



User Manual

H2U-V3

Software Version: 2.20.0.31

Release Date: 2026/9/16

Directory

Directory	2
1 Picture	4
2 Table	5
3 Safety Instruction	6
4 Overview	7
4.1 Overview	7
4.2 Packing Contents	8
5 Desktop Installation	9
5.1 PoE And the use of external power adapters	9
5.2 Wall mounted installation method	10
6 Appendix Table	12
6.1 Appendix I –LED Definition	12
7 Introduction to the User	13
7. 1 Instruction of Keypad	13
7.2 Using Handset / Hands-free Speaker	14
8 Basic Function	15
8.1 Making Phone Calls	15
8.2 Answering Calls	15
8.3 End of the Call	15
8.4 Redial	16
8.5 Auto-Answering	16
8.6 Mute	16
8.6.1 Mute the Call	16
8.6.2 Ringing Mute	17
8.7 Call Hold/Resume	17
8.8 Hotline	17
9 Advance Function	18
9.1 Intercom	18
9.2 MCAST	18
9.3 MWI (Message Waiting Indicator)	19
10 Phone Settings	20
10.1 Basic Settings	20
10.1.1 Language	20

10.2 Function Key	20
11 Web Configurations	22
11.1 Web Page Authentication	22
11.2 System Information	23
11.3 Config Import/Export	24
11.4 Web Firmware Upgrade	24
11.5 Web Ringtone Update	25
11.6 Basic Network Configuration	26
11.7 SIP Registration	27
11.8 Reboot the Phone	28
11.9 Restore Factory Settings	29
11.10 System >> Auto Provision	29
11.11 System >> Tool	31
11.12 Network >> Service Port	32
11.13 Network >> Advanced	33
11.14 Line >> SIP	34
11.15 Line >> Dial Plan	40
11.16 Line >> Basic Settings	42
11.17 Phone settings >> Features	43
11.18 Phone settings >> Media Settings	46
11.19 Phone settings >> MCAST	47
11.20 Phone settings >> Action	48
11.21 Phone settings >> Time/Date	48
11.22 Phone settings >> Tone	49
11.23 Phonebook >> Call List	49
11.24 Function Key >> Function Key	50
11.25 Function Key >> Speed Dial List	50
12 Trouble Shooting	51
12.1 Get Device System Information	51
12.2 Reboot Device	51
12.3 Reset Device to Factory Default	51
12.4 Network Packets Capture	51
12.5 One-click Export Debug Info	52
12.6 Common Trouble Cases	53

1 Picture

Picture 1	- Device installation	10
Picture 2	- Connecting to the Device	11
Picture 3	- Instruction of Keypad	13
Picture 4	- Language setting on Web page	20
Picture 5	- Web Login Page	22
Picture 6	- System Information Page	23
Picture 7	- Import/Export Settings Page	24
Picture 8	- Web Firmware Update Page	25
Picture 9	- Web Ringtone Update	25
Picture 10	- Network Configuration Page	27
Picture 11	- Web Line Registration	28
Picture 12	- Web Reboot	28
Picture 13	- Web Restore Factory Settings	29
Picture 14	- Auto Provision settings	30
Picture 15	- System Tool	32
Picture 16	- Network Advanced	33
Picture 17	- Web Speed Dial List	50
Picture 18	- Web capture	52
Picture 19	- Web One-click Export Debug Info	52

2 Table

Table 1	- DSS KEY LED State	12
Table 2	- Instruction of Keypad	13
Table 3	- Intercom configure	18
Table 4	- Multicast Parameters	18
Table 5	- Auto Provision	30
Table 6	- Service port	32
Table 7	- QoS & VLAN	34
Table 8	- Line Configuration in the Configuration File	34
Table 9	- Phone 4 dialing methods	40
Table 10	- Dial - up rule configuration table	41
Table 11	- Set the line global configuration on the web page	42
Table 12	- General function Settings	43
Table 13	- Voice settings	46
Table 14	- Multicast parameters	47
Table 15	- Time&Date settings	48
Table 16	- Function Key configuration	50
Table 17	- Trouble Cases	53

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 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.
- Avoid exposure the phone to high temperature or below 0°C or high humidity.
- Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

4 Overview

4.1 Overview

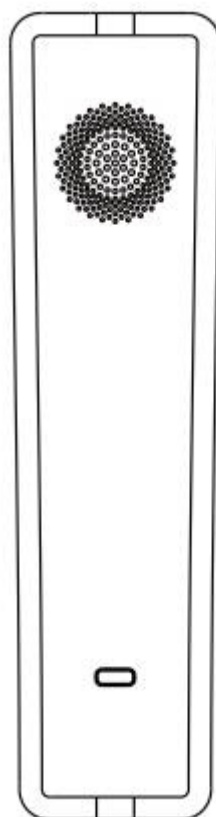
H2U-V3 is a network telephone specially designed for hotels. The simple design of the device brings excellent user experience for users. The equipment is not only a telephone, but also a masterpiece placed in the living room or office.

H2U-V3 is the latest generation of network telephone designed for the hotel, which still continues the excellent performance and specifications of traditional equipment; such as high-definition voice, high-performance echo cancellation, 100M Ethernet, QoS, encrypted transmission, automatic configuration, etc.; new system, smooth operation, flat interface setting and many other advantages.

For enterprise users, the equipment is a cost-effective office equipment, while realizing environmental protection, it also provides convenient operation; For home users, the device is a highly efficient communication device. Users can flexibly configure and define the functions of one DSS keys, saving space and cost. It will be an ideal choice for enterprise users and home users who pursue 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 H2U-V3. 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 H2U-V3 phone, or download and update your user manual from the office website.

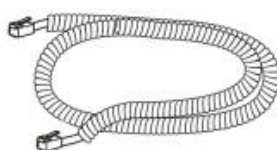
4.2 Packing Contents



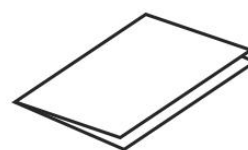
IP Phone



Handset



Handset Cord



QIG

5 Desktop Installation

5.1 PoE And the use of external power adapters

The device 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.

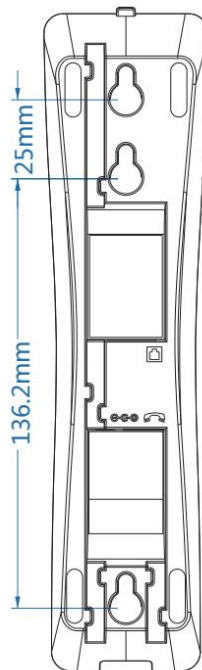
Please use the power adapter supplied by Fanvil and the PoE switch met the specifications to ensure the device work properly.

5.2 Wall mounted installation method

The device supports wall mounted.

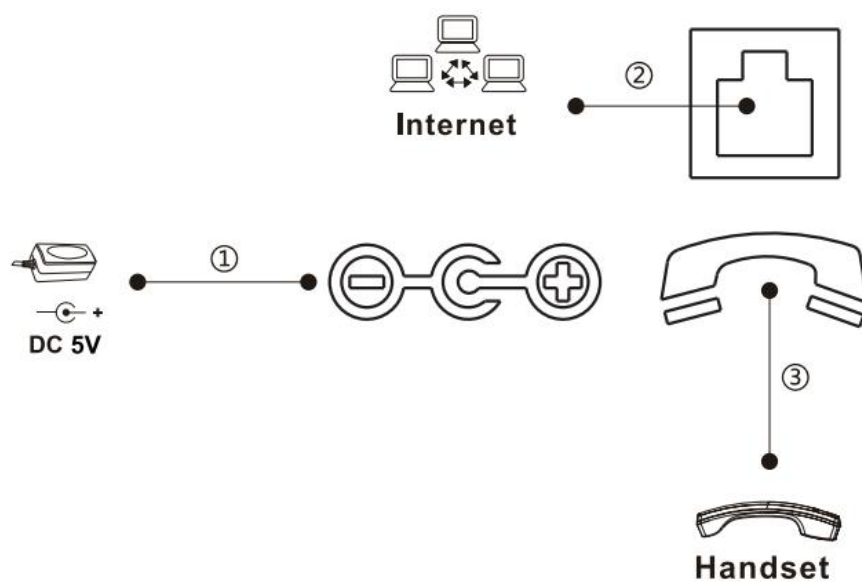
Please follow the instructions in the picture below to install the phone:

- 1) Drill two holes in the wall with a vertical distance of 136.2 or 161.2mm.
- 2) Insert two rubber plugs and screws in turn. Note that 5mm is reserved between the nut and the wall, which is convenient for hanging the phone base.
- 3) Connect the cable, handle cable and power supply.
- 4) Align the wall hole on the base with the screws in step 2 and slide down to complete the installation.



Picture 1 - Device installation

Connect the power adapter, network, PC, phone to the appropriate port as shown in the picture below.



Picture 2 - Connecting to the Device

6 Appendix Table

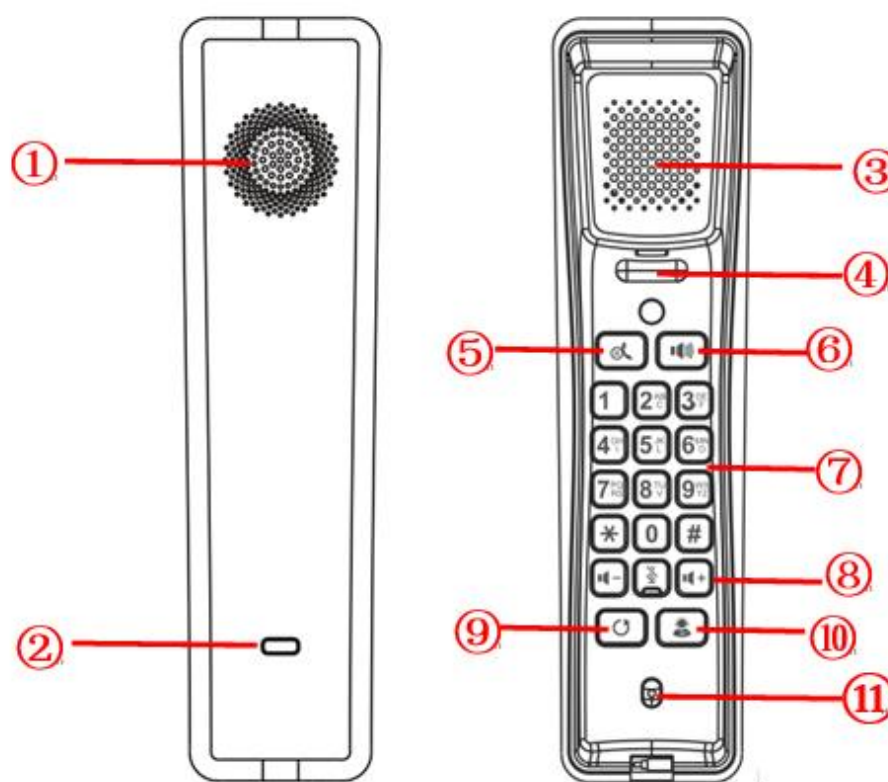
6.1 Appendix I –LED Definition

Table 1 - DSS KEY LED State

Type	LED Light	State
default standby	standby	Green On
	mute	Green slow flash
	Line error (Registration failure)/Network disconnection	Red slow flash
call	calling/Pick up the handle	Red On
	mute	Orange slow flash
	hold/held	Orange slow flash
	Ringing	Red flash

7 Introduction to the User

7.1 Instruction of Keypad



Picture 3 - Instruction of Keypad

The picture above shows the keypad layout of the phone. Each button provides its own specific function. Users can refer to the instructions for the keys in the illustration in this section to operate the phone.

Table 2 - Instruction of Keypad

Number	The keypad names	Instruction
①	Hands-free Speaker	The hands-free channel plays sound
②	Status indicator lamp	Power indication/line status indication
③	Handle the horn	The handle channel plays sound

④	Hook	Hang up the handle and hang up the phone
⑤	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.
⑥	Hands-free Key	The user can press this key to open the audio channel of the speakerphone
⑦	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, Key  - Long-pressed to broadcast IP (Default English) .
⑧	Volumes Key	The volume to add and subtract-In the standby state, ring and ring configuration interface, press this button to increase/reduce the ring volume; Press this button to increase/lower the volume on the call Mute Key-During a call, the user can press this key to mute the microphone.
⑨	Redial	Press the Redial key to redial the last number dialed
⑩	Function Key	User-defined functionality
⑪	Microphone	Listen when the receiver is answering (do not listen when the phone is hands-free)

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, 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.

8 Basic Function

8.1 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)
- The Web end
 - Dial from web fill in number dial
 - Selecting a phone number from call logs

■ Cancel Call

When calling a number, the user can cancel the call by putting back the handle/pressing down the spring.

8.2 Answering Calls

Users can answer the call by picking up the handle or pressing the speakerphone button to open the hands-free channel.

The telephone does not support multiple calls. When there is an established call, the user needs to hang up the current call before answering the second call.

8.3 End of the Call

When the call is over, the user can put the handle back on the phone and press the speakerphone button to 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 on the dial.
- Redial record clearing
After the phone is used, redial will default to the last used number; therefore, it is necessary to clear the records used by the last customer without affecting the use of other customers.

8.5 Auto-Answering

User may enable auto-answering feature on the device and any incoming call will be automatically answered. The auto-answering can be enabled on line basis.

8.6 Mute

You can turn on mute mode during a call and turn off the microphone so that the local voice is not heard. Normally, mute mode is automatically turned off at the end of a call.

You can also turn on mute on any screen (such as the free screen) 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:  the mute lamp is red and the power lamp is orange.

Cancel mute: press  cancel mute on the phone again. When the mute lamp goes out, the power lamp returns to its original state

8.6.2 Ringing Mute

- Mute: press the mute button when the phone is in standby mode: 
mute light red always bright, power lamp green flashing; There is no ringer for incoming calls.

Cancel ring tone mute: On the standby or incoming call screen, press the mute button again  or volume up  + cancel ring tone mute

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 Hotline

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

9 Advance Function

9.1 Intercom

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

Table 3 - Intercom configure

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

9.2 MCAST

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

Table 4 - Multicast Parameters

Parameter	Description
Priority	Define the priority for the current call. The available options are as follows: 1 - 10: Defines the priority for regular calls, with 1 being the highest and 10 being the lowest.
Enable Page Priority	The page priority determines how the phone handles newly received multicast RTP streams while currently engaged in a multicast session. When the page priority switch is enabled, the phone

	automatically ignores low-priority multicast RTP streams, accepts high-priority multicast RTP streams, and places the current multicast session on hold; if it is not enabled, the phone automatically ignores all received multicast RTP streams.
Name	Server name listening for multicast
Host:port	Server address listening for multicast: port number

Multicast Call:

- Set the multicast address and select the encoding
- Set the name, host, and port for receiving the multicast
- Press the DSSkey multicast button
- The recipient will receive the multicast call and automatically play the multicast

9.3 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.

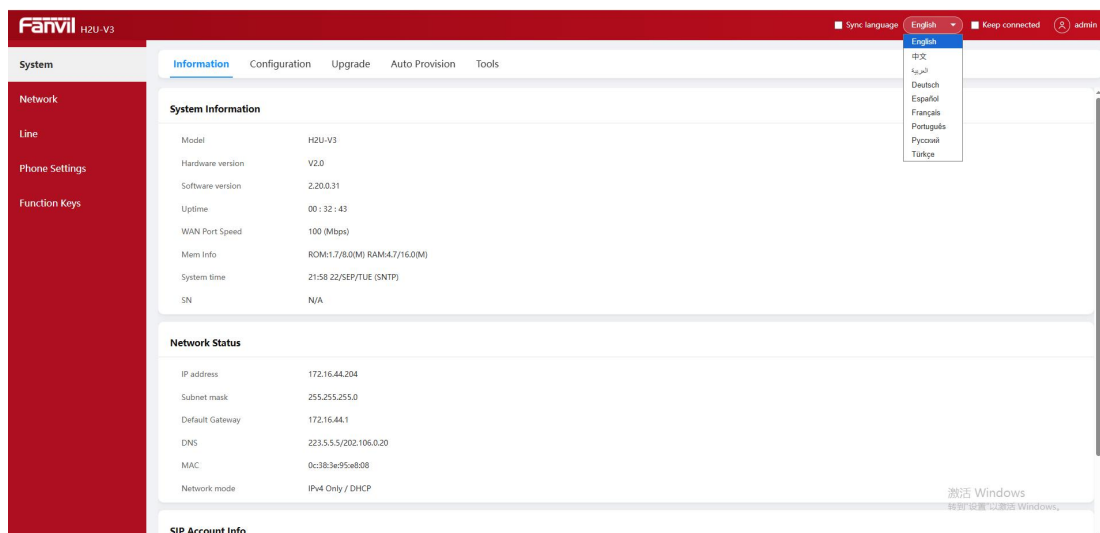
10 Phone Settings

10.1 Basic Settings

10.1.1 Language

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

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



Picture 4 - Language setting on Web page

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

10.2 Function Key

The device has a total of 11 configurable custom function keys; One direct call foreground key and 10 custom digital speed dial keys.

Device direct call key, default configuration as a fixed number; (customizable replacement)

0~9 numeric keys can be used as customized shortcut keys, users can customize the configuration of 0~9 numeric keys in the web page, users can quickly dial the corresponding number by long press each shortcut key.

The DSS Key could be configured as followings,

- ◆ Memory Key
 - Speed Dial/Intercom/Call Park/Call Forward
- ◆ Key Event
 - Call Transfer/Pickup/Join (Passive Invitation)
- ◆ DTMF
- ◆ MCAST Paging
- ◆ Action URL

More detailed information *refers to* [11.24 Function Key >> Function Key](#)

11 Web Configurations

11.1 Web Page Authentication

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

Users can manage and operate the phone via its web interface. To do so, users must first enter the phone's IP address in a web browser to open the phone's web interface. Users can press “#” to retrieve the phone's IP address.

Open a web browser, enter the phone's IP address, and log in to the phone's web interface. The first screen you'll see is the phone's login page.



Picture 5 - Web Login Page

Users must enter their username and password correctly to log in to the website. The default username and password are both "admin."

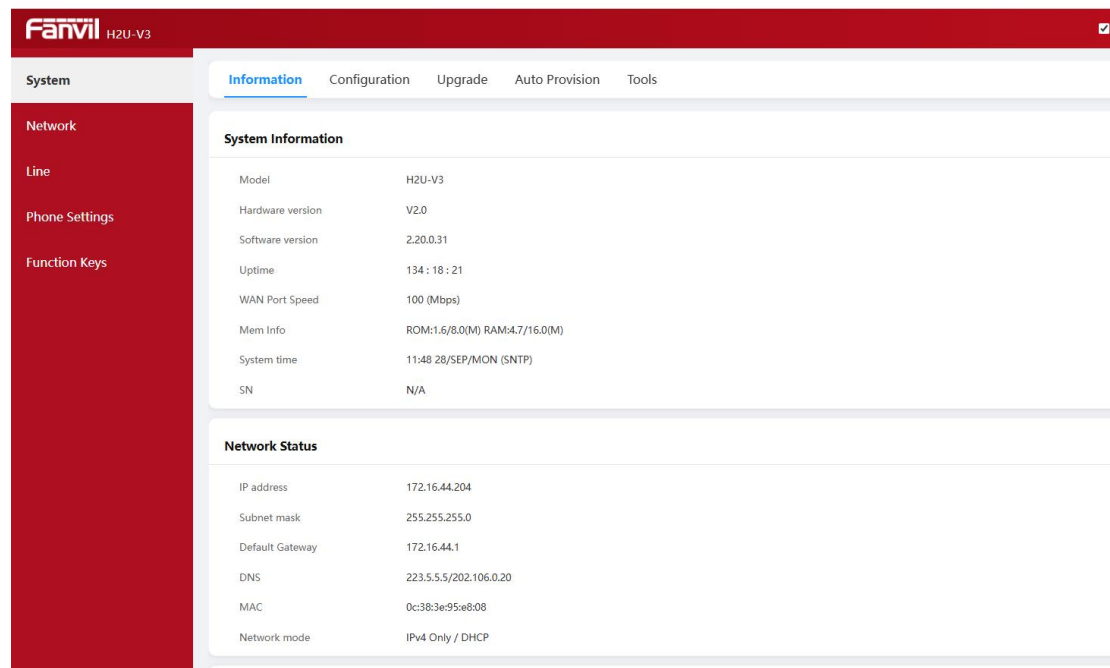
11.2 System Information

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

- Model
- Hardware Version
- Software Version
- Uptime
- WAN Port Speed
- Memory
- System time
- SN

This page also includes information about the network status, including:

- IP address
- Subnet mask
- Gateway
- DNS
- MAC
- Network mode



Fanvil H2U-V3

System

Information | Configuration | Upgrade | Auto Provision | Tools

System Information

Model	H2U-V3
Hardware version	V2.0
Software version	2.20.0.31
Uptime	134 : 18 : 21
WAN Port Speed	100 (Mbps)
Mem Info	ROM:1.6/8.0(M) RAM:4.7/16.0(M)
System time	11:48 28/SEP/MON (SNTP)
SN	N/A

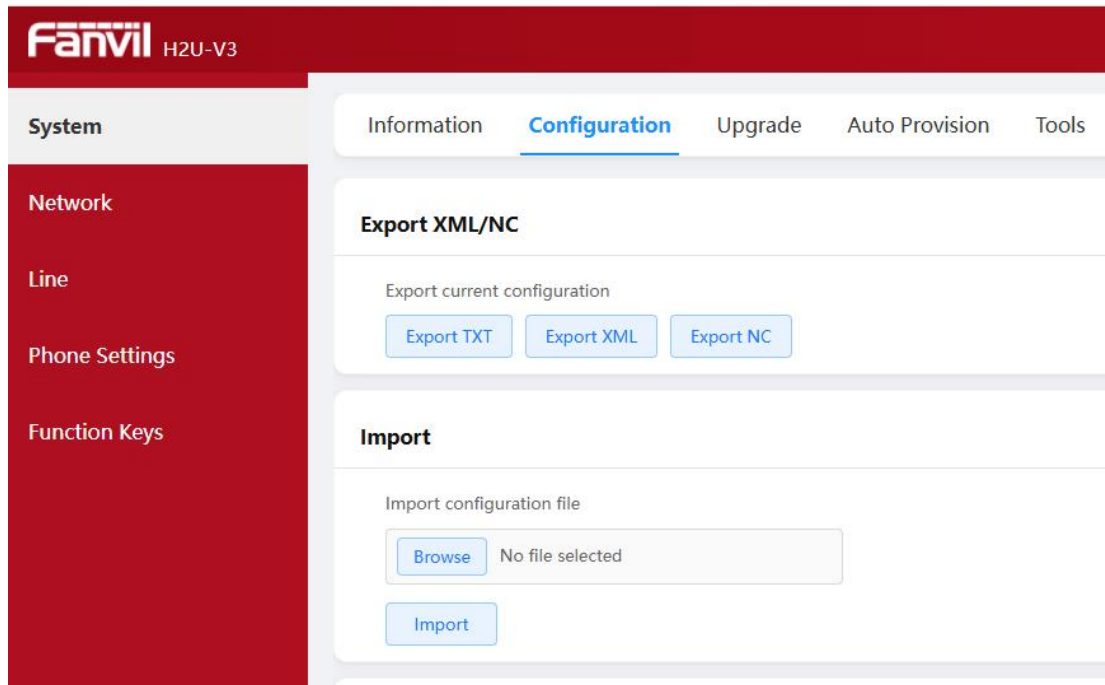
Network Status

IP address	172.16.44.204
Subnet mask	255.255.255.0
Default Gateway	172.16.44.1
DNS	223.5.5.5/202.106.0.20
MAC	0c:38:3e:95:e8:08
Network mode	IPv4 Only / DHCP

Picture 6 - System Information Page

11.3 Config Import/Export

Click "Browse" to select a configuration file, then click "Import" to import the configuration file; Click the "Export TXT," "Export XML," or "Export NC" button to export the configuration file in TXT, XML, or NC format.

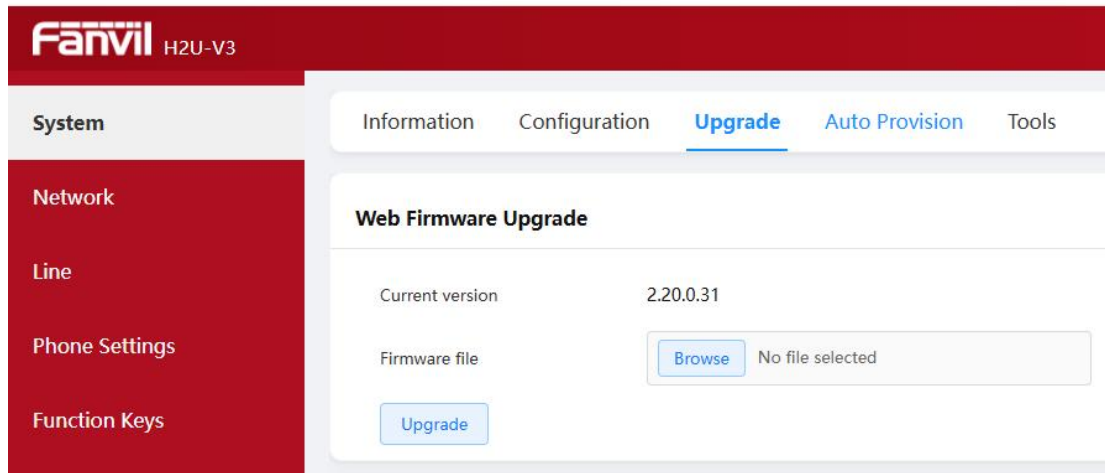


Picture 7 - Import/Export Settings Page

11.4 Web Firmware Upgrade

- Web Interface: Log in to the phone's web interface and navigate to the firmware upgrade module. Click "Browse" to select the target version you want to upgrade to, then click "Upgrade."

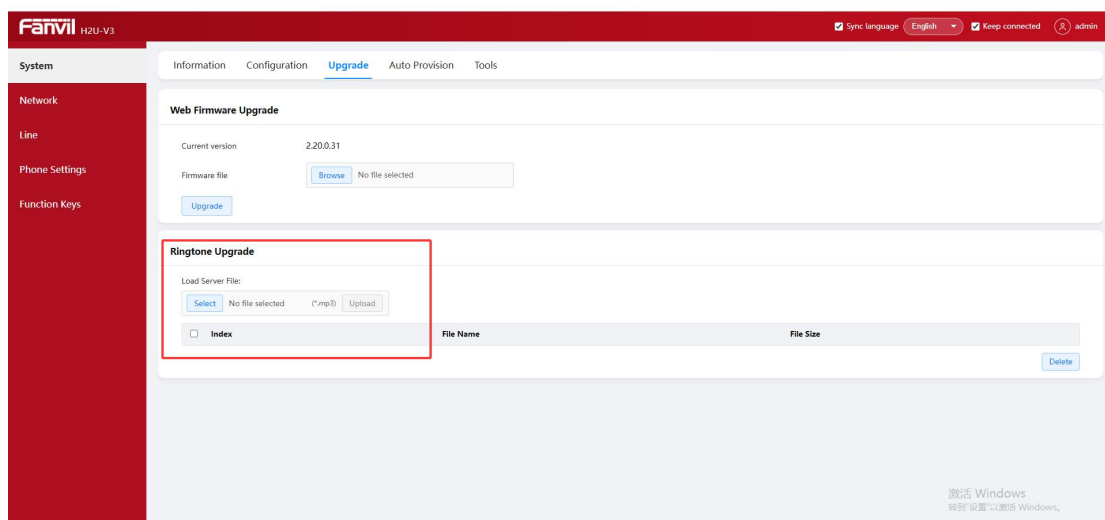
Parameter	Description
Software Version Information	
Current Software Version	Displays the current phone software version number.
Firmware File	Displays the target software version number.



Picture 8 - Web Firmware Update Page

11.5 Web Ringtone Update

- Log in to the phone's web interface, go to System > Upgrade > Ringtone Upgrade, select a file with the .mp3 extension, and click the Upload button to successfully upload the file. In the phone's web interface, go to Lines > Advanced Settings > Ring Type and change it to "Custom Ringtone"; SIP calls to the phone will then play the custom ringtone. In the phone's web interface, go to Phone Settings > Media Settings > Preset Ringtone Type and change it to "Custom Ringtone"; SIP/IP calls to the phone will then play the custom ringtone.
- After upgrading the ringtone via the web interface, you can select the ringtone and click the "Delete" button to remove the custom ringtone.



Picture 9 - Web Ringtone Update

11.6 Basic Network Configuration

The phone supports only wired network connections. It uses an IP network connection to provide services. Unlike traditional phones based on circuit-switched technology, IP phones connect to each other over the network via their IP addresses to exchange data packets and data.

To enable this phone, you must first configure the network settings correctly; the device's default network mode is DHCP/IPv4. If you wish to change to a different mode, you need to access the device's web configuration interface.

- Web Interface: Select the network mode in the Network Configuration module.

■ IP Mode

The phone's IP Mode supports only IPv4. By default, no mode selection is required.

■ IPv4

There are two network types available: DHCP and Static IP.

When the network type is set to DHCP, the phone receives its IP address from the DHCP server (router).

- Use Dynamic DNS: Enabled by default; used for domain name resolution.
- Use Dynamic Time: Disabled by default; controls the time at which the DHCP server assigns the IP address.

When the network is set to Static IP, you must manually configure the phone's IP address.

- IP Address: Enter the IP address you wish to set.
- Subnet Mask: Enter the subnet mask.
- Gateway: Used to connect to the network; enter a value based on your needs.
- Primary DNS: The IP address of the primary DNS server. The default is 8.8.8.8, provided free of charge by Google.
- Secondary DNS: The IP address of the secondary DNS server.

Picture 10 - Network Configuration Page

11.7 SIP Registration

Users can configure lines through the web interface by entering the appropriate information in the fields for Registration Address, Registration Username, Registration Password, SIP User, Display Name, and Registration Port. This information is provided by the SIP server administrator.

- **Web Interface:** After logging in to the phone's web interface, go to Line Registration, select SIP1 or SIP2 for configuration, and click Submit to complete the registration, as shown in the figure below:

Picture 11 - Web Line Registration

11.8 Reboot the Phone

Users can restart the phone via the web interface.

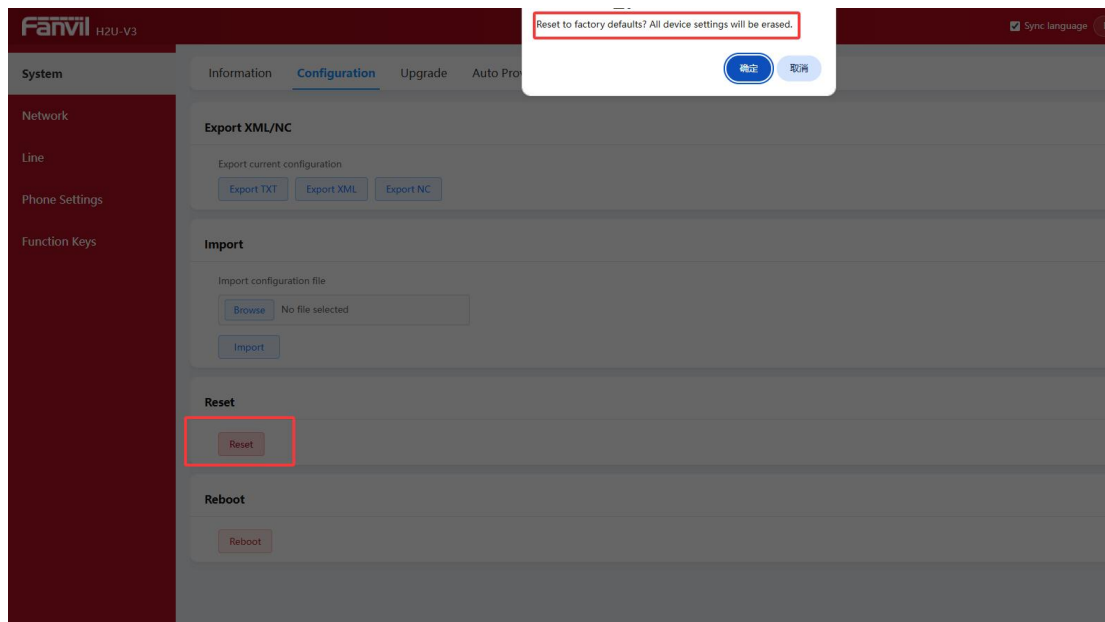
Click the "Reboot" button under the "System-Configuration" module. A prompt will appear asking, "Reboot the device?" Click "OK" to restart the phone. Alternatively, you can simply unplug the power cord to restart the phone.

Picture 12 - Web Reboot

11.9 Restore Factory Settings

Restoring factory settings will delete all configurations, preferences, databases, and profiles from the phone, and the phone will be restored to its factory default state.

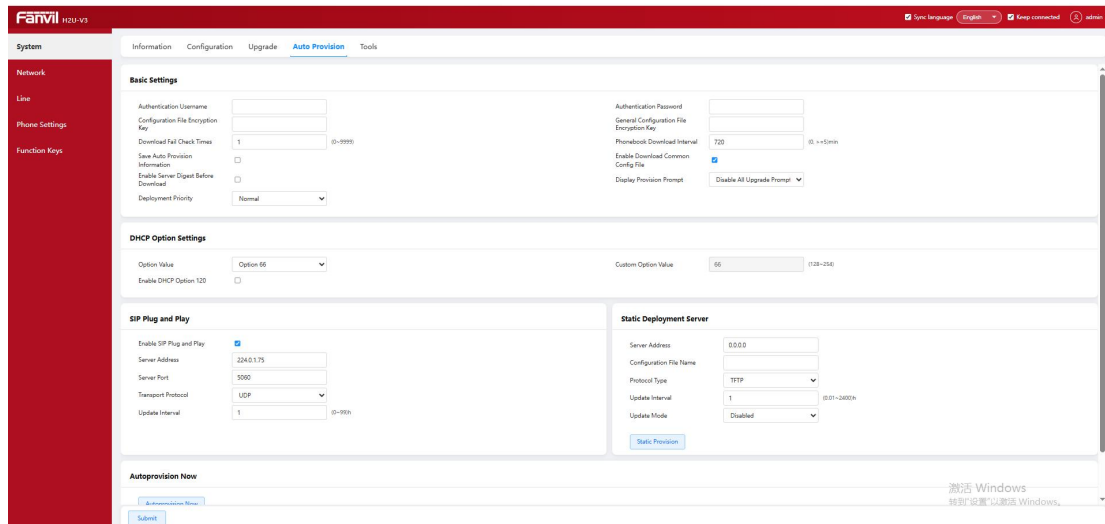
Click the "Reset" button under the System-Configuration module. A web page will prompt you: "Reset to factory defaults? All device settings will be erased." Click "OK" to restore the phone to factory settings.



Picture 13 - Web Restore Factory Settings

11.10 System >> Auto Provision

Page interface: log into the phone page and enter the [system] >> [Auto Provision] page.



Picture 14 - Auto Provision settings

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

PNP>DHCP> Static Provisioning

Transferring protocol: FTP, TFTP, HTTP, HTTPS

Details refer to **Fanvil Auto Provision in**

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	Configure 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.

11.11 System >> Tool

This page provides tools to help users troubleshoot issues.

- System Logs

You can select a log level and export system logs to analyze issues in the event of a

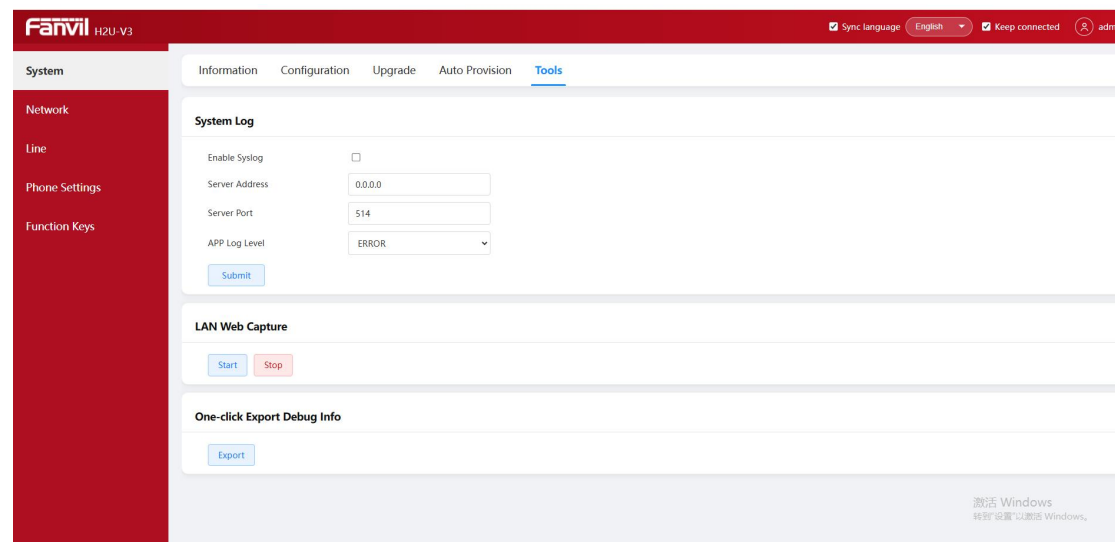
failure.

- Web Packet Capture

Capture packets from network traffic to analyze issues in the event of a failure.

- One-Click Export of Debug Information

When the device experiences an abnormal crash, you can export the crash file to analyze the issue.



Picture 15 - System Tool

11.12 Network >> Service Port

Settings for the web login protocol, protocol port, and RTP port must be modified in the configuration file and then imported into the phone.

Import path: Configuration Import/Export module

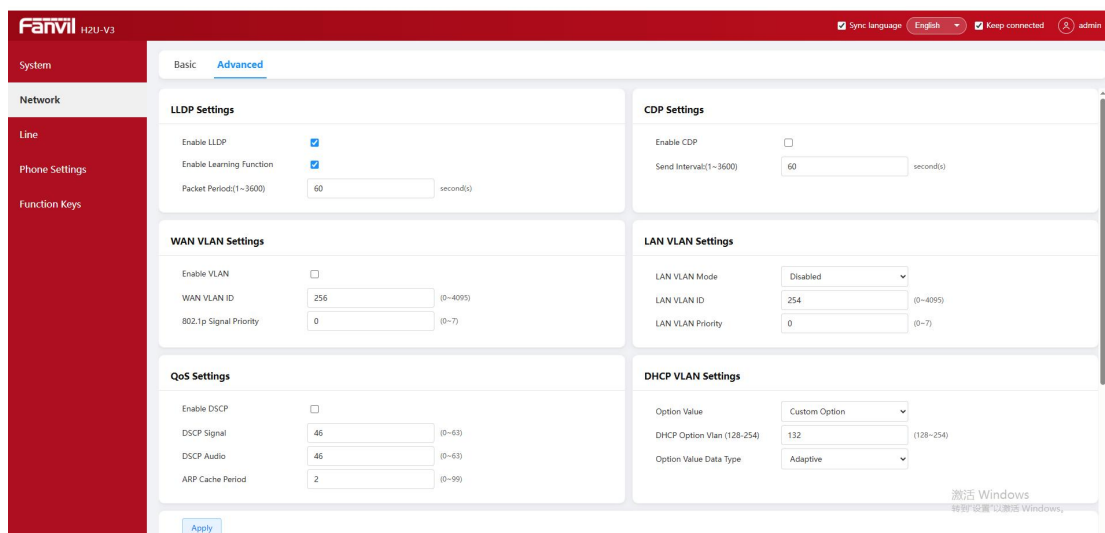
Table 6 - Service port

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.

11.13 Network >> Advanced

The web page layout is shown in the figure below:



Picture 16 - 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

11.14 Line >> SIP

At least one line must be properly configured on the phone to provide telephone service. Line configuration is similar to the role of a SIM card in a mobile phone, which stores the service provider and authentication information for the phone account. When the phone applies these configurations, it automatically registers with the stored service provider—just as when you insert a SIM card into any mobile phone, the phone uses the information on the SIM card rather than its own settings to access services.

Users can configure lines through the web interface by entering the appropriate information in the fields for Registration Address, Registration Username, Registration Password, SIP User, Display Name, and Registration Port. This information is provided by the SIP server administrator.

Table 8 - Line Configuration in the Configuration File

Parameters	Description
Register Settings	

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

Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
Call Forward Unconditional	Enable unconditional call forward, all incoming calls will be forwarded to the number specified in the next field
Call Forward Number for Unconditional	Set the number of unconditional call forward
Call Forward on Busy	Enable call forward on busy, when the phone is busy, any incoming call will be forwarded to the number specified in the next field.
Call Forward Number for Busy	Set the number of call forward on busy .
Call Forward on No Answer	Enable call forward on no answer, when an incoming call is not answered within the configured delay time, the call will be forwarded to the number specified in the next field.
Call Forward Number for No Answer	Set the number of call forward on no answer.
Call Forward Delay for No Answer	Set the delay time of not answered call before being forwarded.
Transfer Timeout	Set the timeout of call transfer process.
Conference Type	Set the type of call conference, Local=set up call conference by the device itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server
Server Conference Number	Set the conference room number when conference type is set to be Server
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Voice Message Number	Set the number for retrieving voice message
Voice Message Subscribe Period	Set the interval of voice message notification subscription
Enable Hotline	Enable hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or turn on hands-free speaker or headphone
Hotline Delay	Set the delay for hotline before the system automatically dialed it

Hotline Number	Set the hotline dialing number
Dial Without Registered	Set call out by proxy without registration
Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.
DTMF Type	Set the DTMF type to be used for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Use VPN	Set the line to use VPN restrict route
Use STUN	Set the line to use STUN for NAT traversal
Enable Failback	Whether to switch to the primary server when it is available.
Failback Interval	A Register message is used to periodically detect the time interval for the availability of the main Proxy.
Signal Failback	Multiple proxy cases, whether to allow the invite/register request to also execute failback.
Signal Retry Counts	The number of attempts that the SIP Request considers proxy unavailable under multiple proxy scenarios.
Codecs Settings	Set the priority and availability of the codecs by adding or remove them from the list.
Video Codecs	Select video code to preview video.
Advanced Settings	
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field.
Enable DND	Set the feature code to dial to the server
Disable DND	Set the feature code to dial to the server
Enable Call Forward Unconditional	Set the feature code to dial to the server

Disable Call Forward Unconditional	Set the feature code to dial to the server
Enable Call Forward on Busy	Set the feature code to dial to the server
Disable Call Forward on Busy	Set the feature code to dial to the server
Enable Call Forward on No Answer	Set the feature code to dial to the server
Disable Call Forward on No Answer	Set the feature code to dial to the server
Enable Blocking Anonymous Call	Set the feature code to dial to the server
Disable Blocking Anonymous Call	Set the feature code to dial to the server
Call Waiting On Code	Set the feature code to dial to the server
Call Waiting Off Code	Set the feature code to dial to the server
Send Anonymous On Code	Set the feature code to dial to the server
Send Anonymous Off Code	Set the feature code to dial to the server
SIP Encryption	Enable SIP encryption such that SIP transmission will be encrypted
RTP Encryption	Enable RTP encryption such that RTP transmission will be encrypted
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if there is not new session timer event update received after the timeout period
Session Timeout	Set the session timer timeout period
Enable BLF List	Enable/Disable BLF List
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Response Single Codec	If setting enabled, the device will use single codec in response to an incoming call request
BLF Server	The registered server will receive the subscription package from ordinary application of BLF phone. Please enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened
Keep Alive Interval	Set the keep alive packet transmitting interval
Keep Authentication	Keep the authentication parameters from previous authentication
Blocking Anonymous Call	Reject any incoming call without presenting

	caller ID
User Agent	Set the user agent, the default is Model with Software Version.
Specific Server Type	Set the line to collaborate with specific server type
SIP Version	Set the SIP version
Anonymous Call Standard	Set the standard to be used for anonymous
Local Port	Set the local port
Ring Type	Set the ring tone type for the line
Enable user=phone	Sets user=phone in SIP messages.
Use Tel Call	Set use tel call
Auto TCP	Using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes
Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
DNS Mode	Select DNS mode, A, SRV, NAPTR
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server, it will use the source IP address, not the address in via field.
Convert URI	Convert not digit and alphabet characters to %hh hex code
Use Quote in Display Name	Whether to add quote in display name, i.e. "Fanvil" vs Fanvil
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
Sync Clock Time	Time Sync with server
Enable Inactive Hold	With the post-call hold capture package enabled, you can see that in the INVITE package, SDP is inactive.
Caller ID Header	Set the Caller ID Header
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Enable Feature Sync	Feature Sync with server
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
CallPark Number	Set the CallPark number.

Server Expire	Set the timeout to use the server.
TLS Version	Choose TLS Version.
uaCSTA Number	Set uaCSTA Number.
Enable Click To Talk	With the use of special server, click to call out directly after enabling.
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	Configure the uaCSTA feature

11.15 Line >> Dial Plan

Table 9 - Phone 4 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
Enable E.164	Please refer to e. 164 standard specification

Add dialing rules: Import via a Web Configuration File

Table 10 - Dial - up rule configuration table

Parameters	Description
Dial rule	<p>There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules.</p> <p>In prefix matching, only part of the number is entered followed by T. The mapping will then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.</p>
<p>Note: Two different special characters are used.</p> <ul style="list-style-type: none"> ■ x -- Matches any single digit that is dialed. ■ [] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits. 	
Destination	Set Destination address. This is for IP direct.
Port	Set the Signal port, and the default is 5060 for SIP.
Alias	Set the Alias. This is the text to be added, replaced or deleted. It is an optional item.
<p>Note: There are four types of aliases.</p> <ul style="list-style-type: none"> ■ all: xxx – xxx will replace the phone number. ■ add: xxx – xxx will be dialed before any phone number. ■ del –The characters will be deleted from the phone number. ■ rep: xxx – xxx will be substituted for the specified characters. 	
Suffix	Characters to be added at the end of the phone number. It is an optional item.
Length	Set the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. It is an optional item.

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

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

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

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

x -- Matches any single digit that is dialed.

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

With these rules, users can easily configure dialing rules that suit their needs and create rules tailored to their specific servers, greatly enhancing the convenience and practicality of the phone.

11.16 Line >> Basic Settings

Set up the register global configuration.

Table 11 - 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
SIP P2P Settings	

Enable Auto Answering	Turn on Auto Answering.
Auto Answering Delay	After the set time has elapsed, the device automatically answers the incoming call.
DTMF Type	Set the DTMF type.
DTMF SIP INFO Mode	Set the expression of #/* when SIP INFO is used as DTMF Send Type.
Use VPN	Turn on VPN.
Call-ID Format	Set the format of the SIP message Call-ID field, which is \$id@\$ip by default

11.17 Phone settings >> Features

Configuration phone features.

Table 12 - General function Settings

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
Enable Call Transfer	Enable Call Transfer.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it
Enable 3-Way Conference	Enable 3-way conference by selecting it
Enable Auto Onhook	The phone will hang up and return to the idle automatically at hands-free mode
Auto Onhook Time	Specify Auto Onhook time, the phone will hang up and return to the idle automatically after Auto Hand down time at hands-free mode, and play dial tone Auto Onhook time at handset mode
Enable Silent Mode	When enabled, the phone is muted, there is no ringing when calls, you can use the volume keys and mute key to unmute.
Disable Mute for Ring	When it is enabled, you can't mute the phone
Enable Default Line	If enabled, user can assign default SIP line for dialing out rather than SIP1.
Enable Auto Switch Line	Enable phone to select an available SIP line as default automatically

Default Ext Line	Select the default line to use for outgoing calls
Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out any number.
Hide DTMF	Configure the hide DTMF mode.
Enable CallLog	Select whether to save the call log.
Enable Restricted Incoming List	Whether to enable restricted call list.
Enable Allowed Incoming List	Whether to enable the allowed call list.
Enable Restricted Outgoing List	Whether to enable the restricted allocation list.
Enable Country Code	Whether the country code is enabled.
Country Code	Fill in the country code.
Area Code	Fill in the area code.
Allow IP Call	If enabled, user can dial out with IP address
P2P IP Prefix	Prefix a point-to-point IP call.
Emergency Call Number	Configure the Emergency Call Number. Despite the keyboard is locked, you can dial the emergency call number
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP address. More details please refer to this link
Push XML Server	Configure the Push XML Server, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.
Enable Pre-Dial	Disable this feature, user enter number will open audio channel automatically. Enable the feature, user enter the number without opening audio channel.
Enable Multi Line	If enabled, up to 10 simultaneous calls can exist on the phone, and if disabled, up to 2 simultaneous calls can exist on the phone.
Line Display Format	Custom line format: SIPn/SIPn: xxx/xxx@SIPn
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.
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	
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.

11.18 Phone settings >> Media Settings

Change voice Settings.

Table 13 - Voice settings

Parameter	Description
Codecs Settings	Select enable or disable voice encoding: G.711A/U,G.722,G.729, G.726-16,G726-24,G726-32,G.726-40, ILBC, Opus
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.
DTMF Payload Type	Enter the DTMF payload type, the value must be

	96~127.
Opus payload type	Set Opus load type, range 96~127.
OPUS Sample Rate	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb (16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length
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	1.mp3-10.mp3

11.19 Phone settings >> MCAST

Using the multicast function, we can simply and conveniently send the announcement to each member of the multicast, and send the multicast RTP stream to the preconfigured multicast address by setting the multicast key on the phone. Listen for and play the RTP stream sent from the multicast address by configuring the listening multicast address on the phone.

Table 14 - 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.
------------	--

Multicast Calls:

- Set the multicast address, and choose the encoding
- Once configured, click “Submit”
- On the web page of the phone receiving the multicast call, go to [Phone Settings] >> [Multicast] to set the multicast name, host, and port
- Press the configured DSSkey multicast key
- The recipient will receive the multicast call and automatically play the multicast content

11.20 Phone settings >> Action

Action URL

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

11.21 Phone settings >> Time/Date

The user can configure the time Settings through the configuration file.

Table 15 - Time&Date settings

Parameters	Description
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Primary Time Server	Set primary time server address
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization.
Time Zone	Select the time zone
Resync Period	Time of re-synchronization with time server
12-Hour Clock	Set the time display in 12-hour mode
Date Format	Select the time/date display format
Daylight Saving Time Settings	
Local	Choose your local, phone will set daylight saving

	time automatically based on the local
DST Set Type	Choose DST Set Type, if Manual, you need to set the start time and end time.
Fixed Type	Daylight saving time rules are based on specific dates or relative rule dates for conversion. Display in read-only mode in automatic mode.
Offset	The offset minutes when DST started
Month Start	The DST start month
Week Start	The DST start week
Weekday Start	The DST start weekday
Hour Start	The DST start hour
Minute Start	The DST start minute
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Minute End	The DST end minute
Manual Time Settings	You can set your time manually

11.22 Phone settings >> Tone

Users can configure the phone's prompt tones in the configuration profile.

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.

11.23 Phonebook >> Call List

■ Restricted Incoming Calls:

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

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

■ Restricted Outgoing Calls:

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

removed from the table.

11.24 Function Key >> Function Key

Table 16 - Function Key configuration

Parameters	Description
Memory Key	<p>Speed Dial: You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed.</p> <p>Intercom: This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments.</p>
Key Event	Users can select a function key as a shortcut to trigger an event
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.
DTMF	It allows user to dial or edit dial number easily.

11.25 Function Key >> Speed Dial List

The user can configure the number button "0~9" to be the speed dial key. After the configuration is completed according to the figure below, the user can press the configured shortcut key for the phone to quickly dial the configuration number. Can more quickly and conveniently exhale, eliminating the need to dial, check the number of the trouble.

Picture 17 - Web Speed Dial List

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 access information through the "System Information" and "Network Status" sections on the web page. The following information will be provided:

Network Status

Device Information (model, software and hardware versions), etc.

12.2 Reboot Device

Users can restart the phone via the web interface.

Click the "Reboot" button under the "System Operations" module. A prompt will appear asking, "Reboot the device?" Click "OK" to restart the phone. Alternatively, you can simply unplug the power cord to restart the phone.

12.3 Reset Device to Factory Default

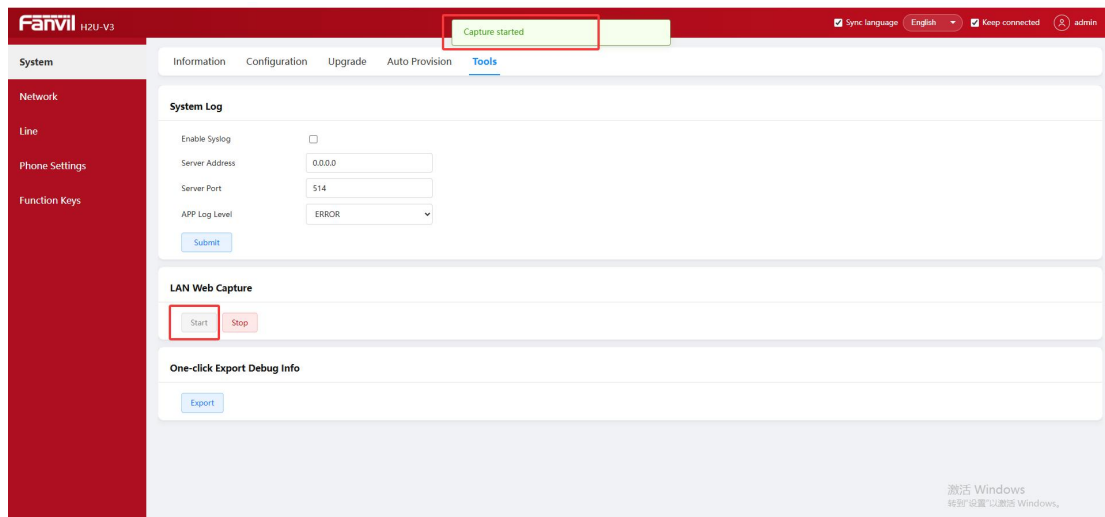
Reset Device to Factory Default will erase all user's configuration, preference, database and profiles on the device and restore the device back to the state as factory default.

Click the "Reset" button under the System Operations module. A web page will prompt you: "Reset to factory defaults? All device settings will be erased." Click "OK" to restore the phone to factory settings.

12.4 Network Packets Capture

Sometimes, network packets from the phone can be helpful. To obtain the phone's packets, users need to log in to the phone's web interface, navigate to [System] >> [Tools], and then click [Start] under "LAN Web Capture." A prompt will indicate that packet capture has begun. At this point, users can perform relevant operations, such as activating or deactivating a line or placing a call. When finished, click the [Stop] button on

the web page. All network packets generated during the phone's operation will be saved to a file.

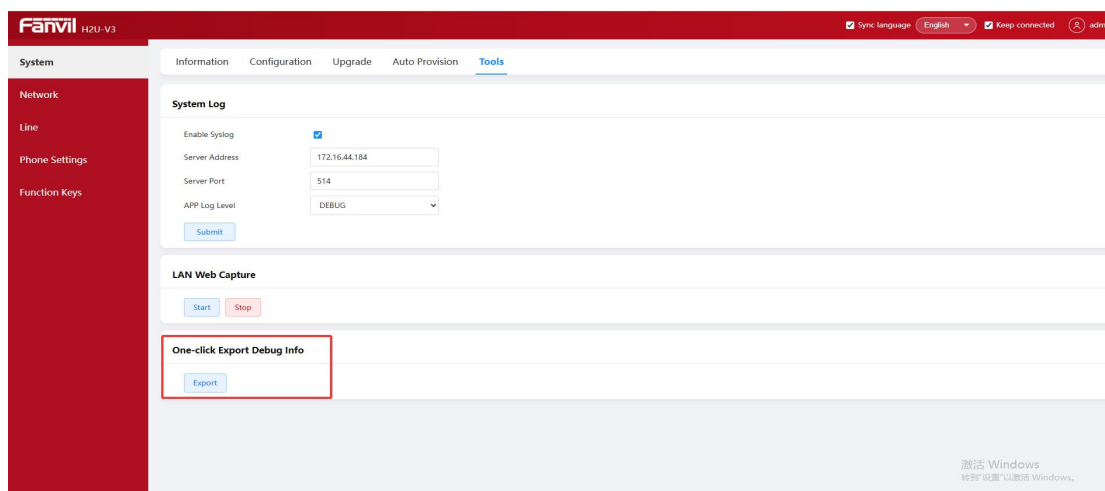


Picture 18 - Web capture

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

12.5 One-click Export Debug Info

If the device experiences an unexpected crash, users can log in to the phone's web interface, navigate to System > Tools > One-Click Export Debug Information, click the "Export" button, and save the file locally for analysis or send it to a technician to help identify the issue.



Picture 19 - Web One-click Export Debug Info

12.6 Common Trouble Cases

Table 17 - 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 Fanvil 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	<ol style="list-style-type: none"> 1. Please check if device is well connected to the network. The network Ethernet cable should be connected to the  [Network] port NOT the  [PC] port. 2. Please check if the device has an IP address. Check the system information, if the IP displays "Negotiating...", the device does not have an IP address. Please check if the network configurations is correct. 3. If network connection is fine, please check again your line configurations. If all configurations are correct, please kindly contact your service provider to get support, or follow the instructions in "13.5 Network Packet Capture" to get the network packet capture of registration process and send it to Fanvil support to analyze the issue.
No Audio or Poor Audio in Handset	<ol style="list-style-type: none"> 1. Please check if Handset is connected to the correct Handset () port NOT Headphone () port. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Poor Audio or Low Volume in Headphone	<ol style="list-style-type: none"> 1. There are two Headphone wire sequence in the market. Please use the Headphone provided by Fanvil, or consult Fanvil the wire sequence if you wish to use a third-party headphone. 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.