Key](#) and [6.2 Appendix ii](#)

10.5 Wi-Fi

The device supports wireless Internet access and has built-in Wi-Fi without external devices.

10.5.1 Wireless network

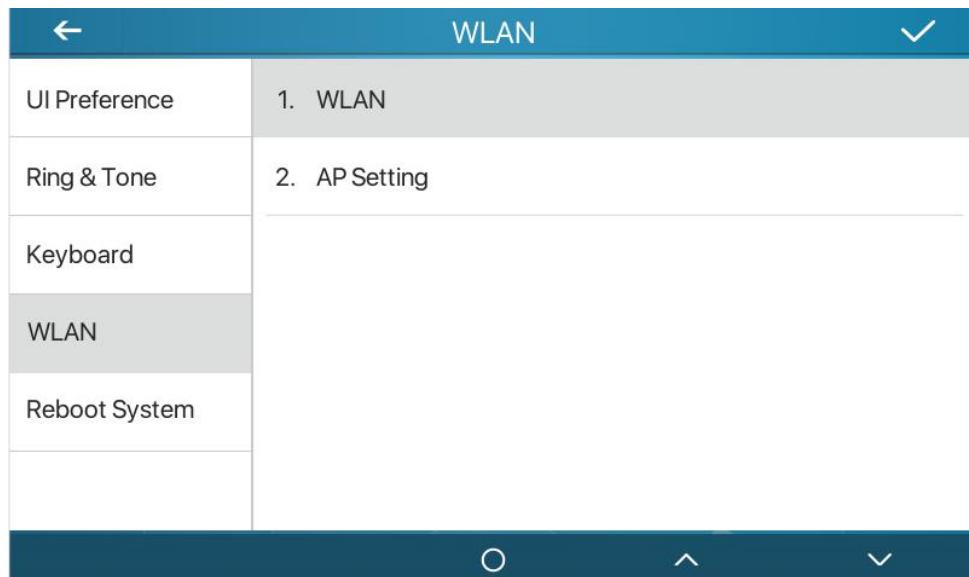
When the device is in the default standby state, search for wireless networks

Press menu [Menu] >> [Basic Settings] .

Click [Basic Settings] >> [WLAN].

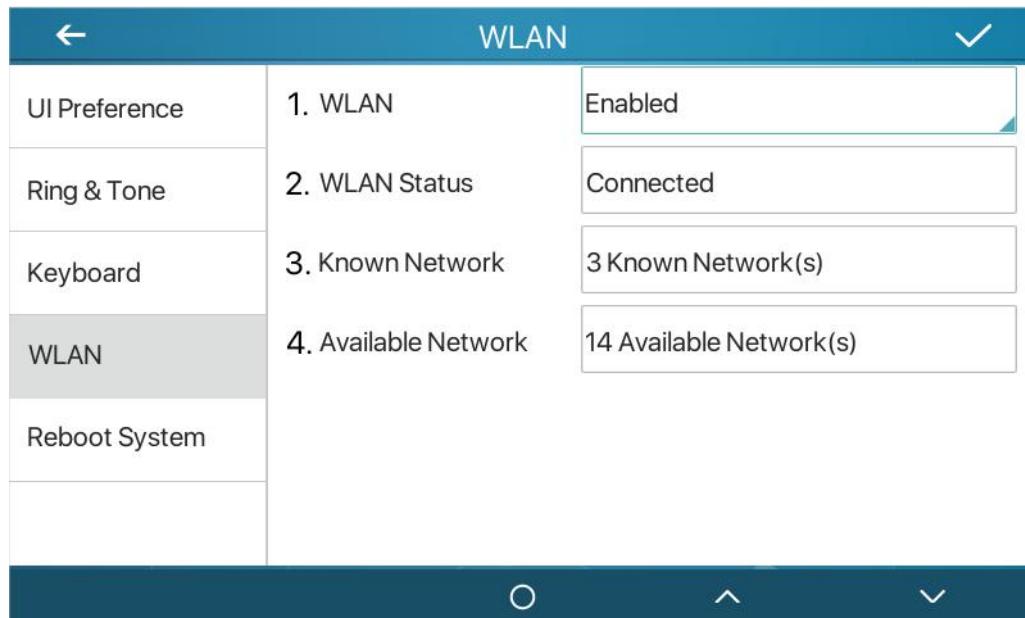
Click [Wireless Network] to enter the setting interface.

Turn on the wireless network, click Save, and the device will automatically search for wireless networks under the current network after enabling.



Picture 58 – WLAN

- The device connects to the wireless network
- Select the available network, select wireless after entering,click , enter username,password to connect
- After connection successful will change to



Picture 59-wireless network

Connection to wireless network

- Log in to the webpage, [Network]>>[Wi-Fi Settings]
- Configure Wi-Fi information, after the configuration is complete, click Add
- Turn on Wi-Fi and click Submit.

	Wi-Fi Name	SSID	Secure Mode	Encryption Type
<input type="checkbox"/>	H3C	H3C_fanvil	WPA/WPA2-SPK	TKIP
<input type="checkbox"/>			None	TKIP
<input type="checkbox"/>	Fanvil	Fanvil	WPA/WPA2-SPK	AES(CMPP)
<input type="checkbox"/>	Fanvil-AP-2.4GHZ	Fanvil-AP-2.4GHZ	WPA/WPA2-SPK	AES(CMPP)

Picture 60- webpage wireless connect

10.5.2 AP setting

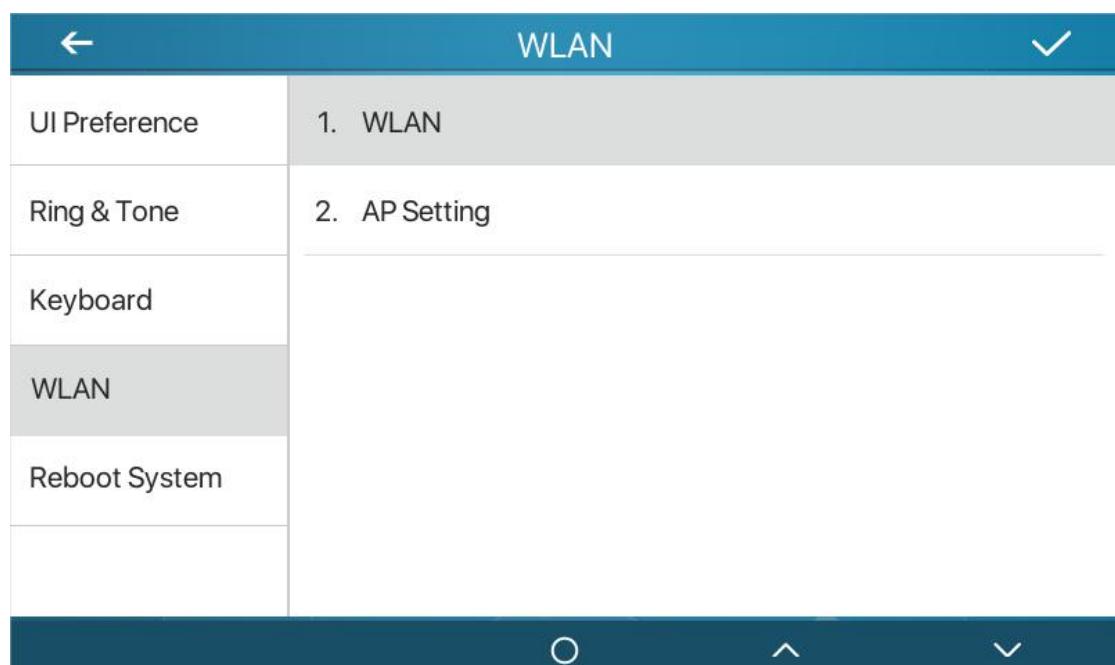
In the absence of a wired network, you can set up a wireless network connection by turning on the AP mode of the device and connecting to the backstage webpage of the device with a mobile phone.

Press menu button [**Menu**] >> [**Basic Settings**] button.

Click [**Basic Settings**] >> [**WLAN**] >> [**AP Settings**]

Enable AP, prompt to restart, it will take effect after restart (cannot be turned on at the same time as the wireless network, if the wireless network is enabled, you need to turn it off)

After restarting, enter the AP setting interface, you can see the SSID and IP address named after the device's MAC address



Picture 61- AP info

Turn on Wi-Fi, you can see the Wi-Fi network named by the device's MAC address, click to connect without a password

After the connection is successful, scan the QR code with the browser of the mobile phone to enter the login interface of the device background

Enter username/password (default admin)

After logging in, select Wi-Fi settings, manually add Wi-Fi and enable Wi-Fi, the device will automatically connect to the Wi-Fi network after the setting is completed

Back to standby, you can see the Wi-Fi icon in the status bar

10.6 Advanced

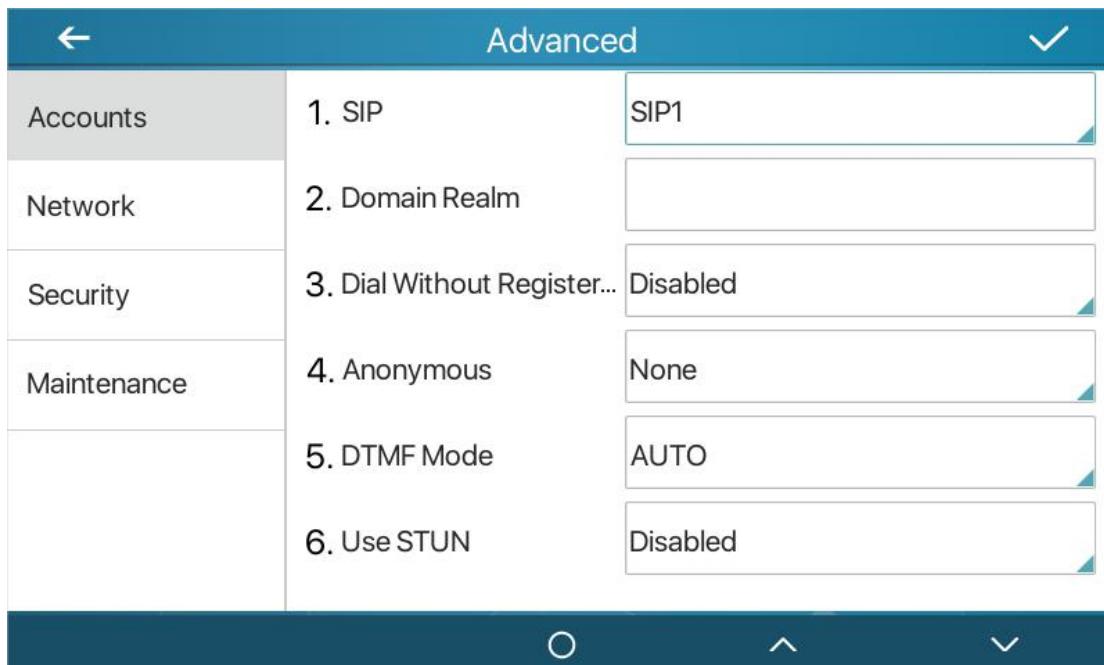
10.6.1 Line Configurations

Phone access [Phone settings] >> [Account] >> [Line], select [Register Account] to configure the SIP line on the phone.

	Basic	
Accounts	1. SIP	SIP1
Network	2. Registration	Enabled
Security	3. Server Address	172.16.1.2
Maintenance	4. Auth. User	
	5. Auth. Password	*****
	6. SIP User	0305

Picture 62-line configurations

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



Picture 63- Configure Advanced Line Options

10.6.1.1 Network Settings

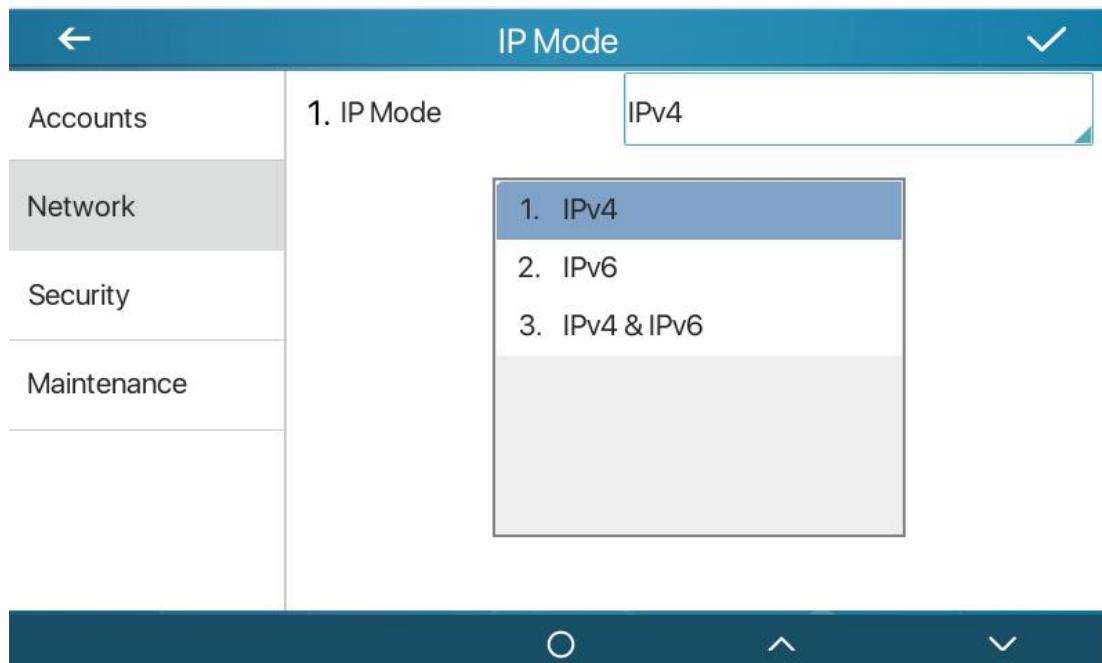
10.6.1.2 Network Settings

Phone access [Phone Settings] >> [Network] >> [Network], you can configure the SIP line on the phone.

■ IP Mode

There are 3 connection mode options: IPv4、IPv6、IPv4&IPv6

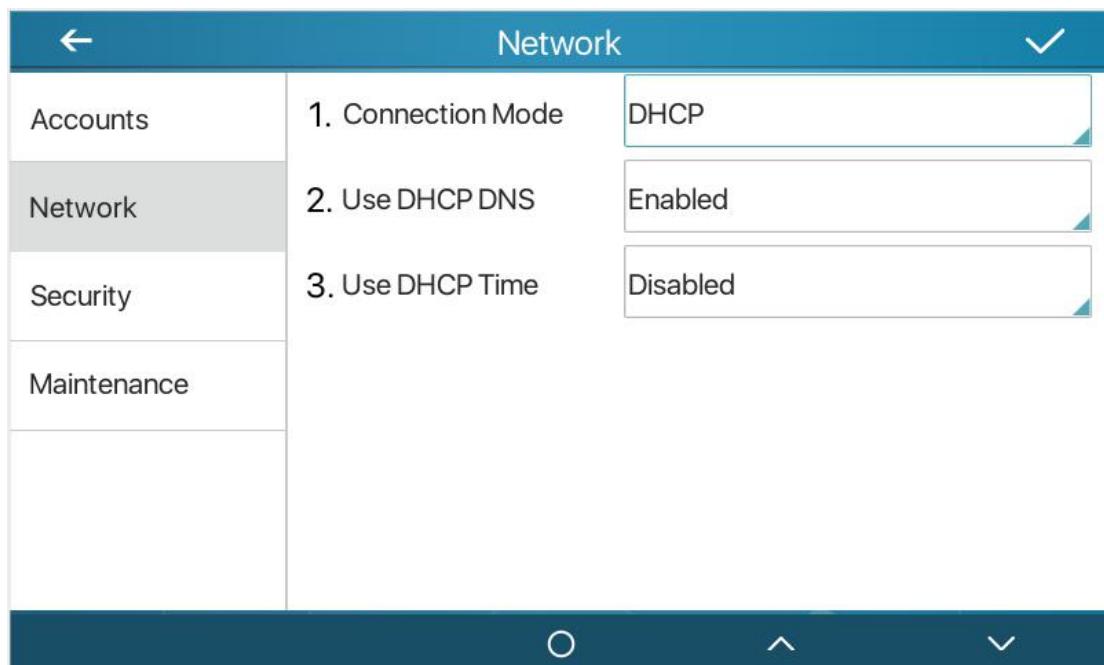
Click to switch IP mode



Picture 64- IP Mode

■ IPv4

The network type has three modes: DHCP, PPPoE, and static IP.



Picture 65- DHCP network mode

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

- Obtain DNS Server automatically: It is enabled as default. “Enable” means phone will get DNS address from DHCP server and “disable” means not.

Step	Setting	Value
1.	Connection Mode	PPPoE
2.	Username	user123
3.	Password	*****

Picture 66 - PPPoE network mode

When the network is set to PPPoE, the PPPoE server issues the network IP address of the device.

User: Fill in the username of the PPPoE server.

Password: Fill in the password of the PPPoE server

Step	Setting	Value
1.	Connection Mode	Static IP
2.	IP Address	192.168.1.179
3.	Mask	255.255.255.0
4.	Gateway	192.168.1.1
5.	Primary DNS	8.8.8.8
6.	Secondary DNS	202.96.134.133

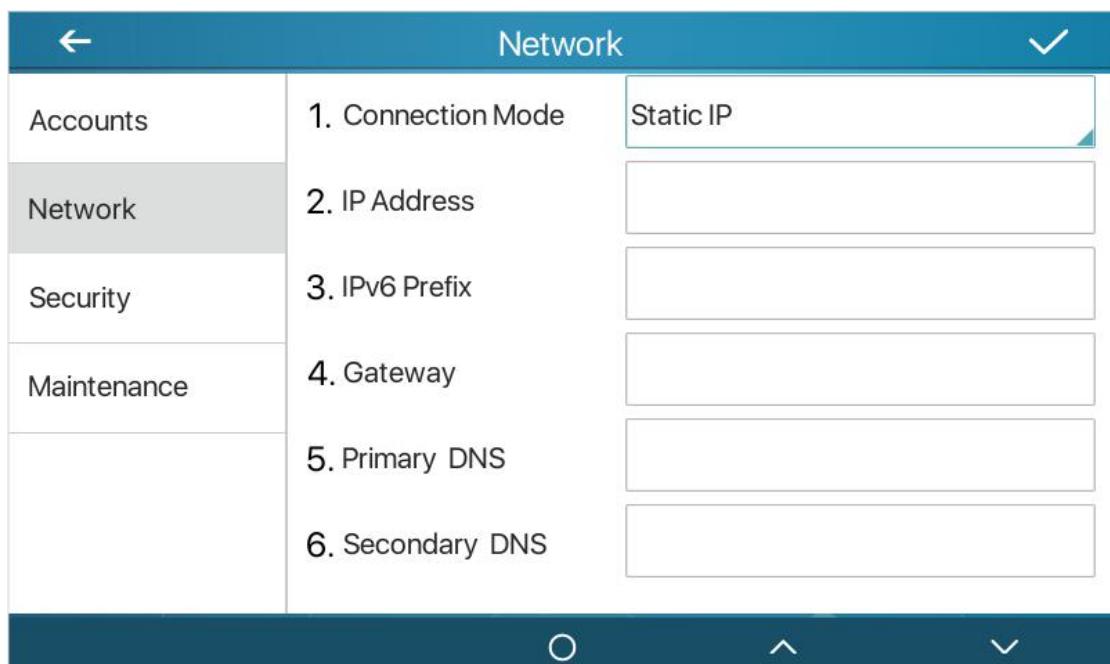
Picture 67- Static IP network mode

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

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

■ IPv6

- The network type has two modes to : DHCP and static IP.
- DHCP network settings are the same as IPv4.
- The static IP network settings are compatible with IPv4, just need to fill in the prefix in IPv6 Prefix.
- IPv6 Prefix: IPv6 prefix digits, the prefix represents the network bit, similar to the IPv4 subnet mask.



Picture 68- IPv6 Static IP network mode

10.6.1.3 QoS & VLAN

■ LLDP

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

segment.

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

■ CDP

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

Table 16- QoS & VLAN

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

Note: *QoS & VLAN details refer to*

<http://www.fanvil.com/Uploads/Tmp/download/20180920/5ba383b56c3ef.pdf>

10.6.1.4 VPN

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

For some users, especially enterprise users, a VPN connection might be required to be established before activate a line registration. The device supports two VPN modes,

Layer 2 Transportation Protocol (L2TP) and OpenVPN.

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

■ L2TP

NOTICE! The device only supports non-encrypted basic authentication and non-encrypted data tunneling. For users who need data encryption, please use OpenVPN instead.

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

When the VPN connection established, the VPN IP Address should be displayed in the VPN status. There may be some delay of the connection establishment. User may need to refresh the page to update the status.

Once the VPN is configured, the device will try to connect to the VPN automatically when the device boots up every time until user disable it. Sometimes, if the VPN connection does not established immediately, user may try to reboot the device and check if VPN connection established after reboot.

■ OpenVPN

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

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

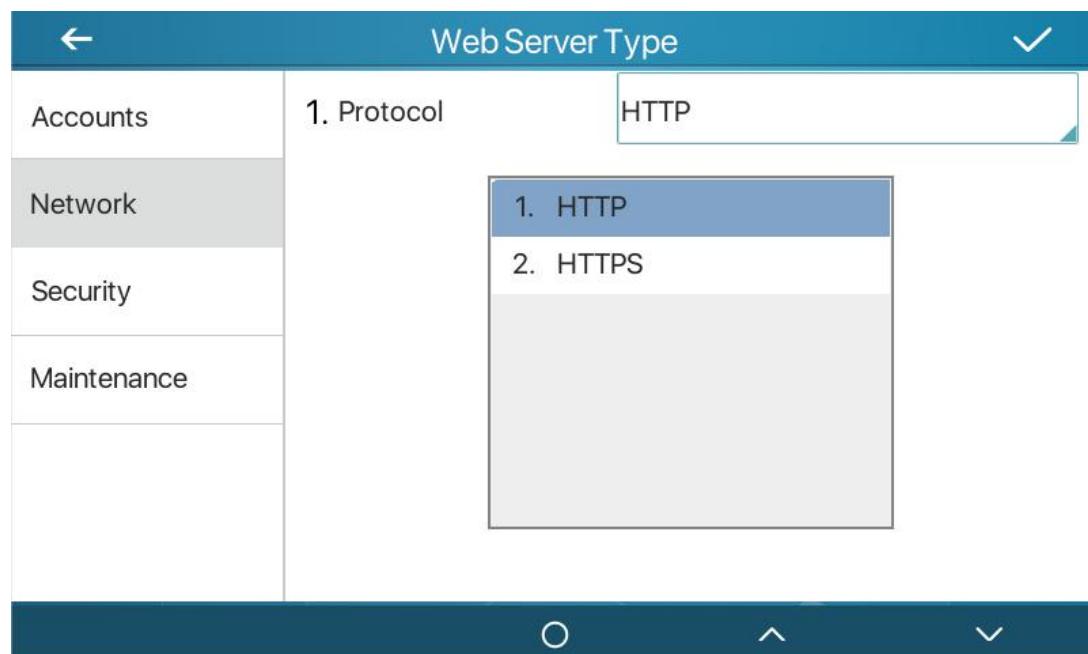
User then upload these files to the device in the web page [Network] -> [VPN], Section OpenVPN Files. Then user should check “Enable VPN” and select “OpenVPN” in VPN Mode and click “Apply” to enable OpenVPN connection.

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

<http://www.fanvil.com/Uploads/Tmp/download/20180920/5ba38303bf0.pdf>

10.6.1.5 Web Server Type

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

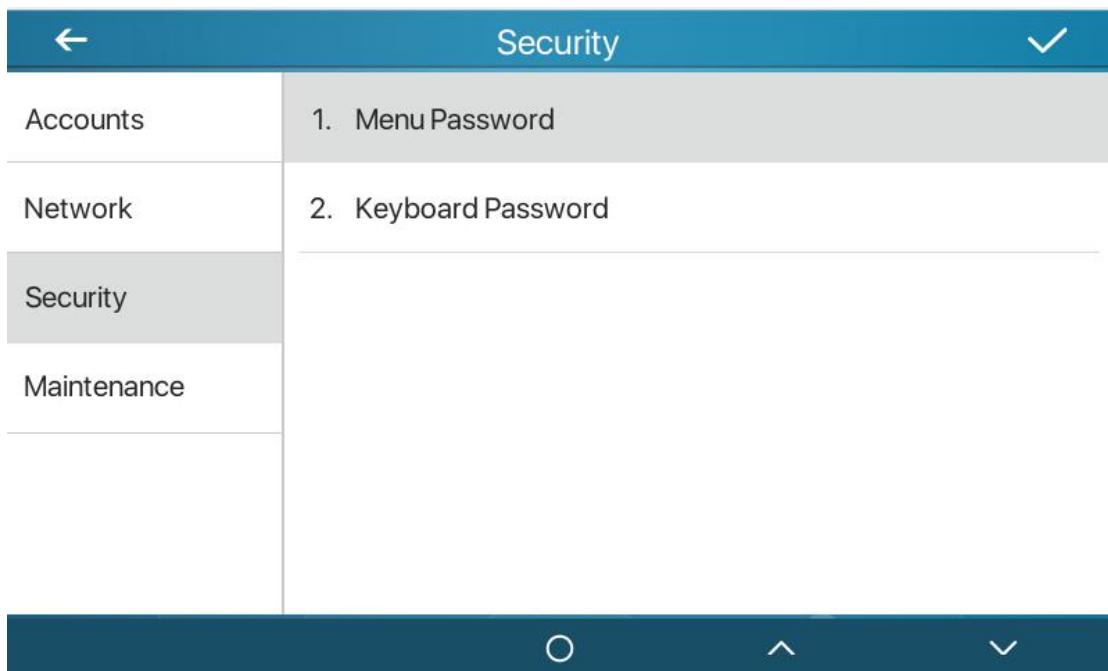


Picture 69 - The phone configures the web server type

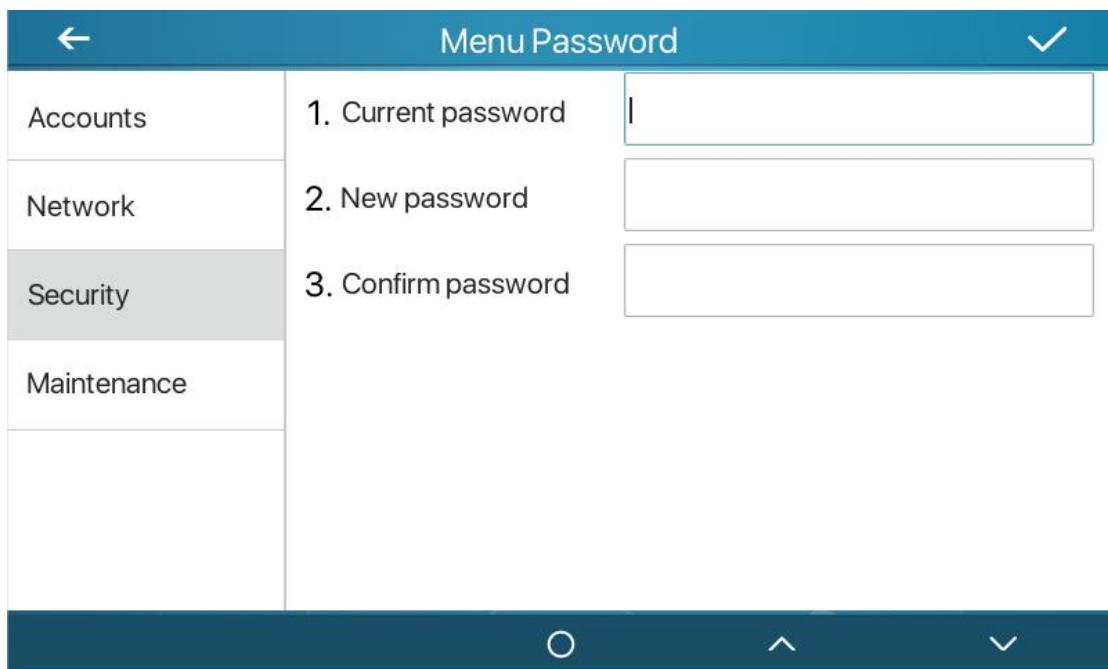
10.6.2 Set The Secret Key

When the device is in the default standby mode,

- Select [Phone Settings]>> [System]>> [Password]
- Click [Password] to change password.



Picture 70- Menu Password



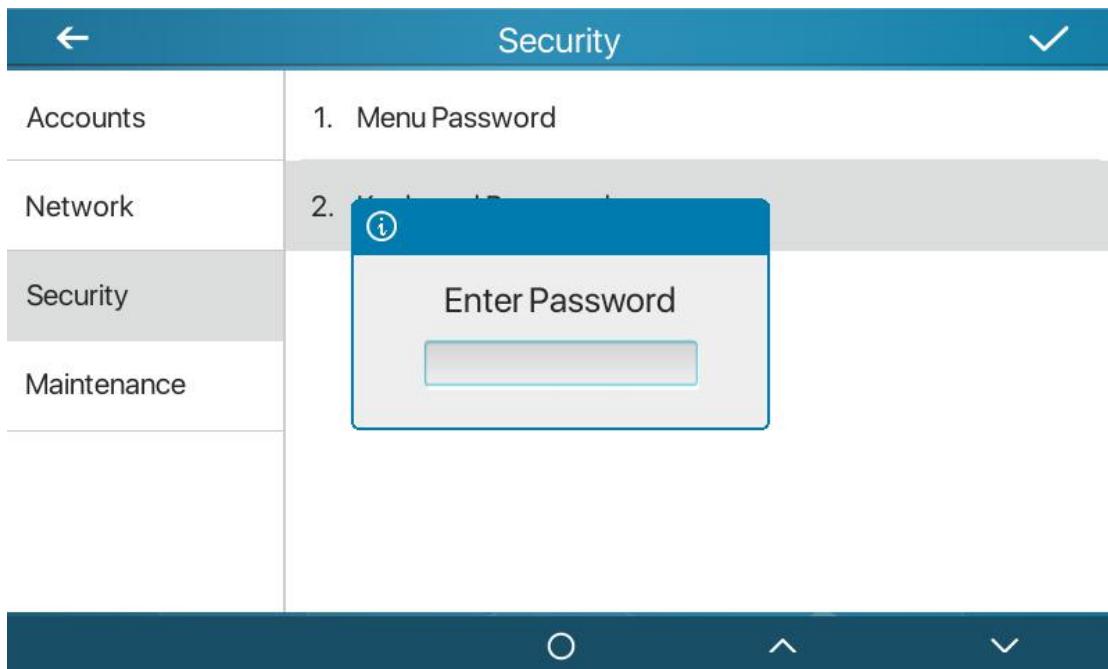
Picture 71 - Menu password setting

The menu password is the advanced setting password.

[Current password] If you not set password, the default password is 123.

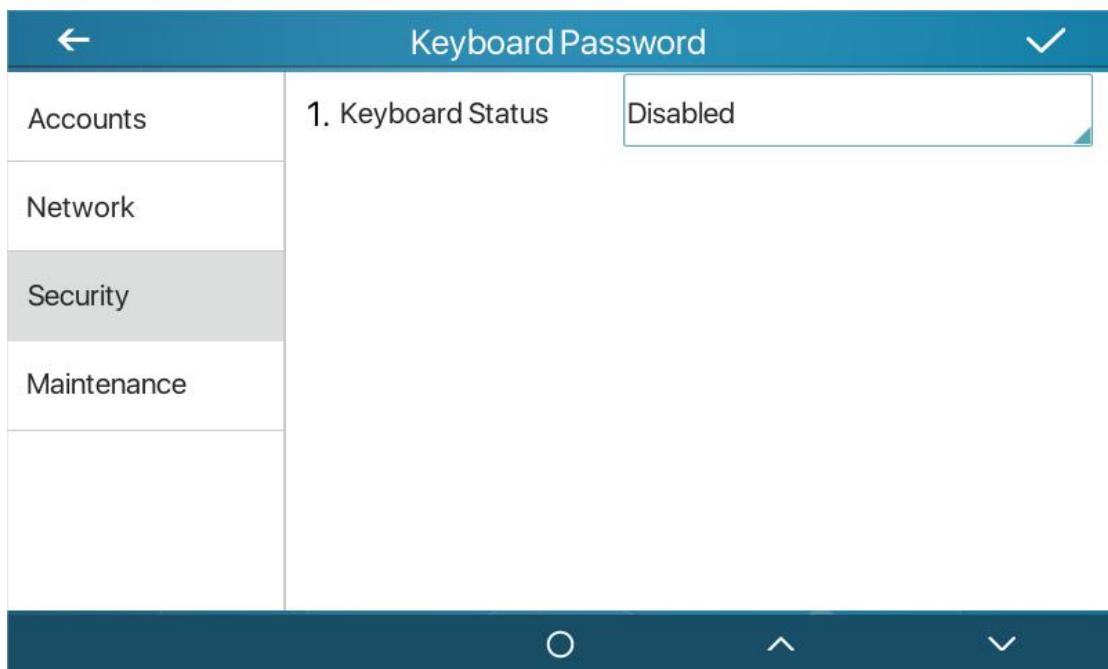
[New password] The password you want to reset.

The password immediately takes effect after the setting is completed, and the password is not displayed in plain text after being entered.



Picture 72- Menu password input

The keyboard password is used to unlock the keyboard after the device is locked.



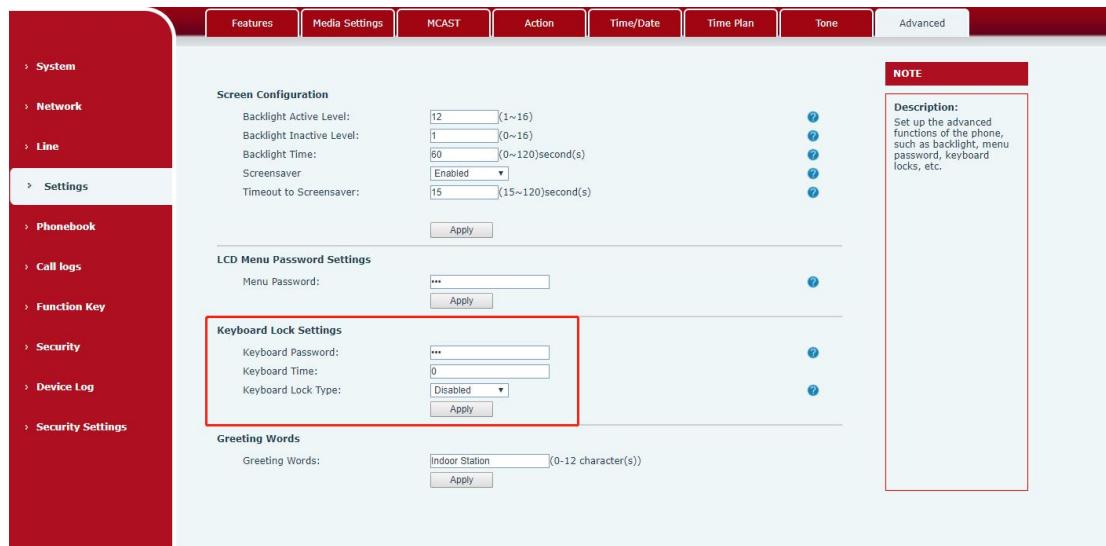
Picture 73- keyboard password setting

The setting of keyboard password only for turn on or turn off on device

Select [Keyboard Password] and click it ,it will pop up a prompt to enter a password, this password is the menu password (the default password is 123).

If the password is correct, you will enter the keyboard lock status interface. The keyboard lock status is off by default, and the timeout period will take effect after you choose to enable it.

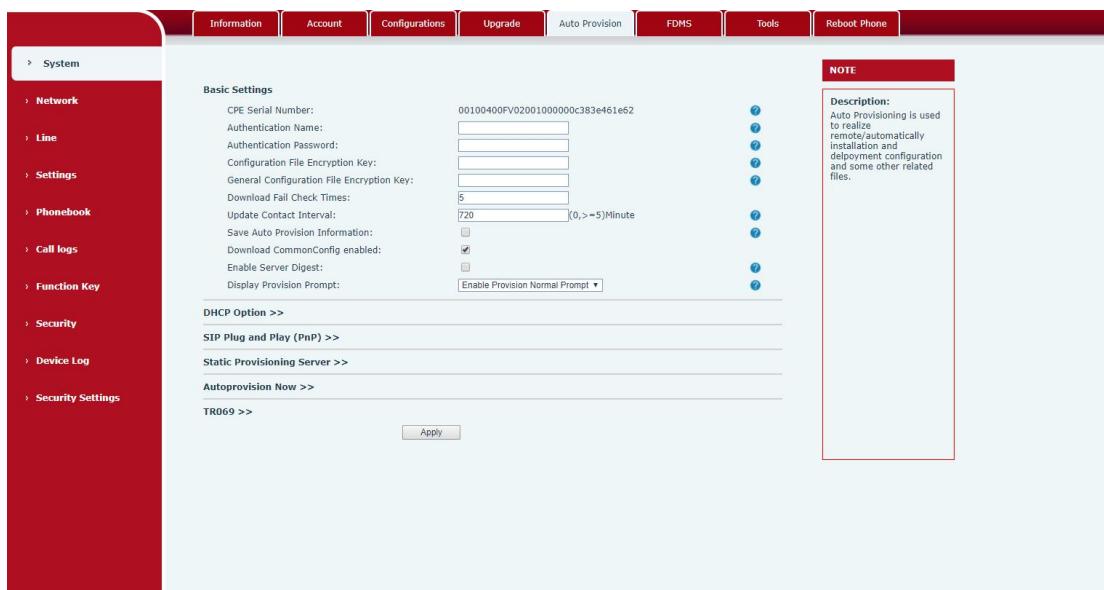
Return to standby,.After the timeout period, the device will lock the keyboard, and there will be a lock icon on the top of the device. Press any key at this time will prompt a password box.



Picture 74- webpage keyboard password setting

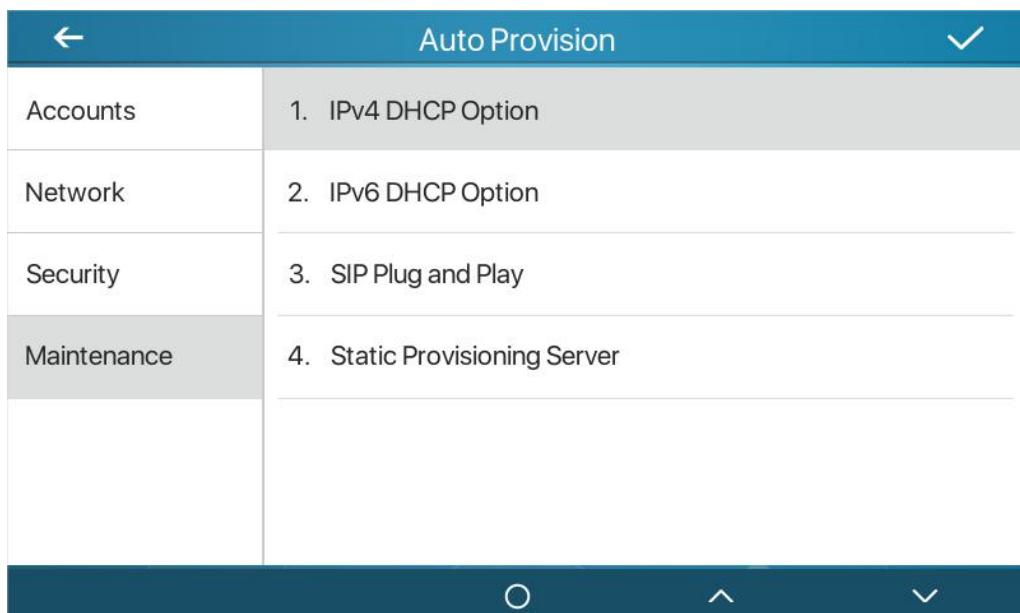
10.6.3 Maintenance

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



Picture 75- Page auto provision Settings

LCD: Enter [Phone Settings] >> [System] >> [Maintain] >> [Auto Provision].



Picture 76- Phone auto provision settings

Fanvil devices support SIP PnP, DHCP options, Static provision, TR069.

Transferring protocol: FTP、TFTP、HTTP、HTTPS

Details refer to **Fanvil Auto Provision** in

<http://www.fanvil.com/Uploads/Temp/download/20180920/5ba3816f8d5f0.pdf>

Table 17 - Auto Provision

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

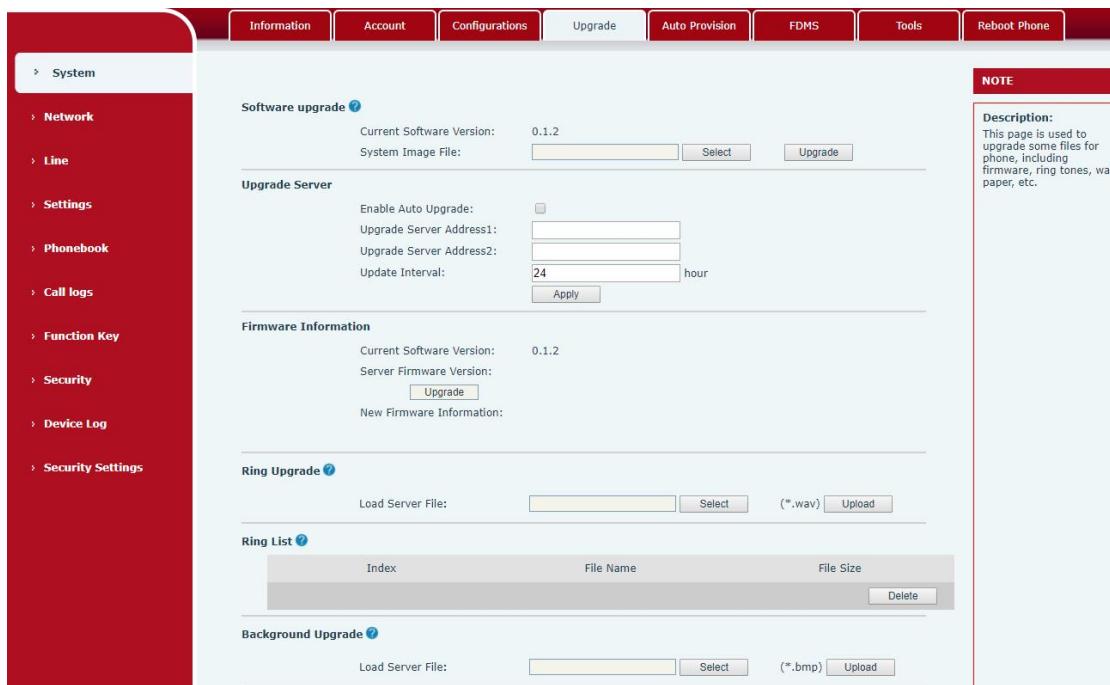
Static Provisioning Server	
Server Address	Provisioning server address. Support both IP address and domain address.
Configuration File Name	The configuration file name. If it is empty, phone will request the common file and device file which is named as its MAC address. The file name could be a common name, \$mac.cfg, \$input.cfg. The file format supports CFG/TXT/XML.
Protocol Type	Transferring protocol type , supports FTP、TFTP、HTTP and HTTPS
Update Interval	Configuration file update interval time. As default it is 1, means phone will check the update every 1 hour.
Update Mode	Provision Mode. 1. Disabled. 2. Update after reboot. 3. Update after interval.

TR069

Enable TR069	Enable TR069 after selection
ACS Server Type	There are 2 options Serve type, common and CTC.
ACS Server URL	ACS server address
ACS User	ACS server username (up to is 59 character)
ACS Password	ACS server password (up to is 59 character)
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.
TLS Version	TLS version (TLS 1.0, TLS 1.1, TLS 1.2)
INFORM Sending Period	INFORM signal interval time. It ranges from 1s to 999s
STUN Server Address	Configure STUN server address
STUN Enable	To enable STUN server for TR069

10.6.4 Firmware Upgrade

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



Picture 77- Web page firmware upgrade

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

Table 18- firmware upgrade

Parameter	Description
Upgrade server	
Enable Auto Upgrade	Enable automatic upgrade, If there is a new version txt and new software firmware on the server, phone will show a prompt upgrade message after Update Interval.
Upgrade Server Address1	Set available upgrade server address.
Upgrade Server Address2	Set available upgrade server address.
Update Interval	Set Update Interval.
Firmware Information	
Current Software Version	It will show Current Software Version.
Server Firmware Version	It will show Server Firmware Version.
[Upgrade] button	If there is a new version txt and new software firmware on the server, the page will display version information and upgrade button will become available; Click [Upgrade] button to upgrade the new firmware.
New version description information	When there is a corresponding TXT file and version on the server side, the TXT and version information will be displayed under the new version description information.

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

名称	修改日期	类型	大小
fanvil_x6_hwv1_0.txt	2018/9/11 17:57	文本文档	1 KB
fanvil_x6_hwv1_1.txt	2018/9/11 17:57	文本文档	1 KB
fanvil_x6_hwv1_2.txt	2018/9/11 17:57	文本文档	1 KB
fanvil_x6_hwv1_3.txt	2018/9/11 17:57	文本文档	1 KB
x6-6904-P0.12.12-1.6.3-2502T2018-0...	2018/8/21 19:52	WinRAR 压缩文...	35,847 KB

- TXT file format must be UTF-8
- vendor_model_hw10.TXT The file format is as follows:
Version=1.6.3 #Firmware
Firmware=xxx/xxx.z #URL, Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers.
BuildTime=2018.09.11 20:00
Info=TXT|XML

Xxxxx

Xxxxx

Xxxxx

Xxxxx

- After the interval of update cycle arrives, if the server has available files and versions, the phone will prompt as shown below. Click [view] to check the version information and upgrade.



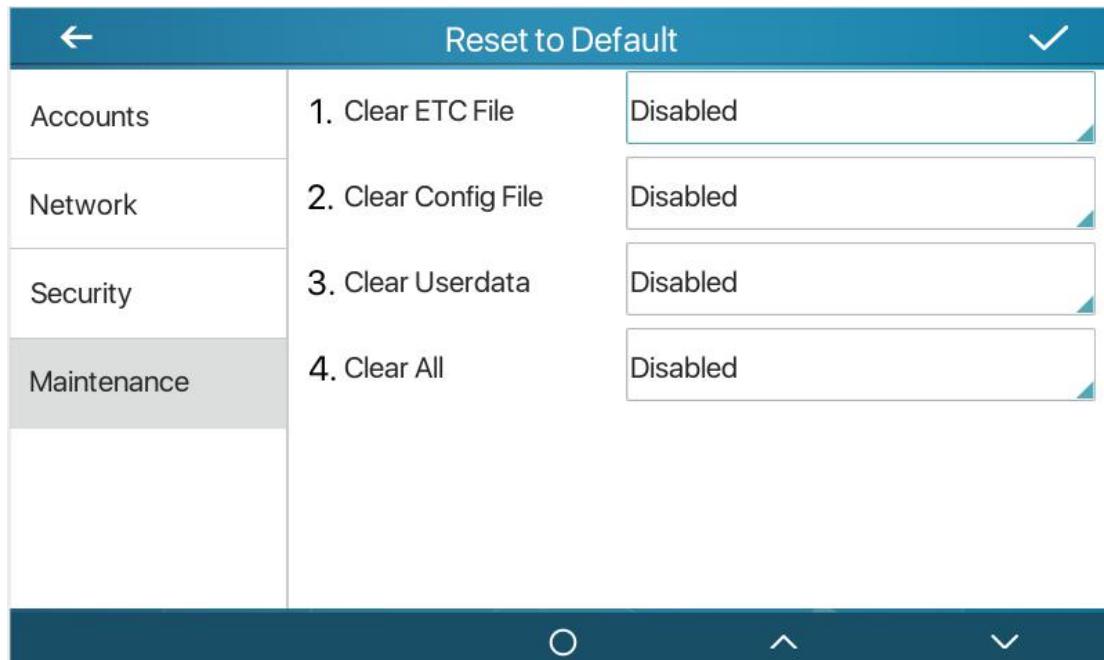
Picture 78- firmware upgrade

10.6.5 Factory Reset

The phone is in default standby mode.

- Press [Phone Settings] to find [System]>> [Maintain]>> [Phone Reset].
- Press the [Reset] button to select the file to be cleared.

Press [OK] to clear after completion. When you select clear configuration file and clear all, the phone will restart automatically after clearing.



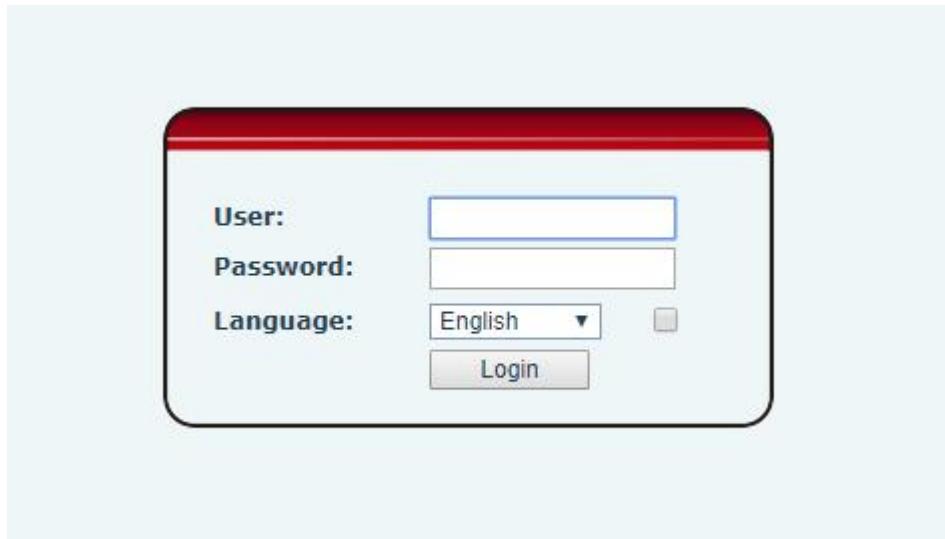
Picture 79 - factory reset

11 Web Configurations

11.1 Web Page Authentication

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

When logging in to the web page with the same or different IP, if the user name/password is entered incorrectly three times, the web page will be locked and you can log in again after 5 minutes.



Picture 80- web login

When the user logs in for the first time, the default user name and password are used. If the password is not changed after login, the web page will prompt "The default password is being used, please change it". After clicking, you can jump to the modify password interface to modify the login password.



Picture 81- default password prompt

11.2 System >> Information

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

- Model
- Hardware Version
- Software Version
- Uptime

And summarization of network status,

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

Besides, summarization of SIP account status,

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

11.3 System >> Account

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

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

11.4 System >> Configurations

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

■ Clear Configurations

Select the module in the configuration file to clear.

SIP: account configuration.

AUTOPROVISION: automatically upgrades the configuration

TR069:TR069 related configuration

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

DSS Key: DSS Key configuration

■ Clear Tables

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

■ Reset Phone

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

11.5 System >> Upgrade

Upgrade the phone software version, customized ringtone, background, DSS Key icon, etc., can also be upgraded to delete the file. Ring tone support “.wav” format.

11.6 System >> Auto Provision

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

<http://www.fanvil.com/Uploads/Temp/download/20180920/5ba3816f8d5f0.pdf>

11.7 System >> Tools

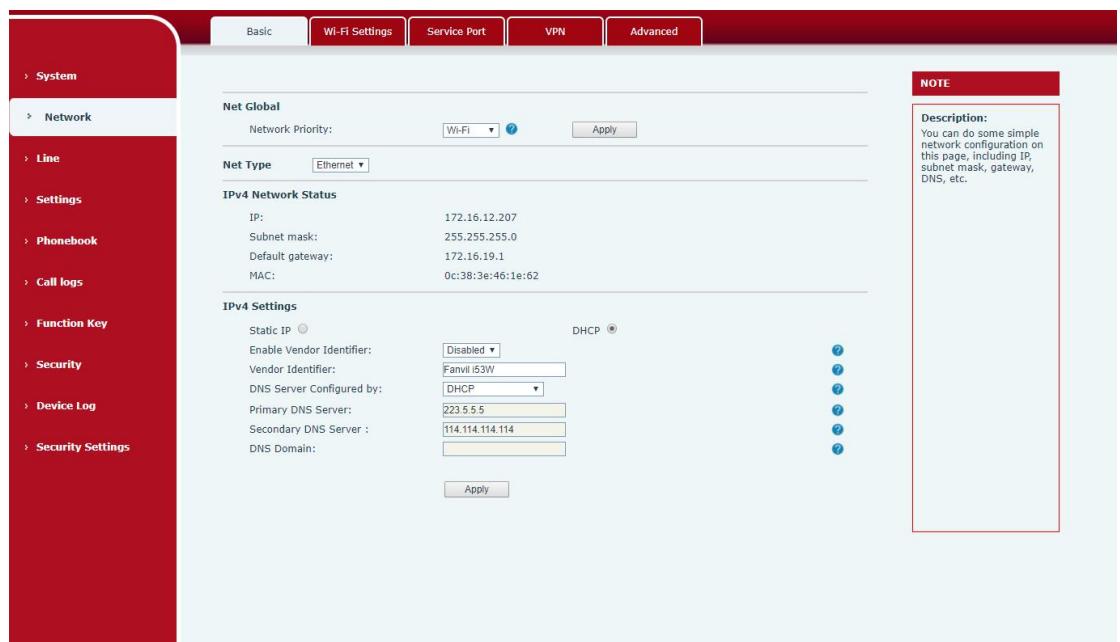
Tools provided in this page help users to identify issues at trouble shooting. Please refer to [13 Trouble Shooting](#) for more detail.

11.8 System >> Reboot Phone

This page can restart the phone.

11.9 Network >> Basic

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



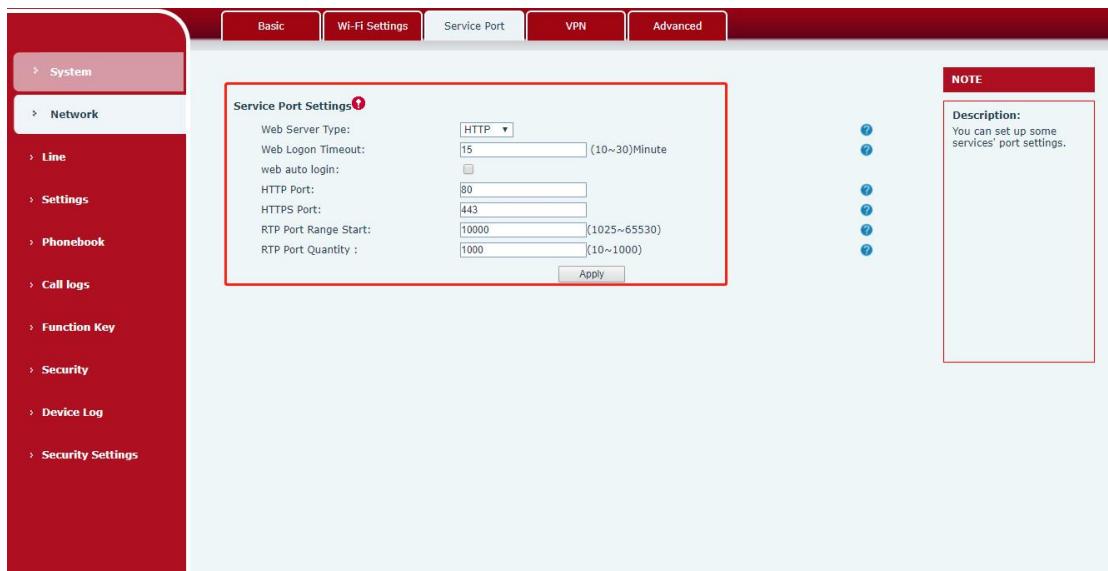
Picture 82- Network settings

Network priority: When wired and wireless are enabled at the same time, you can choose to use wired or wireless first.

Network type: you can view the information of wired/wireless network

11.10 Network >> Service Port

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



Picture 83- Service Port Settings

Table 19 - Service port

Parameter	Description
Web Server Type	Reboot to take effect after settings. Optionally, the web page login is HTTP/HTTPS.
Web Logon Timeout	Default as 15 minutes, the timeout will automatically exit the login page, need to login again.
Web auto login	After the timeout does not need to enter a user name password, will automatically login to the web page.
HTTP Port	The default is 80. If you want system security, you can set ports other than 80. Such as :8080, webpage login: HTTP://ip:8080
HTTPS Port	The default is 443, the same as the HTTP port.
RTP Port Range Start	The value range is 1025 to 65535. The value of RTP port starts from the initial value set. For each call, the value of voice and video port is added 2.
RTP Port Quantity	Number of calls.

11.11 Network >> VPN

Users can configure VPN connections on this page. Please refer to [10.6.2.3 VPN](#)

and get more details.

11.12 Line >> SIP

Configure the Line service configuration on this page.

Table 20- Line configuration on the web page

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

Proxy Server Port	Enter the SIP proxy server port, default is 5060.
Proxy User	Enter the SIP proxy user.
Proxy Password	Enter the SIP proxy password.
Backup Proxy Server Address	Enter the IP or FQDN address of the backup proxy server.
Backup Proxy Server Port	Enter the backup proxy server port, default is 5060.
Basic Settings	
Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
Call Forward Unconditional	Enable unconditional call forward, all incoming calls will be forwarded to the number specified in the next field
Call Forward Number for Unconditional	Set the number of unconditional call forward
Call Forward on Busy	Enable call forward on busy, when the phone is busy, any incoming call will be forwarded to the number specified in the next field.
Call Forward Number for Busy	Set the number of call forward on busy .
Call Forward on No Answer	Enable call forward on no answer, when an incoming call is not answered within the configured delay time, the call will be forwarded to the number specified in the next field.
Call Forward Number for No Answer	Set the number of call forward on no answer.
Call Forward Delay for No Answer	Set the delay time of not answered call before being forwarded.
Transfer Timeout	Set the timeout of call transfer process.
Conference Type	Set the type of call conference, Local=set up call conference by the device itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server
Server Conference Number	Set the conference room number when conference type is set to be Server
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is

	voice message waiting on the server
Voice Message Number	Set the number for retrieving voice message
Voice Message Subscribe Period	Set the interval of voice message notification subscription
Enable Hotline	Enable hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or turn on hands-free speaker or headphone
Hotline Delay	Set the delay for hotline before the system automatically dialed it
Hotline Number	Set the hotline dialing number
Dial Without Registered	Set call out by proxy without registration
Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.
DTMF Type	Set the DTMF type to be used for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Use VPN	Set the line to use VPN restrict route
Use STUN	Set the line to use STUN for NAT traversal
Enable Failback	Whether to switch to the primary server when it is available.
Failback Interval	A Register message is used to periodically detect the time interval for the availability of the main Proxy.
Signal Failback	Multiple proxy cases, whether to allow the invite/register request to also execute failback.
Signal Retry Counts	The number of attempts that the SIP Request considers proxy unavailable under multiple proxy scenarios.
Codecs Settings	Set the priority and availability of the codecs by adding or remove them from the list.
Video Codecs	Select video code to preview video.

Advanced Settings	
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field.
Enable DND	Set the feature code to dial to the server
Disable DND	Set the feature code to dial to the server
Enable Call Forward Unconditional	Set the feature code to dial to the server
Disable Call Forward Unconditional	Set the feature code to dial to the server
Enable Call Forward on Busy	Set the feature code to dial to the server
Disable Call Forward on Busy	Set the feature code to dial to the server
Enable Call Forward on No Answer	Set the feature code to dial to the server
Disable Call Forward on No Answer	Set the feature code to dial to the server
Enable Blocking Anonymous Call	Set the feature code to dial to the server
Disable Blocking Anonymous Call	Set the feature code to dial to the server
Call Waiting On Code	Set the feature code to dial to the server
Call Waiting Off Code	Set the feature code to dial to the server
Send Anonymous On Code	Set the feature code to dial to the server
Send Anonymous Off Code	Set the feature code to dial to the server
SIP Encryption	Enable SIP encryption such that SIP transmission will be encrypted
RTP Encryption	Enable RTP encryption such that RTP transmission will be encrypted
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if there is not new session timer event update received after the timeout period
Session Timeout	Set the session timer timeout period
Enable BLF List	Enable/Disable BLF List
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Response Single Codec	If setting enabled, the device will use single codec in response to an incoming call request
BLF Server	The registered server will receive the subscription package from ordinary application

	of BLF phone. Please enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened
Keep Alive Interval	Set the keep alive packet transmitting interval
Keep Authentication	Keep the authentication parameters from previous authentication
Blocking Anonymous Call	Reject any incoming call without presenting caller ID
User Agent	Set the user agent, the default is Model with Software Version.
Specific Server Type	Set the line to collaborate with specific server type
SIP Version	Set the SIP version
Anonymous Call Standard	Set the standard to be used for anonymous
Local Port	Set the local port
Ring Type	Set the ring tone type for the line
Enable user=phone	Sets user=phone in SIP messages.
Use Tel Call	Set use tel call
Auto TCP	Using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes
Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
DNS Mode	Select DNS mode, A, SRV, NAPTR
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server, it will use the source IP address, not the address in via field.
Convert URI	Convert not digit and alphabet characters to %hh hex code
Use Quote in Display Name	Whether to add quote in display name, i.e. "Fanvil" vs Fanvil
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)

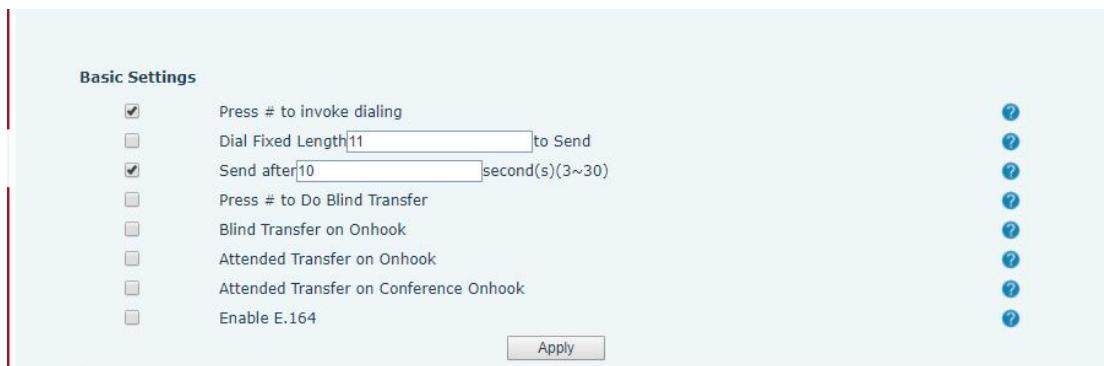
Sync Clock Time	Time Sync with server
Enable Inactive Hold	With the post-call hold capture package enabled, you can see that in the INVITE package, SDP is inactive.
Caller ID Header	Set the Caller ID Header
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Enable Feature Sync	Feature Sync with server
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
CallPark Number	Set the CallPark number.
Server Expire	Set the timeout to use the server.
TLS Version	Choose TLS Version.
uaCSTA Number	Set uaCSTA Number.
Enable Click To Talk	With the use of special server, click to call out directly after enabling.
Enable Chgport	Whether port updates are enabled.
VQ Name	Open the VQ name for VQ RTCP-XR.
VQ Server	Open VQ server address for VQ RTCP-XR.
VQ Port	Open VQ port for VQ RTCP-XR.
VQ HTTP/HTTPS Server	Enable VQ server selection for VQ RTCP-XR.
Flash mode	Chose Flash mode, normal or SIP info.
Flash Info Content-Type	Set the SIP info content type.
Flash Info Content-Body	Set the SIP info content body.
PickUp Number	Set the scramble number when the Pickup is enabled.
JoinCall Number	Set JoinCall Number.
Intercom Number	Set Intercom Number.
Unregister On Boot	Whether to enable logout function.
Enable MAC Header	Whether to open the registration of SIP package with user agent with MAC or not.
Enable Register MAC Header	Whether to open the registration is user agent with MAC or not.
BLF Dialog Strict Match	Whether to enable accurate matching of BLF sessions.
PTime(ms)	Set whether to bring ptime field, default no.
SIP Global Settings	
Strict Branch	Set up to strictly match the Branch field.

Enable Group	Set open group.
Enable RFC4475	Set to enable RFC4475.
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Set the registration failure retry time.
Local SIP Port	Modify the phone SIP port.
Enable uaCSTA	Set to enable the uaCSTA function.

11.13 Line >> SIP Hotspot

Please refer to [9.4 SIP Hotspot](#).

11.14 Line >> Dial Plan



Picture 84- Dial plan settings

Table 21- Phone 7 dialing methods

Parameters	Description
Press # to invoke dialing	The user dials the other party's number and then adds the # number to dial out;
Dial Fixed Length	The number entered by the user is automatically dialed out when it reaches a fixed length
Timeout dial	The system dials automatically after timeout
Press # to Do Blind Transfer	The user enters the number to be transferred and then presses the "#" key to transfer the current call to a third party
Blind Transfer on Onhook	After the user enters the number, hang up the handle or turn off the hands-free function to transfer the current call to a third party.

Attended Transfer on Onhook	Hang up the handle or press the hands-free button to realize the function of attention-transfer, which can transfer the current call to a third party.
Attended Transfer on Conference Onhook	During a three-way call, hang up the handle and the remaining two parties remain on the call.
Enable E.164	Please refer to e. 164 standard specification

Add dialing rules:

Picture 85- Custom setting of dial - up rules

Table 22 - Dial - up rule configuration table

Parameters	Description
Dial rule	<p>There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules.</p> <p>In prefix matching, only part of the number is entered followed by T. The mapping will then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.</p>
<p>Note: Two different special characters are used.</p> <ul style="list-style-type: none"> ■ x -- Matches any single digit that is dialed. ■ [] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated 	

by commas, or a list of digits.	
Destination	Set Destination address. This is for IP direct.
Port	Set the Signal port, and the default is 5060 for SIP.
Alias	Set the Alias. This is the text to be added, replaced or deleted. It is an optional item.
<p>Note: There are four types of aliases.</p> <ul style="list-style-type: none"> ■ all: xxx – xxx will replace the phone number. ■ add: xxx – xxx will be dialed before any phone number. ■ del –The characters will be deleted from the phone number. ■ rep: xxx – xxx will be substituted for the specified characters. 	
Suffix	Characters to be added at the end of the phone number. It is an optional item.
Length	Set the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. It is an optional item.

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

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

User-defined Dial Plan Table						
Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix
1	"123"	Out	No	SIP DIALPEER(172.16.1.15:5560)		Default

Picture 86- Dial rules table (1)

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

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

Picture 87- Dial rules table (2)

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

x -- Matches any single digit that is dialed.

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

11.15 Line >> Action Plan

When calling to a phone, the bounded IP camera synchronously transmits video to the opposite phone (video support).

Table 23- action plan

Parameter	Description
Number	Auxiliary phone number (support video)
Type	Support video display on call.
Direction	For call mode, incoming/outgoing call displays video
Line	Set up outgoing lines.
Username	Bind the user name of the IP camera.
Password	Bind IP camera password.
URL	Video streaming information.
User Agent	Set user agent information

11.16 Line >> Basic Settings

Set up the register global configuration.

Table 24- Set the line global configuration on the web page

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

11.17 Settings >> Features

Configuration phone features.

Table 25- General function Settings

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
Enable Call Transfer	Enable Call Transfer.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it
Enable 3-Way Conference	Enable 3-way conference by selecting it
Enable Auto Onhook	The phone will hang up and return to the idle automatically at hands-free mode
Auto Onhook Time	Specify Auto Onhook time, the phone will hang up and return to the idle automatically after Auto Hand down time at hands-free mode, and play dial tone Auto Onhook time at handset mode
Ring for Headset	Enable Ring for Handset by selecting it, the phone plays ring tone from handset.
Auto Headset	Enable this feature, headset plugged in the phone, user press 'answer' key or line key to answer a call with the headset automatically.

Enable Silent Mode	When enabled, the phone is muted, there is no ringing when calls, you can use the volume keys and mute key to unmute.
Disable Mute for Ring	When it is enabled, you can't mute the phone
Enable Default Line	If enabled, user can assign default SIP line for dialing out rather than SIP1.
Enable Auto Switch Line	Enable phone to select an available SIP line as default automatically
Default Ext Line	Select the default line to use for outgoing calls
Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out any number.
Hide DTMF	Configure the hide DTMF mode.
Enable CallLog	Select whether to save the call log.
Enable Restricted Incoming List	Whether to enable restricted call list.
Enable Allowed Incoming List	Whether to enable the allowed call list.
Enable Restricted Outgoing List	Whether to enable the restricted allocation list.
Enable Country Code	Whether the country code is enabled.
Country Code	Fill in the country code.
Area Code	Fill in the area code.
Enable Number Privacy	Whether to enable number privacy.
Match Direction	Matching direction, there are two kinds of rules from right to left and from left to right.
Start Position	Open number privacy after the start of the hidden location.
Hide Digits	Turn on number privacy to hide the number of digits.
Allow IP Call	If enabled, user can dial out with IP address
P2P IP Prefix	Prefix a point-to-point IP call.
Caller Name Priority	Change caller ID display priority.
Emergency Call Number	
Search path	Select the search path.
LDAP Search	Select from with one LDAP for search
Emergency Call Number	Configure the Emergency Call Number. Despite the keyboard is locked, you can dial the emergency call number
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP address. More details please refer to this link

Push XML Server	Configure the Push XML Server, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.
Enable Pre-Dial	Disable this feature, user enter number will open audio channel automatically. Enable the feature, user enter the number without opening audio channel.
Enable Multi Line	If enabled, up to 10 simultaneous calls can exist on the phone, and if disabled, up to 2 simultaneous calls can exist on the phone.
Line Display Format	Custom line format: SIPn/SIPn: xxx/xxx@SIPn
Contact As White List Type	NONE/BOTH/DND White List/FWD White List
Block XML When Call	Disable XML push on call.
SIP notify	When enabled, the phone displays the information when it receives the relevant notify content.
Tone Settings	
Enable Holding Tone	When turned on, a tone plays when the call is held
Enable Call Waiting Tone	When turned on, a tone plays when call waiting
Play Dialing DTMF Tone	Play DTMF tone on the device when user pressed a phone digits at dialing, default enabled.
Play Talking DTMF Tone	Play DTMF tone on the device when user pressed a phone digits during taking, default enabled.
DND Settings	
DND Option	Select to take effect on the line or on the phone or close.
Enable DND Timer	Enable DND Timer, If enabled, the DND is automatically turned on from the start time to the off time.
DND Start Time	Set DND Start Time
DND End Time	Set DND End Time
Intercom Settings	
Enable Intercom	When intercom is enabled, the device will accept

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

	off, on is always red bright, the default is off.
Ringing	Power lamp status when there is an incoming call, including off/on/slow flash/quick flash, default flash.
Mute	Power lamp status in mute mode, including off/on/slow flash/quick flash, off by default.
Hold/Held	The power lamp state, including off/on/slow flash/quick flash, is turned off by default when left/retained.
Notification Popups	
Display Missed Call Popup	No incoming call popup prompt after opening, no popup prompt when closing, open by default.
Display MWI Popup	Voice message popup prompt is not answered after opening, and it is opened by default if there is no popup prompt when closing.
Display Device Connect Popup	There is a popup prompt when the WIFI adapter is connected. There is no popup prompt when the WIFI adapter is closed. It is on by default.
Display SMS Popup	There is popup prompt for unread messages after opening, and there is no popup prompt when closing. It is opened by default.
Display Other Popup	When the handle is not hung back after opening, registration fails, IP acquisition fails, Tr069 connection fails and other abnormalities, there will be popup prompt when it is opened; otherwise, there will be no prompt when it is closed, and it will be opened by default.

Figure 1

11.18 Settings >> Media Settings

Change voice Settings.

Table 26 - Voice settings

Parameter	Description
Codecs Settings	Select enable or disable voice encoding: G.711A/U,G.722,G.729,

	G.726-16,G726-24,G726-32,G.726-40, ILBC,opus
Video codec	
Video codec	Select to enable video encoding:H264
Media Setting	
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
Headset Mic Gain	Set the earphone's radio volume gain to fit different models of earphones.
Opus payload type	Set Opus load type, range 96~127.
OPUS Sample Rate	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb (16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length
Onhook Time	Configure a minimum response time, which defaults to 200ms
Enable the patting spring to generate Flash	Whether to turn on the plug spring to generate Flash
Video bit rate	Set the bit rate of video:64kbps , 192kbps , 256kbps, 384kbps, 512kbps, 768kbps, 1Mbps, 1.6Mbps, 2Mbps, 3Mbps, 4Mbps
Video frame rate	Set the video frame rate: 5fps, 10fps, 15fps, 20fps, 25fps, 30fps
Video resolution	Set Video resolution: CIF,VGA,4CIF,720P
H.264Payload Type	Set the H264 Payload Type , the value must be 96~127.
Display splicing frame	Whether to start displaying splicing frames
RTP Control Protocol(RTCP) Settings	
CNAME user	Set CNAME user
CNAME host	Set CNAME host
RTP Settings	
RTP keep alive	Hold the call and send the packet after 30s
Alert Info Ring Settings	
Value	Set the value to specify the ring type.
Ring Type	Type1-Type9

11.19 Settings >> MCAST

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

Table 27 - Multicast parameters

Parameters	Description
Normal Call Priority	Define the priority of the active call, 1 is the highest priority, 10 is the lowest.
Enable Page Priority	The voice call in progress shall take precedence over all incoming paging calls.
Name	Listened multicast server name
Host: port	Listened multicast server's multicast IP address and port.

11.20 Settings >> Action

Action URL

Note! Action urls are used for IPPBX systems to submit phone events. Please refer to [Fanvil Action URL](#) for details.

11.21 Settings >> Time/Date

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

◦

Table 28 – Time & Date settings

Parameters	Description
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Primary Time Server	Set primary time server address

Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization.
Time Zone	Select the time zone
Resync Period	Time of re-synchronization with time server
12-Hour Clock	Set the time display in 12-hour mode
Date Format	Select the time/date display format
Daylight Saving Time Settings	
Local	Choose your local, phone will set daylight saving time automatically based on the local
DST Set Type	Choose DST Set Type, if Manual, you need to set the start time and end time.
Fixed Type	Daylight saving time rules are based on specific dates or relative rule dates for conversion. Display in read-only mode in automatic mode.
Offset	The offset minutes when DST started
Month Start	The DST start month
Week Start	The DST start week
Weekday Start	The DST start weekday
Hour Start	The DST start hour
Minute Start	The DST start minute
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Minute End	The DST end minute
Manual Time Settings	You can set your time manually

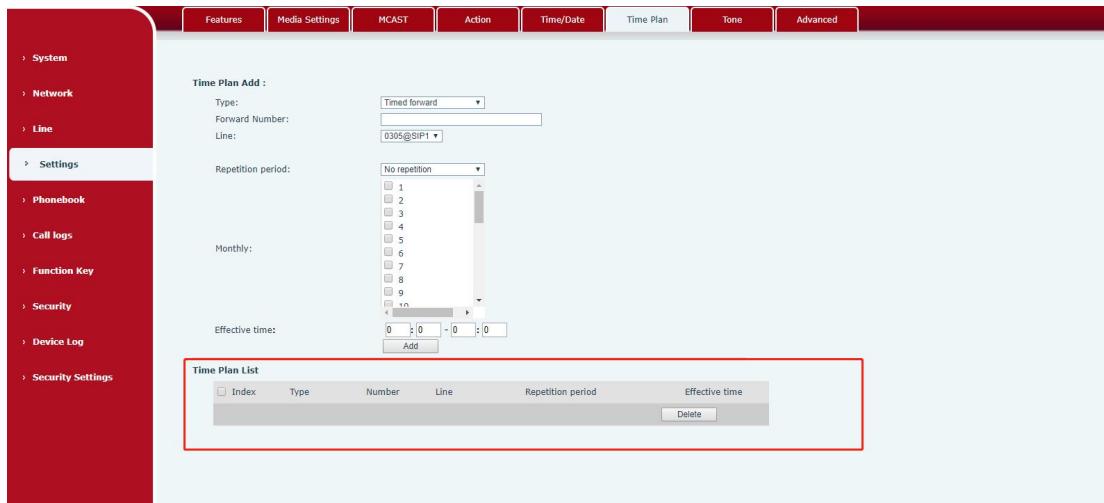
11.22 Setting >> Time plan

Users can configure the time plan to restart and upgrade phone

Table 29 - time plan

parameter	description
Type	Timed restart, timed upgrade, timed forward

Repetition period	Do not repeat: execute once within the set time range Daily: Perform this operation in the same time every day Weekly: Perform this operation in the same time of the week Monthly: Perform this operation in the same time of the Month
Effective time	Set the operation time
Forward Number	Set the SIP number for forwarding in the time range
Line	Set the line for forwarding in the time range

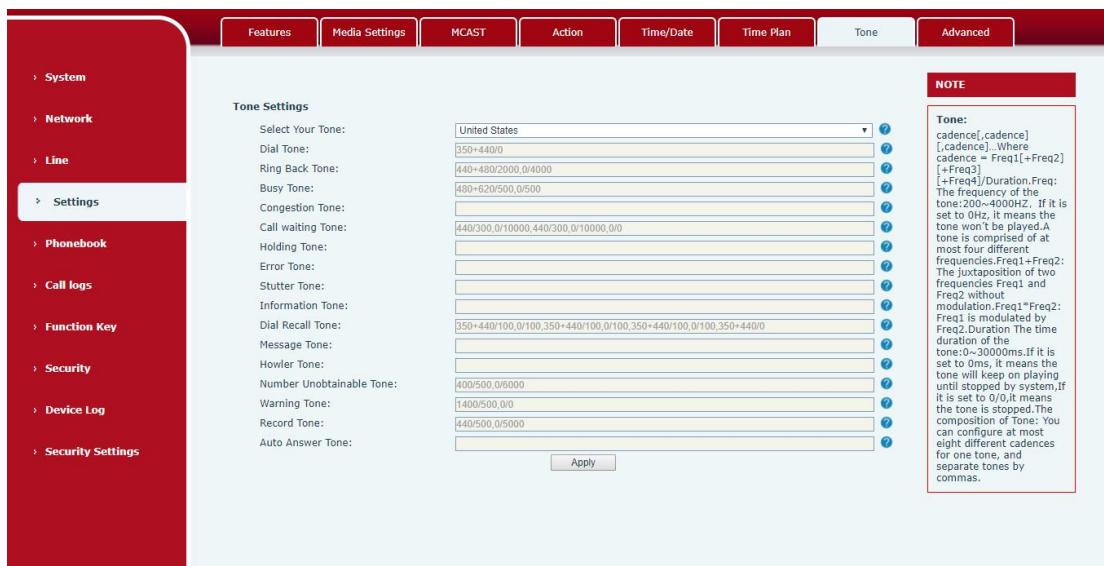


Picture 88- time plan

11.23 Settings > Tone

This page allows users to configure a phone prompt.

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



Picture 89- Webpage Tone

11.24 Settings >> Advanced

User can configure the advanced configuration settings in this page.

- Screen Configuration.
 - Enable Energy Saving
 - Backlight Time
 - Screen Saver
- LCD Menu Password Settings.
- Configure Greeting Words

The greeting message will display on the top left corner of the LCD when the device is idle, which is limited to 16 characters. The default chars are 'Indoor Station'.

11.25 Phonebook >> Contact

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

To add a new contact, user should enter contact's information and press "Add" button to add it.

To edit a contact, click on the checkbox in front of the contact, the contact information will

be copied to the contact edit boxes, press “Modify” button after finished editing.

To delete one or multiple contacts, check on the checkbox in front of the contacts wished to be deleted and click the “Delete” button, or click the “Clear” button with selecting any contacts to clear the phonebook.

User can also add multiple contacts into a group by selecting the group in the dropdown options in front of “Add to Group” button at the bottom of the contact list, selecting contacts with checkbox and click “Add to Group” to add selected contacts into the group.

Similarly, user can select multiple users and add them into blacklist by click “Add to Blacklist” button.

11.26 Phonebook >> Cloud phonebook

Cloud Phonebook

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

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

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

LDAP Settings

The cloud phonebook allows user to retrieve contact list from a LDAP Server through LDAP protocols.

User must configure the LDAP Server information and Search Base to be able to use it on the device. If the LDAP server requests an authentication, user should also provide username and password.

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

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

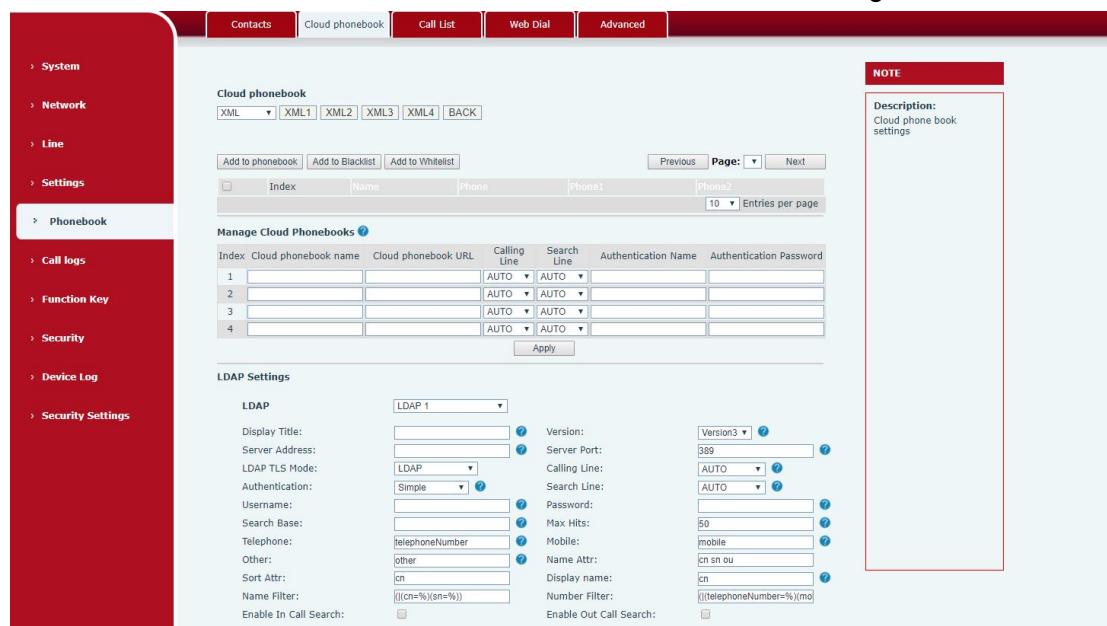
Access password (optional)

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

Web page preview

Phone page supports preview of Internet phone directory and contacts

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



Picture 90- Web cloud phone book Settings

11.27 Phonebook >> Call List

■ Restricted Incoming Calls:

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

Users can add specific Numbers to the blacklist or add specific prefixes to the blacklist to block calls with all Numbers with this prefix.

■ Allowed Incoming Calls:

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

■ Restricted Outgoing Calls:

Adds a number that restricts outgoing calls and cannot be called until the number is

removed from the table.

11.28 Phonebook >> Web Dial

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

11.29 Phonebook >> Advanced

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

Users can also import contacts into the phone book in XML, CSV, and VCF formats.

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

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

11.30 Call Logs

The user can browse the complete call record in this page. The call record can be sorted by time, call number, contact name or line, and the call record can be screened by call record type (incoming call, outgoing call, missed call, forward call).

The user can also save the number in the call record to his/her phone book or add it to the blacklist/whitelist.

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

Users can also download call records conditionally and save them locally.

11.31 Function Key >> Function Key

- Function Key Configuration:

One-key transfer Settings: establish new call, blind transfer, attention-transfer, one-key three-party, Play DTMF.

DSS Key home page: None/Page1/Page2/Page3/Page4

The device provides 112 user-defined shortcuts that users can configure on a web page.

Parameters	Description
Memory Key	<p>BLF (NEW CALL/BXFE /AXFER): It is used to prompt user the state of the subscribe extension, and it can also pick up the subscribed number, which help user monitor the state of subscribe extension (idle, ringing, a call). There are 3 types for one-touch BLF transfer method.</p> <p>p.s. User should enter the pick-up number for specific BLF key to fulfill the pick-up operation.</p> <p>Presence: Compared to BLF, the Presence is also able to view whether the user is online.</p> <p>Note: You cannot subscribe the same number for BLF and Presence at the same time</p> <p>Speed Dial: You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed.</p> <p>Intercom: This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments.</p>
Line	It can be configured as a Line Key. User is able to make a call by pressing Line Key.
Key Event	User can select a key event as a shortcut to trigger. For example: MWI / DND / Release / Headset / Hold / etc.
DTMF	It allows user to dial or edit dial number easily.
URL	Open the specific URL directly.
Multicast	Configure the multicast address and audio codec. User presses the key to initiate the multicast.
Action URL	The user can use a specific URL to make basic calls to the phone.
XML browser	Users can set the DSS Key for specific URL download and other operations.

Table 30- function key

11.32 Function Key>> Side Key

The device has 8 Side Keys, and users can configure each side key on the web

page.

For Side Key ,please refer to [11.31 function key](#).

11.33 Function Key >> Softkey

The User Settings mode and display style, display page.

Table 31- Softkey configuration

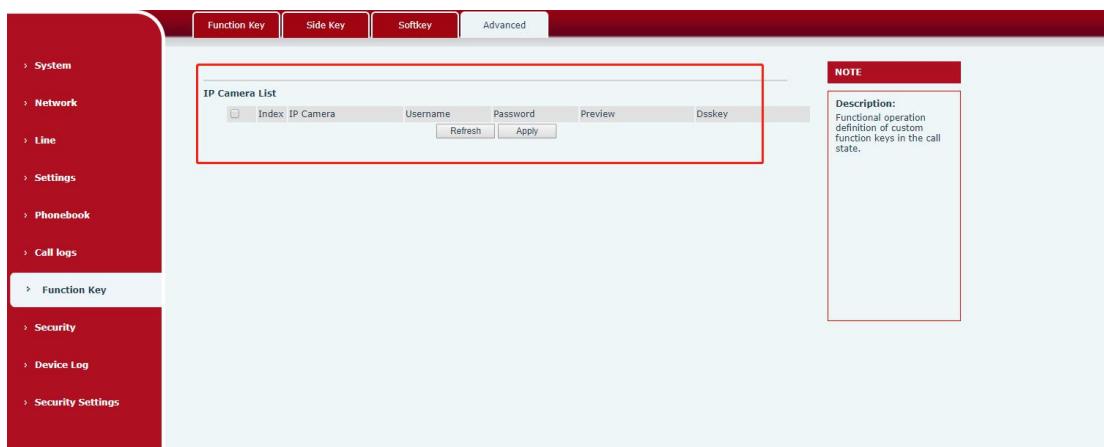
Parameter	Description
Softkey Mode	
Softkey mode	Disabled and More, Default is Disabled
Softkey Style	
Softkey display style	Softkey Exit on Left or Right
Screen	
Call Dialer	Redial/2aB/Delete/Exit/Call Back/Dial/Join/MWI/Local Contacts/Pickup/CallLog/Missed/Clear/In/Dialed/Pause/Next line/Prev line/Headset/Audio/Video/Remote XML/DSS Key
Conference	Hold/Split/End/Release/Mute/DSS Key/Headset
Desktop	CallLog/Menu/Local Contacts/DND/Prev Account/Next Account/Blacklist/Call Back/CallForward/Locked/Memo/Missed/MWI/Dialed/Reboot/Redial/Remote XML/SMS/Headset/Status/DSS Key/In
Divert Dialed	Redial/2aB/Delete/Exit/Forward/Local Contacts/CallLog /Clear/Missed/Dialed/Headset/Video/Audio/Remote XML /DSS Key
Ending	Redial/End/Headset/Release/DSS Key
Predictive Dialer	Dial/2aB/Delete/Exit/Call Back/Local Contacts/Redial /Pickup/MWI/Join/CallLog/Release/Missed/Pause/Dialed/Headset/Video/Audio/Remote XML/DSS Key/In/Next line /Prev line
Ringing	Answer/Forward/Reject/Mute/Release/Headset/Video/Audio/ DSS key
Talking	Hold/Transfer/Conference/End/Mute/Release/New Call/Local Contacts/Listen/CallLog/Next call/Prev call/ Private/Headset/Video/Audio/DSS Key
Transfer Alerting	End/Transfer/Headset/Release/DSS Key

Transfer Dialer	Redial/Delete/Exit/2aB/Dial/Local Contacts/Transfer/ CallLog/Clear/Missed/Dialed/Pause/Headset/Video/Audio/R emote XML/DSS Key
Trying	End/Release/Headset/DSS Key
Waiting	Hold/Transfer/Conference/End/Answer/Forward/Mute/Next call/New call/Prev call/Reject/Release/Headset/Listen/ Video/Audio/DSS Key

11.34 Function Key >> Advanced

■ IP Camera List

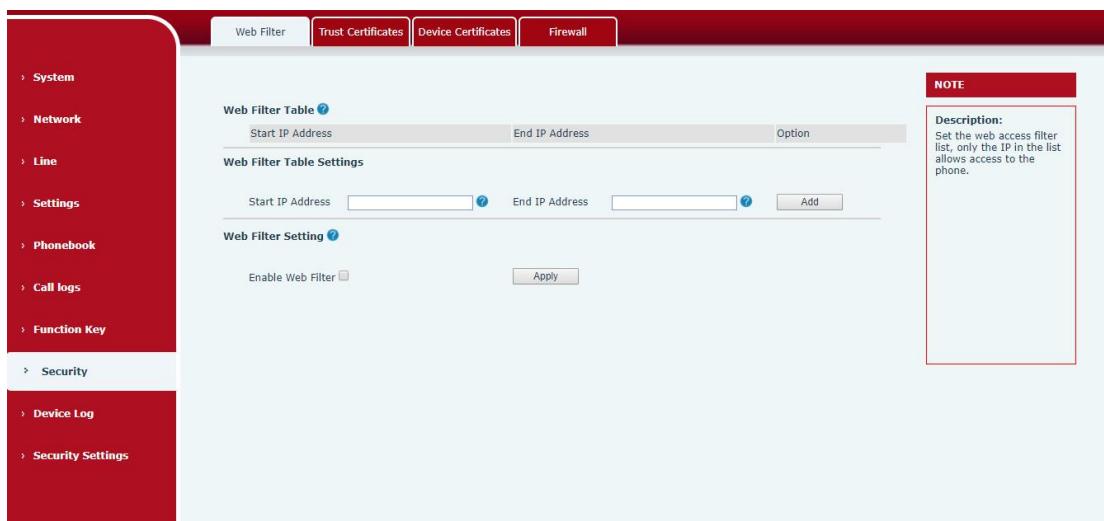
Support to discover the IP Camera in local area network. After scanning, you can bind the camera to the function key and press it to view video



Picture 91- IP Camera List

11.35 Security >> Web Filter

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



Picture 92- Web Filter settings

Start IP Address	End IP Address	Option
172.16.12.14	172.16.12.24	<input type="button" value="Modify"/> <input type="button" value="Delete"/>

Picture 93- Web Filter Table

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

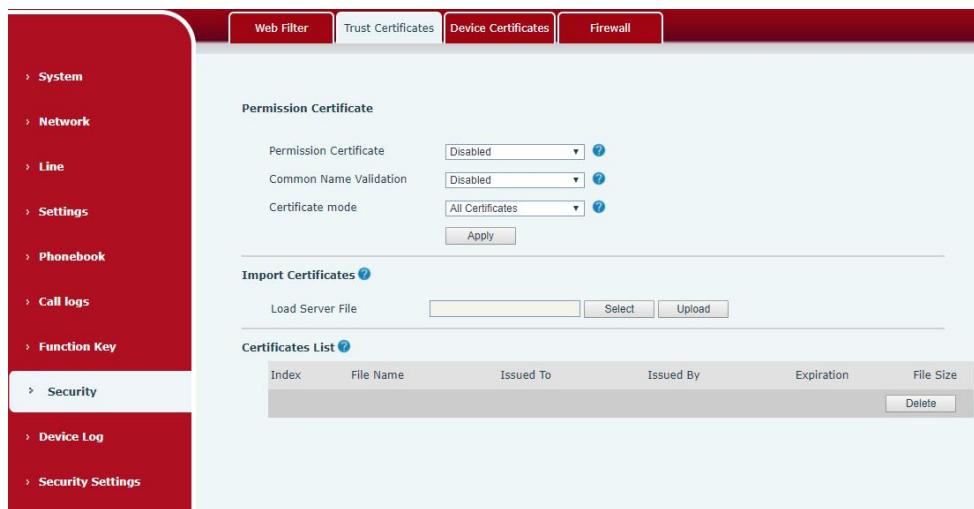
Enable web page filtering: configure enable/disable web page access filtering; Click the "apply" button to take effect.

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

11.36 Security >> Trust Certificates

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

You can upload and delete uploaded certificates.

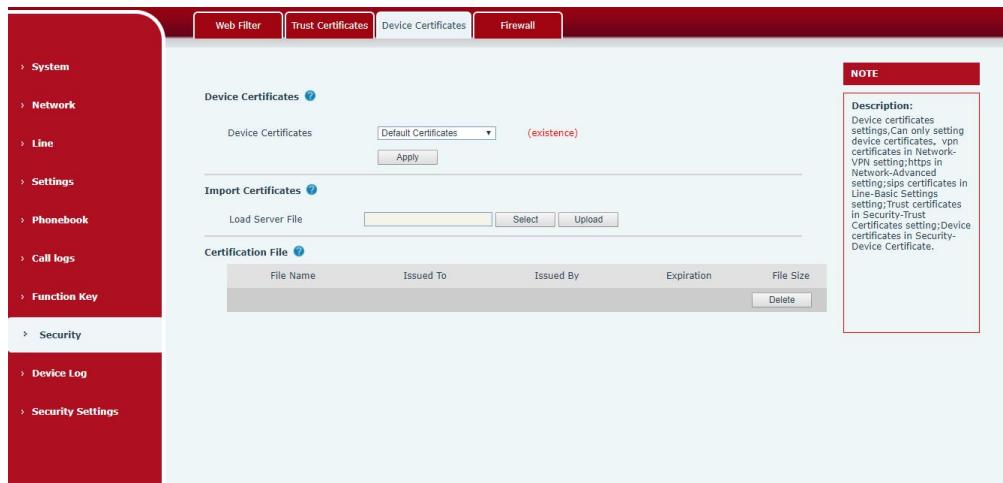


Picture 94- Certificate of settings

11.37 Security >> Device Certificates

Select the device certificate as the default and custom certificate.

You can upload and delete uploaded certificates.

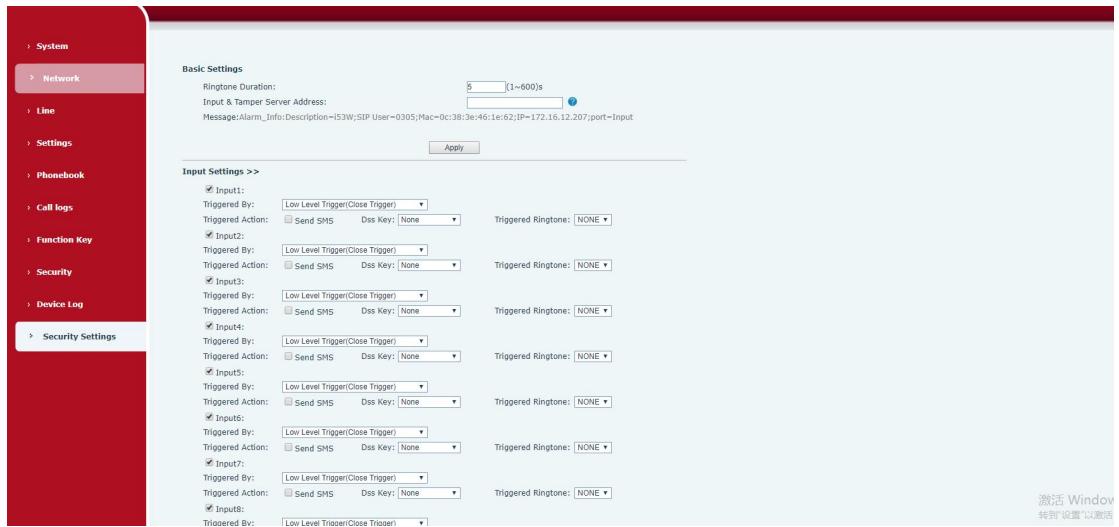


Picture 95 - Device certificate setting

11.38 Device Log >> Device Log

You can grab the device log, and when you encounter an abnormal problem, you can send the log to the technician to locate the problem. See 11.6 [Get log information](#).

11.39 Security Settings



Picture 96- Input and output settings

Security Settings	
parameter	description
Basic Settings	
Ringtone Duration	The duration of the alarm bell
Input & Tamper Server Address	Configure the remote response server address (including the remote response server address and the alarm trigger server address). When the input port is triggered, a short message will be sent to the server, the message format is as follows: Alarm_Info:Description=i51;SIP User=;Mac=0c:38:3e:39:6a:b6;IP=172.16.7.189;port=Input
Input Settings	
Input	Enable or disable the input port
Triggered By	When low level trigger (closed trigger) is selected, the detection input port (low level) closed trigger. When the high level trigger (disconnect trigger) is selected, the detection input port (high level) disconnect trigger.
Send SMS	Enable or disable the input port to send messages to the server
Dss Key	When set to dsskey1 or dsskey2, trigger dsskey to make a call, the default is none
Triggered Ringtone	Support ringtone selection

Table 32 - Input and output parameters

12 Trouble Shooting

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

12.1 Get Device System Information

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

The network information

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

12.2 Reboot Device

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

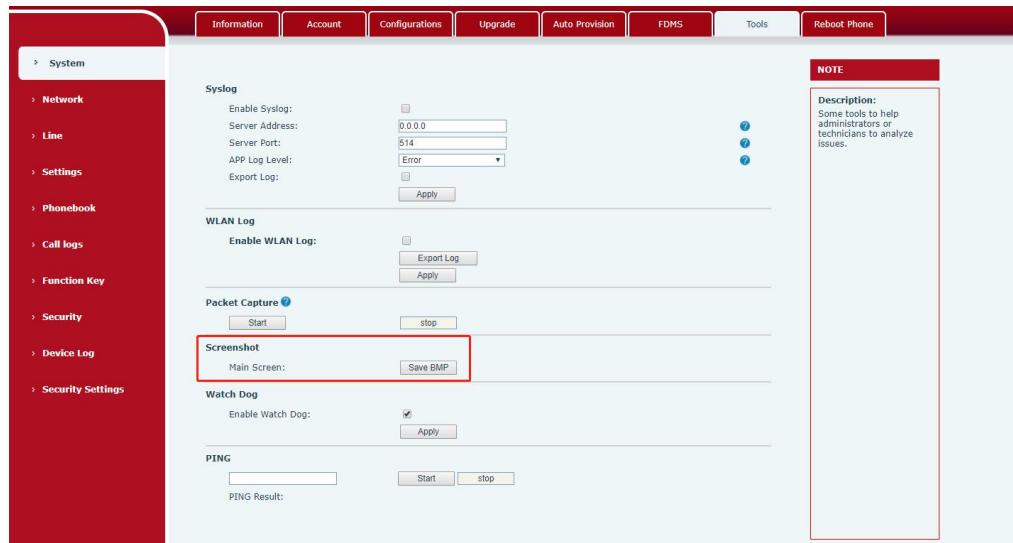
12.3 Reset Device to Factory Default

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

To perform a factory default reset, user should press [**Menu**] >> [**phone setting**]>> [**maintain**], and then input the password to enter the interface. Then choose [**Phone Reset**] and press [**Reset**]. The device will be rebooted into a clean factory default state.

12.4 Screenshot

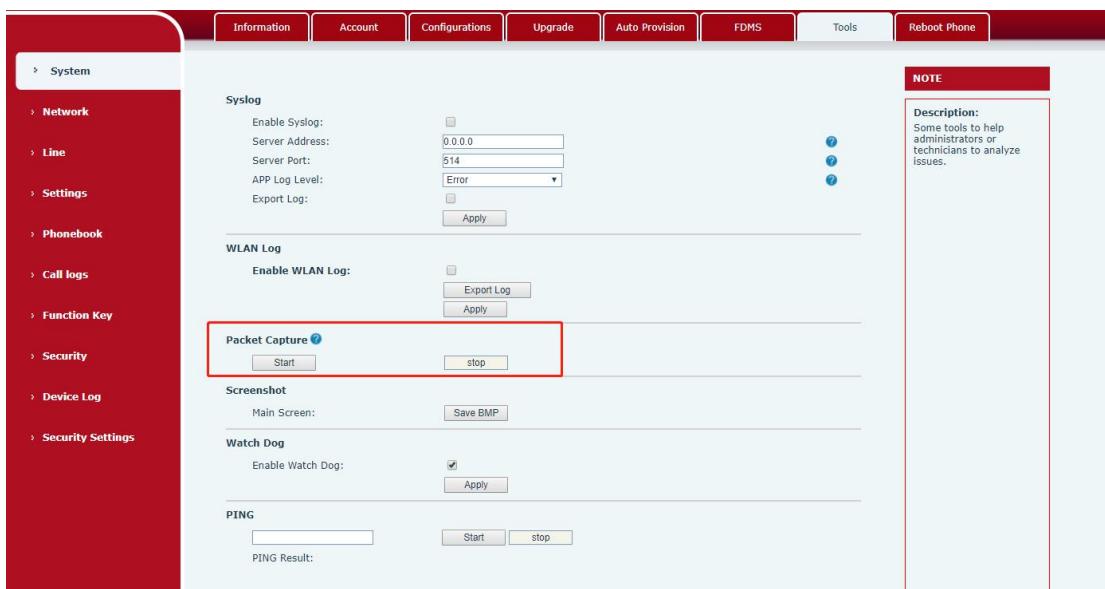
If there is a problem with the phone, the screenshot can help the technician locate the function and identify the problem. In order to obtain screen shots, log in the phone webpage [**System**] >> [**Tools**], and you can capture the pictures of the main screen (you can capture them in the interface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



Picture 97- Input and output settings

12.5 Network Packets Capture

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



Picture 98- Web capture

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

12.6 Get Log Information

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

12.7 Common Trouble Cases

Table 33 - Trouble Cases

Trouble Case	Solution
Device could not boot up	<ol style="list-style-type: none">1. The device is powered by external power supply via power adapter or PoE switch. Please use standard power adapter provided by Fanvil or PoE switch met with the specification requirements and check if device is well connected to power source.2. If you saw “POST MODE” on the device screen, the device system image has been damaged. Please contact location technical support to help you restore the phone system.
Device could not register to a service provider	<ol style="list-style-type: none">1. Please check if device is well connected to the network. The network Ethernet cable should be connected to the  [Network] port NOT the  [PC] port.2. Please check if the device has an IP address. Check the system information, if the IP displays “Negotiating...”, the device does not have an IP address. Please check if the network configurations is correct.3. If network connection is fine, please check again your line configurations. If all configurations are correct, please kindly

	<p>contact your service provider to get support, or follow the instructions in "12.5 Network Packet Capture" to get the network packet capture of registration process and send it to Fanvil support to analyze the issue.</p>
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