PA3
User Manual

Software Version: 1.0.0
Release Date：2020/10/13

Directory

1 Picture

2 Table

3 Safety Instruction

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

- Please use the external power supply that is included in the package. Other power supply may cause damage to the phone and affect the behavior or induce noise.
- Before using the external power supply in the package, please check the home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it because it may cause fire or electric shock.
- Do not drop, knock or shake the phone. Rough handling can break internal circuit boards.
- This phone is designed for indoor 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

PA3 is a SIP broadcast module specially developed for the needs of industry broadcast users. The media stream transmission adopts the standard IP/RTP/RTSP protocol. It integrates multiple functional interfaces: broadcast and intercom. It can realize audio broadcasting by connecting corresponding peripherals, and one-key call intercom and other practical functions. It can adapt to multiple use environments and facilitate rapid device deployment. And the device size is small, suitable for DIY applications of various integrated solutions.

5 Install Guide

5.1 Use POE or external Power Adapter

PA3, 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, PA3 will get power supply from POE switch in priority, and change to external power adapter once the POE power supply fails.

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

5.2 Appendix

5.2.1 Common command modes

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; Report IP after successful switching

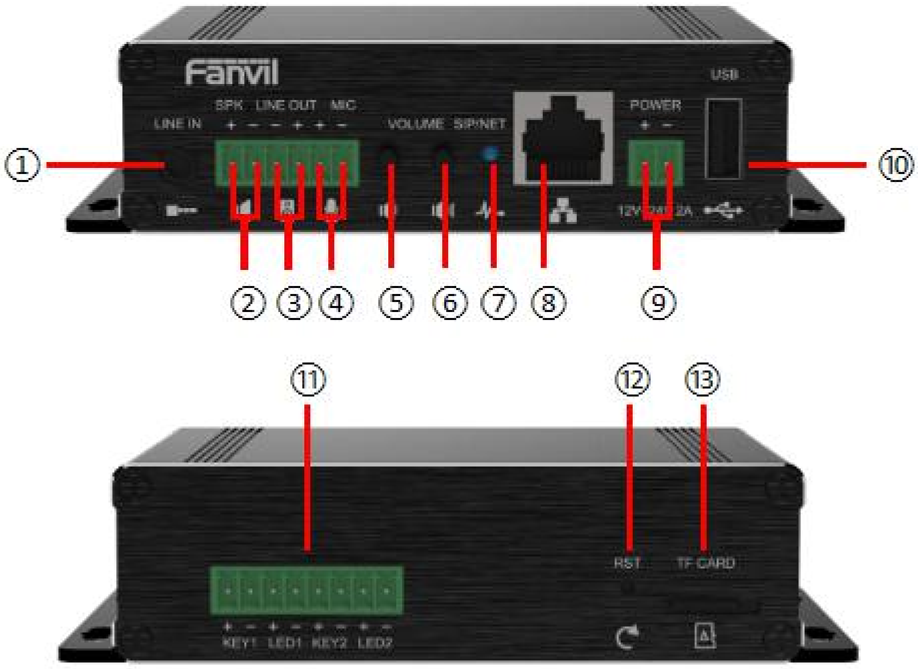
5.2.2 Function key LED status

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	Line in interface	Audio signal input, used to connect external audio input.
②	Speaker interface	Output the maximum power adaptively according to the input voltage of the equipment; 4Ω speakers, POE/10W, 12V/10W, 18V/20W, 24V/30W; Power is related to power supply voltage. The larger the speaker impedance, the smaller the output power. The recommended wire diameter: 18AWG or larger.
③	Line out interface	The audio signal output impedance is 600 Ω, and the single-ended output voltage is 2.54Vpp. Used for external headphones or powered speakers.
④	Microphone interface	It is recommended to use an electret condenser microphone with an impedance of 2.2K Ohm. Sensitivity: -38dB, bias

		voltage 2.2V. It is recommended to use a shielded cable for the microphone signal line. Note: The shielding layer cannot be connected to any ground。
⑤	Volume down	Adjust the ringtone volume/call volume/broadcast volume; Long press the volume down button to report the IP address.
⑥	Volume up	Adjust the ringtone volume/call volume/broadcast volume.
⑦	Network/Registration Indicator	Indicate network status, call status, and registration status. Fast flashing: abnormal network or SIP account; Slow flashing: during a call; Steady on: The network is normal or SIP registration is successful.
⑧	Ethernet interface	WAN port, standard RJ45 interface, 10/100M adaptive, support POE input, it is recommended to use Category 5 or Category 5 network cable.
⑨	Power input interface	12V～24V 2A input, the maximum power output of the power amplifier is determined according to the input voltage
⑩	USB interface	Connect USB peripherals, such as U disk, USB adapter, etc.
⑪	One-key call interface	Connect the speed dial button (with light), you can make a call by pressing the button. It sets the calling number or IP address by logging in to the web page.
⑫	reset	Long press for 6 seconds and the indicator light flashes, the device restarts and restores factory settings.
⑬	TF card interface	TF card slot, used to store local audio files or records.

6.2 Installation instructions

6.2.1 Installation

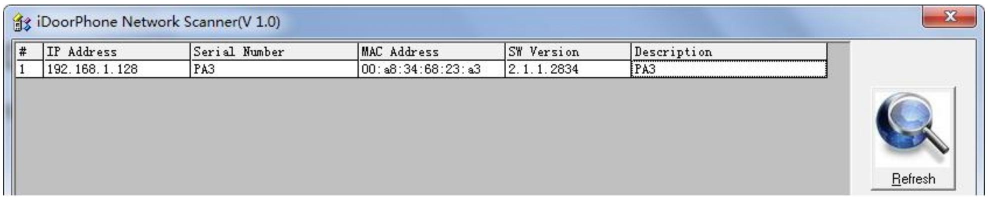
Step 1: Fix the equipment at the installation position with metal strips (provided by the user).
Step 2: Connect peripherals such as one-key call buttons, speakers, and microphones to the corresponding wiring terminals according to the interface definition, and then insert the corresponding interfaces in turn.
Step 3: Plug in the internet cable and power supply, the device indicator flashes to indicate that the power connection is normal.

6.2.2 Device IP address

Method one：

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

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



Method two:

Connect the speaker and press and hold the volume down button for 3 seconds (30 seconds after power-on), the device will automatically announce the IP address of the machine.

Method three:

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

Table 4 - Configuration instructions

Default configuration			
DHCP mode	**Default enable**	**Static IP**	**192.168.1.128**
Voice read IP address	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

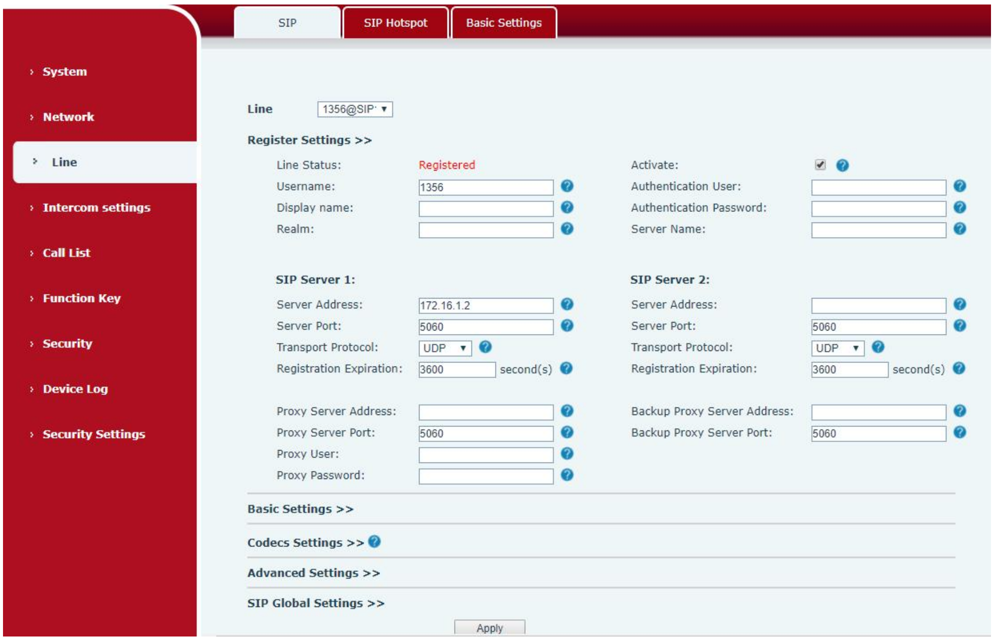
The username and password should be correct to log in to the web page. **The default username and password are "admin"**. For the specific details of the operation of the web page, please refer to 9 Web Configurations

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

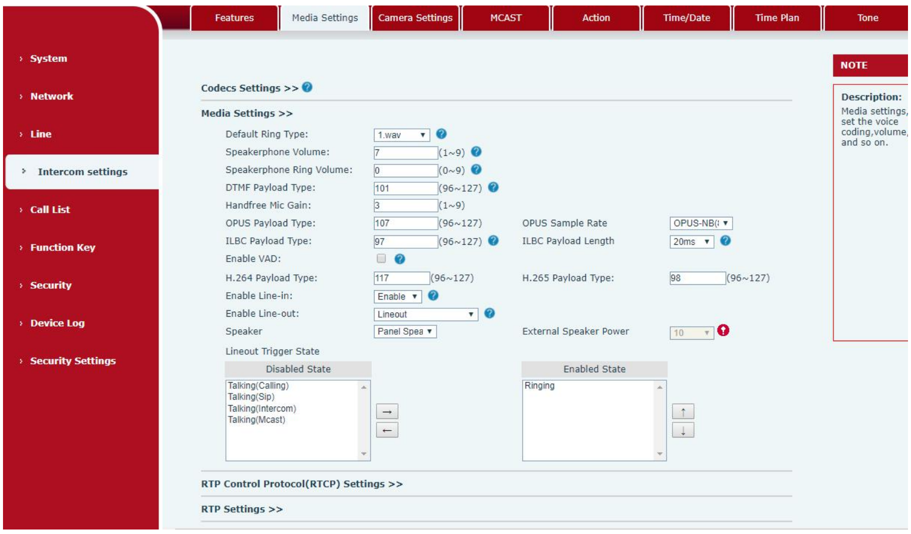
6.5 Volume setting

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

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

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

Hands-free microphone gain: microphone volume level.



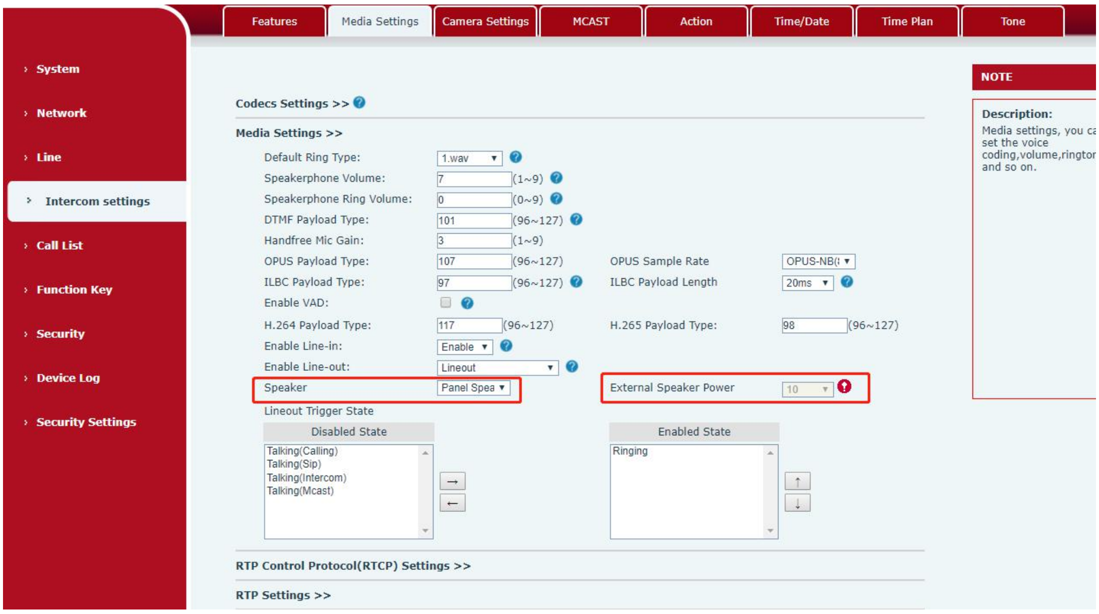
Picture 4- Volume Set

6.6 Set the player type

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

[Intercom 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:

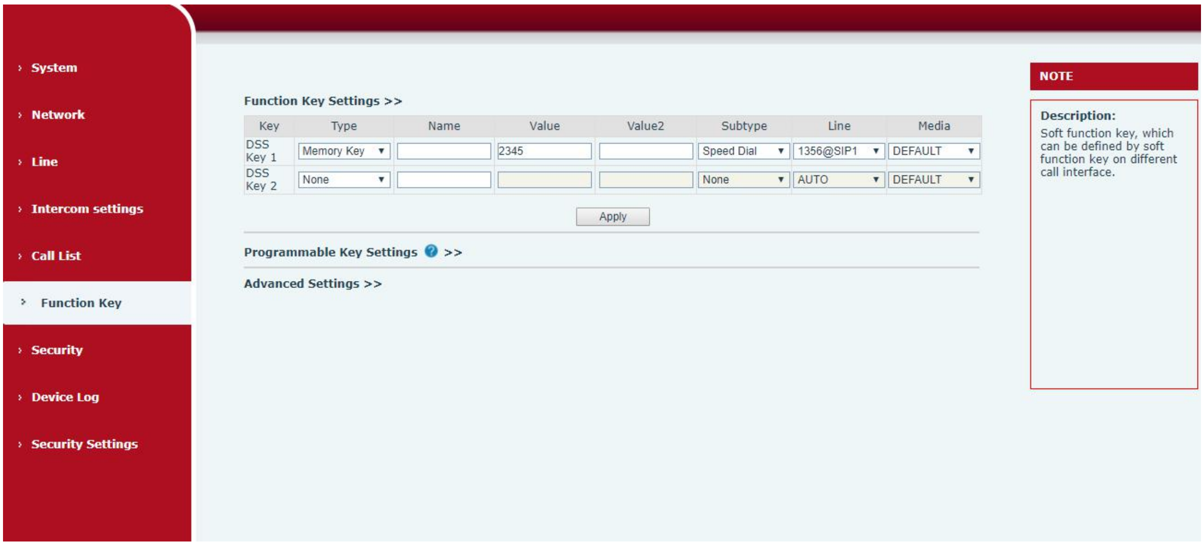
Table 5- Power Supply

	Output Power	Speaker type
POE	10W	10W/4 Ω
12V/2A DC	10W	10W/4Ω
18V/2A DC	20W	20W/4Ω
24V/2A DC	30W	30W/4Ω

7 Basic Function

7.1 Making Calls

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



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

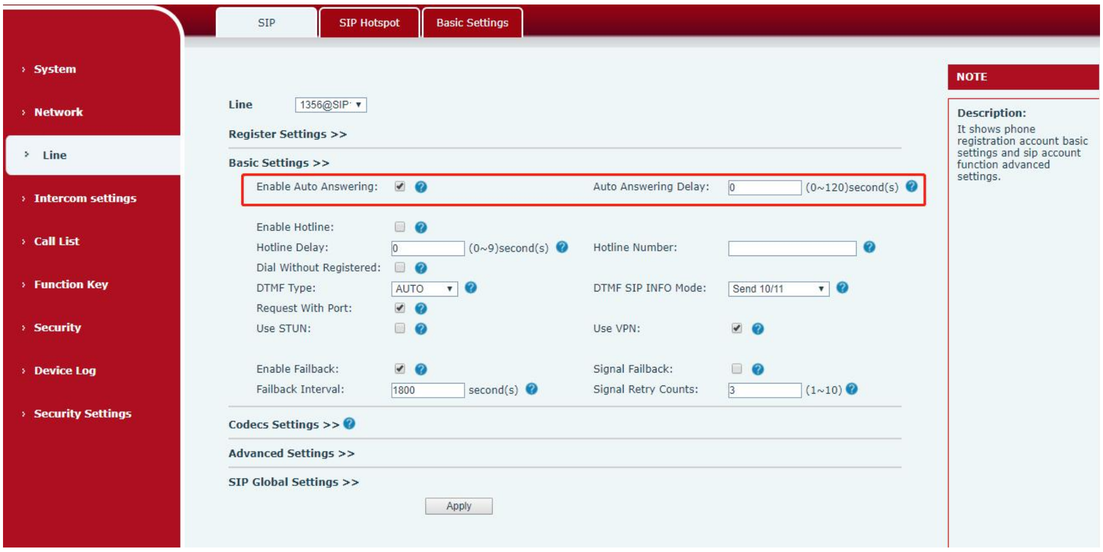
You can hang up the call through the Release key (you can set the function key as the Release key) or turn on the speed dial button to hang up the call. See detailed configuration instructions 9.26 Function Key.

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:

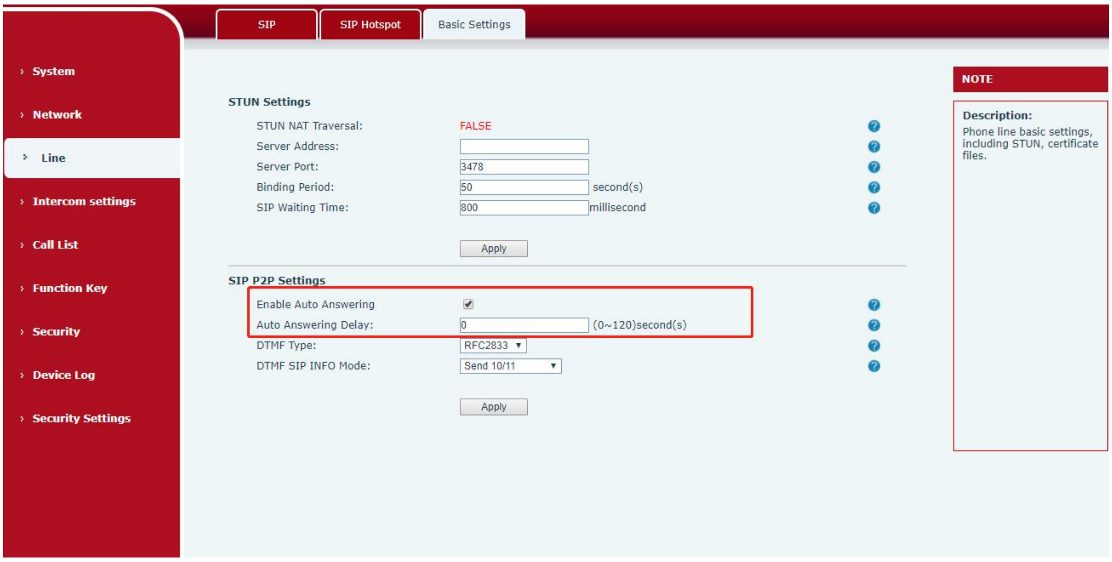
Enter [**Line**] >> [**SIP**], Enable auto answer and set auto answer time and click submit.



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）

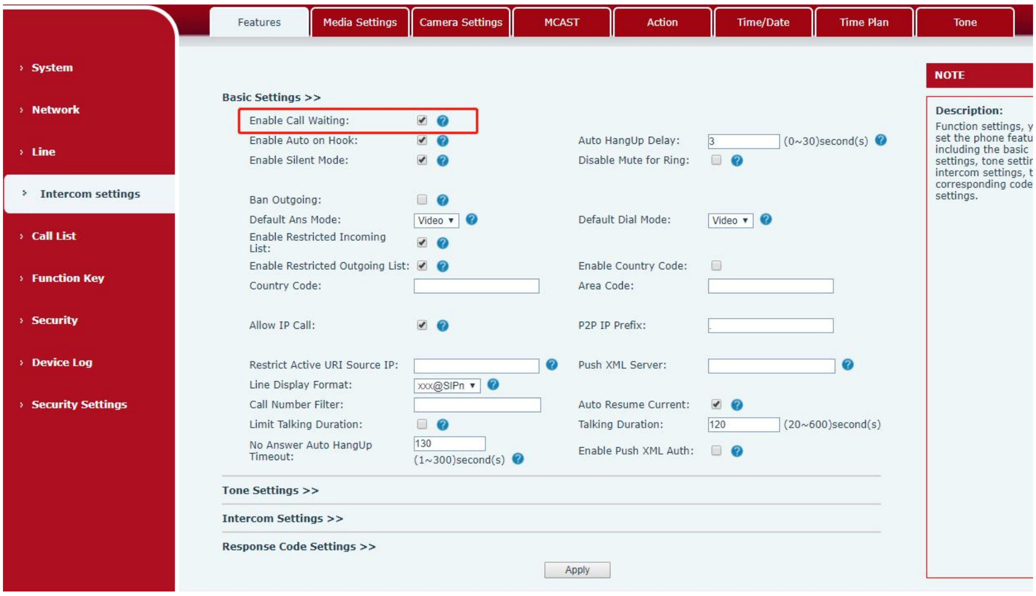
The range can be set to 0~120s, and the call will be answered automatically when the timeout is set.

7.5 Call Waiting

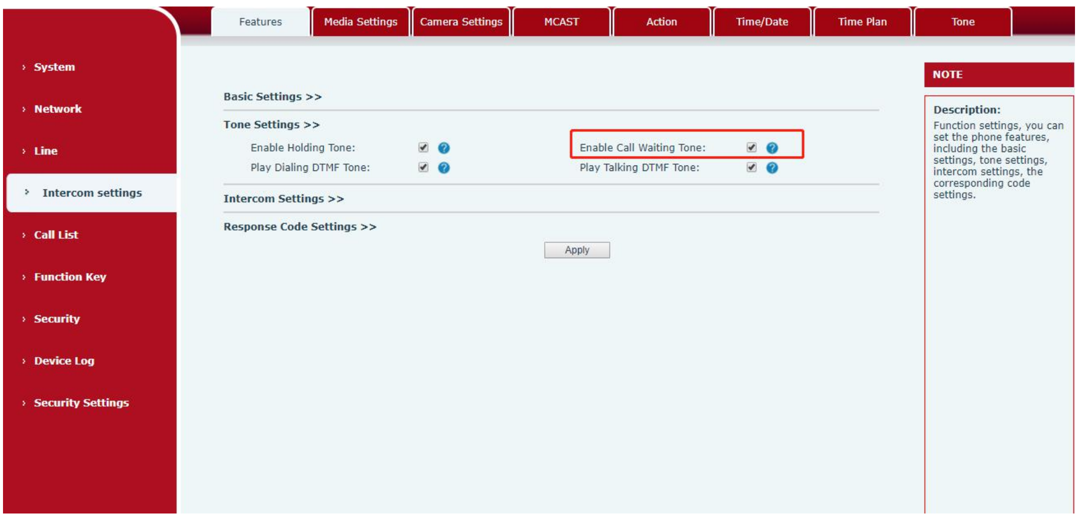
- Enable call waiting: new calls can be accepted during a call.
- Disable call waiting: new calls will be automatically rejected and a busy signal will be prompted
- Enable call waiting tone: when you receive a new call on the line, the device will beep.

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

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



Picture 9 - Call Waiting

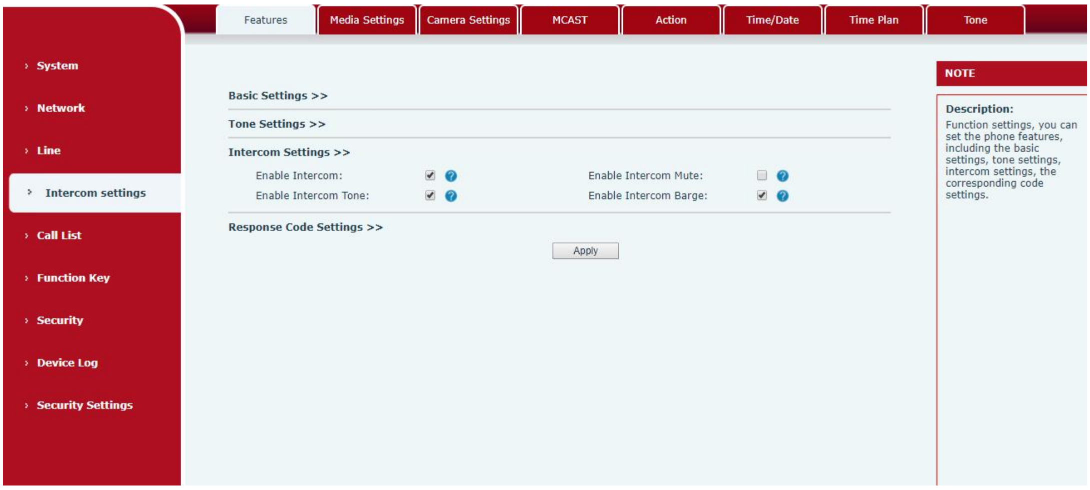


Picture 10 - Call Waiting tone

8 Advance Function

8.1 Intercom

The equipment can answer intercom calls automatically.



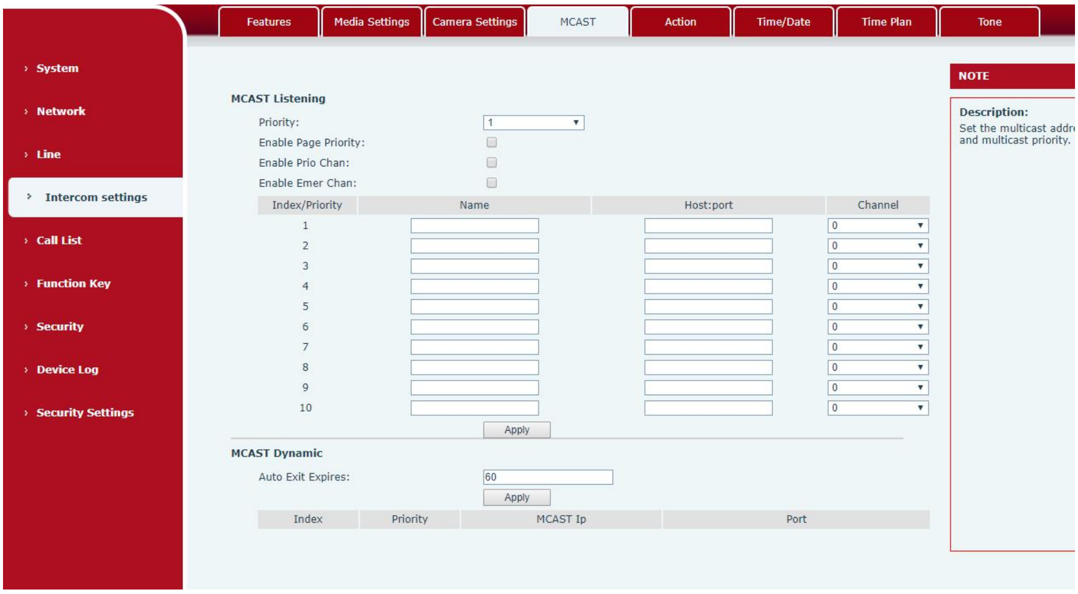
Picture 11 - WEB Intercom

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 Command automatic call
Enable Intercom Barge	If the option is enabled, PA3 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 [**Intercom Settings**] >> [**MCAST**].
- Press the DSSKey of Multicast Key which you set.
- Receive end will receive multicast call and play multicast automatically.

MCAST Dynamic:

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

8.3 Hotspot

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

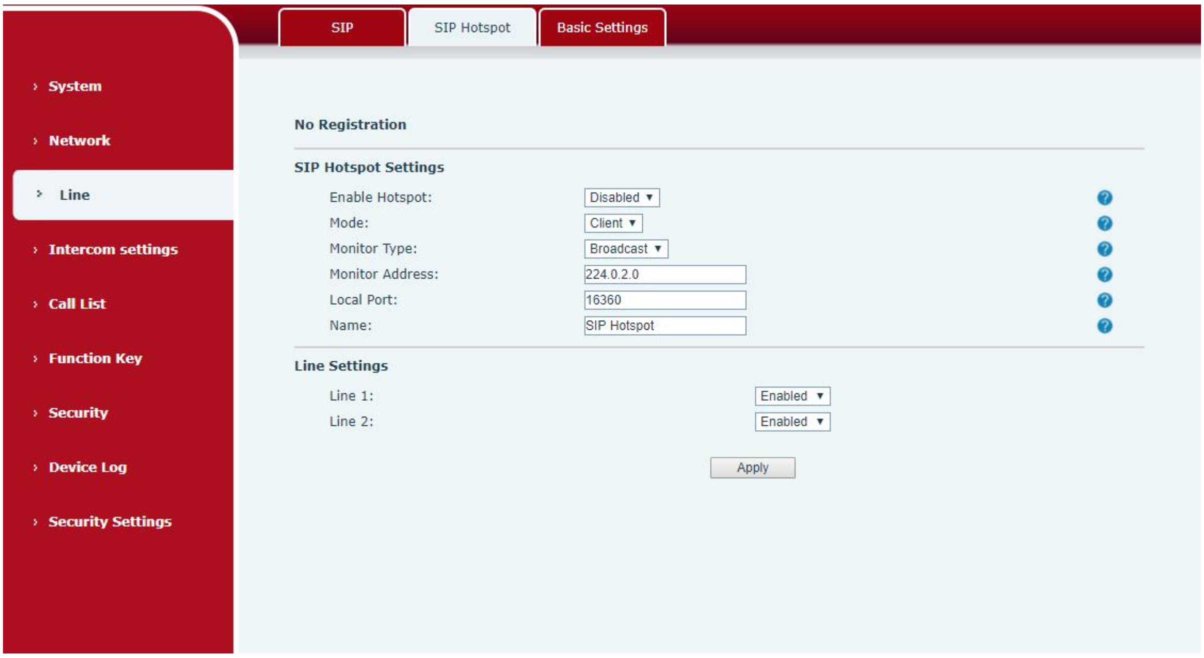
Table 8 - SIP Hotspot

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

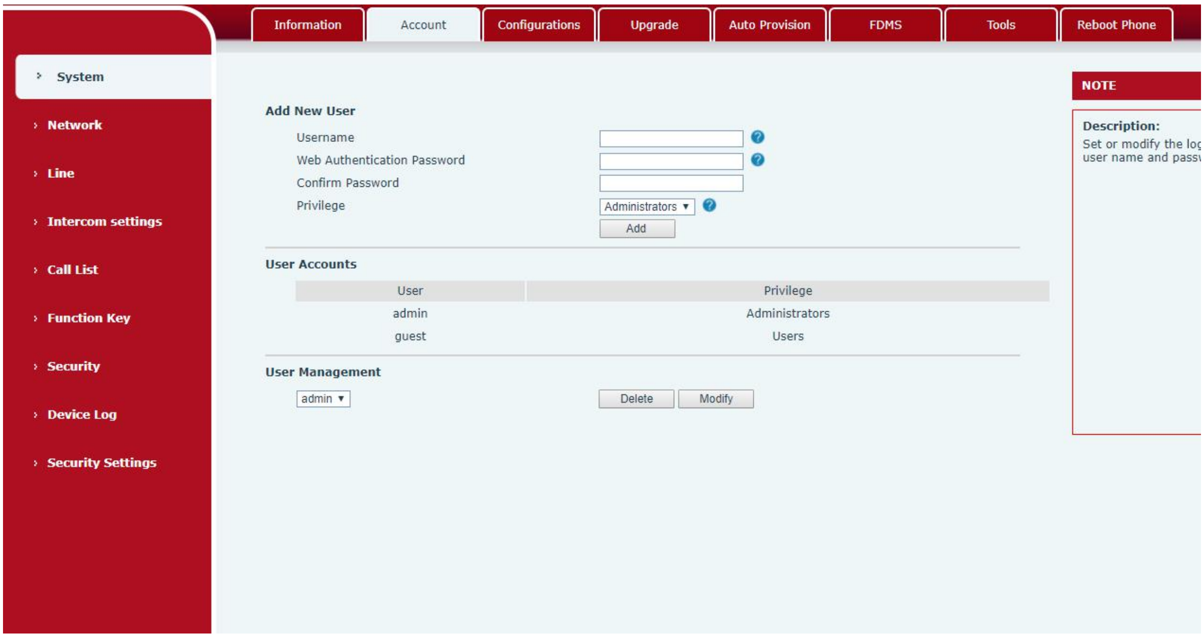
And summarization of network status,

- 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



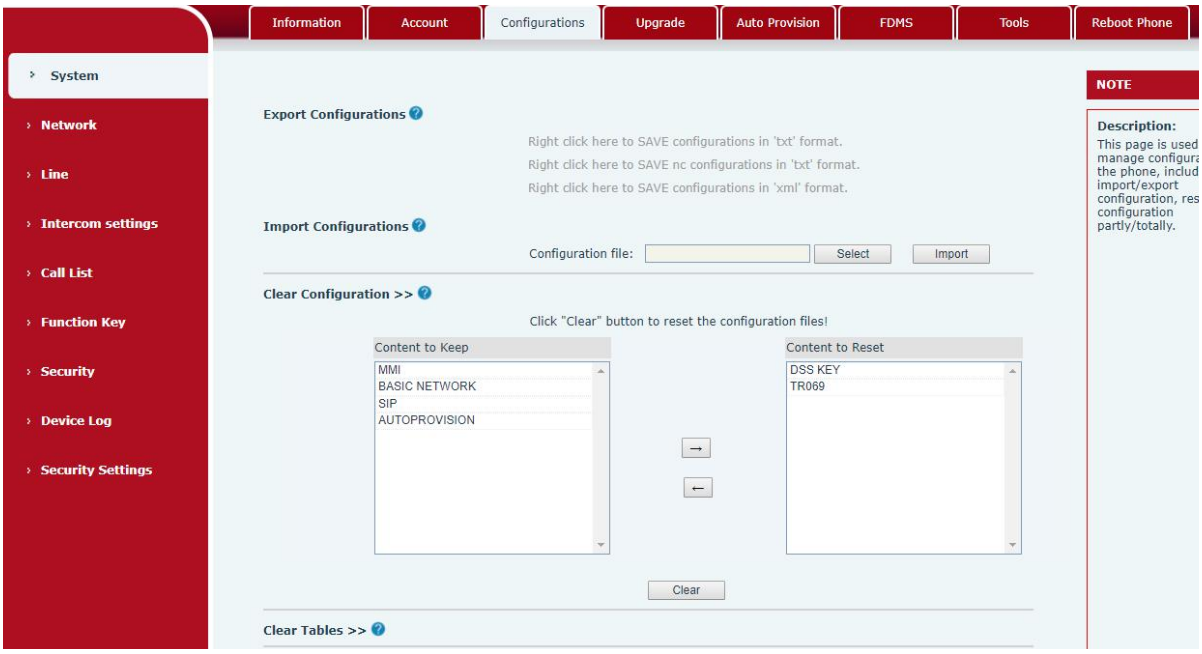
Picture 14- WEB Account

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

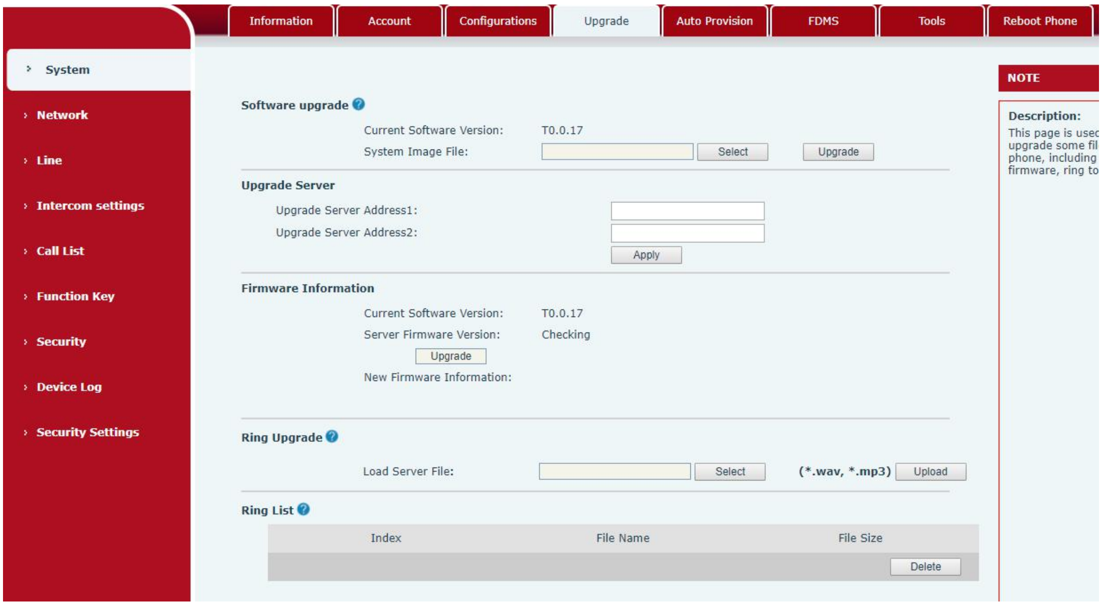
- **Clear Tables**

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



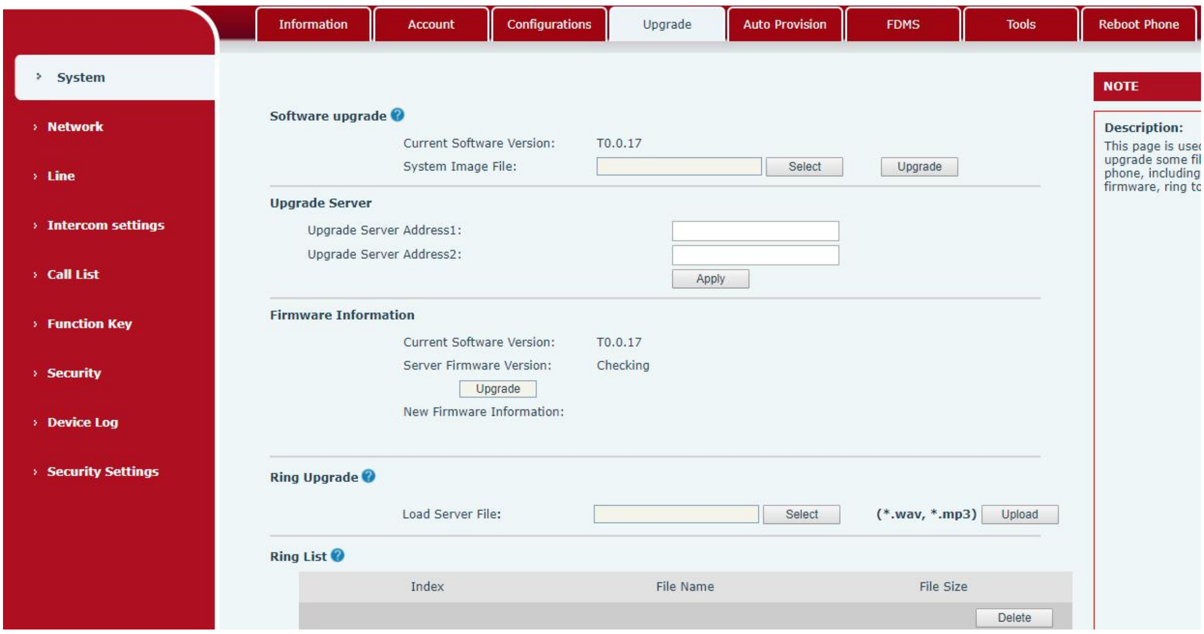
Picture 16- Upgrade

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
Upgrade server	
Enable Auto Upgrade	Enable automatic upgrade, If there is a new version txt and new software firmware on the server, phone will show a prompt upgrade message after Update Interval.
Upgrade Server Address1	Set available upgrade server address.
Upgrade Server Address2	Set available upgrade server address.
Update Interval	Set Update Interval.
Firmware Information	
Current Software Version	It will show Current Software Version.
Server Firmware Version	It will show Server Firmware Version.
[Upgrade] button	If there is a new version txt and new software firmware on the server, the page will display version information and upgrade button will become available; Click [Upgrade] button to upgrade the new firmware.
New version description information	When there is a corresponding TXT file and version on the server side, the TXT and version information will be displayed under the new version description information.

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

- 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

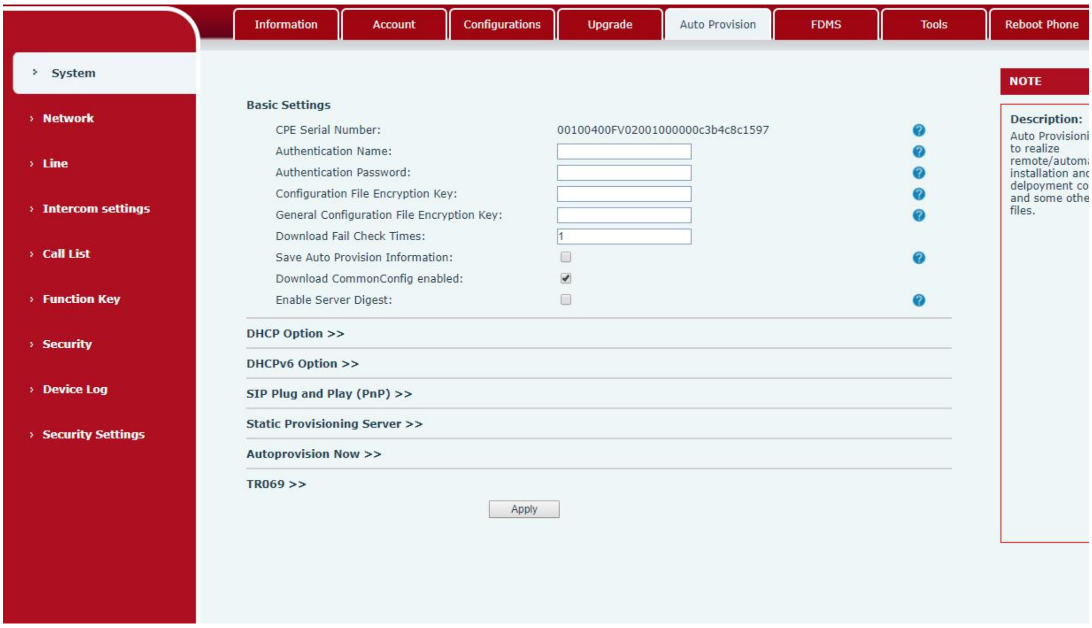
Xxxxx

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

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

PNP>DHCP>TR069> Static Provisioning

Transferring protocol: FTP、 TFTP、 HTTP、 HTTPS

Details refer to **Fanvil Auto Provision**

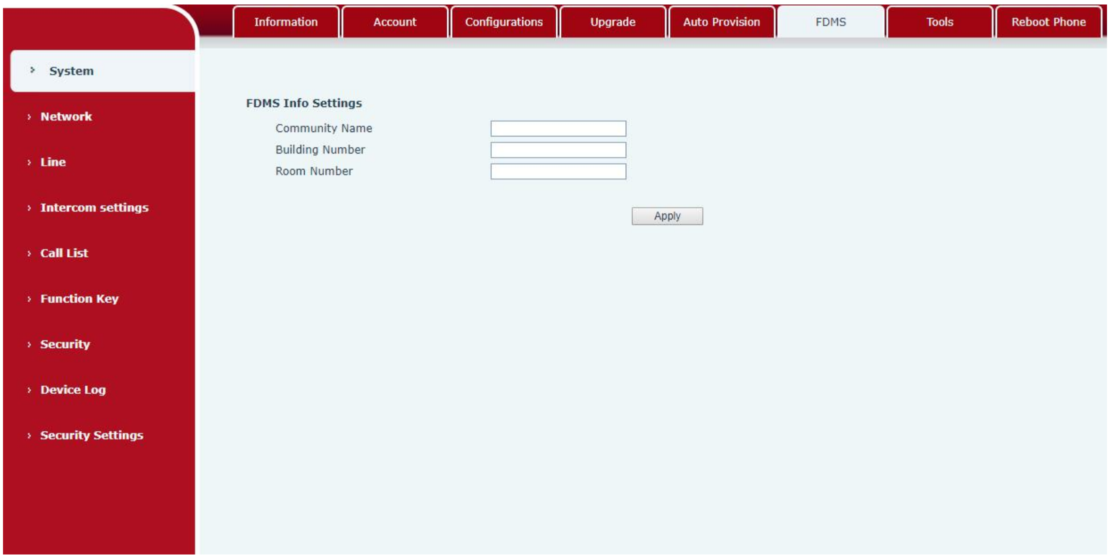
https://www.fanvil.com/Support/download/cid/14.html

Table 10- Auto Provision

Auto provision	
Parameters	**Description**
Basic settings	
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.
DHCP Option	
Option Value	The equipment supports configuration from Option 43, Option 66, or a Custom DHCP option. It may also be disabled.
Custom Option Value	Custom option number. Must be from 128 to 254.
Enable DHCP Option 120	Set the SIP server address through DHCP option 120.
SIP Plug and Play (PnP)	
Enable SIP PnP	Whether enable PnP or not. If PnP is enable, phone will send a SIP SUBSCRIBE message with broadcast method. Any server can

	support the feature will respond and send a Notify with URL to phone. Phone could get the configuration file with the URL.
Server Address	Broadcast address. As default, it is 224.0.0.0.
Server Port	PnP port
Transport Protocol	PnP protocol, TCP or UDP.
Update Interval	PnP message interval.
Static Provisioning Server	
Server Address	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. 1. Disabled. 2. Update after reboot. 3. Update after interval.
TR069	
Enable TR069	Enable TR069 after selection
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.
ACS Server Type	There are 2 options Serve type, common and CTC.
ACS Server URL	ACS server address
ACS User	ACS server username (up to is 59 character)
ACS Password	ACS server password (up to is 59 character)
STUN server address	Enter the STUN address
Enable the STUN	Enable the STUN
TLS Version	TLS Version

9.7 System >> FDMS



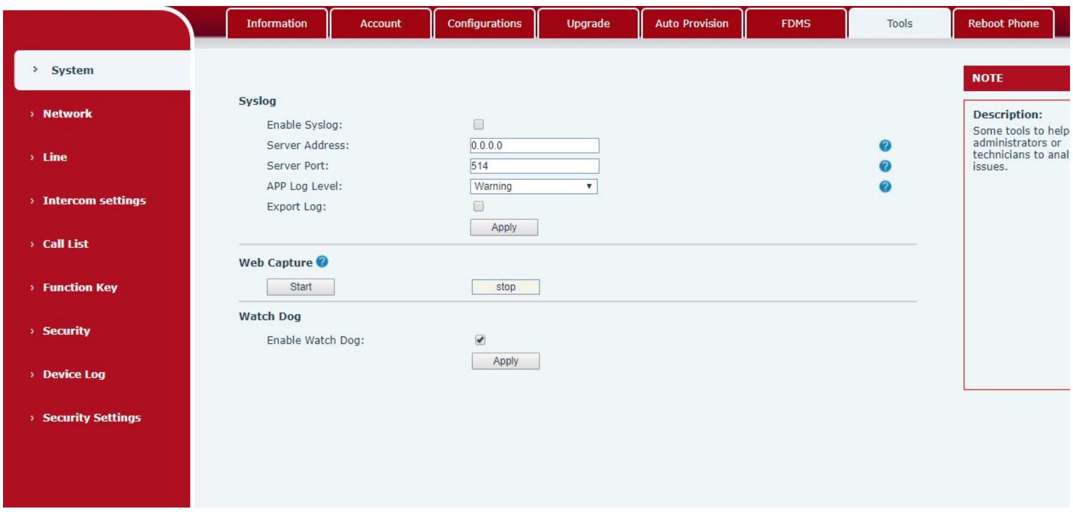
Picture 19 - FDMS

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.

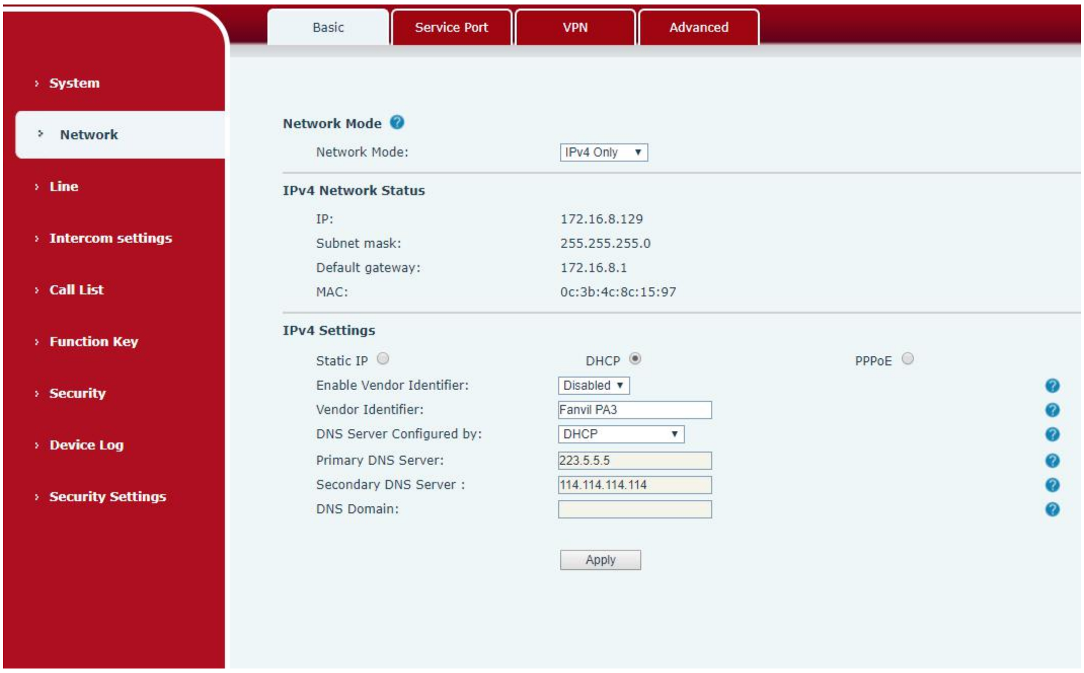


Picture 20 - Tools

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

9.9 Network >> Basic

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



Picture 21 - Network Basic Setting

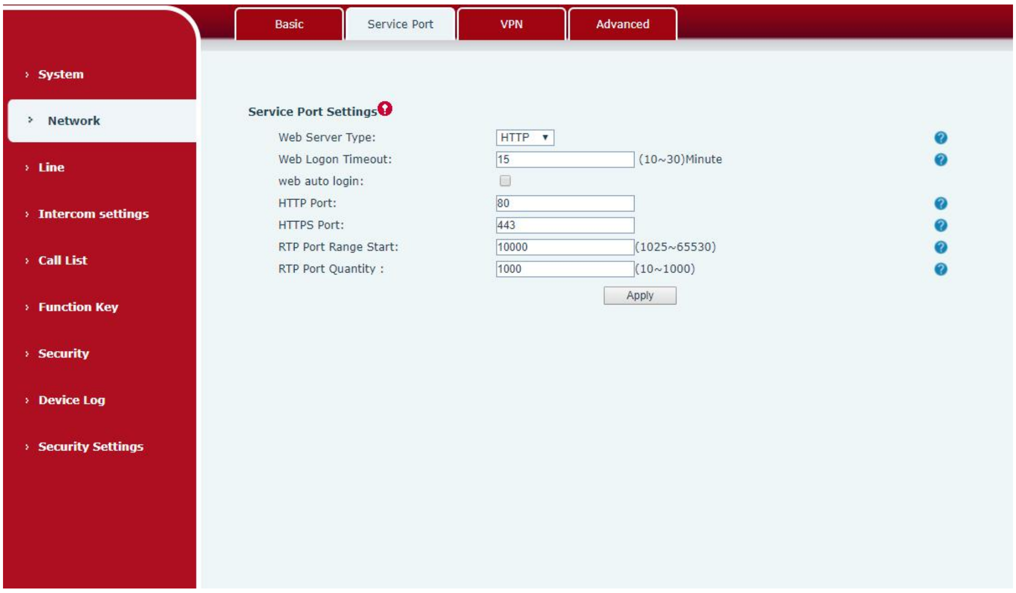
Table 12 - Network Basic Setting

Field Name	Explanation
Network Status	
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
Settings	
Select the appropriate network mode. The equipment supports three network modes:	

Static IP	Network parameters must be entered manually and will not change. All parameters are provided by the ISP.
DHCP	Network parameters are provided automatically by a DHCP server.
PPPoE	Account and Password must be input manually. These are provided by your ISP.
If Static IP is 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.
attention： 1） After setting the parameters, click【Apply】to take effect. 2）If you change the IP address, the webpage will no longer responds, please enter the new IP address in web browser to access the device. 3） If the system USES DHCP to obtain IP when device boots up, and the network address of the DHCP Server is the same as the network address of the system LAN, then after the system obtains the DHCP IP, it will add 1 to the last bit of the network address of LAN and modify the IP address segment of the DHCP Server of LAN. If the DHCP access is reconnected to the WAN after the system is started, and the network address assigned by the DHCP server is the same as that of the LAN, then the WAN will not be able to obtain IP access to the network	

9.10Network >> service port

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

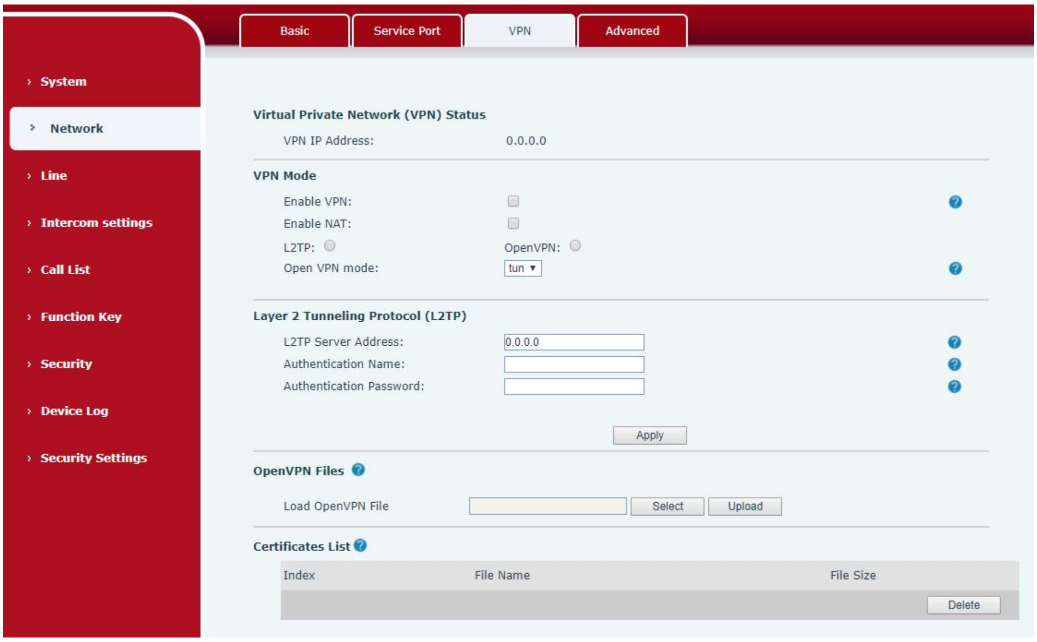


Picture 22- Service port setting interface

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 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



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**

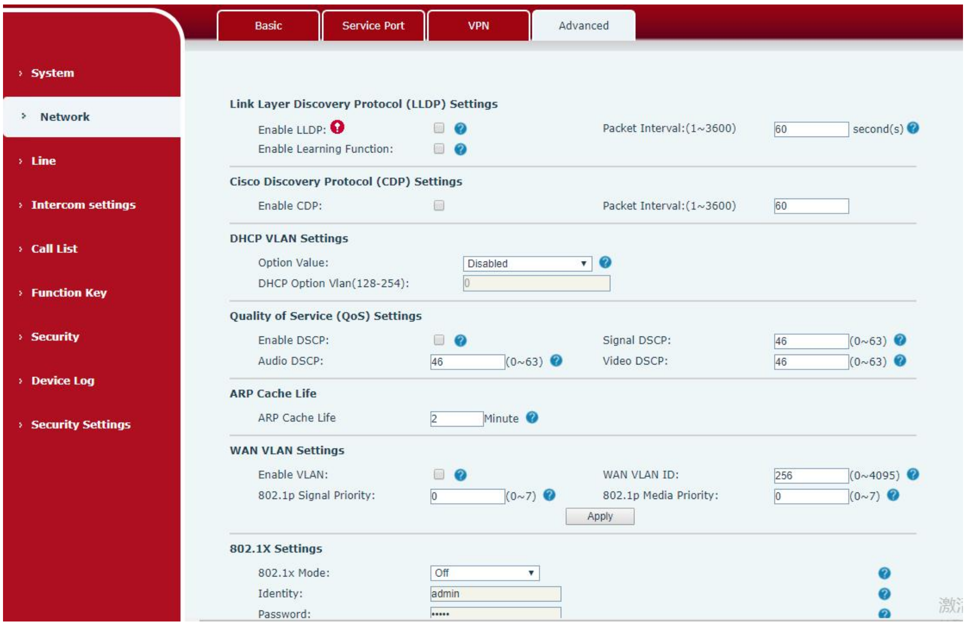
To establish an OpenVPN connection, user should get the following authentication and configuration files from the OpenVPN hosting provider and name them as the following,

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



Picture 24 - Network Setting

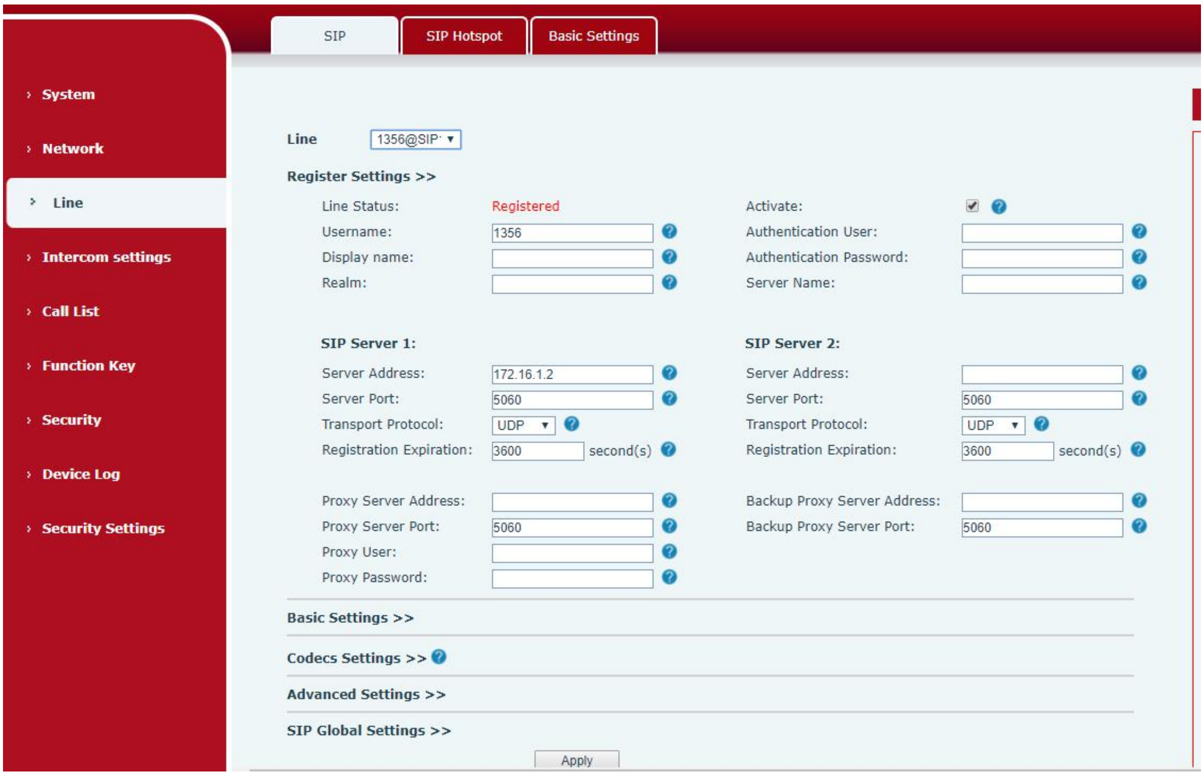
Network advanced Settings are typically configured by IT administrators to improve the quality of device service.

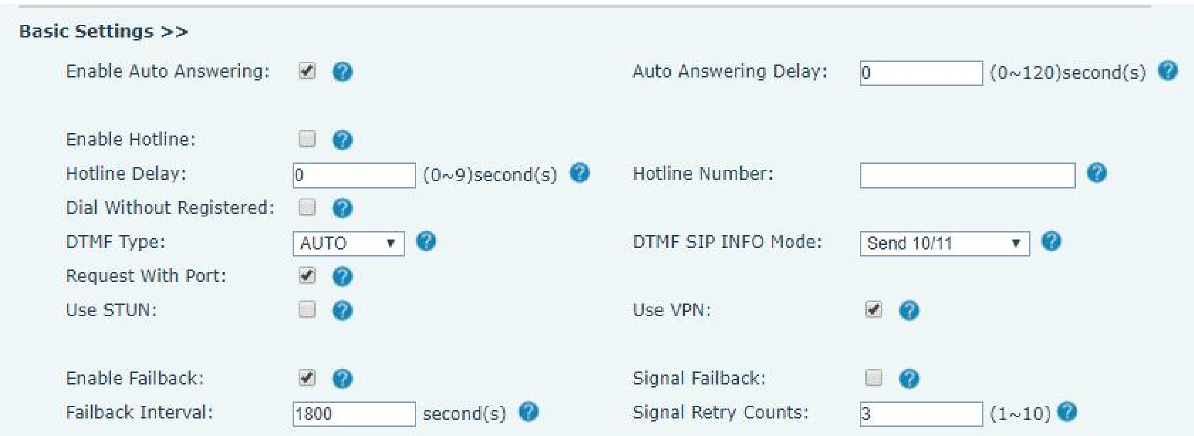
Table 14- Network Setting

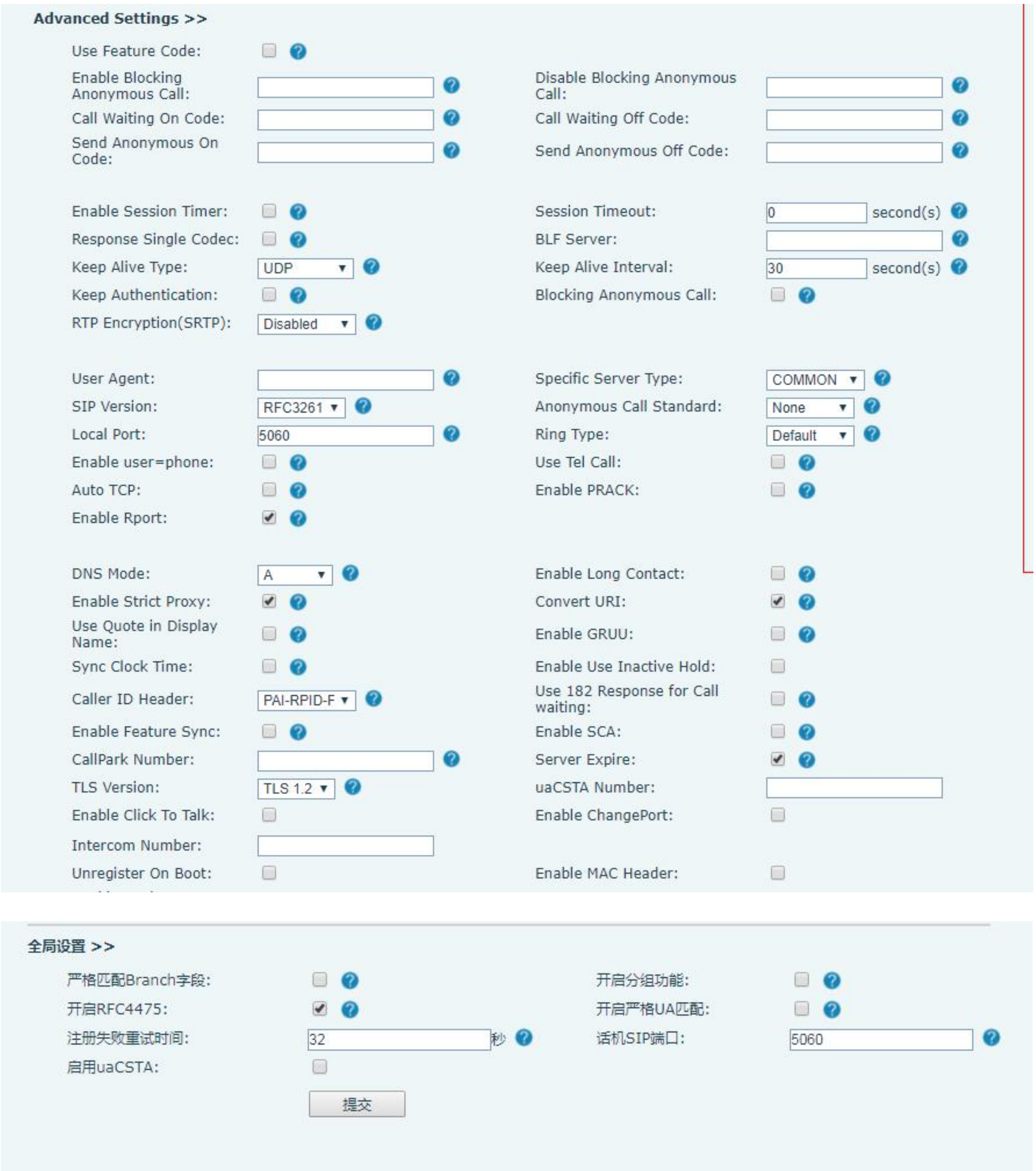
Field Name	Explanation
LLDP Settings	
Enable LLDP	Enable or disable LLDP
Packet Interval	LLDP Send detection cycle
Enable Learning Function	Learn the discovered device information on the device
QoS Settings	
Pattern	Voice quality assurance (off by default)
DHCP VLAN Settings	
parameters values	128-254，Obtain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

Password	Confirm Password

9.13LINES >> SIP







Picture 25- SIP

Table 15 - SIP

Parameters	Description
Register Settings	
Line Status	Display the current line status at page loading. To get the up to date line status, user has to refresh the page manually.
Server Address	Enter the IP or FQDN address of the SIP server

Server Port	Enter the SIP server port, default is 5060
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
Basic Settings	
Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
Call Forward Unconditional	Enable unconditional call forward, all incoming calls will be forwarded to the number specified in the next field
Call Forward Number for Unconditional	Set the number of unconditional call forward
Call Forward on Busy	Enable call forward on busy, when the phone is busy, any incoming call will be forwarded to the number specified in the next field
Call Forward Number for Busy	Set the number of call forward on busy
Call Forward on No Answer	Enable call forward on no answer, when an incoming call is not answered within the configured delay time, the call will be forwarded to the number specified in the next field

Call Forward Number for No Answer	Set the number of call forward on no answer
Call Forward Delay for No Answer	Set the delay time of not answered call before being forwarded
Transfer Timeout	Set the timeout of call transfer process
Conference Type	Set the type of call conference, Local=set up call conference by the device itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server
Server Conference Number	Set the conference room number when conference type is set to be Server
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Voice Message Number	Set the number for retrieving voice message
Voice Message Subscribe Period	Set the interval of voice message notification subscription
Enable Hotline	Enable hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or turn on hands-free speaker or headphone
Hotline Delay	Set the delay for hotline before the system automatically dialed it
Hotline Number	Set the hotline dialing number
Dial Without Registered	Set call out by proxy without registration
Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.
DTMF Type	Set the DTMF type to be used for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically
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
Codec Settings	Set the priority and availability of the codecs by adding or remove them from the list.

Advanced Settings	
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field.
Enable DND	Set the feature code to dial to the server
Disable DND	Set the feature code to dial to the server
Enable Call Forward Unconditional	Set the feature code to dial to the server
Disable Call Forward Unconditional	Set the feature code to dial to the server
Enable Call Forward on Busy	Set the feature code to dial to the server
Disable Call Forward on Busy	Set the feature code to dial to the server
Enable Call Forward on No Answer	Set the feature code to dial to the server
Disable Call Forward on No Answer	Set the feature code to dial to the server
Enable Blocking Anonymous Call	Set the feature code to dial to the server
Disable Blocking Anonymous Call	Set the feature code to dial to the server
Call Waiting On Code	Set the feature code to dial to the server
Call Waiting Off Code	Set the feature code to dial to the server
Send Anonymous On Code	Set the feature code to dial to the server
Send Anonymous Off Code	Set the feature code to dial to the server
SIP Encryption	Enable SIP encryption such that SIP transmission will be encrypted
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, i.e. “Fanvil” vs Fanvil
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
Sync Clock Time	Time Sycn with server
Caller ID Header	Set the Caller ID Header
Use 182 Response for 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. Please enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
Enable Feature Sync	Feature Sycn with server
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
CallPark Number	Set the callPark number
Server Expire	
TLS Version	Choose TLS Version

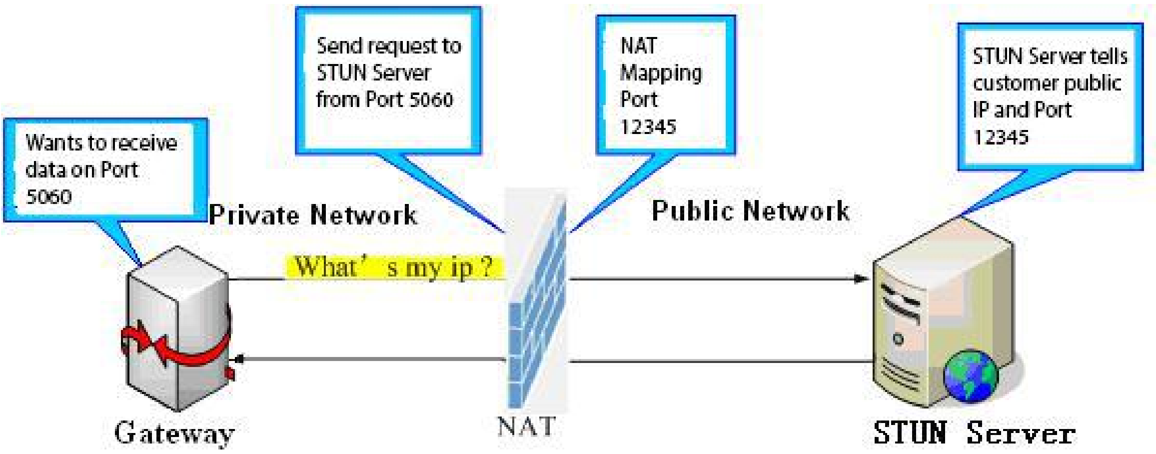
9.14 Line >> SIP Hotspot

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

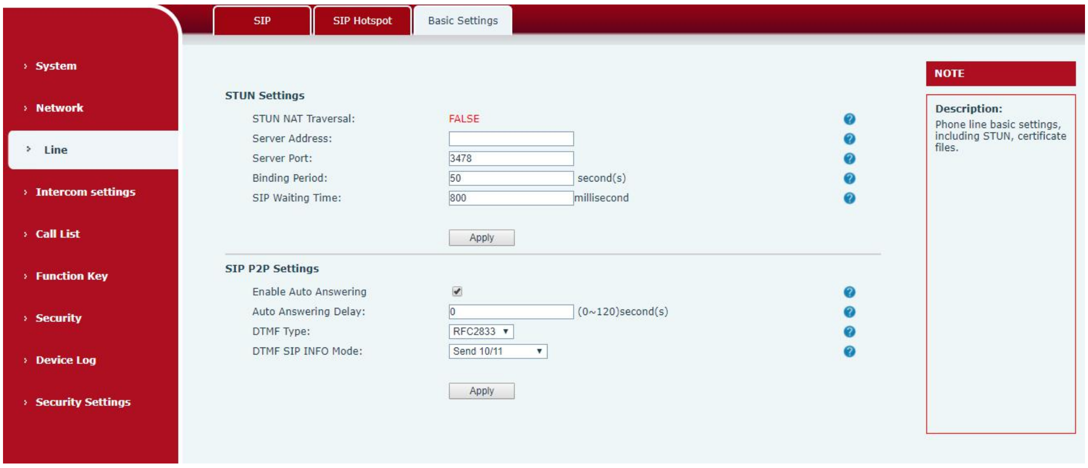
See 8.3 Hotspot for details.

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

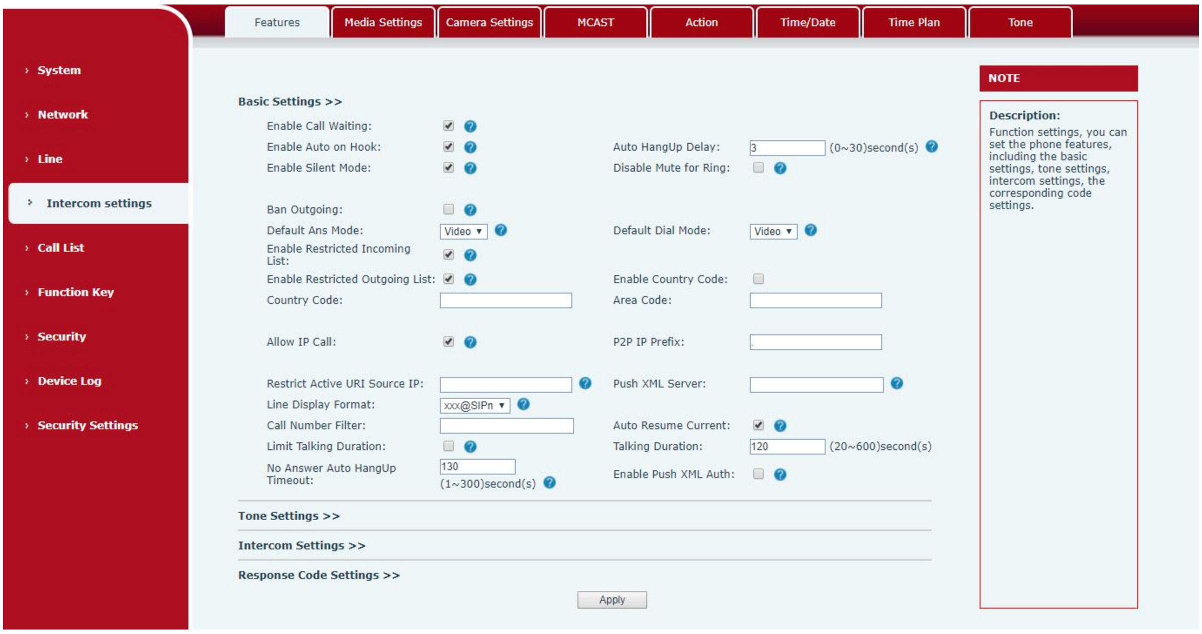


Picture 27 - Line Basic Setting

Table 16- Line Basic Setting

Parameters	Description
STUN Settings	
Server Address	Set the STUN server address
Server Port	Set the STUN server port, default is 3478
Binding Period	Set the STUN binding period which can be used to keep the NAT pinhole opened.
SIP Waiting Time	Set the timeout of STUN binding before sending SIP messages
SIP P2P Settings	
Enable Auto Answering	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 Intercom settings >> Features



Picture 28 - Feature

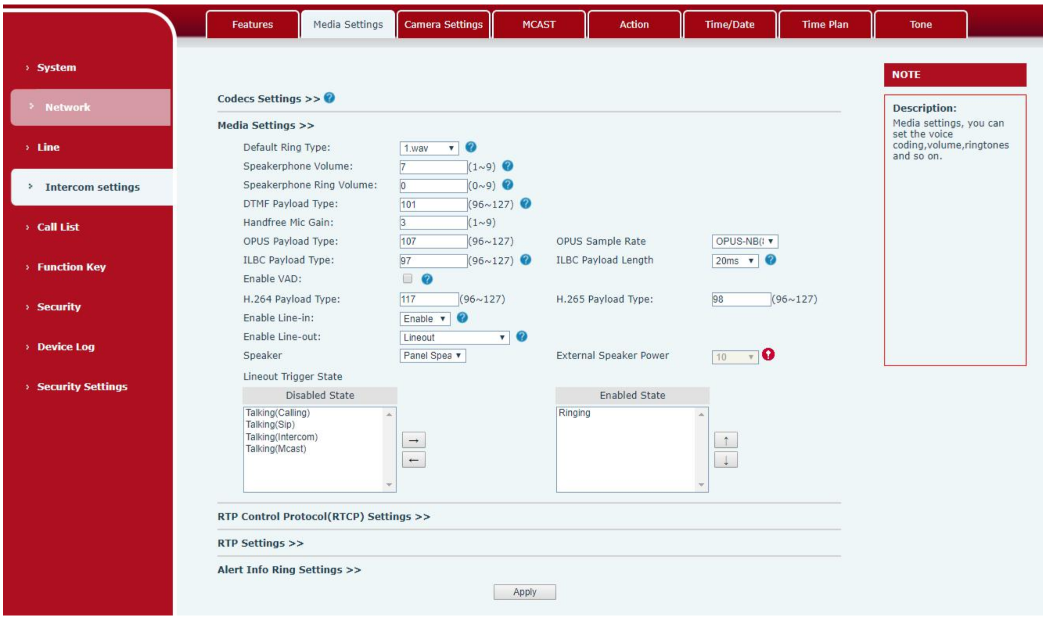
Table 17- Common device function Settings on the web page

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
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.
Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out any number.
Enable Restricted Incoming List	Whether enable Restricted Incoming List
Enable Restricted Outgoing List	Wether enable Restricted Outgoing List
Enable country Code	Wether enable country Code

Country Code	Country Code
Area Code	Area Code
Allow IP Call	If enabled, user can dial out with IP address
P2P IP Prefix	You can set IP call prefix,for example,i set it as “172.16.2.”,then i input #160 in dialpad and press dial key ,it will call 172.16.2.160 automatically
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP address. More details please refer to this link https://www.fanvil.com/Support/download/cid/14.html
Push XML Server	Configure the Push XML Server, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.
Line Display Format	Line display format including SIPn/SIPn：xxx/xxx@SIPn
Call Number Filter	Configure a special character & ,if the number is 78 & 9. The call will be filtered out&
Auto Resume Current	If the current path changes, the hold will be automatically resume
Limit Talking Duration	Automatically hang up the call after enabling the time set for the call
Talking Duration	Call duration ,20-600s
No Answer Auto HangUp 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
Tone Settings	
Enable Holding Tone	When turned on, a tone plays when the call is held
Enable Call Waiting Tone	When turned on, a tone plays when call waiting
Play Dialing DTMF Tone	Play DTMF tone on the device when user pressed a phone digits at dialing, default enabled.
Play Talking DTMF Tone	Play DTMF tone on the device when user pressed a phone digits during taking, default enabled.
Intercom Settings	
Enable Intercom	When intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after specific delay.
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers the intercom call during a call. If the current call is intercom call, the phone will reject the second intercom call
Response Code Settings	
Busy Response Code	Set the SIP response code on line busy

Reject Response Code	Set the SIP response code on call rejection

9.17 Intercom settings >> media



Picture 29- Media Settings

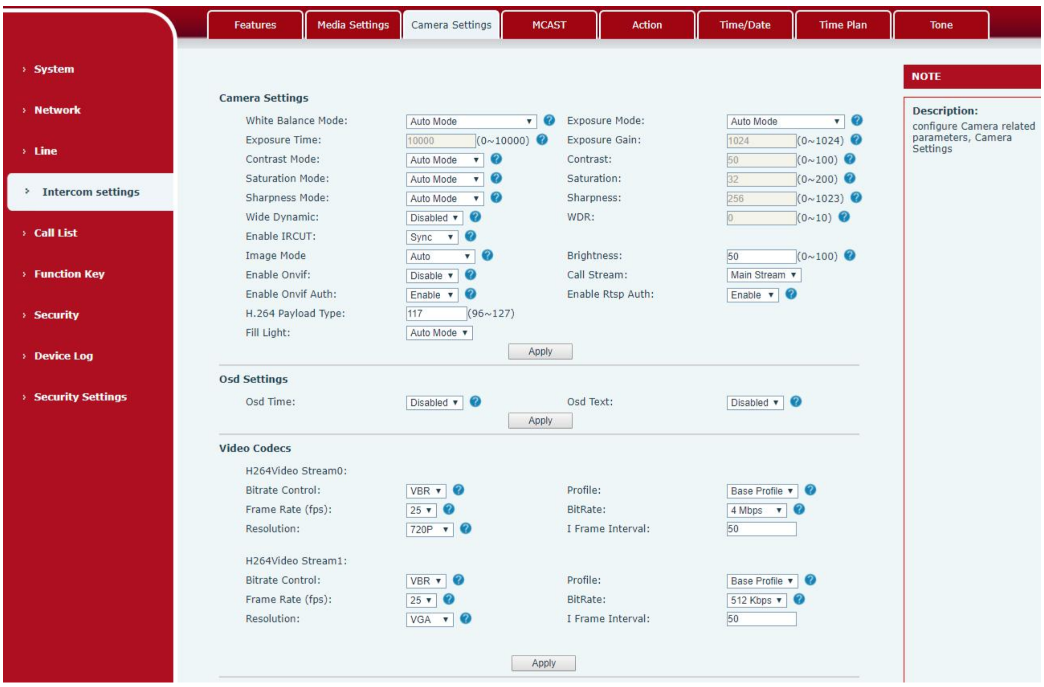
Table 18- Audio Settings

Parameters	**Description**
Codecs Settings	Select the enabled and disabled voice codecs codec:G.711A/U,G.722,G.723,G.729,G.726-32, ILBC,AMR,AMR-WB
Audio Settings	
Default Ring Type	Set the default ring type. If the caller ID of an incoming call was not configured with specific ring type, the default ring will be used.
Speakerphone Volume	Set the speakerphone volume, the value must be 1~9
Speakerphone Ring Volume	Set the ring volume in the speakerphone, the value must be 1~9
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
Opus playload type	Enter the opus payload type, the value must be 96~127.
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
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.
RTP Control Protocol(RTCP) Settings	
CNAME user	Set the CNAME user
CNAME host	Set the CNAME host
RTP	
RTP keep alive	Keep talking, send a packet 30 seconds after enable it
Alert Info Ring Settings （alert-info）	
Value of notification message 1 to 10	Set the value of the specified ring type
ring type	The ring type

9.18Intercom settings>>Camera Settings

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



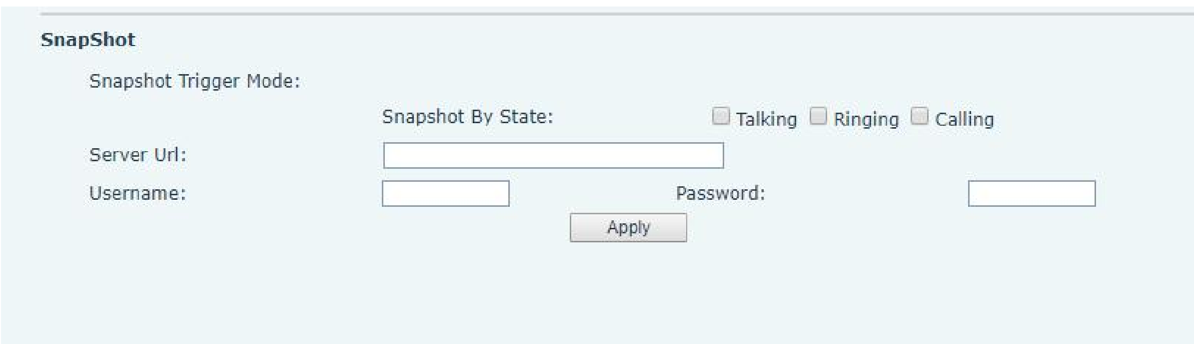
Picture 30- Camera Settings

Table 19- Camera Settings

Parameters	**Description**
camera settings	
White Balance Mode	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.。 Lock mode：Fixed white balance parameters will not be automatically adjusted according to the actual color temperature. Incandescent lamp mode：To compensate for the hue of incandescent lamps, it is suitable for use under beige light sources (bulbs, tungsten lamps, candles) and other light sources of this type.。 Warm light mode：Compensate the hue of warm light, suitable for light sources with a color temperature of about 2700K。 Naturl light mode：It can be used for white balance in outdoor shooting and has a wide range of applications.。 Fluorescent lamp light：Compensate the hue of fluorescent lamps, suitable for use under fluorescent light sources (fluorescent lamps, energy-saving lamps) and other types of light sources。
Exposure Mode	**Auto mode** ：The camera automatically sets the parameters, no need for the operator to adjust. **Manual exposure time** ： Set the exposure time by yourself, the range is 0~10000 **Manual exposure gain**：Set the exposure gain by yourself, the range is 0~1024 All manual ： Manually set the exposure time and gain.
Exposure Time	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.
Exposure Gain	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.
Contrast Mode	Auto mode：The camera automatically sets the contrast according to the

	environment, no need for the operator to adjust Manual mode：Manually set the camera's contrast parameters.
Contrast	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.
Saturation Mode	Auto mode：The camera automatically sets the saturation according to the environment, without the need for the operator to adjust **Manual mode：**Manually set the camera's saturation parameters.
Saturation	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.
Sharpness Mode	Auto mode：The camera automatically sets the sharpness according to the environment, no need for the operator to adjust **Manual mode：**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
Osd Settings	
Osd Time	Turn on/off the date display of the camera image interface.
Color Style	Display colors: black, red, blue, green.
Time Position	Display position: top left, top right, bottom left, bottom right.
Font Size	Display font size：16*16,20*20
Osd Text	Enable/disable the text display of the camera image interface.
Title Message	Text display content of camera image interface
Video Codecs	
H264 Video	Support H.264 encoding format

Stream	
Bitrate Control	VBR：Video call will adapt to the bit rate of the opposite end, so that the video effect is better. CBR：The video call will not change according to the bit rate set by itself.
Resolution	Support 1080P，720P，4CIF,VGA,CIF,QVGA
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 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.
RTSP Information	
Main Stream Url	Display the main stream URL address
Sub Stream Url	Display the sub stream URL address



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 (SD card / USB flash disk)

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

9.19 Intercom Setting >> MCAST

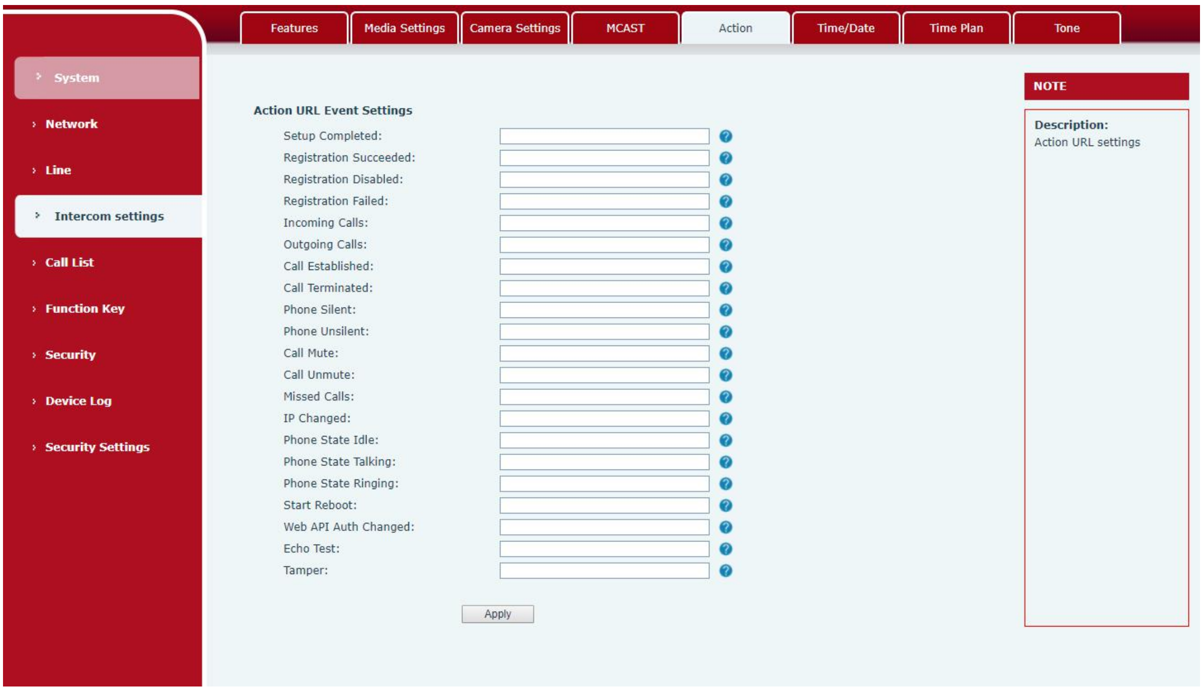
It is easy and convenient to use multicast function to send notice to each member of the multicast via setting the multicast key on the device and sending multicast RTP stream to pre-configured multicast address. By configuring monitoring multicast address on the device, monitor and play the RTP stream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

The detail for 8.2 MCAST

9.20 Intercom Setting >> Action URL

Table 20- Action URL

Action URL Event Settings
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



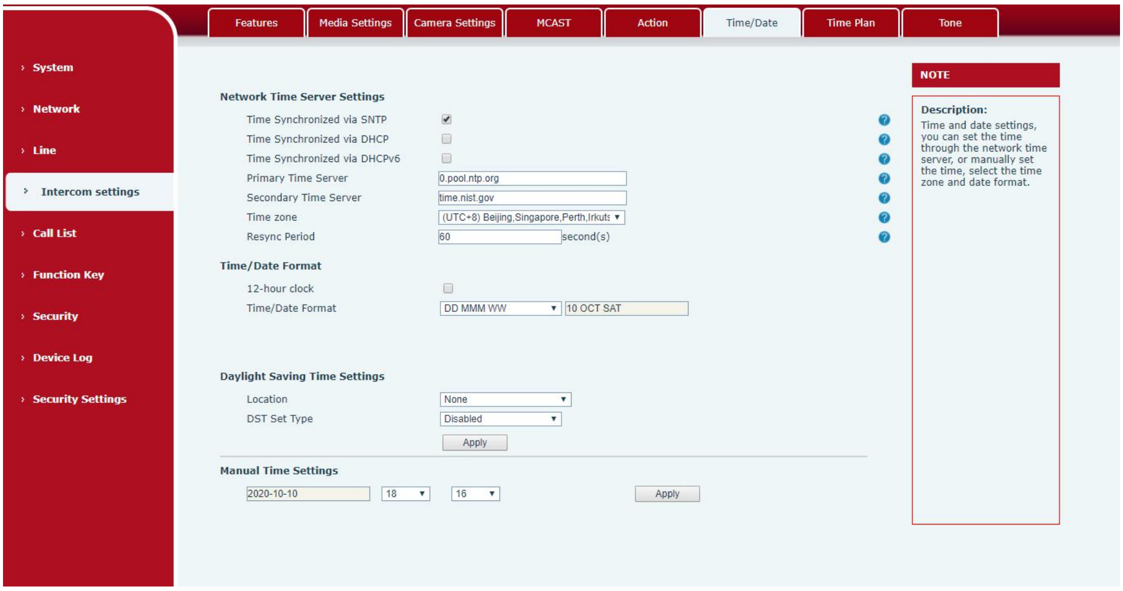
Picture 32- Action URL

Note! The operation URL is used by the IPPBX system to submit device events. Please refer to the details Fanvil Action URL。

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

9.21 Intercom Setting >> Time/Date

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



Picture 33 - Time/Date

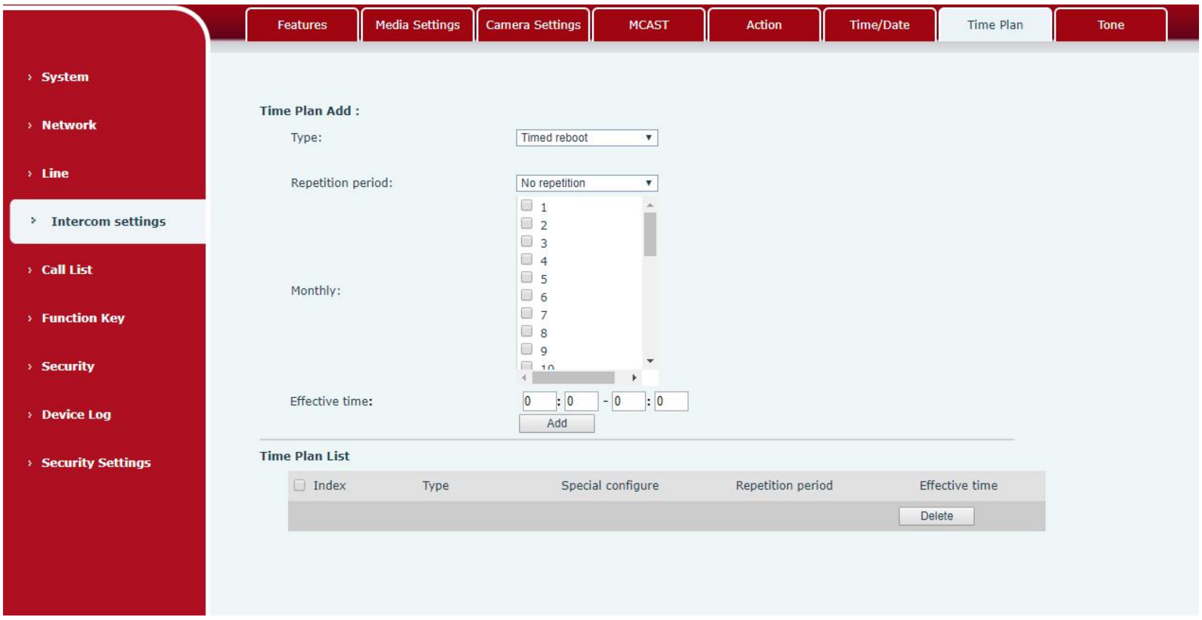
Table 21- Time/Date

Time/Date	
Field Name	**Explanation**
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Primary Time Server	Set primary time server address
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization.
Time zone	Select the time zone
Resync Period	Time of re-synchronization with time server
Daylight Saving Time Settings	
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
Manual Time Settings	
To set the time manually, you need to disable the SNTP service first, and you need to fill in and submit each item of year, month, day, hour and minute in the figure above to make the manual settings successful.	
System time: Display system time and its source (SIP automatic get >SNTP automatic get >manual manual setting)	

9.22Intercom settings>>Time plan

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



Picture 34- Time Plan

Table 22- Time Plan

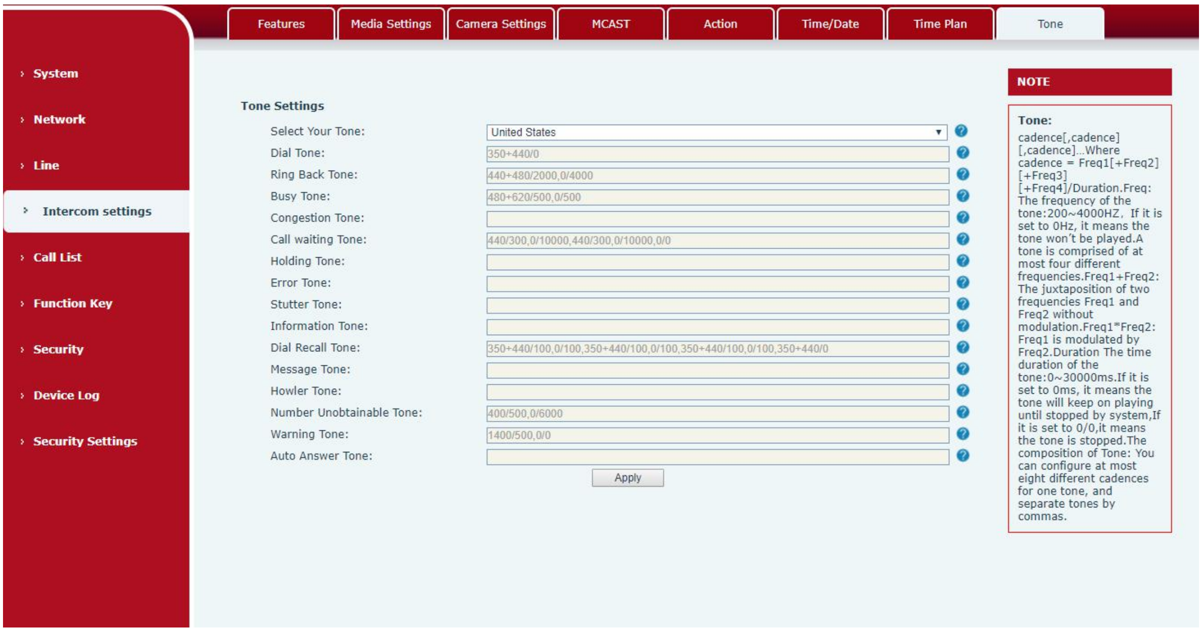
Parameters	Description
type	Timing restart, timing upgrade, timing sound detection, timing playback audio
Audio path	Support local, U disk, SD card

	Local: select the audio file uploaded locally U disk: select the audio file under the U disk SD card: select the audio file under the SD card
Audio settings	Select the audio file you want to play, it supports trial listening, and you can play it immediately after clicking the trial listening
Repeat cycle	Do not repeat: execute once within the set time range Daily: Perform this operation in the same time frame every day Weekly: Do this in the time frame of the day of the week Monthly: the time frame of the month to perform this operation
Effective time	Set the time period for execution

9.23Intercom settings >> Tone

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

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



Picture 35- Tone

9.24Call list >> Call List

- Restricted Incoming Calls

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

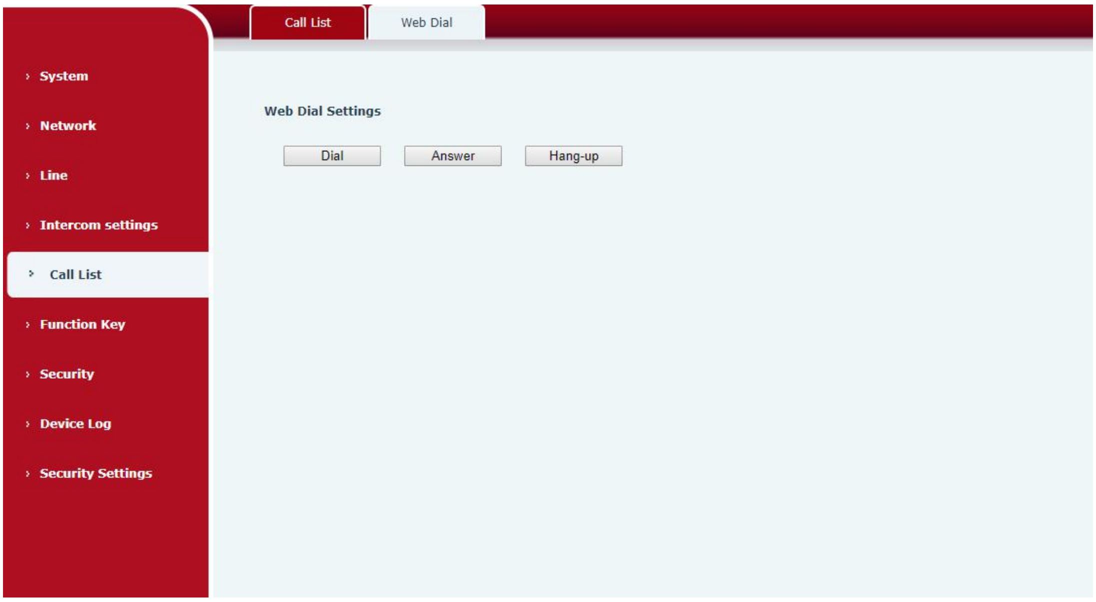
User can add specific number to be blocked, or a prefix where any numbers matched the prefix will all be blocked.

■ Restrict Outgoing Call

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

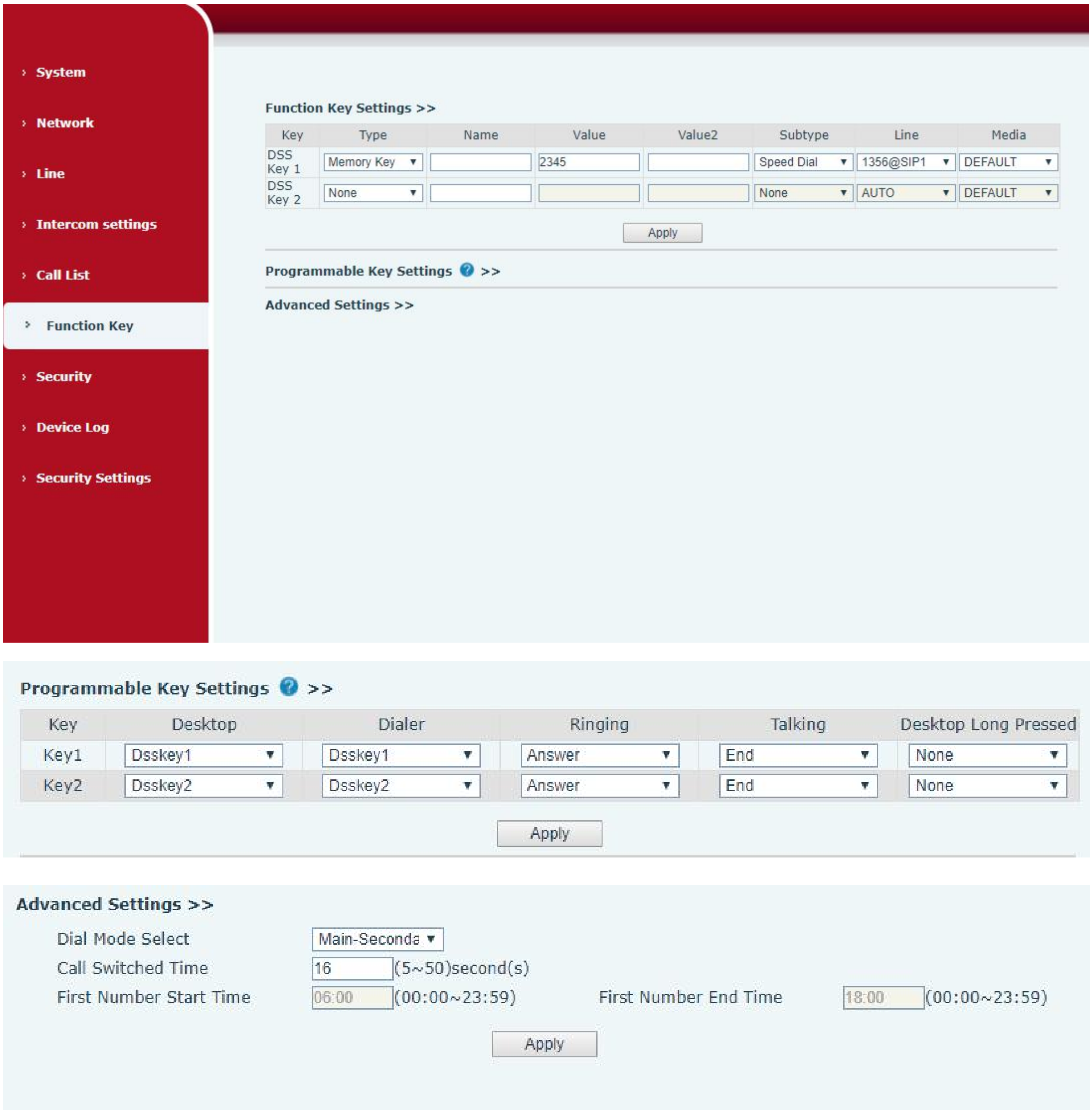
9.25Call list >> Web Dial

Use web page to call, answer and hang up.



Picture 36- Webpage Dial

9.26Function key



Picture 37- Function Key

Table 23- Function Key

Parameters	**Description**
Function key settings	
memory	**Speed Dial**:The user can directly dial the set number. This feature is convenient for customers to dial frequent numbers. Intercom: This feature allows the operator or secretary to quickly connect to the phone, widely used in office environments

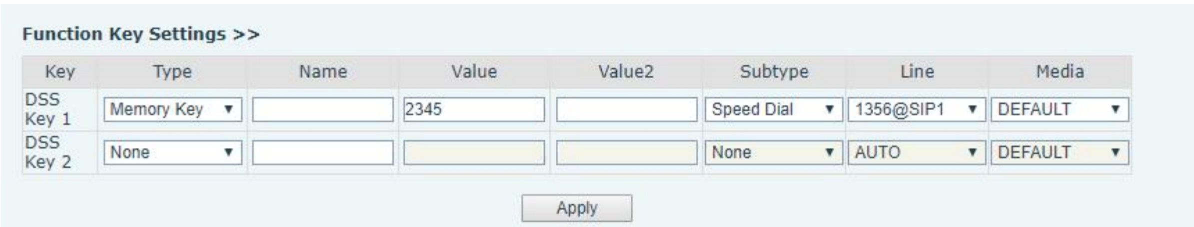
Key event	The user can select a function key as a shortcut to trigger an event for example: None /Handfree
DTMF	Press during a call to send the set DTMF
Mcast Paging	Configure the multicast address and voice encoding. User can initiate multicast by pressing this key
Action URL	The user can use a specific URL to make basic calls to the device, open the door, etc.
Mcast Listening	In standby, press the function key, if the RTP of the multicast is detected, the device will monitor the multicast
PTT	**Speed dial**：Make a call when pressed, and end the call when lifted. Intercom： Start the intercom when pressed, and end the intercom when lifted. Multicast：Initiate multicast when pressed, and end multicast when lifted
Programmable Key Settings	
Desktop	None：Nothing happens when you press the speed dial Dsskey1：When it is set to dsskey1, follow the settings of dsskey1 to make call, answer, etc. Dsskey2： When it is set to dsskey2, perform operations such as calling and answering according to the setting of dsskey2
Dialer	None：Nothing happens when you press the speed dial Dsskey1：When it is set to dsskey1, follow the settings of dsskey1 to make call, answer, etc. Dsskey2： When it is set to dsskey2, perform operations such as calling and answering according to the setting of dsskey2
Ringing	Answer：Set to answer, when there is an incoming call, if auto answer is disabled, press the speed dial key to answer the call End: set to end, when there is an incoming call, press the speed dial button to hang up the call
Talking	End: set to end, when there is a call, press the speed dial key to hang up the call Volume up: set as volume up button, when there is a call, press the speed dial button to increase the volume Volume down: set as volume up button, when there is a call, press the speed dial button to decrease the volume Dsskey1：When it is set to dsskey1, follow the settings of dsskey1 to make call, answer, etc. Dsskey2： When it is set to dsskey2, perform operations such as calling and answering according to the setting of dsskey2

Desktop Long Pressed	None: Long press the speed dial key does not respond 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. <Main/Secondary>: If the first number is not answered within the set time, the second number will be automatically switched. <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.



Picture 38 - Memory Key

Table 24- Memory Key

Type	number	line	Subtype	usage
memory	Fill in the SIP account or IP address	The line corresponding to the SIP	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

	of the called party	account		intercom function, the call can be automatically answered.

➢ Multicast

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

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

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

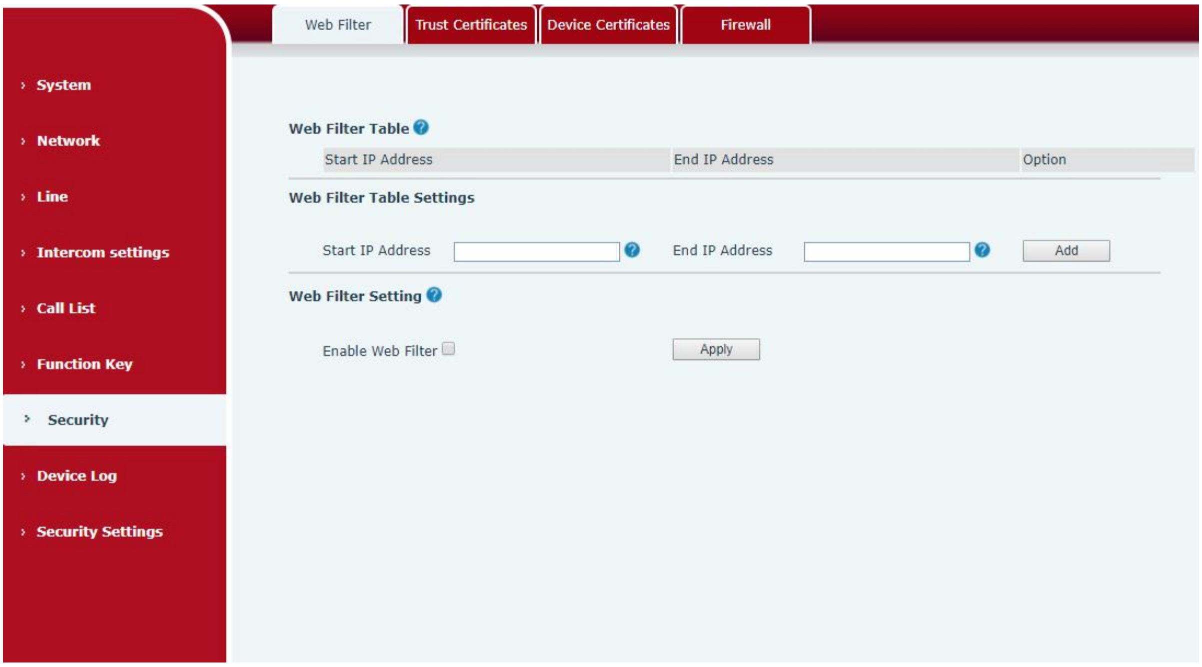
➢ PTT

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

Picture 40 - Advanced Setting

9.27Security >> Web filter

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



Picture 41- WEB filter

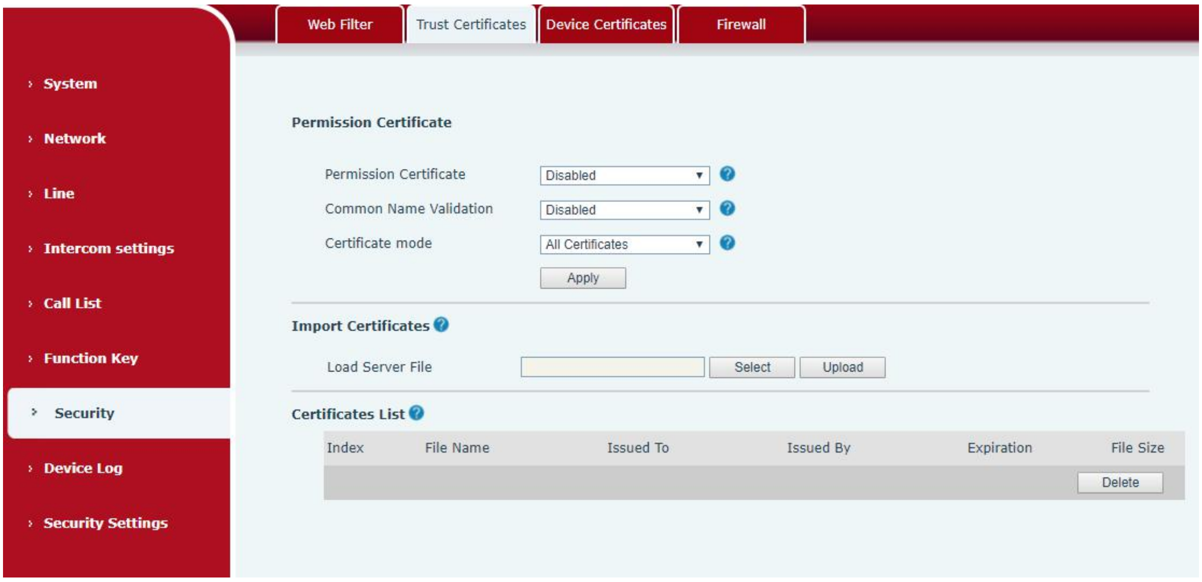
Add and delete the allowed IP network segments; configure the start IP address in the start IP, configure the end IP address in the end IP, and then click [Add] to add successfully. You can set a large network segment or add it into several network segments. When deleting, select the starting IP of the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

9.28Security >> Trust Certificates

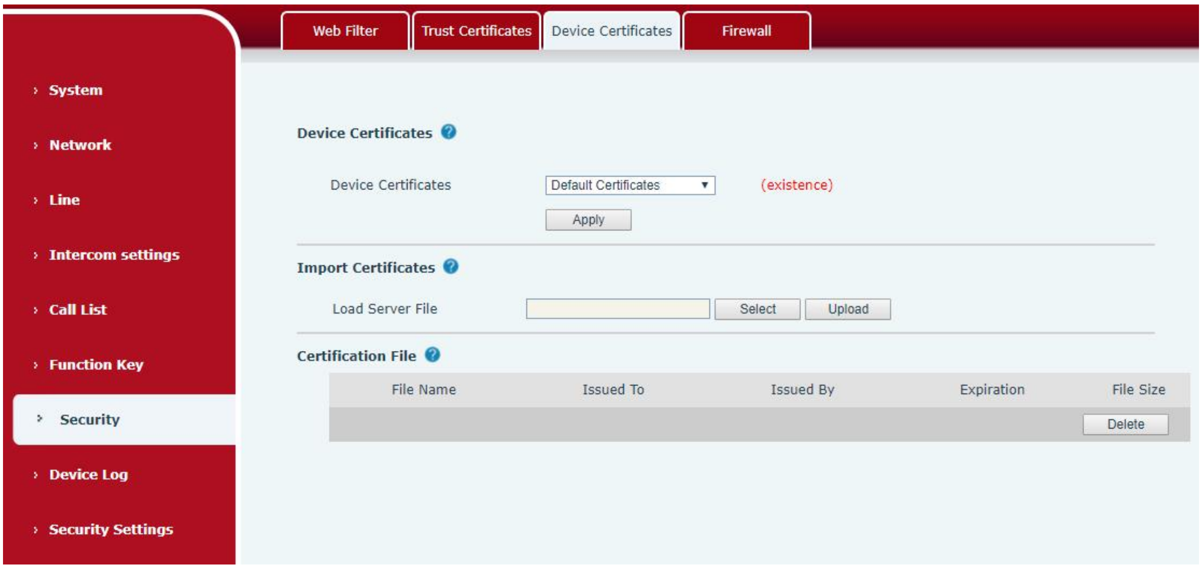
You can upload and delete uploaded trust certificates.



Picture 42 - Trust Certificates

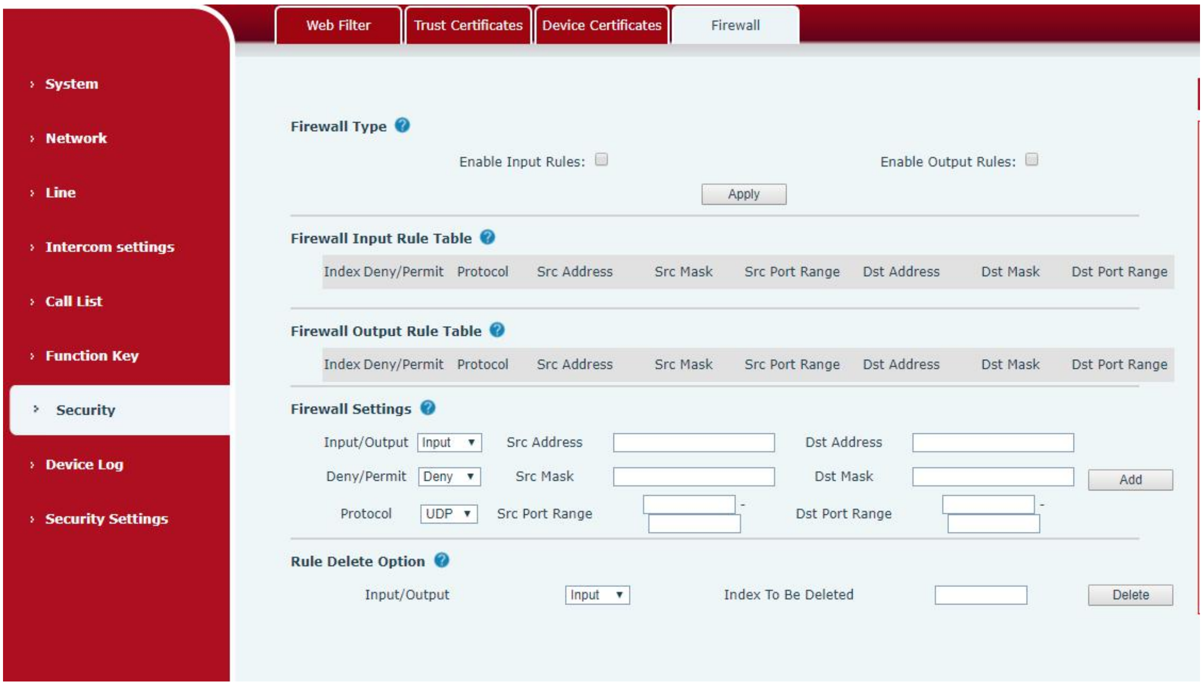
9.29Security >> Device Certificates

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



Picture 43- Device Certificates

9.30Security >> 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.

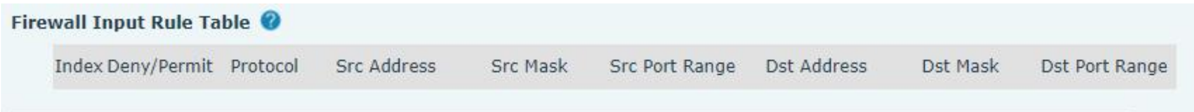
Taking into account the complexity of firewall settings, the following will illustrate with an example:

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

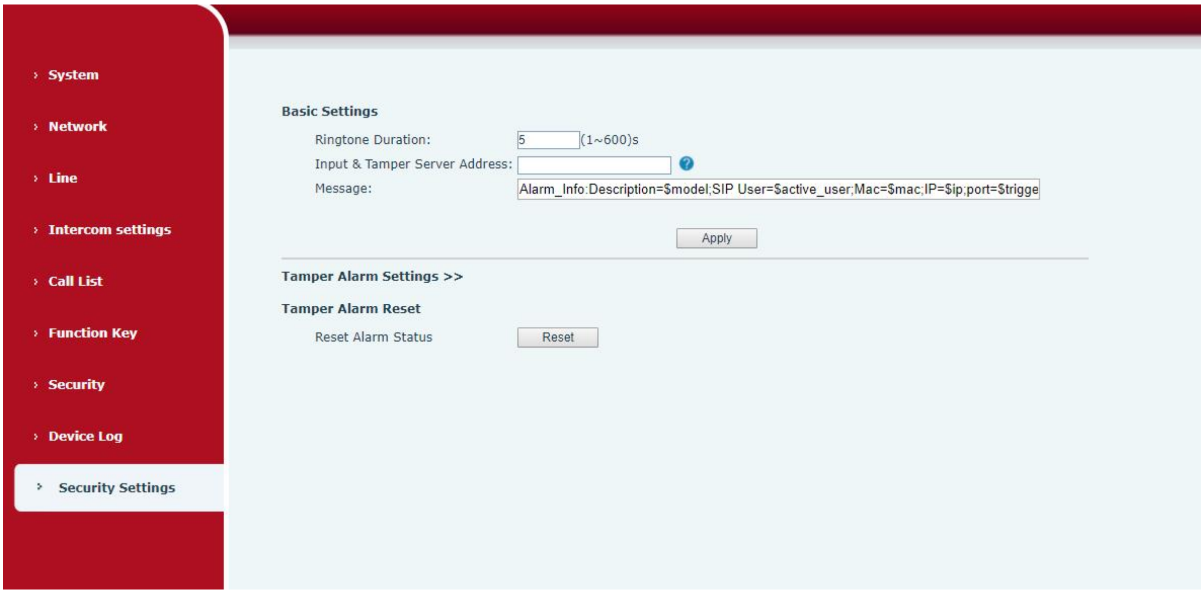
Select the list you want to delete and click [Delete] to delete the selected list.

9.31Device log

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

9.32Security settings

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



Picture 47 - Security Settings

Table 27- Security Settings

Security settings	
Parameters	**Description**
Basic settings	
Ringtone Duration	The Alarm ring duration
Input & Tamper 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 Alarm	If the terminal is forcibly removed, the tamper will be triggered and the set alarm ring will be played all the time
Alarm command	When the alarm is triggered, the server sends the command immediately

Reset 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 Status	Reset to stop the playing of the bell
Alarm Ringtone	The ringtone of alarm

10 Trouble Shooting

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

10.1 Get device system information

Users can obtain information through the [**System**] >> [**Information**] option on the device webpage. The following information will be provided:
Device information (model, software and hardware version) and Internet Information etc.

10.2 Reboot device

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.

10.4 Network Packets Capture

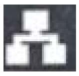
In order to obtain the data packet of the device, the user needs to log in to the webpage of the device, open the webpage [**System**] >> [**Tools**], and click the [**Start**] option in the "Network Packets Capture". A message will pop up asking the user to save the captured file. At this time, the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

10.5Get device log

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

10.6 Common Trouble Cases

Table 25 - Trouble Cases

Trouble Case	**Solution**
Device could not boot up	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. 2. If the device enters "POST mode" (the SIP/NET and function button indicators are always on), the device system is damaged. Please contact your location technical support to help you restore your equipment system.
Device could not register to a service provider	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. 2. If the network connection is good, please check your line configuration again. If all configurations are correct, contact your service provider for support, or follow the instructions in "10.4 Network Data Capture" to obtain a registered network packet and send it to the Fanvil Support Email to help analyze the issue.