



# PA2S User Manual

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### 3 Safety Instruction

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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

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PA2S is a SIP audio&video intercom and Public Announcement system device for industry application. The media stream transmission adopts standard IP/RTP/RTSP protocols. It has various functions and interfaces, such as Intercom, video, broadcast, recording and short circuit trigger, to adapt different application environments. Users can DIY the paging device easily.



## 5 Install Guide

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### 5.1 Use POE or external Power Adapter

PA2S 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, PA2S 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 and the POE switch met the specifications to ensure the device work properly.

### 5.2 Appendix

#### 5.2.1 Common command modes

*Table 1- Common command mode*

| Action behavior     | description   |
|---------------------|---|
| Standby report IP   | Standby long press volume - 3 seconds to report IP  |
| Switch network mode | Long press the volume + 3 seconds to enter the command mode, the beep will sound, and within 5 seconds, press 3 times quickly to switch the network mode; if there is no IP currently, switch to the default static IP (192.168.1.128) DHCP mode; when DHCP obtains IP, it will report IP directly without switching;<br>Report IP after successful switching |

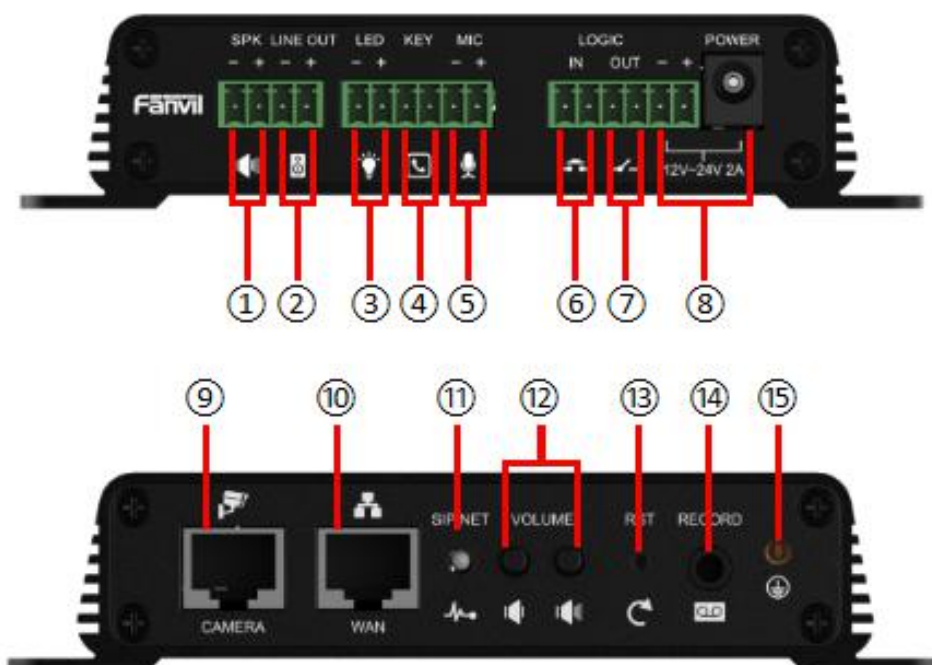
## 5.2.2 Function key LED status

*Table 2- Function key LED status*

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

## 6 User Guide

### 6.1 Interface description



*Picture 1- Interface display*

*Table 3- Interface Description*

| Number | Name                 | Description  |
|--------|----------------------|--|
| ①      | Speaker port         | Speaker port: according to the device input voltage adaptive output maximum power.<br>4Ω spkaker, POE/10W, 12V/10W, 18V/20W, 24V/30W.<br>The greater the horn impedance, the smaller the output power.<br>Suggested wire diameter: 18AWG or larger diameter. |
| ②      | Line out port        | audio signal output, for external headset or active speakers.  |
| ③      | LED port             | 5V / 5mA output, can be an external LED, indicating the network status, call status, registration status.  |
| ④      | Function button port | connection the function button, you can make an emergency call through this button.  |
| ⑤      | Microphone           | the multiplex port is microphone input by default, which can   |

|   |                            |  |
|---|----------------------------|--|
|   | port & line in port        | be configured as audio line input function through software.<br>a)Sensitivity: -38dB, bias voltage 2.2V.<br>b)Microphone signal cable it is recommended to use a shielded cable and connect the shield cable to the grounding screw, improve anti-interference.  |
| ⑥ | Switch input port          | the connection switch, you need to log on to the relevant settings page.   |
| ⑦ | Switch output port         | corresponding to the short-circuit input port, login device security page settings, you can control the alarm light, electric locks and other device, with the ⑧ power port connection for external device power supply.   |
| ⑧ | Power input port           | 12V ~ 24V 2A input, according to the input voltage to determine the maximum output power amplifier.  |
| ⑨ | Camera port                | standard RJ45 port, connect the original camera, the proposed use of five or five sub-network cable.   |
| ⑩ | Ethernet port              | WAN port, standard RJ45 port, 10 / 100M adaptive, support PoE input. It is recommended to use five or super five network cable.  |
| ⑪ | Network / Registration LED | indicates network status, call status, registration status.<br>Fast flashing: network anomaly or SIP account exception.<br>Slow flashing: during a call.<br>Always bright: successful registration.  |
| ⑫ | Volume control keys        | standby to adjust the volume of the ringtone, call only adjust the call volume.<br>a)Volume decrease key: adjust ring volume / call volume / broadcast volume. Long press the volume down key to broadcast the IP address by voice.<br>b)Volume increase key: adjust ring volume / call volume / broadcast volume. |
| ⑬ | Reset key                  | press and hold for 3S, and the indicator light will turn off, and the device will restart and restore the factory settings.  |
| ⑭ | Recording output port      | mixed output of local microphone voice and call voice.   |
| ⑮ | Grounding screw            | the metal shell of external accessories or access control panel shall be connected to this port to prevent abnormal operation of equipment caused by static electricity and other port. (except microphone accessories)  |

## 6.2 Installation instructions

### 6.2.1 Installation

The first step: the device with metal strips (user-owned) fixed in the installation location.

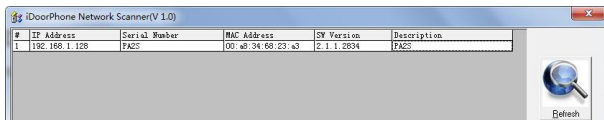
The second step: connect the function key, speaker, microphone and other peripheral devices to the corresponding port. (metal shell with thick wires connected to the device grounding screw.)

The third step: plug the network cable and power, the device light flashes on behalf of the power connection is normal.

### 6.2.2 Device IP address

Method 1:

Open the iDoorPhone Network Scanner. Press the Refresh button to search the device and find the IP address.



Method 2:

Connect the speaker, by pressing the volume down key for 3 seconds (after 30 seconds after power on), the device automatically voice broadcast the machine's IP address.

Method 3:

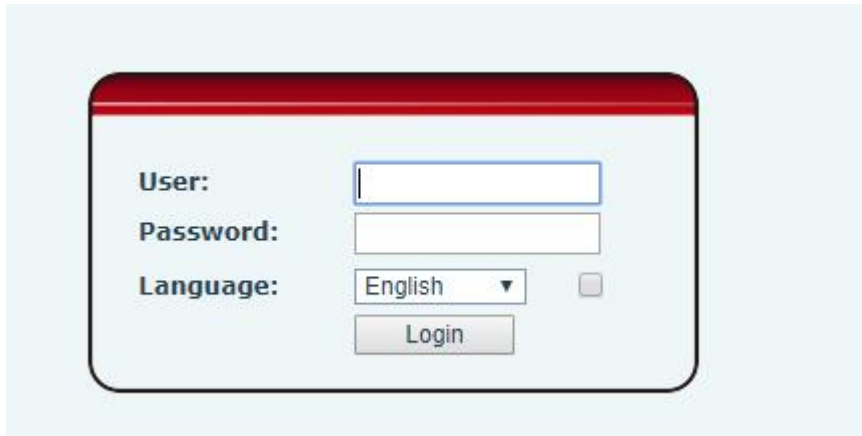
Long press the volume up key for 3 seconds, the speaker issued a rapid beep, and then quickly press the Volume up key three times, beep stopped. Wait 10 seconds, successfully switch to dynamic IP after the system automatically voice broadcast IP address. Switching again will become a fixed IP address.

*Table 4 - Configuration instructions*

| Default configuration        |   |                  |                      |
|------------------------------|---|------------------|----------------------|
| <b>DHCP mode</b>             | <b>Default enable</b>                           | <b>Static IP</b> | <b>192.168.1.128</b> |
| <b>Voice read IP address</b> | Long press the volume down button for 3 seconds | Server port      | 80                   |

## 6.3 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.xxx/` and you can see the login interface of the web page management.



*Picture 2 - WEB Login*

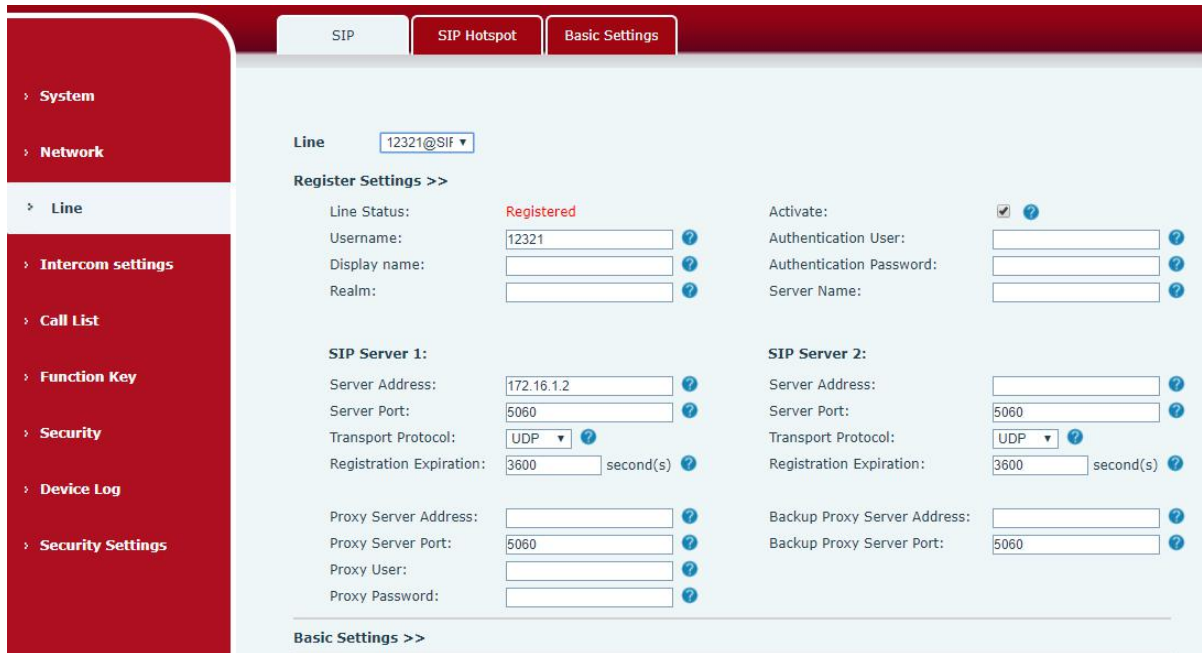
The username and password should be correct to log in to the web page. **The default username and password are "admin"**. For the specific details of the operation of the web page, please refer to [9 Web Configurations](#)

## 6.4 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:



*Picture 3 - SIP Line Configuration*

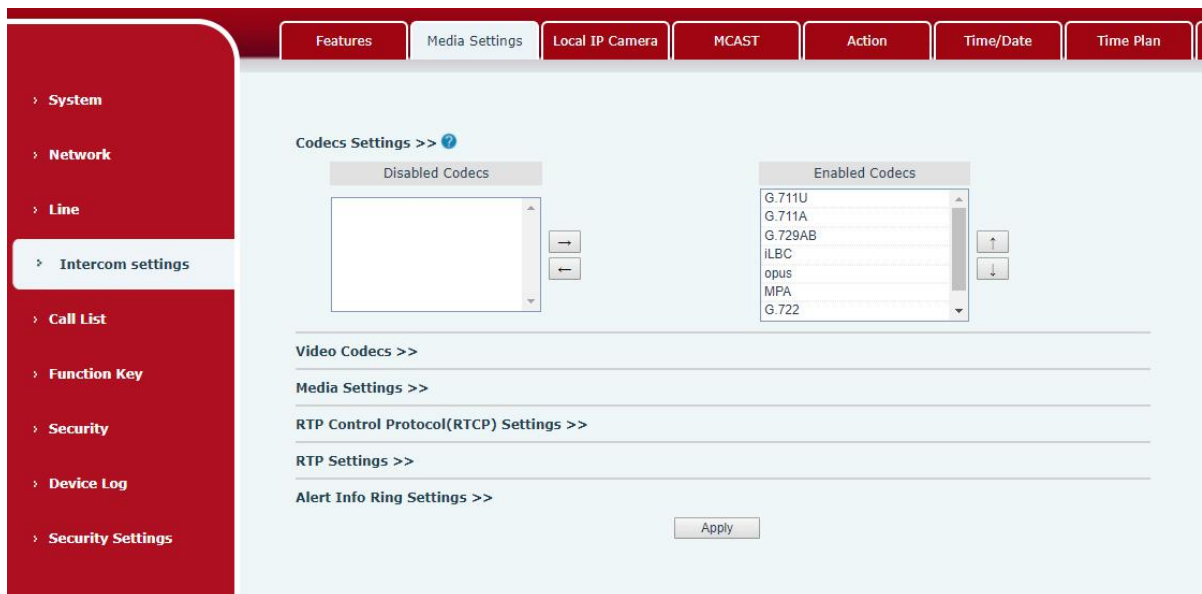
## 6.5 Volume setting

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

**[Phone 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.



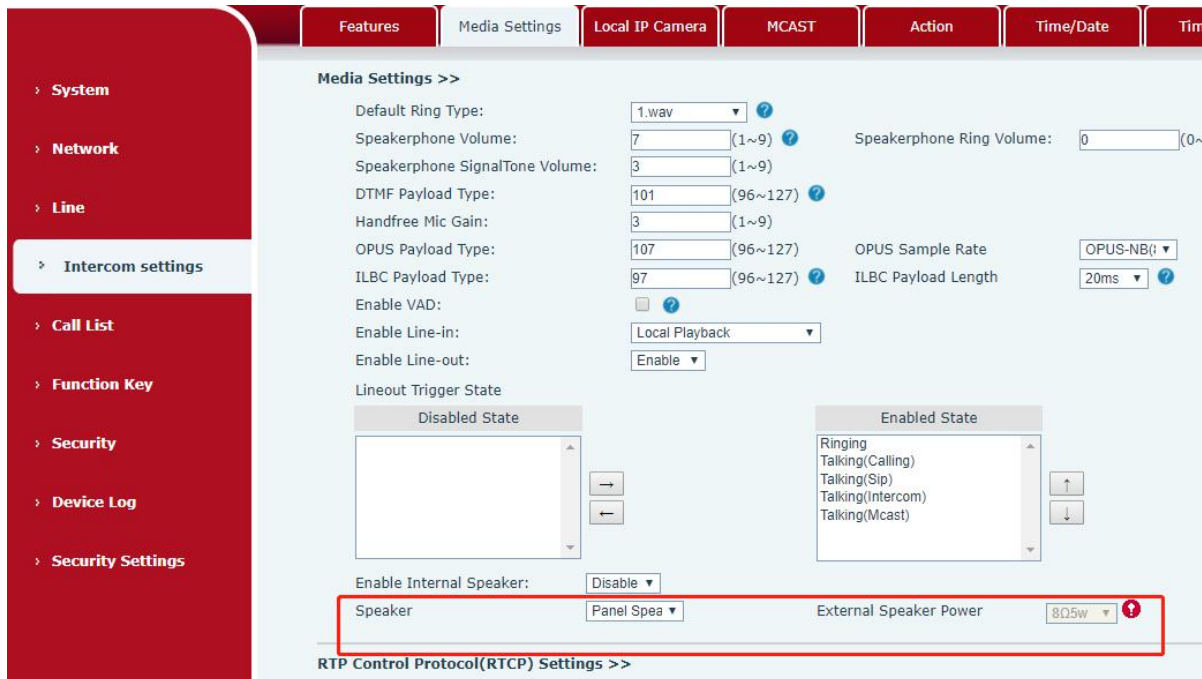
*Picture 4- Volume Set*

## 6.6 Set the player type

Set the player type (the default is panel speaker mode)

**[Phone Settings] >> [Media Settings] >> [Media Settings]**

The system defaults to the <panel speaker> mode, which is an intercom panel terminal with a shell. In order to ensure the voice effect of hands-free intercom and avoid damage to the speaker, When speaking, the output power is limited to less than 10W.



*Picture 5- Speaker*

If you need external speakers for broadcasting, you can adjust to the <external speakers> mode:

At this time, you can select 10W/20W/30W according to the power of the external speaker. Note that the corresponding power supply needs to be matched at this time:

*Table 5- Power Supply*

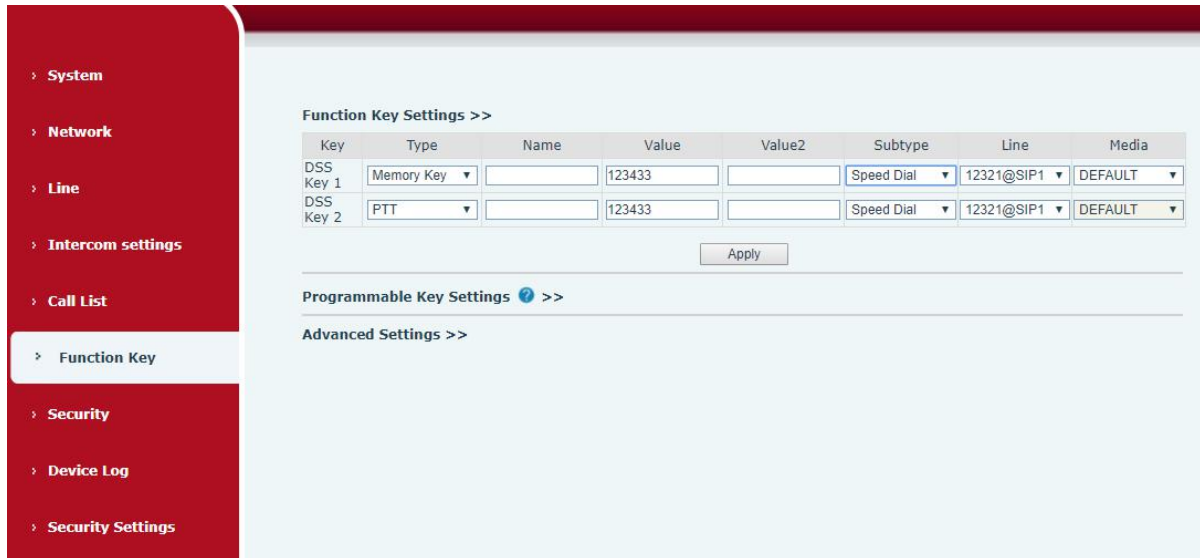
|                  | Output Power | Speaker type |
|------------------|--------------|--------------|
| <b>POE</b>       | 10W          | 10W/4 Ω      |
| <b>12V/2A DC</b> | 10W          | 10W/4Ω       |
| <b>18V/2A DC</b> | 20W          | 20W/4Ω       |
| <b>24V/2A DC</b> | 30W          | 30W/4Ω       |



## 7 Basic Function

### 7.1 Making Calls

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



*Picture 6- 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 [9.26 Function Key](#).

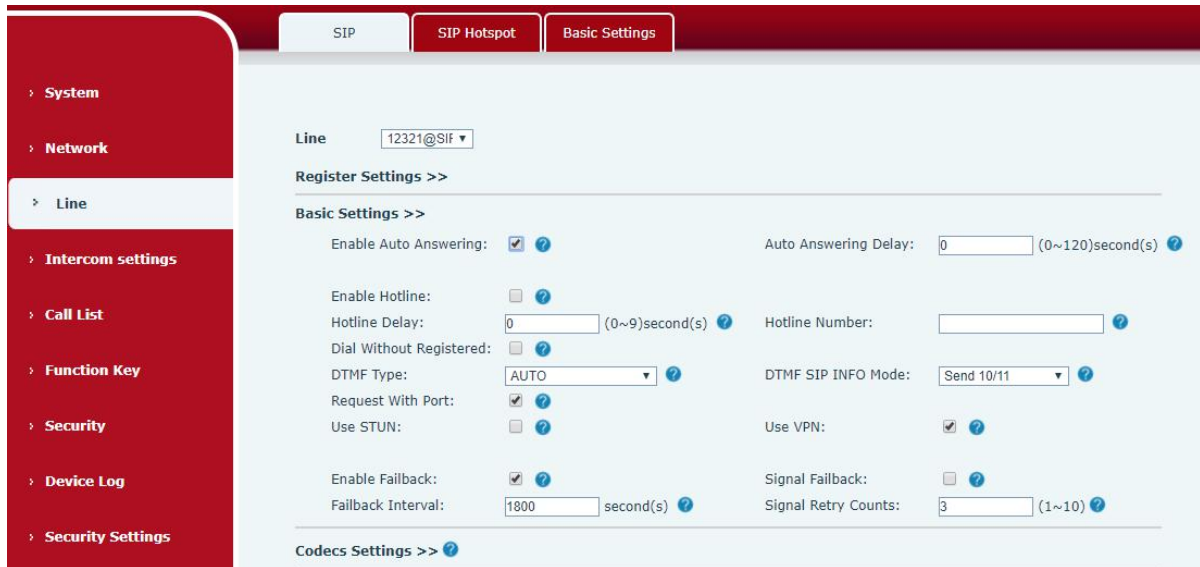
### 7.4 Auto Answer

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:

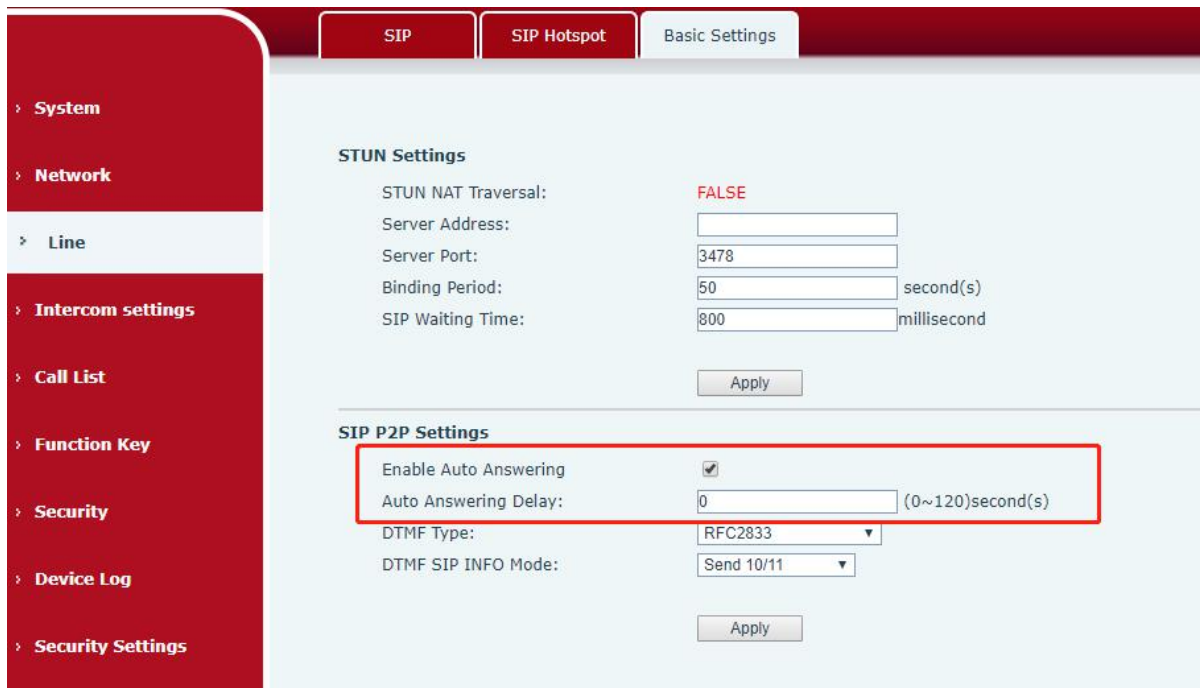
Enter [Line] >> [SIP], Enable auto answer and set auto answer time and click submit.



Picture 7 - WEB line enable auto answer

SIP P2P auto answering:

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



Picture 8- 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

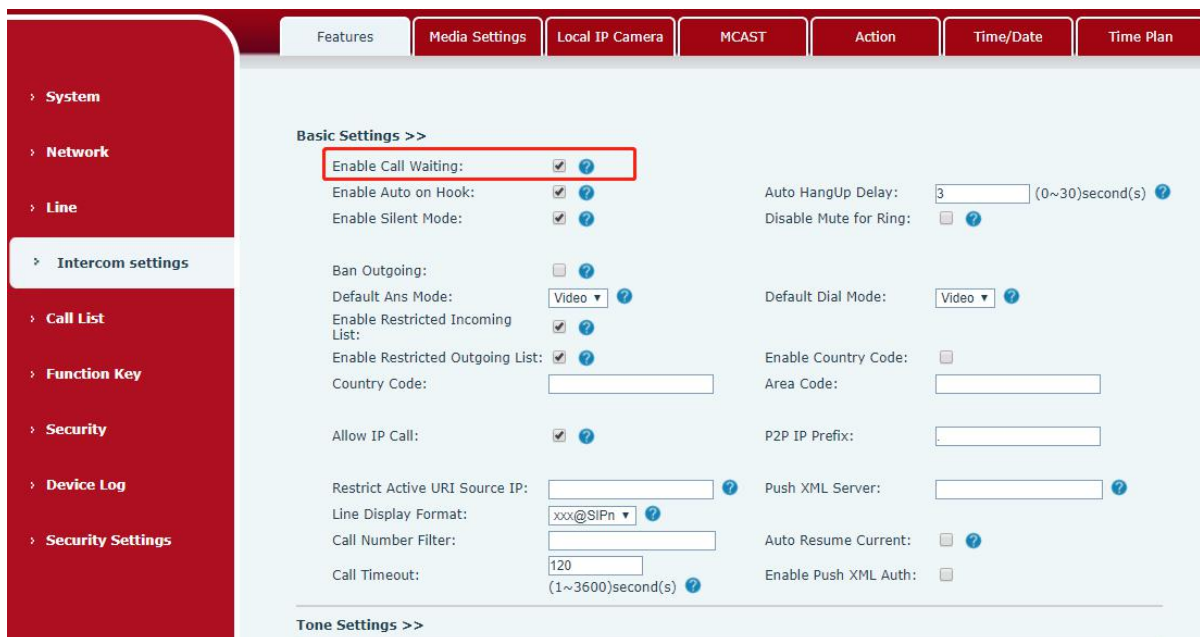
is set.

## 7.5 Call Waiting

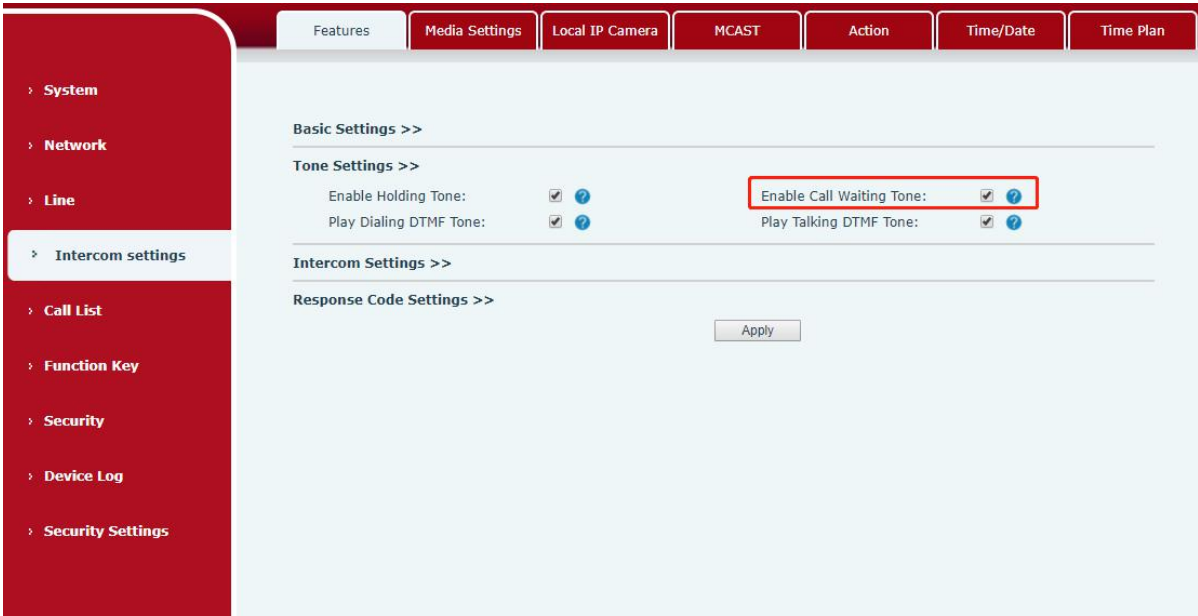
- Enable call waiting: new calls can be accepted during a call.
- Disable call waiting: new calls will be automatically rejected and a busy signal will be prompted
- Enable call waiting tone: when you receive a new call on the line, the device will beep.

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

- Web interface: enter **[Phone Settings]** >> **[Features]**, enable/disable call waiting, enable/disable call waiting tone.



*Picture 9 - Call Waiting*

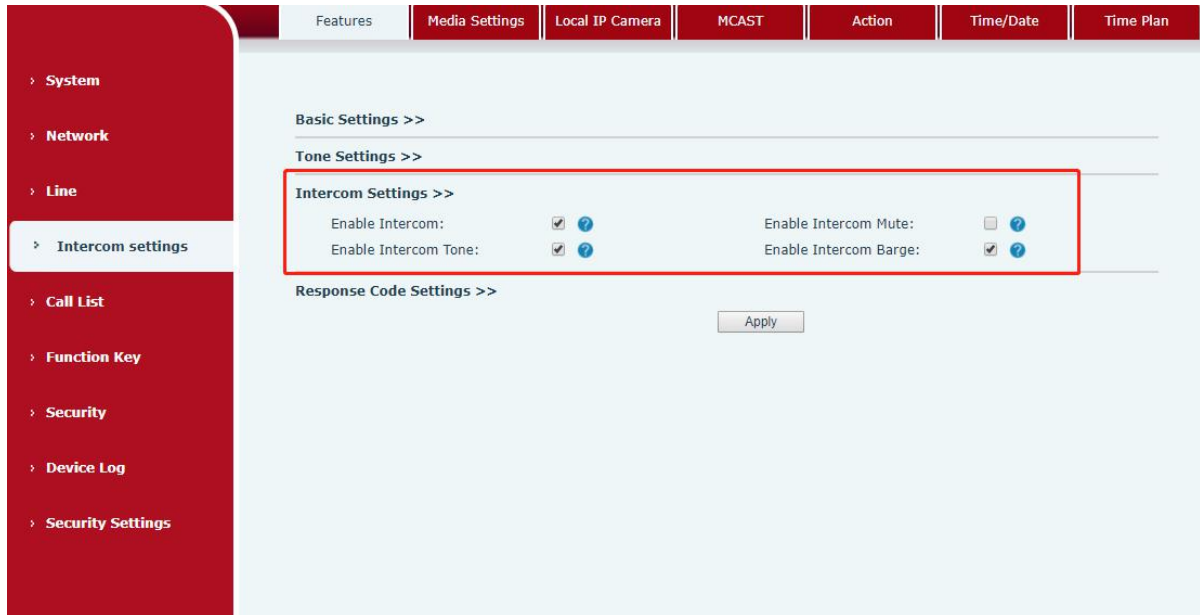


Picture 10 - Call Waiting tone

## 8 Advance Function

### 8.1 Intercom

The equipment can answer intercom calls automatically.



*Picture 11 - WEB Intercom*

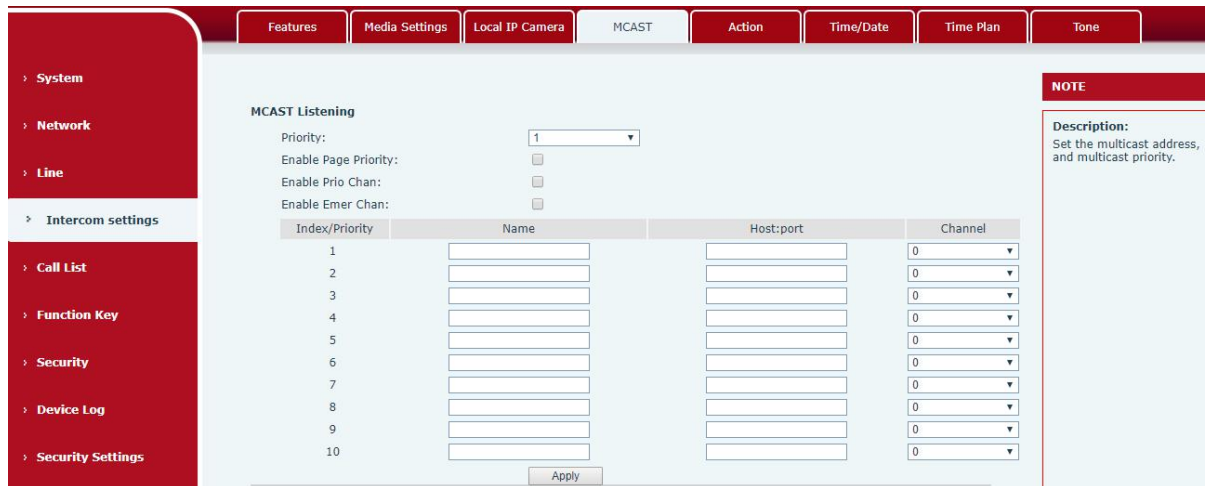
*Table 6- Intercom*

| Parameters              | Description  |
|-------------------------|--|
| Enable Intercom         | When the intercom system is enabled, the device will accept the SIP header call-info of the Call request<br>Command automatic call   |
| Enable Intercom Barge   | If the option is enabled, PA2S 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.



**Picture 12 - MCAST**

**Table 7- 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 Time | When a multicast call does not end normally, but for some reason the 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 **[Phone 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 the quantity 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.

*Table 8 - SIP Hotspot*

| 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 Address | The multicast address used by the client and server when the monitoring 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   |

|  |          |
|--|----------|
|  | 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.



*Picture 13 - 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

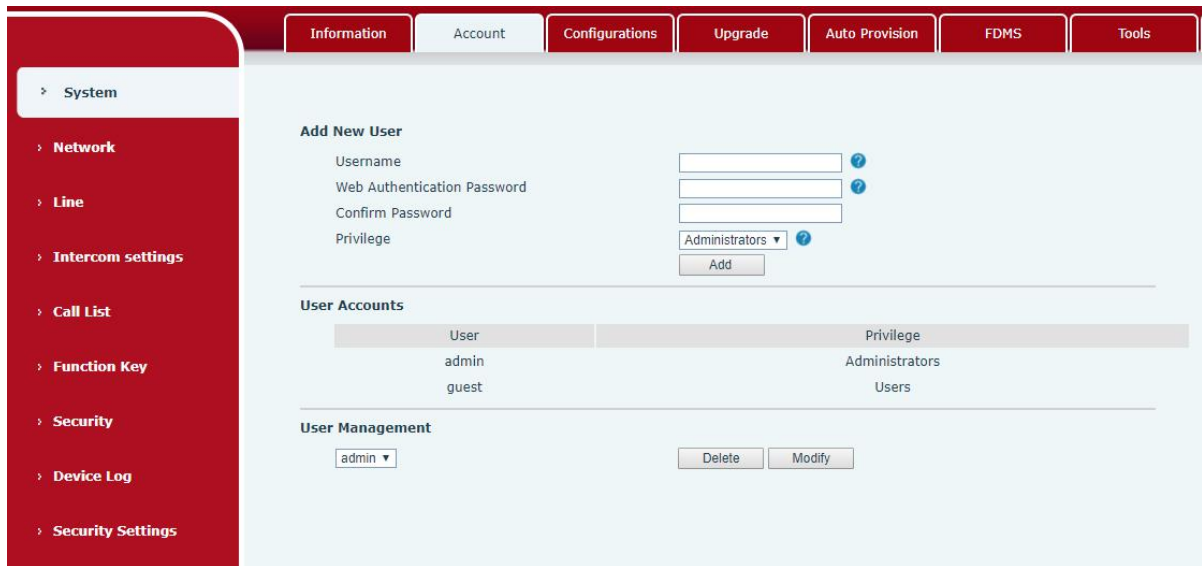
And summarization of network status,

- 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

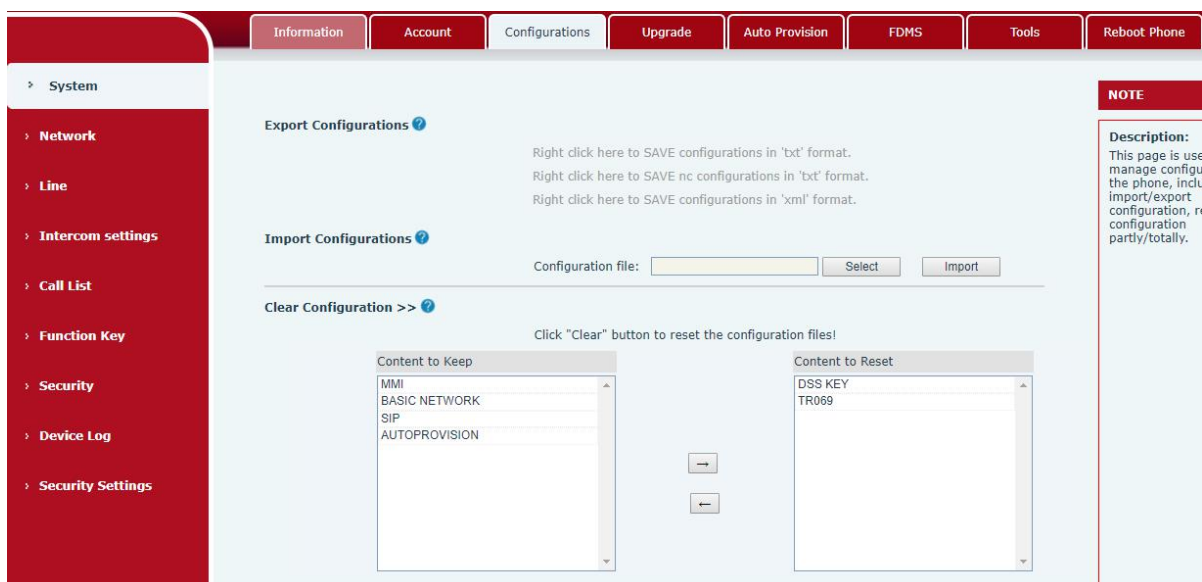


Picture 14- WEB Account

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.



Picture 15 - 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

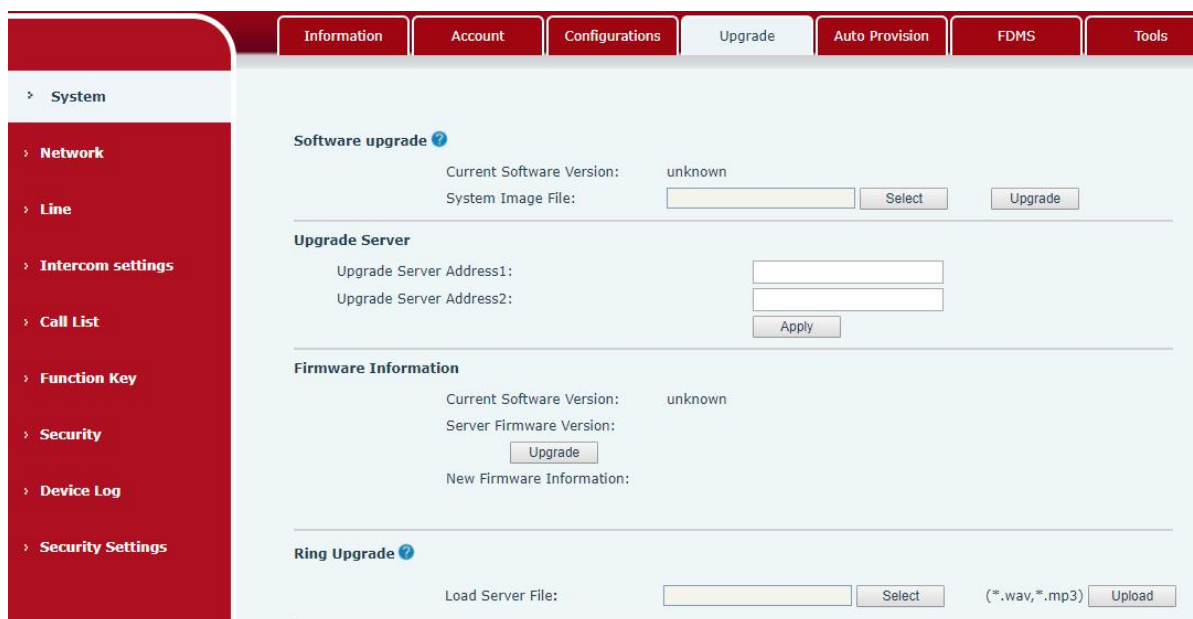
### ■ Clear Tables

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



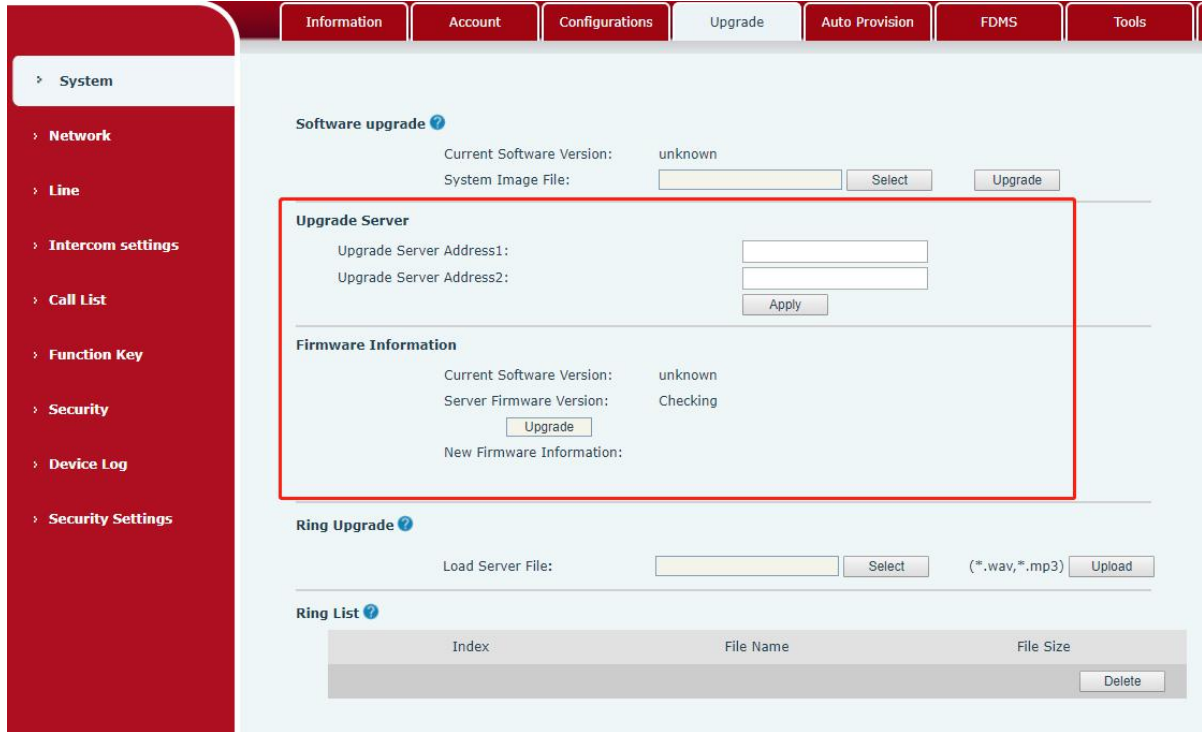
Picture 16- Upgrade

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 wav and MP3 format.

**Firmware Upgrade:**

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



*Picture 17 - Web page firmware upgrade*

*Table 9- Firmware upgrade*

| Parameter                   | Description  |
|-----------------------------|--|
| <b>Upgrade server</b>       |  |
| Enable Auto Upgrade         | Enable automatic upgrade, If there is a new version txt and new software firmware on the server, phone will show a prompt upgrade message after Update Interval. |
| Upgrade Server Address1     | Set available upgrade server address.  |
| Upgrade Server Address2     | Set available upgrade server address.  |
| Update Interval             | Set Update Interval.   |
| <b>Firmware Information</b> |  |
| Current Software Version    | It will show Current Software Version.   |
| Server Firmware Version     | It will show Server Firmware Version.  |
| [Upgrade] button            | If there is a new version txt and new software firmware on the server, the page will display version information   |

|                                     |   |
|-------------------------------------|---|
|                                     | and upgrade button will become available; Click [Upgrade] button to upgrade the new firmware.   |
| New version description information | When there is a corresponding TXT file and version on the server side, the TXT and version information will be displayed under the new version description information. |

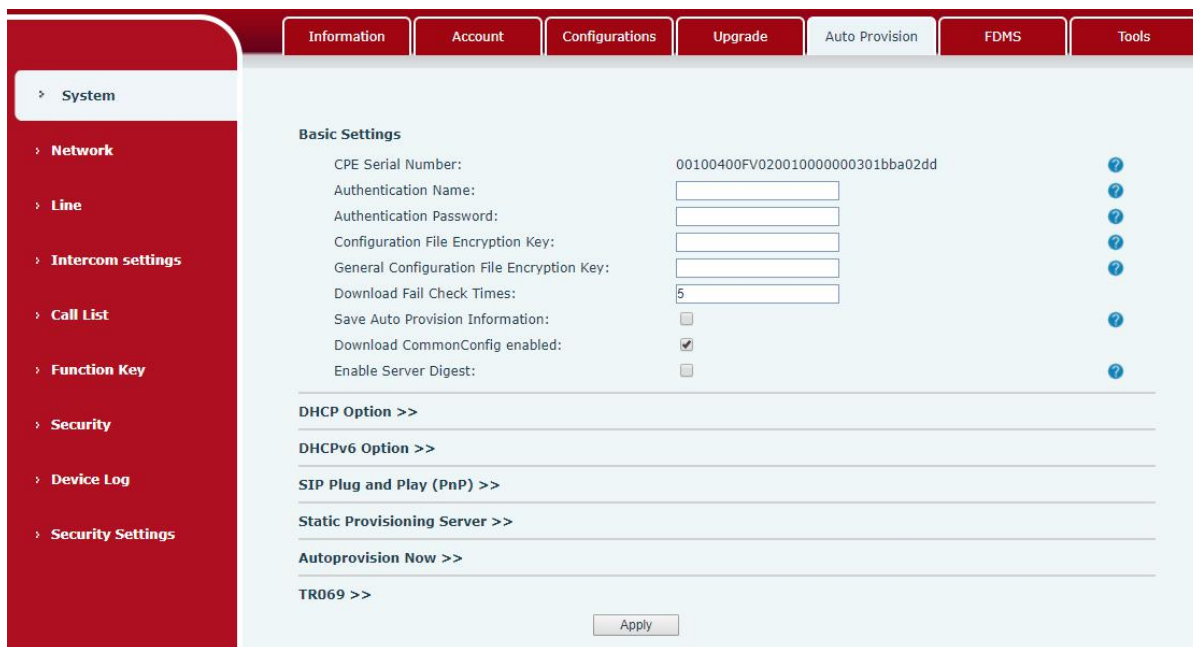
- The file requested from the server is a TXT file called vendor\_model\_hw10.txt.Hw followed by the hardware version number, it will be written as hw10 if no difference on hardware. All Spaces in the filename are replaced by underline.
- The URL requested by the phone is HTTP:// server address/vendor\_Model\_hw10 .txt: The new version and the requested file should be placed in the download directory of the HTTP server, as shown in the figure:

| 名称                                    | 修改日期            | 类型            | 大小        |
|---------------------------------------|-----------------|---------------|-----------|
| fanvil_x6_hww1_0.txt                  | 2018/9/11 17:57 | 文本文档          | 1 KB      |
| fanvil_x6_hww1_1.txt                  | 2018/9/11 17:57 | 文本文档          | 1 KB      |
| fanvil_x6_hww1_2.txt                  | 2018/9/11 17:57 | 文本文档          | 1 KB      |
| fanvil_x6_hww1_3.txt                  | 2018/9/11 17:57 | 文本文档          | 1 KB      |
| x6-6904-P0.12.12-1.6.3-2502T2018-0... | 2018/8/21 19:52 | WinRAR 压缩文... | 35,847 KB |

- TXT file format must be UTF-8
- vendor\_model\_hw10.TXT The file format is as follows:  
Version=1.6.3 #Firmware  
Firmware=xxx/xxx.z #URL , Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers.  
BuildTime=2018.09.11 20:00  
Info=TXT|XML  
  
Xxxxx  
Xxxxx  
Xxxxx  
Xxxxx
- After the interval of update cycle arrives, if the server has available files and versions, the phone will prompt as shown below. Click [view] to check the version information and upgrade.

## 9.6 System >> Auto Provision

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



**Picture 18- Auto provision settings**

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

**Table 10- Auto Provision**

| Auto provision                |   |
|-------------------------------|---|
| Parameters                    | Description   |
| <b>Basic settings</b>         |   |
| Current Configuration Version | Shows the current config file's version. If the version of the downloaded configuration file is same with this one, the configuration file will not be applied. If the device confirm the 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. |
| General Configuration Version | Shows the common config file's version. If the version of the downloaded configuration file is same with this one, the configuration file will not be applied. If the device confirm the 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. If this is blank the phone will use anonymous   |
| Authentication Password                   | Password for configuration server. Used for FTP/HTTP/HTTPS.   |
| Configuration File Encryption Key         | Encryption key for the configuration file   |
| General Configuration File Encryption Key | Encryption key for common configuration file  |
| Download Fail Check Times                 | The default value is 5. If the download configuration fails, it will be downloaded 5 times.   |
| Enable Get Digest From Server             | When the feature is enable, if the configuration of server is changed, phone will download and update.  |
| <b>DHCP Option</b>                        |   |
| 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 120                    | Set the SIP server address through DHCP option 120.   |
| <b>SIP Plug and Play (PnP)</b>            |   |
| Enable SIP PnP                            | Whether enable PnP or not. If PnP is enable, phone will send a SIP SUBSCRIBE message with broadcast method. Any server can support the feature will respond and send a Notify with URL to phone. Phone could get the configuration file with the URL. |
| Server Address                            | Broadcast address. As default, it is 224.0.0.0.   |
| Server Port                               | PnP port  |
| Transport Protocol                        | PnP protocol, TCP or UDP.   |
| Update Interval                           | PnP message interval.   |
| <b>Static Provisioning Server</b>         |   |
| Server Address                            | Set FTP/TFTP/HTTP server IP address for auto update. The address can be an IP address or Domain name with subdirectory.   |
| Configuration File Name                   | The configuration file name. If it is empty, phone will request the common file and device file which is named as its MAC address. The file name could be a common name, \$mac.cfg, \$input.cfg. The file format supports CFG/TXT/XML.                |
| Protocol Type                             | Transferring protocol type, supports FTP、TFTP、HTTP and HTTPS  |
| Update Interval                           | Configuration file update interval time. As default it is 1, means  |

|                           |   |
|---------------------------|---|
|                           | phone will check the update every 1 hour.   |
| Update Mode               | Provision Mode.<br>1. Disabled.<br>2. Update after reboot.<br>3. Update after interval. |
| <b>TR069</b>              |   |
| 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

*Picture 19 - FDMS*

*Table 11- FDMS*

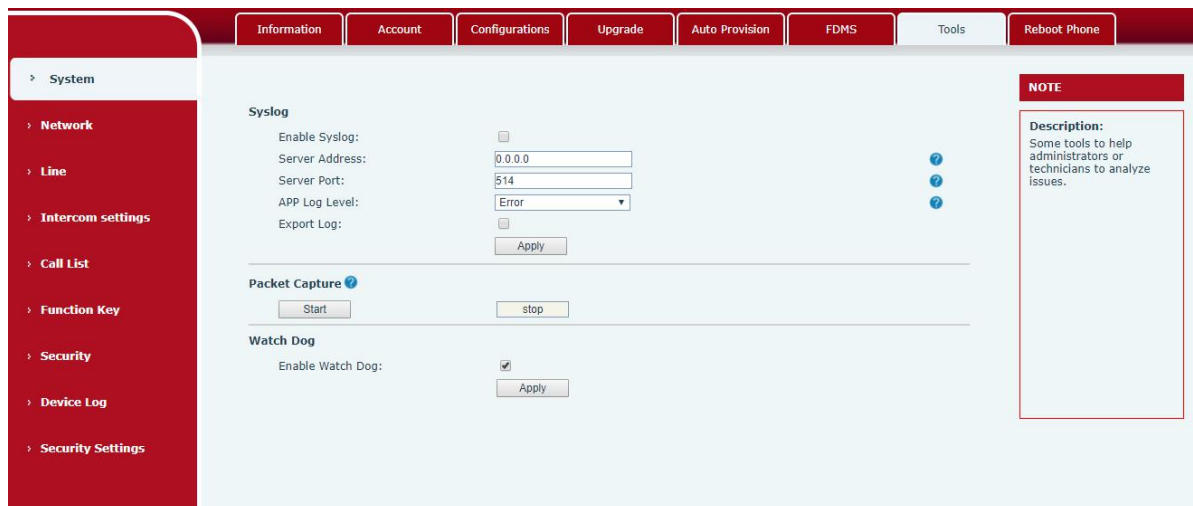
| FDMS information Settings |  |
|---------------------------|--|
| Community Designations    | Name of equipment installation community |



|                          |   |
|--------------------------|---|
| Building a movie theater | Name of equipment installation building |
| room number              | Equipment installation room name        |

## 9.8 System >> Tools

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

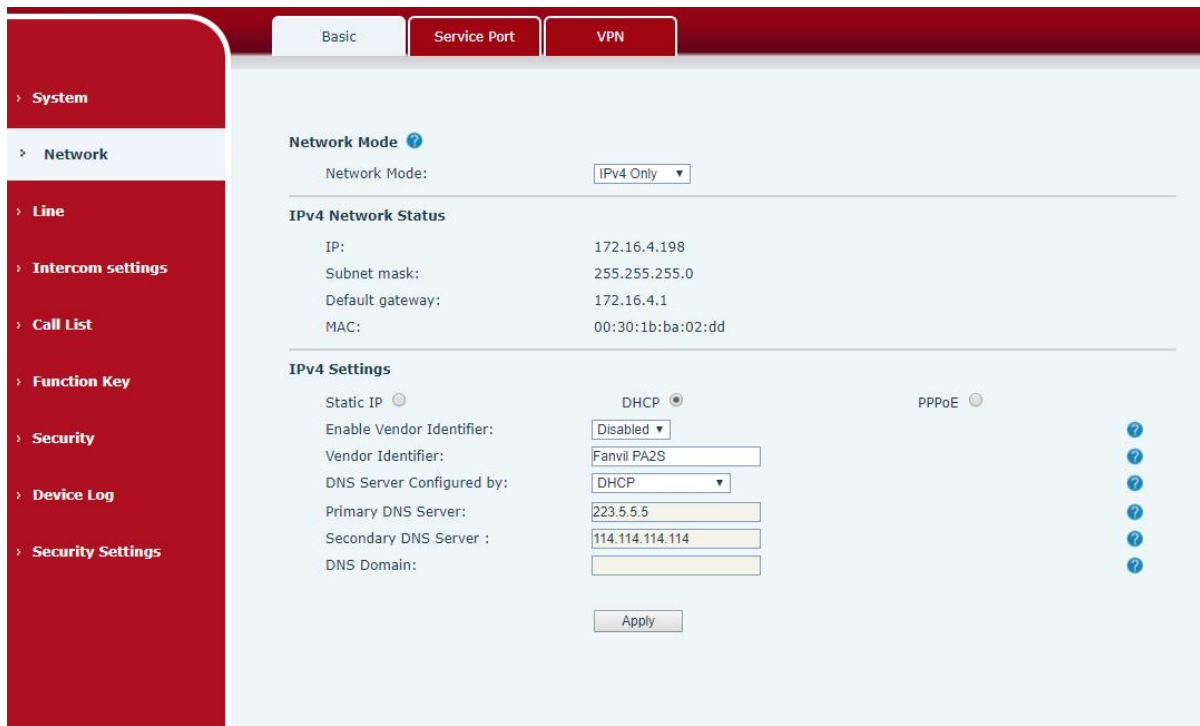


*Picture 20 - Tools*

**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 technical support.

## 9.9 Network >> Basic

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



**Picture 21 - Network Basic Setting**

**Table 12 - Network Basic Setting**

| Field Name   | Explanation  |
|--|--|
| <b>Network Status</b>  |  |
| IP   | The current IP address of the equipment  |
| Subnet mask  | The current Subnet Mask  |
| Default gateway  | The current Gateway IP address   |
| MAC  | The MAC address of the equipment   |
| MAC Time stamp   | Display the time when the device gets the MAC address  |
| <b>Settings</b>  |  |
| 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 chosen, the screen below will appear. Enter values provided by the ISP.   |  |
| DNS Server Configured by  | Select the Configured mode of the DNS Server.  |
| Primary DNS Server  | Enter the server address of the Primary DNS.   |
| Secondary DNS Server  | Enter the server address of the Secondary DNS. |
| <p>attention :</p> <p>1 ) After setting the parameters, click 【Apply】 to take effect.</p> <p>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.</p> <p>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</p> |  |

## 9.10 Network >> service port

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

The screenshot displays the 'Service Port Settings' configuration page. The left sidebar contains a navigation menu with the following items: System, Network, Line, Intercom settings, Call List, Function Key, Security, Device Log, and Security Settings. The main configuration area is titled 'Service Port Settings' and includes the following fields and controls:

- Web Server Type:** A dropdown menu set to 'HTTP'.
- Web Logon Timeout:** A text input field containing '15' with a unit indicator '(10~30)Minute'.
- web auto login:** A checkbox that is currently unchecked.
- HTTP Port:** A text input field containing '80'.
- HTTPS Port:** A text input field containing '443'.
- RTP Port Range Start:** A text input field containing '10000' with a unit indicator '(1025~65530)'.
- RTP Port Quantity :** A text input field containing '1000' with a unit indicator '(10~1000)'.

An 'Apply' button is located at the bottom right of the configuration area. On the right side of the form, there are several blue question mark icons for help.

Picture 22- Service port setting interface

*Table 13- 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 login | No need to enter the user name and password after the timeout, 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<br>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

The screenshot displays the VPN configuration page. On the left is a navigation menu with categories like System, Network, Line, Intercom settings, Call List, Function Key, Security, Device Log, and Security Settings. The main content area has three tabs: Basic, Service Port, and VPN. The VPN tab is selected, showing the following configuration options:

- Virtual Private Network (VPN) Status:** VPN IP Address: 0.0.0.0
- VPN Mode:**
  - Enable VPN:
  - Enable NAT:
  - L2TP:  OpenVPN:
  - Open VPN mode: tun
- Layer 2 Tunneling Protocol (L2TP):**
  - L2TP Server Address: 0.0.0.0
  - Authentication Name: [text input]
  - Authentication Password: [text input]
- OpenVPN Files:** A table listing files for configuration, CA Root Certification, Client Certification, and Client Key, each with a file size of N/A and buttons for Select, Upload, and Delete.

*Picture 23- 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,

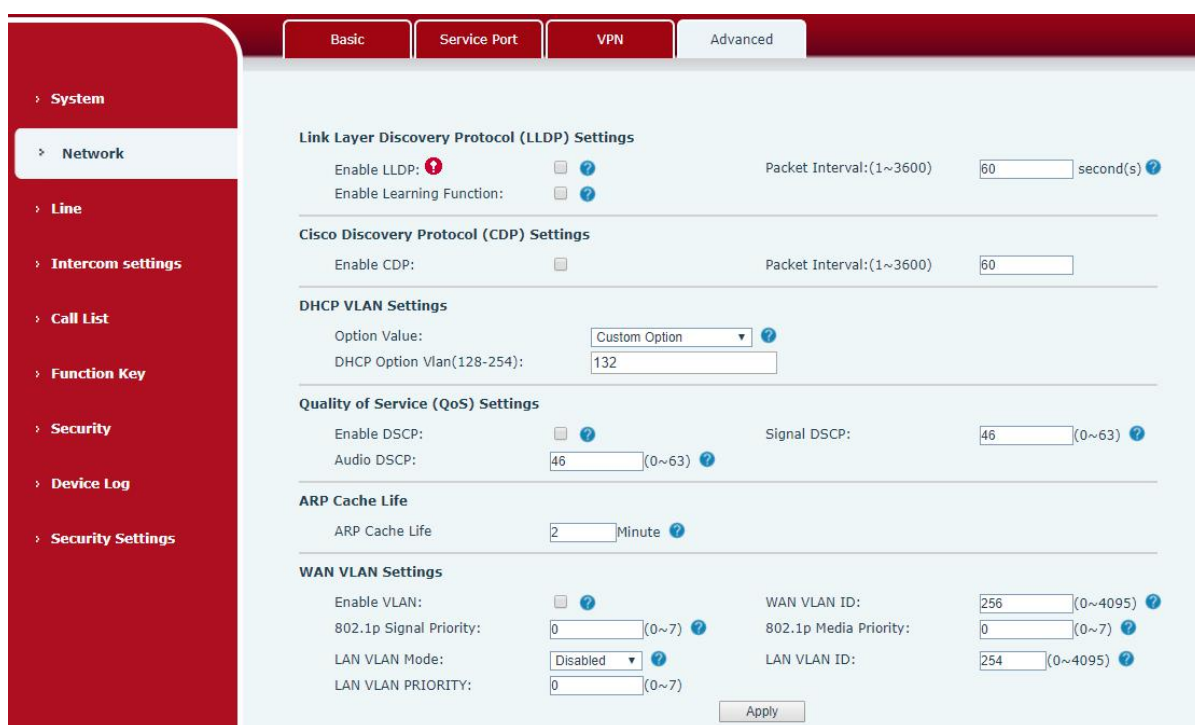
|                             |             |
|-----------------------------|-------------|
| OpenVPN Configuration file: | client.ovpn |
| CA Root Certification:      | ca.crt      |
| Client Certification:       | client.crt  |
| Client Key:                 | client.key  |

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

When the camera connection mode is external, the network advanced can be set. The local camera does not support advanced network settings.



*Picture 24 - Network Setting*

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

*Table 14- Network Setting*

| Field Name           | Explanation               |
|----------------------|---------------------------|
| <b>LLDP Settings</b> |                           |
| Enable LLDP          | Enable or disable LLDP    |
| Packet Interval      | LLDP Send detection cycle |

|                             |   |
|-----------------------------|---|
| Enable Learning Function    | Learn the discovered device information on the device |
| <b>QoS Settings</b>         |   |
| Pattern                     | Voice quality assurance (off by default)              |
| <b>DHCP VLAN Settings</b>   |   |
| parameters values           | 128-254, Obtain the VLAN value through DHCP           |
| <b>WAN port virtual Wan</b> |   |
| WAN port virtual Wan        | WAN port Settings                                     |
| <b>LAN port virtual LAN</b> |   |
| LAN port virtual LAN        | LAN port Settings                                     |
| <b>802.1X</b>               |   |
| Enable 802.1X               | Enable or disable 802.1X                              |
| Username                    | Confirm Username                                      |
| Password                    | Confirm Password                                      |

## 9.13 LINES >> SIP

SIP
SIP Hotspot
Basic Settings

- > System
- > Network
- > Line
- > Intercom settings
- > Call List
- > Function Key
- > Security
- > Device Log
- > Security Settings

Line: 12321@SIF

**Register Settings >>**

|  |   |
|--|---|
| Line Status: <span style="color: red;">Registered</span> | Activate: <input checked="" type="checkbox"/> |
| Username: <input type="text" value="12321"/>             | Authentication User: <input type="text"/>     |
| Display name: <input type="text"/>                       | Authentication Password: <input type="text"/> |
| Realm: <input type="text"/>                              | Server Name: <input type="text"/>             |

|  |  |
|--|--|
| <b>SIP Server 1:</b>   | <b>SIP Server 2:</b>   |
| Server Address: <input type="text" value="172.16.1.2"/>                            | Server Address: <input type="text"/>   |
| Server Port: <input type="text" value="5060"/>                                     | Server Port: <input type="text" value="5060"/>                                     |
| Transport Protocol: <span style="border: 1px solid #ccc; padding: 2px;">UDP</span> | Transport Protocol: <span style="border: 1px solid #ccc; padding: 2px;">UDP</span> |
| Registration Expiration: <input type="text" value="3600"/> second(s)               | Registration Expiration: <input type="text" value="3600"/> second(s)               |
| Proxy Server Address: <input type="text"/>   | Backup Proxy Server Address: <input type="text"/>                                  |
| Proxy Server Port: <input type="text" value="5060"/>                               | Backup Proxy Server Port: <input type="text" value="5060"/>                        |
| Proxy User: <input type="text"/>   |  |
| Proxy Password: <input type="text"/>   |  |

**Basic Settings >>**

### Basic Settings >>

|                          |                                     |                |                       |                                     |   |   |
|--------------------------|-------------------------------------|----------------|-----------------------|-------------------------------------|---|---|
| Enable Auto Answering:   | <input checked="" type="checkbox"/> | ?              | Auto Answering Delay: | <input type="text" value="0"/>      | (0~120)second(s)                        | ? |
| Enable Hotline:          | <input type="checkbox"/>            | ?              | Hotline Number:       | <input type="text"/>                |   | ? |
| Hotline Delay:           | <input type="text" value="0"/>      | (0~9)second(s) | ?                     | DTMF SIP INFO Mode:                 | <input type="text" value="Send 10/11"/> | ? |
| Dial Without Registered: | <input type="checkbox"/>            | ?              | Use VPN:              | <input checked="" type="checkbox"/> |   | ? |
| DTMF Type:               | <input type="text" value="AUTO"/>   | ?              | Signal Failback:      | <input type="checkbox"/>            |   | ? |
| Request With Port:       | <input checked="" type="checkbox"/> | ?              | Signal Retry Counts:  | <input type="text" value="3"/>      | (1~10)                                  | ? |
| Use STUN:                | <input type="checkbox"/>            | ?              |                       |                                     |   |   |
| Enable Failback:         | <input checked="" type="checkbox"/> | ?              |                       |                                     |   |   |
| Failback Interval:       | <input type="text" value="1800"/>   | second(s)      |                       |                                     |   |   |

### Codecs Settings >> ?

Disabled Codecs:

Enabled Codecs:

G.711U  
G.711A  
G.729AB  
iLBC  
opus  
G.722

### Advanced Settings >>

|                                 |   |   |                                    |                                      |           |   |
|---------------------------------|---|---|------------------------------------|--------------------------------------|-----------|---|
| Use Feature Code:               | <input type="checkbox"/>                | ? | Disable Blocking Anonymous Call:   | <input type="text"/>                 | ?         |   |
| Enable Blocking Anonymous Call: | <input type="text"/>                    | ? | Call Waiting Off Code:             | <input type="text"/>                 | ?         |   |
| Call Waiting On Code:           | <input type="text"/>                    | ? | Send Anonymous Off Code:           | <input type="text"/>                 | ?         |   |
| Send Anonymous On Code:         | <input type="text"/>                    | ? | Session Timeout:                   | <input type="text" value="0"/>       | second(s) | ? |
| Enable Session Timer:           | <input type="checkbox"/>                | ? | BLF Server:                        | <input type="text"/>                 | ?         |   |
| Response Single Codec:          | <input type="checkbox"/>                | ? | Keep Alive Interval:               | <input type="text" value="30"/>      | second(s) | ? |
| Keep Alive Type:                | <input type="text" value="UDP"/>        | ? | Blocking Anonymous Call:           | <input type="checkbox"/>             | ?         |   |
| Keep Authentication:            | <input type="checkbox"/>                | ? | Specific Server Type:              | <input type="text" value="COMMON"/>  | ?         |   |
| RTP Encryption(SRTP):           | <input type="text" value="Disabled"/>   | ? | Anonymous Call Standard:           | <input type="text" value="None"/>    | ?         |   |
| User Agent:                     | <input type="text"/>                    | ? | Ring Type:                         | <input type="text" value="Default"/> | ?         |   |
| SIP Version:                    | <input type="text" value="RFC3261"/>    | ? | Use Tel Call:                      | <input type="checkbox"/>             | ?         |   |
| Local Port:                     | <input type="text" value="5060"/>       | ? | Enable PRACK:                      | <input type="checkbox"/>             | ?         |   |
| Enable user=phone:              | <input type="checkbox"/>                | ? | Enable Long Contact:               | <input type="checkbox"/>             | ?         |   |
| Auto TCP:                       | <input type="checkbox"/>                | ? | Convert URI:                       | <input checked="" type="checkbox"/>  | ?         |   |
| Enable Rport:                   | <input checked="" type="checkbox"/>     | ? | Enable GRUU:                       | <input type="checkbox"/>             | ?         |   |
| DNS Mode:                       | <input type="text" value="A"/>          | ? | Enable Use Inactive Hold:          | <input type="checkbox"/>             | ?         |   |
| Enable Strict Proxy:            | <input checked="" type="checkbox"/>     | ? | Use 182 Response for Call waiting: | <input type="checkbox"/>             | ?         |   |
| Use Quote in Display Name:      | <input type="checkbox"/>                | ? | Enable SCA:                        | <input type="checkbox"/>             | ?         |   |
| Sync Clock Time:                | <input type="checkbox"/>                | ? | Server Expire:                     | <input checked="" type="checkbox"/>  | ?         |   |
| Caller ID Header:               | <input type="text" value="PAI-RPID-F"/> | ? | uaCSTA Number:                     | <input type="text"/>                 |           |   |
| Enable Feature Sync:            | <input type="checkbox"/>                | ? | Enable ChangePort:                 | <input type="checkbox"/>             |           |   |
| CallPark Number:                | <input type="text"/>                    | ? | Enable MAC Header:                 | <input type="checkbox"/>             |           |   |
| TLS Version:                    | <input type="text" value="TLS 1.2"/>    | ? |                                    |                                      |           |   |
| Enable Click To Talk:           | <input type="checkbox"/>                |   |                                    |                                      |           |   |
| Intercom Number:                | <input type="text"/>                    |   |                                    |                                      |           |   |
| Unregister On Boot:             | <input type="checkbox"/>                |   |                                    |                                      |           |   |



*Picture 25- SIP*

*Table 15 - SIP*

| <b>Parameters</b>           | <b>Description</b>   |
|-----------------------------|--|
| <b>Register Settings</b>    |  |
| 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   |
| Authentication User         | Enter the authentication user of the service account   |
| Authentication Password     | Enter the authentication password of the service account   |
| Username                    | Enter the username of the service account.   |
| Display Name                | Enter the display name to be sent in a call request.   |
| Activate                    | Whether the service of the line should be activated  |
| Realm                       | Enter the SIP domain if requested by the service provider  |
| SIP Proxy Server Address    | Enter the IP or FQDN address of the SIP proxy server   |
| Proxy Server Port           | Enter the SIP proxy server port, default is 5060   |
| Proxy User                  | Enter the SIP proxy user   |
| Proxy Password              | Enter the SIP proxy password   |
| Backup Proxy Server Address | Enter the IP or FQDN address of the backup proxy server  |
| Backup Proxy Server Port    | Enter the backup proxy server port, default is 5060  |
| <b>Basic Settings</b>       |  |
| Enable Auto Answering       | Enable auto-answering, the incoming calls will be answered automatically after the delay time                              |
| Auto Answering Delay        | Set the delay for incoming call before the system automatically answered it  |
| Call Forward Unconditional  | Enable unconditional call forward, all incoming calls will be forwarded to the number specified in                         |

|                                       |  |
|---------------------------------------|--|
|                                       | the next field   |
| Call Forward Number for Unconditional | Set the number of unconditional call forward   |
| Call Forward on Busy                  | Enable call forward on busy, when the phone is busy, any incoming call will be forwarded to the number specified in the next field   |
| Call Forward Number for Busy          | Set the number of call forward on busy   |
| Call Forward on No Answer             | Enable call forward on no answer, when an incoming call is not answered within the configured delay time, the call will be forwarded to the number specified in the next field                       |
| Call Forward Number for No Answer     | Set the number of call forward on no answer  |
| Call Forward Delay for No Answer      | Set the delay time of not answered call before being forwarded   |
| Transfer Timeout                      | Set the timeout of call transfer process   |
| Conference Type                       | Set the type of call conference, Local=set up call conference by the device itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server |
| Server Conference Number              | Set the conference room number when conference type is set to be Server  |
| Subscribe For Voice Message           | Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server                |
| Voice Message Number                  | Set the number for retrieving voice message  |
| Voice Message Subscribe Period        | Set the interval of voice message notification subscription  |
| Enable Hotline                        | Enable hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or turn on hands-free speaker or headphone                         |
| Hotline Delay                         | Set the delay for hotline before the system automatically dialed it  |
| Hotline Number                        | Set the hotline dialing number   |
| Dial Without Registered               | Set call out by proxy without registration   |
| Enable Missed Call Log                | If enabled, the phone will save missed calls into the call history record.   |

|                                    |  |
|------------------------------------|--|
| DTMF Type                          | Set the DTMF type to be used for the line  |
| DTMF SIP INFO Mode                 | Set the SIP INFO mode to send '*' and '#' or '10' and '11'   |
| Enable DND                         | Enable Do-not-disturb, any incoming call to this line will be rejected automatically   |
| Registration Expiration            | Set the SIP expiration interval  |
| Use VPN                            | Set the line to use VPN restrict route   |
| Use STUN                           | Set the line to use STUN for NAT traversal   |
| <b>Codec Settings</b>              | Set the priority and availability of the codecs by adding or remove them from the list.  |
| <b>Advanced Settings</b>           |  |
| Use Feature Code                   | When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field. |
| Enable DND                         | Set the feature code to dial to the server   |
| Disable DND                        | Set the feature code to dial to the server   |
| Enable Call Forward Unconditional  | Set the feature code to dial to the server   |
| Disable Call Forward Unconditional | Set the feature code to dial to the server   |
| Enable Call Forward on Busy        | Set the feature code to dial to the server   |
| Disable Call Forward on Busy       | Set the feature code to dial to the server   |
| Enable Call Forward on No Answer   | Set the feature code to dial to the server   |
| Disable Call Forward on No Answer  | Set the feature code to dial to the server   |
| Enable Blocking Anonymous Call     | Set the feature code to dial to the server   |
| Disable Blocking Anonymous Call    | Set the feature code to dial to the server   |
| Call Waiting On Code               | Set the feature code to dial to the server   |
| Call Waiting Off Code              | Set the feature code to dial to the server   |
| Send Anonymous On Code             | Set the feature code to dial to the server   |
| Send Anonymous Off Code            | Set the feature code to dial to the server   |
| SIP Encryption                     | Enable SIP encryption such that SIP transmission will be encrypted   |
| 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 SIP 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 Name         | Whether to add quote in display name   |
| Enable GRUU                       | Support Globally Routable User-Agent URI (GRUU)  |
| Sync Clock Time                   | Time Syncn with server   |
| Caller ID Header                  | Set the Caller ID Header   |
| Use 182 Response for Call waiting | Set the device to use 182 response code at call waiting response   |
| 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.<br>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 Syncn with server  |
| Enable SCA                        | Enable/Disable SCA (Shared Call Appearance )   |
| CallPark Number                   | Set the callPark number  |
| Server Expire                     |  |
| TLS Version                       | Choose TLS Version   |

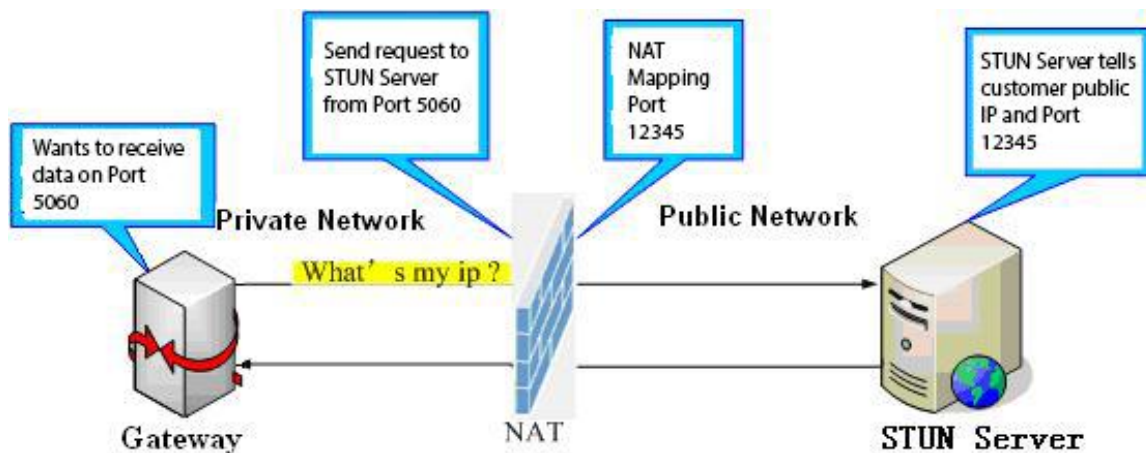
## 9.14 Line >> SIP Hotspot

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.

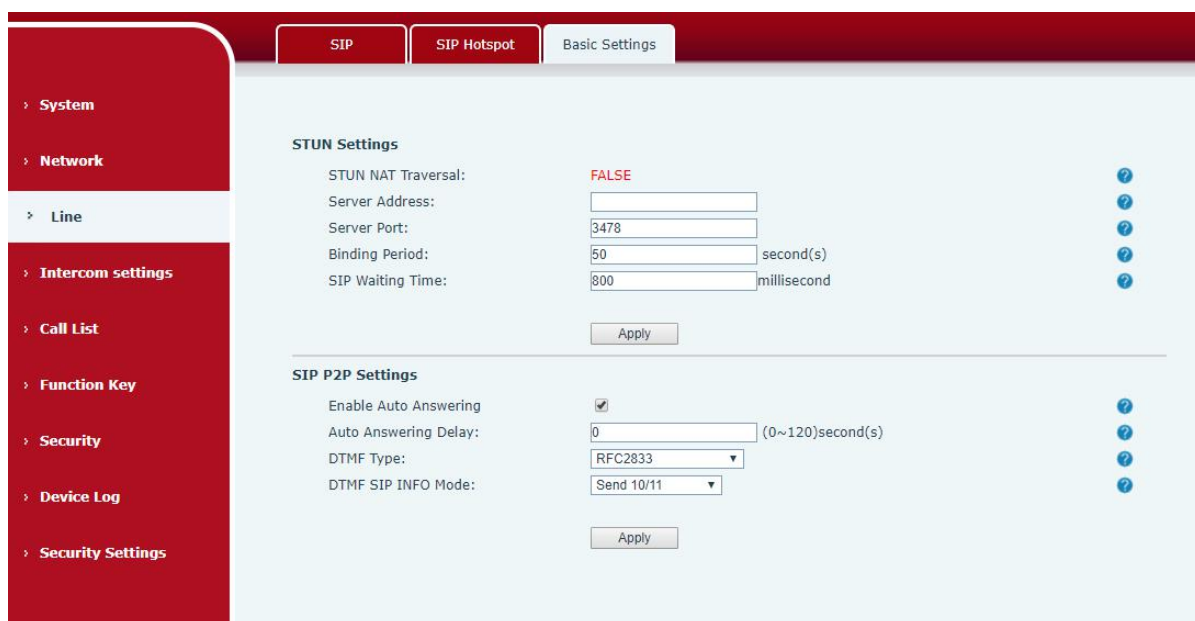
See [8.3 Hotspot](#) for details.

## 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 26- Basic Settings



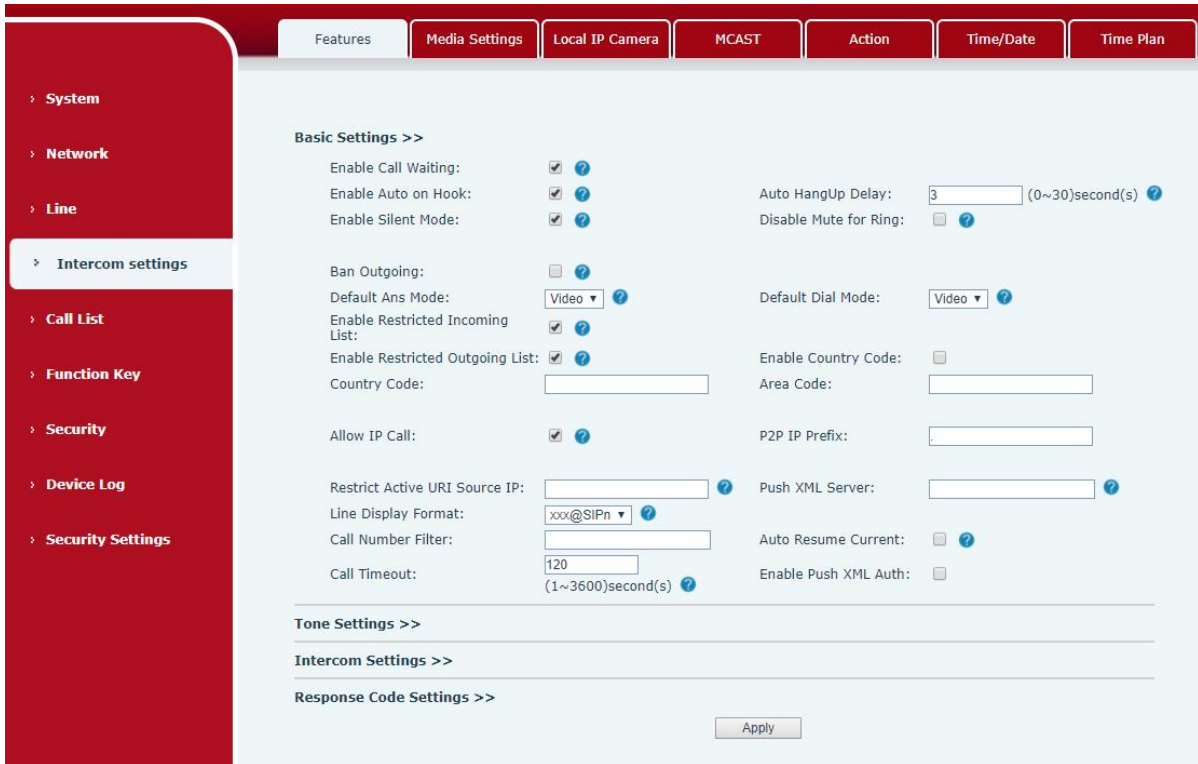
Picture 27 - Line Basic Setting

Table 16- Line Basic Setting

| Parameters              | Description   |
|-------------------------|---|
| <b>STUN Settings</b>    |   |
| 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                   |
| <b>SIP P2P Settings</b> |   |
| Enable Auto Answering   | Automatically answer incoming IP calls after the timeout period is enabled    |

|                      |  |
|----------------------|--|
| Auto Answering Delay | Automatic answer timeout setting                       |
| DTMF Type            | Set the DTMF type of the line.                         |
| DTMF SIP INFO Mode   | Set SIP INFO mode to send '*' and '#' or '10' and '11' |

## 9.16 Phone settings >> Features



Picture 28 - Feature

Table 17- Common device function Settings on the web page

| Parameters            | Description   |
|-----------------------|---|
| <b>Basic Settings</b> |   |
| Enable Call Waiting   | Enable this setting to allow user to take second incoming call during an established call. Default enabled.   |
| Enable Auto Handdown  | The phone will hang up and return to the idle automatically at hands-free mode  |
| Auto Handdown Time    | Specify Auto handdown time, the phone will hang up and return to the idle automatically after Auto Hand down time at hands-free mode, and play dial tone Auto handdown time at handset mode |
| Enable Silent Mode    | When enabled, the phone is muted, there is no ringing when calls, you   |

|                                 |  |
|---------------------------------|--|
|                                 | can use the volume keys and mute key to unmute.  |
| Disable Mute for Ring           | When it is enabled,you can not mute the phone.   |
| Default Ans Mode                | The video mode is used to answer calls by default, and it can be changed to voice answering  |
| Default Dial Mode               | The default outgoing method is video, which can be changed to voice outgoing   |
| 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                        |
| Disable AEC                     | When disabled, the device will not support AEC   |
| Restrict Active URI Source IP   | Set the device to accept Active URI command from specific IP address.  |
| 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   |
| Call Timeout                    | If the call is not answered, the call will be automatically hung up after the timeout  |
| Enable Push XML Auth            | To enable push xml auth, user password is required   |
| Description                     | Display the model of the device  |
| <b>Tone Settings</b>            |  |
| 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.   |
| <b>Intercom Settings</b>      |   |
| 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 |
| <b>Response Code Settings</b> |   |
| Busy Response Code            | Set the SIP response code on line busy  |
| Reject Response Code          | Set the SIP response code on call rejection   |

## 9.17 Phone settings >> media

The screenshot displays the 'Media Settings' configuration page. The interface includes a sidebar with navigation options: System, Network, Line, Intercom settings, Call List, Function Key, Security, Device Log, and Security Settings. The main content area is titled 'Media Settings >>' and contains several settings:

- Default Ring Type:** 1.wav
- Speakerphone Volume:** 7 (1~9)
- Speakerphone Ring Volume:** 0 (0~9)
- Speakerphone SignalTone Volume:** 3 (1~9)
- DTMF Payload Type:** 101 (96~127)
- Handfree Mic Gain:** 3 (1~9)
- OPUS Payload Type:** 107 (96~127)
- OPUS Sample Rate:** OPUS-NB(i)
- ILBC Payload Type:** 97 (96~127)
- ILBC Payload Length:** 20ms
- Enable VAD:**
- Enable Line-in:** Local Playback
- Enable Line-out:** Enable
- Lineout Trigger State:**
  - Disabled State:** (Empty list)
  - Enabled State:** Ringing, Talking(Calling), Talking(Sip), Talking(Intercom), Talking(Mcast)
- Enable Internal Speaker:** Disable
- Speaker:** Panel Spea
- External Speaker Power:** 8Q5w

Picture 29- Media Settings

Table 18- Audio Settings

| <b>Parameters</b>                            | <b>Description</b>   |
|--|--|
| <b>Codecs Settings</b>                       | Select the enabled and disabled voice codecs<br>codec:G.711A/U,G.722,G.729,ILBC,opus,G726-16/24/32/40,G723.1,MPA   |
| <b>Audio Settings</b>                        |  |
| 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 payload type                            | Enter the opus payload type, the value must be 96~127.   |
| OPUS Sample Rate                             | Set the opus sample rate · including OPUS-NB ( 8KHz), 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. |
| Enable Line-in                               | enable or disable the linein function  |
| Enable Line-out                              | enable or disable the lineout function   |
| Lineout Trigger State                        | After enabling, the corresponding operation will trigger the output port   |
| Enable Internal Speaker                      | When enabled, the sound is played from the internal speakers   |
| Speaker                                      | Support panel speaker and external speaker   |
| External Speaker Power                       | External speaker power , support 10W, 20W, 30W, when using the corresponding speaker, you must select the corresponding power supply.                      |
| Video Direction                              | Set Send And Receive,Send Only,Receive Only  |
| <b>RTP Control Protocol(RTCP) Settings</b>   |  |
| CNAME user                                   | Set the CNAME user   |
| CNAME host                                   | Set the CNAME host   |
| <b>RTP</b>                                   |  |
| RTP keep alive                               | Keep talking, send a packet 30 seconds after enable it   |
| <b>Alert Info Ring Settings (alert-info)</b> |  |

|                                       |  |
|---------------------------------------|--|
| Value of notification message 1 to 10 | Set the value of the specified ring type |
| ring type                             | The ring type                            |

## 9.18 Phone settings >> Camera Settings

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

*Picture 30- Camera Settings*

*Table 19- Camera Settings*

| Parameters         | Description   |
|--------------------|---|
| camera settings    |   |
| White Balance Mode | <p>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. .</p> <p>Lock mode: Fixed white balance parameters will not be automatically adjusted according to the actual color temperature.</p> <p>Incandescent lamp mode: To compensate for the hue of incandescent lamps, it is suitable for use under beige light sources (bulbs, tungsten lamps, candles) and other light sources of this type. .</p> <p>Warm light mode: Compensate the hue of warm light, suitable for light sources with a color temperature of about 2700K.</p> |

|                 |   |
|-----------------|---|
|                 | <p>Natural light mode: It can be used for white balance in outdoor shooting and has a wide range of applications.</p> <p>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.</p>   |
| Exposure Mode   | <p><b>Auto mode</b> : The camera automatically sets the parameters, no need for the operator to adjust.</p> <p><b>Manual exposure time</b> : Set the exposure time by yourself, the range is 0~10000</p> <p><b>Manual exposure gain</b>: Set the exposure gain by yourself, the range is 0~1024</p> <p>All manual : Manually set the exposure time and gain.</p>  |
| Exposure Time   | <p>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 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.</p> |
| Exposure Gain   | <p>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 signal. The gain is generally only used when the signal is weak, but you do not want to increase the exposure time.</p>  |
| Contrast Mode   | <p>Auto mode: The camera automatically sets the contrast according to the environment, no need for the operator to adjust</p> <p>Manual mode: Manually set the camera's contrast parameters.</p>  |
| Contrast        | <p>Contrast refers to the contrast between light and dark in the picture. Increase the contrast, the brighter areas will be brighter and the darker areas will be darker, and the contrast between light and dark will increase.</p>  |
| Saturation Mode | <p>Auto mode: The camera automatically sets the saturation according to the environment, without the need for the operator to adjust</p> <p><b>Manual mode</b>: Manually set the camera's saturation parameters.</p>  |
| Saturation      | <p>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 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.</p>  |
| Sharpness Mode  | <p>Auto mode: The camera automatically sets the sharpness according to the environment, no need for the operator to adjust</p>  |

|                     |   |
|---------------------|---|
|                     | <b>Manual mode:</b> Manually set the sharpness parameters of the camera   |
| Sharpness           | Sharpness is sometimes called "sharpness", which is an indicator that reflects the sharpness of the image plane and the sharpness of the edges of 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 Auth   | Is authentication required when using onvif protocol (with username and password)   |
| Enable Rtsp Auth    | When using rtsp protocol, whether authentication is required (with username and password)   |
| H.264 Payload Type  | Set the load type of h.264, the range is 96~127   |
| <b>Osd Settings</b> |   |
| Osd Time            | Turn on/off the date display of the camera image interface.   |
| Osd Text            | Enable/disable the text display of the camera image interface.  |
| Title Message       | Text display content of camera image interface  |
| <b>Video Codecs</b> |   |
| H264 Video Stream   | Support H.264 encoding format   |
| Bitrate Control     | VBR: Video call will adapt to the bit rate of the opposite end, so that the video effect is better.<br>CBR: The video call will not change according to the bit rate set by itself.   |
| Resolution          | Support 1080P, 720P, 4CIF,VGA,CIF,QVGA  |
| Frame Rate (fps)    | The larger the value is, the more fluent the video is, and the higher the requirement for network bandwidth is; adjustment is not recommended   |
| Profile             | Minimum configuration: support I / P frame, only support progressive and CAVLC. It is generally used for low-level or applications requiring additional fault tolerance, such as video call, mobile video, etc<br>Main configuration: provide I / P / B frames, support progressive and interleaved, and support CAVLC and CABAC, |
| BitRate             | It refers to the data flow used by video files in unit time, also known as code 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. |
| <b>RTSP Information</b> |   |
| Main Stream Url         | Display the main stream URL address           |
| Sub Stream Url          | Display the sub stream URL address            |

The screenshot shows a configuration window titled "SnapShot". It contains the following elements:

- Snapshot Trigger Mode:** A label with a corresponding input field.
- Snapshot By State:** A label followed by three checkboxes: "Talking", "Ringing", and "Calling".
- Server Url:** A text input field.
- Username:** A text input field.
- Password:** A text input field.
- Apply:** A button located below the Username and Password fields.

*Picture 31 - Snapshot*

Capture trigger mode: call state trigger

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

Snapshot save path: local

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

## 9.19 Phone Setting >> MCAST

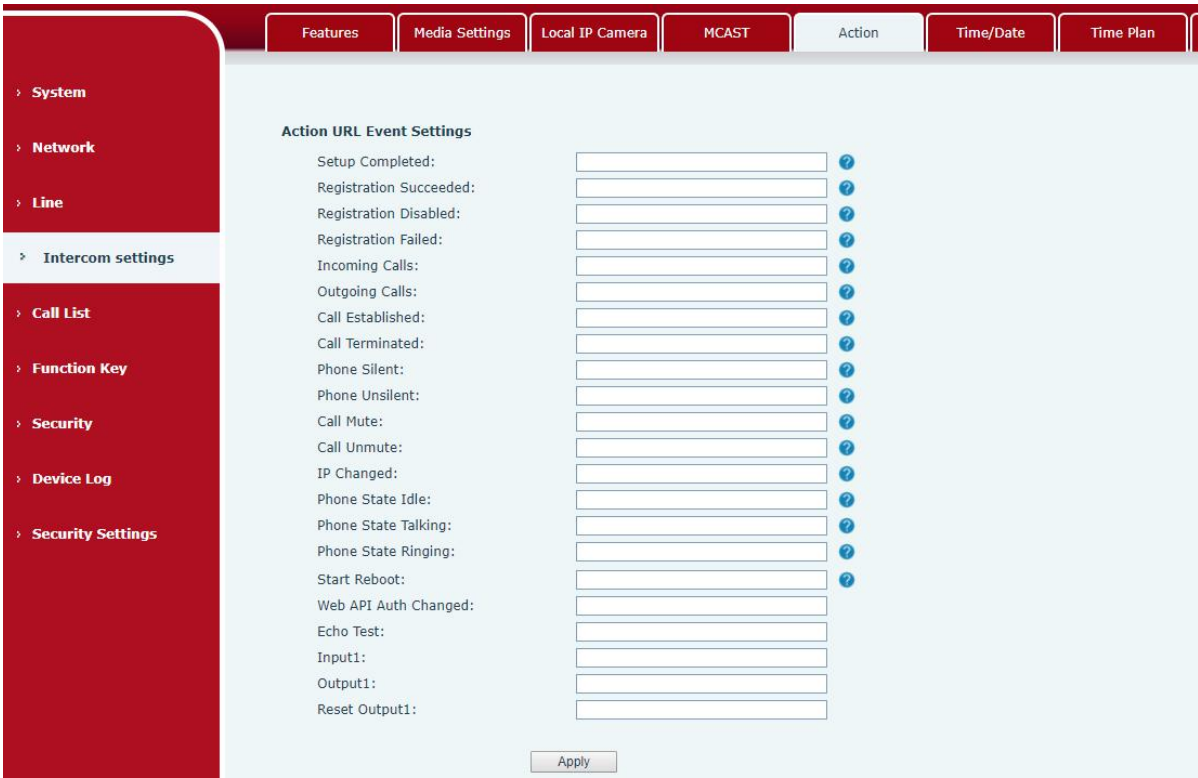
It is easy and convenient to use multicast function to send notice to each member of the multicast via setting the multicast key on the device and sending multicast RTP stream to pre-configured multicast address. By configuring monitoring multicast address on the device, monitor and play the RTP stream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

The detail for [8.2 MCAST](#)

## 9.20 Phone Setting >> Action URL

*Table 20- Action URL*

|  |
|--|
| <b>Action URL Event Settings</b>   |
| URL for various actions performed by the phone. These actions are recorded and sent as xml files to the server. Sample format is http://InternalServer /FileName.xml |

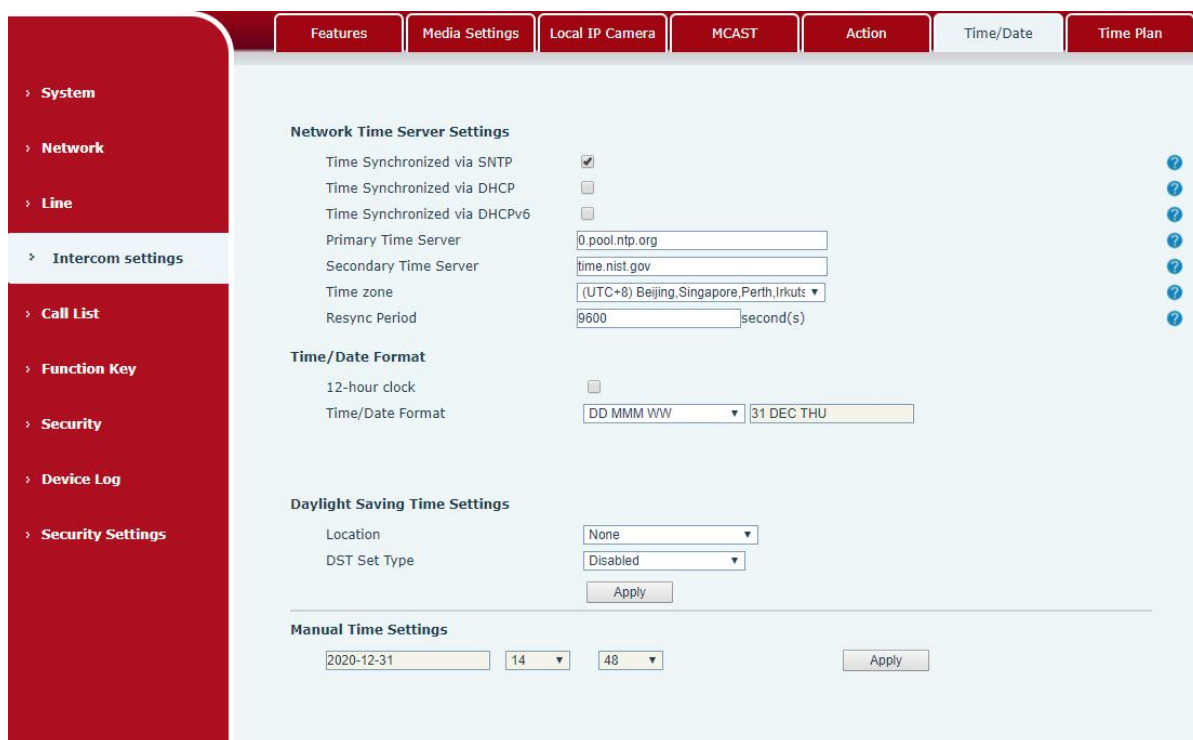


**Picture 32- Action URL**

**Note!** The operation URL is used by the IPPBX system to submit device events.

**9.21 Phone Setting >> Time/Date**

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



Picture 33 - Time/Date

Table 21- Time/Date

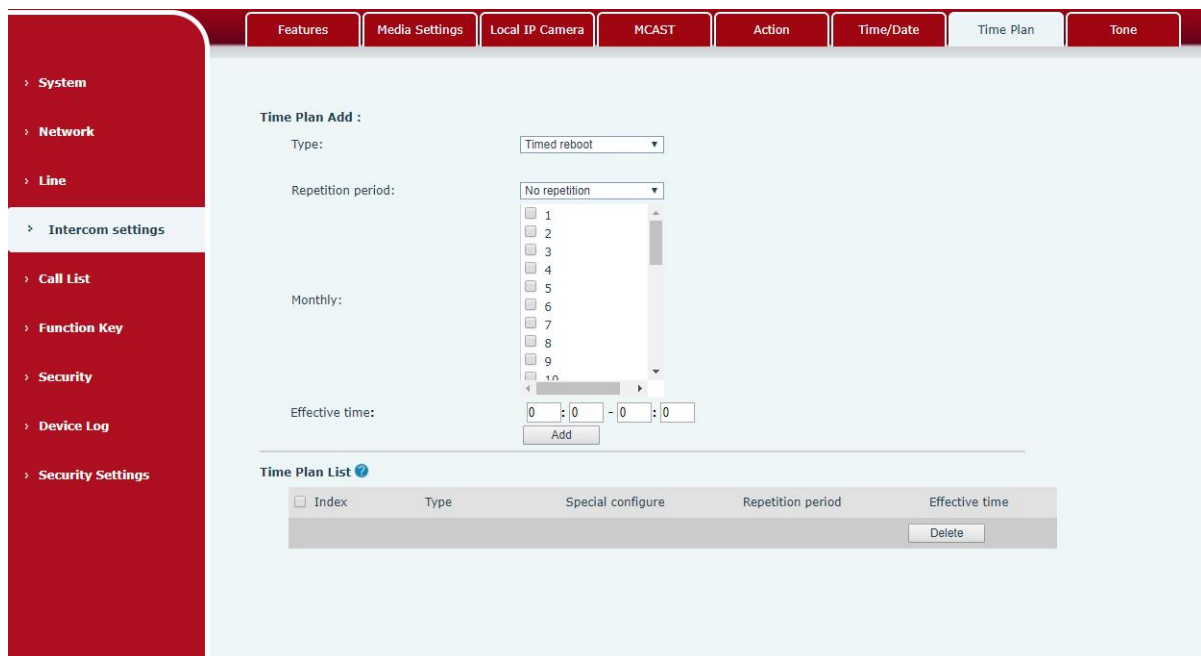
| Time/Date                            |   |
|--------------------------------------|---|
| Field Name                           | Explanation   |
| <b>Network Time Server Settings</b>  |   |
| Time Synchronized via SNTP           | Enable time-sync through SNTP protocol  |
| Time Synchronized via DHCP           | Enable time-sync through DHCP protocol  |
| Primary Time Server                  | Set primary time server address   |
| Secondary Time Server                | Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization. |
| Time zone                            | Select the time zone  |
| Resync Period                        | Time of re-synchronization with time server   |
| <b>Daylight Saving Time Settings</b> |   |
| Location                             | Select the user's time zone specific area   |
| DST Set Type                         | Select automatic DST according to the preset rules of DST, or the 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    |
| <b>Manual Time Settings</b>   |                     |
| 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<br>(SIP automatic get >SNTP automatic get >>manual manual setting)  |                     |

## 9.22 Phone settings >> Time plan

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



*Picture 34- Time Plan*

*Table 22- Time Plan*

| Parameters | Description   |
|------------|---|
| type       | Timing restart, timing upgrade, timing sound detection, timing playback audio |
| Audio path | Support local<br>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<br>Daily: Perform this operation in the same time frame every day<br>Weekly: Do this in the time frame of the day of the week<br>Monthly: the time frame of the month to perform this operation |
| Effective time | Set the time period for execution   |

## 9.23 Phone 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.

The screenshot displays the 'Tone Settings' configuration page. On the left is a navigation menu with categories like System, Network, Line, Intercom settings, Call List, Function Key, Security, Device Log, and Security Settings. The main area contains a list of tone settings, each with a text input field and a help icon. The 'Select Your Tone' dropdown is set to 'United States'. Below the settings is an 'Apply' button. On the right, a 'NOTE' box provides a detailed explanation of the tone configuration syntax, including examples like '350+440/0' and '440+480/2000,0/4000'.

Picture 35- Tone

## 9.24 Call 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

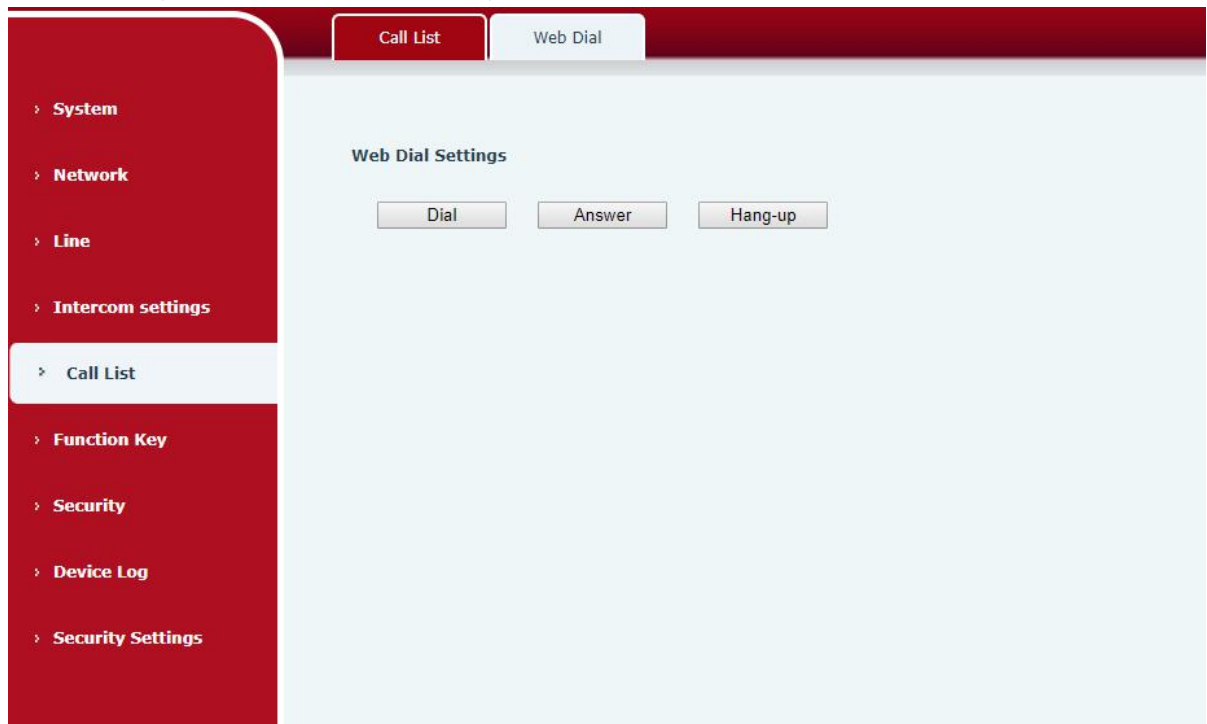
prefix will all be blocked.

■ Restrict Outgoing Call

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

## 9.25 Call list >> Web Dial

Use web page to call, answer and hang up.



*Picture 36- Webpage Dial*

## 9.26 Function key

- > System
- > Network
- > Line
- > Intercom settings
- > Call List
- > Function Key
- > Security
- > Device Log
- > Security Settings

**Function Key Settings >>**

| Key       | Type         | Name                 | Value                | Value2               | Subtype      | Line         | Media     |
|-----------|--------------|----------------------|----------------------|----------------------|--------------|--------------|-----------|
| DSS Key 1 | Memory Key ▼ | <input type="text"/> | 123433               | <input type="text"/> | Speed Dial ▼ | 12321@SIP1 ▼ | DEFAULT ▼ |
| DSS Key 2 | None ▼       | <input type="text"/> | <input type="text"/> | <input type="text"/> | None ▼       | AUTO ▼       | DEFAULT ▼ |

---

**Programmable Key Settings ? >>**

**Advanced Settings >>**

**Programmable Key Settings ? >>**

| Key  | Desktop   | Dialer    | Ringing              | Talking              | Desktop Long Pressed |
|------|-----------|-----------|----------------------|----------------------|----------------------|
| Key1 | Dsskey1 ▼ | Dsskey1 ▼ | Answer ▼             | End ▼                | None ▼               |
| Key2 | None ▼    | None ▼    | <input type="text"/> | <input type="text"/> | None ▼               |

**Advanced Settings >>**

Dial Mode Select

Call Switched Time  (5~50)second(s)

First Number Start Time  (00:00~23:59)      First Number End Time  (00:00~23:59)

*Picture 37- Function Key*

*Table 23- Function Key*

| Parameters                   | Description  |
|------------------------------|--|
| <b>Function key settings</b> |  |
| memory                       | <p><b>Speed Dial:</b>The user can directly dial the set number. This feature is convenient for customers to dial frequent numbers.</p> <p><b>Intercom:</b> This feature allows the operator or secretary to quickly connect to the phone, widely used in office environments</p> |
| 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                       | <b>Speed dial:</b> Make a call when pressed, and end the call when lifted.<br>Intercom: Start the intercom when pressed, and end the intercom when lifted.<br>Multicast: Initiate multicast when pressed, and end multicast when lifted  |
| Programmable Key Settings |  |
| Desktop                   | None: Nothing happens when you press the speed dial<br>Dsskey1: When it is set to dsskey1, follow the settings of dsskey1 to make call, answer, etc.<br>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<br>Dsskey1: When it is set to dsskey1, follow the settings of dsskey1 to make call, answer, etc.<br>Dsskey2: When it is set to dsskey2, perform operations such as calling and answering according to the setting of dsskey2   |
| Ringling                  | Answer: Set to answer, when there is an incoming call, if auto answer is disabled, press the speed dial key to answer the call<br>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<br>Volume up: set as volume up button, when there is a call, press the speed dial button to increase the volume<br>Volume down: set as volume up button, when there is a call, press the speed dial button to decrease the volume<br>Dsskey1: When it is set to dsskey1, follow the settings of dsskey1 to make call, answer, etc.<br>Dsskey2: When it is set to dsskey2, perform operations such as calling and answering according to the setting of dsskey2 |
| Desktop<br>Pressed        | Long<br>None: Long press the speed dial key does not respond<br>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        |   |
| Hot Key Dial Mode Select | Number 1 call number 2 mode selection.<br><Main/Secondary>: If the first number is not answered within the set time, the second number will be automatically switched.<br><Day/Night> : The system time is automatically detected during the call. If 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 "06:00"   |
| Day End Time             | The end time of the day when the <Day/Night> mode is defined. Default "18:00"   |

**table 20 - Function Key**

➤ **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       | Type       | Name | Value  | Value2 | Subtype    | Line       | Media   |
|-----------|------------|------|--------|--------|------------|------------|---------|
| DSS Key 1 | Memory Key |      | 123433 |        | Speed Dial | 12321@SIP1 | DEFAULT |
| DSS Key 2 | None       |      |        |        | None       | AUTO       | DEFAULT |

Apply

**Picture 38 - Memory Key**

**Table 24- Memory Key**

| Type   | number                                       | line                                      | Subtype    | usage   |
|--------|--|---|------------|---|
| memory | Fill in the SIP account or IP address of the | The line corresponding to the SIP account | Speed Dial | Using the speed dial mode, press the button to quickly dial the set number.                                     |
|        |  |   | Intercom   | Using the intercom mode, when the SIP phone at the opposite end supports the intercom function, the call can be |

|  |              |  |  |                         |
|--|--------------|--|--|-------------------------|
|  | called party |  |  | 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:

*Picture 39- Multicast*

*Table 25- Web Multicast*

| Type      | Number  | Subtype |
|-----------|---|---------|
| Multicast | Set the host IP address and port number, they must be separated by a colon (The IP address range is 224.0.0.0 to 239.255.255.255, and the port number is preferably set between 1024 and 65535) | G.711A  |
|           |   | G.711U  |
|           |   | G.729AB |
|           |   | iLBC    |
|           |   | opus    |
|           |   | G.722   |

## ➤ PTT

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

**Function Key Settings >>**

| Key       | Type | Name | Value | Value2 | Subtype    | Line       | Media   |
|-----------|------|------|-------|--------|------------|------------|---------|
| DSS Key 1 | PTT  |      | 123   |        | Speed Dial | 12321@SIP1 | DEFAULT |
| DSS Key 2 | None |      |       |        | None       | AUTO       | DEFAULT |

*Picture 40 - Advanced Setting*

**9.27 Security >> Web filter**

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

**Web Filter Table**

| Start IP Address | End IP Address | Option   |
|------------------|----------------|--|
| 172.16.80.6      | 172.16.80.69   | <input type="button" value="Modify"/><br><input type="button" value="Delete"/> |

*Picture 41- 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.

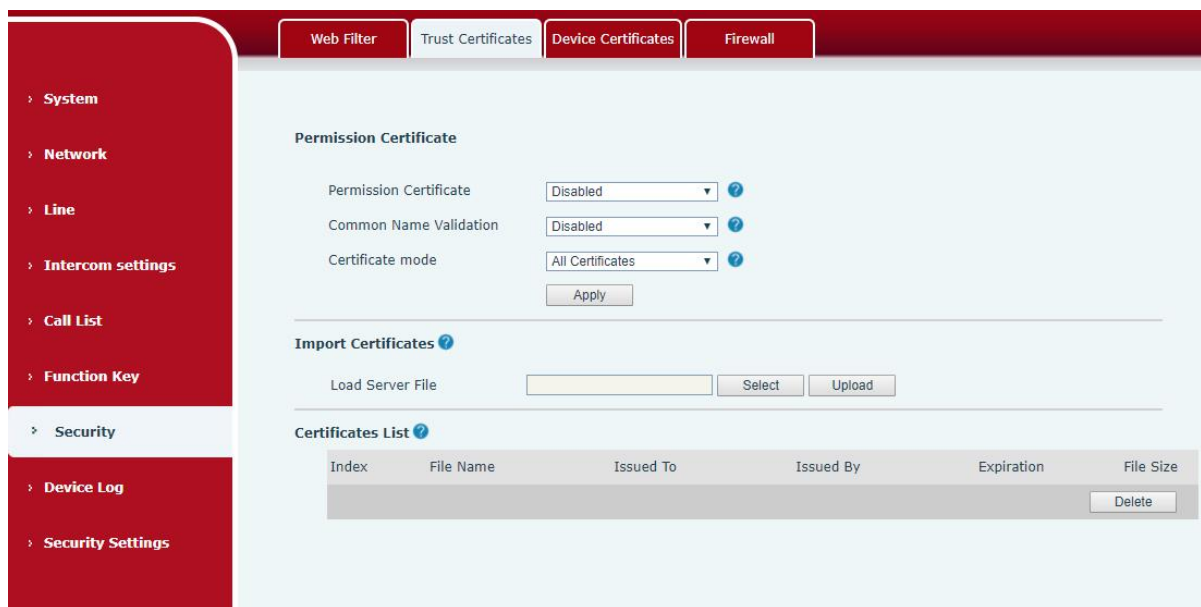
Enable web filtering: configure to enable/disable web access filtering; click the [Submit] button to take effect



**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.

## 9.28 Security >> Trust Certificates

You can upload and delete uploaded trust certificates.

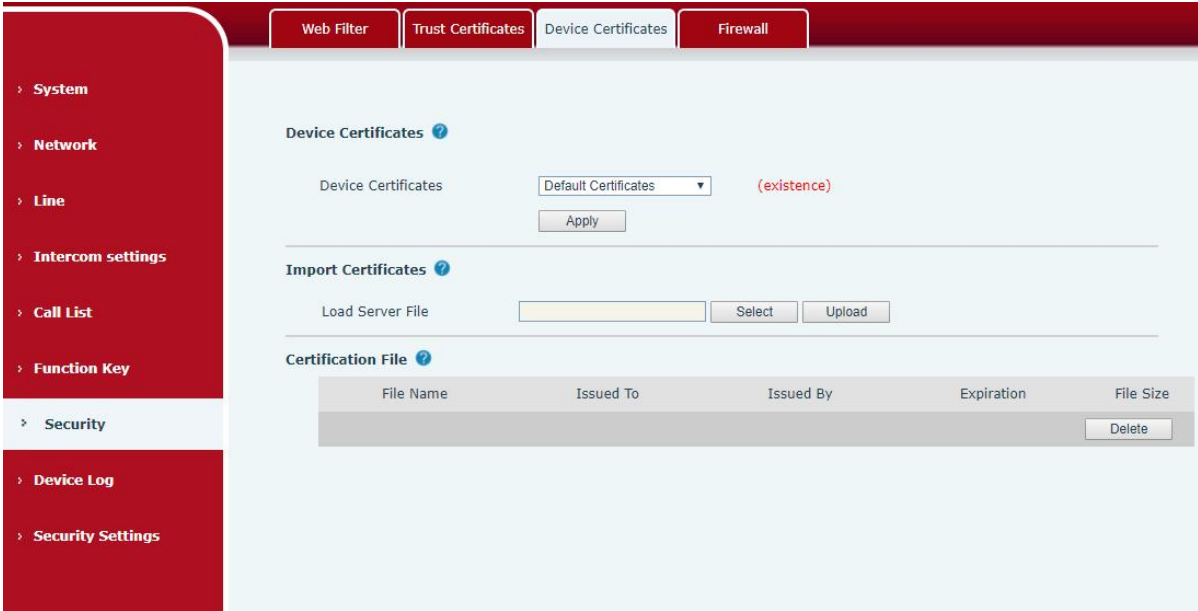


*Picture 42 - Trust Certificates*

## 9.29 Security >> Device Certificates

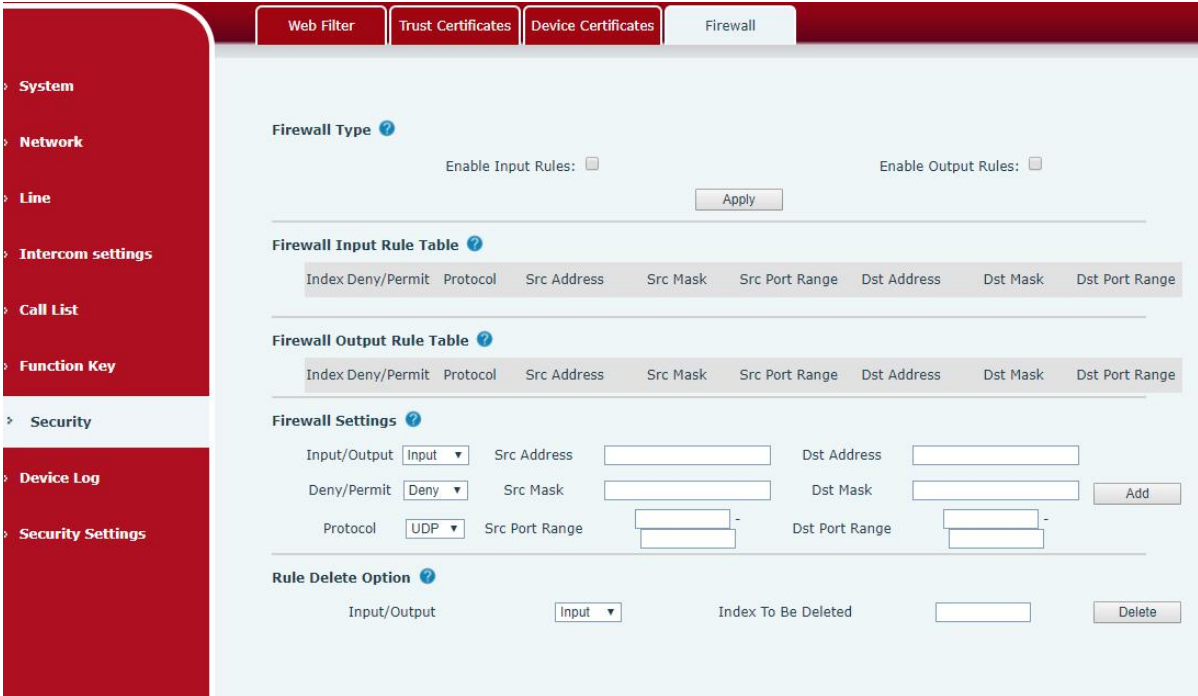
Select the default certificate or the custom certificate as the device certificate.

You can upload and delete uploaded certificates.



Picture 43- Device Certificates

### 9.30 Security >> Firewall



Picture 44 - 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.

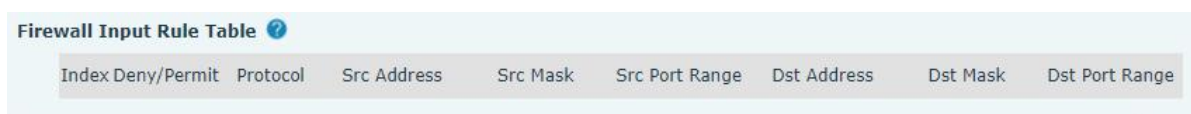
The firewall rule setting is a simple firewall module. This function supports two kinds of rules: input rules and output rules. Each rule will be assigned a serial number, and a maximum of 10 each rule can be set.

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

*Table 26- Web Firewall*

| 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。  |
| Port range          | Port range  |
| Src Address         | The source address can be the host address, network address, or all addresses 0.0.0.0; it can also be a network address similar to *.*.*.0, such as 192.168.1.0.  |
| Dst Mask            | The destination address can be a specific IP address or all 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;    |
| Dst Port Range      | It is the destination 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 255.255.255.0 type, it means that a network segment is filtered; |

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

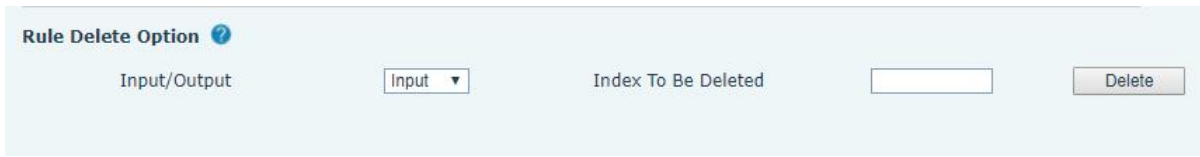


*Picture 45- Firewall rules list*

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.



**Picture 46- Delete firewall rules**

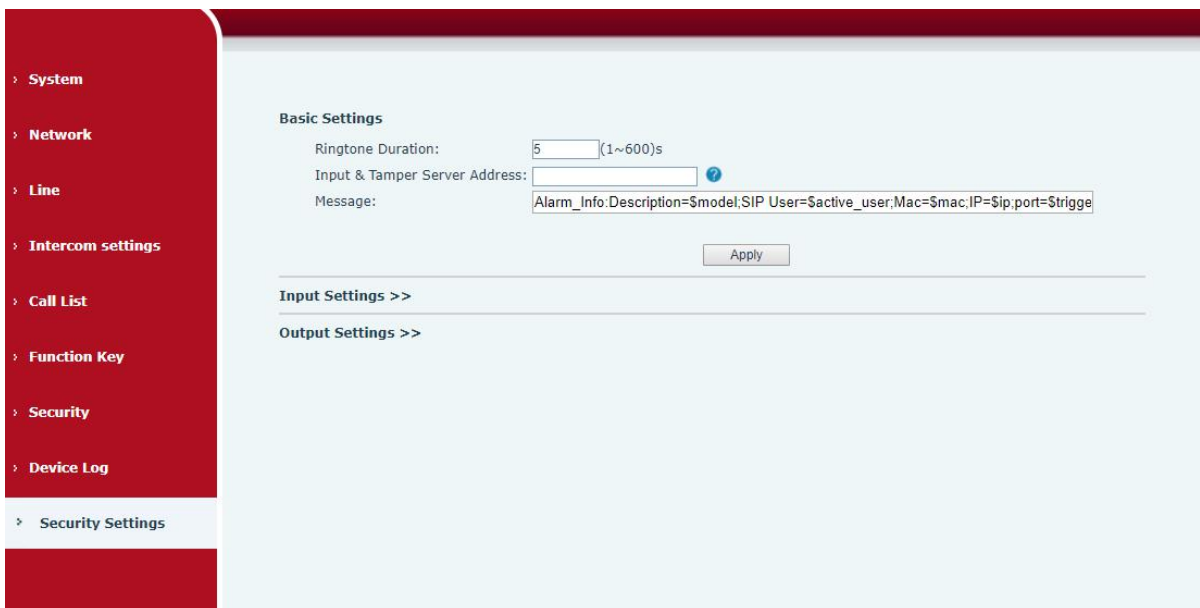
Select the list you want to delete and click [Delete] to delete the selected list.

### 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](#).

### 9.32 Security 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.



**Picture 47 - Security Settings**

**Table 27- Security Settings**

| Security settings                |   |
|----------------------------------|---|
| Parameters                       | Description   |
| Basic settings                   |   |
| Ringtone<br>Duration             | The Alarm ring duration   |
| Input & Tamper<br>Server Address | Configure remote response server address (including remote response server address and trigger alarm server address)  |
| Message                          | When the input port is triggered, a short message will be sent to the server. The message format is as follows : Alarm_Info:Description=\$model;SIP User=\$active_user;Mac=\$mac;IP=\$ip;port=\$trigger                             |
| Enable Tamper<br>Alarm           | If the terminal is forcibly removed, the tamper will be triggered and the set alarm ring will be played all the time  |
| Alarm<br>command                 | When the alarm is triggered, the server sends the command immediately   |
| Reset<br>command                 | If the alarm ring needs to be stopped, the remote end can send a short message to the terminal. The content of the short message is the value set in the reset command. At this time, the terminal will stop playing the alarm bell |
| Reset Alarm<br>Status            | Reset to stop the playing of the bell   |
| Alarm Ringtone                   | The ringtone of alarm   |

## 10 Trouble Shooting

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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 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

The user can restart the device through the webpage, click **[System]** >> **[Tools]** >> **[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 configuration, database and configuration files on the device and the device will be restored to the factory default state.

To restore the factory settings, you need to log in to the webpage **[System]** >> **[Configuration]**, and click **[Reset]** button, the device will return to the factory default state.

Users can press the Reset button on the device and long press 6S to restore factory Settings.

### 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 Technical Support mailbox.

## 10.5 Get 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

**Table 25 - Trouble Cases**

| Trouble Case                                    | Solution   |
|---|--|
| Device could not boot up                        | <ol style="list-style-type: none"><li data-bbox="552 835 1406 1014">1. 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.</li><li data-bbox="552 1025 1406 1193">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.</li></ol>   |
| Device could not register to a service provider | <ol style="list-style-type: none"><li data-bbox="552 1216 1406 1440">1. Please check if the device is connected to the network. The network cable must be connected to the  [Network] interface instead of the  [Camera] interface.</li><li data-bbox="552 1451 1406 1664">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 Support Email to help analyze the issue.</li></ol> |