



i16S&i16SV &i16S-02P&i16SV-02P User Manual

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3 Safety Instruction

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

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



4 Overview

The i16S&i16SV SIP Intercom is designed for outdoor scenes with high reliability, HD Audio/Video, and IP65/IK10 protection grade. It combines security, audio/video intercom and broadcasting functionalities and offer the best communication solution for users.



5 Install Guide

5.1 Use POE or external Power Adapter

i16S&i16SV, called as 'the device' hereafter, supports two power supply modes, power supply from external power adapter or over Ethernet (POE) complied switch.

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

For users who do not have POE equipment, the traditional power adaptor should be used. If the device is connected to both POE switch and external power adapter, i16S&i16SV will get power supply from POE switch in priority, and change to external power adapter once the POE power supply fails.

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

5.2 Appendix

5.2.1 Common command modes

| Action behavior | Description | | |
|-------------------|---|--|--|
| Standby report IP | In standby mode, long press the speed dial button for 3 seconds, | | |
| | there will be a toot sound will 5 seconds, please press the speed | | |
| | dial button once within 5 seconds, the toot sound will stop | | |
| | automatically reporting IP | | |
| | In the standby mode, long-press the speed dial button for 3 | | |
| Switch network | seconds and the beep will last for 5 seconds. Within 5 seconds, | | |
| mode | press the speed dial button three times quickly to switch to the | | |
| mode | network mode. | | |
| | If there is no IP at present, switch to the default static IP | | |

Table 1- Common command mode



| (192.168.1.128). |
|--|
| Then switch to DHCP mode when it is the default static IP |
| (192.168.1.128) |
| When DHCP gets to IP, then do not switch and report the IP |
| directly. |
| Report the IP after the successful switch. |

5.2.2 Function key LED status

Table 2- Function key LED status

| Туре | LED | Status |
|---------|---------------|---|
| SIP/NET | Normally on | Successfully Registered |
| | Fast Flashing | Registration failed/network abnormality |
| | Slow Flashing | In call |

5.2.3 Model List

Table 3 - Model List

| Model | Network | Button | Camera |
|-----------|------------|--------|--------------|
| i16S | 10/100Mbps | 1 | × |
| i16SV | 10/100Mbps | 1 | \checkmark |
| i16S-02P | 10/100Mbps | 2 | × |
| i16SV-02P | 10/100Mbps | 2 | \checkmark |



6 User Guide

6.1 Panel Overview





Picture 1 - Panel



| Number | Name | Description |
|--------|---------------|---|
| 1 | IP Camera | Video signal acquisition and transmission |
| | Infrared lamp | Only i16SV |
| 2 | Speaker | Play sound |
| 2 | DSS kov | For speed dial, multicast, intercom, IP broadcast and |
| 3 | DSS key | other functions |
| 4 | MIC | Audio acquisition |

Table 4 - Panel introduction

6.2 Interface description

Open the rear case of the device, there is a row of terminal blocks for connecting the power supply, electric lock control, etc. The connection is as follows:



Picture 2 - Interface

| Table 5 - | Interface |
|-----------|-----------|
|-----------|-----------|

| SN | Description | Wiring port description(example above) | |
|----|---|--|--|
| | Ethernet interface: standard RJ45 interface, | | |
| 1 | 10/100M adaptive, it is recommended to use | | |
| | five or five types of network cable | | |
| 2 | Power interface: 12V/1A input | Left positive, right grounded | |
| | Two groups of short-circuit input detection | | |
| | interfaces: for connecting switches, infrared | Left IN, right OUT | |
| 35 | probes, door magnets, vibration sensors and | | |
| | other input devices | | |
| | Two groups of short-circuit output control | Left (NC): Normally Close | |
| 46 | interface: used to control electric locks, | Contact | |
| | alarms, etc. | Center (COM): Common Contact | |



| | | Right (NO): Normally Open |
|--------------------------|---|--------------------------------|
| | | Contact |
| | Recording output interface: Mix the device | |
| | and the sound of the far-end call. One is the | Loft grounded right recording |
| $\overline{\mathcal{O}}$ | recording signal line, and the other is the | Left grounded, right recording |
| | ground line (be sure to ground the line, | output |
| | otherwise there will be noise) | |
| | External active speaker interface: external | |
| | active speakers for audio power | |
| 8 | amplification. One is the audio signal line, | Left grounded, right external |
| | and the other is the ground line (be sure to | active speaker |
| | ground the line, otherwise there will be | |
| | noise) | |

6.3 Installation instructions

6.3.1 Installation

1. Use built-in screw tool to remove the surface shell;

2. Based installation dimensions, mounting hole in the wall to draw, use an electric drill holes lay;

3. The white rubber plugged into the wall and the bottom fixed with screws to the wall;

4. After connecting the power cord and network cable, screw the surface shell fixed.



Picture 3 - Installation

6.3.2 Device IP address

Method one:

Open the web page and enter http://download.fanvil.com/tool/iDoorPhoneNetworkScanner.exe



to download and install the IP scanning tool.

Open the IP scanning tool, click the refresh button, search for the device and find the corresponding IP address.

| IP Address | Serial Number | MAC Address | SW Version | Description | |
|--------------|---------------|-------------------|------------|-------------|---------|
| 172.18.70.55 | IP Intercom | 00:a8:34:00:aa:74 | TO. 1. 1 | IP Intercom | |
| | | | | | Refresh |

Method two:

Connect the speaker and press and hold the speed dial button for 3 seconds (30 seconds after power-on), wait for the loudspeaker to beep quickly, press the speed dial button one times within 5 seconds, and the system will automatically announce the IP address by voice after successfully switching to dynamic IP.

Method three:

Press and hold the speed dial button for 3 seconds, wait for the loudspeaker to beep quickly, press the speed dial button three times within 5 seconds, and the system will automatically announce the IP address by voice after successfully switching to dynamic IP.

| Table 6 - | Configuration | instructions |
|-----------|---------------|--------------|
|-----------|---------------|--------------|

| Default configuration | | | | | | | | |
|-----------------------|--|-------------|---------------|--|--|--|--|--|
| DHCP mode | Default enable | Static IP | 192.168.1.128 | | | | | |
| Voice read IF | Long press the speed dial button for 3 | Server port | 80 | | | | | |
| address | seconds, press the speed dial button one | | | | | | | |
| | times within 5 seconds | | | | | | | |

6.4 WEB configuration

When the device and your computer are successfully connected to the network, enter the IP address of the device on the browser as http://xxx.xxx.xxx/ and you can see the login interface of the web page management.



| User: | |
|-----------|-------------|
| Password: | |
| Language: | English 🔹 🔲 |
| | Logon |

Picture 4 - WEB Login

The username and password should be correct to log in to the web page. **The default username and password are "admin"**. For the specific details of the operation of the web page, please refer to <u>9 Web Configurations</u>

6.5 SIP Configurations

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

The SIP line configuration should be set via the WEB configuration page by entering the correct information such as phone number, authentication name/password, SIP server address, server port, etc. which are provided by the SIP server administrator.

• WEB interface: After login into the phone page, enter [Line] >> [SIP] and select SIP1/SIP2 for configuration, click apply to complete registration after configuration, as shown below:



| System | | | | | |
|------------------------|--------------------------|----------------|---|------------------------------|----------------|
| Network | Line 1356@SIP: • | | | | |
| | Register Settings >> | | | | |
| Line | Line Status: | Registered | | Activate: | 2 |
| | Username: | 1356 | 0 | Authentication User: | |
| intercom settings | Display name: | | 0 | Authentication Password: | |
| | Realm: | | 0 | Server Name: | |
| Call List | | | | | |
| | SIP Server 1: | | | SIP Server 2: | |
| unction Key | Server Address: | 172.16.1.2 | 0 | Server Address: | |
| | Server Port: | 5060 | 0 | Server Port: | 5060 |
| ecurity | Transport Protocol: | UDP V | | Transport Protocol: | UDP V |
| | Registration Expiration: | 3600 second(s) | 0 | Registration Expiration: | 3600 second(s) |
| evice Log | | | | | |
| te de construir (de de | Proxy Server Address: | | 0 | Backup Proxy Server Address: | |
| ecurity Settings | Proxy Server Port: | 5060 | 0 | Backup Proxy Server Port: | 5060 |
| | Proxy User: | | 0 | | |
| | Proxy Password: | | 0 | | |
| | Basic Settings >> | | | | |
| | Codecs Settings >> 💡 | | | | |
| | Advanced Settings >> | | | | |

Picture 5 - SIP Line Configuration

6.6 Volume setting

Set the volume (if the speaker or microphone is not connected, you can skip it)

[Intercom Settings] >> [Media Settings] >> [Media Settings], as shown below, click [Submit].

Hands-free volume setting: Set the speaker output volume.

Hands-free microphone gain: microphone volume level.

| | Features Media Setting | S Camera Settings | MCAST | Action | Time/Date | Time Plan | Tone |
|-------------------|-----------------------------------|-------------------|--------------|-------------------|-------------|-----------|--------------------------------|
| System | | | | | | | NOTE |
| Network | Codecs Settings >> 🕜 | | | | | | Description |
| | Media Settings >> | | | | | | Media setting set the voice |
| Line | Default Ring Type: | 1.wav 🔻 🕜 | | | | | coding,volum |
| | Speakerphone Volume: | 7 (1~9) | 0 | | | | and so on. |
| Intercom settings | Speakerphone Ring Volume: | 0 (0~9) | 0 | | | | |
| | DTMF Payload Type: | 101 (96~1 | 27) 🕜 | | | | |
| all List | Handfree Mic Gain: | 3 (1~9) | | | | | |
| | OPUS Payload Type: | 107 (96~1 | 27) OPUS | Sample Rate | OPUS-NB(I V | | |
| unction Key | ILBC Payload Type: | 97 (96~1 | 27) 🕜 ILBC I | Payload Length | 20ms 🔻 🕜 | 6 | |
| incuoir key | Enable VAD: | | | | | | |
| ecurity | H.264 Payload Type: | 117 (96~12) | 7) H.265 | Payload Type: | 98 | 96~127) | |
| curry | Enable Line-in: | Enable 🔻 🕜 | | | | | |
| P2(120) | Enable Line-out: | Lineout | • 📀 | | | | |
| evice Log | Speaker | Panel Spea 🔻 | Extern | nal Speaker Power | 10 🔻 😡 | | |
| | Lineout Trigger State | | | | | | |
| curity Settings | Disabled State | | | Enabled State | | | |
| | Talking(Calling) | * | Ringin | Ig | * | | |
| | Talking(Sip) Talking(Intercom) | | | | | | |
| | Talking(Mcast) | → ← | | | | | |
| | | - | | | | | |
| | | Ψ. | | | Ŧ | | |
| | RTP Control Protocol(RTCP) Se | attings > > | | | | | |
| | RTP CONTROL PROTOCOI(RTCP) Si | ettings >> | | | | | |
| | RTP Settings >> | | | | | | |
| | | | | | | | |

Picture 6- Volume Set



7 Basic Function

7.1 Making Calls

After setting the function key to Hot key and setting the number, press the function key to immediately call out the set number, as shown below:

| > System | | | | | | | | | | | | | NOTE |
|---------------------|--------------|---------------|--------|---------|-------|--------|------------|---|-----------|---|---------|---|--|
| | Functio | n Key Setting | 15 >> | | | | | | | | | | |
| > Network | Кеу | Туре | | Name | Value | Value2 | Subtype | Э | Line | | Media | | Description: Soft function key, which |
| > Line | DSS Key 1 | Memory Key | • | | 2345 | | Speed Dial | ۳ | 1356@SIP1 | ۷ | DEFAULT | • | Soft function key, which can be defined by soft function key on different call interface. |
| | DSS Key 2 | None | • | | | | None | ٣ | AUTO | ۲ | DEFAULT | ٧ | call interface. |
| > Intercom settings | | | | | | Apply | | | | | | | |
| > Call List | Program | mmable Key S | Settin | gs 🕜 >> | | | | | | | | | |
| > Function Key | Advanc | ed Settings > | > | | | | | | | | | | |
| › Security | | | | | | | | | | | | | |
| > Device Log | | | | | | | | | | | | | |
| > Security Settings | | | | | | | | | | | | | |
| | | | | | | | | | | | | | |
| | | | | | | | | | | | | | |
| | | | | | | | | | | | | | |

Picture 7- Function Setting

See detailed configuration instructions 9.26 Function Key

7.2 Answering Calls

After setting up the automatic answer and setting up the automatic answer time, it will hear the ringing bell within the set time and automatically answer the call after timeout. Cancel automatic answering. When a call comes in, you will hear the ringing bell and will not answer the phone over time.

7.3 End of the Call

You can hang up the call through the Release key (you can set the function key as the Release key) or turn on the speed dial button to hang up the call. See detailed configuration instructions <u>9.26 Function Key</u>.

7.4 Auto Answer

The user can turn off the auto-answer function (enabled by default) on the device webpage, and



the ring tone will be heard after the shutdown, and the auto-answer will not time out.

Web interface:

Enter [Line] >> [SIP], Enable auto answer and set auto answer time and click submit.

| | SIP SIP Hots | pot Basic Settings | | | | |
|---------------------|--|---|--|------------|---------------------|---|
| > System | | | | | | NOTE |
| > Network | Line 1356@SIP: • | | | | | Description: It shows phone |
| > Line | Register Settings >> Basic Settings >> | | | | | registration account basic settings and sip account function advanced |
| › Intercom settings | Enable Auto Answering: | | Auto Answering Delay: | 0 |](0~120)second(s) 🕜 | settings. |
| › Call List | Enable Hotline: Hotline Delay: | ② 0 (0~9)second(s) | Hotline Number: | | 0 | |
| Function Key | Dial Without Registered: DTMF Type: Request With Port: | | DTMF SIP INFO Mode: | Send 10/11 | • 0 | |
| > Security | Use STUN: | | Use VPN: | 2 | | |
| > Device Log | Enable Failback: Failback Interval: | ✓ ② 1800 second(s) ② | Signal Failback: Signal Retry Counts: | 3 |] (1~10) 🕑 | |
| Security Settings | Codecs Settings >> 😢 | | | | | |
| | Advanced Settings >> | | | | | |
| | SIP Global Settings >> | Apply | | | | |

Picture 8 - WEB line enable auto answer

SIP P2P auto answering:

Enter [Line]>>[Basic settings], Enable auto answer and set auto answer time and click submit.

| | SIP SIP Hotspot | Basic Settings | | |
|----------------------------------|--|---------------------------------|---|--|
| › System | | | | NOTE |
| > Network | STUN Settings STUN NAT Traversal: | FALSE | 0 | Description: Phone line basic settings, |
| › Line | Server Address: Server Port: | 3478 | 0 | including STUN, certificate files. |
| > Intercom settings | Binding Period: SIP Waiting Time: | 50 second(s) 800 millisecond | Ø | |
| > Call List | | Apply | | |
| › Function Key | SIP P2P Settings Enable Auto Answering Auto Answering Delay: | | 0 | |
| > Security | DTMF Type: DTMF SIP INFO Mode: | RFC2833 V Send 10/11 V | 0 | |
| Device Log Security Settings | | Apply | | |
| · scorry scorings | | | | |
| | | | | |
| | | | | |

Picture 9- Enable auto answer for IP calls

• Auto Answer Timeout (0~120)

The range can be set to 0~120s, and the call will be answered automatically when the timeout is set.



7.5 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 signal will be prompted
- Enable call waiting tone: when you receive a new call on the line, the device will beep.

Users can enable/disable call waiting in the device interface and the web interface.

• Web interface: enter [Intercom Settings] >> [Features], enable/disable call waiting, enable/disable call waiting tone.

| | Features Media Settings | Camera Settings | MCAST | Action | Time/Date | Time Plan | Tone |
|-------------------------------|-----------------------------------|---------------------------|------------|------------------------|-----------|---------------|---|
| > System | | | | | | | NOTE |
| > Network | Basic Settings >> | | | | | | Description: |
| / ITELWOIR | Enable Call Waiting: | ✓ Ø | | | | | Function settings, y |
| > Line | Enable Auto on Hook: | ✓ Ø | P | Auto HangUp Delay: | 3 (0~30 |)second(s) 🕜 | set the phone featu including the basic |
| , Line | Enable Silent Mode: | | 0 | Disable Mute for Ring: | | | settings, tone settin intercom settings, t |
| | | | | | | | corresponding code |
| Intercom settings | Ban Outgoing: | | | | | | settings. |
| | Default Ans Mode: | Video 🔻 🕜 | E. | Default Dial Mode: | Video 🔻 🕜 | | |
| Call List | Enable Restricted Incoming | | | | | | |
| | Enable Restricted Outgoing List: | | E | Enable Country Code: | | | |
| > Function Key | Country Code: | | 4 | Area Code: | | | |
| > Security | | - | | | | | |
| - Bulling | Allow IP Call: | ✓ Ø | F | 2P IP Prefix: | Ł | | |
| > Device Log | Restrict Active URI Source IP: | | Ø F | Push XML Server: | | 0 | |
| | Line Display Format: | xxx@SIPn 🔻 🕜 | | | | | |
| > Security Settings | Call Number Filter: | | 4 | Auto Resume Current: | | | |
| | Limit Talking Duration: | | T | Talking Duration: | 120 (20~6 | 600)second(s) | |
| | No Answer Auto HangUp Timeout: | 130 (1~300)second(s) 2 | E | Enable Push XML Auth: | . 0 | | |
| | Tone Settings >> | | | | | | |
| | Intercom Settings >> | | | | | | |
| | Response Code Settings >> | | | | | | |
| | | | App | dur. | | | |
| | | | wbb | iy . | | | |

Picture 10 - Call Waiting

| | Features Media Setting | gs Camera Settings | MCAST Action | Time/Date | Time Plan | Tone |
|-------------------|---------------------------|--------------------|---------------------------|-----------|-----------|--|
| System | | | | | | NOTE |
| etwork | Basic Settings >> | | | | | Description: |
| | Tone Settings >> | | | | | Function settings, you o set the phone features, |
| ne | Enable Holding Tone: | | Enable Call Waiting Tone: | | | including the basic |
| | Play Dialing DTMF Tone: | | Play Talking DTMF Tone: | I | | settings, tone settings, intercom settings, the corresponding code |
| intercom settings | Intercom Settings >> | | | | | settings. |
| all List | Response Code Settings >> | | Apply | | | |
| unction Key | | | | | | |
| Security | | | | | | |
| evice Log | | | | | | |
| Security Settings | | | | | | |
| | | | | | | |
| | | | | | | |
| | | | | | | |
| | | | | | | |

Picture 11 - Call Waiting tone



8 Advance Function

8.1 Intercom

The equipment can answer intercom calls automatically.

| | Features Media Setting | gs Camera Settings | MCAST Action | Time/Date | Time Plan | Топе |
|---------------------|---------------------------|--------------------|------------------------|-----------|-----------|---|
| › System | | | | | | NOTE |
| > Network | Basic Settings >> | | | | | Description: |
| | Tone Settings >> | | | | | Function settings, you can set the phone features. |
| > Line | Intercom Settings >> | | | | | including the basic settings, tone settings, |
| | Enable Intercom: | | Enable Intercom Mute: | | | intercom settings, the corresponding code |
| > Intercom settings | Enable Intercom Tone: | | Enable Intercom Barge: | | | settings. |
| → Call List | Response Code Settings >> | | Apply | | | |
| > Function Key | | | | | | |
| > Security | | | | | | |
| > Device Log | | | | | | |
| Security Settings | | | | | | |
| | | | | | | |
| | | | | | | |

Picture 12 - WEB Intercom

| Table 7 | - Intercom |
|---------|------------|
|---------|------------|

| Parameters | Description | | | | |
|-------------------------|--|--|--|--|--|
| | When the intercom system is enabled, the device will accept | | | | |
| Enable Intercom | the SIP header call-info of the Call request | | | | |
| | Command automatic call | | | | |
| Enable Intercom Barge | If the option is enabled, device will answer the intercom call automatically while it is in a normal call, and it will reject new intercom call if there is already one intercome call | | | | |
| Enable Intercom Mute | Enable mute during intercom mode | | | | |
| Enable Intercom Ringing | If the incoming call is intercom call, the device plays the intercom tone. | | | | |

8.2 MCAST

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



multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.

| | Features | Media Settings | Camera Settings | MCAST | Action | Time/Date | Time Plan | Tone |
|---------------------------------------|---------------------------------|----------------|-----------------|----------|-----------|-----------|--------------------------|---------------------------------------|
| › System | | | | | | | | NOTE |
| > Network | MCAST Listening Priority: | | 1 | T | | | | Description: Set the multicast add |
| > Line | Enable Page P Enable Prio Ch | ian: | | | | | | and multicast priority. |
| > Intercom settings | Enable Emer C Index/Priori | | Name | | Host:port | | Channel | |
| > Call List | 1 2 | | | | | | 0 v | |
| > Function Key | 3 4 5 | | | | | | 0 V 0 V 0 V | |
| > Security | 6 | | | | | | 0 v 0 v | |
| > Device Log | 8 9 | | | | | | 0 v | |
| Security Settings | 10 | | Appl | (| | | • | |
| | MCAST Dynamic | | | | | | | |
| | Auto Exit Expir | res: | 60 Apply | (| | | | |
| | Index | Priori | ty | MCAST Ip | | Port | | |
| | | | | | | | | |

Picture 13 - MCAST

Table 8- MCAST

| Parameters | Description |
|---------------------------|---|
| Enable Auto Mcast | Send the multicast configuration information by Sip Notify signaling, |
| | and the device will configure the information to the system for |
| | multicast listening or cancel the multicast listening in the system after |
| | receiving the information |
| Auto Mcast Timeout Delete | When a multicast call does not end normally, but for some reason the |
| Time | device can no longer receive a multicast RTP packet, this |
| | configuration cancels the listening after a specified time |
| SIP Priority | Defines the priority in the current call, with 1 being the highest priority |
| | and 10 the lowest. |
| Intercom Priority | Compared with multicast and SIP priority, high priority is pluggable |
| | and low priority is rejected |
| Enable Page Priority | Regardless of which of the two multicast groups is called in first, the |
| | device will receive the higher priority multicast first. |
| Enable Mcast Tone | When enabled, play the prompt sound when receiving multicast |
| Name | Listened multicast server name |
| Host:port | Listened multicast server's multicast IP address and port. |

Multicast:

• Go to web page of [Function Key] >> [Function Key], select the type to multicast, set



the multicast address, and select the codec.

- Click Apply.
- Set up the name, host and port of the receiving multicast on the web page of [Intercom Settings] >> [MCAST].
- Press the DSSKey of Multicast Key which you set.
- Receive end will receive multicast call and play multicast automatically.

MCAST Dynamic:

Description: send multicast configuration information through SIP notify signaling. After receiving the message, the device configures it to the system for multicast monitoring or cancels multicast monitoring in the system.

8.3 Hotspot

SIP hotspot is a simple utility. Its configuration is simple, which can realize the function of group vibration and expand thequantity of sip account. Take one device A as the SIP hotspot and the other devices (B, C) as the SIP hotspot client. When someone calls device A, devices A, B, and C will ring, and if any of them answer, the other devices will stop ringing and not be able to answer at the same time. When A B or C device is called out, it is called out with A SIP number registered with device A.

| Parameters | Description |
|----------------|---|
| Enable Hotspot | Enable or disable hotspot |
| Mode | This device can only be used as a client |
| Monitor Type | The monitoring type can be broadcast or multicast. If you want to restrict |
| | broadcast packets in the network, you can choose multicast. The type of |
| | monitoring on the server side and the client side must be the same, for |
| | example, when the device on the client side is selected for multicast, the |
| | device on the SIP hotspot server side must also be set for multicast |
| Monitor | The multicast address used by the client and server when the monitoring |
| Address | type is multicast. If broadcasting is used, this address does not need to |
| | be configured, and the system will communicate by default using the |
| | broadcast address of the device's wan port IP |
| Remote Port | Fill in a custom hotspot communication port. The server and client ports |
| | need to be consistent |
| Name | Fill in the name of the SIP hotspot. This configuration is used to identify |
| | different hotspots on the network to avoid connection conflicts |
| Line Settings | Sets whether to enable the SIP hotspot function on the corresponding |

Table 9 - SIP Hotspot



| | | SIP line |
|--|--|----------|
|--|--|----------|

Client Settings:

As a SIP hotspot client, there is no need to set up a SIP account, which is automatically acquired and configured when the device is enabled. Just change the mode to "client" and the other options are set in the same way as the hotspot.

| | SIP SIP Hotspot | Basic Settings | |
|---------------------|----------------------|-------------------|---|
| › System | | | |
| > Network | No Registration | | |
| | SIP Hotspot Settings | | |
| > Line | Enable Hotspot: | Disabled v | 0 |
| | Mode: | Client * | 0 |
| > Intercom settings | Monitor Type: | Broadcast V | 0 |
| | Monitor Address: | 224.0.2.0 | 0 |
| > Call List | Local Port: | 16360 | 0 |
| | Name: | SIP Hotspot | 0 |
| > Function Key | Line Settings | | |
| | Line 1: | Enabled • | |
| > Security | Line 2: | Enabled V | |
| | | | |
| > Device Log | | Apply | |
| | | | |
| > Security Settings | | | |
| | | | |
| | | | |
| | | | |
| | | | |
| | | | |

Picture 14 - SIP hotspot

The device is the hotspot server, and the default extension is 0. The device ACTS as a client, and the extension number is increased from 1 (the extension number can be viewed through the [SIP hotspot] page of the webpage).

Calling internal extension:

- The hotspot server and client can dial each other through the extension number before
- Extension 1 dials extension 0



9 Web Configurations

9.1 Web Page Authentication

Users can log into the device's web page to manage user device information and operate the device. Users must provide the correct user name and password to log in. If the password is entered incorrectly three times, it will be locked and can be entered again after 5 minutes. The details are as follows:

- If an IP is logged in more than the specified number of times with a different user name, it will be locked
- If a user name logs in more than a specified number of times on a different IP, it is also locked

9.2 System >> Information

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

- Model
- Hardware Version
- Software Version
- Uptime
- Last uptime
- MEMinfo
- System Time

And summarization of network status,

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

Besides, summarization of SIP account status,

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



9.3 System >> Account

| | Information | Account | Configurations | Upgrade | Auto Provision | FDMS | Tools | Reboot Phone |
|---------------------|-----------------------------|--------------------------|----------------|-------------------------|-------------------------|------|-------|---------------------------------------|
| > System | | | | | | | | NOTE |
| > Network | Add New User Username | | | | 0 | | | Description: Set or modify the log |
| > Line | Web Authent Confirm Pass | ication Password word | | | 0 | | | user name and pass |
| > Intercom settings | Privilege | | | Administrators v | 0 | | | |
| › Call List | User Accounts | User | | | Privilege | | | |
| > Function Key | | admin guest | | | Administrators Users | | | |
| > Security | User Managemer | nt | | | | | | |
| > Device Log | admin 🔻 | | | Delete | Aodify | | | |
| > Security Settings | | | | | | | | |
| | | | | | | | | |
| | | | | | | | | |
| | | | | | | | | |

Picture 15- WEB Account

On this page the user can change the password for the login page. Users with administrator rights can also add or delete users, manage users, and set permissions and passwords for new users

9.4 System >> Configurations

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



| | Information Account | Configurations | Upgrade | Auto Provision | FDMS | Tools | Reboot Phone |
|------------------|-------------------------|----------------|-----------------------|--|---------|-------|--|
| System | | | | | | | NOTE |
| Network | export Configurations 🕖 | | | | | | Description: |
| | | | | ations in 'txt' format. Jurations in 'txt' format | | | This page is used manage configura |
| Line | | | | ations in 'xml' format. | | | the phone, includ import/export |
| ntercom settings | mport Configurations 🕜 | | | | | | configuration, res configuration partly/totally. |
| ** | inport comgurations | Configuration | file: | Sel | ect Imp | ort | purary county. |
| List – | | | | | | | |
| | lear Configuration >> 🕜 | Click "Close" | outton to reset the | configuration files | | | |
| tion Key | Content to Keep | | outton to reset the (| Content to | Reset | | |
| v. | MMI | | | DSS KEY | | * | |
| | BASIC NETWORI | < | | TR069 | | | |
| g | AUTOPROVISION | 4 | | | | | |
| Settings | | | _→] | | | | |
| | | | + | | | | |
| | | | | | | | |
| | | * | | | | * | |
| | | | Clear | | | | |
| | :lear Tables >> 🕜 | | | | | | |

Picture 16 - System Setting

Export Configurations

Right click to select target save as, that is, to download the device's configuration file, suffix ".txt". (note: profile export requires administrator privileges)

Import Configurations

Import the configuration file of Settings. The device will restart automatically after successful import, and the configuration will take effect after restart

Clear Configurations

Select the module in the configuration file to clear.

SIP: account configuration.

AUTOPROVISION: automatically upgrades the configuration

TR069:TR069 related configuration

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

DSS Key: DSS Key configuration

Clear Tables

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

Reset Phone

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



9.5 System >> Upgrade

| | Information Account Configurations Upgrade Auto Provision FDMS Tools | Reboot Phone |
|-------------------|--|---|
| > System | | NOTE |
| Network | Software upgrade Current Software Version: T0.0.17 | Description: |
| Line | System Image File: Select Upgrade | This page is use upgrade some f phone, includin firmware, ring t |
| Intercom settings | Upgrade Server Upgrade Server Address1: | initiate, ring e |
| Call List | Upgrade Server Address2: | |
| Function Key | Firmware Information | |
| | Current Software Version: T0.0.17 Server Firmware Version: Checking | |
| Security | Upgrade New Firmware Information: | |
| Device Log | New Firmware Information: | |
| Security Settings | Ring Upgrade 🍘 | |
| | Load Server File: Select (*.wav, *.mp3) Upload | |
| | Ring List 🜒 | |
| | Index File Name File Size | |
| | Delete | |

Picture 17- Upgrade

Upgrade the software version of the device, and upgrade to the new version through the webpage. After the upgrade, the device will automatically restart and update to the new version. Click select, select the version and then click upgrade. Upgrade the ringtone, support way and MP3 format.

Firmware Upgrade:

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

| | Information | Account Configural | ions Upgrade | Auto Provision | FDMS | Tools | Reboot Phone |
|---------------------|-----------------|-------------------------------------|--------------|----------------|---------------|-----------|---|
| > System | | | | | | | NOTE |
| > Network | Software upgra | de 🕜 Current Software Version: | T0.0.17 | | | | Description: This page is used |
| › Line | | System Image File: | | Select | Upgrade | | upgrade some fil phone, including firmware, ring to |
| > Intercom settings | | rver Address1: rver Address2: | | | | | |
| › Call List | opgrade Se | river Addressz: | Appl | y | | | |
| › Function Key | Firmware Inform | nation Current Software Version: | T0.0.17 | | | | |
| › Security | | Server Firmware Version: | Checking | | | | |
| > Device Log | | New Firmware Information: | | | | | |
| > Security Settings | Ring Upgrade 💡 |). | | | | | |
| | | Load Server File: | | Select | (*.wav, *.mp3 | 3) Upload | |
| | Ring List 🕜 | | | | | | |
| | | Index | File Name | | File Size | | |
| | | | | | | Delete | |



Picture 18 - Web page firmware upgrade

Table 10- Firmware upgrade

| Parameter | Description | | |
|--------------------------|--|--|--|
| Upgrade server | | | |
| | Enable automatic upgrade, If there is a new version txt | | |
| Enable Auto Upgrade | 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. | | |
| | If there is a new version txt and new software firmware | | |
| [] Ingrada] button | on the server, the page will display version information | | |
| [Upgrade] button | and upgrade button will become available; Click | | |
| | [Upgrade] button to upgrade the new firmware. | | |
| New version description | When there is a corresponding TXT file and version on | | |
| information | the server side, the TXT and version information will be | | |
| | displayed under the new version description information. | | |

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

| 名称 ^ | 修改日期 | 类型 | 大小 |
|------------------------------------|-----------------|------------|-----------|
| fanvil_x6_hwv1_0.txt | 2018/9/11 17:57 | 文本文档 | 1 KB |
| fanvil_x6_hwv1_1.txt | 2018/9/11 17:57 | 文本文档 | 1 KB |
| fanvil_x6_hwv1_2.txt | 2018/9/11 17:57 | 文本文档 | 1 KB |
| fanvil x6 hwv1 3.txt | 2018/9/11 17:57 | 文本文档 | 1 KB |
| a6-6904-P0.12.12-1.6.3-2502T2018-0 | 2018/8/21 19:52 | WinRAR 压缩文 | 35,847 KB |

- TXT file format must be UTF-8
- vendor_model_hw10.TXT The file format is as follows: Version=1.6.3 #Firmware



Firmware=xxx/xxx.z #URL , Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers. BuildTime=2018.09.11 20:00 Info=TXT|XML

Xxxxx Xxxxx Xxxxx

Xxxxx

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

9.6 System >> Auto Provision

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

| | Information | Account | Configurations | Upgrade | Auto Provision | FDMS | Tools | Reboot Phone |
|---------------------------------------|--------------------------------|---|----------------|-----------------|-------------------|------|-------|---|
| > System | | | | | | | | NOTE |
| > Network | Basic Settings CPE Serial M | lumber: | | 00100400FV02001 | 000000c3b4c8c1597 | | 0 | Description: Auto Provision |
| > Line | | ion Password: | | | | | 0 | to realize remote/autom installation and delpoyment co |
| Intercom settings | General Cor | on File Encryption & Infiguration File Enc ail Check Times: | | 1 | | | 0 | and some othe files. |
| › Call List | Save Auto F | Provision Informatio | | | | | 0 | |
| > Function Key | Enable Serv DHCP Option >: | | | 0 | | | 0 | |
| > Security | DHCPv6 Option >> | | | | | | | |
| > Device Log | SIP Plug and Pl | | | | | | | |
| Security Settings | Autoprovision M | low >> | | | | | | |
| | TR069 >> | | Apply | | | | | |
| | | | | | | | | |
| | | | | | | | | |
| | | | | | | | | |

Picture 19- 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

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

Table 11- Auto Provision



| Auto provision | | | | |
|-----------------------|---|--|--|--|
| Parameters | Description | | | |
| Basic settings | | | | |
| | Shows the current config file's version. If the version of the | | | |
| | downloaded configuration file is same with this one, the | | | |
| Current Configuration | configuration file will not be applied. If the device confirm the | | | |
| Version | configuration by the Digest method, once the configuration of | | | |
| | server is modified or the device's configurations are different from | | | |
| | server's, the device will download and apply the configurations. | | | |
| | Shows the common config file's version. If the version of the | | | |
| | downloaded configuration file is same with this one, the | | | |
| General | configuration file will not be applied. If the device confirm the | | | |
| Configuration Version | configuration by the Digest method, once the configuration of | | | |
| | server is modified or the device's configurations are different from | | | |
| | server's, the device will download and apply the configurations. | | | |
| CPE Serial Number | Serial number of the equipment | | | |
| Authentication Name | Username for configuration server. Used for FTP/HTTP/HTTPS. | | | |
| Authentication Name | If this is blank the phone will use anonymous | | | |
| Authentication | Password for configuration server. Used for FTP/HTTP/HTTPS. | | | |
| Password | | | | |
| Configuration File | Encryption key for the configuration file | | | |
| Encryption Key | | | | |
| General | | | | |
| Configuration File | Encryption key for common configuration file | | | |
| Encryption Key | | | | |
| Download Fail Check | The default value is 5. If the download configuration fails, it will be | | | |
| Times | downloaded 5 times. | | | |
| Enable Get Digest | When the feature is enable, if the configuration of server is | | | |
| From Server | changed, phone will download and update. | | | |
| DHCP Option | | | | |
| Option Value | The equipment supports configuration from Option 43, Option 66, | | | |
| | or a Custom DHCP option. It may also be disabled. | | | |
| Custom Option Value | Custom option number. Must be from 128 to 254. | | | |
| Enable DHCP Option | Set the SIP server address through DHCP option 120. | | | |
| 120 | | | | |
| SIP Plug and Play (F | nP) | | | |
| Enable SIP PnP | Vhether 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 | g Server |
| | Set FTP/TFTP/HTTP server IP address for auto update. The address |
| Server Address | can be an IP address or Domain name with subdirectory. |
| | The configuration file name. If it is empty, phone will request the |
| Configuration File | common file and device file which is named as its MAC address. |
| Name | 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 |
| | Configuration file update interval time. As default it is 1, means |
| Update Interval | phone will check the update every 1 hour. |
| | Provision Mode. |
| | 1. Disabled. |
| Update Mode | 2. Update after reboot. |
| | 3. Update after interval. |
| TR069 | |
| Enable TR069 | Enable TR069 after selection |
| Enable TR069 | |
| Warning Tone | If TR069 is enabled, there will be a prompt tone when connecting. |
| 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) |
| STUN | |
| server address | Enter the STUN address |
| Enable the STUN | Enable the STUN |
| TLS Version | TLS Version |
| | · · · · · · · · · · · · · · · · · · · |



9.7 System >> FDMS

| | Information | Account | Configurations | Upgrade | Auto Provision | FDMS | Tools | Reboot Phone |
|---------------------|------------------------------|---------|----------------|---------|----------------|------|-------|--------------|
| > System | | | | | | | | |
| > Network | FDMS Info Setti Community | | | | | | | |
| > Line | Building Nur Room Numb | | | | | | | |
| › Intercom settings | | | | ŀ | Apply | | | |
| › Call List | | | | | | | | |
| › Function Key | | | | | | | | |
| › Security | | | | | | | | |
| > Device Log | | | | | | | | |
| > Security Settings | | | | | | | | |
| | | | | | | | | |
| | | | | | | | | |

Picture 20 - FDMS

| Table | 12- | FDMS |
|-------|-----|------|
| 1 | | |

| FDMS information Settings | | |
|---------------------------|--|--|
| Community Designations | Name of equipment installation community | |
| Building a movie theater | Name of equipment installation building | |
| room number | Equipment installation room name | |

9.8 System >> Tools

Account Configurations Upgrade Auto Provision FDMS Tools Reboot Phone Information > System NOTE Syslog > Network Description: Some tools to help administrators or technicians to anal issues. Enable Syslog: 0.0.0.0 000 Server Address: > Line Server Port: APP Log Level: 514 Warning • > Intercom settings Export Log: Apply > Call List Web Capture 🕜 > Function Key Start stop Watch Dog > Security Apply Enable Watch Dog: > Device Log > Security Settings

This page gives the user the tools to solve the problem.

Picture 21 - Tools



Syslog: When enabled, set the syslog software address, and log information of the device will be recorded in the syslog software during operation. If there is any problem, log information can be analyzed by Fanvil technical support.

9.9 Network >> Basic

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

| | Basic Service Port | VPN Advanced | | |
|---------------------|--|---|-------|--|
| › System | | | | |
| > Network | Network Mode 🥝 Network Mode: | IPv4 Only • | | |
| > Line | IPv4 Network Status | | | |
| > Intercom settings | IP: Subnet mask: Default gateway: | 172.16.8.129 255.255.255.0 172.16.8.1 | | |
| › Call List | MAC: | 0c:3b:4c:8c:15:97 | | |
| › Function Key | IPv4 Settings Static IP | DHCP () | PPPoE | |
| > Security | Enable Vendor Identifier: Vendor Identifier: | Disabled v Fanvil PA3 | 6 | |
| > Device Log | DNS Server Configured by: Primary DNS Server: | DHCP • | 6 | |
| > Security Settings | Secondary DNS Server : DNS Domain: | 114.114.114 | 0 | |
| | | Apply | | |
| | | | | |
| | | | | |

Picture 22 - Network Basic Setting

| Table 1 | 13 - | Network | Basic | Setting |
|---------|------|---------|-------|---------|
|---------|------|---------|-------|---------|

| Field | Explanation | | |
|--|---|--|--|
| Name | | | |
| Network Sta | atus | | |
| IP | The current IP address of the equipment | | |
| Subnet | The current Subnet Mask | | |
| mask | | | |
| Default | The surrent Cotoway ID address | | |
| gateway | The current Gateway IP address | | |
| MAC | The MAC address of the equipment | | |
| MAC Time | Display the time when the device gets the MAC address | | |
| stamp | Display the time when the device gets the MAC address | | |
| Settings | | | |
| Select the appropriate network mode. The equipment supports three network modes: | | | |



| Static IP | Network parameters must be entered manually and will not change. All parameters are provided by the ISP. | |
|-------------------|--|--|
| DHCP | Network parameters are provided automatically by a DHCP server. | |
| PPPoE | Account and Password must be input manually. These are provided by | |
| | your ISP. | |
| If Static IP is c | hosen, the screen below will appear. Enter values provided by the ISP. | |
| DNS Server | | |
| Configured | Select the Configured mode of the DNS Server. | |
| by | | |
| Primary DNS | Enter the conver address of the Primary DNS | |
| Server | Enter the server address of the Primary DNS. | |
| Secondary | | |
| DNS Server | Enter the server address of the Secondary DNS. | |
| attention: | | |

1) After setting the parameters, click 【Apply】 to take effect.

2) If you change the IP address, the webpage will no longer responds, please enter the new IP address in web browser to access the device.

3) If the system USES DHCP to obtain IP when device boots up, and the network address of the DHCP Server is the same as the network address of the system LAN, then after the system obtains the DHCP IP, it will add 1 to the last bit of the network address of LAN and modify the IP address segment of the DHCP Server of LAN. If the DHCP access is reconnected to the WAN after the system is started, and the network address assigned by the DHCP server is the same as that of the LAN, then the WAN will not be able to obtain IP access to the network

9.10Network >> service port

This page provides the settings of webpage login protocol, protocol port and RTP port.



| | Basic Service Port | VPN Advanced | |
|---------------------|-----------------------|--------------------|---|
| › System | | | |
| > Network | Service Port Settings | | |
| | Web Server Type: | HTTP V | 0 |
| > Line | Web Logon Timeout: | 15 (10~30)Minute | 0 |
| | web auto login: | | |
| > Intercom settings | HTTP Port: | 80 | 0 |
| 7 Intercom setungs | HTTPS Port: | 443 | 0 |
| › Call List | RTP Port Range Start: | 10000 (1025~65530) | 0 |
| | RTP Port Quantity : | 1000 (10~1000) | 0 |
| › Function Key | | Apply | |
| › Security | | | |
| > Device Log | | | |
| > Security Settings | | | |
| | | | |
| | | | |
| | | | |
| | | | |
| | | | |



Table 14- Server Port

| parameter | description |
|----------------------|--|
| Web server type | Restart after setting takes effect. Optional web login as |
| | HTTP/HTTPS |
| Web login timeout | The default is 15 minutes, the timeout will automatically log out of |
| | the login page, and you need to log in again |
| Web page automatic | No need to enter the user name and password after the timeout, |
| login | it will automatically log in to the web page. |
| HTTP port | The default is 80, if you want system security, you can set other |
| | port |
| | Such as: 8080, web page login: HTTP://ip:8080 |
| HTTPS port | The default is 443, same as HTTP port usage |
| RTP port start range | The value range is 1025-65535. The value of rtp port starts from |
| | the initial value set. Each time a call is made, the value of the |
| | voice and video ports is increased by 2 |
| RTP port quantity | Number of calls |



9.11 VPN

| | Basic Service Port | VPN Advanced | |
|-------------------|--|------------------|-----------|
| System | | | |
| Network | Virtual Private Network (VPN) S VPN IP Address: | tatus 0.0.0.0 | |
| Line | VPN Mode | | |
| | Enable VPN: | | 0 |
| Intercom settings | Enable NAT: | | • |
| | L2TP: | OpenVPN: | |
| Call List | Open VPN mode: | tun 🔻 | 0 |
| Function Kou | Layer 2 Tunneling Protocol (L2T | p) | |
| Function Key | | | |
| | L2TP Server Address: | 0.0.0.0 | 0 |
| Security | Authentication Name: | | 0 |
| | Authentication Password: | | 0 |
| Device Log | | | |
| | | Apply | |
| Security Settings | OpenVPN Files 🕜 | | |
| | Load OpenVPN File | Select | Upload |
| | Certificates List 🕖 | | |
| | Index | File Name | File Size |
| | | | Delete |

Picture 24- 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 page [Network] -> [VPN]. In VPN Mode, check the "Enable VPN" option and select "L2TP", then fill in the L2TP server address, Authentication Username, and Authentication Password in the L2TP section. Press "Apply" then the device will try to connect to the L2TP server.

When the VPN connection established, the VPN IP Address should be displayed in the VPN


status. There may be some delay of the connection establishment. User may need to refresh the page to update the status.

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

OpenVPN

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

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

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

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



9.12 Network >> Advanced

| | Basic Service Port | VPN Adv | anced | |
|-------------------|---------------------------------|-----------------|--------------------------|----------------|
| ystem | | | | |
| Network | Link Layer Discovery Protocol | (LLDP) Settings | Packet Interval:(1~3600) | 60 second(s) 🔮 |
| ine | Enable Learning Function: | | | occorra(o) |
| | Cisco Discovery Protocol (CDP |) Settings | | |
| Intercom settings | Enable CDP: | | Packet Interval:(1~3600) | 60 |
| Call List | DHCP VLAN Settings | | | |
| con cist | Option Value: | Disabled | v 🔞 | |
| Function Key | DHCP Option Vlan(128-254) | : 0 | | |
| | Quality of Service (QoS) Settin | igs | | |
| Security | Enable DSCP: | . 🕜 | Signal DSCP: | 46 (0~63) 🕜 |
| | Audio DSCP: | 46 (0~63) 🔮 | Video DSCP: | 46 (0~63) 🕜 |
| Device Log | ARP Cache Life | | | |
| Security Settings | ARP Cache Life | 2Minute 🕜 | | |
| | WAN VLAN Settings | | | |
| | Enable VLAN: | | WAN VLAN ID: | 256 (0~4095) 🕑 |
| | 802.1p Signal Priority: | 0 (0~7) | 802.1p Media Priority: | 0 (0~7) 🕜 |
| | | | Apply | |
| | 802.1X Settings | | | |
| | 802.1x Mode: | Off 🔻 | | 0 |
| | Identity: | admin | | 0 |
| | Password: | | | 0 |

Picture 25 - Network Setting

Network advanced Settings are typically configured by IT administrators to improve the quality of device service.

| Field Name | Explanation |
|--------------------------|---|
| LLDP Settings | |
| Enable LLDP | Enable or disable LLDP |
| Packet Interval | LLDP Send detection cycle |
| Enable Learning Function | Learn the discovered device information on the device |
| QoS Settings | |
| Pattern | Voice quality assurance (off by default) |
| DHCP VLAN Settings | |
| parameters values | 128-254, Obtain the VLAN value through DHCP |
| WAN port virtual Wan | |
| WAN port virtual Wan | WAN port Settings |
| LAN port virtual LAN | |
| LAN port virtual LAN | LAN port Settings |
| 802.1X | |
| Enable 802.1X | Enable or disable 802.1X |
| Username | Confirm Username |



9.13LINES >> SIP

| | Line 1356@SIP· • | | | | |
|--|---|---|---|----------------|------|
| Network | Line 1356@SIP· • | | | | |
| Network | Line Isso@SIP V | | | | |
| > Line | | | | | |
| | Register Settings >> Line Status: Registered | Ac | tivate: | | |
| | Username: 1356 | | ithentication User: | | 0 |
| Intercom settings | Display name: | | uthentication Password: | | 0 |
| | Realm: | 🕜 Se | erver Name: | | 0 |
| Call List | | 1.11 | 020070-2 | | |
| Function Key | SIP Server 1: Server Address: 172.16.1.2 | | P Server 2: erver Address: | | 0 |
| 2 A M CHUININ (1998) | Server Address: 172.16.1.2 Server Port: 5060 | | erver Port: | 5060 | 0 |
| Security | Transport Protocol: |) Tra | ansport Protocol: | UDP V | 1 |
| | Registration Expiration: 3600 | second(s) 🕜 Re | gistration Expiration: | 3600 second(s) | 0 |
| Device Log | Proxy Server Address: | Ba | ackup Proxy Server Address: | | 0 |
| Security Settings | Proxy Server Port: 5060 | | ackup Proxy Server Port: | 5060 | 0 |
| | Proxy User: | 0 | | | |
| | Proxy Password: | 0 | | | |
| | Basic Settings >> | | | | |
| | | | | | |
| | Codecs Settings >> 😵 | | | | |
| | | | | | |
| | Advanced Settings >> SIP Global Settings >> | pply | | | |
| Basic Settings >> Enable Auto Answering: | Advanced Settings >> SIP Global Settings >> | Auto Answering | g Delay: 0 | (0~120)second | 1(s) |
| | Advanced Settings >> SIP Global Settings >> | | g Delay: 0 | (0~120)second | d(s) |
| Enable Auto Answering: Enable Hotline: | Advanced Settings >> SIP Global Settings >> | Auto Answering | | (0~120)second | d(s) |
| Enable Auto Answering: Enable Hotline: Hotline Delay: | Advanced Settings >> SIP Global Settings >> | Auto Answering | | | d(s) |
| Enable Auto Answering: Enable Hotline: Hotline Delay: Dial Without Registered: | Advanced Settings >> SIP Global Settings >> | Auto Answering O Hotline Numbe | er: | | d(s) |
| Enable Auto Answering: Enable Hotline: Hotline Delay: Dial Without Registered: DTMF Type: | Advanced Settings >> SIP Global Settings >> | Auto Answering | er: | | d(s) |
| Enable Auto Answering: Enable Hotline: Hotline Delay: Dial Without Registered: | Advanced Settings >> SIP Global Settings >> | Auto Answering O Hotline Numbe | er: | | d(s) |
| Enable Auto Answering: Enable Hotline: Hotline Delay: Dial Without Registered: DTMF Type: Request With Port: Use STUN: | Advanced Settings >> SIP Global Settings >> | Auto Answering Hotline Numbe DTMF SIP INFO Use VPN: | er: D Mode: Send 10/1 | | d(s) |
| Enable Auto Answering: Enable Hotline: Hotline Delay: Dial Without Registered: DTMF Type: Request With Port: | Advanced Settings >> SIP Global Settings >> | Auto Answering Hotline Numbe DTMF SIP INFO | er: D Mode: Send 10/1 ☑ ? :: □ ? | | d(s) |





Picture 26- SIP

Table 16 - SIP

| 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. |
| Server Address | Enter the IP or FQDN address of the SIP server |
| Server Port | Enter the SIP server port, default is 5060 |



| Server Conference | Set the conference room number when conference type is set to be | |
|-----------------------------------|---|--|
| | conference by dialing to a conference room on the server | |
| | device itself, maximum supports two remote parties, Server=set up call | |
| Conference Type | Set the type of call conference, Local=set up call conference by the | |
| Transfer Timeout | Set the timeout of call transfer process | |
| Answer | | |
| Call Forward Delay for No | Set the delay time of not answered call before being forwarded | |
| Call Forward Number for No Answer | Set the number of call forward on no answer | |
| Coll Converd Number (| the number specified in the next field | |
| Answer | answered within the configured delay time, the call will be forwarded to | |
| Call Forward on No | Enable call forward on no answer, when an incoming call is not | |
| Busy | | |
| Call Forward Number for | will be forwarded to the number specified in the next field Set the number of call forward on busy | |
| Call Forward on Busy | Enable call forward on busy, when the phone is busy, any incoming call | |
| Unconditional | | |
| Call Forward Number for | Set the number of unconditional call forward | |
| Unconditional | to the number specified in the next field | |
| Call Forward | Enable unconditional call forward, all incoming calls will be forwarded | |
| | answered it | |
| Auto Answering Delay | Set the delay for incoming call before the system automatically | |
| | automatically after the delay time | |
| Enable Auto Answering | Enable auto-answering, the incoming calls will be answered | |
| Basic Settings | | |
| Backup Proxy Server Port | Enter the backup proxy server port, default is 5060 | |
| Address | | |
| Backup Proxy Server | Enter the IP or FQDN address of the backup proxy server | |
| Proxy User Proxy Password | Enter the SIP proxy user Enter the SIP proxy password | |
| Proxy Server Port | Enter the SIP proxy user | |
| SIP Proxy Server Address | Enter the IP or FQDN address of the SIP proxy server | |
| | Enter the SIP domain if requested by the service provider | |
| Activate | Whether the service of the line should be activated | |
| Display Name | Enter the display name to be sent in a call request. | |
| Username | Enter the username of the service account. | |
| Authentication Password | Enter the authentication password of the service account | |
| | | |



| Messageenabled, the device will receive notification from the server if there is voice message NumberVoice Message NumberSet the number for retrieving voice messageVoice Message SubscribeSet the interval of voice message notification subscriptionPeriodEnable HotlineEnable HotlineEnable 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 headphoneHotline DelaySet the delay for hotline before the system automatically dialed it Hotline NumberDial Without RegisteredSet call out by proxy without registrationEnable Missed Call LogIf enabled, the phone will save missed calls into the call history record. DTMF TypeDTMF SIP INFO ModeSet the SIP INFO mode to send "" and "# or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the line to use VPN restrict routeUse VPNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 | | | | |
|--|---|--|--|--|
| Message 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 Set the interval of voice message notification subscription Period Enable Hotline 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 dolar for headphone 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 expiration interval Use VPN Set the SIP expiration interval Use VPN Set the line to use VPN restrict route Use STUN Set the priority and availability of the codecs by adding or remove them from the list. Advanced Settings Set the feature code to dial to the server 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 ord | Number | Server | | |
| voice message waiting on the serverVoice Message NumberSet the number for retrieving voice messageVoice Message SubscribeSet the interval of voice message notification subscriptionPeriodEnable HotlineEnable HotlineEnable 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 headphoneHotline DelaySet the delay for hotline before the system automatically dialed it Hotline NumberDial Without RegisteredSet call out by proxy without registrationEnable Missed Call LogIf enabled, the phone will save missed calls into the call history record.DTMF TypeSet the DTMF type to be used for the lineDTMF SIP INFO ModeSet the SIP INFO mode to send ** and '#' or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the line to use VPN restrict routeUse VPNSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsSet the feature code to dial to the serverUse Feature CodeWhen 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 to the server by dialing the number specified in each feature code to the server by dialing the number serverDisable DNDSet the feature code to dial to the serverDisable Call Forward Uncondition | Subscribe For Voice | Enable the device to subscribe a voice message waiting notification, if | | |
| Voice Message Number Set the number for retrieving voice message Voice Message Subscribe Set the interval of voice message notification subscription Period 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 Hotine Number Set the hotline dialing number Dial Without Registered Set the DTMF type to be used for the line DTMF Type 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 Use VPN Set the line to use VPN restrict route Use STUN Set the line to use STUN for NAT traversal Codec Settings Set the feature code to dial to 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 to the server by dialing the number specified in each feature code to the server by dialing the number server Unconditional Set the feature code to dial to the server Disable DND Set the feature code to dial to the server Disable DND Set the feature code to dial to the server | Message | enabled, the device will receive notification from the server if there is | | |
| Voice Message Subscribe Set the interval of voice message notification subscription Period 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 Registration Expiration Set the SIP expiration interval Use VPN Set the line to use VPN restrict route Use STUN Set the priority and availability of the codecs by adding or remove them from the list. Advanced Settings Set the feature code to dial to the server 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 control the | | voice message waiting on the server | | |
| Period Enable hotline 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 Registration Expiration Set the SIP expiration interval Use VPN Set the line to use VPN restrict route Use STUN Set the priority and availability of the codecs by adding or remove them from the list. Advanced Settings Set the feature code to dial to the server 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 to the server by dialing the numbe | Voice Message Number | Set the number for retrieving voice message | | |
| Enable HotlineEnable 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 headphoneHotline DelaySet the delay for hotline before the system automatically dialed itHotline NumberSet the hotline dialing numberDial Without RegisteredSet call out by proxy without registrationEnable Missed Call LogIf enabled, the phone will save missed calls into the call history record.DTMF TypeSet the DTMF type to be used for the lineDTMF SIP INFO ModeSet the SIP INFO mode to send '*' and '#' or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 field.Enable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call ForwardSet the feature code to dial to the server | Voice Message Subscribe | Set the interval of voice message notification subscription | | |
| immediately at audio channel opened by off-hook handset or turn on hands-free speaker or headphoneHotline DelaySet the delay for hotline before the system automatically dialed itHotline NumberSet the hotline dialing numberDial Without RegisteredSet call out by proxy without registrationEnable Missed Call LogIf enabled, the phone will save missed calls into the call history record.DTMF TypeSet the DTMF type to be used for the lineDTMF SIP INFO ModeSet the SIP INFO mode to send ** and *# or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsSet the features is 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 field.Enable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call Forward | Period | | | |
| Hands-free speaker or headphoneHotline DelaySet the delay for hotline before the system automatically dialed itHotline NumberSet the hotline dialing numberDial Without RegisteredSet call out by proxy without registrationEnable Missed Call LogIf enabled, the phone will save missed calls into the call history record.DTMF TypeSet the DTMF type to be used for the lineDTMF SIP INFO ModeSet the SIP INFO mode to send '*' and '#' or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsSet the feature code to dial to the serverUse Feature CodeWhen 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 serverDisable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverDisable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverBusySet the feature code to dial to the server | Enable Hotline | Enable hotline configuration, the device will dial to the specific number | | |
| 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 Registration Expiration Set the SIP expiration interval Use VPN Set the line to use STUN for NAT traversal Codec Settings Set the priority and availability of the codecs by adding or remove them from the list. Advanced Settings Set the feature 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 Disable DND Set the feature code to dial to the server Disable DND Set the feature code to dial to the server Disable Call Forward Set the feature code to dial to the server Unconditional Set the feature code to dial to the server Disable Call Forward on Set the feature code to d | | immediately at audio channel opened by off-hook handset or turn on | | |
| Hotline NumberSet the hotline dialing numberDialWithoutRegisteredSet call out by proxy without registrationEnable Missed Call LogIf enabled, the phone will save missed calls into the call history record.DTMF TypeSet the DTMF type to be used for the lineDTMF SIP INFO ModeSet the SIP INFO mode to send '*' and '#' or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsSet the faiture code to dial to the serverUse Feature CodeWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward on BusySet the feature code to dial to the server | | hands-free speaker or headphone | | |
| DialWithoutRegisteredSet call out by proxy without registrationEnable Missed Call LogIf enabled, the phone will save missed calls into the call history record.DTMF TypeSet the DTMF type to be used for the lineDTMF SIP INFO ModeSet the SIP INFO mode to send ** and *# or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward on BusySet the feature code to dial to the server | Hotline Delay | Set the delay for hotline before the system automatically dialed it | | |
| Enable Missed Call LogIf enabled, the phone will save missed calls into the call history record.DTMF TypeSet the DTMF type to be used for the lineDTMF SIP INFO ModeSet the SIP INFO mode to send '*' and '#' or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverDisable Call Forward on BusySet the feature code to dial to the server | Hotline Number | Set the hotline dialing number | | |
| DTMF TypeSet the DTMF type to be used for the lineDTMF SIP INFO ModeSet the SIP INFO mode to send '*' and '#' or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverDisable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverBusySet the feature code to dial to the server | Dial Without Registered | Set call out by proxy without registration | | |
| DTMF SIP INFO ModeSet the SIP INFO mode to send '*' and '#' or '10' and '11'Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverDisable Call Forward on BusySet the feature code to dial to the server | Enable Missed Call Log | If enabled, the phone will save missed calls into the call history record. | | |
| Enable DNDEnable Do-not-disturb, any incoming call to this line will be rejected automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward BusySet the feature code to dial to the server | DTMF Type | Set the DTMF type to be used for the line | | |
| automaticallyRegistration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 DNDSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverEnable Call Forward on BusySet the feature code to dial to the server | DTMF SIP INFO Mode | Set the SIP INFO mode to send '*' and '#' or '10' and '11' | | |
| Registration ExpirationSet the SIP expiration intervalUse VPNSet the line to use VPN restrict routeUse STUNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward on BusySet the feature code to dial to the server | Enable DND | Enable Do-not-disturb, any incoming call to this line will be rejected | | |
| Use VPNSet the line to use VPN restrict routeUse STUNSet the line to use STUN for NAT traversalCodec SettingsSet the priority and availability of the codecs by adding or remove them from the list.Advanced SettingsWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward on BusySet the feature code to dial to the server | | automatically | | |
| Use STUN Set the line to use STUN for NAT traversal Codec Settings Set the priority and availability of the codecs by adding or remove them from the list. Advanced Settings Very the priority and availability of the codecs by adding or remove them from the list. Advanced Settings 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 Disable Call Forward Unconditional Set the feature code to dial to the server Disable Call Forward on Busy Set the feature code to dial to the server | Registration Expiration | Set the SIP expiration interval | | |
| Codec Settings Set the priority and availability of the codecs by adding or remove them from the list. Advanced Settings Men 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 Disable Call Forward Set the feature code to dial to the server Disable Call Forward on Set the feature code to dial to the server Busy Set the feature code to dial to the server | Use VPN | Set the line to use VPN restrict route | | |
| 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 Disable Call Forward Set the feature code to dial to the server Disable Call Forward on Set the feature code to dial to the server Enable Call Forward on Set the feature code to dial to the server | Use STUN | Set the line to use STUN for NAT traversal | | |
| Advanced Settings Use Feature Code When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field. Enable DND Set the feature code to dial to the server Disable DND Set the feature code to dial to the server Enable Call Forward Set the feature code to dial to the server Disable Call Forward Set the feature code to dial to the server Enable Call Forward on Set the feature code to dial to the server Busy Set the feature code to dial to the server | Codec Settings | Set the priority and availability of the codecs by adding or remove them | | |
| Use Feature CodeWhen 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverEnable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward on BusySet the feature code to dial to the server | | from the list. | | |
| 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverEnable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverEnable Call Forward on BusySet the feature code to dial to the server | Advanced Settings | | | |
| 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 DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverEnable Call Forward UnconditionalSet the feature code to dial to the serverDisable Call Forward UnconditionalSet the feature code to dial to the serverEnable Call Forward UnconditionalSet the feature code to dial to the serverEnable Call Forward on BusySet the feature code to dial to the server | Use Feature Code | When this setting is enabled, the features in this section will not be | | |
| to the server by dialing the number specified in each feature code field.Enable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverEnable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverDisable Call ForwardSet the feature code to dial to the serverEnable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverBusySet the feature code to dial to the server | | handled by the device itself but by the server instead. In order to | | |
| Enable DNDSet the feature code to dial to the serverDisable DNDSet the feature code to dial to the serverEnable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverDisable Call ForwardSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverUnconditionalSet the feature code to dial to the serverBusySet the feature code to dial to the server | | control the enabling of the features, the device will send feature code | | |
| Disable DND Set the feature code to dial to the server Enable Call Forward Set the feature code to dial to the server Unconditional Disable Call Forward Set the feature code to dial to the server Disable Call Forward Set the feature code to dial to the server Unconditional Set the feature code to dial to the server Unconditional Set the feature code to dial to the server Enable Call Forward on Set the feature code to dial to the server Set the feature code to dial to the server Busy Set the feature code to dial to the server | | to the server by dialing the number specified in each feature code field. | | |
| Enable Call Forward Set the feature code to dial to the server Unconditional Disable Call Forward Set the feature code to dial to the server Unconditional Set the feature code to dial to the server Set the feature code to dial to the server Enable Call Forward on Set the feature code to dial to the server Busy Set the feature code to dial to the server | Enable DND | Set the feature code to dial to the server | | |
| Unconditional Set the feature code to dial to the server Disable Call Forward On Set the feature code to dial to the server Enable Call Forward on Set the feature code to dial to the server Busy Set the feature code to dial to the server | Disable DND | Set the feature code to dial to the server | | |
| Disable Call Forward Set the feature code to dial to the server Unconditional Enable Call Forward on Busy Set the feature code to dial to the server | nable Call Forward Set the feature code to dial to the server | | | |
| Unconditional Enable Call Forward on Busy Set the feature code to dial to the server | Unconditional | | | |
| Enable Call Forward on Set the feature code to dial to the server Busy | Disable Call Forward | Set the feature code to dial to the server | | |
| Busy | Unconditional | | | |
| | Enable Call Forward on | Set the feature code to dial to the server | | |
| Disable Call Forward on Set the feature code to dial to the server | Busy | | | |
| | Disable Call Forward on | Set the feature code to dial to the server | | |



| Busy | |
|-------------------------|--|
| Enable Call Forward on | Set the feature code to dial to the server |
| No Answer | |
| Disable Call Forward on | Set the feature code to dial to the server |
| No Answer | |
| Enable Blocking | Set the feature code to dial to the server |
| Anonymous Call | |
| Disable Blocking | Set the feature code to dial to the server |
| Anonymous Call | |
| 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 | Set the feature code to dial to the server |
| Code | |
| Send Anonymous Off | Set the feature code to dial to the server |
| Code | |
| SIP Encryption | Enable SIP encryption such that SIP transmission will be encrypted |
| SIP Encryption Key | Set the pass phrase for SIP encryption |
| RTP Encryption | Enable RTP encryption such that RTP transmission will be encrypted |
| RTP Encryption Key | Set the pass phrase for RTP encryption |
| 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. |
| 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 SI | | |
| | messages above 1500 bytes | | |
| Transport Protocol | Set the line to use TCP or UDP for SIP transmission | | |
| 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 | Whether to add quote in display name, i.e. "Fanvil" vs Fanvil | | |
| Name | | | |
| Enable GRUU | Support Globally Routable User-Agent URI (GRUU) | | |
| Sync Clock Time | Time Sycn with server | | |
| Caller ID Header | Set the Caller ID Header | | |
| Use 182 Response for | Set the device to use 182 response code at call waiting response | | |
| Call waiting | | | |
| 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. | | |
| Enable Feature Sync | Feature Sycn with server | | |
| Enable SCA | Enable/Disable SCA (Shared Call Appearance) | | |
| CallPark Number | Set the callPark number | | |
| Server Expire | | | |
| TLS Version | Choose TLS Version | | |
| | | | |

9.14 Line >> SIP Hotspot

SIP hotspot is a simple and practical function. It is simple to configure, can realize the function of group vibration, and can expand the number of SIP accounts.



See <u>8.3 Hotspot</u> for details.

9.15 Line >> Basic Settings

STUN -Simple Traversal of UDP through NAT -A STUN server allows a phone in a private network to know its public IP and port as well as the type of NAT being used. The equipment can then use this information to register itself to a SIP server so that it can make and receive calls while in a private network.



Picture 27- Basic Settings

| | SIP SIP Hotspot | Basic Settings | | |
|---------------------------------------|--|--|--------|--|
| › System | | | | NOTE |
| > Network | STUN Settings STUN NAT Traversal: | FALSE | 0 | Description: Phone line basic settings, |
| > Line | Server Address: Server Port: | 3478 | 0 0 | including STUN, certificate files. |
| › Intercom settings | Binding Period: SIP Waiting Time: | 50 second(s) 800 millisecond | 0 0 | |
| › Call List | | Apply | | |
| > Function Key | SIP P2P Settings Enable Auto Answering | × | 0 | |
| > Security | Auto Answering Delay: DTMF Type: DTMF SIP INFO Mode: | 0 (0~120)second(s) RFC2833 Send 10/11 | 0 0 | |
| > Device Log | DTHE SIP INFO HOUR. | Apply | v | |
| Security Settings | | | | |

Picture 28 - Line Basic Setting

Table 17- Line Basic Setting

| 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 | Automatically answer incoming IP calls after the timeout period is |
| Answering | enabled |
| Auto Answering | Automatic answer timeout setting |
| Delay | |
| DTMF Type | Set the DTMF type of the line. |
| DTMF SIP INFO | Set SIP INFO mode to send '*' and '#' or '10' and '11' |
| Mode | |

9.16 Intercom settings >> Features

| | Features Media Settings | Camera Settings | MCAS | T Action | Time/ | Date Tim | e Plan | Tone | |
|-----------------------|-------------------------------------|---------------------------|------|------------------------|-----------|---------------|--------|----------------------------------|------------|
| System | | | | | | | | NOTE | |
| CHARLES IN CONTRACTOR | Basic Settings >> | | | | | | | | |
| Network | Enable Call Waiting: | | | | | | | Description: Function setting | is you can |
| | Enable Auto on Hook: | | | Auto HangUp Delay: | 3 | (0~30)second(| s) 🕜 | set the phone f | eatures, |
| Line | Enable Silent Mode: | I | | Disable Mute for Ring: | . 0 | | | settings, tone s | ettings, |
| | | | | | | | | intercom setting | |
| Intercom settings | Ban Outgoing: | | | | | | | settings. | |
| | Default Ans Mode: | Video 🔻 🕜 | | Default Dial Mode: | Video 🔻 🤇 | | | | |
| Call List | Enable Restricted Incoming List: | Ø | | | | | | | |
| | Enable Restricted Outgoing List | :: 🗹 🕜 | | Enable Country Code: | | | | | |
| Function Key | Country Code: | | | Area Code: | | | | | |
| Security | Allow IP Call: | I | | P2P IP Prefix: | - | | | | |
| Device Log | Restrict Active URI Source IP: | | 0 | Push XML Server: | | 0 | | | |
| | Line Display Format: | xxx@SIPn v 🚷 | _ | | | | | | |
| Security Settings | Call Number Filter: | | | Auto Resume Current: | | | | | |
| | Limit Talking Duration: | | | Talking Duration: | 120 | (20~600)secor | nd(s) | | |
| | No Answer Auto HangUp Timeout: | 130 (1~300)second(s) 📀 | | Enable Push XML Auth: | • • | | | | |
| | Tone Settings >> | | | | | | | | |
| | Intercom Settings >> | | | | | | | | |
| | Response Code Settings >> | | | | | | | | |
| | | | A | oply | | | | | |

Picture 29 - Feature

| Table | 18- | Common | device | function | Settings on | the web page |
|-------|-----|--------|--------|----------|-------------|--------------|
| | | | | | | |

| Parameters | Description |
|-----------------------|--|
| Basic Settings | |
| Enable Call Waiting | Enable this setting to allow user to take second incoming call during an |
| | established call. Default enabled. |
| Enchle Auto Llanddoum | The phone will hang up and return to the idle automatically at |
| Enable Auto Handdown | hands-free mode |
| Auto Hondoloum Times | Specify Auto handdown time, the phone will hang up and return to the |
| Auto Handdown Time | idle automatically after Auto Hand down time at hands-free mode, and |



| | play dial tone Auto handdown time at handset mode |
|------------------------------------|--|
| | When enabled, the phone is muted, there is no ringing when calls, you |
| Enable Silent Mode | can use the volume keys and mute key to unmute. |
| Disable Mute for Ring | When it is enabled,you can not mute the phone. |
| Ban Outgoing | If you select Ban Outgoing to enable it, and you cannot dial out any number. |
| Enable Restricted Incoming List | Whether enable Restricted Incoming List |
| Enable Restricted Outgoing List | Wether enable Restricted Outgoing List |
| Enable country Code | Wether enable country Code |
| Country Code | Country Code |
| Area Code | Area Code |
| Allow IP Call | If enabled, user can dial out with IP address |
| P2P IP Prefix | You can set IP call prefix, for example, i set it as "172.16.2.", then i input |
| | #160 in dialpad and press dial key ,it will call 172.16.2.160 automatically |
| Restrict Active URI | Set the device to accept Active URI command from specific IP address. |
| Source IP | More details please refer to this link |
| | https://www.fanvil.com/Support/download/cid/14.html |
| 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. |
| Line Display Format | Line display format including SIPn/SIPn: xxx/xxx@SIPn |
| Call Number Filter | Configure a special character & ,if the number is 78 & 9. The call will be filtered out& |
| Auto Resume Current | If the current path changes, the hold will be automatically resume |
| Limit Talking Duration | Automatically hang up the call after enabling the time set for the call |
| Talking Duration | Call duration ,20-600s |
| No Answer Auto HangUp | If the call is not answered, the call will be automatically hung up after the |
| Timeout | timeout |
| Enable Push XML Auth | To enable push xml auth, user password is required |
| 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 | | |

9.17 Intercom settings >> media



Picture 30- Media Settings

Table 19- Audio Settings

| Parameters | Description |
|-------------------|---|
| Codecs Settings | Select the enabled and disabled voice codecs |
| | codec:G.711A/U,G.722,G.723,G.729,G.726-32, |
| | ILBC,AMR,AMR-WB |
| Audio Settings | |
| Default Ring Type | Set the default ring type. If the caller ID of an incoming call |



| | was not configured with specific ring type, the default ring |
|----------------------------|--|
| | will be used. |
| Speakerphone Volume | Set the speakerphone volume, the value must be 1~9 |
| Speakerphone Ring Volume | Set the ring volume in the speakerphone, the value must |
| | be 1~9 |
| DTMF Payload Type | Enter the DTMF payload type, the value must be 96~127. |
| Opus playload type | Enter the opus payload type, the value must be 96~127. |
| | Set the opus sample rate, including OPUS-NB(8KHz), |
| OPUS Sample Rate | OPUS-WB(16KHz) |
| | |
| ILBC Payload Type | Set the ILBC Payload Type |
| ILBC Payload Length | Set the ILBC Payload Length |
| Enable VAD | Enable Voice Activity Detection. When enabled, the |
| | device will suppress the audio transmission with artificial |
| | comfort noise signal to save the bandwidth. |
| RTP Control Protocol(RTC | P) Settings |
| CNAME user | Set the CNAME user |
| CNAME host | Set the CNAME host |
| RTP | |
| RTP keep alive | Keep talking, send a packet 30 seconds after enable it |
| Alert Info Ring Settings (| alert-info) |
| Value of notification | Set the value of the specified ring type |
| message 1 to 10 | |
| ring type | The ring type |
| - | |

9.18 Intercom settings>>Camera Settings

Customers can configure camera related parameters and adjust video coding related settings.



| <u>)</u> | Features Media Settin | gs Camera Settings | MCAST Action | Time/Date Time Plan | Tone |
|-----------------------|-----------------------|--------------------|------------------------------------|---------------------|---|
| System | | | | | NOTE |
| | Camera Settings | | | | in the second |
| Network | White Balance Mode: | Auto Mode | Exposure Mode: | Auto Mode 🔹 🕐 | Description: configure Camera rela |
| 1998) | Exposure Time: | 10000 (0~10000) |) 🕜 Exposure Gain: | 1024 (0~1024) | parameters, Camera |
| Line | Contrast Mode: | Auto Mode 🔻 🕜 | Contrast: | 50 (0~100) | Settings |
| | Saturation Mode: | Auto Mode 🔻 🕜 | Saturation: | 32 (0~200) 🥝 | |
| Intercom settings | Sharpness Mode: | Auto Mode 🔻 🕜 | Sharpness: | 256 (0~1023) | |
| | Wide Dynamic: | Disabled 🔻 🕜 | WDR: | 0 (0~10) 🕜 | |
| Call List | Enable IRCUT: | Sync 🔻 🕜 | | | |
| | Image Mode | Auto 🔻 🕜 | Brightness: | 50 (0~100) 🕜 | |
| Function Key | Enable Onvif: | Disable 🔻 🕜 | Call Stream: | Main Stream V | |
| | Enable Onvif Auth: | Enable 🔻 🕜 | Enable Rtsp Auth: | Enable 🔻 🕜 | |
| Security | H.264 Payload Type: | 117 (96~127) | | | |
| CONTRACTOR CONTRACTOR | Fill Light: | Auto Mode 🔻 | | | |
| Device Log | | | Apply | | |
| | Osd Settings | | | | |
| Security Settings | Osd Time: | Disabled 🔻 🕜 | Osd Text: | Disabled v | |
| | | | Apply | | |
| | Video Codecs | | | | |
| | H264Video Stream0: | | | | |
| | Bitrate Control: | VBR 🔻 🕜 | Profile: | Base Profile V | |
| | Frame Rate (fps): | 25 🔻 🕜 | BitRate: | 4 Mbps 🔻 🕜 | |
| | Resolution: | 720P 🔻 🕜 | I Frame Interval: | 50 | |
| | H264Video Stream1: | | | | |
| | Bitrate Control: | VBR V | Profile: | Base Profile V | |
| | Frame Rate (fps): | 25 • 0 | BitRate: | 512 Kbps 🔻 🕜 | |
| | Resolution: | VGA VGA | I Frame Interval: | 50 | |
| | | | | | |

Picture 31- Camera Settings

Table 20- Camera Settings

| Parameters | Description |
|-----------------|---|
| camera settings | |
| | Auto mode: The camera automatically makes the most appropriate |
| | adjustments according to the color temperature of the shooting scene, and |
| | automatically compensates for the color of the light source. $_{\circ}$ |
| | Lock mode: Fixed white balance parameters will not be automatically |
| | adjusted according to the actual color temperature. |
| | Incandescent lamp mode: To compensate for the hue of incandescent lamps, |
| | it is suitable for use under beige light sources (bulbs, tungsten lamps, |
| White Balance | candles) and other light sources of this type. $_{\circ}$ |
| Mode | Warm light mode: Compensate the hue of warm light, suitable for light |
| | sources with a color temperature of about 2700K $_{\circ}$ |
| | Naturl light mode: It can be used for white balance in outdoor shooting and |
| | has a wide range of applications. |
| | Fluorescent lamp light: Compensate the hue of fluorescent lamps, suitable |
| | for use under fluorescent light sources (fluorescent lamps, energy-saving |
| | lamps) and other types of light sources。 |
| | Auto mode : The camera automatically sets the parameters, no need for the |
| Exposure Mode | operator to adjust. |
| | Manual exposure time : Set the exposure time by yourself, the range is |



| | 0~10000 |
|-----------------|---|
| | Manual exposure gain: Set the exposure gain by yourself, the range is |
| | 0~1024 |
| | All manual : Manually set the exposure time and gain. |
| | It refers to the time to press the shutter. Increasing the exposure time can |
| | increase the signal-to-noise ratio and make the image clear. The longer the |
| | time, the more the sum of photons to the CCD\CMOS surface, the brighter |
| Exposure Time | the captured image will be, but if it is overexposed, the photo will be too |
| | bright and lose the image details; if it is underexposed, the photo will be too |
| | dark. |
| | It refers to the amplification gain of the analog signal after double sampling, |
| | but the noise signal is also amplified in the process of amplifying the image |
| Exposure Gain | signal. The gain is generally only used when the signal is weak, but you do |
| | not want to increase the exposure time. |
| | Auto mode: The camera automatically sets the contrast according to the |
| Contrast Mode | environment, no need for the operator to adjust |
| | Manual mode: Manually set the camera's contrast parameters. |
| | Contrast refers to the contrast between light and dark in the picture. Increase |
| Contrast | the contrast, the brighter areas will be brighter and the darker areas will be |
| | darker, and the contrast between light and dark will increase. |
| | Auto mode: The camera automatically sets the saturation according to the |
| Saturation Mode | environment, without the need for the operator to adjust |
| | Manual mode: Manually set the camera's saturation parameters. |
| | Saturation refers to the color. Adjusting the saturation will change the color. |
| | The greater the adjustment, the more distorted the image color. Adjusting the |
| Saturation | saturation is only suitable for pictures with insufficient colors. When the |
| | saturation is adjusted to the lowest, the image will lose its color and become |
| | a black and white image. |
| | Auto mode: The camera automatically sets the sharpness according to the |
| Sharpness Mode | environment, no need for the operator to adjust |
| | Manual mode: Manually set the sharpness parameters of the camera |
| | Sharpness is sometimes called "sharpness", which is an indicator that |
| Sharphase | reflects the sharpness of the image plane and the sharpness of the edges of |
| Sharpness | the image. If you increase the sharpness, the contrast of the details on the |
| | image plane is also higher and it looks clearer. |
| Enable Onvif | Enable or disable the onvif protocol, after enabling it, the device can be |
| | discovered through a recorder that supports ONVIF |
| Call Stream | Main stream or sub stream used in video call |



| Enable Onvif | Is authentication required when using onvif protocol (with username and |
|----------------------|---|
| Auth | password) |
| Enable Pten Auth | When using rtsp protocol, whether authentication is required (with username |
| Enable Rtsp Auth | and password) |
| H.264 Payload | Sat the load type of h 264, the range is 06-127 |
| Туре | Set the load type of h.264, the range is 96~127 |
| Osd Settings | |
| Osd Time | Turn on/off the date display of the camera image interface. |
| Color Style | Display colors: black, red, blue, green. |
| Time Position | Display position: top left, top right, bottom left, bottom right. |
| Font Size | Display font size: 16*16,20*20 |
| Osd Text | Enable/disable the text display of the camera image interface. |
| Title Message | Text display content of camera image interface |
| Video Codecs | |
| H264 Video Stream | Support H.264 encoding format |
| | VBR: Video call will adapt to the bit rate of the opposite end, so that the video |
| Bitrate Control | effect is better. |
| | CBR: The video call will not change according to the bit rate set by itself. |
| Resolution | Support 1080P,720P,4CIF,VGA,CIF,QVGA |
| France Data (fra) | The larger the value is, the more fluent the video is, and the higher the |
| Frame Rate (fps) | requirement for network bandwidth is; adjustment is not recommended |
| | Minimum configuration: support I / P frame, only support progressive and |
| | CAVLC. It is generally used for low-level or applications requiring additional |
| Profile | fault tolerance, such as video call, mobile video, etc |
| | Main configuration: provide I / P / B frames, support progressive and |
| | interleaved, and support CAVLC and CABAC, |
| | It refers to the data flow used by video files in unit time, also known as code |
| BitRate | rate or code flow rate. Generally speaking, sampling rate is the most |
| | important part of picture quality control in video coding. Generally, the unit we |
| | use is KB / s or MB / s |
| I Frame Interval | The larger the value, the worse the video quality, otherwise the better the |
| | video quality; adjustment is not recommended. |
| RTSP Information | h |
| Main Stream Url | Display the main stream URL address |
| Sub Stream Url | Display the sub stream URL address |



| Snapshot Trigger Mode: | | |
|------------------------|--------------------|-------------------------------|
| | Snapshot By State: | 🔲 Talking 🔲 Ringing 🔲 Calling |
| Server Url: | | |
| Username: | | Password: |
| | Ap | ply |
| | | |

Picture 32 - Snapshot

Capture trigger mode: call state trigger

Call status trigger: save the screenshot to the server when the status of outgoing call, incoming call and call is triggered.

Server address (supports uploading via FTP / TFTP / HTTP / HTTPS) ftp://IP : Port @ user name: password / path

9.19 Intercom Setting >> MCAST

It is easy and convenient to use multicast function to send notice to each member of the multicast via setting the multicast key on the device and sending multicast RTP stream to pre-configured multicast address. By configuring monitoring multicast address on the device, monitor and play the RTP stream which sent by the multicast address.

The detail for 8.2 MCAST

9.20 Intercom Setting >> Action URL

Table 21- Action URL

Action URL Event Settings

Set URL for the device to report its action to server. These actions are recorded and sent as xml files to the server. Sample format is http://InternalServer /FileName.xml.

(Internal Server: The IP address of server; File Name: the device's xml file used to report action.)



| | Features Media Settings | Camera Settings MCAST | Action | Time/Date | Time Plan | Tone |
|--------------------------------------|---------------------------|-----------------------|--------|-----------|-----------|---------------------|
| > System | | | | | | NOTE |
| > Network | Action URL Event Settings | | | | | Description: |
| | Setup Completed: | | 0 | | | Action URL settings |
| > Line | Registration Succeeded: | | 0 | | | |
| < Ling | Registration Disabled: | | 0 | | | |
| > Intercom settings | Registration Failed: | | 0 | | | |
| Intercom setungs | Incoming Calls: | | | | | |
| | Outgoing Calls: | | 0 | | | |
| > Call List | Call Established: | | 0 | | | |
| 2.00000000000 | Call Terminated: | | 0 | | | |
| Function Key | Phone Silent: | | 0 | | | |
| | Phone Unsilent: | | 0 | | | |
| > Security | Call Mute: | | 0 | | | |
| | Call Unmute: | | 0 | | | |
| > Device Log | Missed Calls: | | 0 | | | |
| | IP Changed: | | 0 | | | |
| > Security Settings | Phone State Idle: | | 0 | | | |
| | Phone State Talking: | | 0 | | | |
| | Phone State Ringing: | | 0 | | | |
| | Start Reboot: | | 0 | | | |
| | Web API Auth Changed: | | 0 | | | |
| | Echo Test: | | 0 | | | |
| | Tamper: | | 0 | | | |
| | | | | | | |
| | | Apply | | | | |
| | | | | | | |
| | | | | | | |
| | | | | | | |

Picture 33- Action URL

Note! The operation URL is used by the IPPBX system to submit device events. Please refer to the details Fanvil Action URL . https://www.fanvil.com/Support/download/cid/14.html

9.21 Intercom Setting >> Time/Date

Users can configure the device's time Settings on this page.

| | Features | Media Settings | Camera Settings | MCAST | Action | Time/Date | Time Plan | Tone | |
|-------------------------|------------------------------|--|---------------------------------|----------------------|--------|-----------|-----------|---|-----------------------------------|
| › System | | | | | | | | NOTE | |
| > Network | Network Time Synch | Server Settings | ۲ | | | | 0 | Description: Time and date s | settings, |
| › Line | Time Synch | ronized via DHCP ronized via DHCPv6 | 0 | | | | 0 | you can set the through the net server, or manu the time, select | work time ally set the time |
| > Intercom settings | Primary Tin Secondary | ne Server Time Server | 0.pool.ntp.org time.nist.gov | | | | 0 | zone and date f | ormat. |
| › Call Li st | Time zone Resync Peri | od | (UTC+8) Beijing 60 | Singapore,Perth,Irku | | | 0 | | |
| › Function Key | Time/Date For 12-hour clo | | | | | | | | |
| › Security | Time/Date | Format | DD MMM WW | • 10 OCT : | SAT | | | | |
| > Device Log | Daylight Saving | g Time Settings | | | | | | | |
| Security Settings | Location DST Set Ty | pe | None Disabled Apply | T | | | | | |
| | Manual Time Se 2020-10-10 | ettings 18 | ▼ 16 ▼ | | Apply |] | | | |
| | | | | | | | | | |

Picture 34 - Time/Date



| Time/Date | | |
|-------------------|--------------|---|
| Field Name | Explanat | tion |
| Network Time | Server Setti | ings |
| Time Synchroniz | ed via SNTP | Enable time-sync through SNTP protocol |
| Time Synchroniz | ed via DHCP | Enable time-sync through DHCP protocol |
| Primary Time Se | rver | Set primary time server address |
| | | Set secondary time server address, when primary server is not |
| Secondary Time | Server | 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 |
| Daylight Savi | ng Time Sett | ings |
| Location | | Select the user's time zone specific area |
| | | Select automatic DST according to the preset rules of DST, or the |
| DST Set Type | | manually input rules |
| Offset | | The DST offset time |
| Month Start | | The DST start month |
| Week Start | | The DST start week |
| Weekday Start | | The DST start weekday |
| Hour Start | | The DST start hour |
| Month End | | The DST end month |
| Week End | | The DST end week |
| Weekday End | | The DST end weekday |
| Hour End | | The DST end hour |
| Manual Time | Settings | |
| To sot the time m | | eed to disable the SNTP service first, and you need to fill in and subm |

To set the time manually, you need to disable the SNTP service first, and you need to fill in and submit each item of year, month, day, hour and minute in the figure above to make the manual settings successful.

System time: Display system time and its source

(SIP automatic get >SNTP automatic get >manual manual setting)

9.22Intercom settings>>Time plan

The user can set the time point and time period for the device to perform a certain action.



| | Features Media Settings | Gamera Settings MCAST | Action Time | e/Date Time Plan |
|---------------------|--------------------------|-----------------------|-------------------|------------------|
| System | | | | |
| › Network | Time Plan Add : Type: | Timed reboot | | |
| › Line | Repetition period: | No repetition • | | |
| > Intercom settings | | 1 2 | | |
| → Call List | Monthly: | 3 4 5 6 | | |
| › Function Key | | 7 | | |
| Security | | 9 | | |
| Device Log | Effective time: | 0 : 0 - 0 : 0 Add | | |
| Security Settings | Time Plan List | | | |
| | Index Type | Special configure | Repetition period | Effective time |
| | | | | Delete |

Picture 35- Time Plan

Table 23- Time Plan

| Parameters | Description |
|----------------|--|
| type | Timing restart, timing upgrade, timing sound detection, timing playback |
| | audio |
| Audio path | Support local |
| | Local: select the audio file uploaded locally |
| Audio settings | Select the audio file you want to play, it supports trial listening, and you can |
| | play it immediately after clicking the trial listening |
| Repeat cycle | Do not repeat: execute once within the set time range |
| | Daily: Perform this operation in the same time frame every day |
| | Weekly: Do this in the time frame of the day of the week |
| | Monthly: the time frame of the month to perform this operation |
| Effective time | Set the time period for execution |

9.23Intercom settings >> Tone

The user can configure the prompt tone of the device on this page.

You can select the country area or customize the area. The selected area can directly appear the default information, and the customized one can modify the key tone, callback tone and other information.



| | Features Media Settings | Camera Settings MCAST | Action | Time/Date | Time Plan | Tone |
|-------------------|---------------------------|--------------------------------|-------------------------|-----------|-----------|---|
| System | | | | | | NOTE |
| usausa 🕴 | Tone Settings | | | | | |
| letwork | Select Your Tone: | United States | | | • 🕜 | Tone: cadence[.cadence] |
| | Dial Tone: | 350+440/0 | | | 0 | [,cadence]Where |
| ne | Ring Back Tone: | 440+480/2000,0/4000 | | | 0 | cadence = Freq1[+Frec [+Freq3] |
| | Busy Tone: | 480+620/500,0/500 | | | 0 | [+Freq4]/Duration.Freq The frequency of the |
| Intercom settings | Congestion Tone: | | | | 0 | tone:200~4000HZ, If i set to 0Hz, it means the |
| | Call waiting Tone: | 440/300,0/10000,440/300,0/1000 | 0,0/0 | | 0 | tone won't be played.A |
| ill List | Holding Tone: | | | | 0 | tone is comprised of at most four different |
| | Error Tone: | | | | 0 | frequencies.Freq1+Freq The juxtaposition of two |
| nction Key | Stutter Tone: | | | | 0 | frequencies Freq1 and |
| | Information Tone: | | | | 0 | Freq2 without modulation.Freq1*Freq2 |
| curity | Dial Recall Tone: | 350+440/100.0/100.350+440/100 | 0/100 350+440/100 0/100 | 350+440/0 | 0 | Freq1 is modulated by Freq2.Duration The time |
| | Message Tone: | | | | 0 | duration of the |
| vice Log | Howler Tone: | | | | 0 | tone:0~30000ms.If it is set to 0ms, it means the |
| evice Log | Number Unobtainable Tone: | 400/500.0/6000 | | | Ő | tone will keep on playin until stopped by system |
| | Warning Tone: | 1400/500.0/0 | | | 0 | it is set to 0/0, it means |
| ecurity Settings | Auto Answer Tone: | 1 1001000,000 | | | ő | the tone is stopped. The composition of Tone: Yo |
| | | Apply | | | | can configure at most eight different cadences |
| | | ларау | | | | for one tone, and |
| | | | | | | separate tones by commas. |
| | | | | | | |
| | | | | | | |
| | | | | | | |
| | | | | | | |



9.24Call list >> Call List

Restricted Incoming Calls

It same as blacklist.By adding a number into the blacklist, user will no longer receive phone call from that number and it will be rejected automatically by the device until user delete it from the blacklist.

User can add specific number to be blocked, or a prefix where any numbers matched the prefix will all be blocked.

Restrict Outgoing Call

You can set the rule to restrict some numbers from dialing out, until you remove the number from the table.

9.25Call list >> Web Dial

Use web page to call, answer and hang up.



| | Call List Web Dial |
|---------------------|---------------------|
| > System | |
| > Network | Web Dial Settings |
| › Line | Dial Answer Hang-up |
| › Intercom settings | |
| > Call List | |
| > Function Key | |
| › Security | |
| › Device Log | |
| › Security Settings | |
| | |
| | |



9.26Function key

| ¢ | Key | on Key Settings >: Type | Name | Value | Value2 | Subtype | Line | | Media | |
|----------|--------------|----------------------------|----------|-------|--------|------------|-------------|---|---------|--|
| | DSS Key 1 | Memory Key 🔻 | | 2345 | | Speed Dial | ▼ 1356@SIP1 | ۲ | DEFAULT | |
| | DSS Key 2 | None 🔻 | | | | None | ▼ AUTO | Ŧ | DEFAULT | |
| settings | | | | er er | Apply | | | | | |
| | Progra | mmable Key Setti | nas @ >> | | | | | | | |
| n Key | | | | | | | | | | |
| | | | | | | | | | | |
| | 2. | | | | | | | | | |
| | | | | | | | | | | |
| | | | | | | | | | | |
| | | | | | | | | | | |
| | | | | | | | | | | |
| | | | | | | | | | | |
| 5 | | | | | | | | | | |
| | | | | | | | | | | |
| į | | | | | | | | | | |
| igs | | | | | | | | | | |

| Кеу | Deskto | p | Dialer | - | Ringin | g | Talki | ng | Desktop Lon | g Pressed |
|------|---------|---|---------|---|--------|---|-------|----|-------------|-----------|
| Key1 | Dsskey1 | T | Dsskey1 | • | Answer | T | End | ٣ | None | • |
| Key2 | Dsskey2 | v | Dsskey2 | Ŧ | Answer | • | End | | None | T |



| Dial Mode Select | Main-S | econda 🔻 | | | |
|-------------------------|--------|-----------------|-----------------------|-------|--------------|
| Call Switched Time | 16 | (5~50)second(s) | | | |
| First Number Start Time | 06:00 | (00:00~23:59) | First Number End Time | 18:00 | (00:00~23:59 |

Picture 38- Function Key

Table 24- Function Key

| Parameters | Description |
|------------------|--|
| Function key set | tings |
| memory | Speed Dial: The user can directly dial the set number. This feature is |
| | convenient for customers to dial frequent numbers. |
| | Intercom: This feature allows the operator or secretary to quickly connect |
| | to the phone, widely used in office environments |
| Key event | The user can select a function key as a shortcut to trigger an event for |
| | example: None /Handfree |
| DTMF | Press during a call to send the set DTMF |
| Mcast Paging | Configure the multicast address and voice encoding. User can initiate |
| | multicast by pressing this key |
| Action URL | The user can use a specific URL to make basic calls to the device, open |
| | the door, etc. |
| Mcast Listening | In standby, press the function key, if the RTP of the multicast is detected, |
| | the device will monitor the multicast |
| PTT | Speed dial: Make a call when pressed, and end the call when lifted. |
| | Intercom: Start the intercom when pressed, and end the intercom when |
| | lifted. |
| | Multicast: Initiate multicast when pressed, and end multicast when lifted |
| Programmable Ke | ey Settings |
| Desktop | None: Nothing happens when you press the speed dial |
| | Dsskey1: When it is set to dsskey1, follow the settings of dsskey1 to make |
| | call, answer, etc. |
| | Dsskey2: When it is set to dsskey2, perform operations such as calling |
| | and answering according to the setting of dsskey2 |
| Dialer | None: Nothing happens when you press the speed dial |
| | Dsskey1: When it is set to dsskey1, follow the settings of dsskey1 to make |
| | call, answer, etc. |



| | Dsskey2: When it is set to dsskey2, perform operations such as calling |
|--------------------|--|
| | and answering according to the setting of dsskey2 |
| Ringing | Answer: Set to answer, when there is an incoming call, if auto answer is |
| | disabled, press the speed dial key to answer the call |
| | End: set to end, when there is an incoming call, press the speed dial |
| | button to hang up the call |
| Talking | End: set to end, when there is a call, press the speed dial key to hang up |
| | the call |
| | Volume up: set as volume up button, when there is a call, press the speed |
| | dial button to increase the volume |
| | Volume down: set as volume up button, when there is a call, press the |
| | speed dial button to decrease the volume |
| | Dsskey1: When it is set to dsskey1, follow the settings of dsskey1 to make |
| | call, answer, etc. |
| | Dsskey2: When it is set to dsskey2, perform operations such as calling |
| | and answering according to the setting of dsskey2 |
| Desktop Long | None: Long press the speed dial key does not respond |
| Pressed | Main menu: Long press the speed dial key to enter the command line |
| | mode, see 5.2.1 Common Command Mode for details |
| Advanced Settings | |
| | Number 1 call number 2 mode selection. |
| | <main secondary="">: If the first number is not answered within the set time,</main> |
| Hot Key Dial Mode | the second number will be automatically switched. |
| Select | <day night="">: The system time is automatically detected during the call. If</day> |
| | it is daytime, the first number is called, otherwise the second number is |
| | called. |
| Call Switched Time | Set number 1 to call number 2 time, default 16 seconds |
| Day Start Time | The start time of the day when the <day night=""> mode is defined. Default</day> |
| | "06:00" |
| Day End Time | The end time of the day when the <day night=""> mode is defined. Default</day> |
| | "18:00 |
| | |

> Memory

Enter the phone number in the input box. When you press the function key, the device will call out the set phone number. This button can also be used to set the IP address, press the function key to make an IP direct call.



| Key | Туре | Name | Value | Value2 | Subtype | Line | Media |
|--------------|--------------|------|-------|--------|--------------|-----------|---------|
| DSS Key 1 | Memory Key 🔻 | | 2345 | | Speed Dial 🔻 | 1356@SIP1 | DEFAULT |
| DSS Key 2 | None 🔻 | | |] [| None | AUTO | DEFAULT |

Picture 39 - Memory Key

Table 25- Memory Key

| Туре | number | line | Subtype | usage |
|------------|---|--|---------------|--|
| | Fill in the SIP | The line | Speed Dial | Using the speed dial mode, press the button to quickly dial the set number. |
| memor y | account or IP address of the called party | correspon ding to the SIP account | Intercom | Using the intercom mode, when the SIP phone at the opposite end supports the intercom function, the call can be automatically answered. |

> Multicast

Multicast function is to deliver voice streams to configured multicast address; all equipment monitored the multicast address can receive and play the broadcasting. Using multicast functionality would make deliver voice one to multiple which are in the multicast group simply and conveniently.

The DSS Key multicast web configuration for calling party is as follow:

| Key | Туре | Name | Value | Value2 | Subtype | | Line | Media | |
|--------------|---------------|------|----------------|--------|---------|---|-------------|---------|---|
| DSS Key 1 | MCAST Pagin 🔻 | | 239.1.1.1:1366 |) | G.711U | ۲ | 1356@SIP1 • | DEFAULT | ٧ |
| DSS Key 2 | None 🔻 | | | | None | ۳ | AUTO | DEFAULT | ٣ |

Picture 40- Multicast

Table 26- Web Multicast

| Туре | Number | Subtype |
|-----------|--|---------|
| | Set the host IP address and port number, they must | G.711A |
| Multicast | be separated by a colon (The IP address range is | G.711U |
| | 224.0.0.0 to 239.255.255.255, and the port number | G.729AB |



| is preferably set between 1024 and 65535) | iLBC |
|---|-------|
| | opus |
| | G.722 |

> PTT

Keep pressing the shortcut key set to make a call, release it and hang up

| Key | Type | Name | Value | Value2 | Subtype | Line | Media |
|--------------|-------|------|-------|--------|--------------|-------------|-----------|
| DSS Key 1 | PTT T | | 2345 | | Speed Dial 🔻 | 1356@SIP1 • | DEFAULT |
| DSS Key 2 | None | | | | None v | AUTO 🔻 | DEFAULT V |

Picture 41 - Advanced Setting

9.27Security >> Web filter

Users can set up to allow only a certain network segment IP to access the device

| | Web Filter Trust Certifica | tes Device Certificates Firewall | | |
|---------------------|----------------------------|----------------------------------|----------|------------------|
| > System | | | | |
| > Network | Web Filter Table 🕜 | End IP Address | | Option |
| > Line | Web Filter Table Settings | | | |
| > Intercom settings | Start IP Address | End IP Address | @ | Add |
| › Call List | Web Filter Setting 🕜 | | | |
| > Function Key | Enable Web Filter 🗐 | Apply | | |
| > Security | | | | |
| > Device Log | | | | |
| Security Settings | | | | |
| | | | | |
| | | | | |
| | | | | |
| | | | | |
| Web Filter Table 🕜 | | | | |
| Start IP Address | | End IP Address | Opt | ion |
| 172.16.80.6 | | 172.16.80.69 | | Modify Delete |

Picture 42- WEB filter



Add and delete the allowed IP network segments; configure the start IP address in the start IP, configure the end IP address in the end IP, and then click [Add] to add successfully. You can set a large network segment or add it into several network segments. When deleting, select the starting IP of the network segment to be deleted in the list, and then click [Delete] to take effect.

Enable web filtering: configure to enable/disable web access filtering; click the [Submit] button to take effect

Note: If the device you access to the device is on the same network segment as the device, do not configure the web filtering network segment to be outside your own network segment, otherwise you will not be able to log in to the web page.

9.28Security >> Trust Certificates

You can upload and delete uploaded trust certificates.

| | Web Filter | Trust Certificates | Device Certificates | Firewall | 1 | | |
|---------------------|-----------------------------|--------------------|----------------------|----------|--------|------------|-----------|
| > System | | | | | | | |
| > Network | Permission Certi | ficate | | | | | |
| > Line | Permission Ce Common Nan | | Disabled Disabled | 0 | | | |
| > Intercom settings | Certificate mo | | All Certificates | 0 | | | |
| › Call List | | | Apply | | | | |
| › Function Key | Import Certificat | | | Select | Upload | | |
| > Security | Certificates List | 0 | | | | | |
| > Device Log | Index | File Name | Issued To | Iss | ued By | Expiration | File Size |
| Security Settings | | | | | | | Deigle |

Picture 43 - Trust Certificates

9.29Security >> Device Certificates

Select the default certificate or the custom certificate as the device certificate. You can upload and delete uploaded certificates.



| | Web Filter Trust Certificate | Device Certificates | Firewall | | |
|---------------------|------------------------------|----------------------|---------------|------------|---------------------|
| > System | | | | | |
| > Network | Device Certificates 💡 | | | | |
| › Line | Device Certificates | Default Certificates | (existence) | | |
| › Intercom settings | Import Certificates 💡 | | | | |
| › Call List | Load Server File | | Select Upload | | |
| > Function Key | Certification File 🕜 | | | | |
| > Security | File Name | Issued To | Issued By | Expiration | File Size Delete |
| > Device Log | | | | | |
| > Security Settings | | | | | |

Picture 44- Device Certificates

9.30Security >> Firewall

| | Web Filter Trust Certificates Firewall |
|---------------------|--|
| › System | |
| > Network | Firewall Type 🕜 |
| › Line | Apply |
| > Intercom settings | Firewall Input Rule Table 🔮 Index Deny/Permit Protocol Src Address Src Mask Src Port Range Dst Address Dst Mask Dst Port Range |
| › Call List | |
| › Function Key | Firewall Output Rule Table 🛿 Index Deny/Permit Protocol Src Address Src Mask Src Port Range Dst Address Dst Mask Dst Port Range |
| > Security | Firewall Settings 🕖 |
| > Device Log | Input/Output Input Src Address Deny/Permit Deny Src Mask Add |
| › Security Settings | Protocol UDP Src Port Range Dst Port Range |
| | Rule Delete Option ? Input/Output Input * Index To Be Deleted Delete |

Picture 45 - Firewall

Through this page, you can set whether to enable the input and output firewalls, and at the same time, you can set the input and output rules of the firewall. Use these settings to prevent malicious network access, or restrict internal users from accessing some resources of the external network, and improve safety.

The firewall rule setting is a simple firewall module. This function supports two kinds of rules:



input rules and output rules. Each rule will be assigned a serial number, and a maximum of 10 each rule can be set.

Taking into account the complexity of firewall settings, the following will illustrate with an example:

| parameter | Description | |
|---------------------|--|--|
| Enable Input Rules | whether enable Input Rules | |
| Enable Output Rules | Whether enable Output Rules | |
| input/output | Select the current rule as an input or output rule | |
| Deny/permit | Choose the current rule is deny or allowed; | |
| protocol | There are four types of protocols: TCP, UDP, ICMP, IP $_{\circ}$ | |
| Port range | Port range | |
| | The source address can be the host address, network address, or | |
| Src Address | all addresses 0.0.0.0; it can also be a network address similar to | |
| | *.*.*.0, such as 192.168.1.0. | |
| | The destination address can be a specific IP address or all | |
| Dst Mask | addresses 0.0.0.0; it can also be a network address similar to | |
| | *.*.*.0, such as 192.168.1.0. | |
| Src Port Range | It is the source address mask. When it is configured as | |
| | 255.255.255.255, it means it is a specific host. When it is set as a | |
| | subnet mask of type 255.255.255.0, it means that the filter is a | |
| | network segment; | |
| | It is the destination address mask. When it is configured as | |
| Dst Port Range | 255.255.255.255, it means it is a specific host. When it is set as a | |
| Dotroit Nange | subnet mask of 255.255.255.0 type, it means that a network | |
| | segment is filtered; | |

Table 27- Web Firewall

After setting, click [Add], a new item will be added to the firewall output rules, as shown in the figure below:



Picture 46- Firewall rules list

Then select and click the button [Submit].

In this way, when the device runs: ping 192.168.1.118, it will not be able to send data packets to 192.168.1.118 because of the prohibition of the output rule. But ping other IPs in the



192.168.1.0 network segment can still receive the response packets from the destination host normally.

| Rule Delete Option 🕜 | | | |
|----------------------|---------|---------------------|--------|
| Input/Output | Input 🔻 | Index To Be Deleted | Delete |
| | | | |
| | | | |

Picture 47- Delete firewall rules

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

9.31 Device log

You can crawl the device log, when you encounter unusual problems, please send the device log to the technical staff for positioning problem. For more detail 10.5 get device log.

9.32Security settings

Enable Tamper: after enable, when the device is removed by force, the alarm information will be sent to the server and the alarm ring will be played.

| Ringtone Duration: | | | 5 | (1~600)s | | |
|--------------------------------|---------------------|-------------------|-----------------|----------------------|-----------|--|
| Input & Tamper Server Address: | | | | | | |
| Message:Alarm_In | fo:Description=;SI | P User=512;Mac=00 | ::38:3e:3a:06:6 | 5;IP=172.18.60.192;p | ort=Input | |
| | | | | | | |
| | | | Apply | | | |
| put Settings >> | | | | | | |
| ✓ Input1: | | | | | | |
| Triggered By: | Low Level Trigger(C | lose Trigger) 🔻 | | | | |
| Triggered Action: | Send SMS | Dss Key: None | • | Triggered Ringtone: | 2.wav 🔻 | |
| | | | | | | |
| | | | Apply | | | |
| utput Settings >> | | | | | | |

Picture 48 - Security Settings

Table 28- Security Settings

| Security Settings | |
|-------------------|-------------|
| Parameters | Description |
| Basic Settings | |



| Ringtone Duration | Set the ringtone duration, default value is 5 seconds. | | |
|---------------------|--|--|--|
| | Set remote server address. The device will send message to the | | |
| Input & Tamper | server when the alarm is triggered. The message format is : | | |
| Server Address | Alarm_Info:Description=i16SV;SIP | | |
| | User=;Mac=0c:38:3e:3a:06:65;IP=; port=Input . | | |
| Input settings | | | |
| Input Detect | Enable or disable Input Detect | | |
| | When choosing the low level trigger (closed trigger), detect the input | | |
| Triggorod by | port (low level) closed trigger. | | |
| Triggered by | When choosing the high level trigger (disconnect trigger), detect the | | |
| | input port (high level) disconnected trigger. | | |
| | Send SMS: Set the alert message send to server if selected. | | |
| Triggorod Action | Dss Key: The device will perform corresponding Dss Key | | |
| Triggered Action | configurations if any key is selected, by default the value is none. | | |
| | Triggered Ringtone: Select triggered ring tone. | | |
| Output Settings | | | |
| Output Response | Enable or disable Output Response | | |
| Triggered by DTMF | Calcat the DTME trippen ring tage | | |
| Ring tone | Select the DTMF trigger ring tone. | | |
| Triggered by URI | Coloct the LIDI trigger ring tone | | |
| Ringtone | Select the URI trigger ring tone. | | |
| Triggered By SMS | Select the SMS trigger ring tone. | | |
| Ringtone | | | |
| Triggered By Dsskey | Select the Dsskey trigger ring tone. | | |
| Ringtone | | | |
| Standard Status | When choosing the low level trigger (NO: normally open), when meet | | |
| | the trigger condition, trigger the NO port disconnected. | | |
| | When choosing the high level trigger (NC: normally close), when meet | | |
| | the trigger condition, trigger the NC port close. | | |
| Output Duration | Set the output change duration time, the default is 5 seconds. | | |
| | Enable or disable trigger by DTMF. The device will check the received | | |
| Trigger by DTMF | DTMF sent by remote device, if it matches the DTMF trigger code, the | | |
| | device will trigger corresponding output port. | | |
| DTMF Trigger Code | Input the DTMF trigger code, default value is 1234. | | |
| DTMF Reset Code | Input the DTMF reset code, default value is 4321. | | |
| | Reset the output port mode by duration or state. | | |
| Reset By | By duration: Reset the output port status when output duration occurs. | | |
| | By state: Reset the output port status when device's call state | | |
| | 1 | | |



| | changes. |
|-----------------------|---|
| Trigger by URI | Enable or disable trigger by URI. |
| | User can send commands from remote device or server to i16SV |
| | series device, if the command is correct, then device will trigger |
| | corresponding output port. |
| Trigger Message | Input trigger message for trigger by URI mode. |
| Rest Message | Input reset message for trigger by URI mode. |
| | Enable or disable trigger by SMS. |
| Trigger by SMS | User can send ALERT command to i16SV series device, if the |
| | command is correct, then device will trigger corresponding output port. |
| Trigger SMS | Input trigger message for trigger by SMS mode. |
| Reset SMS | Input reset message for trigger by SMS mode. |
| | Select the input port, when the input port meets the trigger condition, |
| Trigger by Input | the output port will be triggered (The Port level time change, By < |
| | Output Duration > control) |
| | Select call state to trigger the output port, options are: |
| Trigger By Call state | Talking: When the device's talking status changes, trigger the output |
| | port. |
| | Ringing: When the device's ringing status changes, trigger the output |
| | port. |
| | Calling: When the device's calling status changes, trigger the output |
| | port. |
| | Enable or disable trigger by dsskey. If any of the dsskey is selected, |
| Trigger By DssKey | when the dsskey application performs, the output port will be |
| | triggered. |



10 Trouble Shooting

When the device is not working properly, users can try the following methods to restore the device to normal operation or collect relevant information to send a problem report to the Fanvil technical support mailbox.

10.1 Get device system information

Users can obtain information through the [**System**] >> [**Information**] option on the device webpage. The following information will be provided:

Device information (model, software and hardware version) and Internet Information etc.

10.2 Reboot device

User can restart the device through the webpage, click [**System**] >> [**Reboot Phone**] and click [**Reboot**] button, or directly unplug the power to restart the device.

10.3 Device factory reset

Restoring the factory settings will delete all configurations, database and configuration files on the device and the device will be restored to factory default state.

To restore the factory settings, please go to [**System**] >> [**Configuration**] >> [**Reset Phone**] page, and click [**Reset**] button, the device will return to the factory default state.

10.4 Network Packets Capture

In order to obtain the data packet of the device, the user needs to log in to the webpage of the device, open the webpage [**System**] >> [**Tools**], and click the [**Start**] option in the "Network Packets Capture". A message will pop up asking the user to save the captured file. At this time, the user can perform related operations, such as starting/deactivating the line or making a call, and clicking the [**Stop**] button on the webpage after completion. Network packets during the device are saved in a file. Users can analyze the packet or send it to the Fanvil Technical Support mailbox.



10.5Get device log

Log information is helpful when encountering abnormal problems. In order to obtain the log information of the device, the user can log on to the device web page, open the web page [device log], click the "start" button, follow the steps of the problem until the problem appears, and then click the "end" button, "save" to the local for analysis or send the log to the technician to locate the problem.

10.6 Common 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 |
| | or POE switch met with the specification requirements and check if |
| | device is well connected to power source. |
| | 2. If the device enters "POST mode" (the SIP/NET and function |
| | button indicators are always on), the device system is damaged. |
| | Please contact your location technical support to help you restore |
| | your equipment system. |
| Device could not register to a | 1. Please check if the device is connected to the network. |
| service provider | 2. If the network connection is good, please check your line |
| | configuration again. If all configurations are correct, contact your |
| | service provider for support, or follow the instructions in "10.4 Network |
| | Data Capture" to obtain a registered network packet and send it to the |
| | Fanvil Support Email to help analyze the issue. |

Table 29 - Trouble Cases