i51W&i52W User Manual

Software Version: 1.0.0

Release Date：2020/06/15

Directory

1 Picture

2 Table

Safety Instruction

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

- Please use the external power supply that is included in the package. Other power supply may cause damage to the phone and affect the behavior or induce noise.
- Before using the external power supply in the package, please check the home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it because it may cause fire or electric shock.
- Do not drop, knock or shake the phone. Rough handling can break internal circuit boards.
- This phone is design for indoor use. Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Before using the product, please confirm that the temperature and humidity of the environment meet the working requirements of the product.
- 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.

3 Overview

3.1 Overview

Fanvil i51W is a 4.3-inch indoor station and i52W is a 7-inch indoor station, which support 8-way alarm input and industrial power socket input interface. It is mainly used in residential areas, villas, office buildings and other places to receive calls from the door phone, and communicate with the door phone, and achieve the remote access control. It provides reliable security and convenient access to door control, creating a safe and comfortable living environment for the majority of users.

To help the users better understand the product details, this user manual can be used as a reference guide for i51W&i52W. This document may not be applicable to the latest version of the software. If you have any questions, you can use the i51W&i52W device's built-in help tips interface, or download and update your user manual from the Official Fanvil website.

3.2 Product Introduction

Table 1 – Model Configuration

Model	Configuration
i51W	4.3-inch display，Support WiFi、RS-485
i52W	7-inch display，Support WiFi、RS-485

4 Install Guide

4.1 Use PoE or external Power Adapter

i51W&i52W, called as ‘the device’ hereafter, supports two power supply modes, power supply from external power adapter or over Ethernet (PoE) complied switch.

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

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

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

5 Appendix Table

5.1 Appendix I - Icon

Table 2 - Keypad Icons

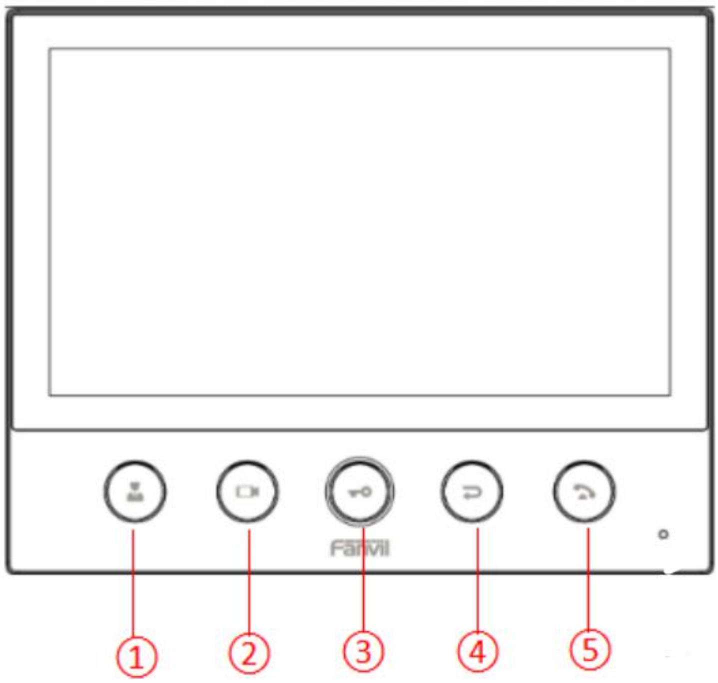
	Speed Dial
	Live View
	Open Door
	Return/End
	Answer

Table 3- Status Prompt and Notification Icons

	Call out
	Call in
	Network Disconnected
	Auto-answering activated
	The Voice quality of calling
	The Voice encryption of calling
	Connecting WIFI
	Open SIP Hotspot

6 Introduction to the User

6.1 Instruction of Keypad



Picture 1 – Key Instruction

The image above shows the device's key layout. Each key provides its own specific function. The user may refer to the key instructions in the illustrations in this section to operate the device.

Some keys can be long pressed. Users can hold down the button for 3 seconds to trigger the function.

Table 4 - Key Instruction

Key Index	Key Name	Description
1	Management Center	One press to call out the set number. If not set, press down and the pop-up prompts
2	Video Monitoring	One press to check the outdoor situation. If not set, press down and the pop-up prompts
3	Unlock Key	One press to unlock the door. Press 3s to enter the menu interface.
4	Return Key	Hang up when making a call

		Return to the parent directory
5	Answer Key	Answer the call

6.2 Idle Screen

Picture2 – Idle Screen

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

The upper half of the home screen displays the welcome message, time and date, and status information (such as automatic response, network connection status, etc.).

Bottom half icon area shows function key icon.

The icon description is described in Appendix I 6.1.

6.3 Web Management

Users can use the webpage of the device to manage and operate the device. The user first needs to enter the device's IP address in the browser to open the device's web page. Users can enter the menu [system Information] to see the device's IP address by long pressing the key.

1) Open the browser, enter the device IP, and log in the device web page. The first thing you see is the device login page.

Picture3 – Login Page

2) Long press the key to enter the menu [**System Message**] >> [**QrCode**], and log in to the web page with the QR code scanned by the phone's own browser.
Users must enter the correct username and password to log into the web page. The default username and password are "admin". Refer to the 10 Page Configuration for details on how to operate the page

6.4 Network Configurations

Fanvil i51W &i52W support two network connection modes: wired network connection and wireless network connection. Users should choose the corresponding connection mode according to their own situation.

Devices use IP network connections to provide services. Unlike traditional devices based on line circuit technology, IP devices exchange packets and data based on device IP addresses connected to each other over the network.

6.4.1 Wired Network

To enable the device, you must first configure the network configuration correctly. Users need to press long to enter the menu to configure the network. After entering the menu, users should follow the prompts to enter [**Network Settings**] >> [**Network**].

Notice: If the user sees the  *"Network not connected" icon at the top of the screen, that means the cable is not connected to the device's network port. Check that the cable connects the device to the network switch, router, or modem.*

There are two common IP configuration types for IPv4:

1) DHCP – This is the mode to automatically get the network configuration from the server. The user does not need to manually configure any parameters. Suitable for most users.
2) Static IP –This option allows the user to manually configure each IP parameter, including IP address,

mask, gateway, and primary and secondary DNS servers. This usually applies to some professional web user environments.

3) The default configuration of the device is the automatically configured network mode

6.4.2 WLAN

If there is no wired network, you can connect to the background web page of the device by turning on the AP mode of the device and set up the wireless network connection.

1) Long press to enter the menu. After entering, follow the prompts to enter [**Network Settings**] >> [**AP Settings**]
2) After entering, press to enable AP according to the interface prompt, prompt restart, and restart will take effect
3) After restarting, enter the AP setting interface, and you can see the SSID and IP address named after the MAC address of the device

Picture4– AP Information

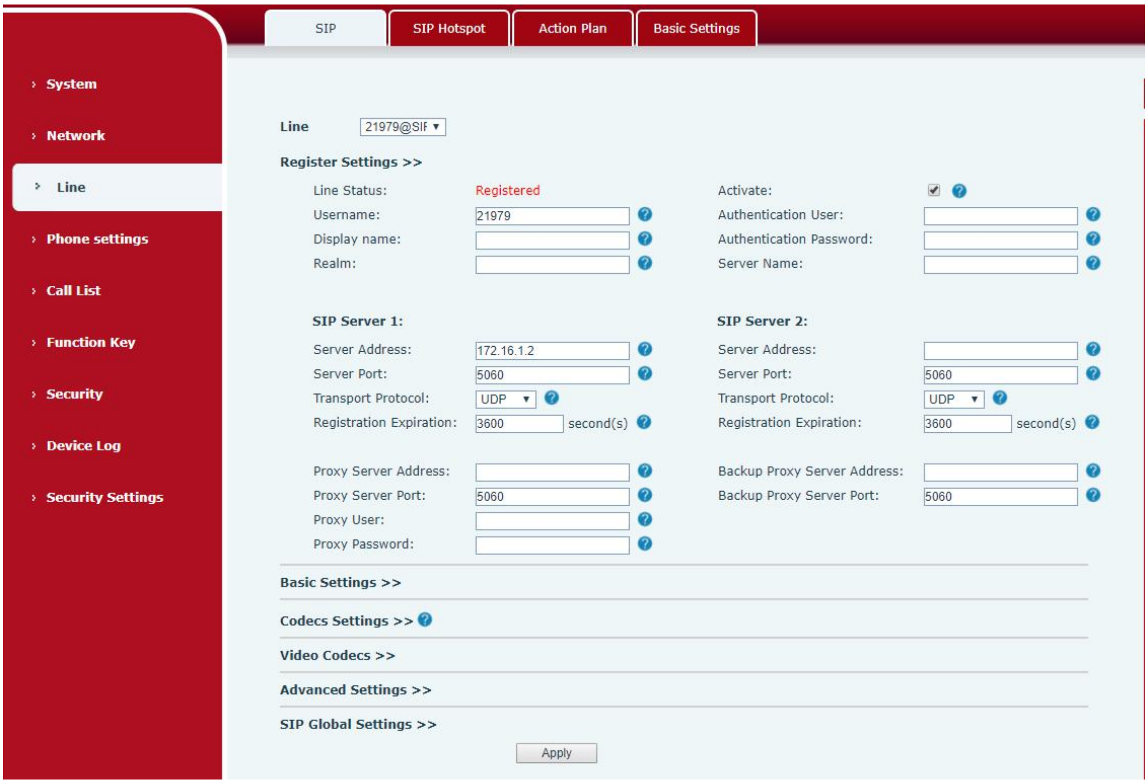
1) Turn on the mobile phone Wi-Fi, and you will see the Wi-Fi network named after the device's MAC address. Click to connect without a password
2) After successful connection, scan the QR code with the phone's own browser and enter the background login interface of the device
3) Enter user name/password (default admin)
4) After logging in, select Wi-Fi Settings, manually add Wi-Fi and enable Wi-Fi, and the device will automatically connect to the Wi-Fi network after setting
5) Return to the superior directory and enter [**WLAN**]. The wireless state is connected, or go back to standby to see the Wi-Fi icon in the status bar

6.5 Lines Configuration

At least one line of equipment must be configured correctly to provide telephone service. The line configuration works like a virtual SIM card that holds a mobile phone with service provider and telephone account authentication. When the device applies these configurations, the device automatically registers with a stored information provider, just as you can insert a SIM card into any mobile phone, and the phone USES the information in the SIM card to apply services, rather than the phone itself.

The user can configure the line in the calling device interface or the web interface and enter the corresponding information in the registered address, registered user name, registered password and SIP user, display name and registered port respectively, which are provided by the SIP server administrator.

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



Picture 5 – Line Registration

7 Basic Function

7.1 Call

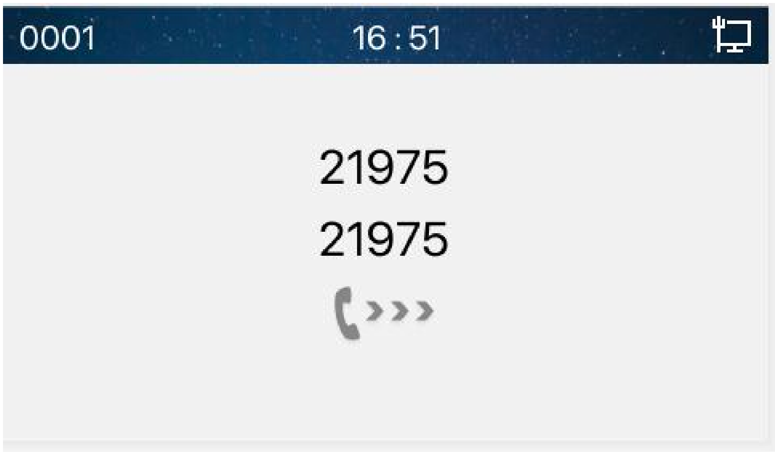
The DSS key for webpage setting is the memory key, the type is speed dial, the value is the number to call, the line is set as the registered line, and the media is set as the default, as shown below:

Function Key Settings

Key	Type	Name	Value	Subtype	Line	Media	PickUp Number
DSS Key 1	Memory Key		21975	Speed Dial	21979@SIP1	DEFAULT	
DSS Key 2	None			None	AUTO	DEFAULT	
DSS Key 3	DTMF			None	AUTO	DEFAULT	
DSS Key 4	Key Event			End	AUTO	DEFAULT	
DSS Key 5	Key Event			Handfree	AUTO	DEFAULT	

Apply

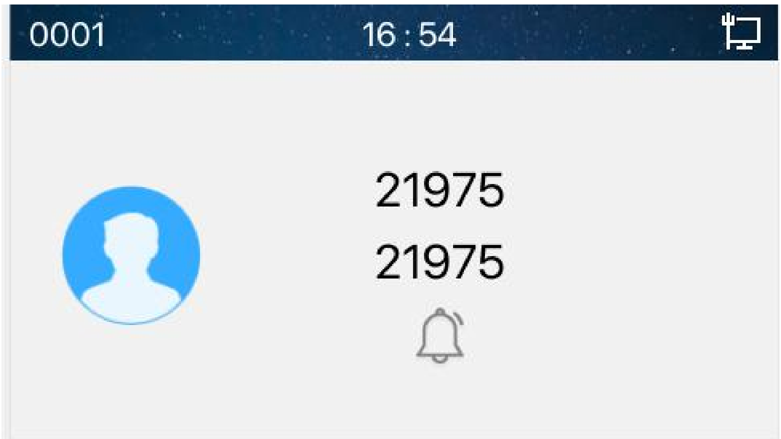
After setting, press the button to exhale, and the exhale interface is as follows:



Picture 6 – Dialing Interface

7.2 Answer

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



Picture 7 – Incoming Call Interface

The user can answer the phone by pressing the button . To reject a call, the user presses the button .

7.3 Unlock

The DSS key for webpage setting is DTMF, and the value is the opening password for the caller of access control

You can open the door by pressing the button while talking.

7.4 End the call

Picture8 – Dialing Interface

When users end the call

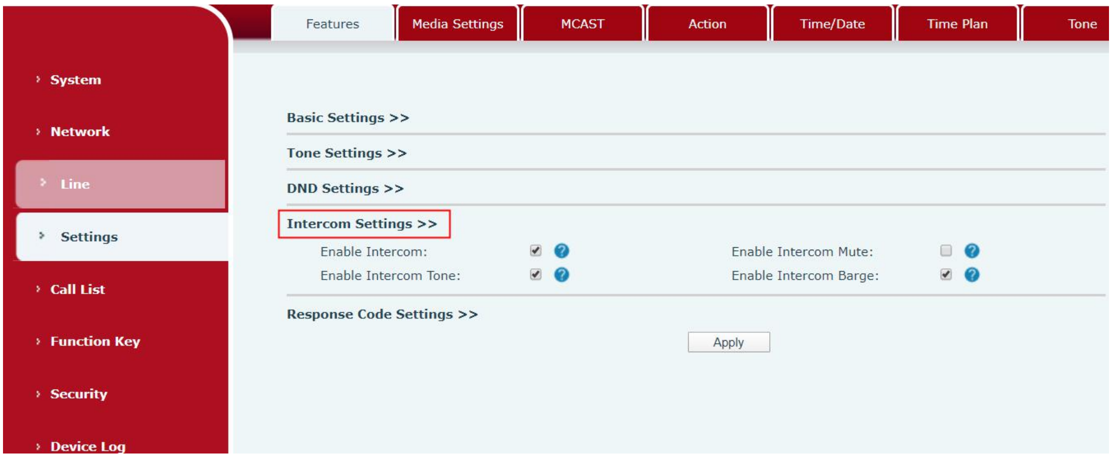
1） Press

2） Press

8 Advanced Function

8.1 Intercom

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



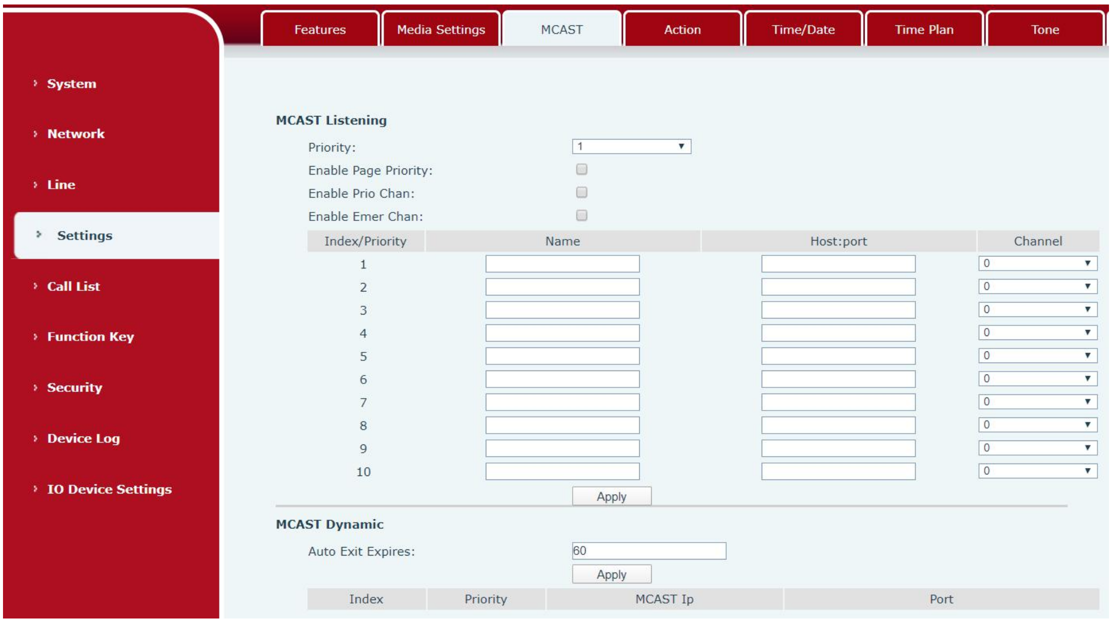
Picture 9 - Web Intercom configure

Table 5 - 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

8.2 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 10 - Multicast Settings Page

Table 6 - 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 [**Settings**] >> [**MCAST**].
- Press the DSSKY of Multicast Key which you set.
- Receive end will receive multicast call and play multicast automatically.

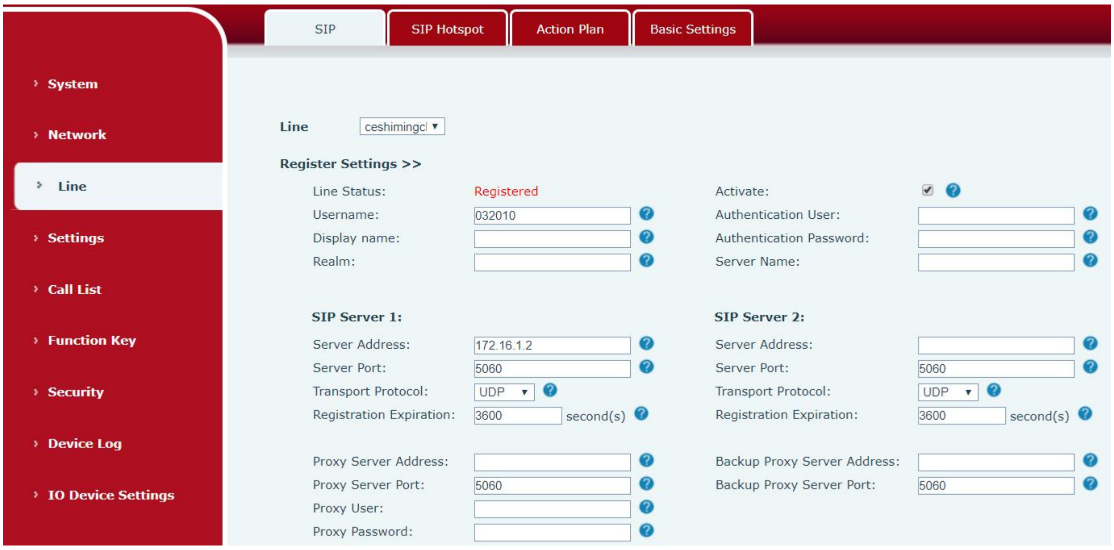
Dynamic multicast:

- Function description: send multicast configuration information through Sip Notify signaling. After receiving the information, the device will be configured to perform multicast monitoring in the system or cancel multicast monitoring in the system

8.3 SIP Hotspot

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

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

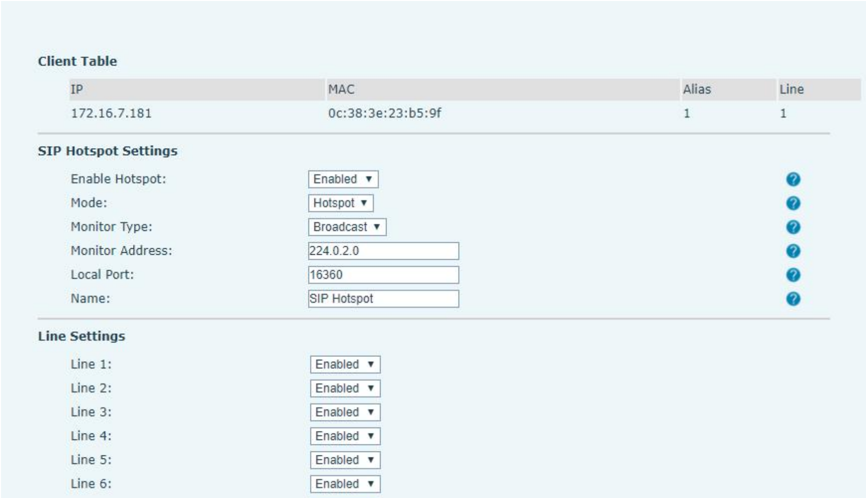


Picture 11 - Register SIP account

Table 7- 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.

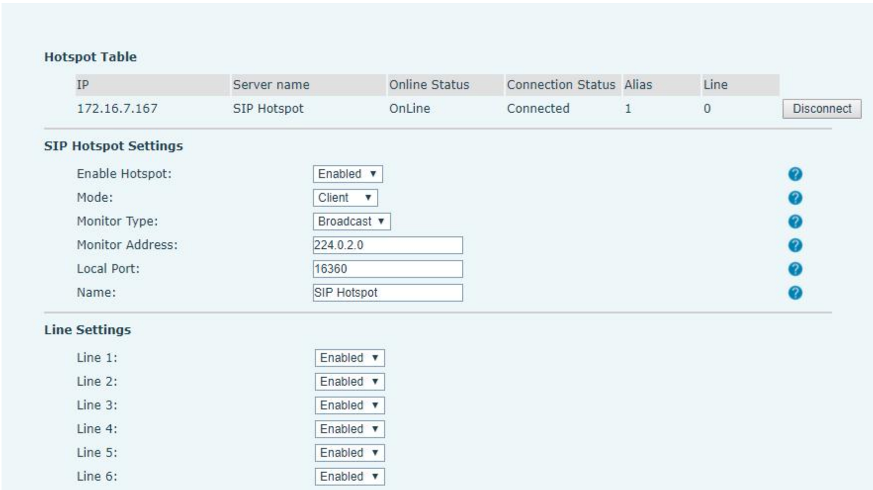
Configure SIP hotspot server:



Picture 12 - SIP hotspot server configuration

Configure SIP hotspot client:

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



Picture13- 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.

9 Web Configurations

9.1 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.

9.2 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)

9.3 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.

9.4 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.

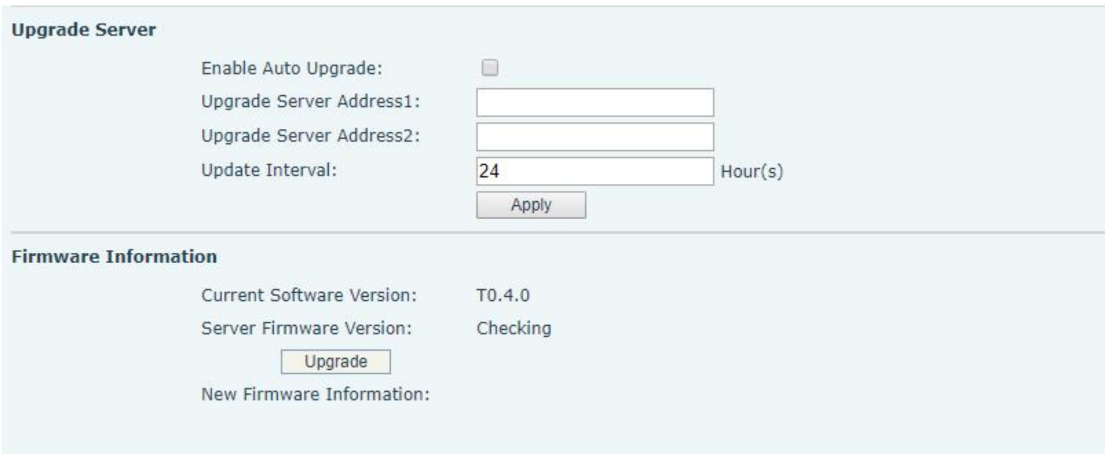
9.5 System >> Upgrade

Upgrade the software version of the device. Upgrade to the new version through the web page. After the upgrade, the device will automatically restart and update to the new version. Click select, select version and click Upgrade.

Support to upgrade ring tones and background.

Online upgrade:

Firmware online upgrade is for the device to send an HTTP request to the server. It returns with a corresponding description file or 404 or timeout. After the device receives the file, it will parse the version of description file and prompt the user for a new version and whether to upgrade



Picture 14 – Online Upgrade Configuration

Table8 – Online Upgrade Parameter

Parameter	Description
Upgrade Service	
Upgrade Service Address 1	Fill in the address of the available primary upgrade server (HTTP server).
Upgrade Service Address 2	Fill in the address of the available backup upgrade server (HTTP server) and request the backup server when the primary server is unavailable.
Software Version	
Present Software Version	Display the current device software version number.
Present Server Version	Display the current server software version number.
[Upgrade]Button	When the server side has the corresponding TXT file and version, the [Upgrade] button changes from ash to available state, click [Upgrade] to choose whether to upgrade.
New Version Description Information	When the server side has the corresponding TXT file and version, the new version description information will show the version of TXT information.

- The file that the device requests from the server is a TXT file called vendor_model_hw1_0.txt.Hw is followed by the hardware version number.Underline all Spaces in file names.
- The URL requested by the device is HTTP:// server address /, and the new version and the requested file are placed in the download directory of the HTTP server.
- TXT file format must be UTF-8
- vendor_model_hw1_0.txt document format is as below：

  Version=1.6.3 #software version

  Firmware=xxx/xxx.z #xxx.z or http://server IP：ports/Content/xxx.z

  BuildTime=2018.09.11 20:00

  Info=TXT|XML

  Xxxxx

  Xxxxx

  Xxxxx

  Xxxxx

9.6 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。

https://www.fanvil.com/Support/download/cid/14.html

9.7 System >> Tools

Tools provided in this page help users to identify issues at trouble shooting. Please refer to 10 Trouble Shooting for more detail.

9.8 System >> Reboot Phone

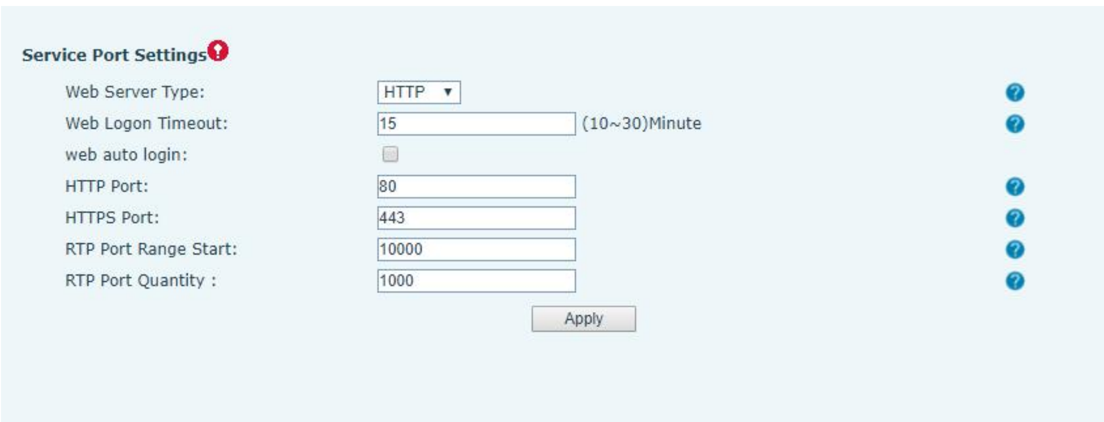
This page can restart the phone.

9.9 Network >> Basic

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

9.10 Network >> Service Port

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



Picture 15 - Service Port Settings

Table 9 - 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.

9.11 Line >> SIP

Configure the Line service configuration on this page.

Table 10 - 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.
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	Set the feature code to dial to the server

Unconditional	
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.

9.12 Line >> SIP Hotspot

Please refer to 8.3 SIP Hotspot.

9.13 Line >> Action Plan

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

Table 11 - IP camera

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

9.14 Line >> Basic Settings

Set up the register global configuration.

Table 12- 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
The TLS authentication	
TLS Certification File	Upload or delete the TLS certification file used for encrypted SIP transmission.
SIP P2P Configuration	
Enable automatic answer	Answer the call automatically after the timeout
Automatic answer waiting time	Timeout setting of auto-answer
DTMF Types	Set the DTMF types of lines
DTMF SIP info Types	Set SIP INFO mode to send '*' and '#' or '10' and '11'
Enable preview	Video preview after IP call
Preview mode	Set preview mode to 18X or 2xx

9.15 Settings >> Features

Configuration phone features.

Table 13 - General function Settings

Parameters	Description
Basic Settings	
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
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.
Default response mode	Set the default voice/video answer
Default dialing mode	Set the default voice/video outgoing call
Hide DTMF	Configure the hide DTMF mode.
Enable CallLog	Select whether to save the call log.
Enable Country Code	Whether the country code is enabled.
Country Code	Fill in the country code.
Area Code	Fill in the area code.
Enable Number Privacy	Whether to enable number privacy.
Match Direction	Matching direction, there are two kinds of rules from right to left and from left to right.
Start Position	Open number privacy after the start of the hidden location.
Hide Digits	Turn on number privacy to hide the number of digits.
Allow IP Call	If enabled, user can dial out with IP address
P2P IP Prefix	Prefix a point-to-point IP call.
Caller Name Priority	Change caller ID display priority.
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP address. More details please refer to this link https://www.fanvil.com/Support/download/cid/14.html。
Line Display Format	Custom line format：SIPn/SIPn：xxx/xxx@SIPn
Call number filtering	Configure a special character &, the number of the other party is 78&9, the call will be filtered out &
Automatically restore the current call	Automatically releases HOLD if current path changes
Call timeout	Stop the call after the call timeout
Tone Settings	

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

9.16 Settings >> Media Settings

Change voice Settings.

Table 14 - Voice settings

Parameter	**Description**
Codecs Settings	Select enable or disable voice encoding: G.711A/U,G.722,G.729, ILBC, Opus
Audio Settings	
Default Ring Type	Configure default ringtones. If no special ringtone is set for the phone number, the default ringtone will be used.
Speakerphone Volume	Set the hands-free volume to 1-9.
Speakerphone Ring Volume	Set the volume of hands-free ringtone to 1~9.

DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
Opus playload type	Set Opus load type, range 96~127.
OPUS Sample Rate	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb (16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length
Enable MWI Tone	When there is a new voice message, the phone will start a special dial tone.
Enable VAD	Whether voice activity detection is enabled.
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

9.17 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 15 - 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.

9.18 Settings >> Action

Action URL

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

https://www.fanvil.com/Support/download/cid/14.html

9.19 Settings >> Time/Date

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

Table 16– 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

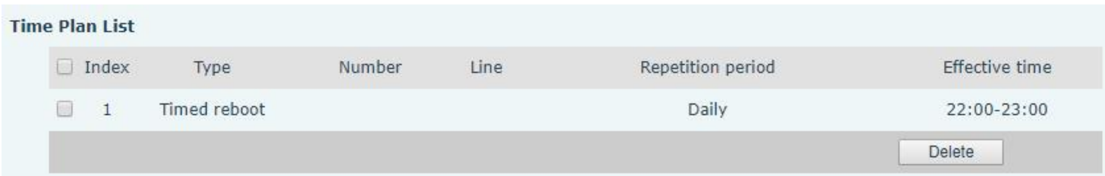
Week Start	The DST start week
Weekday Start	The DST start weekday
Hour Start	The DST start hour
Minute Start	The DST start minute
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Minute End	The DST end minute
Manual Time Settings	You can set your time manually

9.20 Settings >> Time Management

Users can configure the time period and time point in the secondary interface to restart and upgrade operations

Table 17 - Time Management Settings

Parameters	**Description**
Types	Regular restart, regular upgrade, regular forwarding
Repetition Period	Do not repeat: Set the time range to execute once Daily: Perform this operation for the same time frame each day Weekly: Do this on a time scale of what day of the week Monthly: Perform this operation within the time range on what day of the month
Effective time	Set the time period for execution
Forwarding Number	Select the forwarding SIP number in the period
Line	Select the forwarding SIP line in the period

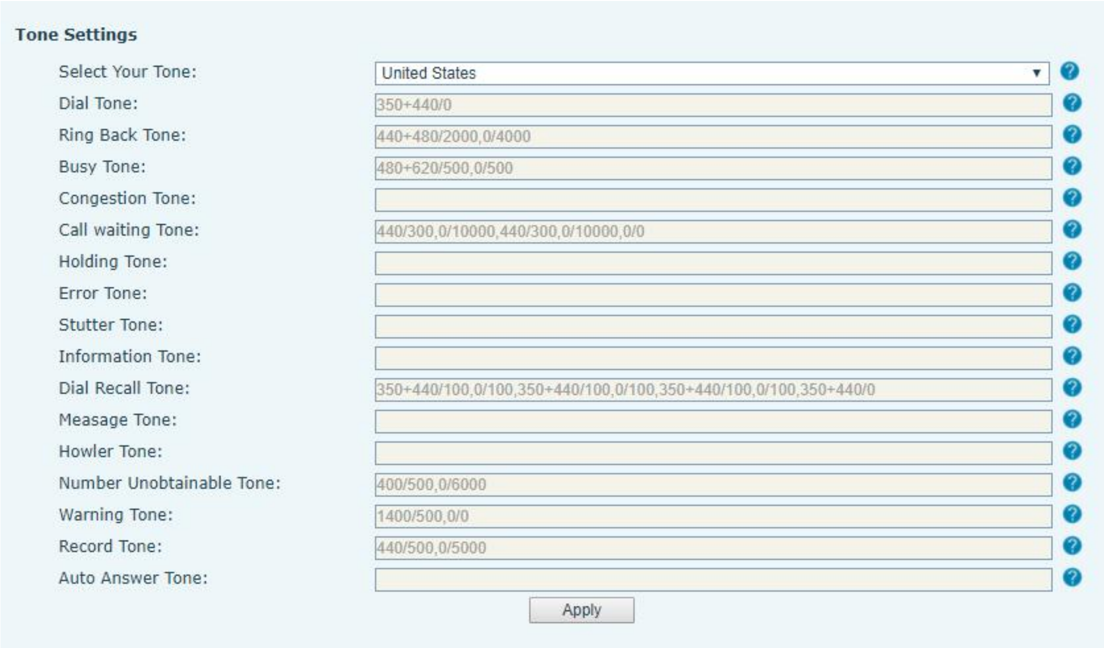


Picture 16- Time Management Settings

9.21 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 17 - Tone settings on the web

9.22 Settings >> Advanced

User can configure the advanced configuration settings in this page.

- Screen Configuration.
  - Enable Energy Saving
  - Backlight Time
  - Screen Saver
- 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’.

9.23 Call List >> 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.

9.24 Call List >> Web Dial

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

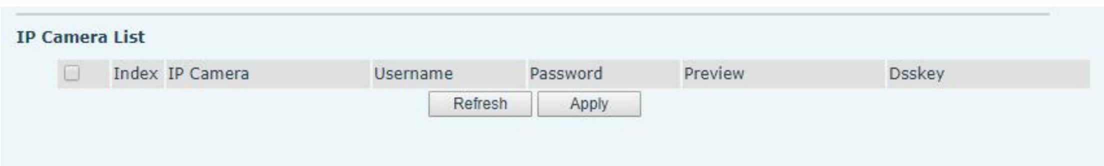
9.25 Function Key >> Function Key

Table 18 - Function Key configuration

Parameters	Description
Memory Key	**Speed Dial**: You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed. **Intercom**: This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments.
Line	It can be configured as a Line Key. User is able to make a call by pressing Line Key.
Key Event	User can select a key event as a shortcut to trigger. For example: DND/ Handfree / intercom / 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.

■ **IP Camera List**

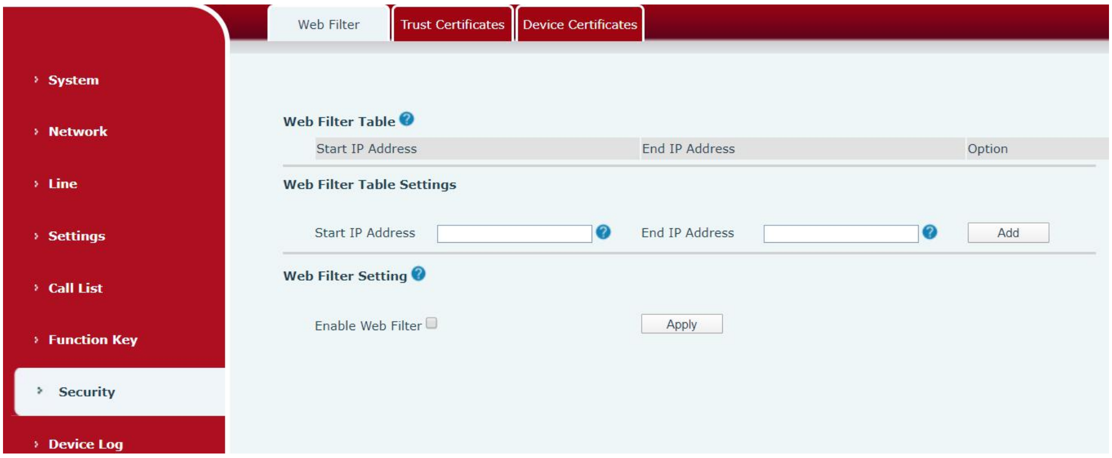
It supports IP Camera to find LAN. After scanning, the camera can be bound to the function key. One press to view the video when standby.



Picture 18 - IP Camera List

9.26 Security >> Web Filter

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



Picture 19- Web Filter settings

Picture 20 - Web Filter Table

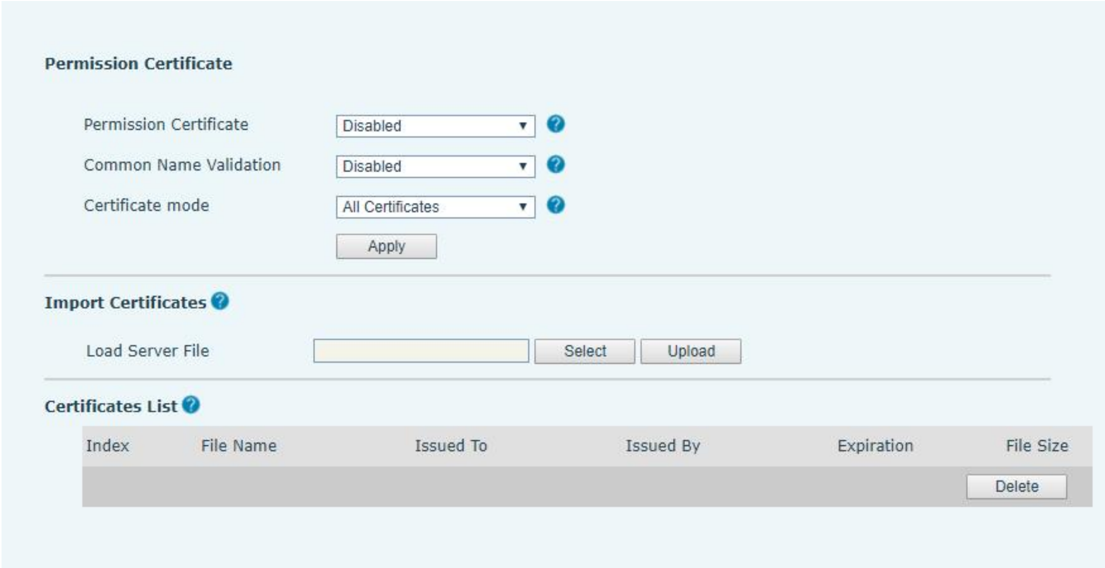
Add and remove IP segments that are accessible; Configure the starting IP address within the start IP, end the IP address within the end IP, and click [**Add**] to submit to take effect. A large network segment can be set, or it can be divided into several network segments to add. When deleting, select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

9.27 Security >> Trust Certificates

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

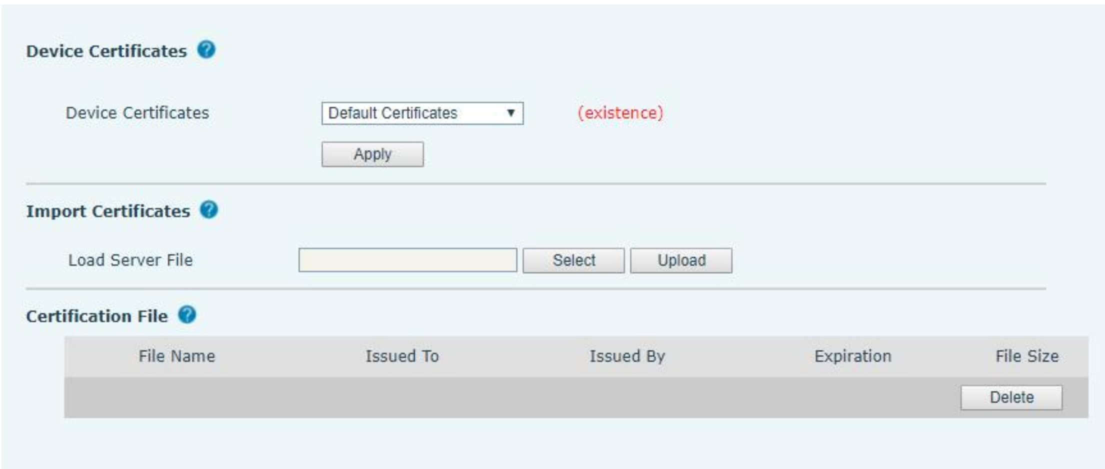


Picture 21 - Certificate of settings

9.28 Security >> Device Certificates

Select the device certificate as the default and custom certificate.

You can upload and delete uploaded certificates.

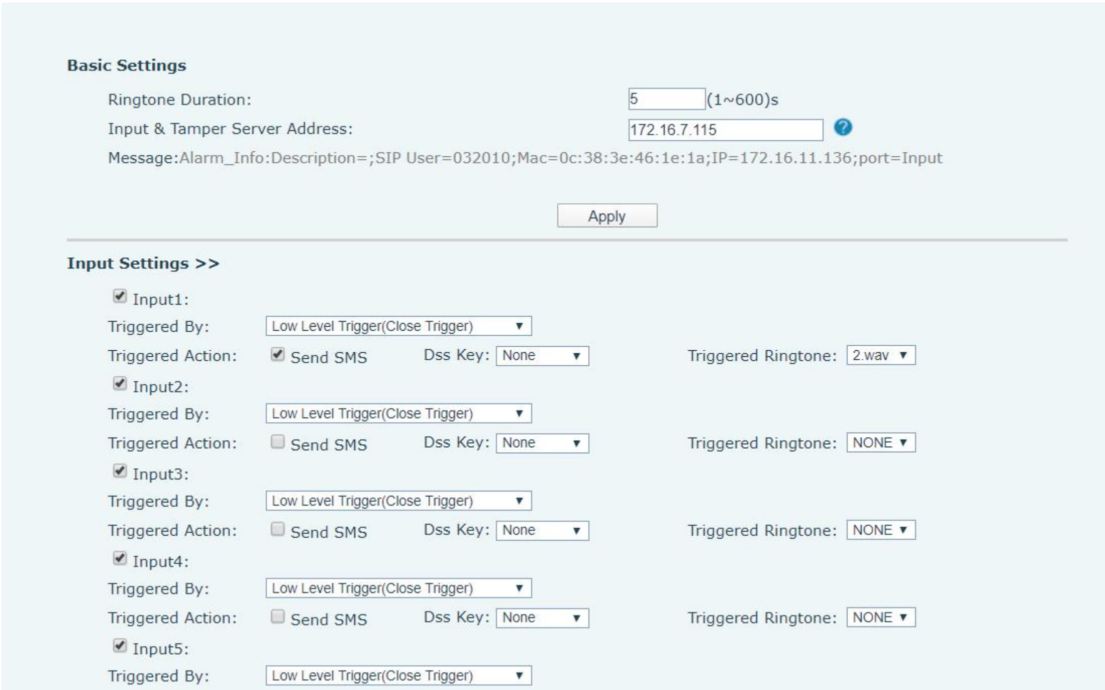


Picture 22 - Device certificate setting

9.29 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.

9.30 Security Settings



Picture 23- Security Settings

Table 19 – Input/ Output Parameters

Security Products Settings	
Paraments	**Description**
Basic Settings	
Ring Time	The duration of the alarm
Server Address	Configure the remote response server address (including the remote response server address and the triggering alarm server address). When the input port is triggered, a short message is sent to the server in the following format: Alarm Info: Description=i51;SIP User=;Mac=0c:38:3e:39:6a:b6;IP=172.16.7.189;port=Input
Input port settings	
Input port	Enable or disable the input port
Tigger Mode	When low level trigger (close trigger) is selected, the input port (low level) close trigger is detected.

	When the high-level trigger (disconnect trigger) is selected, the input port (high level) disconnect trigger is detected.
Send short messages	Send Short Messages enables or disables input ports to Send messages to the server
DSS keys	When set to DSSKey1 or DSSKey2, the DSS key is triggered for the call, which defaults to NONE
Ring Trigger	Support to choose the ring

10 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.

10.1 Receive the System Info of Device

Users can enter the menu interface by long pressing. After entering the menu, users can view the [**System Message**] according to the interface prompt. The following information will be provided:
The network information
Equipment information (model, hardware and software version), etc

10.2 Reboot Device

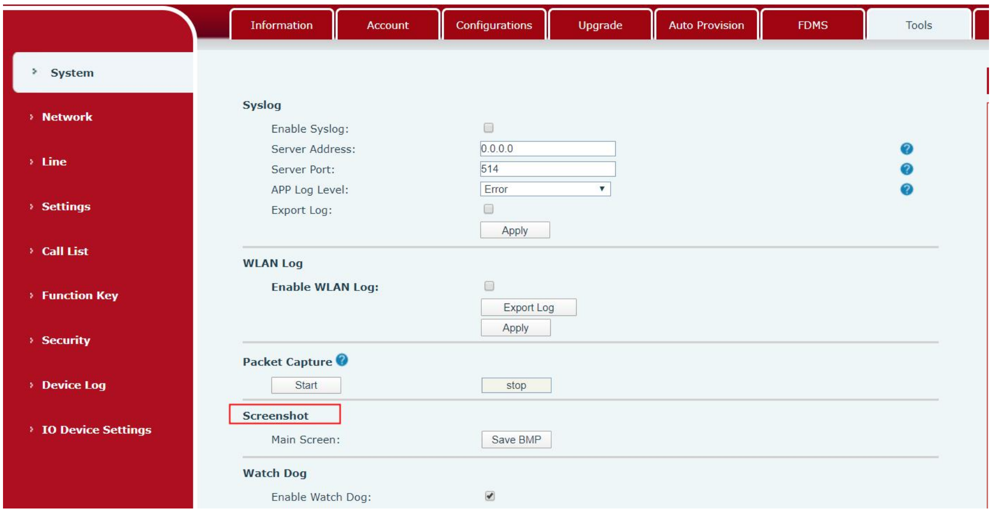
After entering the menu interface , the user can select [**Reboot System**] according to the interface prompt, press "OK" after the pop-up prompt, and press "OK" again to restart the device.

10.3 Reset Device

Restoring factory Settings will remove all configuration, preferences, databases, and configuration files on the device, and the device will revert to its factory default state.
After entering the menu interface , the user can select [**Factory Reset**]" according to the interface prompt, press "OK" after the pop-up prompt, and press "OK" again to restore the device.

10.4 Screenshot

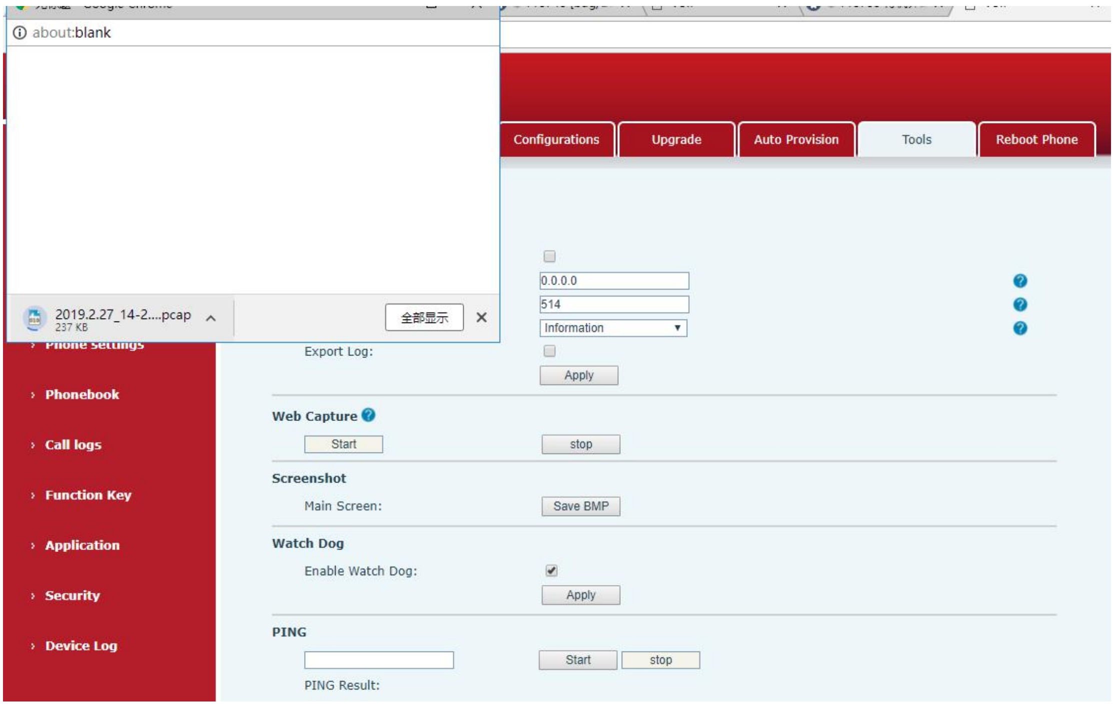
If there is a problem with the phone, the screenshot can help the technician locate the function and identify the problem. In order to obtain screen shots, log in the phone webpage [**System**] >> [**Tools**], and you can capture the pictures of the main screen (you can capture them in the interface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



Picture 24- Screenshot

10.5 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 "Network Packets Capture" section. A pop-up message will be prompt to ask user to save the capture file. User then should perform relevant operations such as activate/deactivate line or making phone calls and click [**Stop**] button in the web page when operation finished. The network packets of the device during the period have been dumped to the saved file.



Picture 25- Web capture

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

10.6 Get Log Information

Log information is helpful when encountering an exception problem. In order to get the log information of the phone, the user can log in the phone web page, open the page [**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.

10.7 Common Trouble Cases

Table 20- Trouble Cases

Trouble Case	Solution
Device could not boot up	1. The equipment is powered by an external power source from the power adapter or PoE exchange. Use a Fanvil power adapter or a PoE switch that conforms to standard specifications and check that the device is connected to the power supply. 2. If the device enters "POST mode", the device system is damaged. Please contact location technical support to help you restore the equipment system.
Device could not register to a service provider	1. Please check whether the device is connected to the network. The network cable should be properly connected to [network]. 2. Please check whether the device has AN IP address. Check the system information, if the IP address is Negotiating..., indicating that the device did not get the IP address. Check that the network configuration is correct. 3. If the network connection is good, please check your line configuration again. If all configurations are correct, contact your service provider for support, or follow the instructions in "10.5 Network Data Capture" to get a registered network packet and send it to the Fanvil support mailbox to help analyze the problem.