H1 User Manual

Software Version: 2.12.1

Release Date：2023/02/03

Directory

1 Picture

2 Table

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 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.
- Avoid exposure the phone to high temperature or below 0℃ 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.

FCC Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation.

This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications.

However, there is no guarantee that interference will not occur in a particular installation.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

-Reorient or relocate the receiving antenna.

-Increase the separation between the equipment and receiver.

-Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

-Consult the dealer or an experienced radio/TV technician for help. To assure continued compliance, any changes or modifications not expressly approved by the party.

Responsible for compliance could void the user's authority to operate this equipment. (Example- use only shielded interface cables when connecting to computer or peripheral devices).

This equipment complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

(1) This device may not cause harmful interference, and

(2) This device must accept any interference received, including interference that may cause undesired operation.

FCC Radiation Exposure Statement: The equipment complies with FCC Radiation exposure limits set forth foruncontrolled enviroment. This equipment should be installed and operated withminimum distance 20cm between the radiator and your body.

4 Overview

4.1 Overview

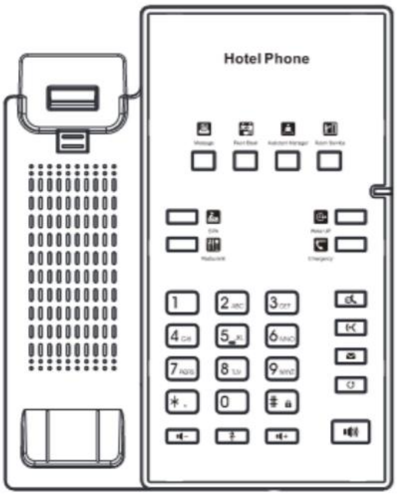
H1 is a WIFI product phone, with advanced design, high cost performance, paperless office, which greatly improves the production efficiency of the enterprise; not only a desk phone, but also a boutique placed in the living room or office .

H5W, which are the latest generation of IP phone developed on the basis of the H series, inheriting many excellent features of the previous H series traditional phone, such as high-definition voice and high-performance echo cancellation full duplex speaker, fast / gigabit Ethernet, QoS, encryption transmission, automatic configuration, new system, smooth operation, flat interface settings and many other advantages.

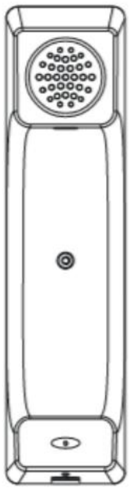
For enterprise users,while realizing environmental protection, they also provide convenient operation. Users can flexibly configure and define the functions of two DSS keys, space saving and cost. It will be an ideal choice for enterprise users and family users who pursue the high quality and high efficiency.

In order to help some interested users better understand the details of the product, this user manual can be used as a reference guide for the use of H5W series. This document may not be applicable to the latest version of the software. If you have any questions, you can use the help prompt interface of the device phone, or download and update your user manual from the official website.

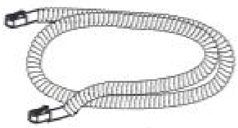
4.2 H1 Packing Contents



IP Phone

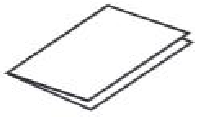


Handset

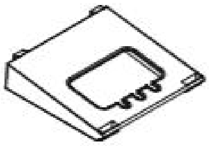


Handset Cord

Ethernet Cable

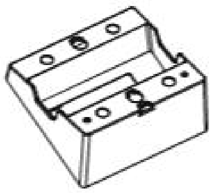


Quick Installation Guide



Stand

Power Adapter
(Optional)



Wall Stand
(Buy separately)

5 Desktop Installation

5.1 PoE and the use of external power adapters

The devices support two power supply modes from external power adapter or over Ethernet (PoE) complied switch.

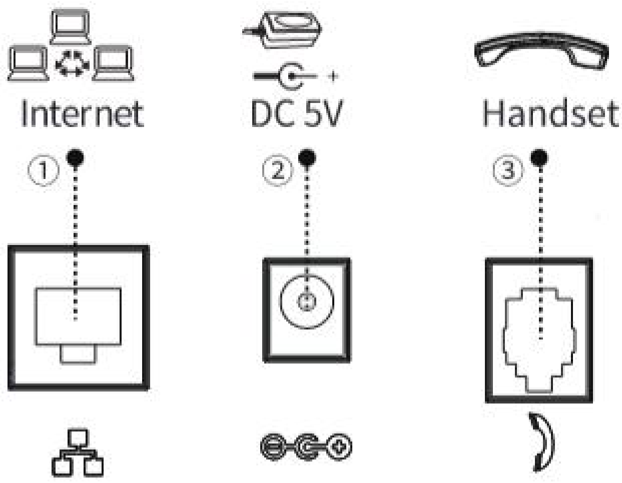
PoE power supply saves the space and cost of providing the device additional power outlet. With a PoE switch, the device can be powered through a single Ethernet cable which is also used for data transmission. By attaching UPS system to PoE switch, the device can keep working at power outage just like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

5.2 Desktop and wall mounted method

Please connect the power adapter, network, and phone to the corresponding ports in the description below.



picture 1-Connecting to the Device

Table 1 - Hardware Interface Description

Index	Interface	Description
①	Network port	Connect LAN or Internet
②	Power port	Connect the power adapter
③	Handset port	Connect IP Phone handset

6 Appendix Table

6.1 Appendix I - Icon

Table 2 - Keypad Icons

Icon	Instruction
	Redial
	MWI
	Hands-free (HF) speaker
	Mute Microphone (During Call)
	Volume down
	Volume up
	Hold
	Transfer

6.2 Appendix II - Keyboard character query table

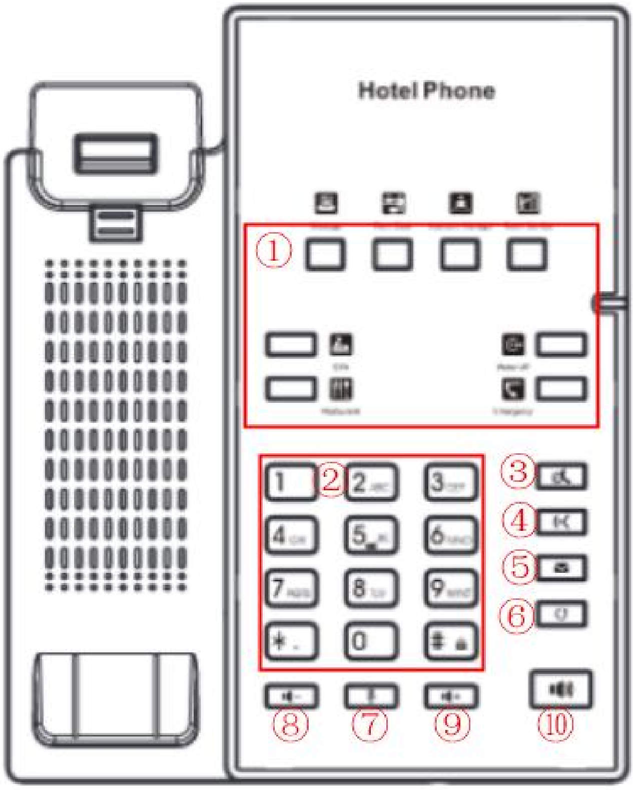
Table 3 - Look-up Table of Characters

Mode Icon	Text Mode	Key Button	Characters Of Each Press
123	Numeric	1	1
		2	2
		3	3
		4	4
		5	5
		6	6
		7	7
		8	8
		9	9
		0	0
		*	*.+
		#	#
abc	Lower Case Alphabets	1	@:;()<>
		2	a b c
		3	d e f
		4	g h i

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

7 Introduction to the User

7.1 H1 Key description



picture 2-Instruction of Keypad

Table 4 - H1 Instruction of Keypad

Number	The keypad names	Instruction
①	Softkey	These 8 buttons provide the function that corresponds to the website
②	Standard Telephone Keys	The 12 standard telephone keys provide the same function as standard telephones, but further to the standard function, some keys also provide special function by long-pressing the key,
③	Hold Key	Press the "Hold" key during the call, the user can hold the call, and press it again to cancel the holding and restore the normal call state.
④	Transfer Key	Press the "Transfer" button, the user can transfer the current call to other numbers.

⑤	Voice Mail	Press the Voice Mail button to listen to the voice mail
⑥	Redial	Press the Redial key to redial the last number dialed
⑦	Mute Key	During a call, the user can press this key to mute the microphone.
⑧	Volume Down Key	In the standby or ringing state, press this button to reduce the ringing volume; Press this button to lower the volume on the call.
⑨	Volume Up Key	In the standby or ringing state, press this button to increase the ringing volume; Press this button to increase the volume on the call.
⑩	Hands-free Key	The user can press this key to open the audio channel of the speakerphone.

7.2 Using Handset / Hands-free Speaker

■ **Using Handset**

About the use of the handle, the user can pick up the handle to dial the number, press the "#" button after pressing the number, and the number will be dialed. Users can switch audio channels of the phone by pressing the hands-free button.

■ **Using Hands-free Speaker**

For the use of the speakerphone, the user can dial the number by pressing the speakerphone button, or by dialing the number and then pressing the speakerphone button. When the voice channel of the handle is opened, the user can switch the audio channel of the phone by pressing the button of the hands-free speaker.

7.3 Making Phone Calls

■ **Default Line**

The device provides two line services (1 main line and 1 standby line). if both lines are configured successfully, the user uses line 1 to make or receive calls by default.

■ **Dialing Methods**

Users can dial a number in the following ways:

➢ The Device end

- Dial directly, pick up the handle and input the number, then press "#" to call out
- Redialing the last dialed number（Redial）

■ **Dialing Number then Opening Audio**

To make a phone call, user can firstly dial a number by one of the above methods. When the dialed number is completed, user can press [**Dial**] button on the soft-menu, or press hand-free button to turn on the speaker or headphone, or lift the handset to call out with the current line.

■ **Opening Audio then Dialing the Number**

Another alternative is the traditional way to firstly open the audio channel by lifting the handset, then turn on the hands-free speaker by pressing hands-free button, or line key, and then dial the number with one of the above methods. When completing the number dial, user can press [**Dial**] button to call out, or the number can also be dialed out automatically after timeout.

■ **Cancel Call**

While calling the number, user can stop the audio channel by putting back the handset or pressing the hands-free button to drop the call.

7.4 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 and open the web page of the phone firstly. H5W users can access the menu page and check the IP address of their phone by pressing the "#" key for more than 3 seconds.H3W users can voice dial the phone's IP address by long pressing the "#" key (3 seconds or more).

Picture 3 - Landing 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 configuration.

The device relies on IP network connection to provide service. Unlike traditional phone system based on a circuit switched wire technology, IP devices are connected to each other over the network and exchange data in packet basis based on the devices' IP address.

NOTICE! If the user long press # and hear "0.0.0.0", it means the network is disconnected. Please check the cable is connected correctly to the device and to the network switch, router, or modem.

The device supports three types of networks，IPv4/IPv6/IPv4&IPv6

There are three common IP configuration modes about IPv4

- Dynamic Host Configuration Protocol (DHCP) – This is the automatic configuration mode by getting network configurations from a DHCP server. Users don't need to configure any parameters manually. All configuration parameters will be getting from DHCP server and applied to the device. This is recommended for the most users.
- Static IP Configuration – This option allows user to configure each IP parameters manually, including IP Address, Subnet Mask, Default Gateway, and DNS servers. This is usually used in a technical environment of network users.
- PPPoE - This option is often used by users who connect the device to a broadband modem or router. To establish a PPPoE connection, user should configure username and password provided by the service provider.

The device is default configured in DHCP mode.

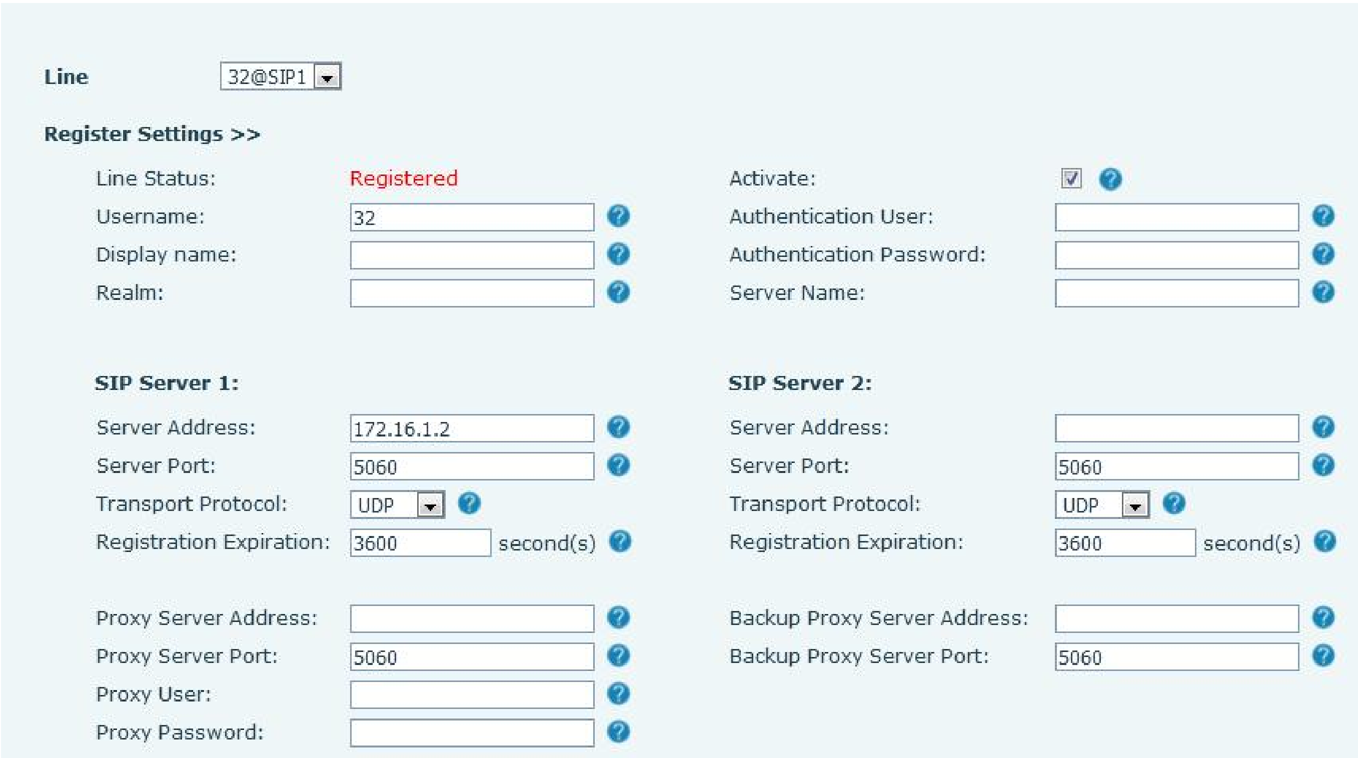
There are three common IP configuration modes about IPv6

- DHCP - This is the automatic configuration mode by getting network configurations from a DHCP server. Users need not to configure any parameters manually. All configuration parameters will be getting from DHCP server and applied to the device. This is recommended for most users.
- Static IP configuration - this option allows users to manually configure each IP parameter, including IP address, mask, gateway, and primary and secondary domains. This usually applies to some professional network user environments.

7.5 SIP Configurations

A line must be configured properly to be able to provide telephony service. The line configuration is like a virtualized SIM card on a mobile phone which stores the service provider and the account information used for registration and authentication. When the device is applied with the configuration, it 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 and registered port respectively, which are provided by the SIP server administrator.

- WEB interface: After logging into the phone page, enter [**Line**] >> [**SIP**] and select **SIP** for configuration, click apply to complete registration after configuration, as shown below:



Picture 4 - Web SIP registration

8 Basic Function

8.1 Answering Calls

When there is an incoming call while the device is idle, user will hear the ringing and see the power light blinking fast.

User can answer the call by lifting the handset, open speaker phone by pressing the hands-free button, or the [Answer] button. To reject the incoming call, user should press [**Reject**] button.

8.2 Make / Receive Second Call

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

- **Second Incoming Call**

When there is another incoming call during a phone call, this call will be waiting for user to answer. The device will not be ringing but playing a call waiting tone in the audio channel of the current call and the LED will be flashing in red. User can accept or reject the call as same as normal incoming call. When the waiting call is answered, the first call will be held on automatically.

- **Switching Between Two Calls**

When there are two calls established, user can press [Next Call] button and [**Resume**] button to switch between two calls.

- **Ending One Call**

User may hang up the current talking call by closing the audio channel or press [**End**] button. The device will return to single call mode in holding state.

8.3 End of the Call

After the user finishes the call, the user can put the handle back on the phone, press the hands-free button or press the [End] button to close the voice channel and end the call.

8.4 Redial

- Redial the last outgoing number:

  When the phone is in standby mode, press the redial button and the phone will call out the last number dialed.
- Call out any number with the redial key:

  Enter the number, press the redial key, and the phone will call out the number just entered.
- Clear the Redial record:

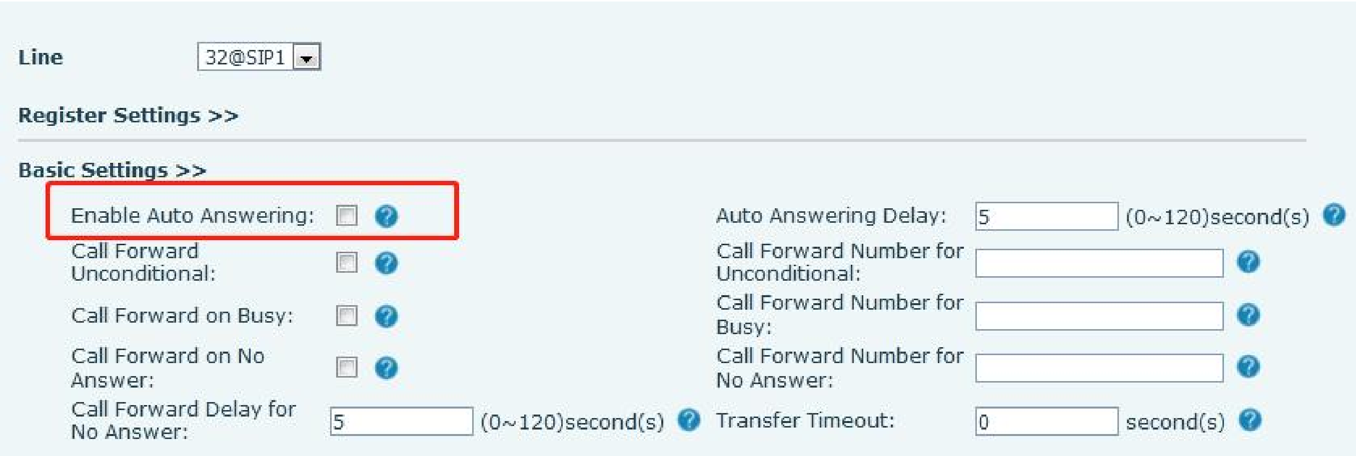
After the phone is used, redial will call out the last used number; therefore it is necessary to clear the records of the last customer so that they won't affect the use of other customers.

8.5 Auto-Answering

Users can enable the automatic answer function in the web page, the phone will be able to answer automatically after the call. Automatic answer can be enabled by line.

- **WEB interface:**

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



picture 5-Enable auto-answering

8.6 Mute

You can turn on the mute mode during a call which will turn off the microphone so that the local voice can not be heard. Normally, mute mode is automatically turned off at the end of a call. You can also turn on the mute mode anytime (such as idle status) and mute the ringtone automatically when there is an incoming call. Mute mode can be turned on in all call modes (handles or hands-free).

8.6.1 Mute the Call

- During the conversation, press the [Mute] button on the phone.

When the [Mute] button is pressed, the LED on the phone will be always red.

- Cancel mute: press the [Mute] button again to cancel mute on the phone.

The red LED is off.

8.6.2 Ringing Mute

- Mute: press the mute button when the phone is in standby mode.

  When there is an incoming call, the LED will turn red and slowly blink but the phone will not ring.
- Cancel ring tone mute: On the standby or incoming call screen, press the mute button again or [volume up] can cancel ring tone mute, and the red light is off .

8.7 Call Hold/Resume

The user can press the [**Hold**] button to maintain the current call, and this button will become the **[Resume**] button, and the user can press the **[Resume]** button to restore the call.

8.8 Call Transfer

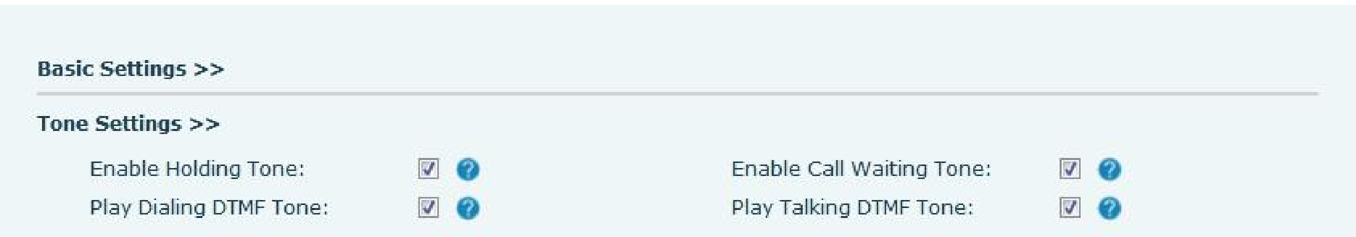
When the user is talking with a remote party and wish to transfer the call to another remote party.
During the call, the user presses transfer button on the phone, Enter the number to transfer or to press the contact button or the history button to select the number, press the transfer key again or blind transfer to a third party. After the third party rings, the phone will show that the transfer is successful and hang up.

8.9 Call Waiting

- Enable call waiting: new calls can be accepted during a call.
- Disable call waiting: new calls will be automatically rejected and a busy tone will be prompted.
- Enable call waiting tone: when you receive a new call on the line, the tone will beep.
- The user can enable/disable the call waiting function in the phone interface and the web interface.

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

picture 6-Web call waiting setting



picture 7-Web call waiting tone setting

8.10 Anonymous Call

8.10.1 Anonymous Call

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

- On the web page [**Line**] >> [**SIP**] >> [**Advanced Settings**] can open the mode of anonymous calls.
- Setting to enable anonymous calls also corresponds to the SIP line. That is, the setting under the SIP1 page can only take effect on the SIP1 line.

picture 8-Enable Anonymous web page call

8.10.2 Ban Anonymous Call

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

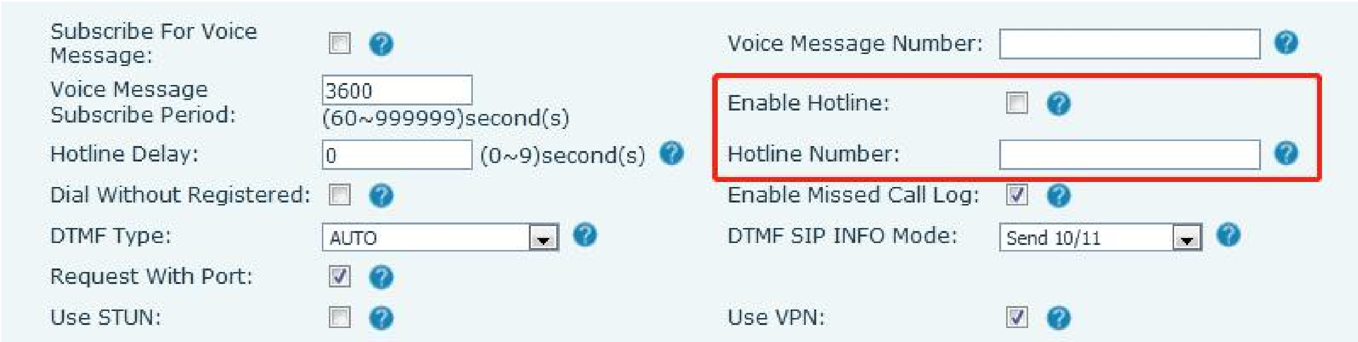
- On the web page [**Line**] >> [**SIP**] >> [**Advanced Settings**], also can disable anonymous calls.
- The setup to disable anonymous calls also corresponds to the SIP line. That is, the setting under the SIP1 page can only take effect on the SIP1 line.

picture 9-Page Settings blocking anonymous call

8.11 Hotline

The device supports hotline dialing. After setting up the hotline dialing, directly pick up the handset, hands-free etc., and the phone will automatically call according to the hotline delay time.

- On the website [**Line**] >> [**SIP**] >> [**Basic Settings**], can also set up a hotline.
- The setup hotline also corresponds to the SIP line. That is, the hotline set in the SIP1 webpage can only be activated in the SIP1 line.



picture 10-Hotline set up on webpage

9 Advance Function

9.1 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.

9.2 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. The user will be notified of the server voice message and the status of the power lamp.

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.

10 Web Configurations

10.1 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.

10.2 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)

10.3 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.

10.4 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 Data Tables**

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

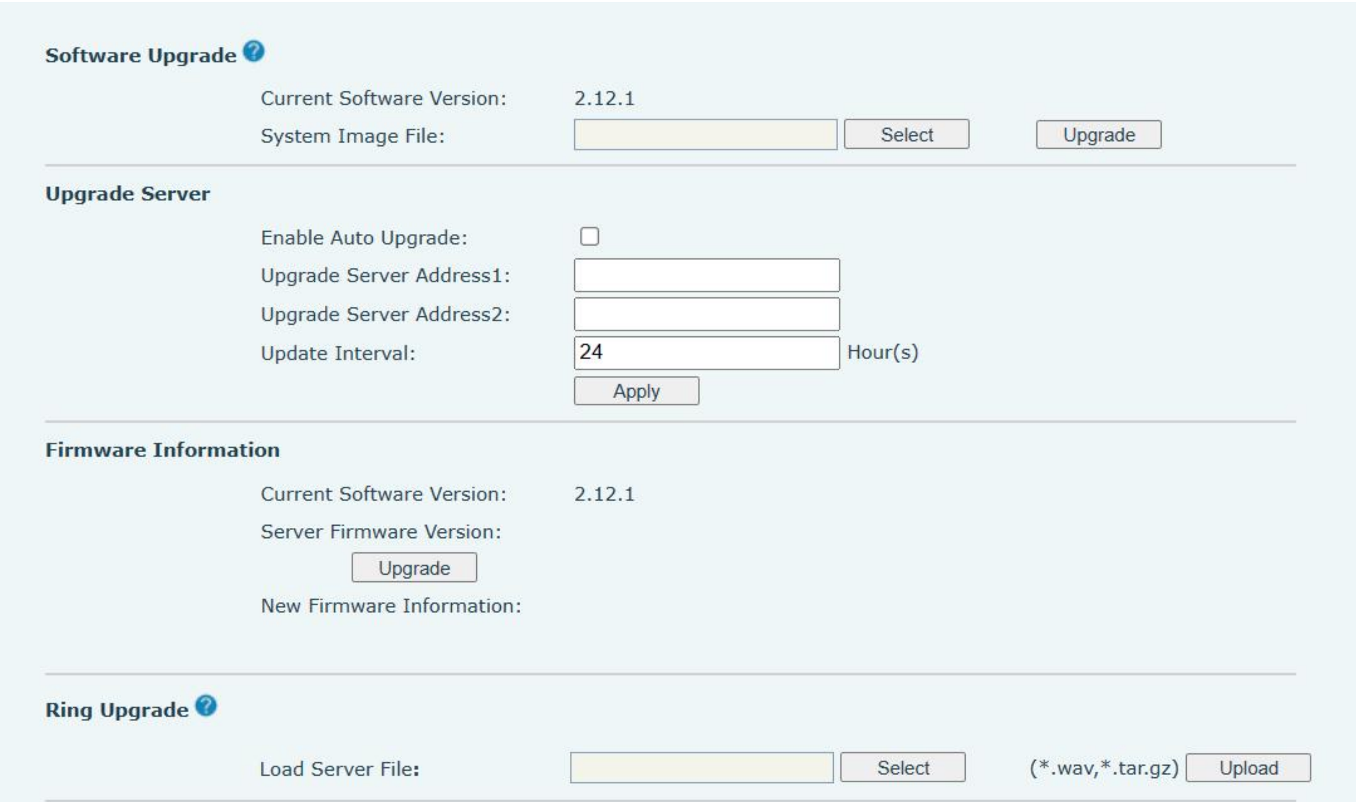
- **Reset Phone**

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

10.5 System >> Upgrade

Upgrade the phone software version, customize the ringtone, or delete the upgrade file.Ringtone support. Wav format.

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



picture 11-Web page firmware upgrade

Table 4 - Firmware upgrade

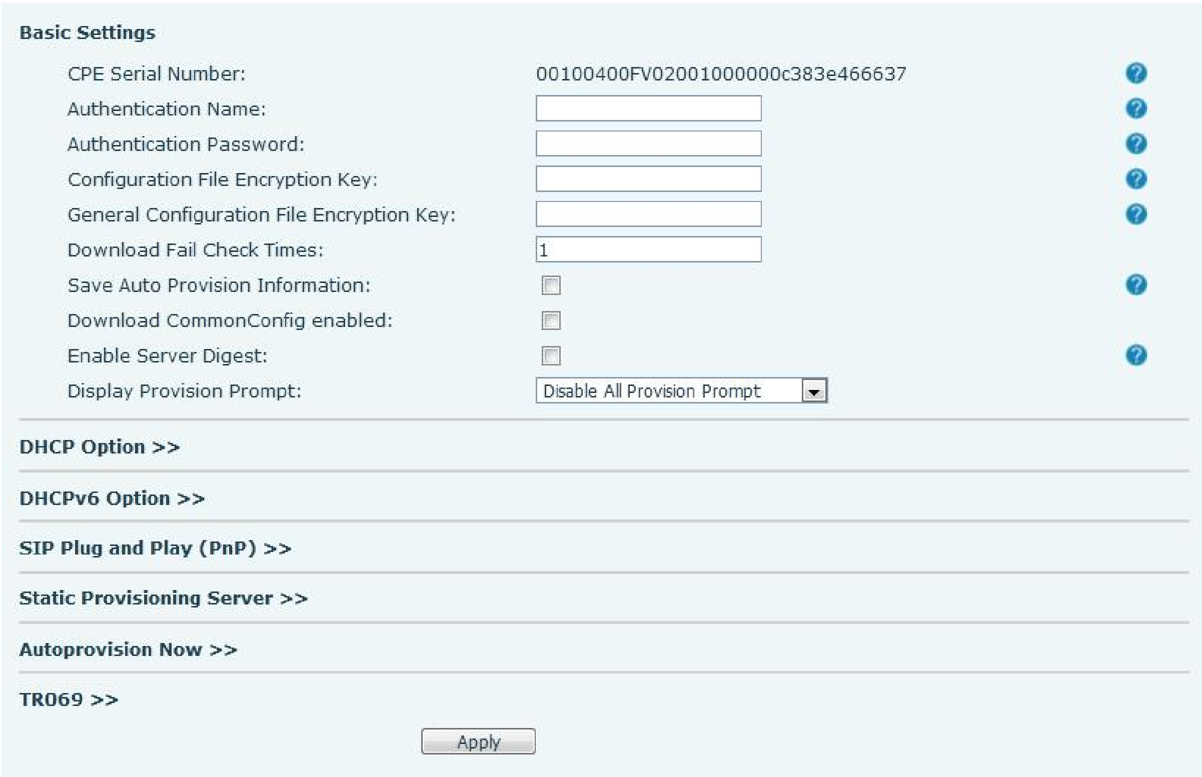
Parameter	Description
Upgrade server	
Enable Auto Upgrade	Enable automatic upgrade, If there is a new version txt and new software firmware on the server, phone will show a prompt upgrade message after Update Interval.
Upgrade Server Address1	Set available upgrade server address.
Upgrade Server Address2	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
- TXT file format must be UTF-8
- vendor_model_hw10.TXT The file format is as follows：
  Version=2.12.1 #Firmware
  Firmware=xxx/xxx.z #URL,Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers.
  BuildTime=2023.01.01 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.

10.6 System >> Auto Provision

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



picture 12-Page auto provision Settings

Fanvil devices support SIP PnP, DHCP options, Static provision, TR069. If all of the 4 methods are enabled, the priority from high to low as below:

PNP>DHCP>TR069> Static Provisioning

Transferring protocol: FTP, TFTP, HTTP, HTTPS

Details refer to **Fanvil Auto Provision**

Table 5 - 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.

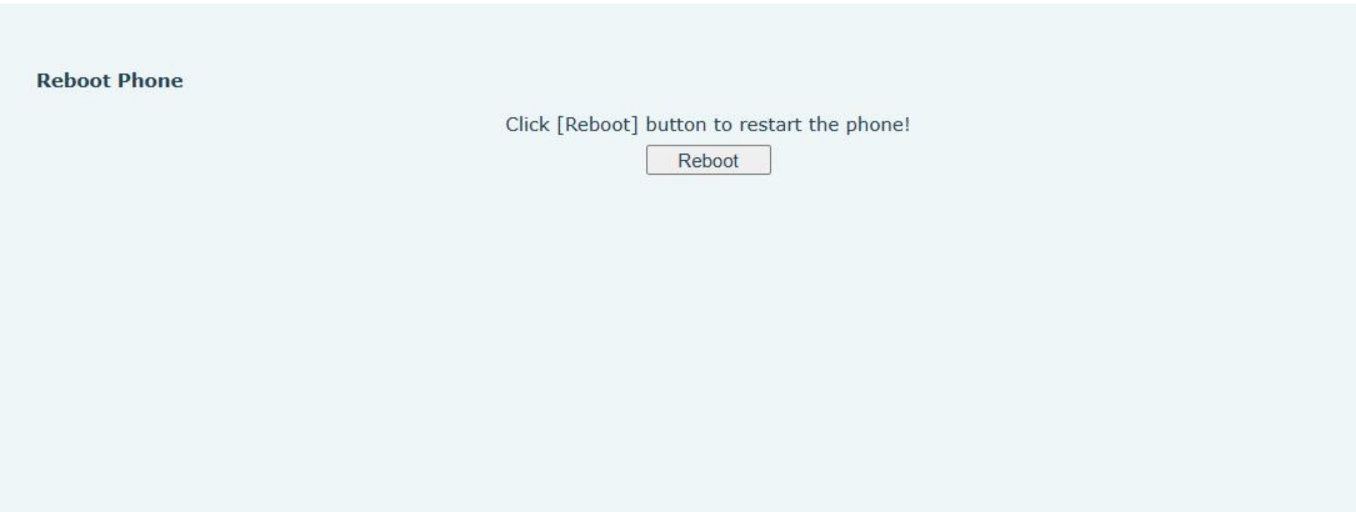
ACS Server URL	ACS server address
ACS User	ACS server username (up to is 59 character)
ACS Password	ACS server password (up to is 59 character)
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.
TLS Version	TLS version (TLS 1.0, TLS 1.1, TLS 1.2)
INFORM Sending 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

10.7 System >> Tools

Tools provided in this page help users to identify issues at trouble shooting. Please refer to 12 Trouble Shooting for more detail.

10.8 System >> Reboot Phone

This page can restart the phone.



picture 13-Web page reboot

11 Network

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

11.1 Network >> Basic

■ **IP Mode**

There are 3 network protocol mode options, IPv4, IPv6 and IPv4 & IPv6.

■ **IPv4**

In IPv4 mode, there are 3 connection mode options: DHCP, PPPoE and Static IP.

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

- Use DHCP DNS: It is enabled as default. “Enable” means phone will get DNS address from DHCP server and “disable” means not.
- Use DHCP time: It is disabled as default. “Enable” to manage the time of get DNS address from DHCP server and “disable” means not

When using PPPoE, phone will get the IP address from PPPoE server.

- Username: PPPoE user name.
- Password: PPPoE password.

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

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

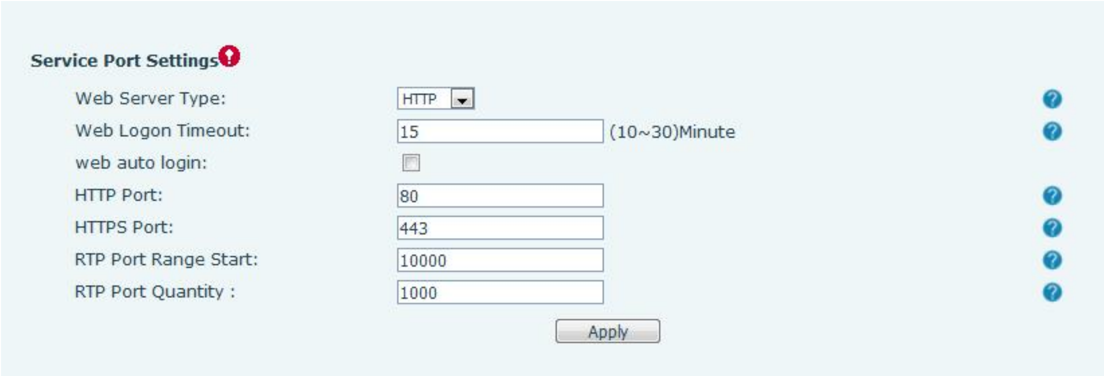
■ **IPv6**

In IPv6, there are 2 connection mode options, DHCP and Static IP.

- DHCP configuration refers to IPv4 introduction in last page.
- Static IP configuration is almost same as IPv4’s, except the IPv6 Prefix.
- IPv6 Prefix: IPv6 prefix, it is similar with mask of IPv4.

11.2 Network >> Service Port

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



picture 14-Service Port Settings

Table 6 - 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.

11.3 Network >> 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 webpage [**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 the 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 with the VPN automatically when the device boots up every time until user disable it. Sometimes, if the VPN connection does not establish 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**], select 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.

11.4 Network >> Advanced

- **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 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 7 - 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

11.5 Line >> SIP

Configure the Line service configuration on this page.

Table 8 - 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.
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	Enabling 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
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'
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 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
Session Timeout	Set the session timer timeout period
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.
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.
Unregister On Boot	Whether to enable logout function.
Enable MAC Header	When opening the registration, are IP package and user agent with MAC.
Enable Register MAC Header	When opening the registration, is user agent with MAC.
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.

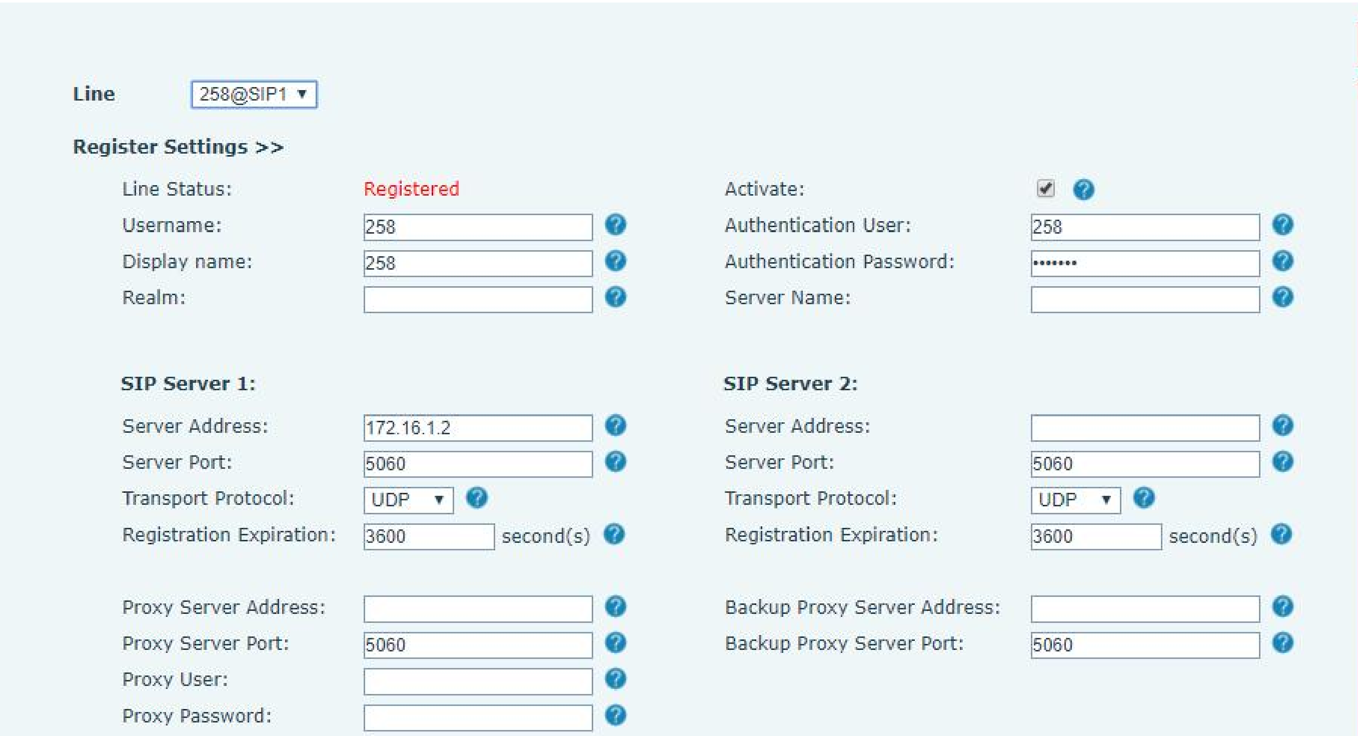
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Set the registration failure retry time.
Local SIP Port	Modify the phone SIP port.

11.6 Line >> 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.

The users can set functions as a SIP hotspot and other phones set (B and C) function as SIP hotspot clients. When somebody calls phone set A, phone sets A, B, and C all ring at the same time. 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.

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



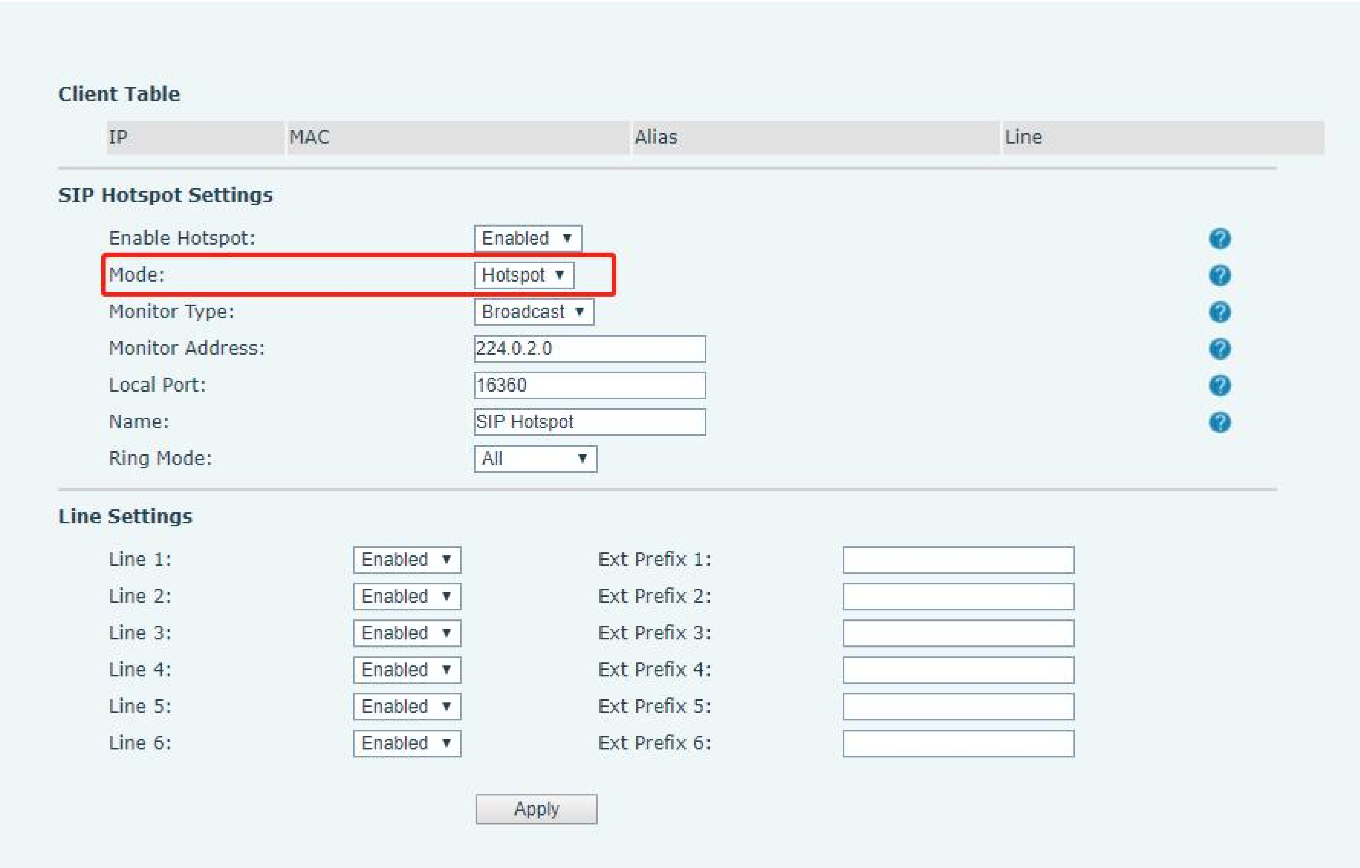
picture 15-SIP hotspot

Table 9 - 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.

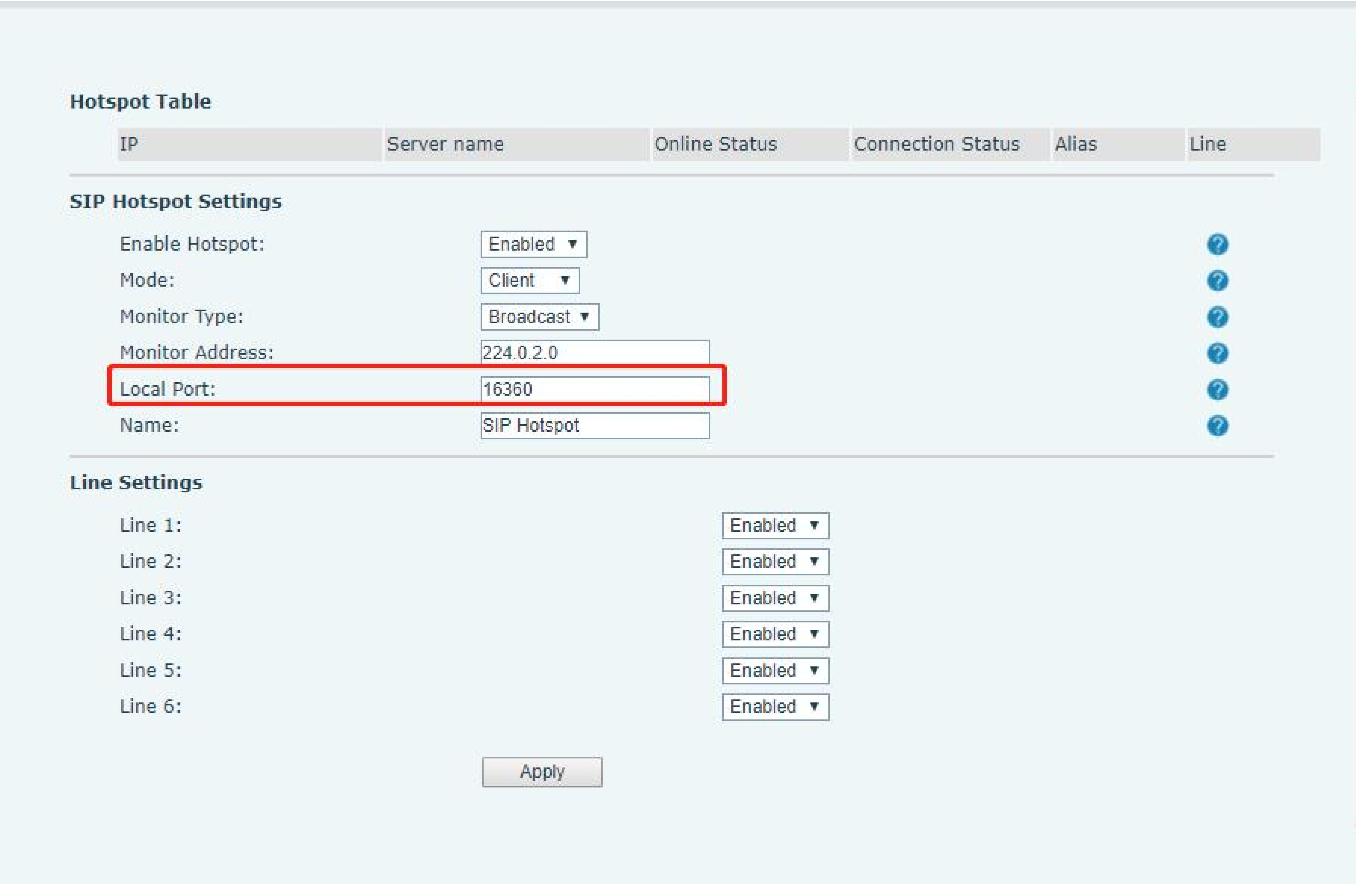
服务器端设置：



picture 16-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.



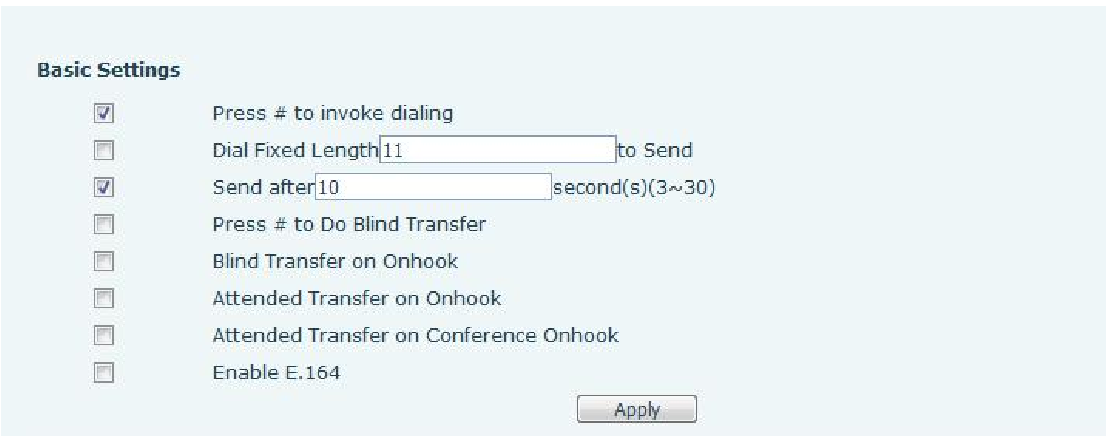
picture 17-SIP hotspot client configuration

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.

11.7 Line >> Dial Plan

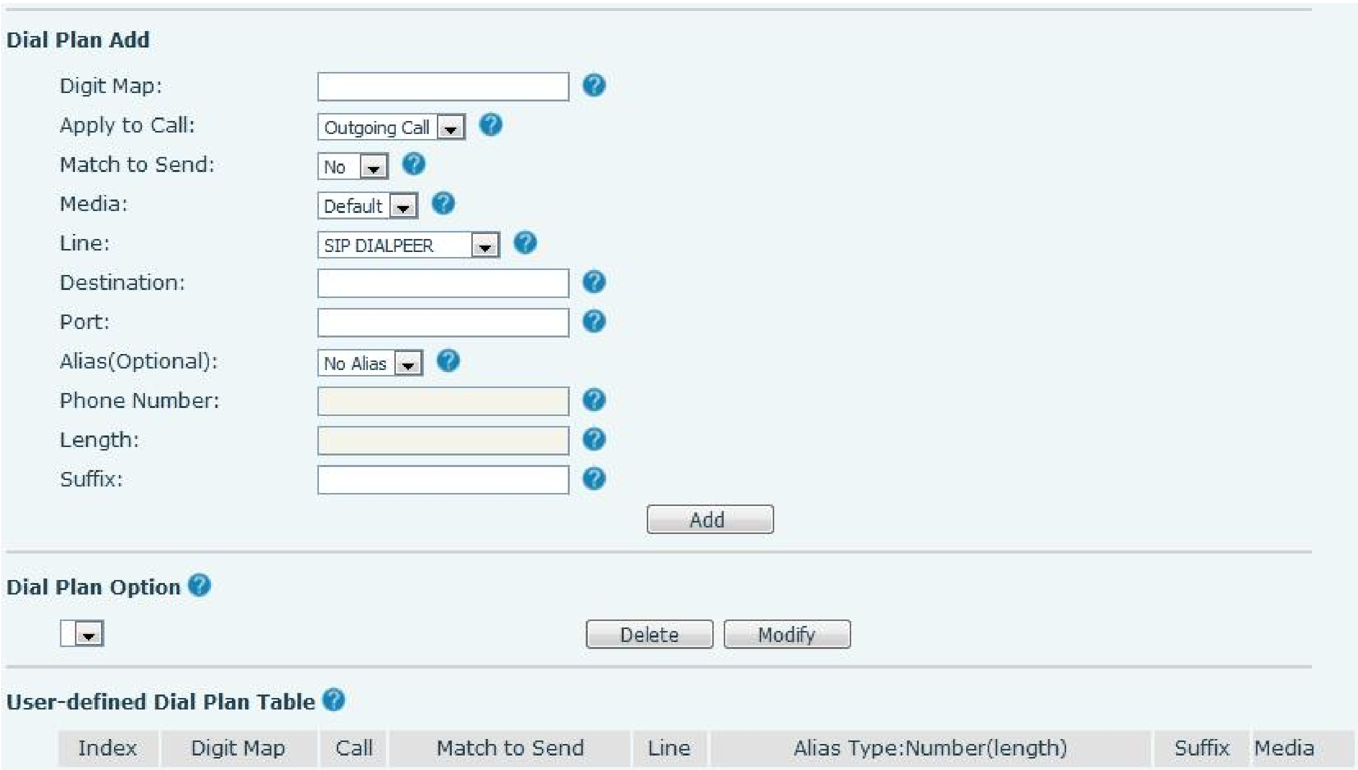


picture 18-Dial plan settings

Table 10 - Phone 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.
Enable E.164	Please refer to e. 164 standard specification

添加拨号规则：



picture 19-Custom setting of dial-up rules

Table 11 - Dial - up rule configuration table

Parameters	Description
Dial rule	There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules. In prefix matching, only part of the number is entered followed by T. The mapping with then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.
Note: Two different special characters are used. ■ x -- Matches any single digit that is dialed. ■ [ ] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.	
Destination	Set Destination address. This is for IP direct.
Port	Set the Signal port, and the default is 5060 for SIP.
Alias	Set the Alias. This is the text to be added, replaced or deleted. It is an optional item.
Note: There are four types of aliases. ■ all: xxx – xxx will replace the phone number. ■ add: xxx – xxx will be dialed before any phone number. ■ del –The characters will be deleted from the phone number. ■ rep: xxx – xxx will be substituted for the specified characters.	
Suffix	Characters to be added at the end of the phone number. It is an optional item.
Length	Set the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. It is an optional item.

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

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

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media
1	"123"	Out	No	SIP DIALPEER(172.16.1.15:5560)			Default

picture 20-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 21-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.

11.8 Line >> Action Plan

1. When a IP phone calls a phone, the bound IP camera synchronously transmits video to the other phone (video is supported)
2. When SIP calls, multicast calls or intercom calls are made, the device converts calls that conform to the number rules into group calls.

Table 12 - 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;Mcast Address (mcast://IP:port)
User Agent	Set user agent information

Details refer to **Fanvil Action Plan**

11.9 Line >> Basic Settings

Set up the register global configuration.

Table 13 - 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.

11.10 Line >> RTCP-XR

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

Table 14 - VQ RTCP-XR Settings

Parameters	Description
VQ RTCP-XR Settings	
VQ RTCP-XR Session Report	VQ report on whether session mode is enabled or not.
VQ RTCP-XR Interval Report	Whether to turn on Interval mode for VQ report sending.
Period for Interval Report(5~99)	The time interval at which VQ reports are sent periodically.
Warning threshold for Moslq(15~40)	When the phone calculated the Moslq value x10 below the set threshold, a warning was issued.
Critical threshold for Moslq(15~40)	When the phone calculates the Moslq value x10 below the set threshold, the critical report is issued.
Warning Threshold for Delay(10~2000)	When the one-way delay of the phone is greater than the set threshold, warning is issued.
Critical Threshold for Delay(10~2000)	When the phone computes that the one-way delay is greater than the set threshold, the critical report is issued.
Display Report Options on web	Whether to display the VQ report data for the last call through the web page.

11.11 Phone settings >> Features

Configuration phone features.

Table 15 - 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 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.
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
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.
Response Code Settings	
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 stand 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	

11.12 Phone settings >> Media Settings

Change voice Settings.

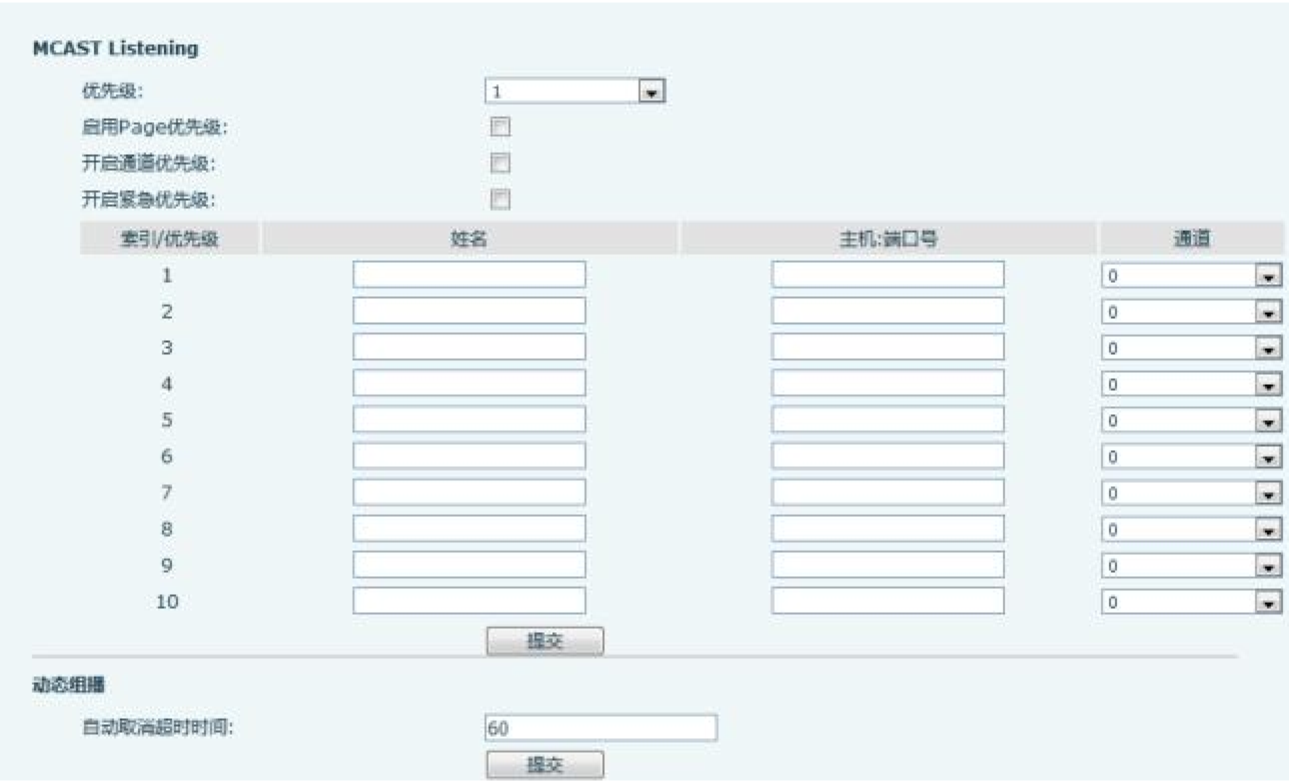
Table 16 - Voice settings

Parameters	**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
Audio Settings	
Handset Volume	Set the Handset volume, the value must be 1~9
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.
G.723.1 Bit Rate	5.3kb/s or 6.3kb/s is available.
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
AMR Payload Type	Set AMR load type, range 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 message, the phone will start a special dial tone.
Enable VAD	Whether voice activity detection is enabled.
Onhook Time	Configure a minimum response time, which defaults to 200ms
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

11.13 Phone settings >> MCAST

This feature allows user to make some kind of broadcast call to people who are in multicast group. User can configure a multicast 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 22-MCAST

Table 17 - 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.

11.14 Phone settings >> Action

Action URL

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

11.15 Phone settings >> Time/Date

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

Table 18 - 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

11.16 Phone Settings >> Time Plan

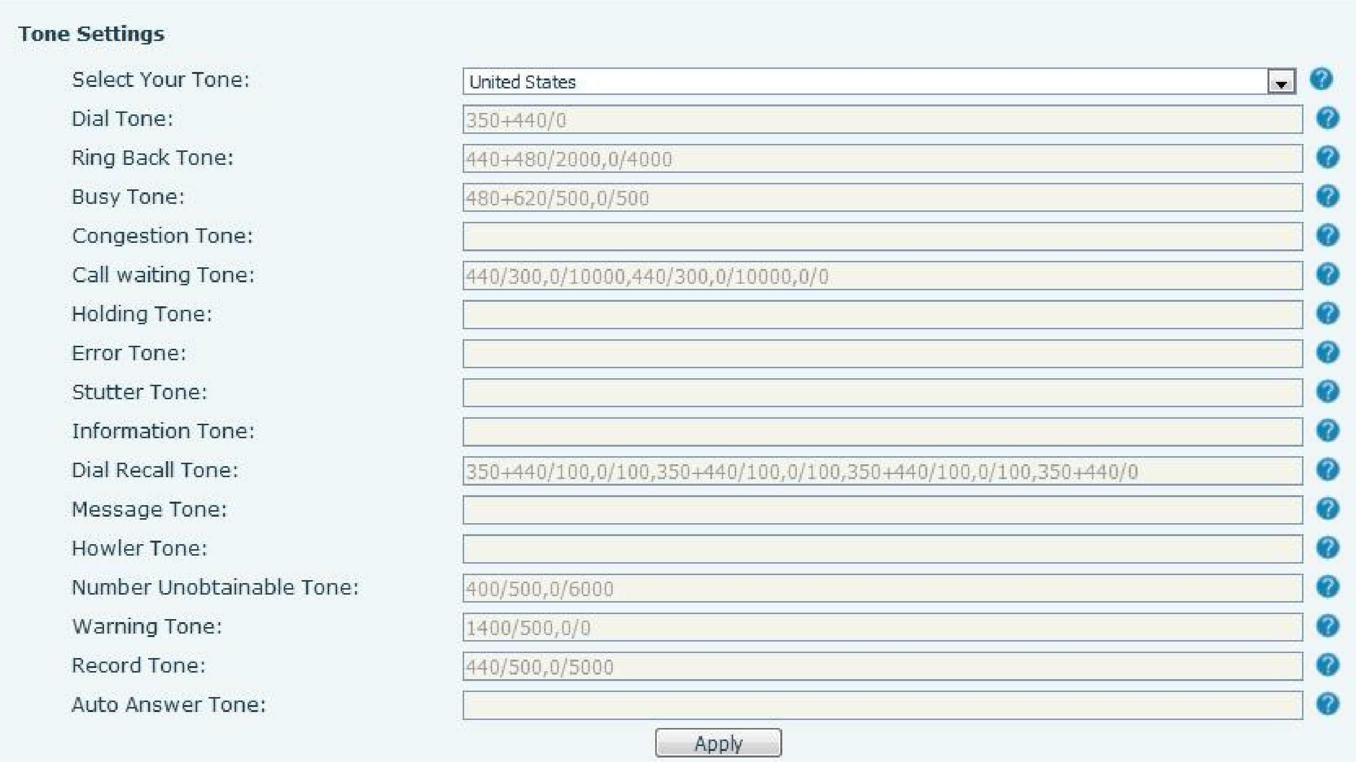
Details refer to **Fanvil Time Plan**

picture 23-Time Plan

11.17 Phone settings >> Tone

This page allows users to configure a phone prompt.

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



picture 24-Tone settings on the web

11.18 Call Log

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).

11.19 Function Key >> Softkey

The User Settings mode and display style, display page.

Table 19 - 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	Dial/None/Delete/Exit
Desktop	DSSkey1(Message)/DSSkey2(Front Desk)/DSSkey3(Assistant Manager)/DSSkey4(Room Server)/DSSkey5(SPA)/DSSkey6(WakeUp)/DSSkey7(Restaurant)/DSSkey8(

	Emergency)
Divert Dialed	Send/None/Delete/Exit
Ending	Redial/None/None/End
Predictive Dialer	Dial/None/Delete/Exit
Ringing	Prev call/Next call/Answer/Reject
Talking	Prev call/Next call/Hold/Release/None/End/None/None/
Transfer Alerting	End/None/None/Transfer(XFER)
Transfer Dialer	Delete/Transfer(XFER)/Dial/Exit
Trying	None/None/None/End
Waiting	Prev call/Next call/Release/End

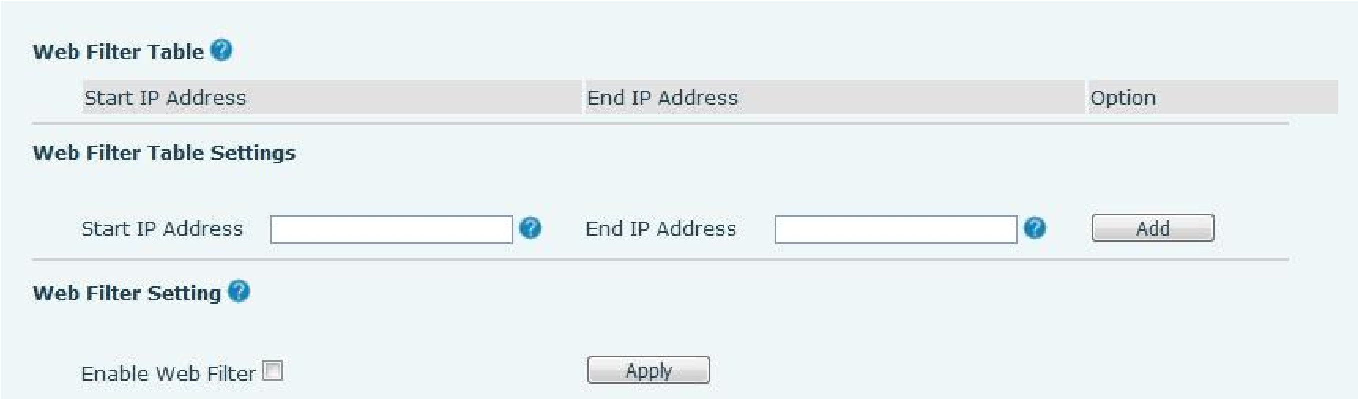
You can customize the configuration, Softkey functions and Settings on the web page.

Table 20 - Side Key configuration

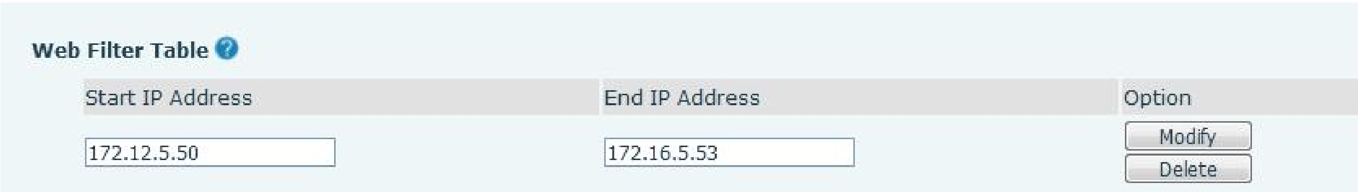
Parameters	Description
Memory Key	**Presence**: the Presence is able to view whether the user is online. Note: You cannot subscribe the same number for BLF and Presence at the same time **Speed Dial**: You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed. **Intercom**: This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments.
Line	It can be configured as a Line Key. User is able to make a call by pressing Line Key.
Key Event	User can select a key event as a shortcut to trigger. For example: Redail / END / 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.
XML browser	Users can set the DSS Key for specific URL download and other operations.

11.20 Security >> Web Filter

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



picture 25-Web Filter settings



picture 26-Web Filter Table

Adding and removing IP segments 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. If the user wants to delete, 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.

11.21 Security >> Trust Certificates

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

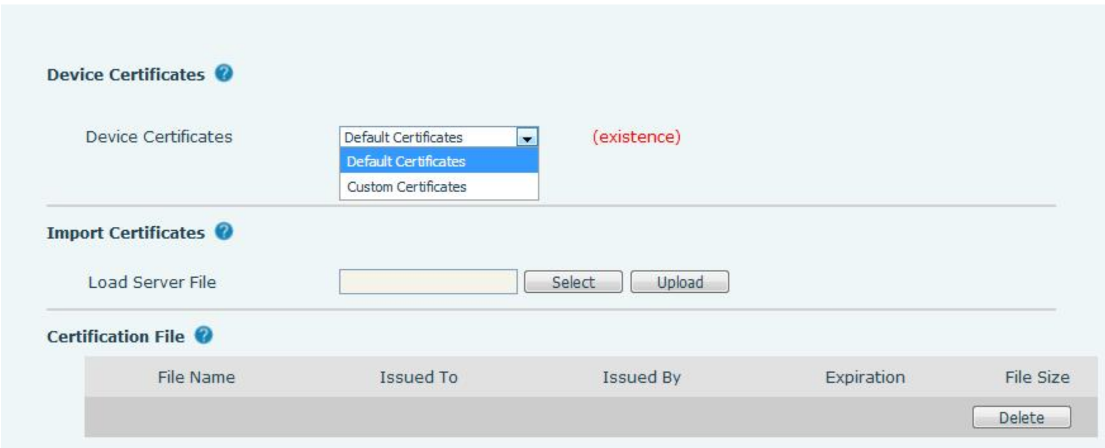
You can upload and delete uploaded certificates.

picture 27-Certificate of settings

11.22 Security >> Device Certificates

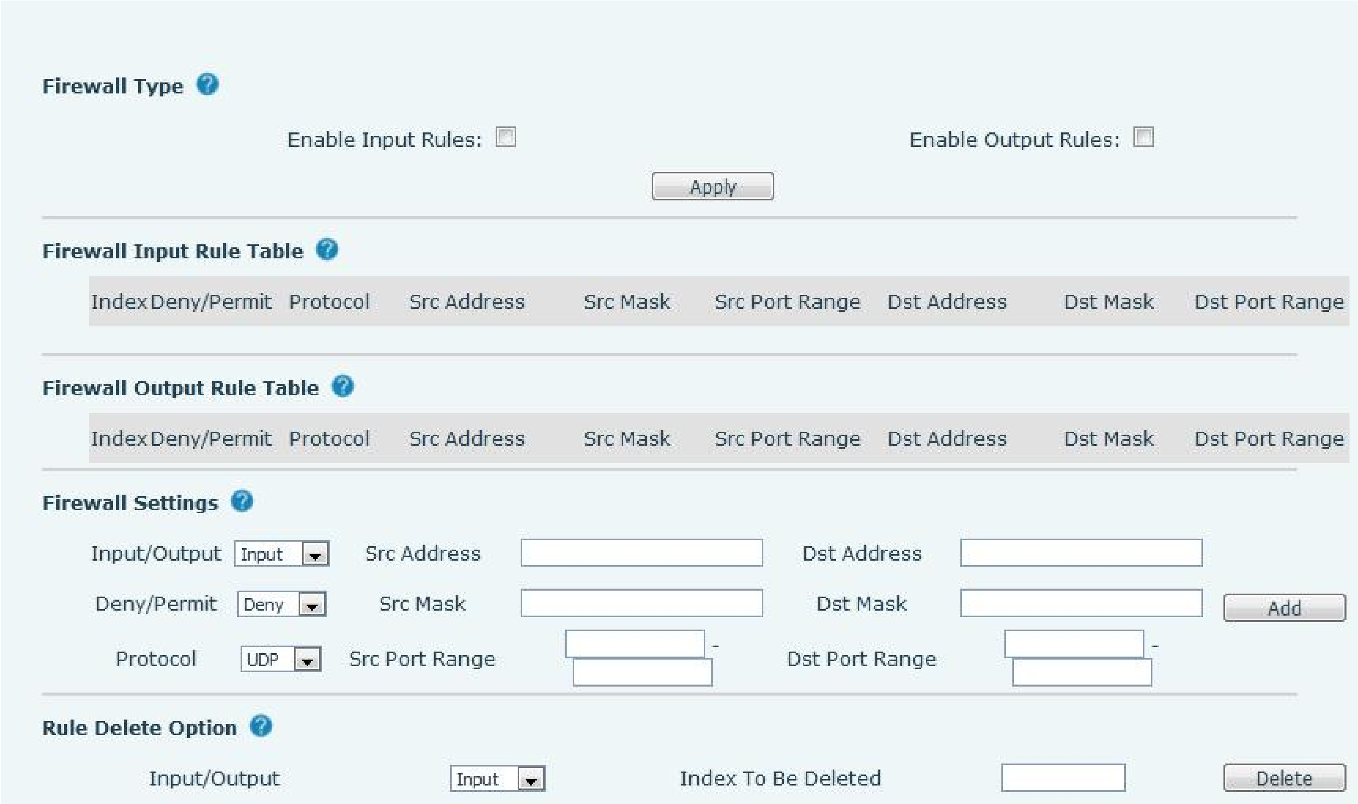
Select the device certificate as the default and custom certificate.

You can upload and delete uploaded certificates.



picture 28-Device certificate setting

11.23 Security >> Firewall



picture 29-Network firewall Settings

The user can set whether to enable the input through this page, output firewall and set the firewall input and output rules. Using these Settings can prevent some malicious network access, or restrict internal users access to some resources of the external network, which can improve security.

Firewall rule set is a simple firewall module. This feature supports two types of rules: input rules and output rules. Each rule is assigned an ordinal number, allowing up to 10 for each rule.

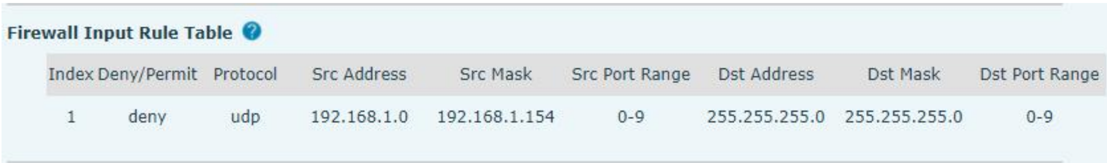
Considering the complexity of firewall Settings, the following is an example to illustrate:

Table 21 - Network Firewall

Parameter	Description
Enable Input Rules	Indicates that the input rule application is enabled.
Enable Output Rules	Indicates that the output rule application is enabled.
Input/Output	To select whether the currently added rule is an input or output rule.
Deny/Permit	To select whether the current rule configuration is disabled or allowed;
Protocol	There are four types of filtering protocols: TCP | UDP | ICMP | IP.
Src Port Range	Filter port range
Src Address	Source address can be host address, network address, or all addresses 0.0.0.0; It can also be a network address similar to *.*.*.0, such as: 192.168.1.0.

Dst Address	The destination address can be either the specific IP address or the full address 0.0.0.0; It can also be a network address similar to *.*.*.0, such as: 192.168.1.0.
Src Mask	Is the source address mask. When configured as 255.255.255.255, it means that the host is specific. When set as 255.255.255.0, it means that a network segment is filtered.
Dst Mask	Is the destination address mask. When configured as 255.255.255.255, it means the specific host. When set as 255.255.255.0, it means that a network segment is filtered.

After setting, click [**Add**] and a new item will be added in the firewall input rule, as shown in the figure below:


Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range
1	deny	udp	192.168.1.0	192.168.1.154	0-9	255.255.255.0	255.255.255.0	0-9

picture 30-Firewall Input rule table

Then select and click the button [**Apply**].

In this way, when the device is running: ping 192.168.1.118, the packet cannot be sent to 192.168.1.118 because the output rule is forbidden. However, the other IP of the ping 192.168.1.0 network segment can still receive the response packet from the destination host normally.

picture 31-Delete firewall rules

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

11.24 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 12.5 Get log information.

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 [**Network**] >> [**Phone**] 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 use the webpage **[system**] >> [**Reboot System**] and press [**ok**], or simply remove the power supply and restore it again.

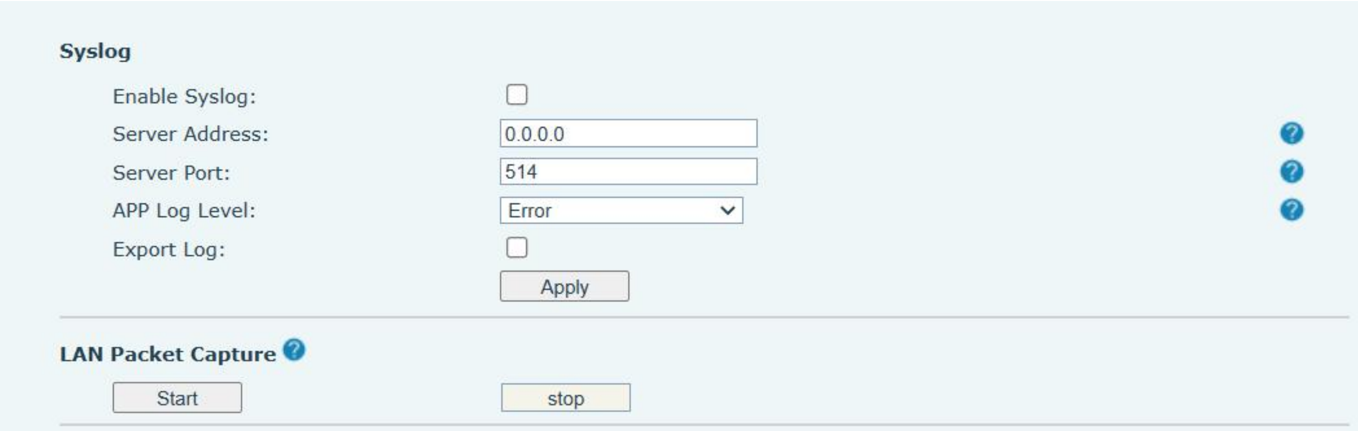
12.3 Reset Device to Factory Default

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

User restore factory reset press **[system]** >> **[Configuration]** >> **[Reset Phone]** and press **[reset]**.The phone will revert to the factory default state.

12.4 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 the relevant operations such as activating/deactivating 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.

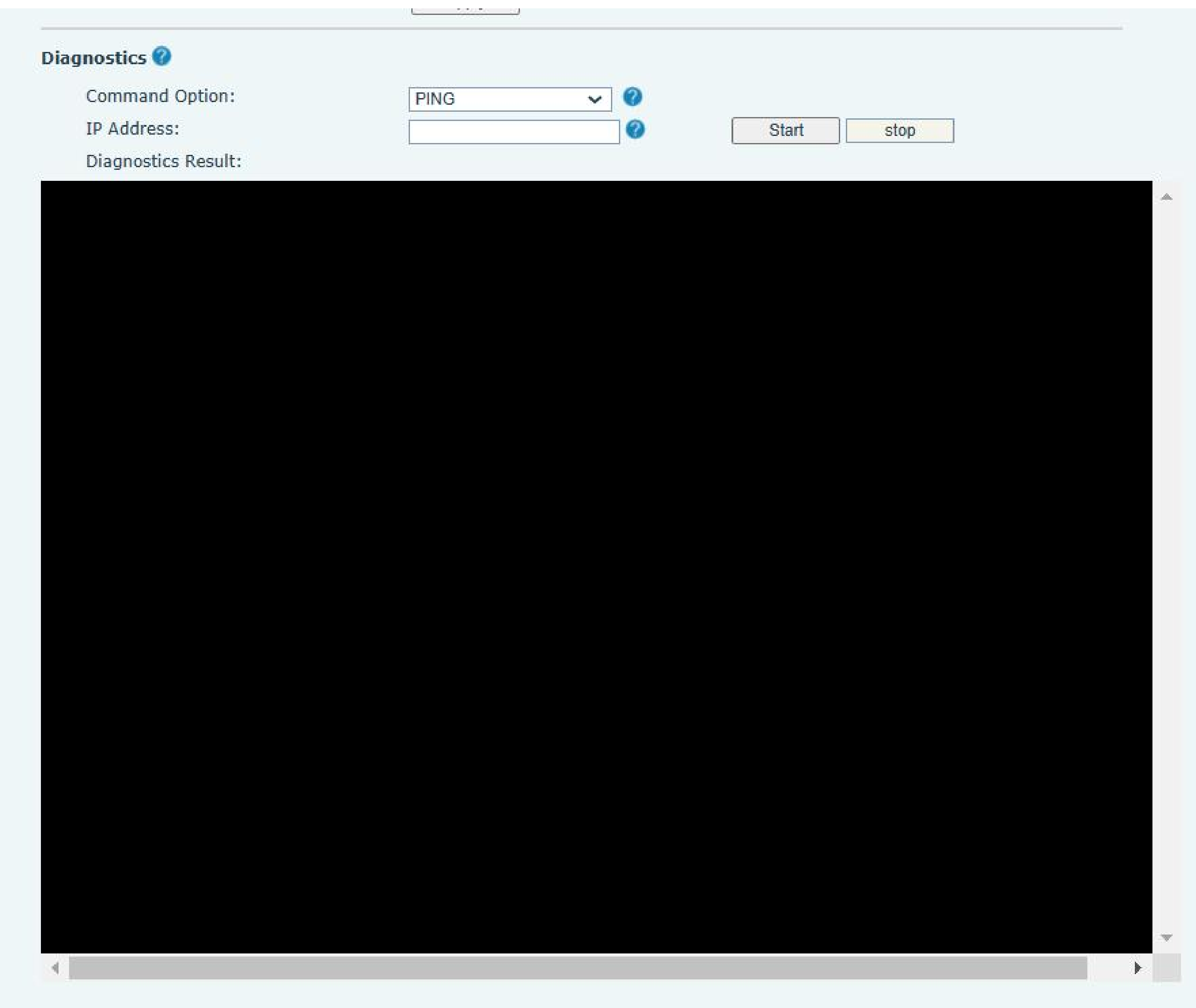


picture 32-Web capture

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

12.5 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 [**Diagnostics**], 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.



picture 33-Diagnostics

12.6 Common Trouble Cases

Table 22 - Trouble Cases

Trouble Case	Solution
Device could not boot up	1. The device is powered by external power supply via power adapter or PoE switch. Please use standard power adapter provided by manufacturer or PoE switch met with the specification requirements and check if device is well connected to power source.
Device could not register to a service provider	1. Please check if device is well connected to the network. The network Ethernet cable should be connected to the [Network] port NOT the [PC] port. 2. Please check if the device has an IP address. Long press #. If the

	device broadcast “0.0.0.0”, the device does not have an IP address. Please check if the network configuration is correct. 3. If network connection is fine, please check your line configurations again. If all configurations are correct, please kindly contact your service provider to get support, or follow the instructions in “12.4 Network Packet Capture” to get the network packet capture of registration process and send it to manufacturer support to analyze the issue.
No Audio or Poor Audio in Handset	1. Please check if Handset is connected to the correct port. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Audio is chopping at far-end in Hands-free speaker mode	This is usually due to loud volume feedback from speaker to microphone. Please lower down the speaker volume a little bit, the chopping will be gone.