User Manual

i10&i10V&i10D

Software Version: 1.0.0

Release Date：2019/07/24

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 the power cord or plug is impaired, do not use it because it may cause fire or electric shock.
- Do not drop, knock or shake the phone. Rough handling can break internal circuit boards.
- This phone is designed for indoor use. Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- 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 the power plug, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

4 Overview

The i10/i10V/i10D SIP mini intercom is designed for indoor scenes with IP54 waterproof and dustproof. Supports wall mounting installation. It combines security, audio/video intercom and broadcasting functionalities and offers a qualified communication solution for users.

5 Install Guide

5.1 Use POE or external Power Adapter

i10/i10V/i10D, called as ‘the device’ hereafter, supports two power supply modes, power supply from an external power adapter or over Ethernet (POE) complied switch.

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

For users who do not have POE equipment, the traditional power adaptor should be used. If the device is connected to a POE switch and power adapter at the same time, the power adapter will be used in priority and will switch to POE power supply once it fails.

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

5.2 Appendix Table

5.2.1 Common command mode

Table 1 - Common command mode

Action	Description
IP Broadcast under standby mode	In standby mode, long press the speed dial button for 3 seconds, there will be a toot sound will 5 seconds, please press the speed dial button once within 5 seconds, the toot sound will stop automatically reporting IP
Switch network mode	In the standby mode, long-press the speed dial button for 3 seconds and the beep will last for 5 seconds. Within 5 seconds, press the speed dial button three times quickly to switch to the network mode. If there is no IP at present, switch to the default static IP (192.168.1.128). Then switch to DHCP mode when it is the default static IP (192.168.1.128) When DHCP gets to IP, then do not switch and report the IP directly. Report the IP after the successful switch.

5.2.2 Function key LED state

Table 2 - Function key LED state

Type	LED	State
Speed dial	Normally on	Successfully registered
	Quick flashing	Registration failed/ network abnormal
	Slow flashing	In call

6 Basic Introduction

6.1 Panel Overview

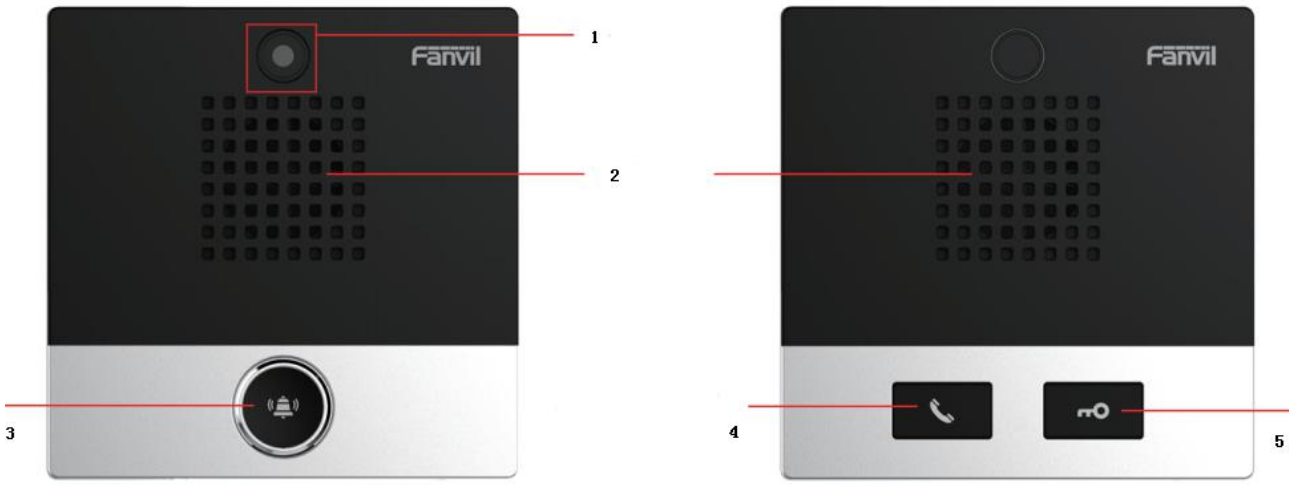


Figure 1 - Panel

Table 3 - Panel introduction

Number	Name	Description
1	IP Camera	Video signal acquisition and transmission
2	Speaker	Play sound
3	Speed dial button	For speed dial, multicast, intercom, IP broadcast and other functions
4	Speed dial/Answer button	For speed dial/answer button, multicast, intercom, IP broadcast and other functions
5	Unlock	Unlock door

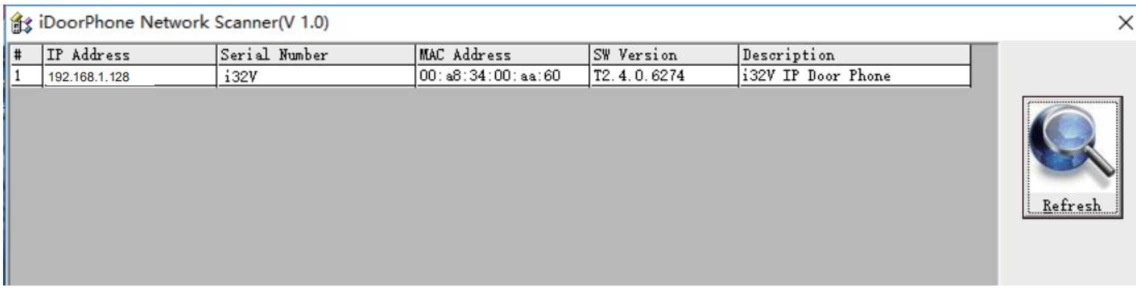
6.2 Quick Setting

Before proceeding with this step, make sure your Internet broadband connection is working properly and complete the network hardware connection. The default factory mode is DHCP. IP address can be viewed by.

- In standby mode, long press the speed dial button for 3 seconds, there will be a toot sound will 5 seconds, please press the speed dial button once within 5 seconds (please do not operate within 30 seconds when power on), the toot sound will stop automatically reporting IP.
- Of the device or use the "IP scanning tool. exe"

software to find the IP address of the device.

（Download http://download.fanvil.com/tool/iDoorPhoneNetworkScanner.exe）



iDoorPhone Network Scanner(V 1.0)

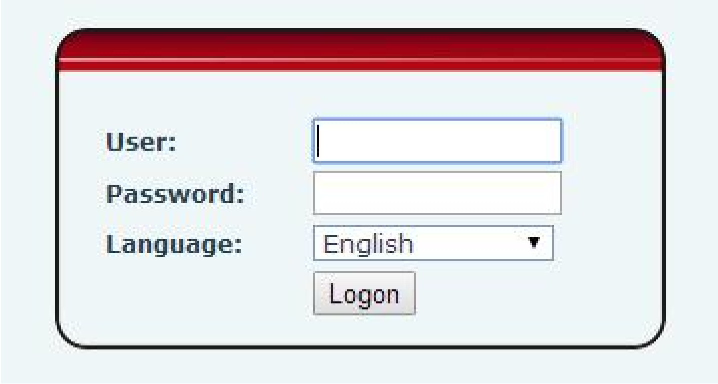
#	IP Address	Serial Number	MAC Address	SW Version	Description
1	192.168.1.128	i32V	00:a8:34:00:aa:60	T2.4.0.6274	i32V IP Door Phone

Figure 2 - Quickly setting

- In the standby mode, long-press the speed dial button for 3 seconds and the beep will last for 5 seconds. Within 5 seconds, press the speed dial button three times quickly to switch to the network mode.
- Login to the device's WEB page for configuration according to the IP address
- Configure the account, user name, server address and other parameters required for registration provided by the service provider on the WEB configuration page;

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.



User:

Password:

Language: English

Figure 3 - WEB Login

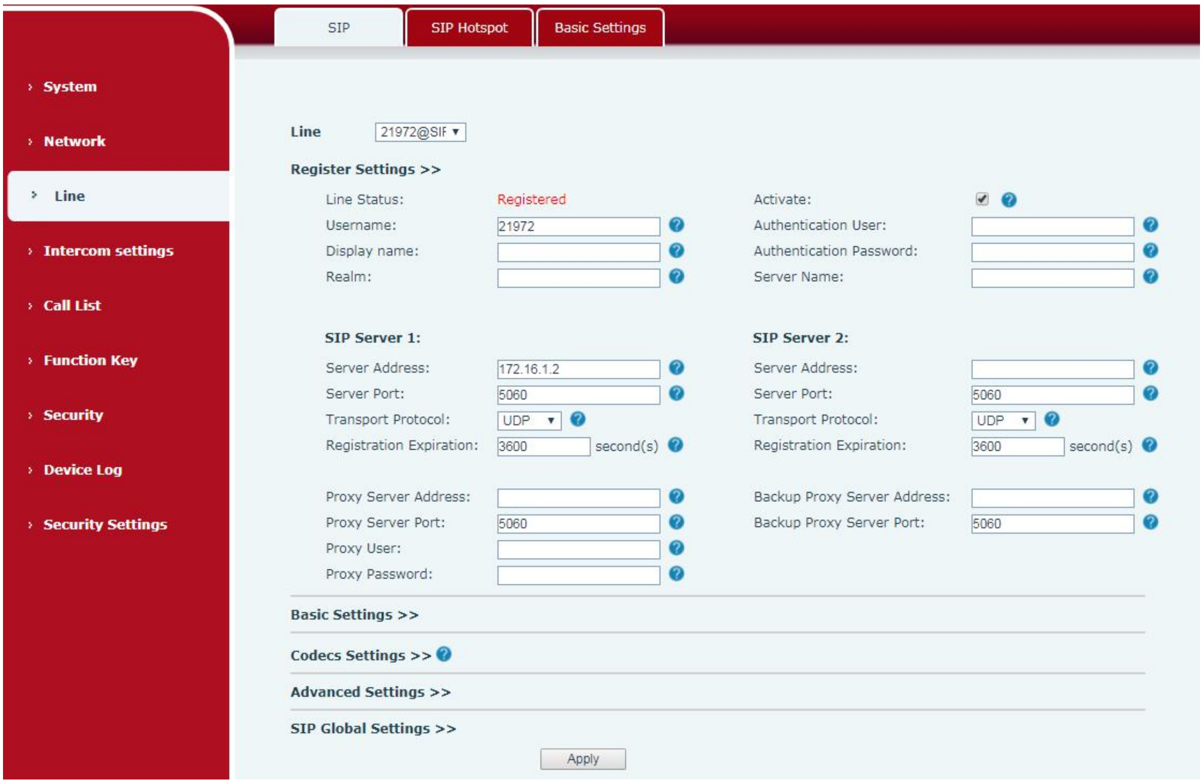
The username and password should be correct to log in to the web page. **The default username and password are "admin"**. For the specific details of the operation of the web page, please refer to 9 Web Configurations

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 is 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:



Line: 21972@SIF ▼

Register Settings >>

Line Status:	Registered	Activate:	☑
Username:	21972	Authentication User:	
Display name:		Authentication Password:	
Realm:		Server Name:	
SIP Server 1:		**SIP Server 2:**	
Server Address:	172.16.1.2	Server Address:	
Server Port:	5060	Server Port:	5060
Transport Protocol:	UDP ▼	Transport Protocol:	UDP ▼
Registration Expiration:	3600 second(s)	Registration Expiration:	3600 second(s)
Proxy Server Address:		Backup Proxy Server Address:	
Proxy Server Port:	5060	Backup Proxy Server Port:	5060
Proxy User:			
Proxy Password:			

Basic Settings >>

Codecs Settings >>

Advanced Settings >>

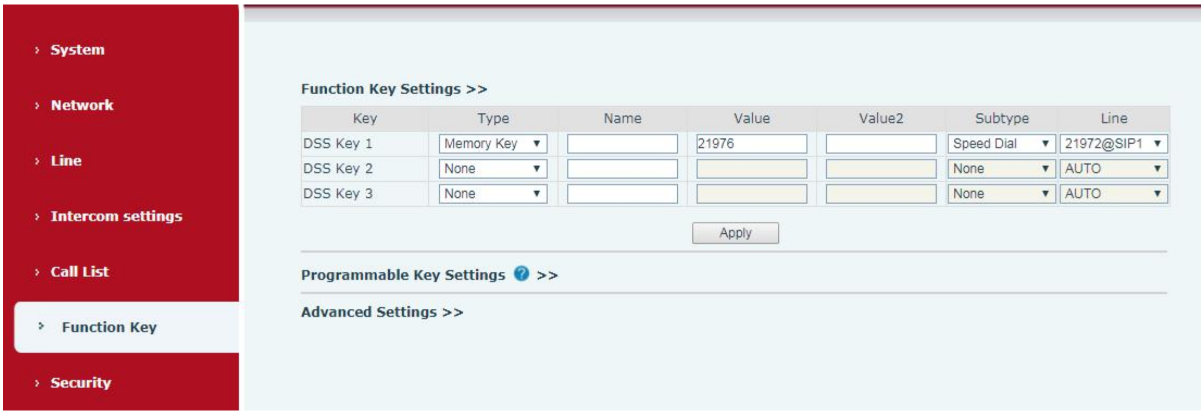
SIP Global Settings >>

Figure 4 - SIP Line Configuration

7 Basic Function

7.1 Making Calls

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



Function Key Settings >>

Key	Type	Name	Value	Value2	Subtype	Line
DSS Key 1	Memory Key ▾		21976		Speed Dial ▾	21972@SIP1 ▾
DSS Key 2	None ▾				None ▾	AUTO ▾
DSS Key 3	None ▾				None ▾	AUTO ▾

Figure 5 - Function Setting

See detailed configuration instructions 9.23 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 the 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

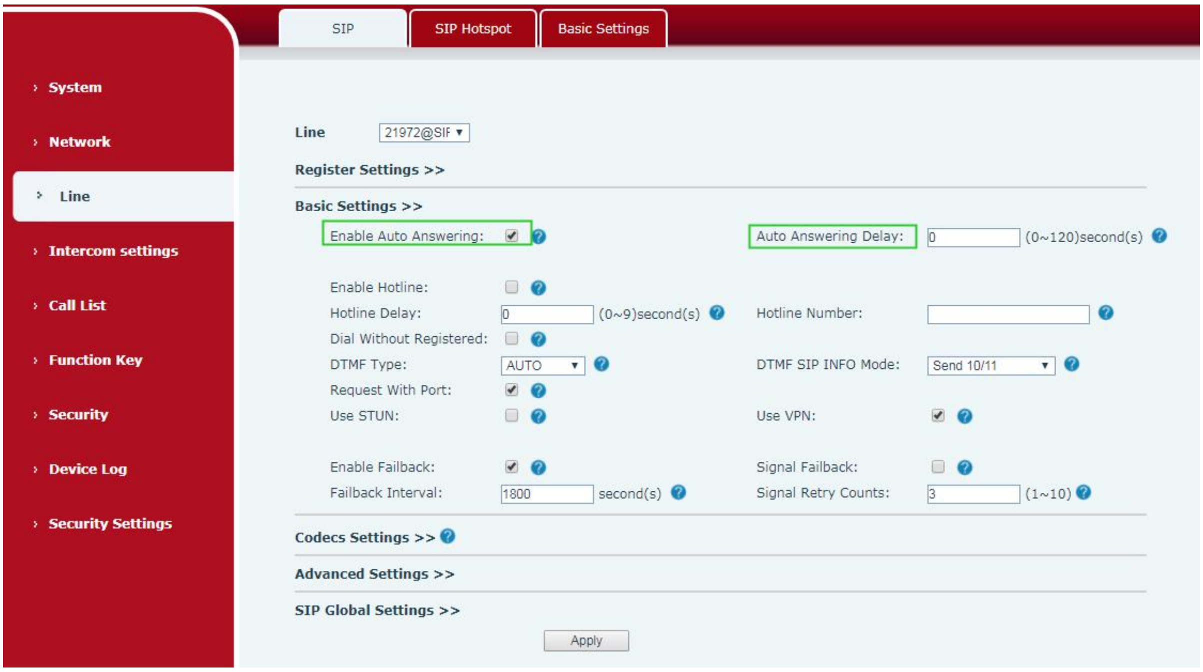
When there is a call, you can press the speed dial button to hang up the call, the default setting is to end the call. See detailed configuration instructions 9.23 Function Key.

7.4 Auto-Answering

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.

- **Enable auto answering on the line：**

Web interface: enter [**Line**] >> [**SIP**], Enable auto answer, set mode and auto answer time and click submit.



Line: 21972@SIP

Register Settings >>

Basic Settings >>

Enable Auto Answering: ☑

Auto Answering Delay: 0 (0~120)second(s)

Enable Hotline: □

Hotline Delay: 0 (0~9)second(s)

Hotline Number:

Dial Without Registered: □

DTMF Type: AUTO

DTMF SIP INFO Mode: Send 10/11

Request With Port: ☑

Use STUN: □

Use VPN: ☑

Enable Failback: ☑

Signal Failback: □

Failback Interval: 1800 second(s)

Signal Retry Counts: 3 (1~10)

Codecs Settings >>

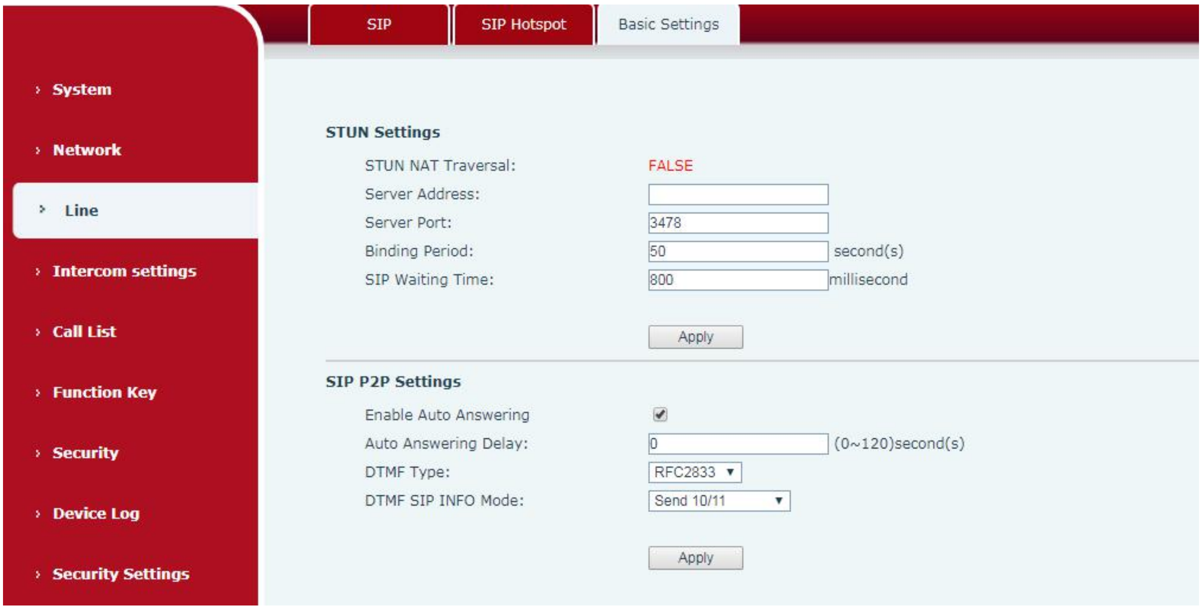
Advanced Settings >>

SIP Global Settings >>

Figure 6 - Enable Auto Answer

- **Enable auto answering P2P**:

Web interface: enter [**line**] >> [**Basic Settings**] >> [**SIP P2P Settings**], enable automatic answering, setting mode and automatic answering time, and click submit.



STUN Settings

STUN NAT Traversal: FALSE

Server Address:

Server Port: 3478

Binding Period: 50 second(s)

SIP Waiting Time: 800 millisecond

SIP P2P Settings

Enable Auto Answering ☑

Auto Answering Delay: 0 (0~120)second(s)

DTMF Type: RFC2833

DTMF SIP INFO Mode: Send 10/11

Figure 7 - Enable Auto Answer

- Auto Answer Timeout（0~60）

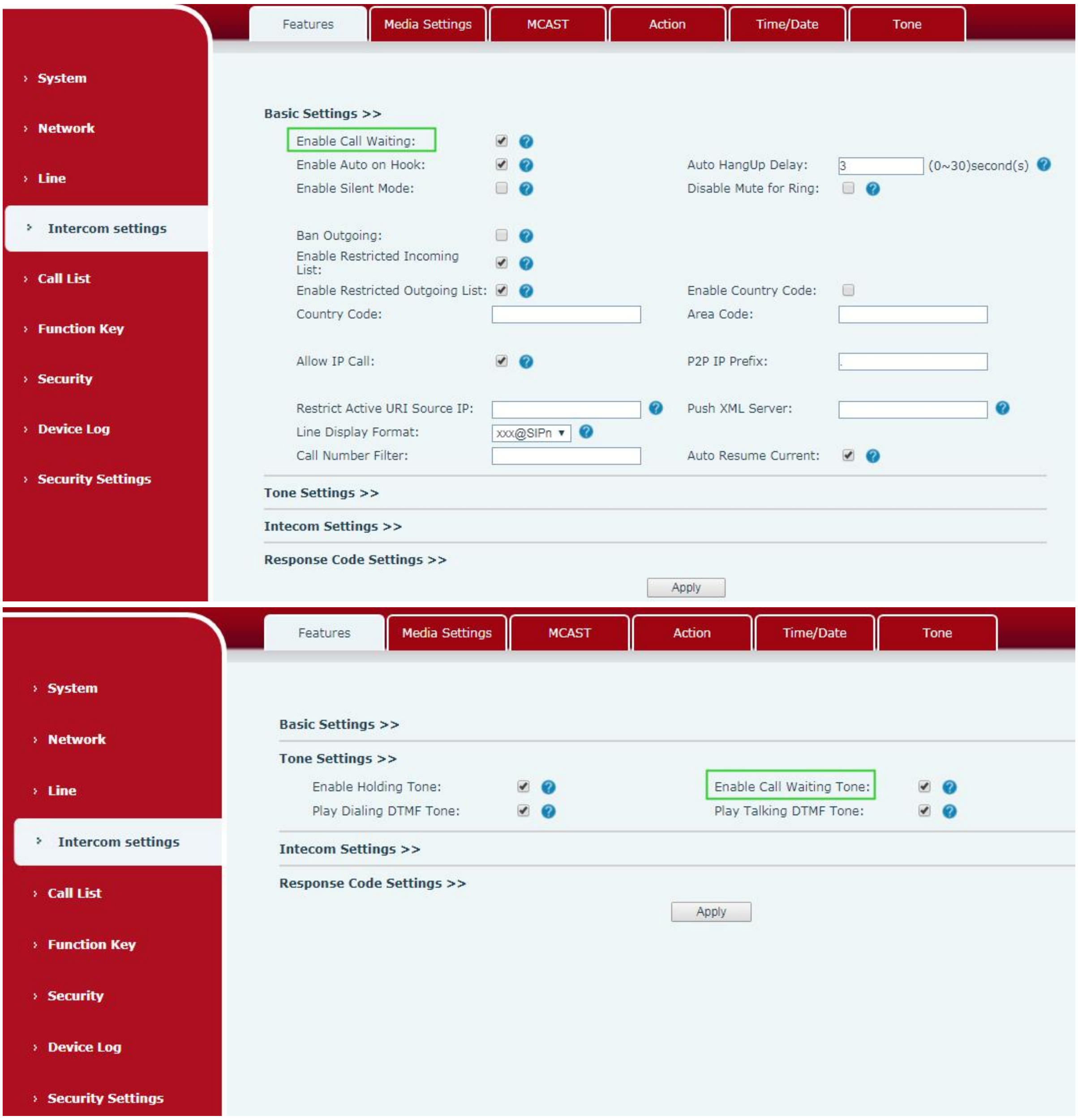
The range can be set to 0~60s, and the call will be answered automatically when the timeout is set.

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 Setting**] >> [**Features**], enable/disable call waiting, enable/disable call waiting tone.



Basic Settings >>

Enable Call Waiting: ☑
Enable Auto on Hook: ☑
Auto HangUp Delay: 3 (0~30)second(s)
Enable Silent Mode: □
Disable Mute for Ring: □
Ban Outgoing: □
Enable Restricted Incoming List: ☑
Enable Restricted Outgoing List: ☑
Enable Country Code: □
Country Code:
Area Code:
Allow IP Call: ☑
P2P IP Prefix:
Restrict Active URI Source IP:
Push XML Server:
Line Display Format: xxx@SIPn
Call Number Filter:
Auto Resume Current: ☑

Tone Settings >>

Intecom Settings >>

Response Code Settings >>

Basic Settings >>

Tone Settings >>

Enable Holding Tone: ☑
Enable Call Waiting Tone: ☑
Play Dialing DTMF Tone: ☑
Play Talking DTMF Tone: ☑

Intecom Settings >>

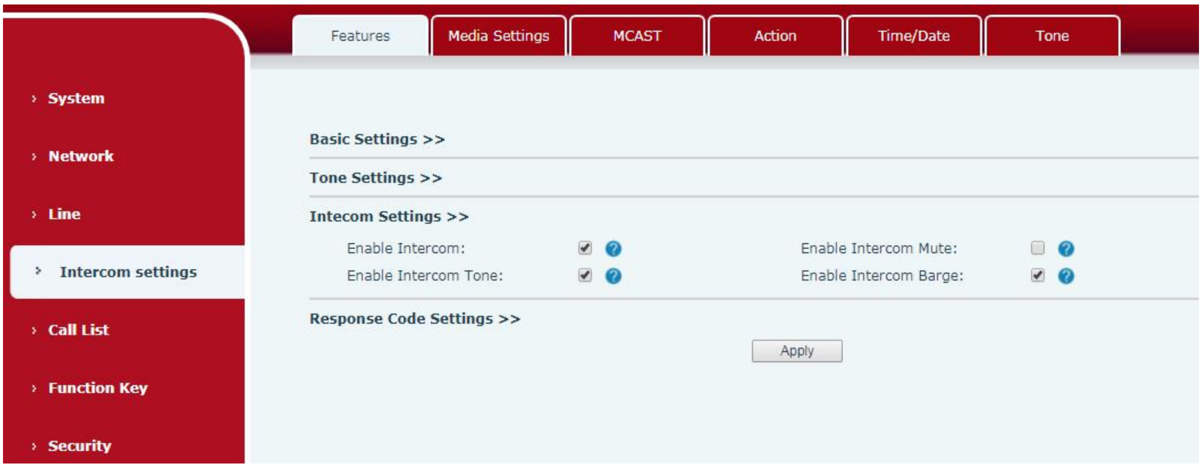
Response Code Settings >>

Figure 8 - Call Waiting

8 Advance Function

8.1 Intercom

The equipment can answer intercom calls automatically.



Intecom Settings >>

Enable Intercom: ☑

Enable Intercom Mute: □

Enable Intercom Tone: ☑

Enable Intercom Barge: ☑

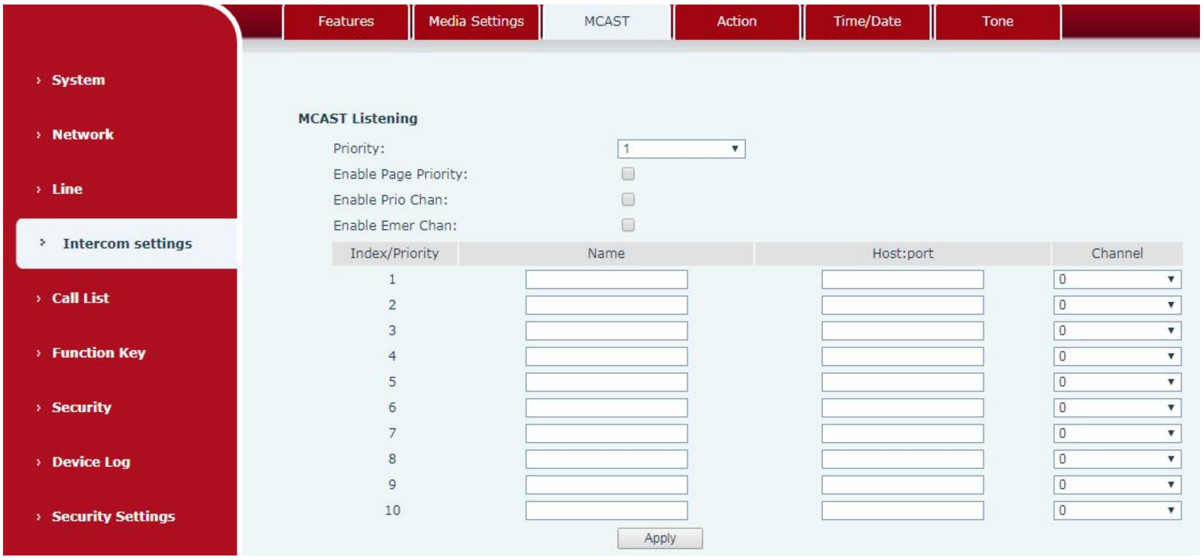
Figure 9 - Intercom

Table 4 - Intercom

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

8.2 MCAST

This feature allows the user to make some kind of broadcast call to people who are in the multicast group. The user can configure a multicast DSS Key on the device, which allows the 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 device to receive an RTP stream from the pre-configured multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.



MCAST Listening

Priority: 1

Enable Page Priority: □

Enable Prio Chan: □

Enable Emer Chan: □

Index/Priority	Name	Host:port	Channel
1			0
2			0
3			0
4			0
5			0
6			0
7			0
8			0
9			0
10			0

Figure 10 - MCAST

Table 5 - MCAST

Parameters	**Description**
Priority	Define the current call's priority, 1 means the highest priority and 10 means the lowest.
Enable Page Priority	If enable page priority, the device will receive the multicast from address with higher priority, regardless of which of the two multicast groups sending the multicast first.
Enable Prio Chan	If enable this option, the only multicast with the same port and channel can be connected. Channel 24 has the higher priority, its priority is higher than 1-23; Set channel value to be 0, it means no channel is used.
Enable Emer Chan	When enabled, channel 25 has the highest priority
Name	Set the multicast server name.
Host:port	Set the multicast server's multicast IP address and port.
Channel	0-25 (24 priority channel,25 emergency channel).

Multicast:

Send multicast:

- Go to web page of [**Function Key**] >> [**Function Key Settings**], select the type to be multicast, set the multicast address, and select the codec.
- Click Apply.
- Press the DssKey of Multicast Key which you set.

Receive multicast:

- Set up the name, host and port of the receiving multicast on the web page of

[**Intercom Settings**] >> [**MCAST**].

- When the remote server sends the multicast, the device will receive multicast call and play multicast automatically.

8.3 Hotspot

SIP hotspot is a simple utility. Its configuration is simple, which can realize the function of group vibration and expand the number of SIP account.

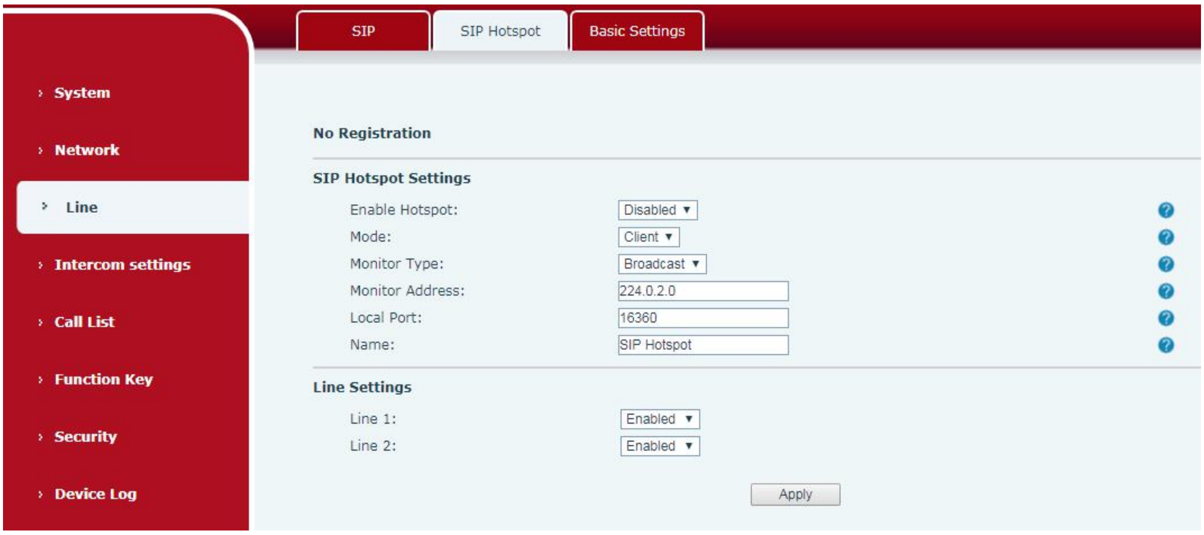
Take one device A as the SIP hotspot and the other devices (B, C) as the SIP hotspot client. When someone calls device A, devices A, B, and C will ring, and if any of them answers, the other devices will stop ringing and not be able to answer at the same time. When A B or C device is called out, it is called out with A SIP number registered with device A.

Table 6 - SIP Hotspot

Parameters	**Description**
Enable Hotspot	Set the enable hotspot option in the SIP hotspot configuration TAB to enabled
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.



No Registration

SIP Hotspot Settings

Enable Hotspot: Disabled

Mode: Client

Monitor Type: Broadcast

Monitor Address: 224.0.2.0

Local Port: 16360

Name: SIP Hotspot

Line Settings

Line 1: Enabled

Line 2: Enabled

Figure 11 - 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 login the device's webpage to manage and operate the device. User must provide the correct username and password to login. If the password is incorrect for three times, the webpage will be locked for 5 minutes and then the user can try to login again.

The details as following:

- If one IP logins more than the specified number of times with different username/passwords, web login will be locked.
- If a same user login more than a specified number of times from different IP addresses, web login will be locked too.

9.2 System >> Information

Users can get the following information in **System>>Information** page:

Basic system information:

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

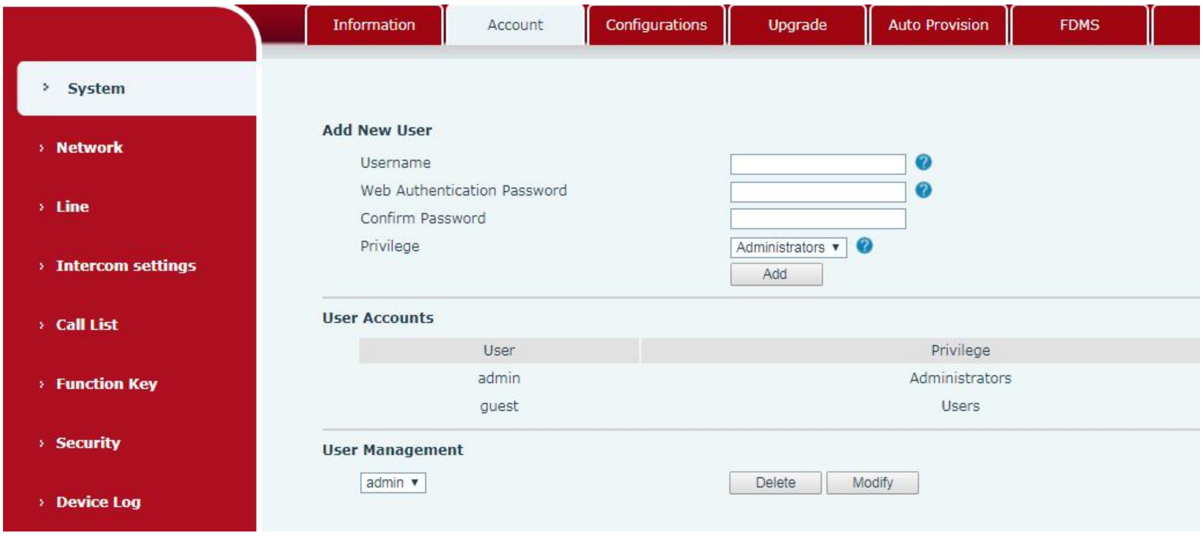
And summarization of network status:

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

Besides, the summarization of SIP account status:

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

9.3 System >> Account



Add New User

Username

Web Authentication Password

Confirm Password

Privilege: Administrators ▾

Add

User Accounts

User	Privilege
admin	Administrators
guest	Users

User Management

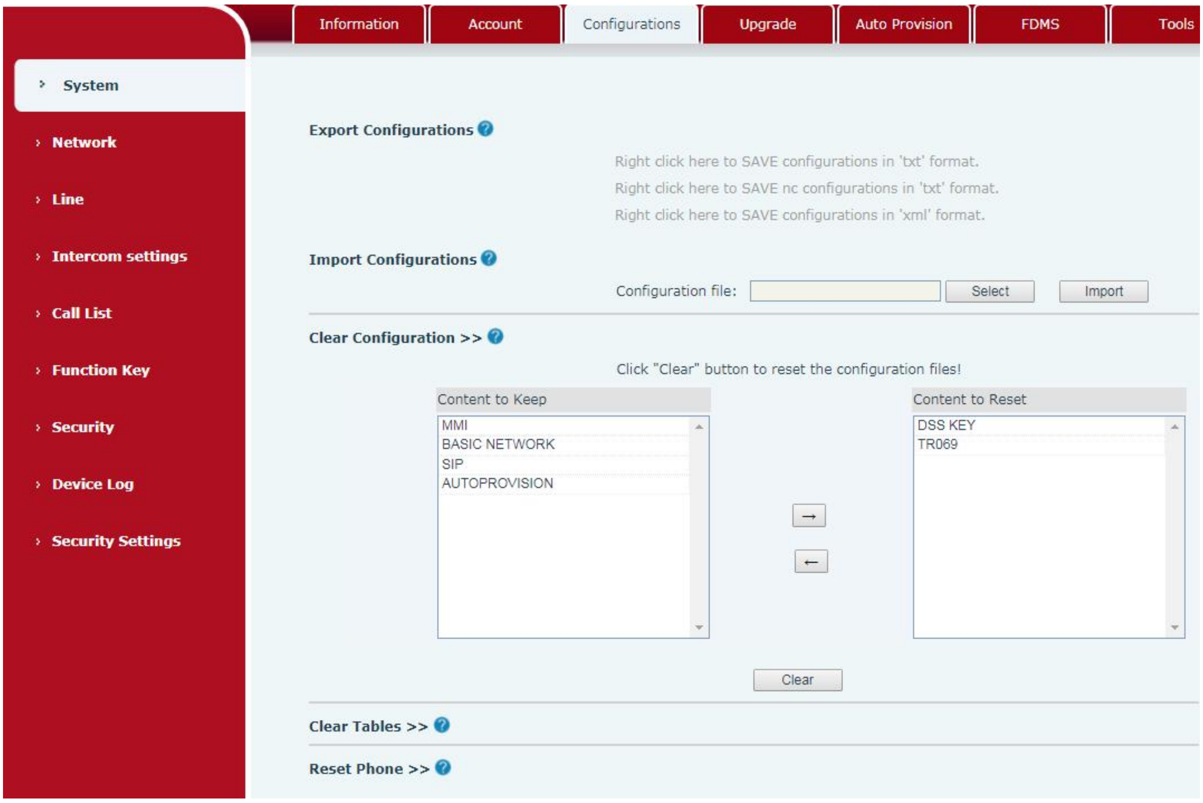
admin ▾ Delete Modify

Figure 12 - WEB Account

On this page, user can change the webpage login password.

Administration user can also add or delete users, manage users, set permissions and passwords for new users.

9.4 System >> Configurations



Export Configurations

Right click here to SAVE configurations in 'txt' format.

Right click here to SAVE nc configurations in 'txt' format.

Right click here to SAVE configurations in 'xml' format.

Import Configurations

Configuration file: Select Import

Clear Configuration >>

Click "Clear" button to reset the configuration files!

Content to Keep	Content to Reset
MMI BASIC NETWORK SIP AUTOPROVISION	DSS KEY TR069

→ ←

Clear

Clear Tables >>

Reset Phone >>

Figure 13 - System Setting

In this page, administration user can view, export, or import the configuration file, or restore the device to factory settings.

- **Export Configurations**

Right click to download the device's configuration file to your PC, the file format is “.txt”. (Notice: only administrator user can export the configuration file.)

- **Import Configurations**

Import the configuration file of settings. The device will restart automatically after successful importation, and the configuration will take effect after a restart

- **Clear Configurations**

Select the module in the configuration file to clear.

SIP: SIP account configuration

AUTOPROVISION: Provision related configuration

TR069:TR069 related configuration

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

DSS Key: DSS key configuration

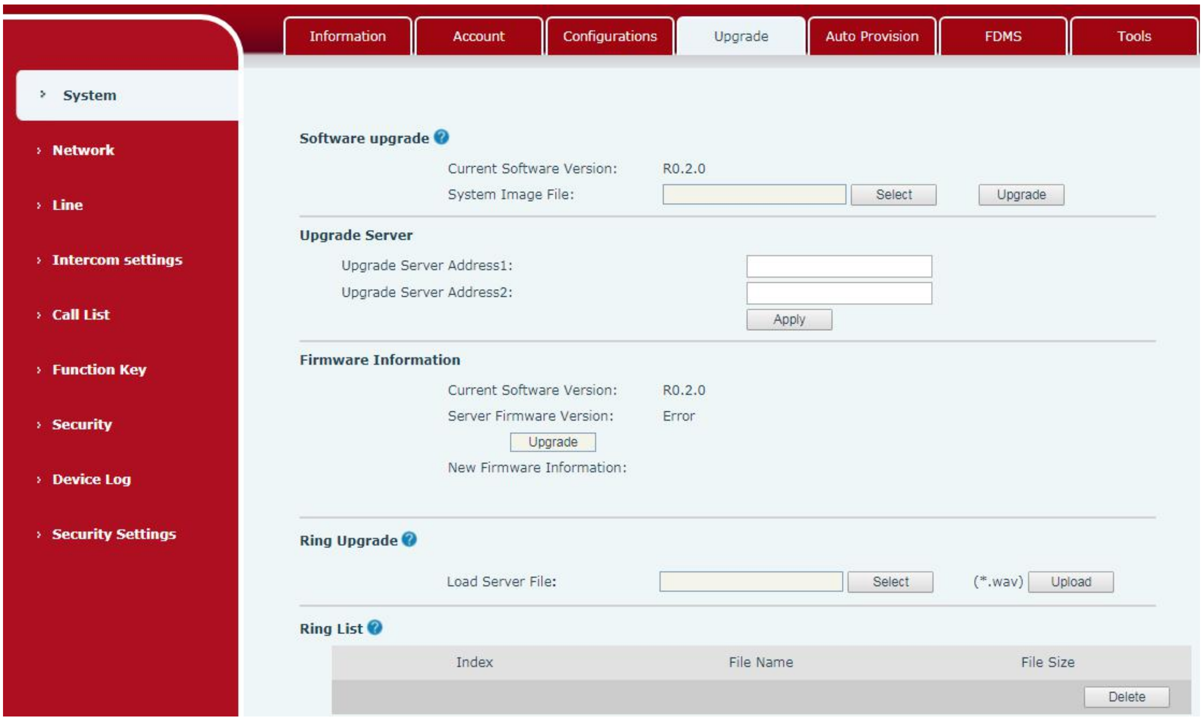
- **Clear Tables**

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

- **Reset Phone**

The device data will be cleared, including configurations and database tables.

9.5 System >> Upgrade



Software upgrade

Current Software Version: R0.2.0

System Image File:

Upgrade Server

Upgrade Server Address1:

Upgrade Server Address2:

Firmware Information

Current Software Version: R0.2.0

Server Firmware Version: Error

New Firmware Information:

Ring Upgrade

Load Server File: (*.wav)

Ring List

Index	File Name	File Size

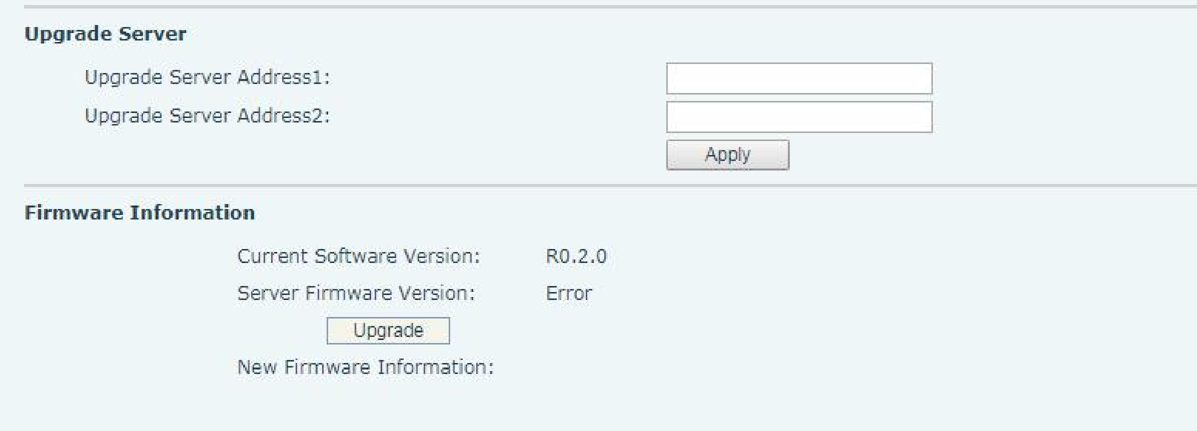
Figure 14 - Upgrade

In this page, user can upgrade the software for the device. After the upgrade, the device will automatically restart and update to the new version.

Click select to select the software file from local PC and then click upgrade to start upgrading.

Online upgrade:

Online Firmware update is when a device sends an HTTP request to a server, the server replies with a corresponding description file or 404 or timeout. After device gets the reply, it analyzes the version description file and prompts the user whether to upgrade the new version or not.



Upgrade Server

Upgrade Server Address1:

Upgrade Server Address2:

Firmware Information

Current Software Version: R0.2.0

Server Firmware Version: Error

New Firmware Information:

Figure 15 - Online Upgrade

Table 7 - Online Upgrade

Parameters	**Description**
Upgrade Server	
Upgrade Server Address1	Fill in the available primary upgrade server (HTTP server) address.
Upgrade Server Address2	Fill in the available backup upgrade server (HTTP server) address, when the primary server is not available, device will send the request to backup server.
Firmware Information	
Current Software Version	Displays the current device software version information.
Server Firmware Version	Displays the server software version information.
[**Upgrade**] button	When there is a corresponding TXT file and firmware file on the server side, the "upgrade" button changes from gray to available state. Click "upgrade" to choose whether to upgrade or not.
New Firmware Information	When the server side has the corresponding TXT file and firmware file, the new firmware information will display the version information in TXT.

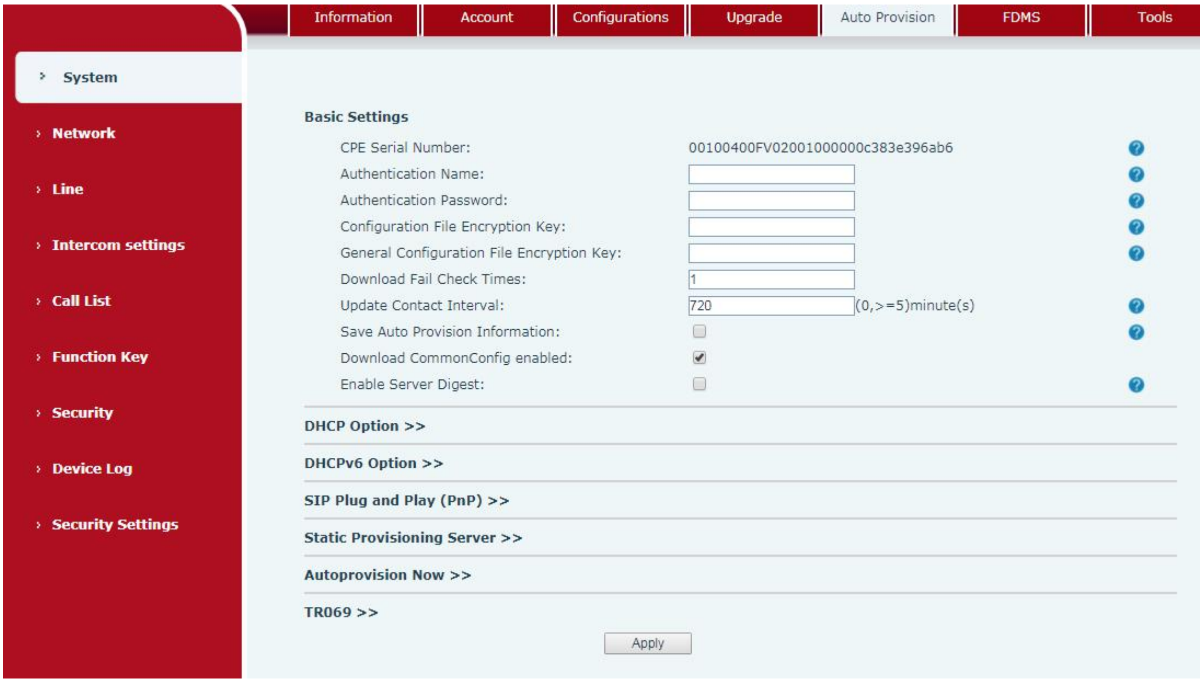
- The device requests TXT file to the server, the TXT file named with vendor_model_hw1_0.txt. Hw is followed by the hardware version information. All spaces in file names are changed to underlined.

- The URL requested by the device is HTTP:// server address /, and both the new version and the requested file are placed in the download directory of the HTTP server.
- The TXT file format must be UTF-8.
- Vendor_model_hw1_0.txt file format is as following:

```
Version=1.6.3      # software Version
Firmware=xxx/xxx.z      #xxx.z or http:// server IP: port/directory /xxx.z
BuildTime = 2018.09.11 20:00
Info = TXT | XML
Xxxxx
Xxxxx
Xxxxx
Xxxxx
```

9.6 System >> Auto Provision

Webpage: Login device's webpage and go to [**System**] >> [**Auto provision**].



Basic Settings

CPE Serial Number:	00100400FV02001000000c383e396ab6
Authentication Name:	
Authentication Password:	
Configuration File Encryption Key:	
General Configuration File Encryption Key:	
Download Fail Check Times:	1
Update Contact Interval:	720 (0,>=5)minute(s)
Save Auto Provision Information:	□
Download CommonConfig enabled:	☑
Enable Server Digest:	□

DHCP Option >>

DHCPv6 Option >>

SIP Plug and Play (PnP) >>

Static Provisioning Server >>

Autoprovision Now >>

TR069 >>

Figure 16 - Auto Provision

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

PNP>DHCP>TR069> Static Provisioning

Transferring protocol: FTP/ TFTP/HTTP/HTTPS

More details, please refer to **Fanvil Auto Provision**.

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

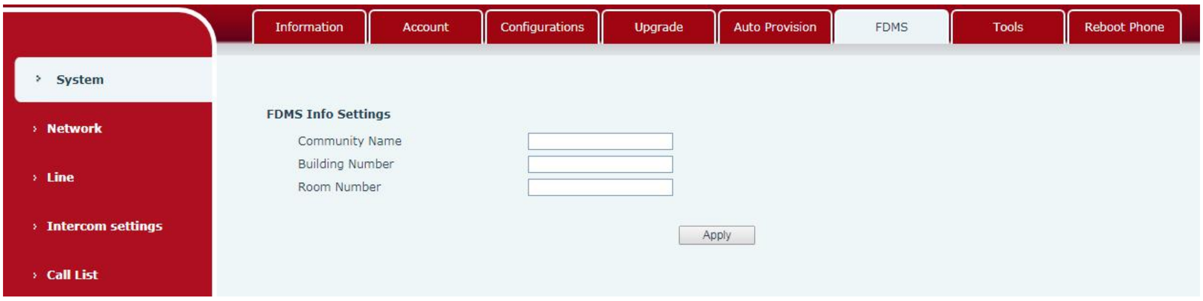
Table 8 - Auto Provision

Parameters	**Description**
Basic settings	
Current Configuration Version	Show the current config file's version. If the device confirm the downloaded .CFG configuration file is same with the one it uses, the device won't perform auto provision. Or if the device is matching the configuration file's context via Digest method, when the configuration file of server is modified or configuration file of device is different with server's, the device will perform provision.
General Configuration Version	Show the common config file's version. If the device confirm the downloaded .CFG configuration file is same with the one it uses, the device won't perform auto provision. Or if the device is matching the configuration file's context via Digest method, when the configuration file of server is modified or configuration file of device is different with server's, the device will perform provision.
CPE Serial Number	Serial number of the equipment
Authentication Name	Configure FTP server's username, TFTP server does not require this option. When use FTP server, device uses anonymous as authentication name if user leave this option blank.
Authentication Password	Corresponding password for FTP server.
Configuration File Encryption Key	Encryption key for the encrypted configuration file.
General Configuration File Encryption Key	Encryption key for encrypted common configuration file.
Save Auto Provision Information	Configure whether to save the auto provision information or not.
Download Fail Check Times	The default value is 5. When device fails to download configuration file, it will retry until it counts to fail check times.
Enable Server Digest	When the feature is enabled, if the configuration file of server is changed, or device's configuration is different from server's, the

	device will download and update.
DHCP Option	
Option Value	The equipment supports configuration from Option 43, Option 66, or a Custom DHCP option. User can select any of the three method to perform auto provision, by default, the option is disabled.
Custom Option Value	The custom option value should be same with the one of server, it can be any number 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, i10 series device will send a SIP SUBSCRIBE message with broadcast method. Any server which can support the feature will respond and send a Notify with URL to phone. The device could get the configuration file with the URL.
Server Address	Input SIP PnP server address.
Server Port	Input SIP PnP server port.
Transport Protocol	Select SIP PnP protocol, TCP or UDP.
Update Interval	Configure SIP 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, for example ftp.domain.com. And the device supports to access server subdirectory, 192.168.1.1/ftp/config or ftp.domain.com/ftp/config, it means server address is 192.168.1.1 or ftp.domain.com, file path is /ftp/config/.
Configuration File Name	Input the configuration file name. If it is empty, i10 series device will request the file which is named as its MAC address.
Protocol Type	Select transportation protocol type, i10 series support FTP/TFTP/HTTP and HTTPS
Update Interval	Set configuration file update interval time. As default it is 1, which means i10 series will check the update every 1 hour.
Update Mode	Select Provision Mode: 1. Disabled. 2. Update after reboot. 3. Update at a time interval.
TR069	
Enable TR069	Select it to enable TR069.
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting to TR069 server successfully.

ACS Server Type	There are 4 kinds of ACS server: China Unicom, China Telecom, common and esight.
ACS Server URL	Input ACS server address.
ACS User	Input ACS server username.
ACS Password	Input ACS server password.
TR069 Auto Login	Enable/Disable TR069 Auto Login. If TR069 auto login is enabled, every time user reboot the device, it will use the previous correct username or password to connect ACS server instead of asking TR069 username or password.
STUN server address	Enter the STUN address
Enable the STUN	Select it to enable STUN.

9.7 System >> FDMS



FDMS Info Settings

Community Name

Building Number

Room Number

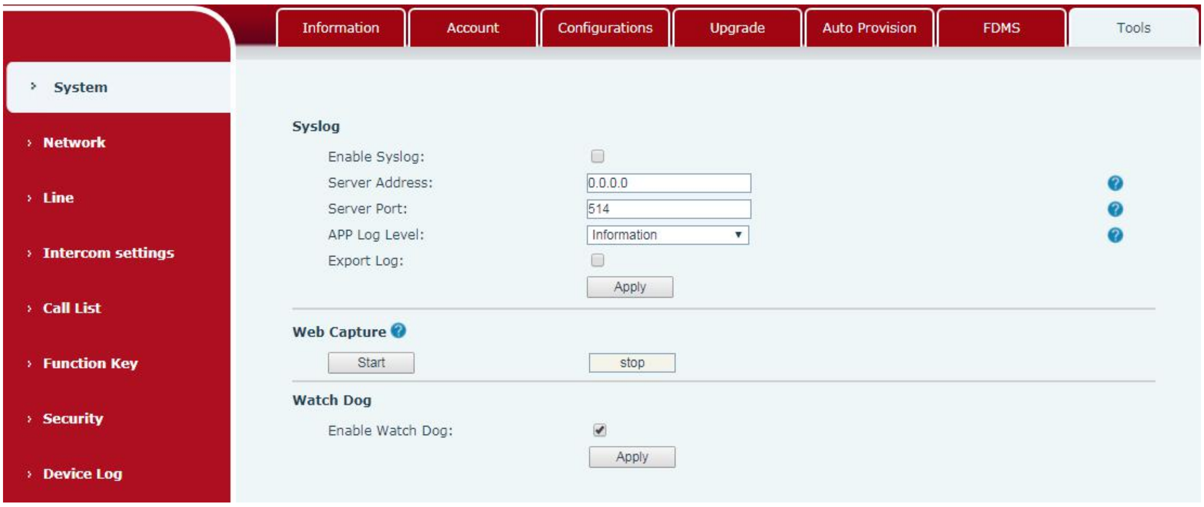
Figure 17 - FDMS

Table 9 - FDMS

FDMS information Settings	
Community Name	Name of equipment installation community.
Building Number	Name of equipment installation building.
Room Number	Name of equipment installation room.

9.8 System >> Tools

This page provides users the tools to check the problems.



Syslog

Enable Syslog: □

Server Address: 0.0.0.0

Server Port: 514

APP Log Level: Information

Export Log: □

Web Capture

Watch Dog

Enable Watch Dog: ☑

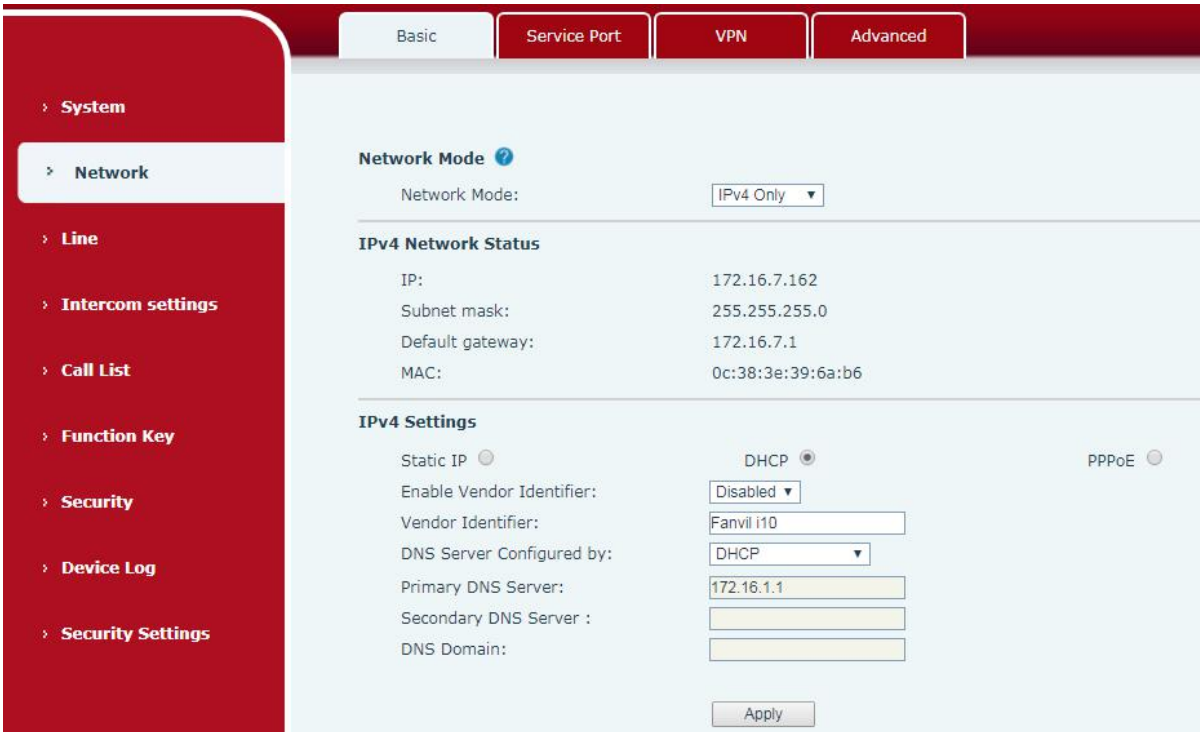
Figure 18 - Tools

Syslog： When the user open syslog and set syslog server address, the log information of the device will be recorded in the syslog server during operation. If there is any problem, send the logs to Fanvil support team to analyze.

For other details, please refer to 10 Trouble Shooting.

9.9 Network >> Basic

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



Network Mode

Network Mode: IPv4 Only

IPv4 Network Status

IP: 172.16.7.162

Subnet mask: 255.255.255.0

Default gateway: 172.16.7.1

MAC: 0c:38:3e:39:6a:b6

IPv4 Settings

Static IP ○ DHCP ● PPPoE ○

Enable Vendor Identifier: Disabled

Vendor Identifier: Fanvil i10

DNS Server Configured by: DHCP

Primary DNS Server: 172.16.1.1

Secondary DNS Server :

DNS Domain:

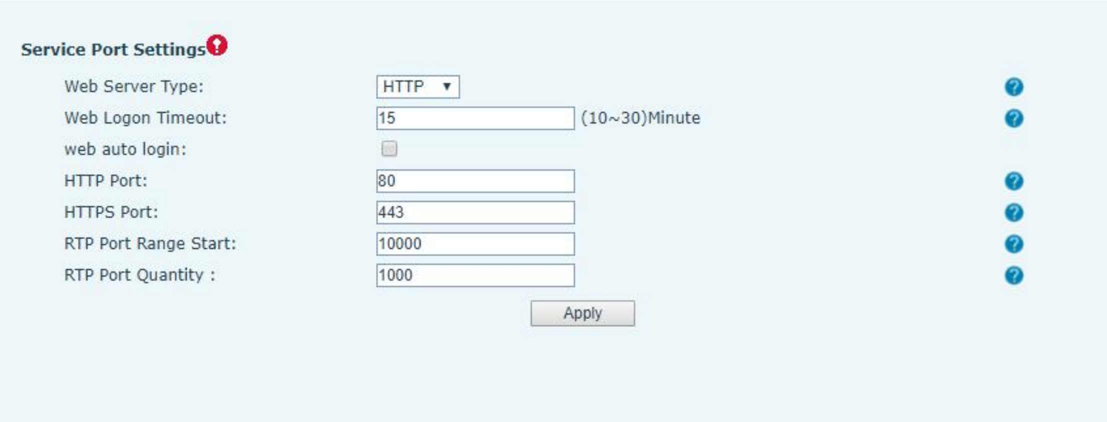
Figure 19 - Network Basic Settings

Table 10 - Network Basic Setting

Parameters	Description
Network Mode	IPv4 only、IPv6 only、IPv4&IPv6
Network Status	
IP	The current IP address of the equipment.
Subnet mask	The current Subnet Mask of the device.
Default gateway	The current Gateway IP address.
MAC	The MAC address of the equipment.
Settings	
Select the appropriate network mode. The equipment supports three kinds of network mode:	
Static IP	Network parameters must be entered manually and will not change. All parameters are provided by the ISP. Please contact ISP or network administrator for help if you do not know these information.
DHCP	Network parameters are provided automatically by a DHCP server.
PPPoE	Account and Password must be input manually. These are provided by your ISP.
Enable Vendor Identifier	When enabled, you will see the vendor identifier information in the DHCP option60 field
Vendor Identifier	Support for customization. When vendor identity is enabled, you will see the vendor identifier information in the DHCP option60 field
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.
Notice： 1）After set the parameters, click【Apply】to make settings take effect. 2）If you change the IP address, the current webpage will no longer respond, user should enter new IP address in URL to re-connect and re-login to the device's webpage.	

9.10 Network >> Service Port

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



Service Port Settings

Web Server Type: HTTP

Web Logon Timeout: 15 (10~30)Minute

web auto login: □

HTTP Port: 80

HTTPS Port: 443

RTP Port Range Start: 10000

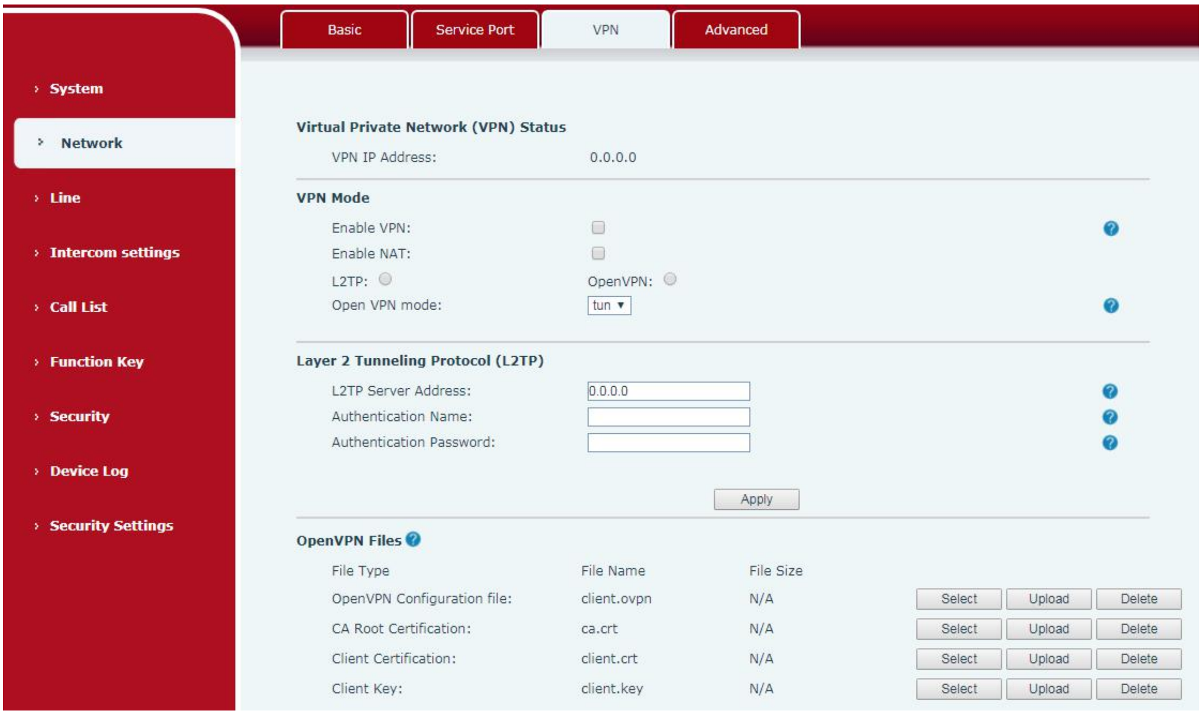
RTP Port Quantity : 1000

Figure 20 - Service Port

Table 11 - Service port

Parameter	Description
Web Server Type	Reboot the device to make settings take effective. i10 series supports two kinds of web login: HTTP and HTTPS.
Web Logon Timeout	Default value is 15 minutes, when login time expires, web login will exit automatically, user need to login again.
Web auto login	If enable web auto login, after web login exits, refresh the webpage to login, user does not need to input username and password.
HTTP Port	The default value is 80. If you want more secure system management, you can set other value. Such as :8080, webpage login URL is: HTTP://ip:8080
HTTPS Port	The default is 443, using method is similar to HTTP port.
RTP Port Range Start	The value range is 1025 to 65535. The value of RTP port starts from the initial value, each call, the value of voice and video port will added 2.
RTP Port Quantity	Number of calls.

9.11 Network >> VPN



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:

Authentication Password:

OpenVPN Files

File Type	File Name	File Size			
OpenVPN Configuration file:	client.ovpn	N/A	Select	Upload	Delete
CA Root Certification:	ca.crt	N/A	Select	Upload	Delete
Client Certification:	client.crt	N/A	Select	Upload	Delete
Client Key:	client.key	N/A	Select	Upload	Delete

Figure 21 - VPN

Virtual Private Network (VPN) is a technology to allow the 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 activating 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 webpage.

■ **L2TP**

NOTICE: The device only supports non-encrypted basic authentication and non-encrypted data tunneling. For users who need data encryption, please use OpenVPN.

To establish a L2TP connection, the user should log in to the device webpage, go to 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 corresponding option. Press "Apply" to save changes and 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 option. 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 with the VPN server automatically every time it boots up, unless user disable VPN. Sometimes, if the VPN connection does not establish immediately, user may try to reboot the device and check again.

- **OpenVPN**

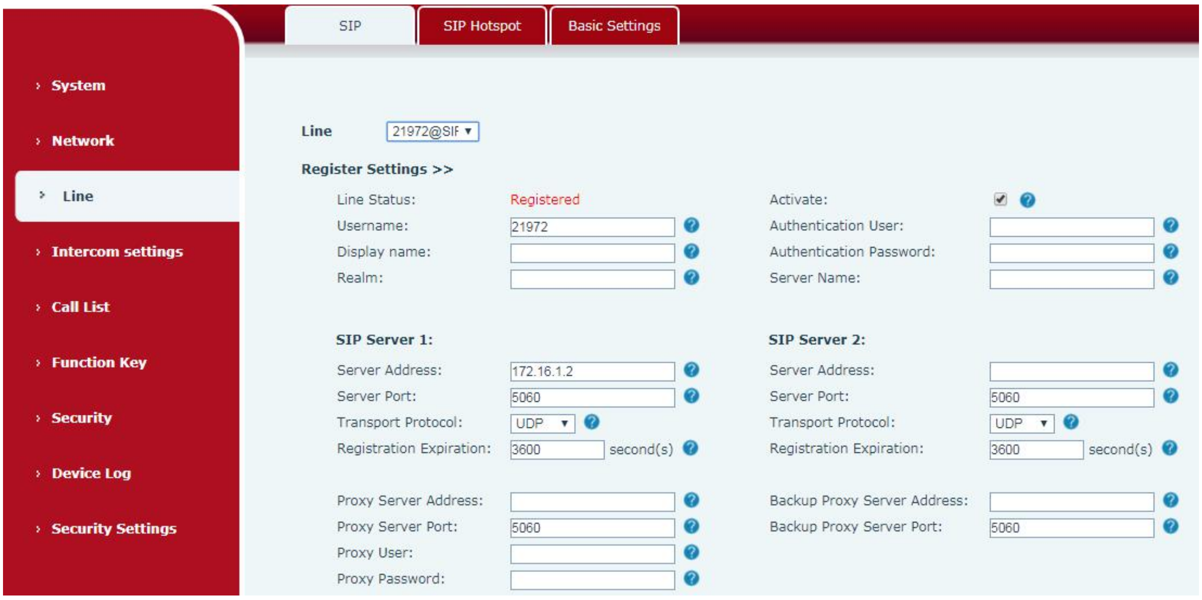
To establish an OpenVPN connection, user should get the following authentication and configuration files from the OpenVPN service provider and name them as the following,

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

Select OpenVPN files and then click upload to upload these files to the device in the webpage [Network] >> [VPN]. 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 system boots up unless the user disable it manually.

9.12 Line >> SIP

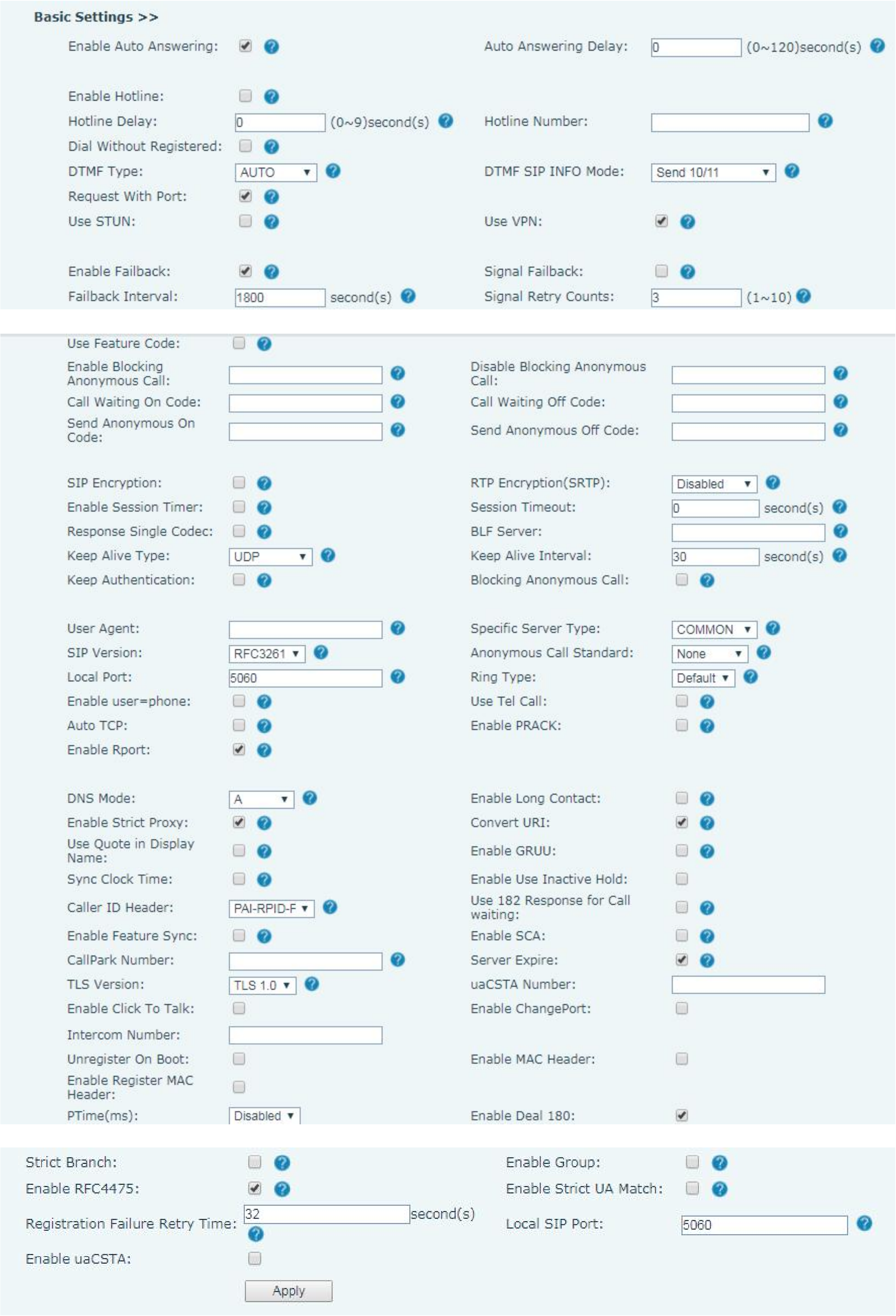


Line 21972@SIF ▾

Register Settings >>

Line Status:	Registered	Activate:	☑
Username:	21972	Authentication User:	
Display name:		Authentication Password:	
Realm:		Server Name:	

SIP Server 1:		SIP Server 2:	
Server Address:	172.16.1.2	Server Address:	
Server Port:	5060	Server Port:	5060
Transport Protocol:	UDP ▾	Transport Protocol:	UDP ▾
Registration Expiration:	3600 second(s)	Registration Expiration:	3600 second(s)
Proxy Server Address:		Backup Proxy Server Address:	
Proxy Server Port:	5060	Backup Proxy Server Port:	5060
Proxy User:			
Proxy Password:			



Basic Settings >>

Enable Auto Answering:	☑	Auto Answering Delay:	0 (0~120)second(s)
Enable Hotline:	□		
Hotline Delay:	0 (0~9)second(s)	Hotline Number:	
Dial Without Registered:	□		
DTMF Type:	AUTO	DTMF SIP INFO Mode:	Send 10/11
Request With Port:	☑		
Use STUN:	□	Use VPN:	☑
Enable Failback:	☑	Signal Failback:	□
Failback Interval:	1800 second(s)	Signal Retry Counts:	3 (1~10)

Use Feature Code:	□		
Enable Blocking Anonymous Call:		Disable Blocking Anonymous Call:	
Call Waiting On Code:		Call Waiting Off Code:	
Send Anonymous On Code:		Send Anonymous Off Code:	
SIP Encryption:	□	RTP Encryption(SRTP):	Disabled
Enable Session Timer:	□	Session Timeout:	0 second(s)
Response Single Codec:	□	BLF Server:	
Keep Alive Type:	UDP	Keep Alive Interval:	30 second(s)
Keep Authentication:	□	Blocking Anonymous Call:	□
User Agent:		Specific Server Type:	COMMON
SIP Version:	RFC3261	Anonymous Call Standard:	None
Local Port:	5060	Ring Type:	Default
Enable user=phone:	□	Use Tel Call:	□
Auto TCP:	□	Enable PRACK:	□
Enable Rport:	☑		
DNS Mode:	A	Enable Long Contact:	□
Enable Strict Proxy:	☑	Convert URI:	☑
Use Quote in Display Name:	□	Enable GRUU:	□
Sync Clock Time:	□	Enable Use Inactive Hold:	□
Caller ID Header:	PAI-RPID-F	Use 182 Response for Call waiting:	□
Enable Feature Sync:	□	Enable SCA:	□
CallPark Number:		Server Expire:	☑
TLS Version:	TLS 1.0	uaCSTA Number:	
Enable Click To Talk:	□	Enable ChangePort:	□
Intercom Number:			
Unregister On Boot:	□	Enable MAC Header:	□
Enable Register MAC Header:	□		
PTime(ms):	Disabled	Enable Deal 180:	☑

Strict Branch:	□	Enable Group:	□
Enable RFC4475:	☑	Enable Strict UA Match:	□
Registration Failure Retry Time:	32 second(s)	Local SIP Port:	5060
Enable uaCSTA:	□		

Figure 22 - SIP

Table 111 - SIP

Parameters	Description
Register Settings	
Line Status	Display the current line status. To get the latest line status, user has to refresh the page manually.
Activate	Whether to activate the line or not.
Username	Enter the username of the service account.
Authentication User	Enter the authentication user name of the service account.
Display Name	Enter the display name which will be sent in a call request.
Authentication Password	Enter the authentication password of the service account.
Realm	Enter the SIP domain provided by the service provider.
Server Name	Input server name.
SIP Server 1	
Server Address	Enter the IP or FQDN address of the SIP server
Server Port	Enter the SIP server port, default is 5060
Transport Protocol	Set up the SIP transportation protocol: TCP or UDP or TLS.
Registration Expiration	Set SIP registration expiration time.
SIP Server 2	
Server Address	Enter the IP or FQDN address of the SIP server
Server Port	Enter the SIP server port, default is 5060
Transport Protocol	Set up the SIP transportation protocol: TCP or UDP or TLS.
Registration Expiration	Set SIP registration expiration time.
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 username.
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 time for incoming call before the system automatically answers it.
Enable Hotline	Enable hotline configuration, the device will dial the specific number immediately once audio channel is opened.
Hotline Delay	Set the delay time for hotline before the call sends out.
Hotline Number	Set the hotline dialing number

Dial Without Registered	Enable call out without registration.
DTMF Type	Set the DTMF type for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'
Use VPN	Set the line to use VPN restrict route
Use STUN	Set the line to use STUN for NAT traversal
Enable Failback	Whether to switch to the primary server when it is available.
Failback Interval	The time interval of detecting the availability of the main Proxy using Register message.
Signal Failback	When there are multiple proxy, whether to allow the invite/register request to execute failback or not.
Signal Retry Counts	When there are multiple proxy, the attempt times that the SIP Request considers proxy is unavailable.
Codecs Settings	Set the priority and availability of the codecs by adding or removing 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 device, the device will send feature code to the server by dialing the number specified in each feature code field.
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, and SIP transmission will be encrypted.
RTP Encryption	Enable RTP encryption, and RTP transmission will be encrypted.
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if there is no new session timer event update received after the timeout period.
Session Timeout	Set the session timer timeout period.
Response Single Codec	If enable this option, the device will use single codec to respond to incoming call request.
BLF Server	Input BLF server address. Ordinary BLF application is that device sends subscription message to SIP server. If your SIP server does not support subscription, please input BLF server address to

	separate SIP registration server and BLF server.
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT opened.
Keep Alive Interval	Set the keep alive packet transmitting interval.
Keep Authentication	Keep the previous authentication parameters.
Blocking Anonymous Call	Reject any incoming call without presenting caller ID.
User Agent	Set the user agent, the default value is device 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 for anonymous call.
Local Port	Set the local port.
Ring Type	Set the ring tone type for the line.
Enable user=phone	In SIP invite message, there is user=phone field.
Use Tel Call	Enable or disable use tel call.
Auto TCP	Using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes.
Enable Rport	Set the line to add Rport in SIP headers.
Enable PRACK	Set the line to support PRACK SIP message.
DNS Mode	Select DNS mode, options are A, SRV and NAPTR.
Enable Long Contact	Enable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.
Enable Strict Proxy	This is used for matching special server. When the i10 series receives packets from the server， it will reply with the source IP address, not the address in via field.
Convert URI	Whether to enable convert URI or not.
Use Quote in Display Name	Whether to add quote in display name, i.e. “Fanvil” vs Fanvil.
Enable GRUU	Enable Globally Routable User-Agent URI (GRUU) or not.
Sync Clock Time	Time Sync with server.
Enable Inactive Hold	With inactive hold enabled, you can see SDP is inactive in the SDP packet.
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.
Enable Feature Sync	Enable or disable Feature Sync with server.
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
CallPark Number	Set the CallPark number.

Server Expire	Set the timeout of using the server.
TLS Version	Choose TLS Version.
uaCSTA Number	Set uaCSTA Number.
Enable Click To Talk	This is used to match special server, click to call out directly after enable this option.
Enable Change port	Whether to enable change port or not.
Intercom Number	Set intercom number.
Unregister On Boot	Whether to enable logout function.
Enable MAC Header	Whether to enable MAC header. When enable, there is MAC information in SIP packet and user agent when register
BLF Dialog Strict Match	Whether to enable accurate matching of BLF sessions.
PTime(ms)	Set whether to bring ptime field, by default it is no.
Enable Deal 180	Enable: after receives183+ SDP, device will play ivr; and after receives180, device will play local tone. Disable: after receives 183+ SDP, device will play ivr. After receives180, device does not play local tone.
SIP Global Settings	
Strict Branch	Enable or disable this to strictly match the Branch field.
Enable Group	Enable or disable SIP group server function as server backup.
Enable RFC4475	Enable or disable RFC4475.
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Set the registration failure retry time.
Local SIP Port	Modify the device SIP port.
Enable uaCSTA	Set to enable the uaCSTA function.

9.13 Line >> SIP Hotspot

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

Please check 8.3 Hotspot for more details.

9.14 Line >> Basic Settings

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

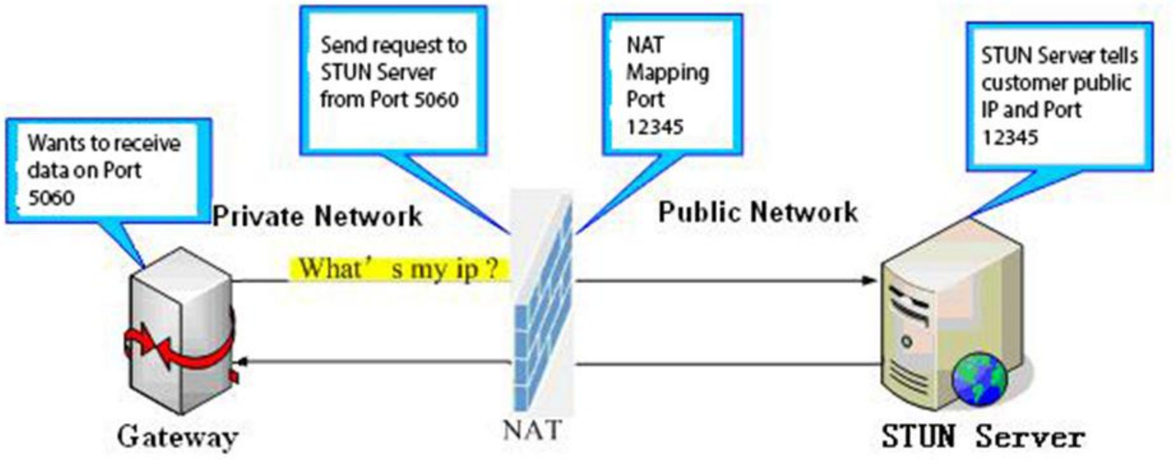
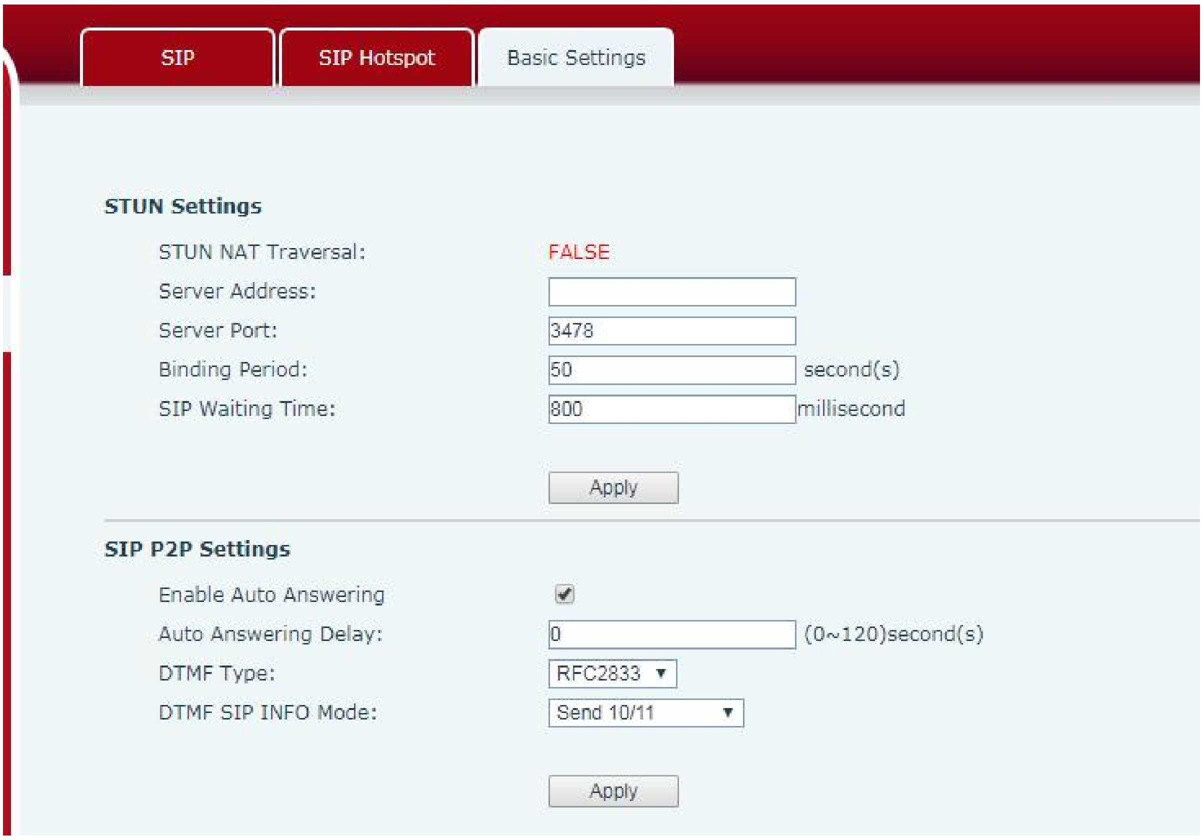

Wants to receive data on Port 5060
Send request to STUN Server from Port 5060
NAT Mapping Port 12345
STUN Server tells customer public IP and Port 12345
Private Network
Public Network
What' s my ip ?
Gateway
NAT
STUN Server

Figure 23 - Network Basic



STUN Settings

STUN NAT Traversal:	FALSE
Server Address:	
Server Port:	3478
Binding Period:	50 second(s)
SIP Waiting Time:	800 millisecond

Apply

SIP P2P Settings

Enable Auto Answering	☑
Auto Answering Delay:	0 (0~120)second(s)
DTMF Type:	RFC2833 ▼
DTMF SIP INFO Mode:	Send 10/11 ▼

Apply

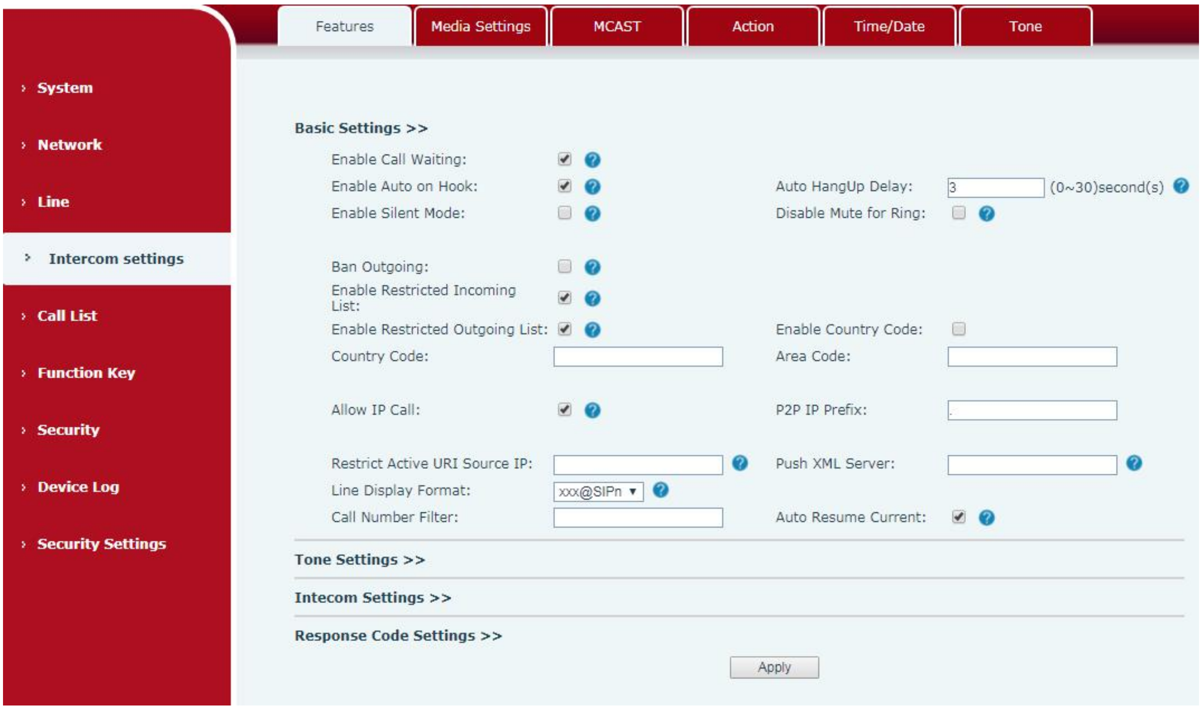
Figure 24 - Line Basic Setting

Table 12 - Line Basic Setting

Parameters	Description
STUN Settings	

Server Address	Input the STUN server address.
Server Port	Input the STUN server port, default is 3478.
Binding Period	Set the STUN binding period which can be used to keep the NAT open.
SIP Waiting Time	Set the timeout of STUN binding before sending SIP messages.
SIP P2P Settings	
Enable Auto Answering	Enable timeout to automatically answer IP calls
Auto Answering Delay	Automatic answer timeout setting
DTMF Type	Set the DTMF type of the line.
DTMF SIP INFO Mode	Set up SIP INFO mode to send '*' and '#' or '10' and '11'

9.15 Intercom Setting >> Features



Basic Settings >>

Enable Call Waiting: ☑

Enable Auto on Hook: ☑ Auto HangUp Delay: 3 (0~30)second(s)

Enable Silent Mode: □ Disable Mute for Ring: □

Ban Outgoing: □

Enable Restricted Incoming List: ☑

Enable Restricted Outgoing List: ☑ Enable Country Code: □

Country Code: Area Code:

Allow IP Call: ☑ P2P IP Prefix: .

Restrict Active URI Source IP: Push XML Server:

Line Display Format: xxx@SIPn

Call Number Filter: Auto Resume Current: ☑

Tone Settings >>

Intecom Settings >>

Response Code Settings >>

Apply

Figure 25 - Intercom Setting

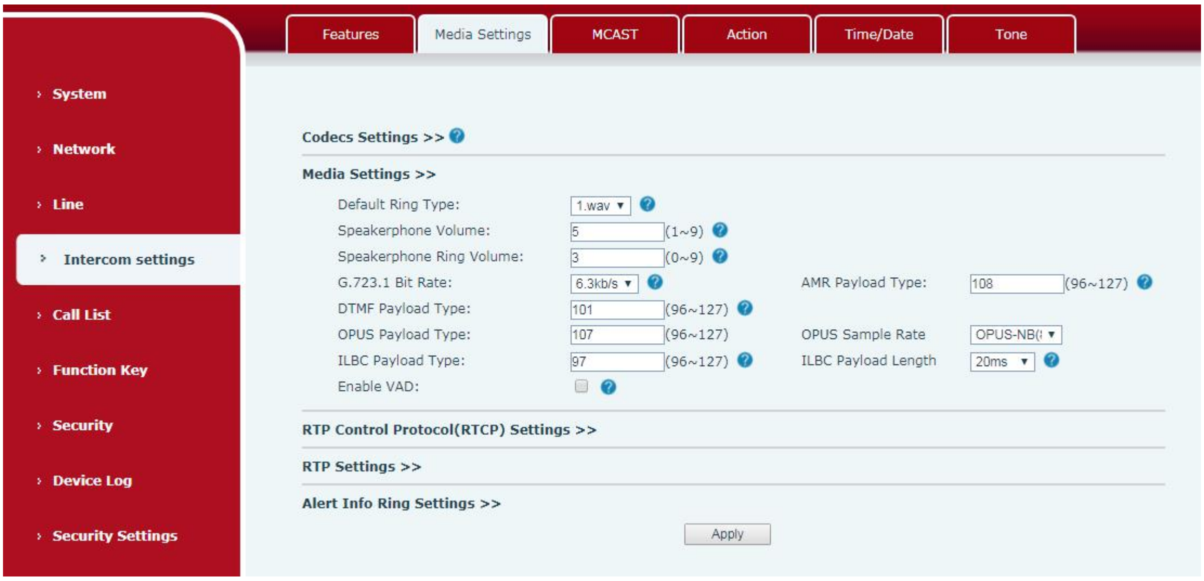
Table 13 - Intercom Setting

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. By default it is enabled.
Enable Auto Onhook	Enable auto onhook or not. If enable, the device will hang up the call and return to the idle status automatically.

Auto Onhook Time	Specify Auto Onhook time, the device will hang up and return to the idle automatically after Auto onhook time.
Enable Silent Mode	When enabled, the device is muted, there is no ringing when calls, you can use the volume keys and mute key to unmute.
Disable Mute for Ring	Disable the mute mode, if this option is clicked, mute button on device does not take effect.
Ban Outgoing	Enable or disable ban outgoing, if enable, the device can not dial out any number.
Enable Restricted Incoming List	Whether to enable restricted incoming call list.
Enable Restricted Outgoing List	Whether to enable the restricted outgoing list.
Enable Country Code	Whether to enable the country code.
Country Code	Fill in the country code.
Area Code	Fill in the area code.
Allow IP Call	If enabled, user can dial out with IP address.
P2P IP Prefix	Set prefix for point-to-point IP calls.
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP address. Notice: this function is usually used to manage device.
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.
Line Display Format	Custom line format：SIPn or SIPn:xxx or xxx@SIPn
Call Number Filter	Configure a special ampersand, the called number is 78-9, the ampersand will be filtered when device sends the call out.
Auto Resume Current	Automatically break HOLD if current call changes.
Tone Settings	
Enable Holding Tone	Whether to enable call holding tone.
Enable Call Waiting Tone	Whether to enable call waiting tone.
Play Dialing DTMF Tone	Play DTMF tone on the device when user presses a digits wehen dial the call, by default it is enabled.
Play Talking DTMF Tone	Play DTMF tone on the device when user presses a phone digits during taking, by default it is enabled.
Intercom Settings	
Enable Intercom	When intercom is enabled, the device will accept the incoming call which requests with a SIP header of Alert-Info automatically.
Enable Intercom Mute	Enable mute mode during the intercom call.
Enable Intercom Tone	If the incoming call is intercom call, the device plays the intercom

	tone.
Enable Intercom Barge	While enable intercom barge, the device will auto answer the intercom call during a call. If the current call is intercom call, the device will reject the second intercom call.
Response Code Settings	
Busy Response Code	Set the SIP response code when line is busy.
Reject Response Code	Set the SIP response code when device reject one call.

9.16 Intercom Setting >> Audio



Codecs Settings >>

Media Settings >>

Default Ring Type: 1.wav

Speakerphone Volume: 5 (1~9)

Speakerphone Ring Volume: 3 (0~9)

G.723.1 Bit Rate: 6.3kb/s

AMR Payload Type: 108 (96~127)

DTMF Payload Type: 101 (96~127)

OPUS Payload Type: 107 (96~127)

OPUS Sample Rate OPUS-NB(

ILBC Payload Type: 97 (96~127)

ILBC Payload Length 20ms

Enable VAD: □

RTP Control Protocol(RTCP) Settings >>

RTP Settings >>

Alert Info Ring Settings >>

Figure 26 - Media Setting

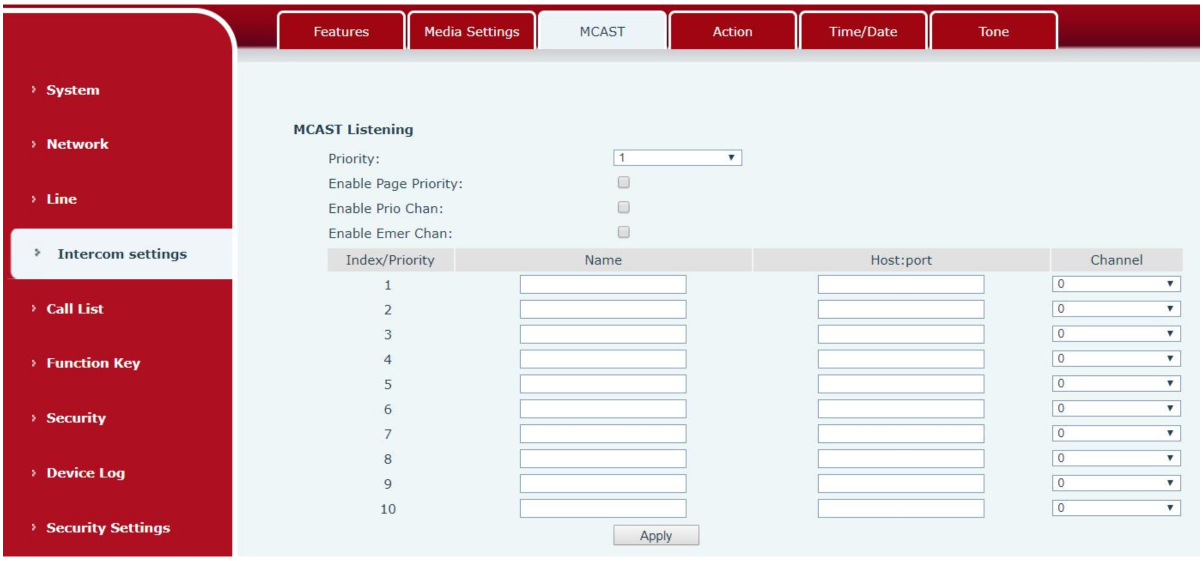
Table 14 - Media Setting

Parameter	**Description**
Codecs Settings	Select enable or disable voice codecs: G.711A/U, G.722, G.729AB, iLBC, opus.
Media Settings	
Default Ring Type	Configure default ringtones. If no special ringtone is set, the default ringtone will be used.
Speakerphone Volume	Set the speaker volume, value can be 1~9.
Speakerphone Ring Volume	Set the speaker ring volume, value can be 1~9.
G.723.1 Bit Rate	5.3kb/s or 6.3kb/s is available.
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
AMR Payload Type	Set AMR load type, range is 96~127.
Opus playload type	Set Opus load type, range is 96~127.

OPUS Sample Rate	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb (16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length.
Enable VAD	Whether to enable voice activity detection.
RTP Control Protocol(RTCP) Settings	
CNAME user	Set CNAME user
CNAME host	Set CNAME host
RTP Settings	
RTP keep alive	Hold the call and send the packet every 30s.
Alert Info Ring Settings	
Value	Set the value to specify the ring type.
Ring Type	Select ring type.

9.17 Intercom Setting >> MCAST

It is easy and convenient to use multicast function to send notice to each member of the multicast group 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, the device will receive multicast from the configured monitoring multicast address.



MCAST Listening

Priority: 1

Enable Page Priority: □

Enable Prio Chan: □

Enable Emer Chan: □

Index/Priority	Name	Host:port	Channel
1			0
2			0
3			0
4			0
5			0
6			0
7			0
8			0
9			0
10			0

Figure 27 - MCAST

9.18 Intercom Setting >> Action

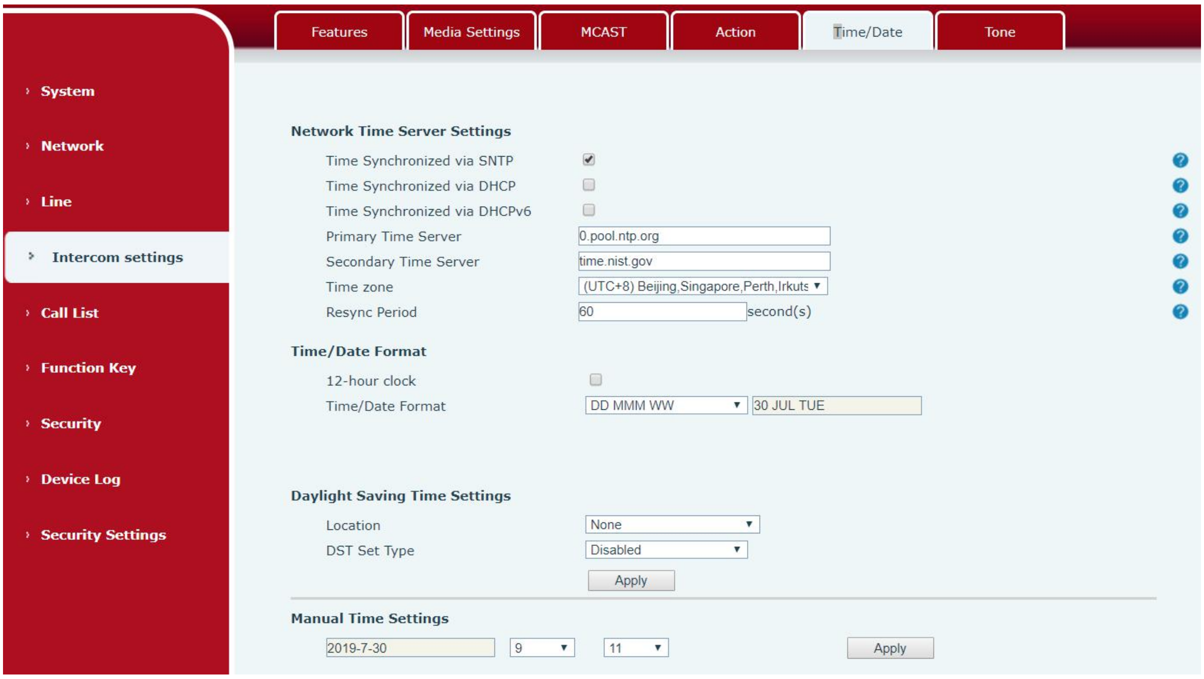
Action URL Event Settings
Set URL for the device to report its action to server. These actions are recorded and sent as xml files to the server. Sample format is http://InternalServer /FileName.xml. (Internal Server: The IP address of server; File Name: the device’s xml file used to report action.)

Table 15 - action URL

Notice: The operation URL is used by the IPPBX system to submit device events. Please refer to the details Fanvil Action URL。
http://www.fanvil.com/Uploads/Temp/download/20190122/5c46debfbde37.pdf

9.19 Intercom Setting >> Time/Date

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



Network Time Server Settings
Time Synchronized via SNTP ☑
Time Synchronized via DHCP □
Time Synchronized via DHCPv6 □
Primary Time Server 0.pool.ntp.org
Secondary Time Server time.nist.gov
Time zone (UTC+8) Beijing,Singapore,Perth,Irkuts
Resync Period 60 second(s)

Time/Date Format
12-hour clock □
Time/Date Format DD MMM WW 30 JUL TUE

Daylight Saving Time Settings
Location None
DST Set Type Disabled

Manual Time Settings
2019-7-30 9 11

Figure 28 - time/date

Table 16 - time/date

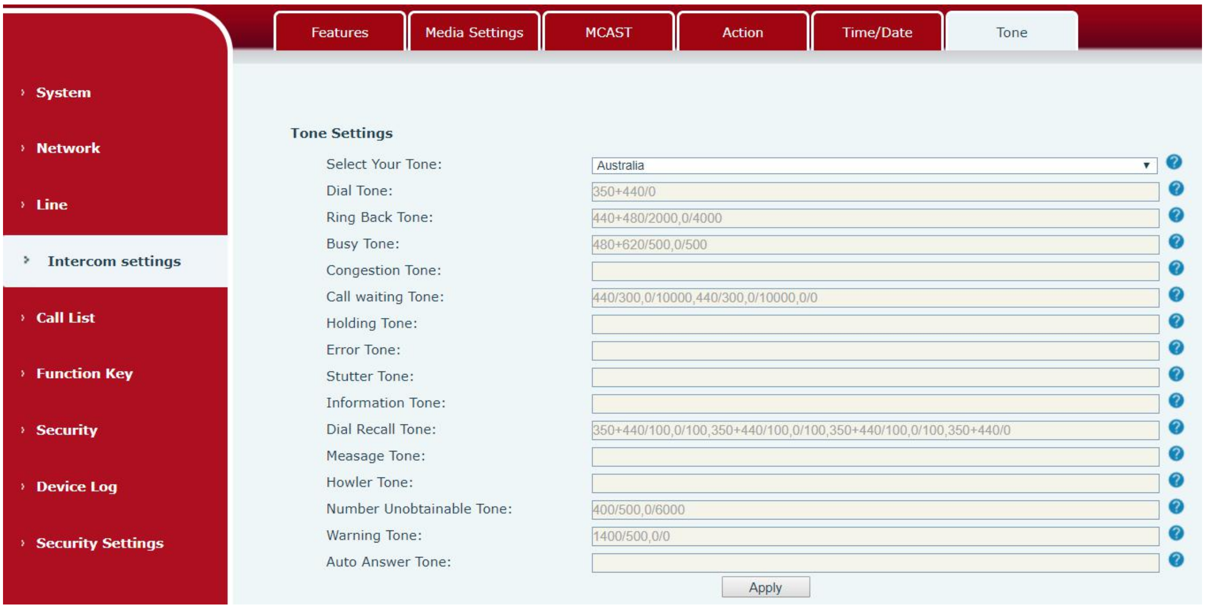
Parameter	Description
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol.
Time Synchronized via DHCP	Enable time-sync through DHCP protocol.
Time Synchronized via	Enable time-sync through DHCPv6 protocol.

DHCPv6	
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 interval of re-synchronization with time server
Time/Date Format	
12-hour clock	Enable or disable 12-hour clock.
Time/Date Format	Set time/date format.
Daylight Saving Time Settings	
Location	Select the user's time zone specific area
DST Set Type	Select DST type, and set the DST rules.
Fixed Type	Select DST fixed type.
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	
Manual Time Settings	Set time manually, please disable SNTP service first.

9.20 Intercom settings >> Tone

User can set device’s tone in this page.

You can select the corresponding country and use the settings directly, or select custom and set the tone manually.



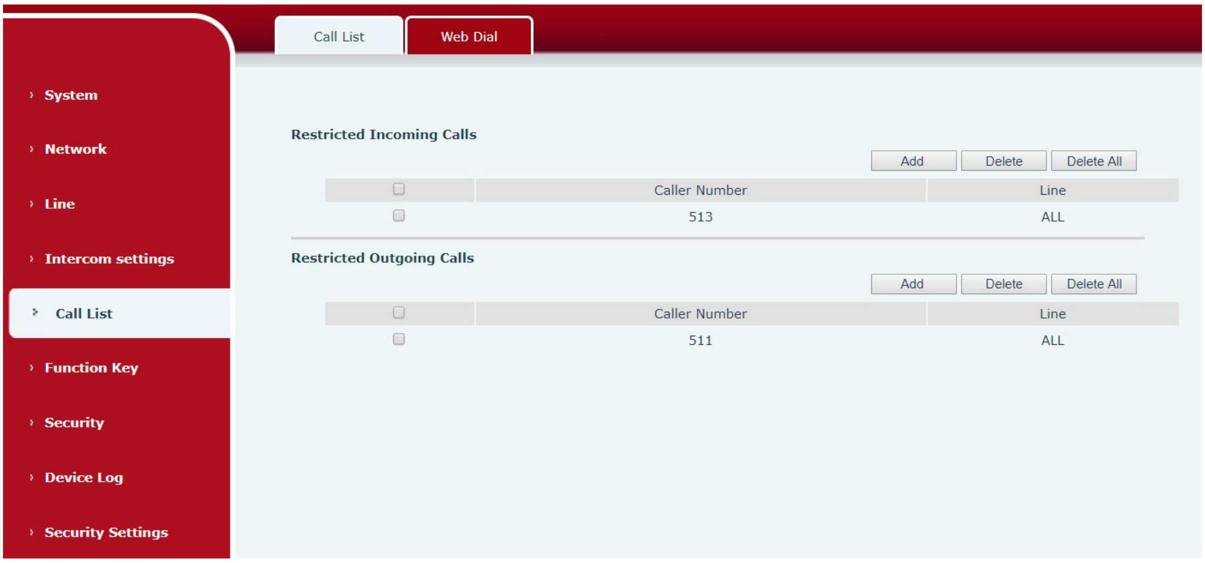
Tone Settings

Select Your Tone:	Australia
Dial Tone:	350+440/0
Ring Back Tone:	440+480/2000,0/4000
Busy Tone:	480+620/500,0/500
Congestion Tone:	
Call waiting Tone:	440/300,0/10000,440/300,0/10000,0/0
Holding Tone:	
Error Tone:	
Stutter Tone:	
Information Tone:	
Dial Recall Tone:	350+440/100,0/100,350+440/100,0/100,350+440/100,0/100,350+440/0
Measage Tone:	
Howler Tone:	
Number Unobtainable Tone:	400/500,0/6000
Warning Tone:	1400/500,0/0
Auto Answer Tone:	

Figure 29 - Tone

9.21 Call List >> Call List

User can set restricted incoming calls list and restricted outgoing calls list in this page.



Restricted Incoming Calls

□	Caller Number	Line
□	513	ALL

Restricted Outgoing Calls

□	Caller Number	Line
□	511	ALL

Figure 30 - Call List

- Restricted Incoming Calls:

The function is same with blacklist. Add the numbers in restricted incoming calls list, the device will reject all the calls from these blacklist numbers, unless user deletes the numbers from the list.

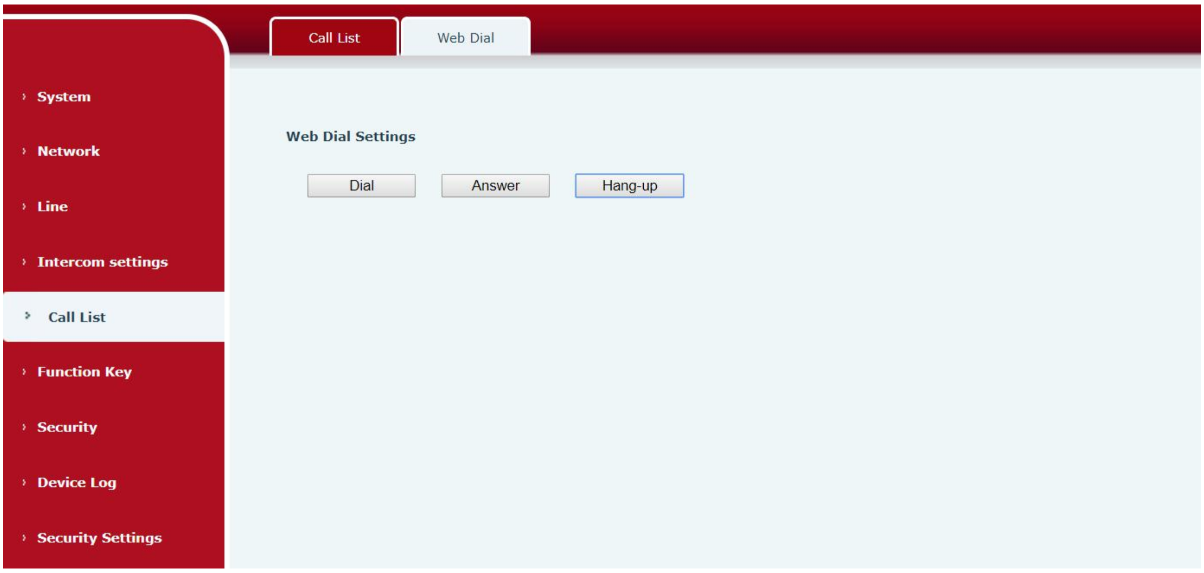
User can add both numbers and prefix in the restricted incoming calls list, the i10 series device will reject all the calls from the blacklist numbers or calls from numbers with blacklist prefix.

- Restricted Outgoing Calls:

Add numbers to restricted outgoing calls list, the i10 series device will end the calling when user dial these numbers, unless user removes the numbers from the list.

9.22 Web Dial

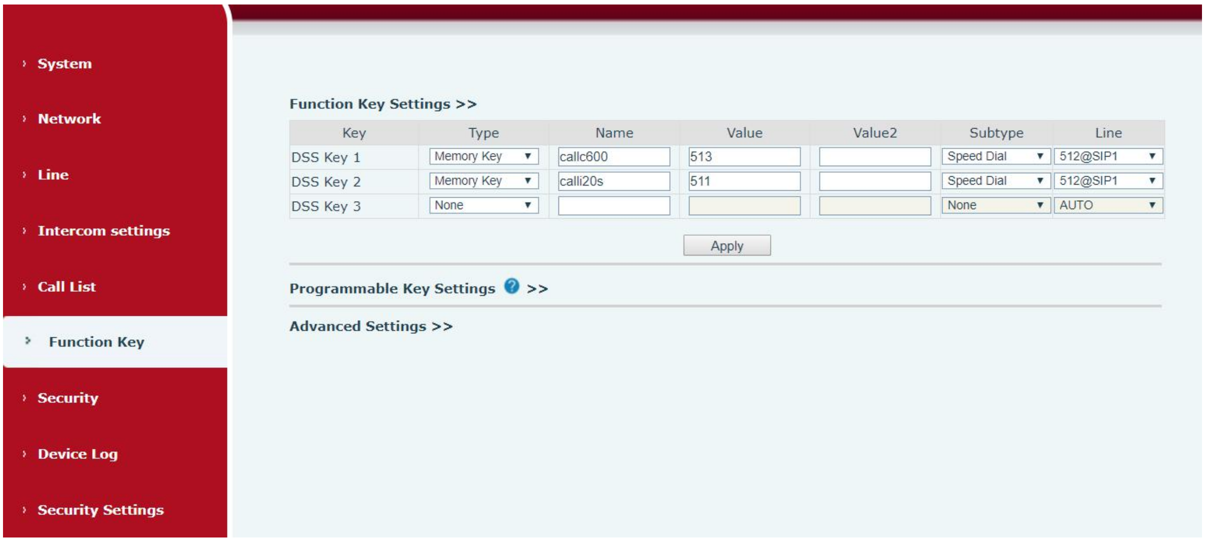
In this page, user can make calls, answer the calls or hang up the calls.



Web Dial Settings

Figure 31 - Web Dial

9.23Function Key

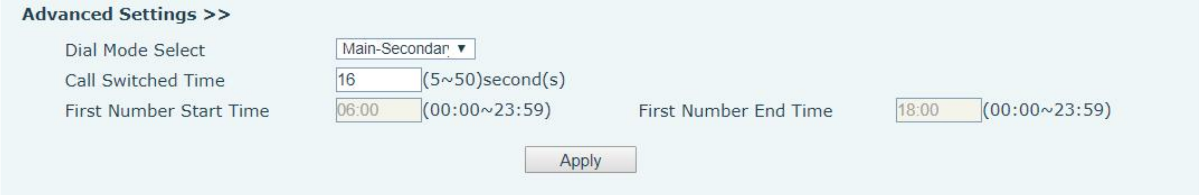


Function Key Settings >>

Key	Type	Name	Value	Value2	Subtype	Line
DSS Key 1	Memory Key	callc600	513		Speed Dial	512@SIP1
DSS Key 2	Memory Key	calli20s	511		Speed Dial	512@SIP1
DSS Key 3	None				None	AUTO

Programmable Key Settings >>

Advanced Settings >>



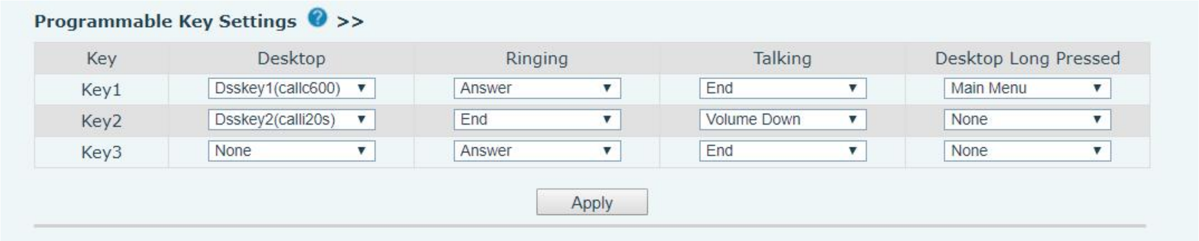
Advanced Settings >>

Dial Mode Select: Main-Secondar

Call Switched Time: 16 (5~50)second(s)

First Number Start Time: 06:00 (00:00~23:59)

First Number End Time: 18:00 (00:00~23:59)



Programmable Key Settings >>

Key	Desktop	Ringing	Talking	Desktop Long Pressed
Key1	Dsskey1(callc600)	Answer	End	Main Menu
Key2	Dsskey2(calli20s)	End	Volume Down	None
Key3	None	Answer	End	None

Figure 32 - Function Key

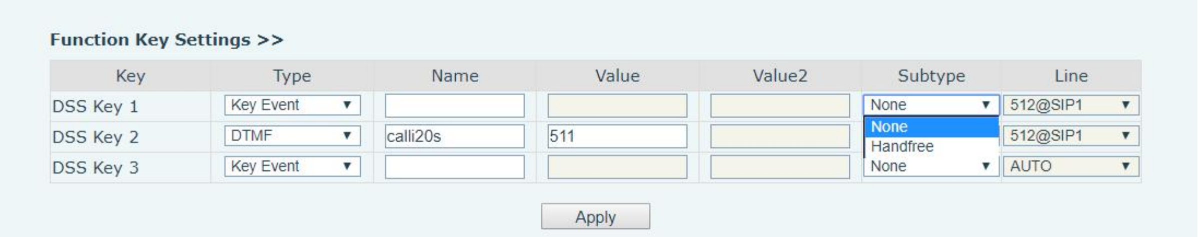
Table 17 - Function Key

Parameters	**Description**
Function Key Settings	
Memory Key	Speed dial: User can set one number or IP address in Value option. It is convenient to use this function to make calls to specified numbers/IP address used continually. Intercom: The intercom function make the operator or secretary to answer calls directly, which is popular in office.
Key Event	Use key event function to activate one application directly. Example: None/Handfree.
DTMF	Send the DTMF directly with the corresponding settings.
MCAST Paging	Set paging IP address and voice codec, user can initiate paging directly by pressing the button.
Action URL	User can use specified URL to make calls or open the doors.
MCAST Listening	When device is idle, use the MCAST listening key to monitor the MCAST from the paging IP address user set.
Programmable Key Settings	
Desktop	None: None. Dsskey1: Call out or pick up calls according to dsskey1's settings. Dsskey2: Call out or pick up calls according to dsskey2's settings. Dsskey3: Call out or pick up calls according to dsskey3's settings.
Ringing	Answer: When there is one incoming call and auto answer is disabled, use this key to pick up the call. End: When there is one incoming call, use this key to end the call.
Talking	End: Press the key to end the call when device is in one call. Volume Up: Press the key to increase volume when the device is in one call. Volume Down: Press the key to decrease volume when the device is in one call. Dsskey1: Call out or pick up calls according to dsskey1's settings. Dsskey2: Call out or pick up calls according to dsskey2's settings.

	Dsskey3: Call out or pick up calls according to dsskey3's settings.
Desktop Long pressed	None: None. Main Menu: Long press the key to make device go to command mode, details please check chapter 5.2.1 Common command mode.
Advanced Settings	
Dial Mode Select	Set the dial mode between calling 1st number and calling 2nd number. Main-Secondary: If the 1st number does not pick up the call in specified time, then the device will call 2nd number. Time Period: Device check the system time and send the call to 1st number in 1st number's time period, or the device will send call to the 2nd number.
Call Switched Time	Set the switched time between 1st number and 2nd number when device calls out, by default the value is 16 seconds.
First Number Start Time	Set 1st called number's start time, by default it is 06:00am.
First Number End Time	Set 1st called number's end time, by default it is 18:00pm.

Key Event

The speed dial key type could be set as Key Event.



Function Key Settings >>

Key	Type	Name	Value	Value2	Subtype	Line
DSS Key 1	Key Event				None	512@SIP1
DSS Key 2	DTMF	calli20s	511			512@SIP1
DSS Key 3	Key Event				None	AUTO

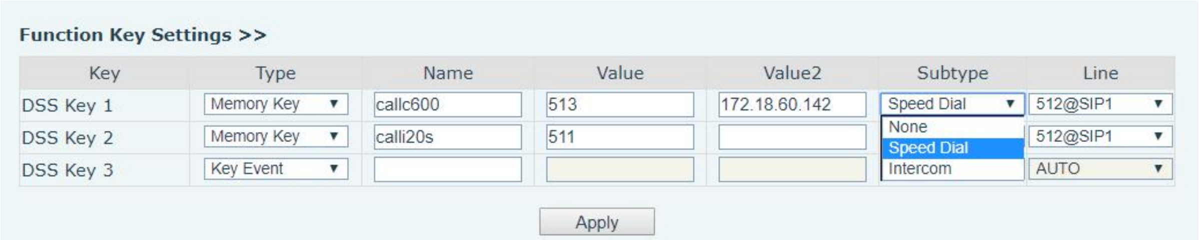
Figure 33 - Function Key Settings

Table 18 - Function Key Settings

Type	Subtype	Usage
Key Event	None	No responding
	Handfree	Handfree

Memory Key

When the speed dial key set as Memory Key, the device would dial preset telephone number. This button can also be used to set the IP address: you can press the speed dial button to directly make an IP call.



Function Key Settings >>

Key	Type	Name	Value	Value2	Subtype	Line
DSS Key 1	Memory Key	callc600	513	172.18.60.142	Speed Dial	512@SIP1
DSS Key 2	Memory Key	calli20s	511			512@SIP1
DSS Key 3	Key Event					AUTO

Figure 34 - Memory Key Settings

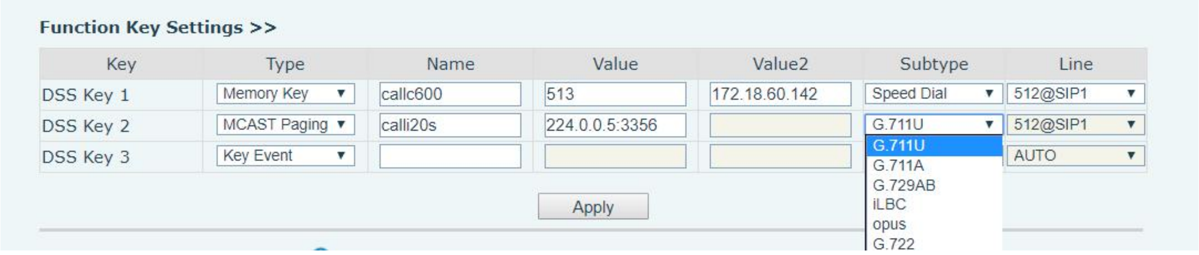
Table 19 - Memory Key Settings

Type	**Number**	**Line**	**Subtype**	**Usage**
Hot Key	Fill the called party's SIP account or IP address	The SIP account corresponding lines	Speed Dial	Set speed dial, press the key to call out the number.
			Intercom	In Intercom mode, if the caller's IP phone supports Intercom feature, the device can automatically answer the Intercom calls

➢ 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:



Function Key Settings >>

Key	Type	Name	Value	Value2	Subtype	Line
DSS Key 1	Memory Key	callc600	513	172.18.60.142	Speed Dial	512@SIP1
DSS Key 2	MCAST Paging	calli20s	224.0.0.5:3356		G.711U	512@SIP1
DSS Key 3	Key Event					AUTO

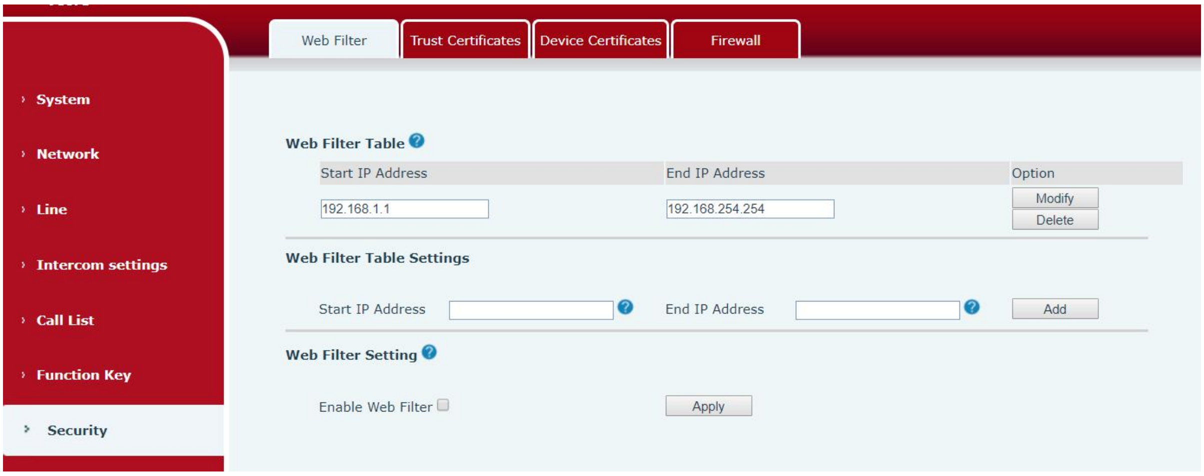
Figure 35 - Multicast Settings

Table 20 - Multicast Settings

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.711U G.711A G729AB iLBC opus G.722

9.24 Security >> Web Filter

In this page, user can set the IP address segment which is allowed to access the device.



Web Filter Table

Start IP Address	End IP Address	Option
192.168.1.1	192.168.254.254	Modify Delete

Web Filter Table Settings

Start IP Address [ ] End IP Address [ ] Add

Web Filter Setting

Enable Web Filter □ Apply

Figure 36 - Multicast Settings

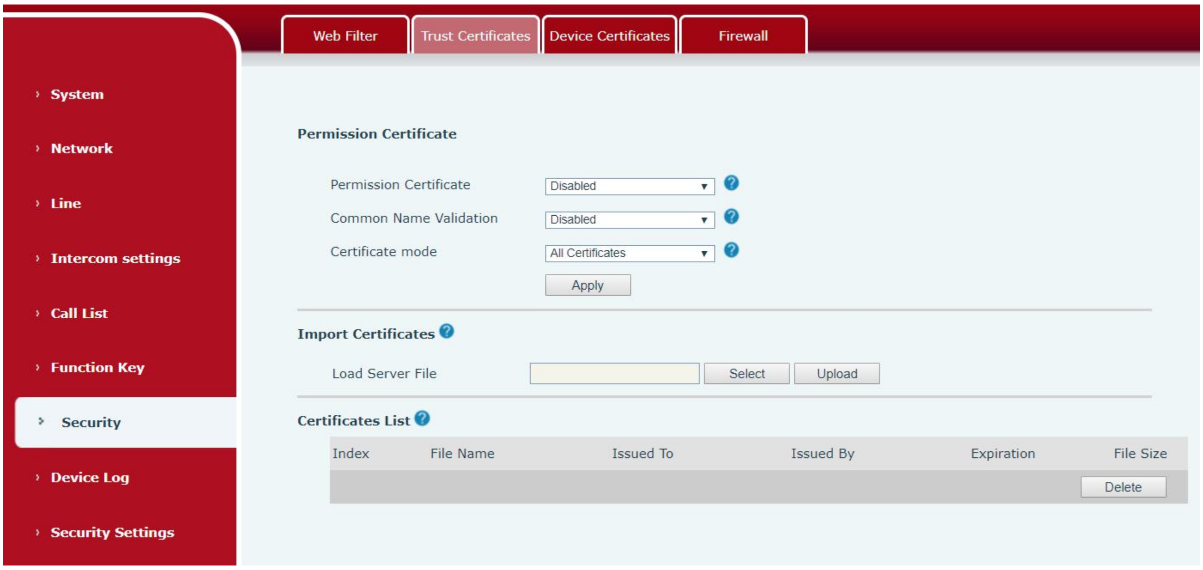
Add or delete the allowed IP address segment. Please input start IP address in Start IP address option, and input end IP address in End IP address option and click Apply to save the settings. Click Delete to remove the corresponding IP address segment.

Enable Web Filter: Enable or disable web filter, select it and click apply to save the settings.

Notice: Please remove your PC's IP address from the filter IP address list, or your PC is not able to access the device's webpage.

9.25 Security >> Trusted Certificates

User can upload or delete the trusted certificates in this page.



Permission Certificate

Permission Certificate	Disabled
Common Name Validation	Disabled
Certificate mode	All Certificates

Import Certificates

Load Server File

Certificates List

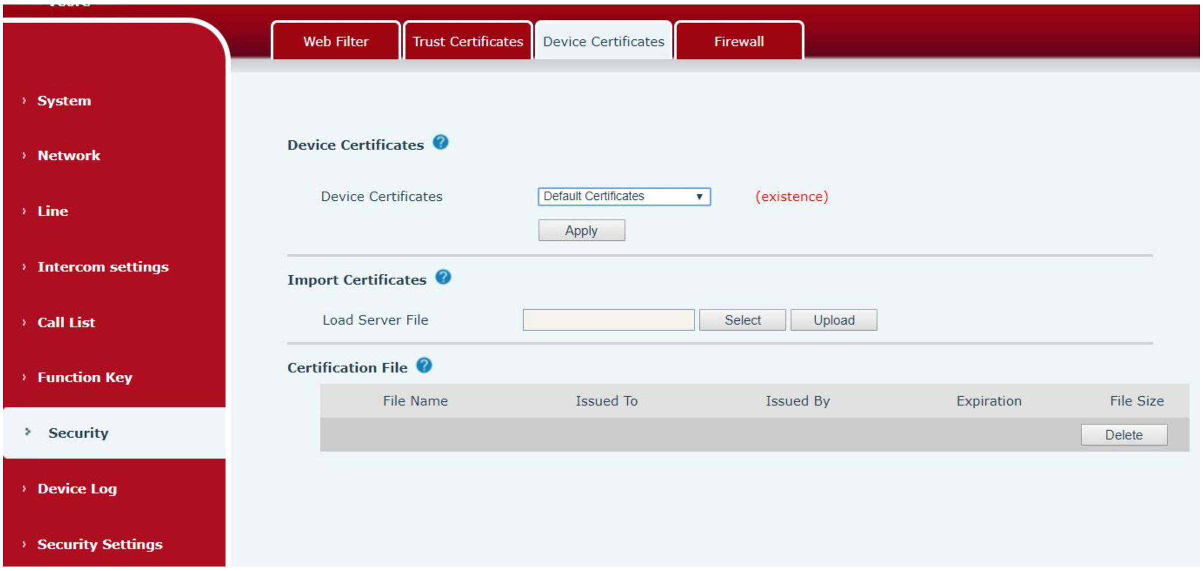
Index	File Name	Issued To	Issued By	Expiration	File Size

Figure 37 - Trusted Certificates

9.26Security >> Device Certificates

Select the device certificate to use default certificates or custom certificate.

You can upload and delete uploaded certificates.



Device Certificates

Device Certificates	Default Certificates	(existence)

Import Certificates

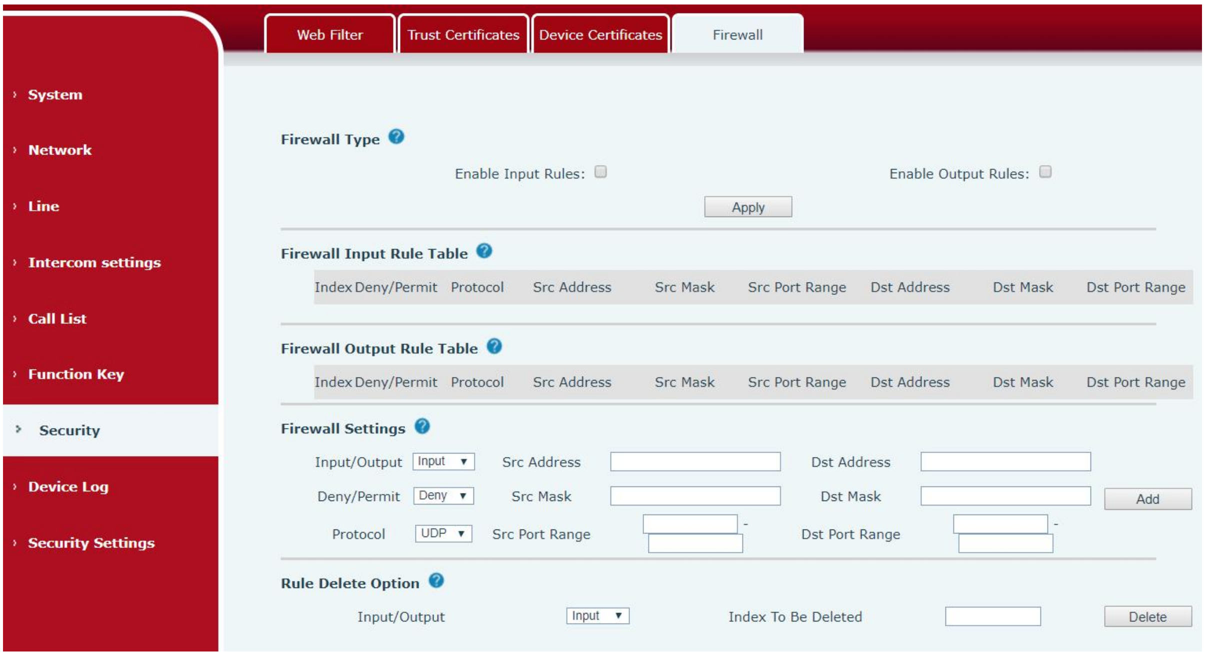
Load Server File

Certification File

File Name	Issued To	Issued By	Expiration	File Size

Figure 38 - Device Certificates

9.27Security >> Firewall



Firewall Type

Enable Input Rules: □ Enable Output Rules: □

Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range

Firewall Output Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range

Firewall Settings

Input/Output: Input | Src Address | Dst Address

Deny/Permit: Deny | Src Mask | Dst Mask

Protocol: UDP | Src Port Range - | Dst Port Range -

Rule Delete Option

Input/Output: Input | Index To Be Deleted

Figure 39 - Firewall

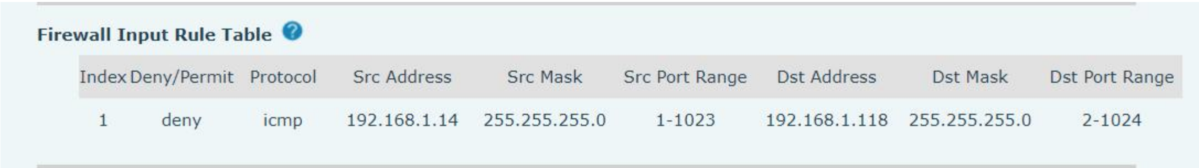
In this page, user can select whether to enable input or output firewall, and set the detailed rules. These settings are used to prevent illegal network access, limit the internal user to access Internet sources, enhance the security.

Table 21 - Firewall

Parameters	Description
Firewall Type	
Enable Input Rules	Enable or disable input rules.
Enable Output Rules	Enable or disable output rules.
Input/Output	Set the rule to be input rule or output rule.
Deny/Permit	Set the rule to be denying rule or permitting rule.
Protocol	Select the firewall protocol, options are UDP, TCP and ICMP.
Src Address	Input source address. The source IP can be one host address or network address, you can input 0.0.0.0 to represent all the IP addresses, or input one *.*.*.0 network IP, like 192.168.1.0.
Src Mask	Input source mask. If user configure source mask to be 255.255.255.255, the source IP should be one detailed IP address; if user configure source mask to be 255.255.255.0, the source IP contains a segment of IP addresses.
Src Port Range	Input source port range.
Dst Address	Input destination address. The destination IP can be one specified IP

	address, or 0.0.0.0 which represents all the IP addresses, or network IP address *.*.*.0, like 192.168.1.0.
Dst Mask	Input destination mask. If user configure destination mask to be 255.255.255.255, the destination IP should be one detailed IP address; if user configure source mask to be 255.255.255.0, the destination IP contains a segment of IP addresses.
Dst Port Range	Input destination port range.

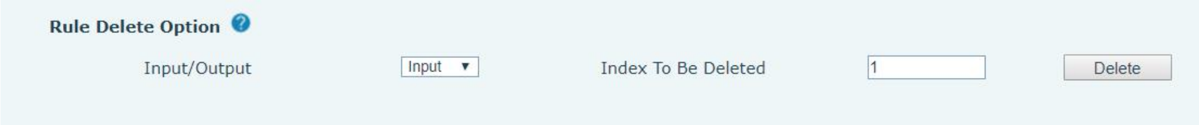
Input the parameters and click Add, the new rules will be added to the firewall list, for example:



Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range
1	deny	icmp	192.168.1.14	255.255.255.0	1-1023	192.168.1.118	255.255.255.0	2-1024

Select Input/Output rule, and enter the index of the rule in Index to Be deleted option, click delete then the correponding rule will be removed.



Rule Delete Option

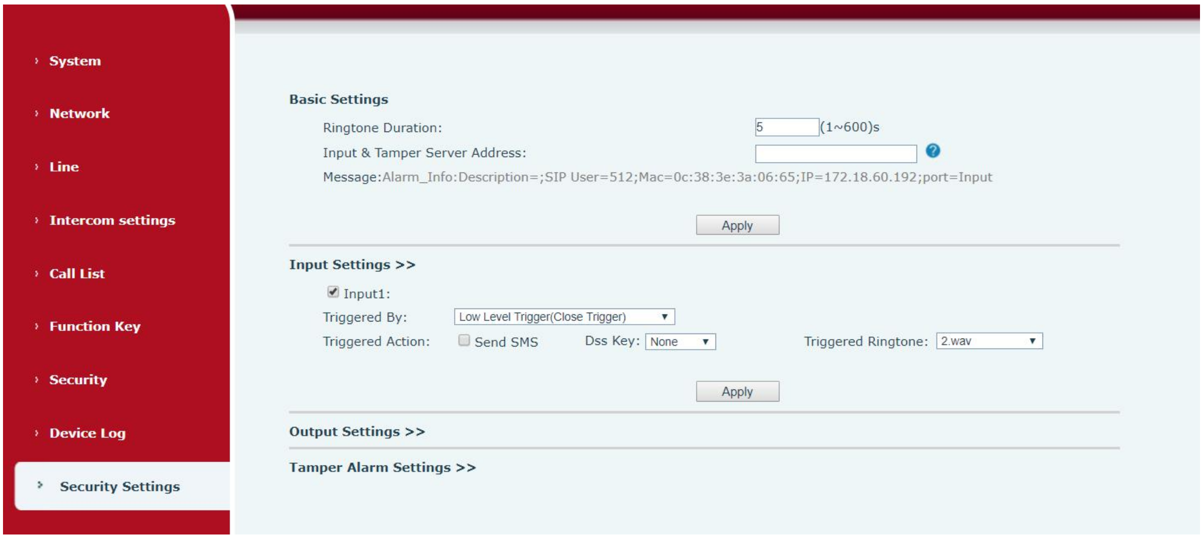
Input/Output: Input

Index To Be Deleted: 1

9.28 Device Log

In this page, user can get the device's logs. When device works abnormally, user can get the logs and send to Fanvil support team. Details please check chapter **10.5 Get Log Information**.

9.29 Security Settings



Basic Settings

Ringtone Duration: 5 (1~600)s

Input & Tamper Server Address:

Message:Alarm_Info:Description=;SIP User=512;Mac=0c:38:3e:3a:06:65;IP=172.18.60.192;port=Input

Input Settings >>

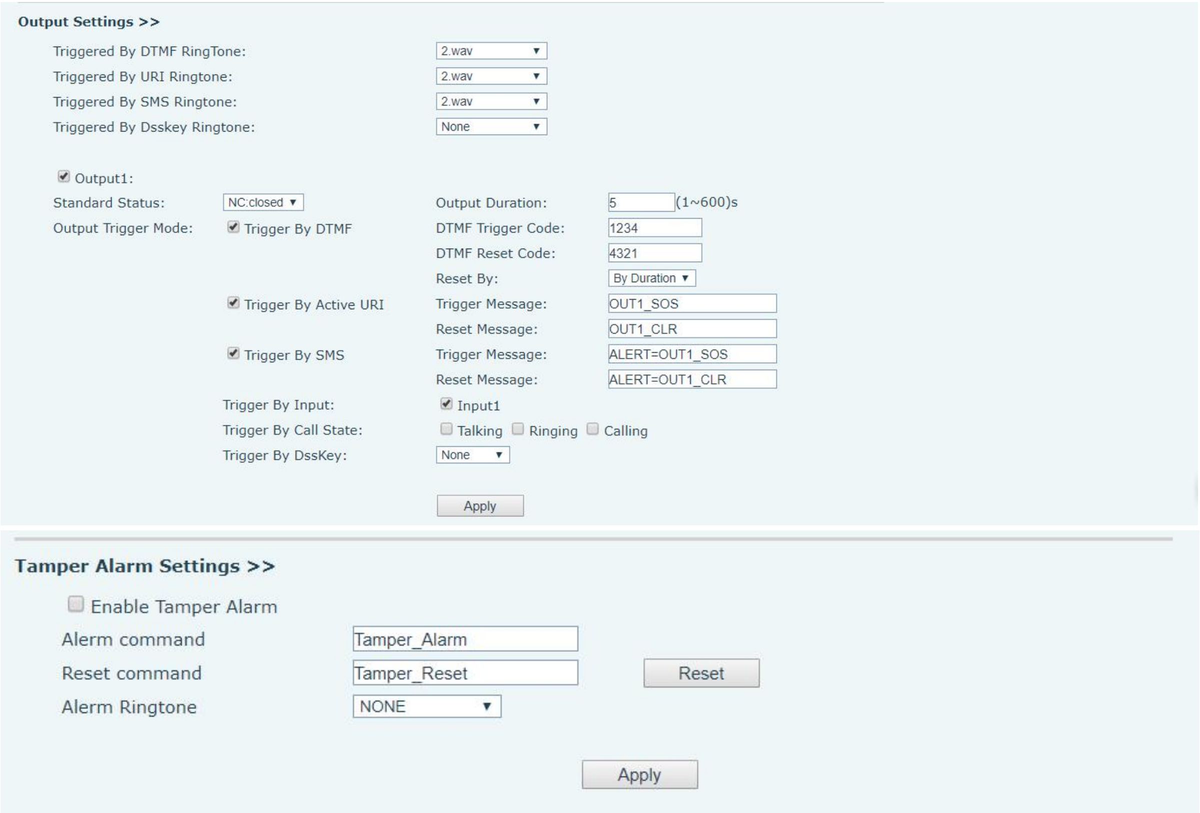
☑ Input1:

Triggered By: Low Level Trigger(Close Trigger)

Triggered Action: □ Send SMS Dss Key: None Triggered Ringtone: 2.wav

Output Settings >>

Tamper Alarm Settings >>



Output Settings >>

Triggered By DTMF RingTone: 2.wav

Triggered By URI Ringtone: 2.wav

Triggered By SMS Ringtone: 2.wav

Triggered By Dsskey Ringtone: None

☑ Output1:

Standard Status: NC:closed

Output Duration: 5 (1~600)s

Output Trigger Mode: ☑ Trigger By DTMF

DTMF Trigger Code: 1234

DTMF Reset Code: 4321

Reset By: By Duration

☑ Trigger By Active URI

Trigger Message: OUT1_SOS

Reset Message: OUT1_CLR

☑ Trigger By SMS

Trigger Message: ALERT=OUT1_SOS

Reset Message: ALERT=OUT1_CLR

Trigger By Input: ☑ Input1

Trigger By Call State: □ Talking □ Ringing □ Calling

Trigger By DssKey: None

Apply

Tamper Alarm Settings >>

□ Enable Tamper Alarm

Alerm command: Tamper_Alarm

Reset command: Tamper_Reset

Reset

Alerm Ringtone: NONE

Apply

Figure 40 - Security Settings

Table 22 - Security Settings

Security Settings	
Parameters	**Description**
Basic Settings	
Ringtone Duration	Set the ringtone duration, default value is 5 seconds.
Input & Tamper Server Address	Set remote server address. The device will send message to the server when the alarm is triggered. The message format is : Alarm_Info:Description=i10;SIP User=;Mac=0c:38:3e:3a:06:65;IP=;port=Input .
Input settings	
Input Detect	Enable or disable Input Detect
Triggered by	When choosing the low level trigger (closed trigger), detect the input port (low level) closed trigger.
	When choosing the high level trigger (disconnect trigger), detect the input port (high level) disconnected trigger.
Triggered Action	**Send SMS:** Set the alert message send to server if selected. **Dss Key:** The device will perform corresponding Dss Key configurations if any key is selected, by default the value is none. **Triggered Ringtone:** Select triggered ring tone.

Output Settings	
Output Response	Enable or disable Output Response
Triggered by DTMF Ring tone	Select the DTMF trigger ring tone.
Triggered by URI Ringtone	Select the URI trigger ring tone.
Triggered By SMS Ringtone	Select the SMS trigger ring tone.
Triggered By Dsskey Ringtone	Select the Dsskey trigger ring tone.
Standard Status	When choosing the low level trigger (NO: normally open), when meet the trigger condition, trigger the NO port disconnected.
	When choosing the high level trigger (NC: normally close), when meet the trigger condition, trigger the NC port close.
Output Duration	Set the output change duration time, the default is 5 seconds.
Trigger by DTMF	Enable or disable trigger by DTMF. The device will check the received DTMF sent by remote device, if it matches the DTMF trigger code, the device will trigger corresponding output port.
DTMF Trigger Code	Input the DTMF trigger code, default value is 1234.
DTMF Reset Code	Input the DTMF reset code, default value is 4321.
Reset By	Reset the output port mode by duration or state. By duration: Reset the output port status when output duration occurs. By state: Reset the output port status when device's call state changes.
Trigger by URI	Enable or disable trigger by URI. User can send commands from remote device or server to i10 series device, if the command is correct, then device will trigger corresponding output port.
Trigger Message	Input trigger message for trigger by URI mode.
Rest Message	Input reset message for trigger by URI mode.
Trigger by SMS	Enable or disable trigger by SMS. User can send ALERT command to i10 series device, if the command is correct, then device will trigger corresponding output port.
Trigger SMS	Input trigger message for trigger by SMS mode.
Reset SMS	Input reset message for trigger by SMS mode.
Trigger by Input	Select the input port, when the input port meets the trigger condition, the output port will be triggered (The Port level time change, By < Output Duration > control)

Trigger By Call state	Select call state to trigger the output port, options are: Talking: When the device’s talking status changes, trigger the output port. Ringing: When the device’s ringing status changes, trigger the output port. Calling: When the device’s calling status changes, trigger the output port.
Trigger By DssKey	Enable or disable trigger by dsskey. If any of the dsskey is selected, when the dsskey application performs, the output port will be triggered.

10 Trouble Shooting

When the device doesn't work properly, user 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

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

10.2 Reboot device

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

When the device has problems and user can't access the web page, you can disassemble the surface shell and press the "**RESET**" button. The device will restart and the configuration will not change.

10.3 Device factory reset

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

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

10.4 Network Packets Capture

Sometimes, when the device has problems, the data packet is very helpful. In order to obtain the data packet of the device, please log in the device's webpage, and go to [**System**] >> [**Tools**] page, and click the [**Start**] option in the "Web Capture". A message will inform user that capturing starts and at this time, user can perform related operations, such as starting/deactivating the line or making a call, please click the [**Stop**] button on the webpage after complete. Network packets are saved in a file, users can analyze the packet or send it to Fanvil Technical Support team.

10.5 Get Log Information

Log information is helpful when encountering an abnormal problem. In order to get the log, the user can login device's webpage, and go to page [**Device Log**], click the [**Start**] option, and perform device until the problem appears, click [**Save**] to save the logs to local PC, user can analyze the logs or send the log file to the technician to check the problem.

10.6 Common Trouble Cases

Table 23 - Common Trouble Cases

Trouble Case	Solution
Device could not boot up	1. Please check device's power connection and confirm the power adapter or PoE switch is in Fanvil list. 2. If the device go to "POST mode" (the LED flashes slowly), it means the device system is damaged. Please contact Fanvil technical support to help you restore.
Device could not register to a service provider	1. Please check network cable connection and confirm the device is connected to Internet well. 2. If the network connection is good, please check your SIP line configuration again. If all configurations are correct, please contact your service provider for support, or obtain a registration network packet according to the instructions in "10.4 Network Data Capture", and send the packet to Fanvil support team to help analyze the issue.