ESI Access

AC10v Video Intercom Programming & User's Manual for eSIP & eCloud

This document provides instructions for installing and programming the ESI AC10v Video Intercom.

Content

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.

- Do not handle the device roughly. Rough handling can break internal circuit boards.
- This device 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.
- Avoid exposing the device to high temperature, low temperature or high humidity.
- Avoid getting the device wet.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean device. Wipe device with a soft cloth that has been slightly dampened in a mild soap and water solution.
- Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

Overview

The AC10v Video Intercom is designed for indoor and is IP54 waterproof and dustproof. It combines security, audio/video intercom and broadcasting functionality.

Install Guide

Use POE or external Power Adapter

AC10v, called as ‘the device’ hereafter, supports two power supply modes, power supply from an external power adapter or over Ethernet (POE) complied switch.

A Power Over Ethernet saves 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 a POE switch, the device can keep working during power outage.

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 when it fails.

Common command mode

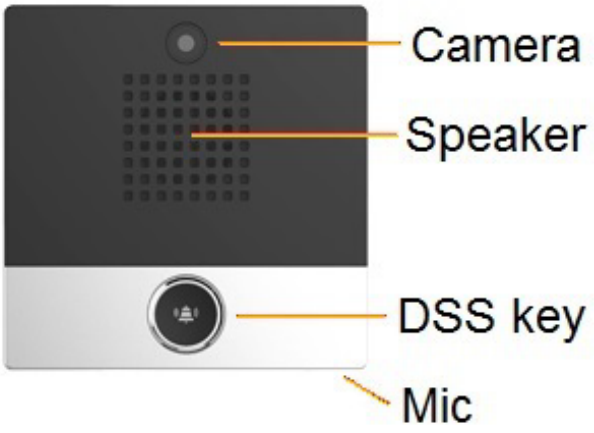
Action	Description
IP Broadcast under standby mode	In standby mode, long press the speed dial button for 3 seconds, there will be a beep for 5 seconds. Press the speed dial button once within 5 seconds, the beep will stop automatically reporting IP.
Switch network mode	In standby mode, long-press the 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 the network mode.

Function key LED state

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

Basic Introduction

Panel Overview



Name	Description
Camera	Video signal acquisition and transmission
Speaker	Play sound
DSS key	For speed dial, multicast, intercom, IP broadcast and other functions
Mic	Sound capture

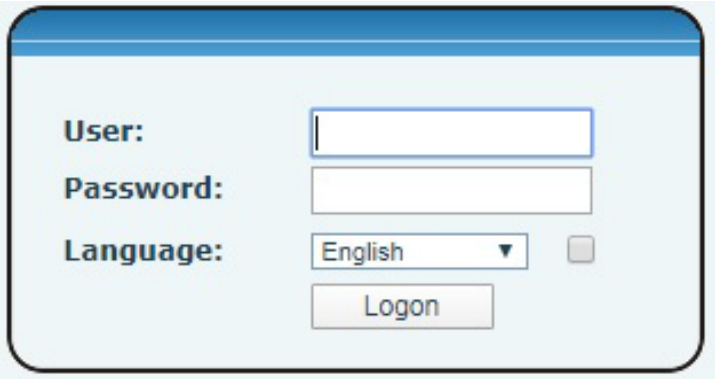
Quick Setting

Before proceeding with this step, make sure your Internet broadband connection is working properly and have completed 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 beep sound for 5 seconds. Press the speed dial button once within 5 seconds, the beep sound will stop automatically and report the device IP.
- In 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.
- Configure the account, user name, server address and other parameters required for registration provided by the service provider on the WEB configuration page;

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 will see the login as shown below.



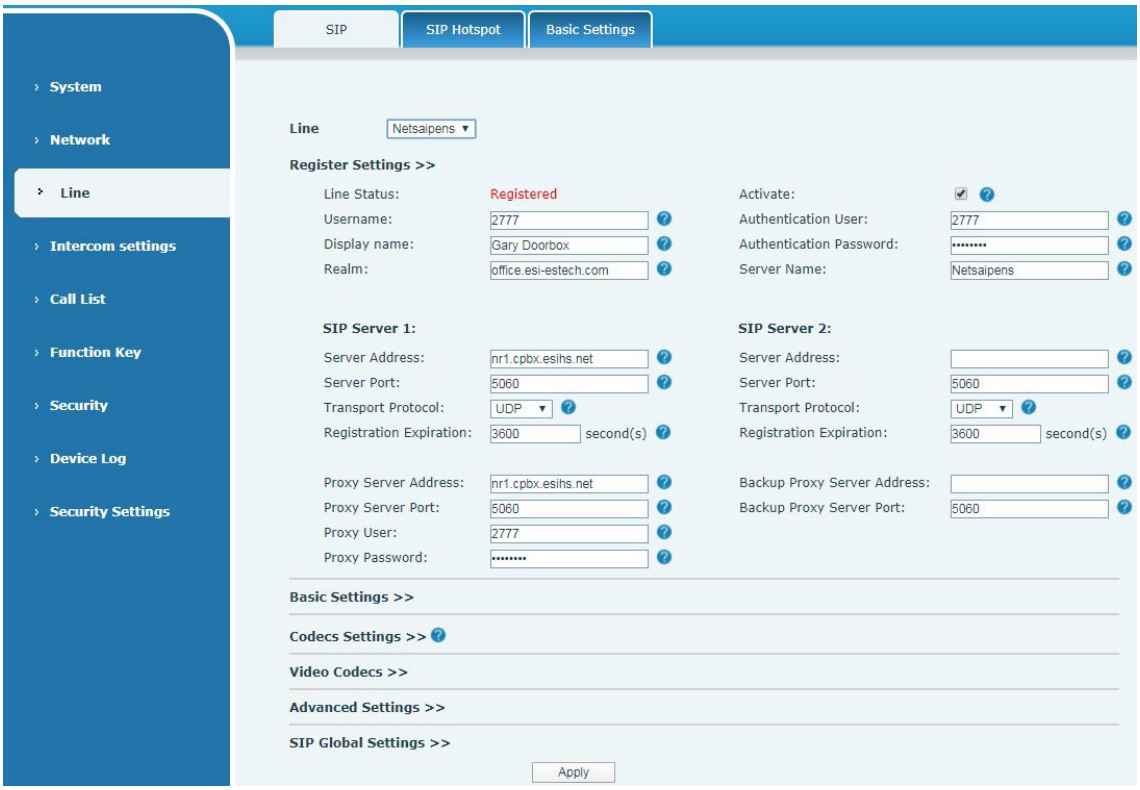
The username and password should be correct to log in to the web page. **The default username is "admin". Default password for eSIP is "admin" Default password for eCloud is SIPstn@ESI**. For the specific details of the operation of the web page, please refer to the **Web Configuration** section.

SIP Configurations

At least one SIP line should be configured properly to enable the telephony service. The line configuration 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.

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.

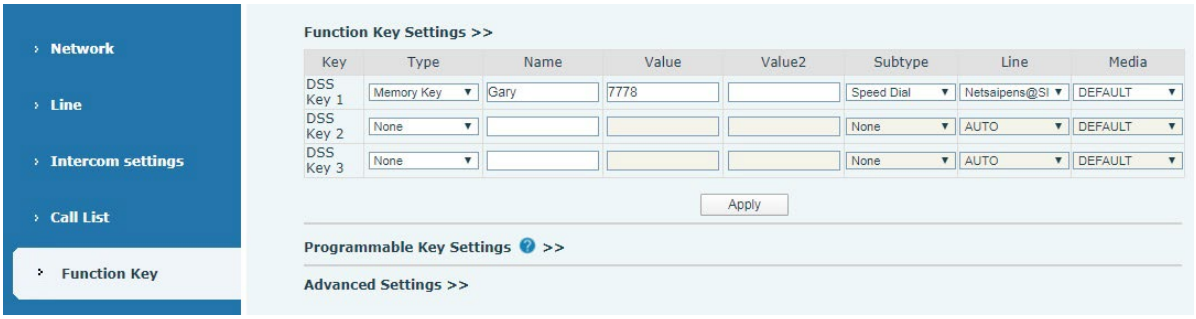
- WEB interface: After login into the devices web page, enter [**Line**] >> [**SIP**] and select **SIP1/SIP2** for configuration, click apply to complete registration after configuration, as shown below:



Basic Function

Making Calls

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



See detailed configuration instructions in **Function Key**

Answering Calls

After setting up the automatic answer and setting up the automatic answer delay, it will automatically answer the call after the timeout, cancel automatic answering or will not answer the phone.

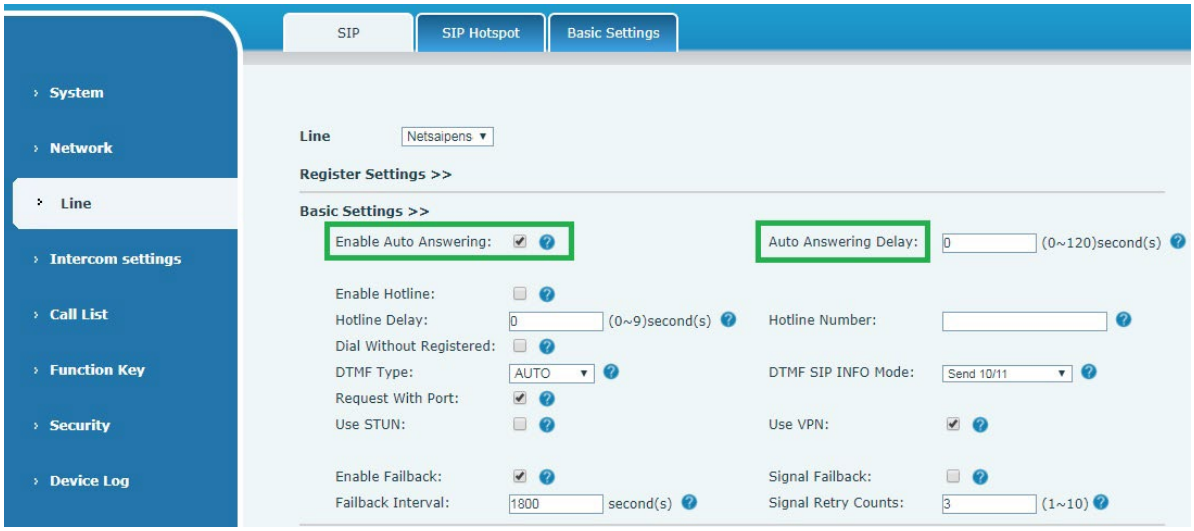
End of the Call

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 **Function Key**.

Auto-Answering

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

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

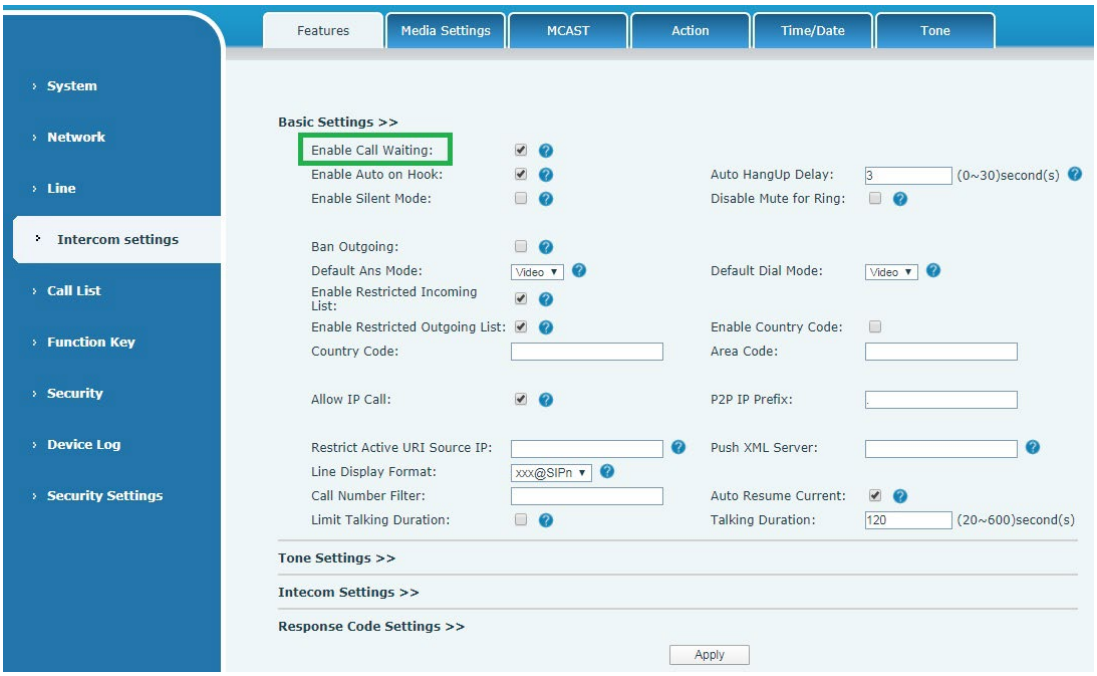


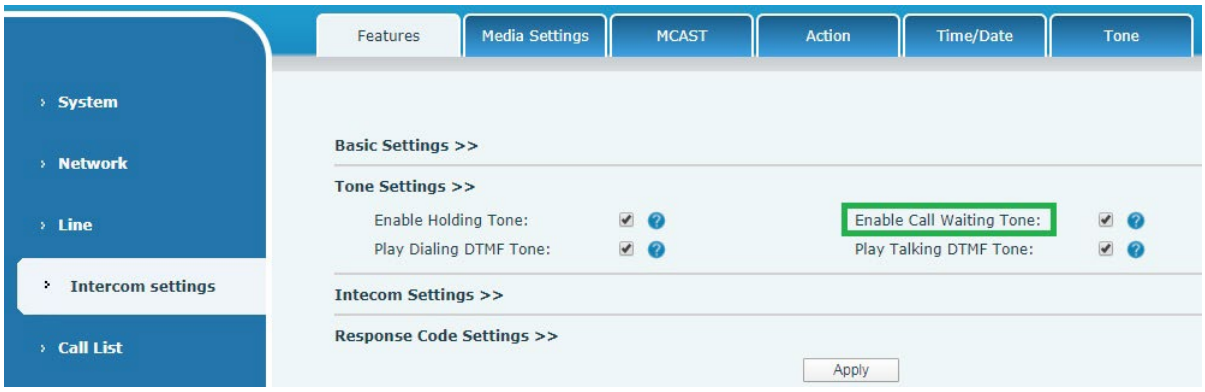
To enable auto answer P2P, at the web interface: enter [**line**] >> [**Basic Settings**] >> [**SIP P2P Settings**], enable automatic answering, setting mode and automatic answer delay, and click APPLY.

The Auto Answer Delay range can be set to 0~60 seconds, and the call will be answered automatically after the delay time has elapsed.

Call Waiting

- Enable call waiting: New calls can be accepted during a call.
- Disable call waiting: New calls will be automatically rejected and caller will hear a busy signal.
- 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 web interface.
- Web interface: Enter [**Intercom Setting**] >> [**Features**], enable/disable call waiting, enable/disable call waiting tone.

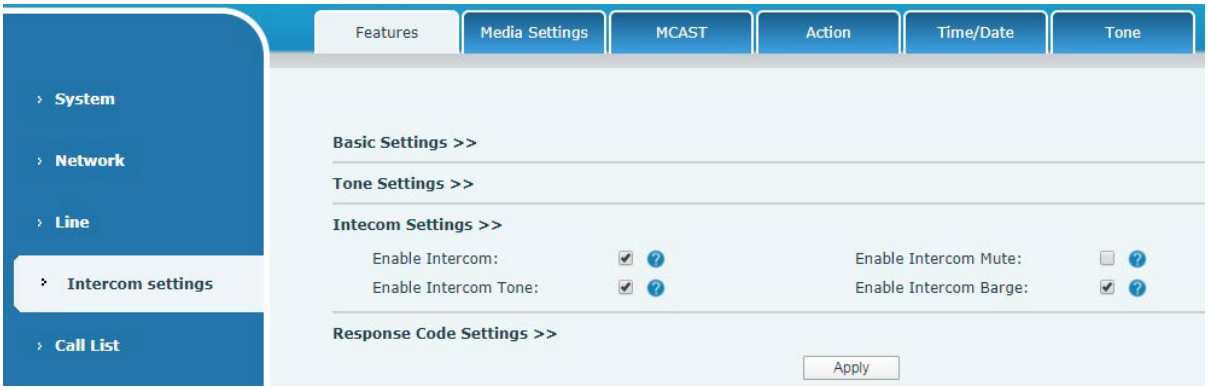




Advance Function

Intercom

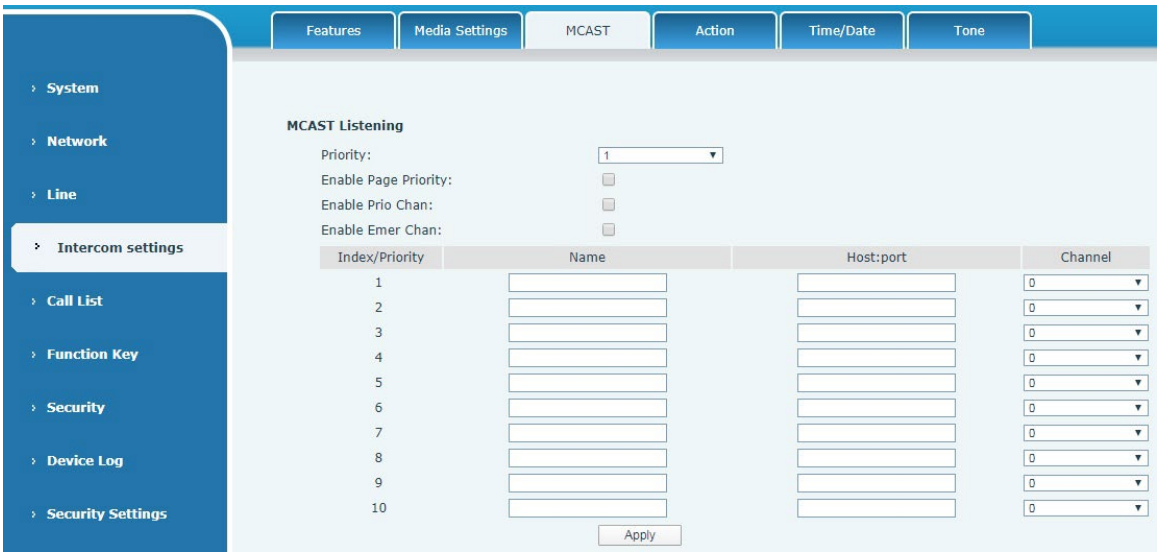
The equipment can answer intercom calls automatically.



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	When intercom barge is enabled, 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.

MCAST

This feature allows the user to make a 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.



Parameters	Description
Priority	Define the current call's priority, 1 means the highest priority and 10 means the lowest.
Enable Page Priority	If page priority is enabled, the device will receive the multicast from address with higher priority, regardless of which of the two multicast groups sent the multicast first.
Enable Prio Chan	If this option is enabled, the only multicast with the same port and channel can be connected. Channel 24 has the higher priority than 1-23; Set channel value to 0, meaning 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 multicast type, set the multicast address (AUTO), 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.

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

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

SIP Hotspot

SIP hotspot is a feature that will allow multiple devices to share the same number.
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 calls out, it calls out using the number registered to device A.

Parameters	Description
Enable Hotspot	Enable hotspot option in the SIP hotspot configuration TAB.
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 is 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
Local 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.
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).

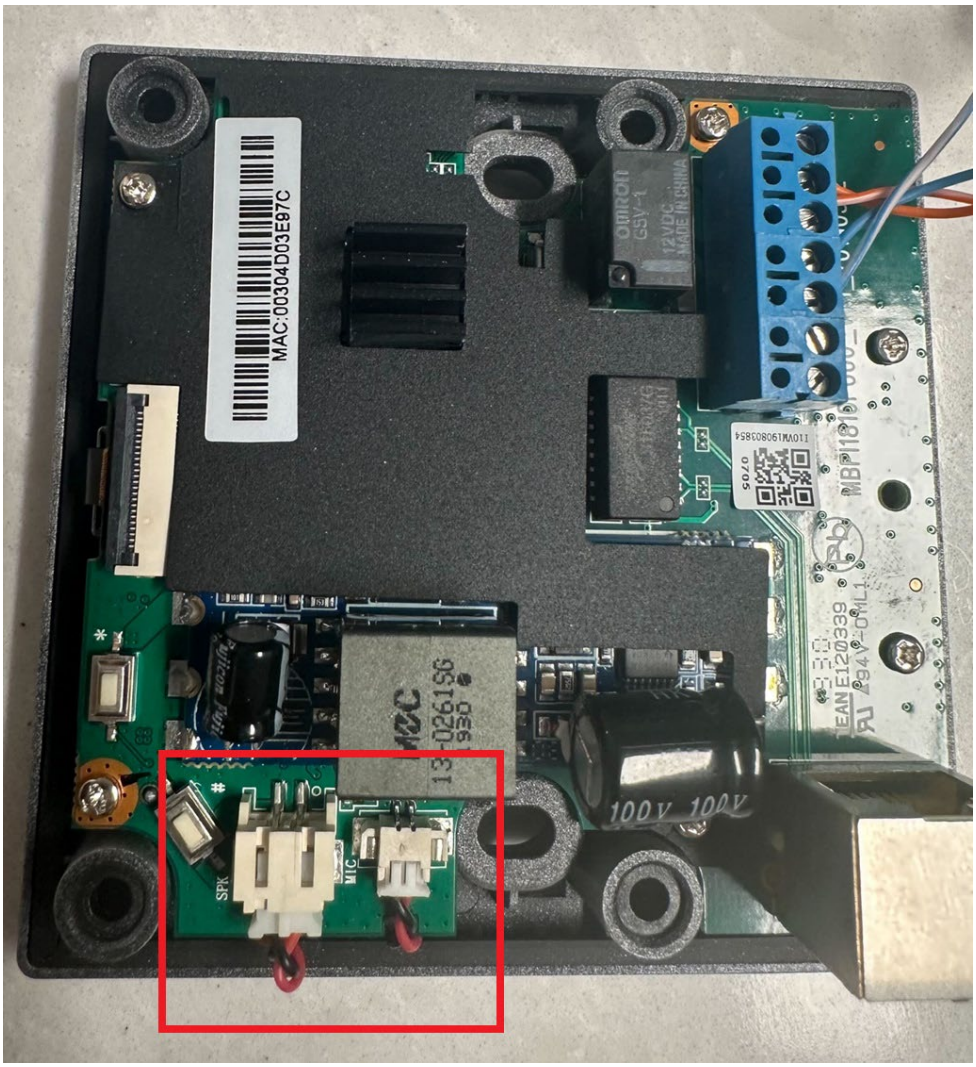
Calling internal extension:
The hotspot server and client can dial each other using the same extension number.

AC10v Silent Alarm

In case of an emergency, many customer environments are in need of a silent alarm call button. Doctors' Offices, School Classrooms, and Face to Face customer service are just some of the environments that may require silent alarm buttons to alert security. The ESI AC10v Access Device is a small form factor intercom capable of two-way, one-way, and no-way communications with or without video.

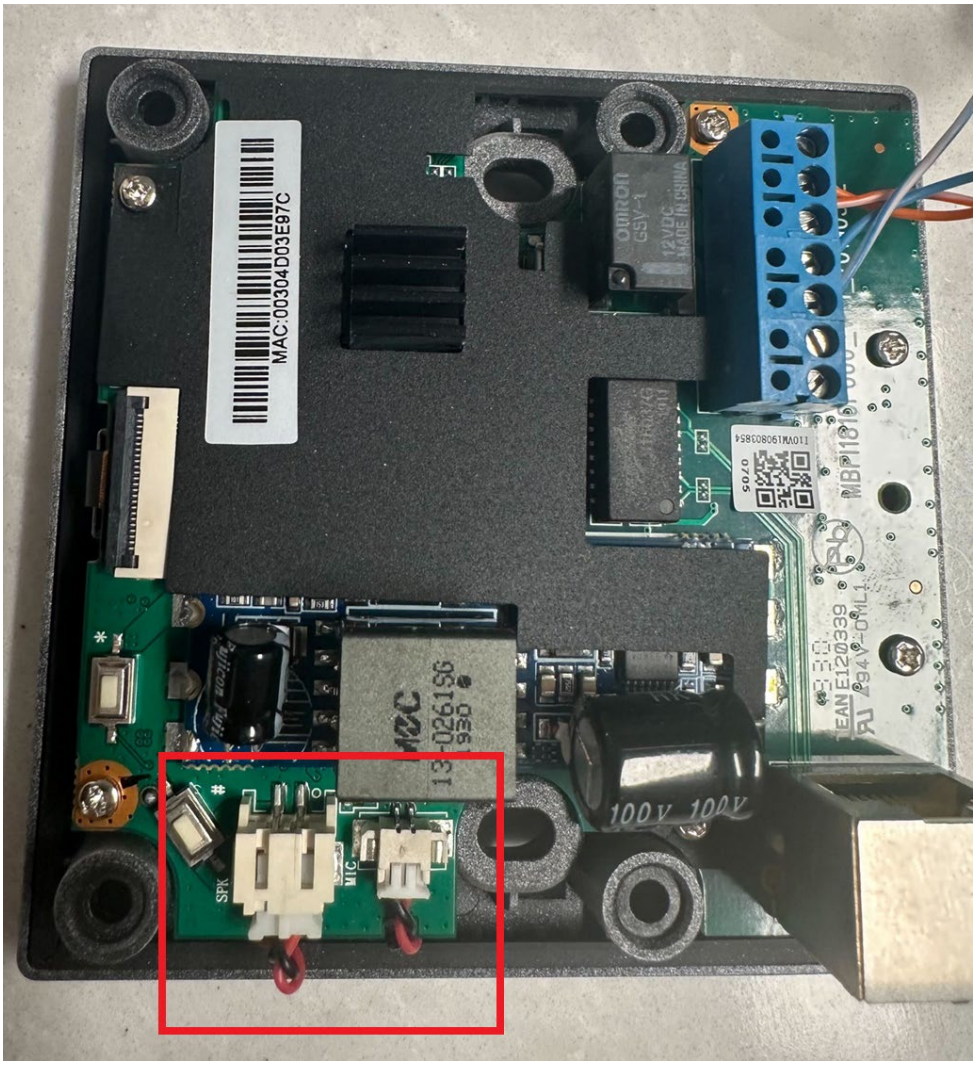
How to set up

On the AC10v circuit board there are two connectors, one for the Speaker (labeled SPK) and the other for the Microphone (labeled MIC).



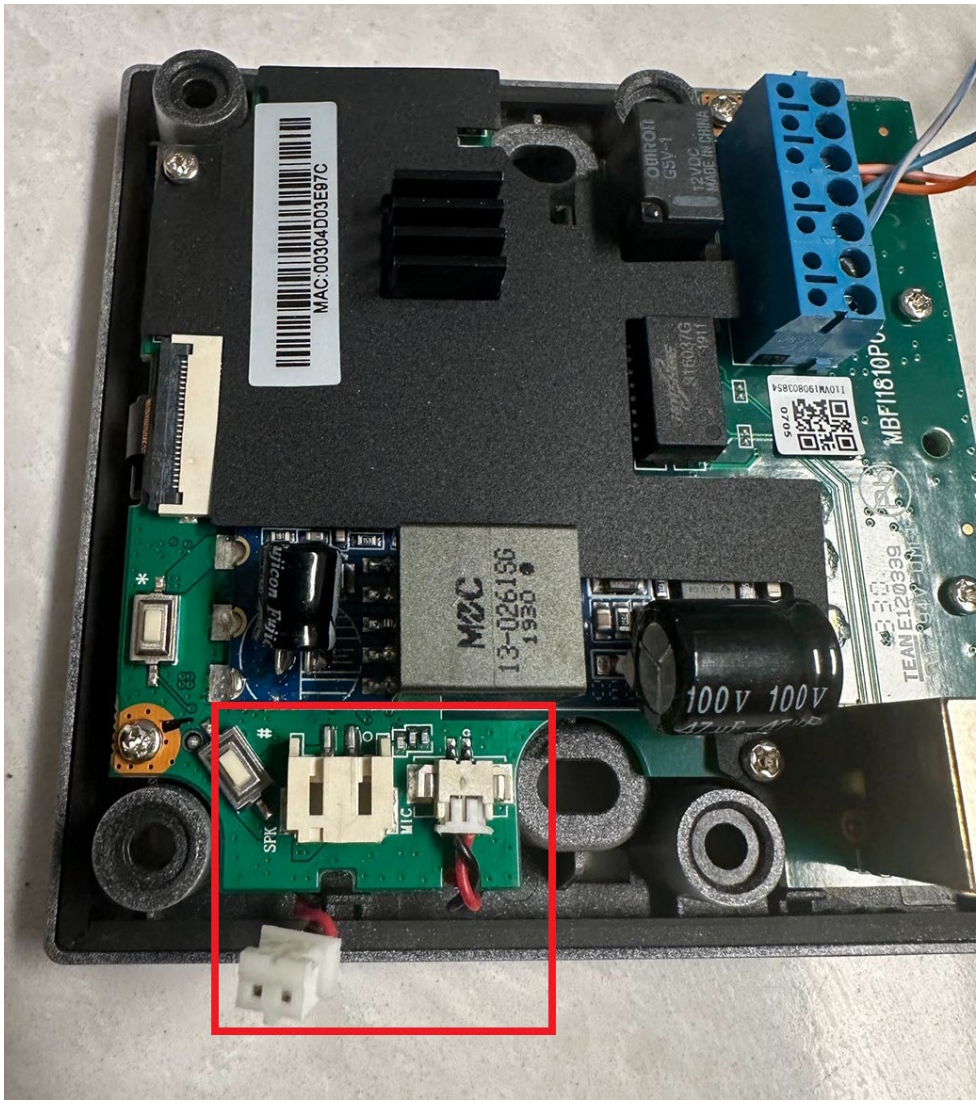
Two-Way Audio

Connect both Speaker and Mic.



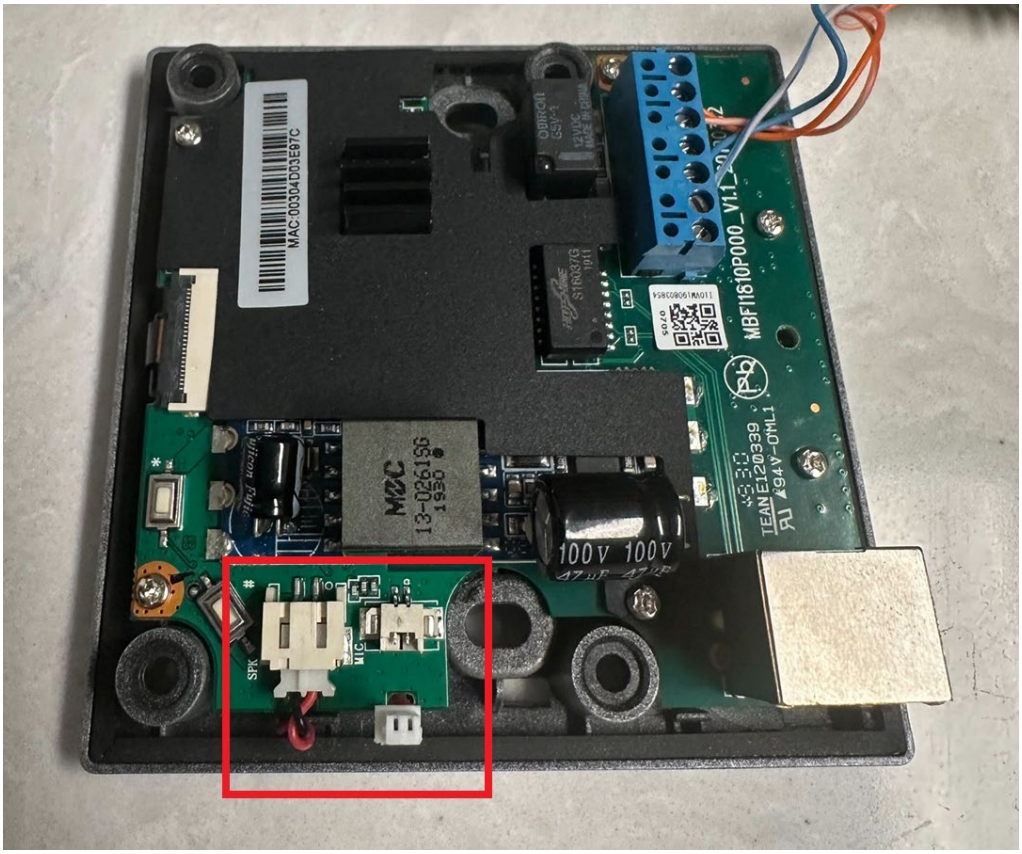
One-Way Audio (Send Only)

Disconnect Speaker and Connect Mic.



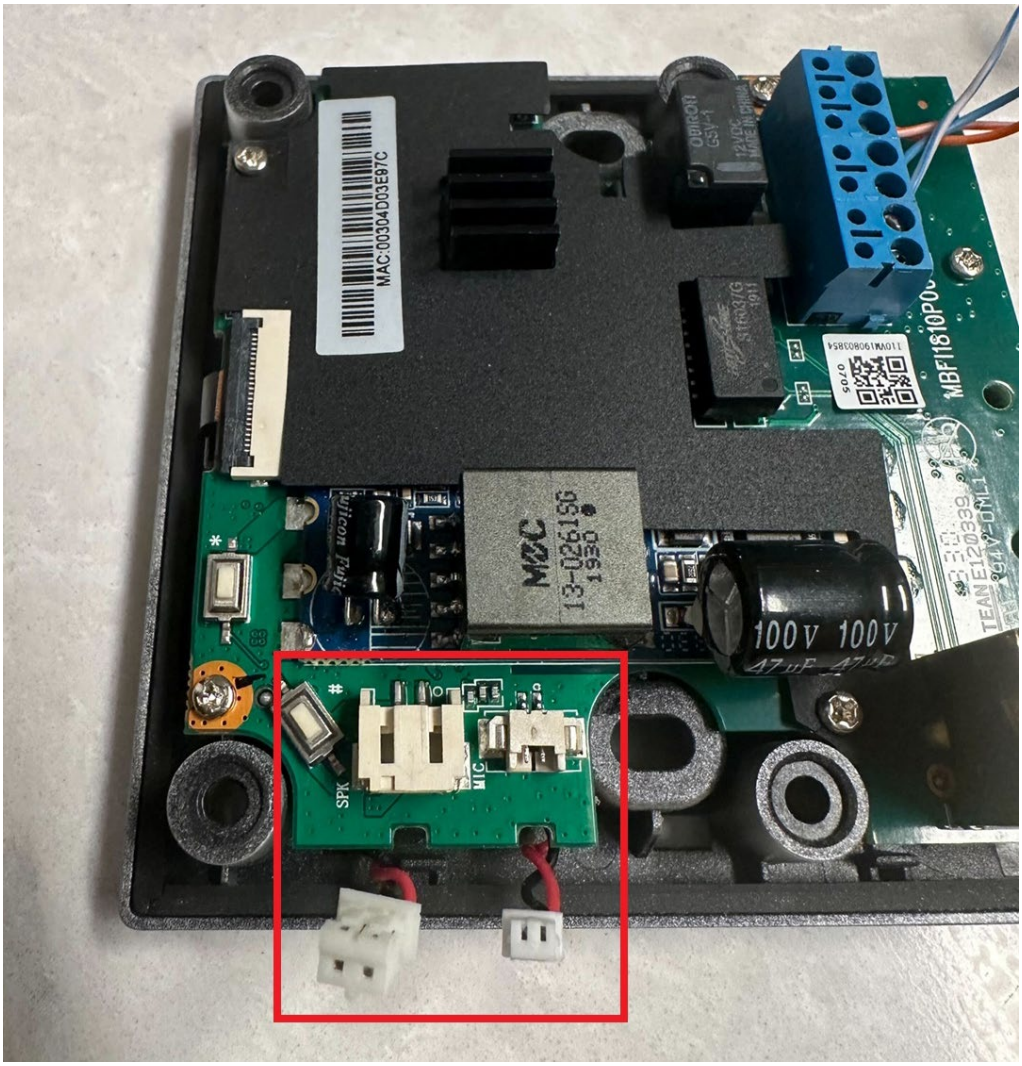
One-Way Audio (Receive Only)

Connect Speaker and Disconnect Mic.



No Audio

Disconnect Speaker and Mic.



Before installing the back cover, make sure the disconnected connector and its wires are tucked away so as not to interfere with reassembly

Web Configurations

Web Page Authentication

Users can log in to 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 three times, the webpage will be locked for 5 minutes and then the user can try to log in again.
The details as following:

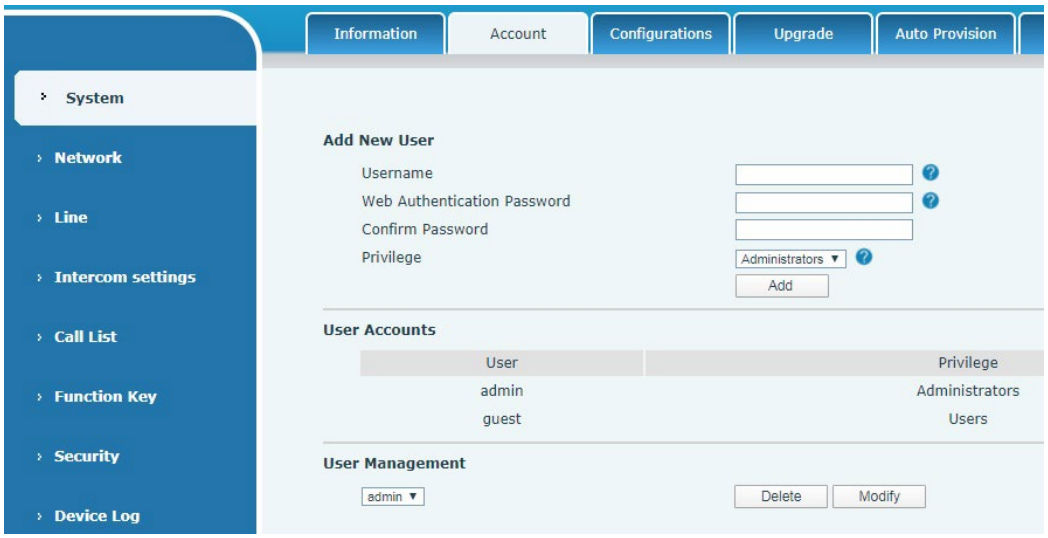
- If one IP logs in more than the specified number of times with different username/passwords, web login will be locked.
- If a same user logs in more than a specified number of times from different IP addresses, web login will be locked too.

System >> Information

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

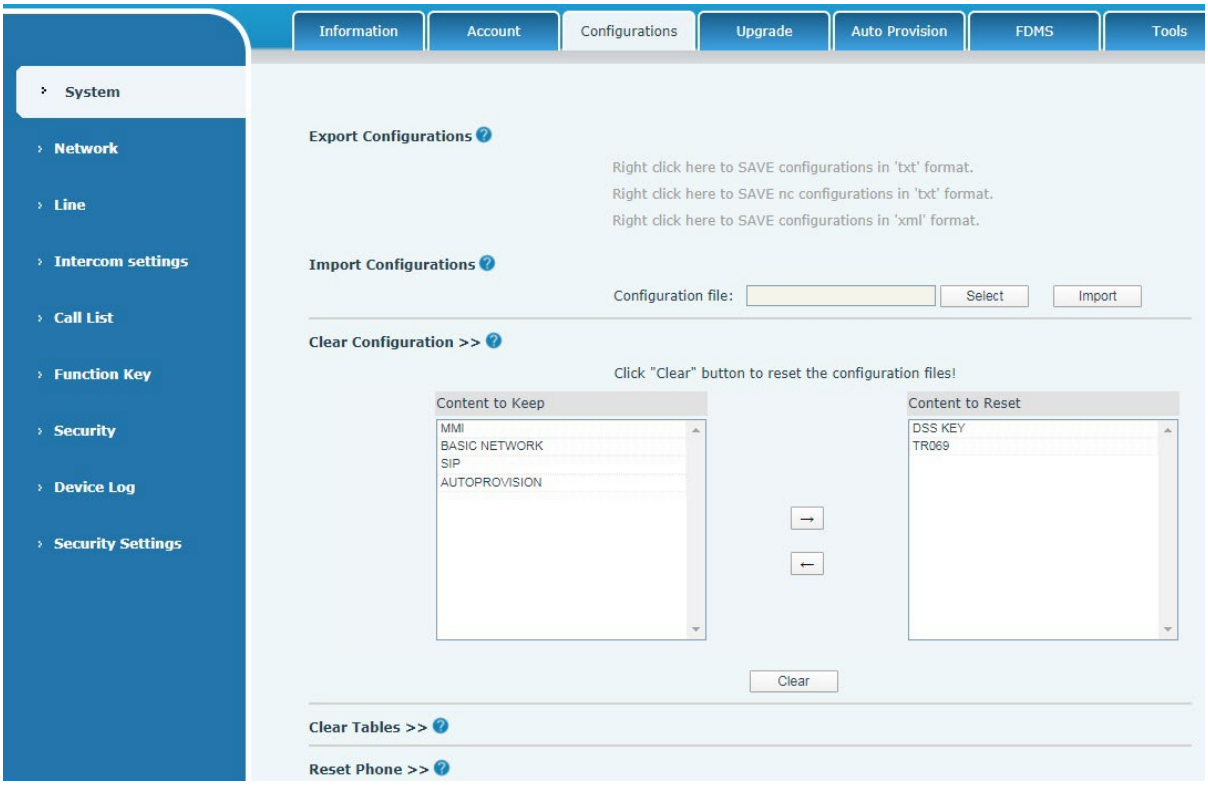
- Model
- Hardware Version
- Software Version
- Uptime
- Last uptime
- MEMinfo
- Network Mode (DHCP/Static)
- MAC Address
- IP
- Subnet Mask
- Default Gateway
- SIP User
- SIP account status (Registered/Inactive/Trying/Timeout)

System >> Account



On this page, user can change the webpage login password.
Administrator can also add or delete users, manage users, set permissions and passwords for new users.

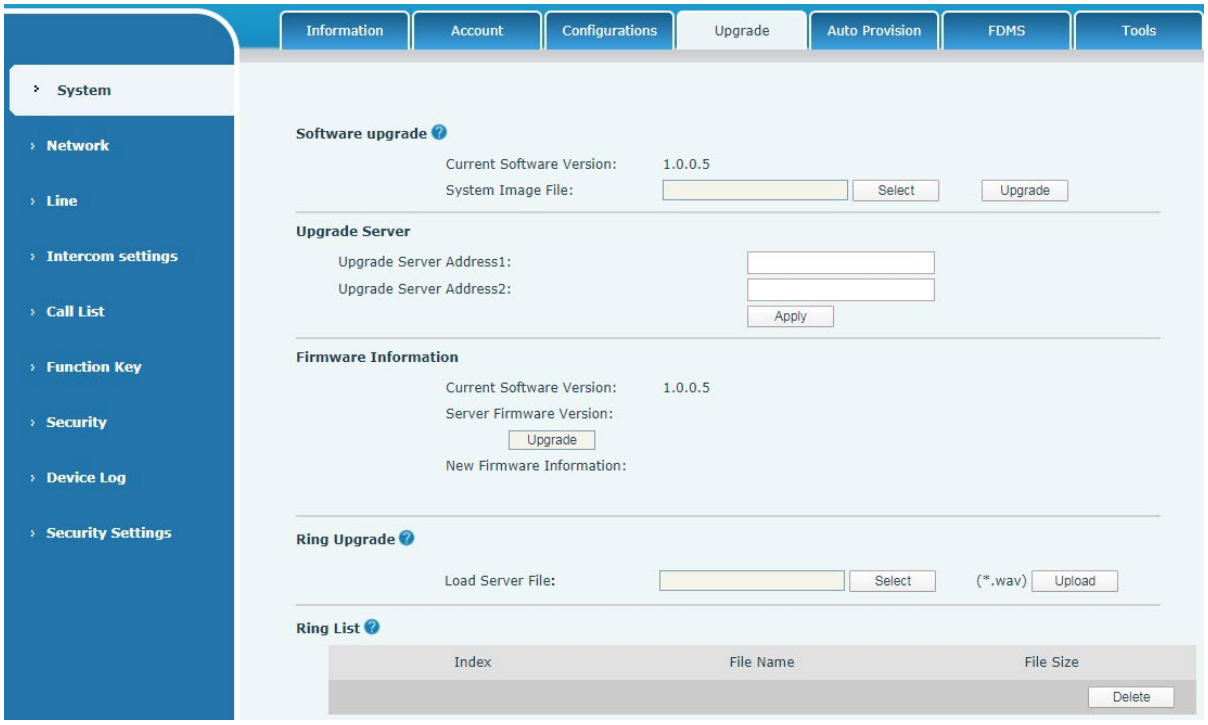
System >> Configurations



In this page, administrator 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 the administrator 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.

System >> Upgrade



On 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.

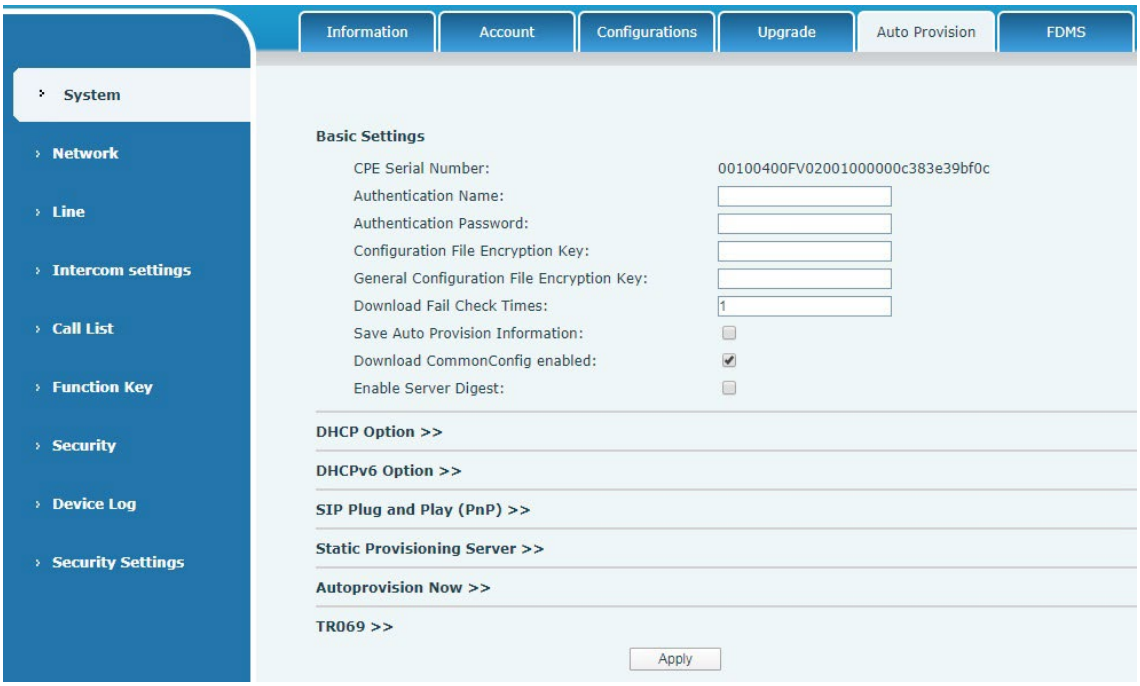
Online upgrade via Upgrade Server:
Online Firmware update is when a device sends an HTTP request to a server, the server replies with a corresponding description file or 404 or timeout. After device gets the reply, it analyzes the version description file and prompts the user whether to upgrade to the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

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

System >> Auto Provision

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



This device supports 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

Transfer protocol: FTP/TFTP/HTTP/HTTPS.

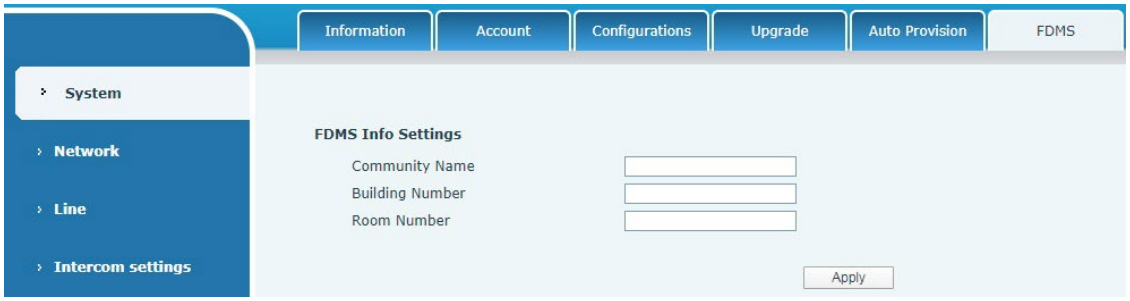
Parameters	Description
Basic settings	
CPE Serial Number	Serial number of the equipment
Authentication Name	Configure FTP server's username, TFTP server does not require this option. When using FTP server, device uses anonymous authentication name if user leaves this option blank.
Authentication Password	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 reaches the Download Fail Check Times number.
Enable Server Digest	When the feature is enabled, if the configuration file of server is changed or device's configuration is different from the 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 methods to perform auto provision. The option is disabled by default.
Custom Option Value	The custom option value should be the same the 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 to enable PnP or not. If PnP is enables, AC10v 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 the device. 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 access to a 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, the AC10v device will request the file which is named as its MAC address.
Protocol Type	Select transportation protocol type, the AC10v supports FTP/TFTP/HTTP and HTTPS
Update Interval	Set configuration file update interval time. As default it is 1, which means AC10v 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	Choose the ACS server type from the drop down list.
ACS Server URL	Input ACS server address.
ACS User	Input ACS server username.
ACS Password	Input ACS server password.
STUN Server Address	Enter the STUN address
STUN Enable	Select it to enable STUN.

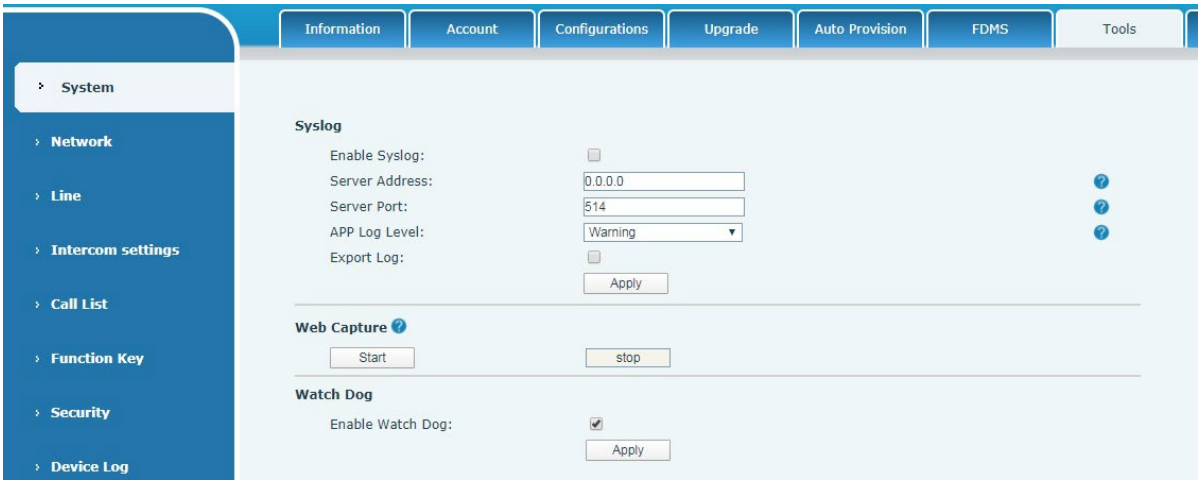
System >> FDMS



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

System >> Tools

This page provides users with tools for helping to diagnose issues.

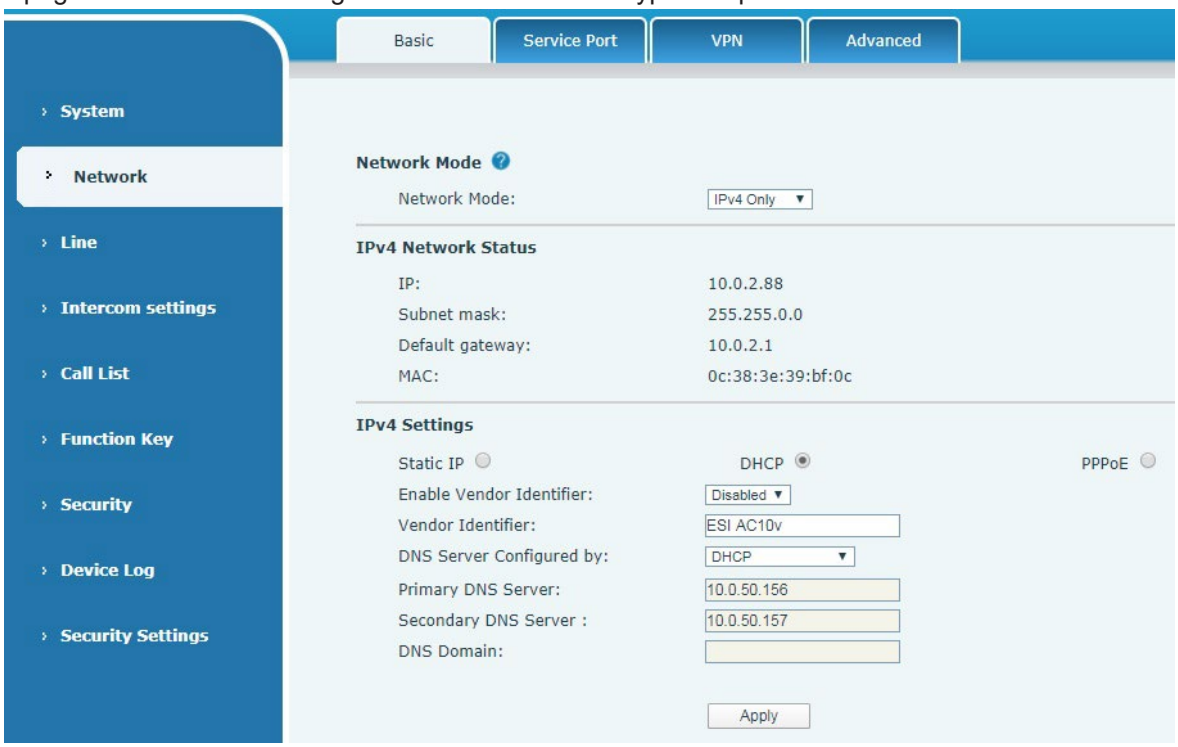


Syslog: When the user open syslog and sets the syslog server address, the log information of the device will be recorded in the syslog server during operation. If there is a problem that is not related to your internal network, send the logs to ESI support team to analyze.

For other details, please refer to the **Trouble Shooting** section.

Network >> Basic

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

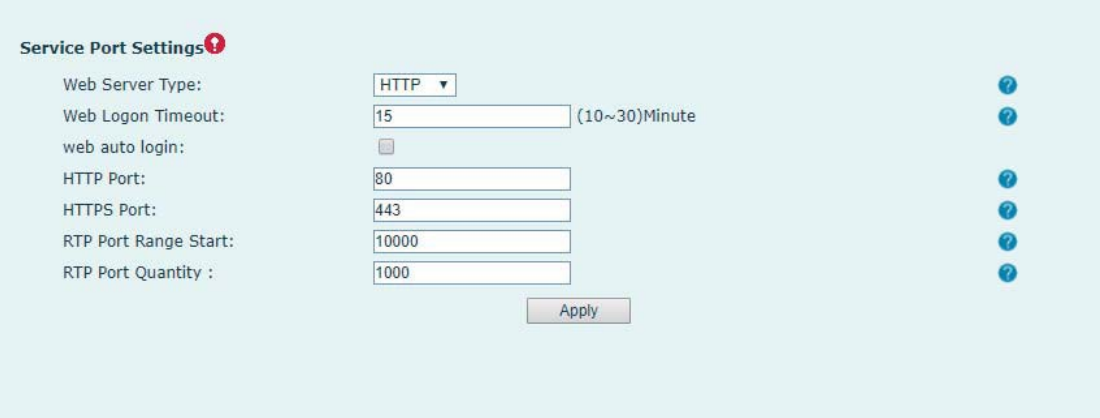


Parameters	Description
Network Mode	IPv4 only，IPv6 only，IPv4&IPv6
Network Status	
IP	The current IP address of the device.
Subnet mask	The current Subnet Mask of the device.
Default gateway	The current Gateway IP address.
MAC	The MAC address of the device.
Settings	
Select the appropriate network mode. 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. Please contact ISP or network administrator for this information.
DHCP	Network parameters are provided automatically by a DHCP server.
PPPoE	Account and Password must be input manually, which 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 identifier is enabled, you will see the vendor identifier information in the DHCP option60 field
DNS Server Configured by	Select the Configuration 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 setting 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.	

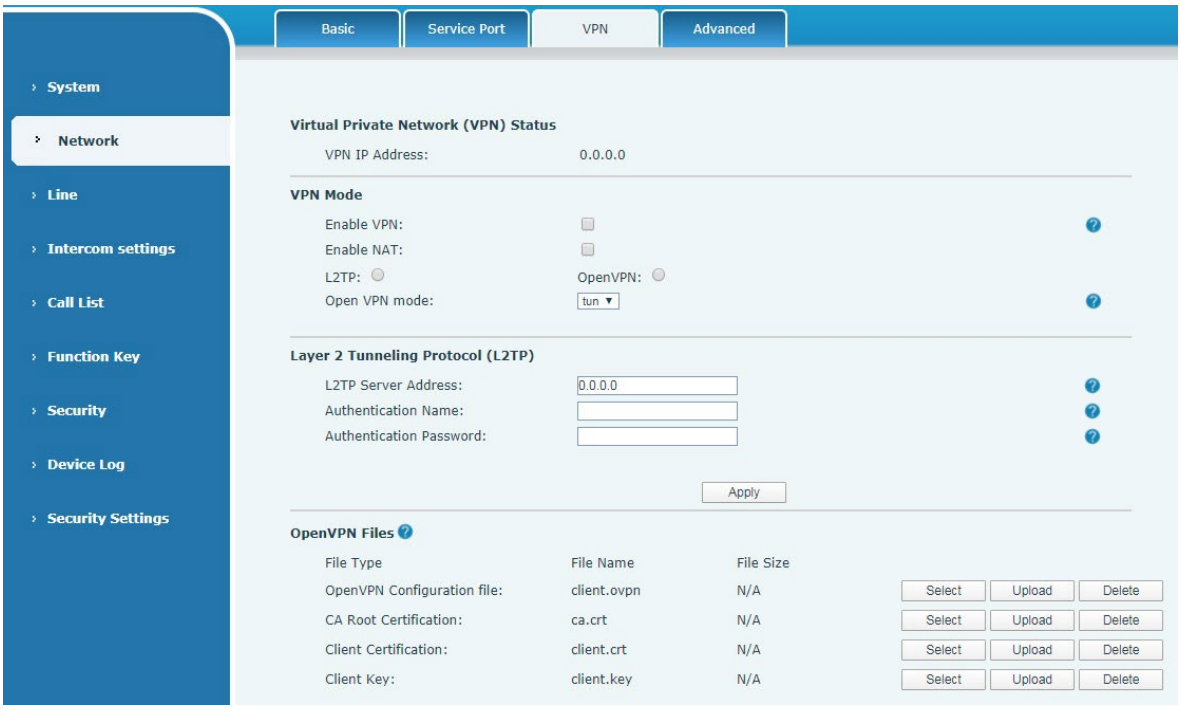
Network >> Service Port

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



Parameter	Description
Web Server Type	AC10v supports two kinds of web login: HTTP and HTTPS. Reboot the device for new setting to take effect.
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 Web Auto Login is enabled, 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 values,.such as :8080. Webpage login URL is: HTTP://ip:8080.
HTTPS Port	The default is 443. Using this method is similar to HTTP.
RTP Port Range Start	The value range is 1025 to 65535. The value of RTP port starts from the initial value. With each call, the value of the voice and video port will increment by 2.
RTP Port Quantity	Number of calls.

Network >> VPN



Virtual Private Network (VPN) is a technology that allows 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 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. Click “**Apply**” to save changes and device will try to connect to the L2TP server.

When the VPN connection is established, the VPN IP Address should be displayed in the VPN status option. There may be a connection delay. User may need to refresh the page to update the status.

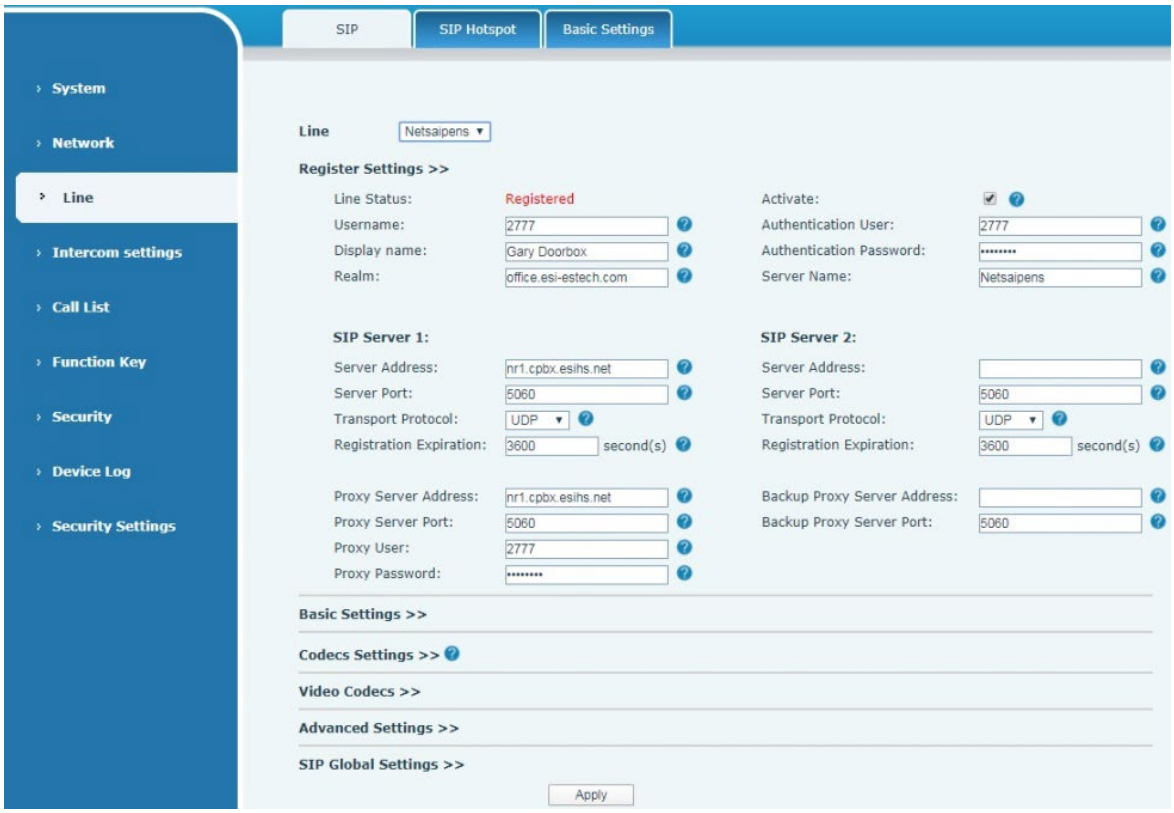
Once the VPN is configured, the device will try to connect with the VPN server automatically every time it boots up, unless user disable VPN. Sometimes, if the VPN connection does not establish immediately, user may try to reboot the device and check again.

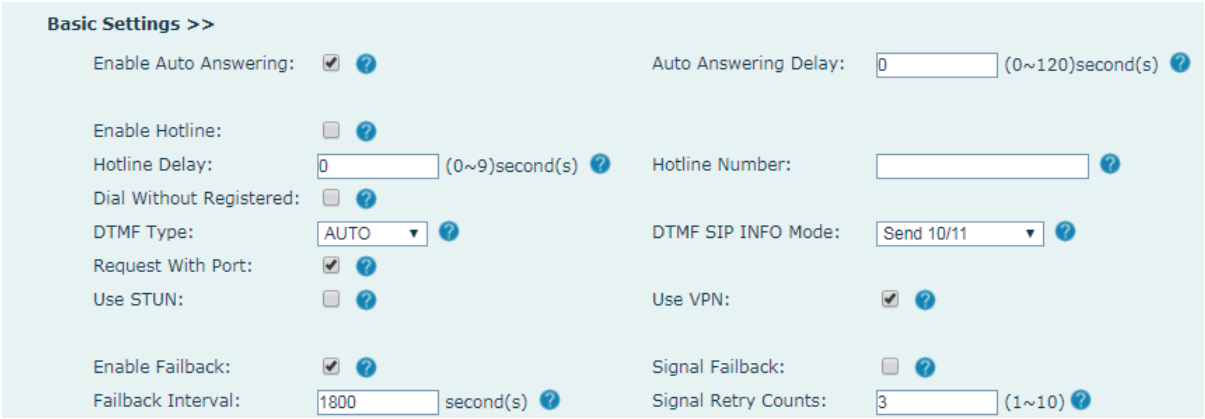
OpenVPN

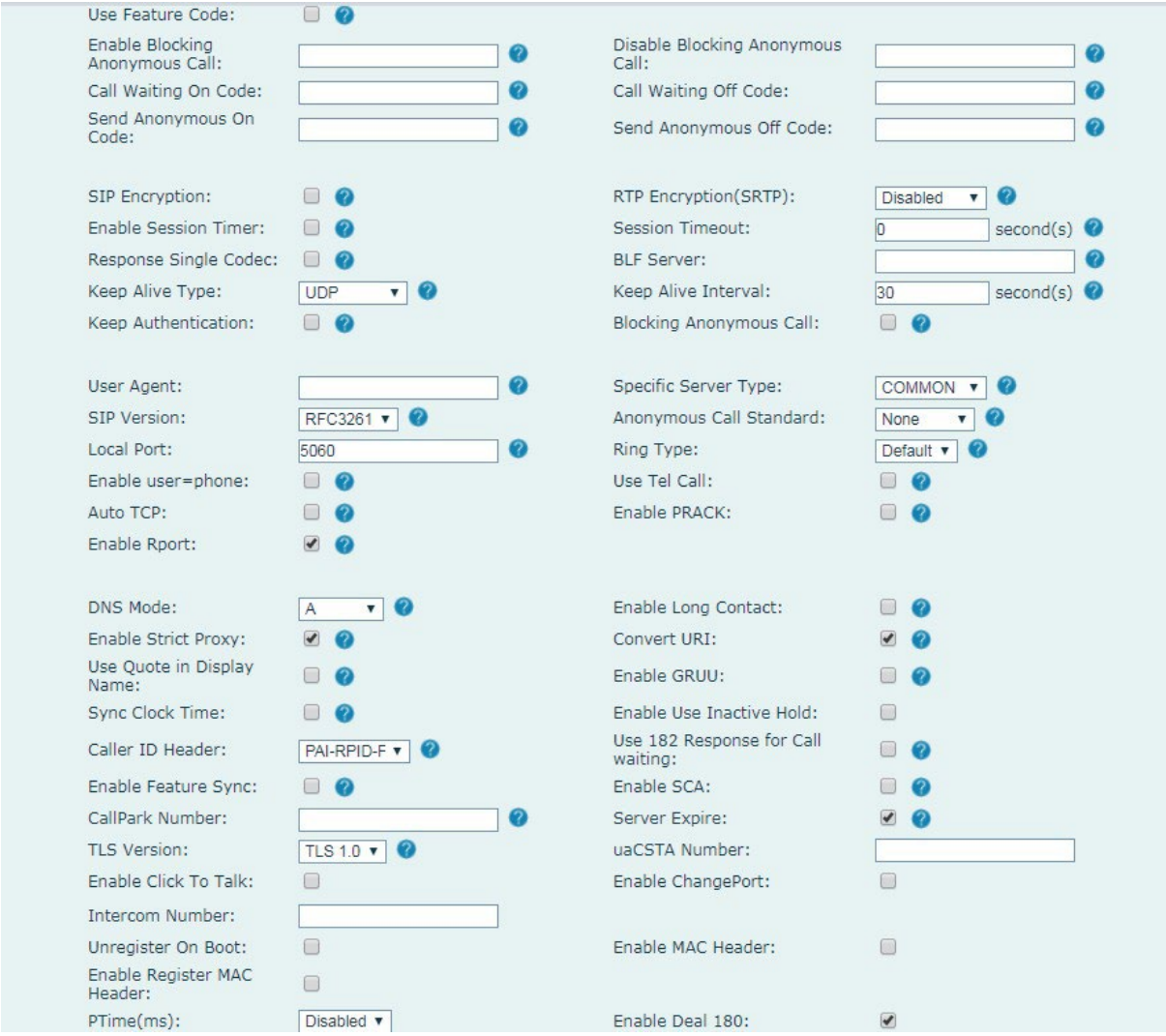
To establish an OpenVPN connection, user should get the following authentication and configuration files from the OpenVPN service provider and name them as follows,
OpenVPN Configuration file: client.ovpn
CA Root Certification: ca.crt
Client Certification: client.crt
Client Key: client.key

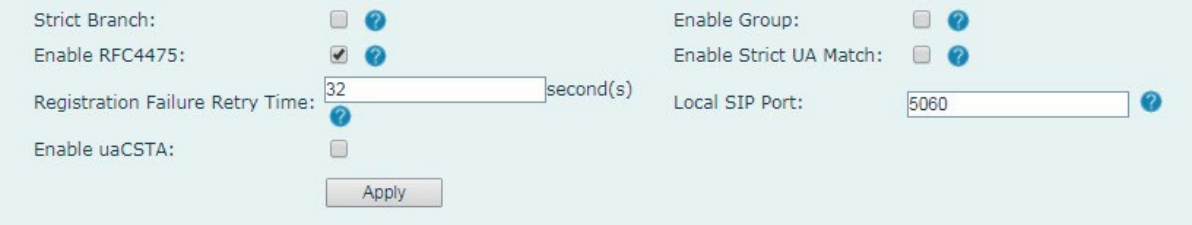
Select OpenVPN files and then click upload to upload these files to the device in the webpage **[Network] >> [VPN]**. Then user should check “**Enable VPN**” and select “**OpenVPN**” in VPN Mode and click “**Apply**” to enable OpenVPN connection. The connection will be established every time system boots up unless the user disable it manually.

Line >> SIP


SIP
SIP Hotspot
Basic Settings
System
Network
Line
Intercom settings
Call List
Function Key
Security
Device Log
Security Settings
Line
Netsaipens
Register Settings >>
Line Status:
Registered
Activate:
Username:
2777
Authentication User:
2777
Display name:
Gary Doorbox
Authentication Password:
Realm:
office.esi-estech.com
Server Name:
Netsaipens
SIP Server 1:
SIP Server 2:
Server Address:
nr1.cpbx.esihs.net
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:
nr1.cpbx.esihs.net
Backup Proxy Server Address:
Proxy Server Port:
5060
Backup Proxy Server Port:
5060
Proxy User:
2777
Proxy Password:
Basic Settings >>
Codecs Settings >>
Video Codecs >>
Advanced Settings >>
SIP Global Settings >>
Apply


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)





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	Check the box to activate the line.
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 is sent 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	Check the box to enable VPN
Use STUN	Check the box to enable STUN for NAT traversal
Enable Failback	Switch back to primary server when it becomes available.
Failback Interval	The time interval of detecting the availability of the main Proxy using Register message.
Signal Failback	When there are multiple proxy, check this box to allow the invite/register request to execute failback.

Signal Retry Counts	The number of times that the SIP will attempt to connect to proxy server.
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 (SRTP)	Enable RTP encryption, and RTP transmission will be encrypted.
Enable Session Timer	When the call timer is enabled, when the configuration item is switched on, the phone periodically sends the message and terminates the call without a reply.
Session Timeout	Set the session timer timeout period.
Response Single Codec	If this option is enabled, the device will use a single codec to respond to incoming call request.
BLF Server	Input BLF server address. 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 transmission interval.
Keep Authentication	Keep the previous authentication parameters.
Blocking Anonymous Call	Reject any incoming call that doesn't present caller ID.
User Agent	Set the user agent, the default value is the device model and 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	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 AC10v 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. “ESI” vs ESI.
Enable GRUU	Enable Globally Routable User-Agent URI (GRUU) or not.
Sync Clock Time	Time Sync with server.
Enable Use 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 for 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	Use the timeout of 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	Check box to enable Change Port.
Intercom Number	Set intercom number.
Unregister On Boot	Check box to enable logout function.

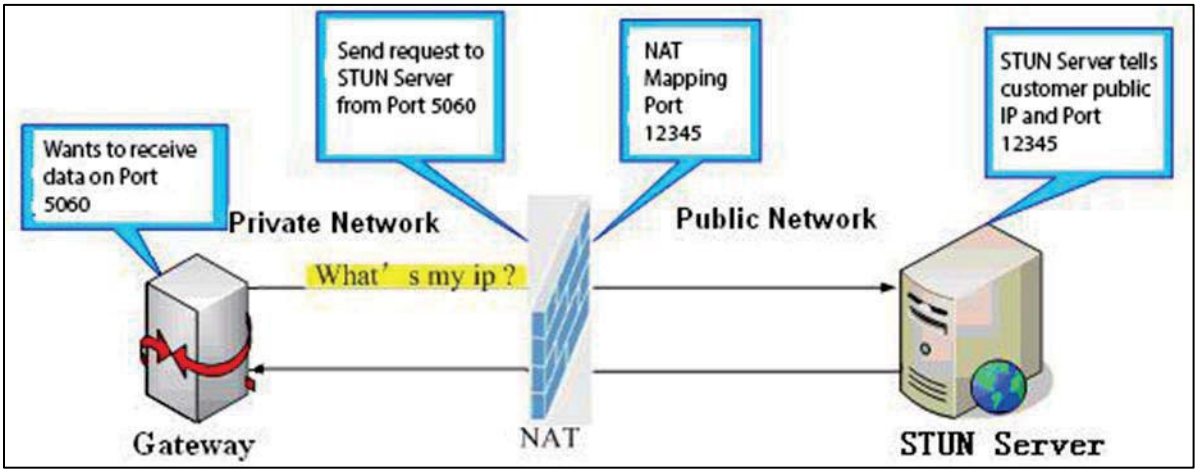
Enable MAC Header	Whether to enable MAC header. When enable, there is MAC information in SIP packet and user agent when register
Enable Register MAC Header	Register MAC Header
PTime(ms)	Select a PTime interval from 10-60 ms.
Enable Deal 180	Enable: After receiving183+ SDP, device will play ivr. After receiving180, device will play local tone. Disable: After receiving 183+ SDP, device will play ivr. After receiving180, 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.

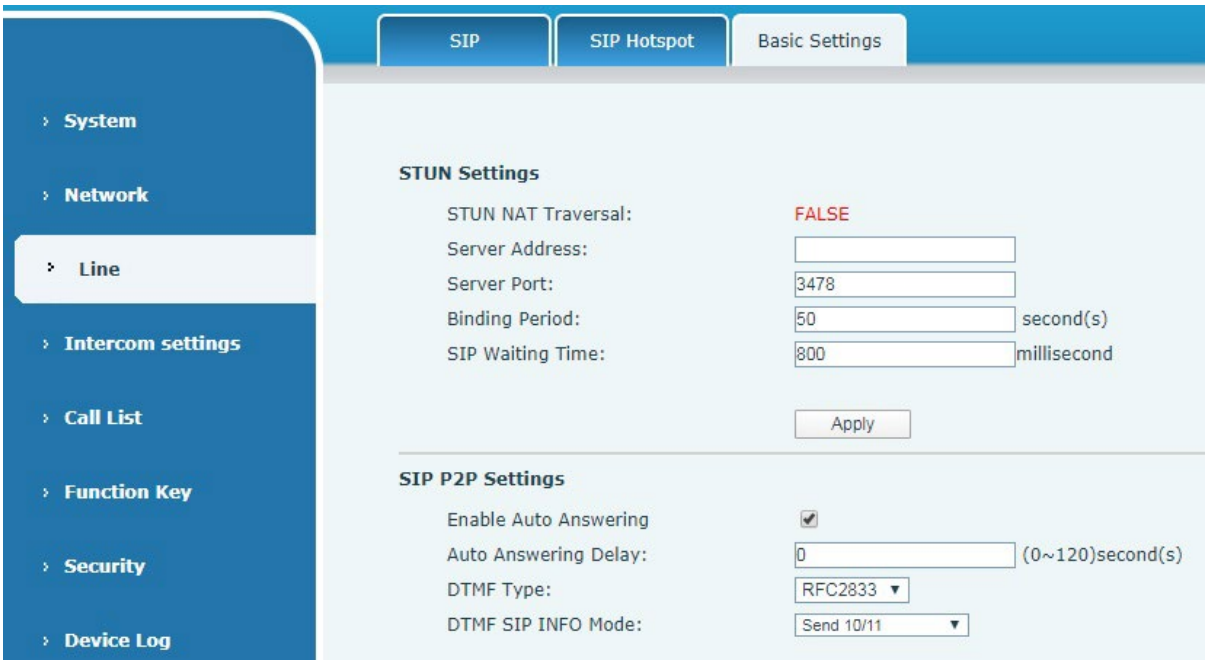
Line >> SIP Hotspot

SIP hotspot is a feature that will allow multiple devices to share the same number.
Please see **Hotspot** for more details.

Line >> Basic Settings

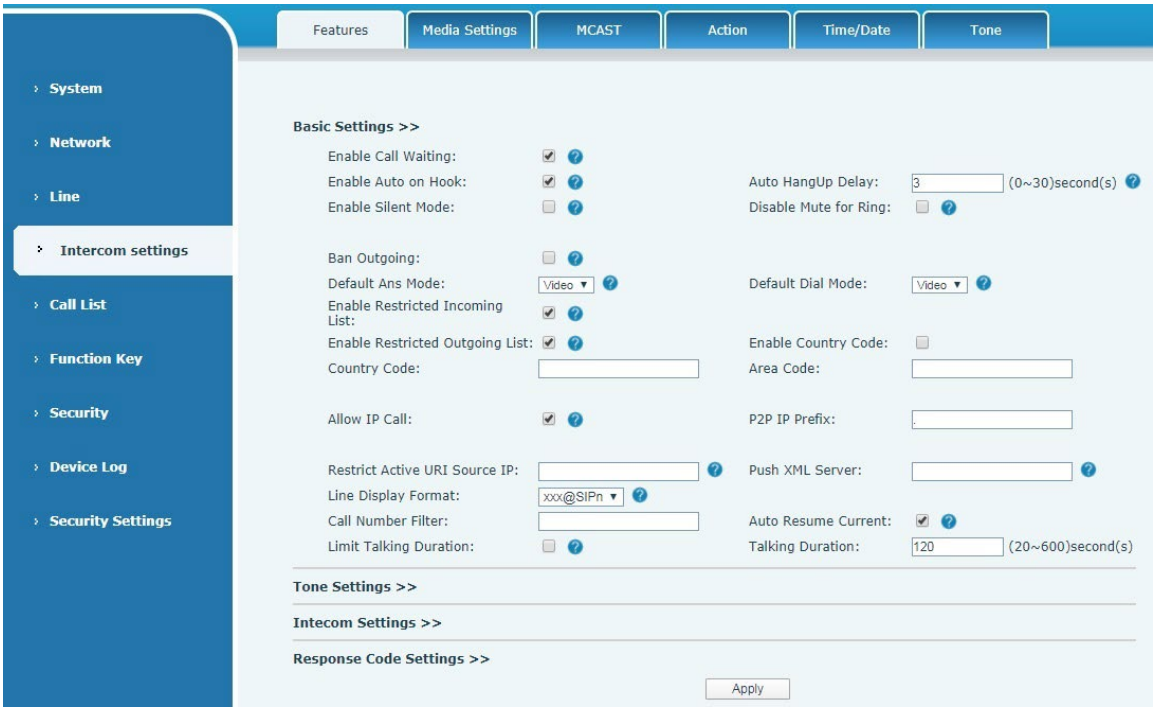
STUN -Simple Traversal of UDP through NAT. A STUN server allows the device in a private network to know its public IP and port as well as the type of NAT being used. The equipment can use this information to register itself to a SIP server so that it can receive calls from public network while it is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





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'

Intercom Setting >> Features



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 HangUp Delay	Specify a time for the device to hang up and return to an idle state automatically.
Enable Silent Mode	When enabled, the device is muted, there is no ringing for incoming 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 is disabled.
Ban Outgoing	If enabled, the device cannot dial out.
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 was 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 dials digits for a call, by default it is enabled.
Play Talking DTMF Tone	Play DTMF tone on the device when user presses 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	When enabled, 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 rejects call.

Configuring ePhoneX phone for intercom video

Preparing your eX for intercom video on ESI eSIP

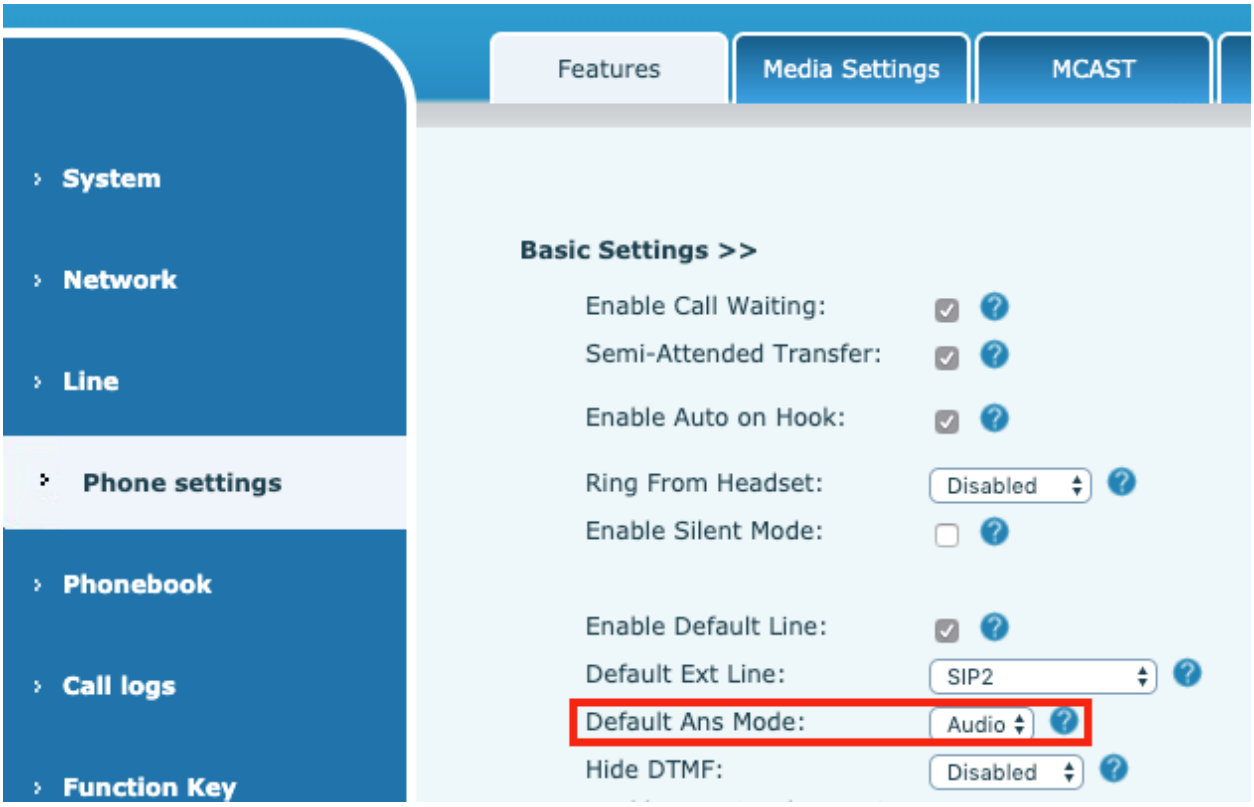
There are a few GUI changes that need to be made to the ePhoneX that can streamline the performance of viewing a video call and setting a button to open the door lock.

Note, when a video call is received by the ePhoneX the 10 main screen DSS keys will not function once the call is answered and video is played.

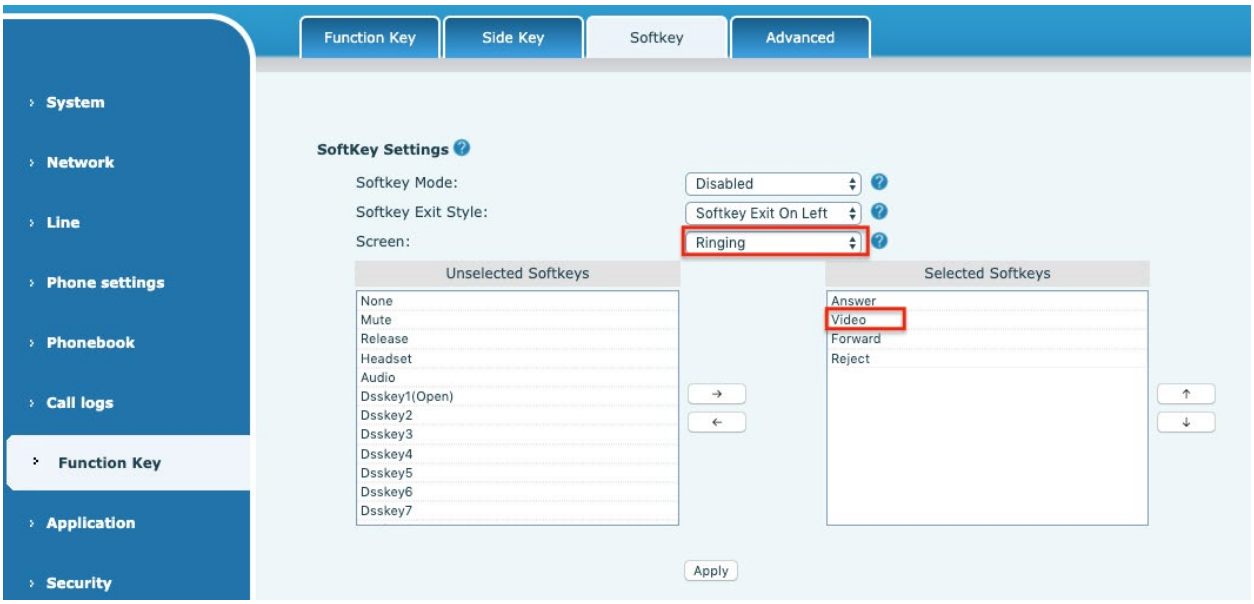
Create Video Answer key

This key allows the user to select between video and audio calls in an inbound state. Since the eSIP PBX is a true B2BUA the H264 media codec is present on all invite SDP attributes even if the call is sourced from a non-video capable phone.

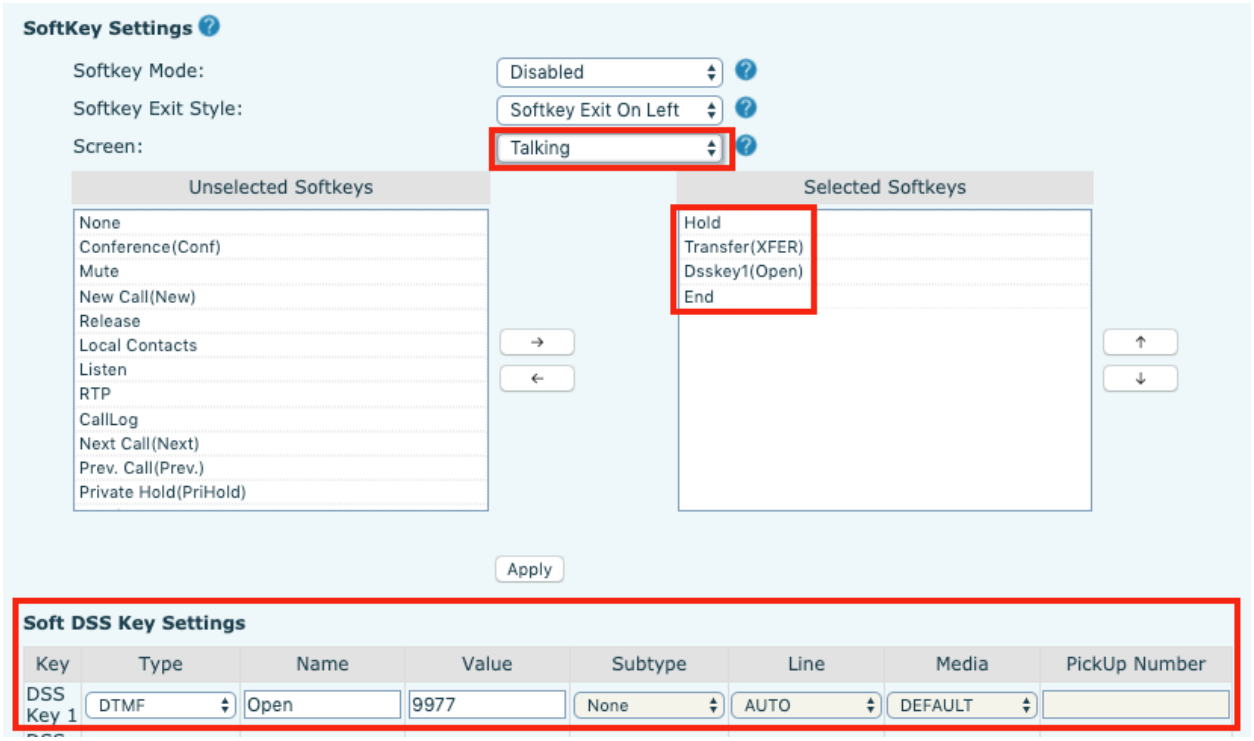
1: Navigate to Phone Settings >> Features and change Default Ans Mode to Audio.



2: Navigate to Function Key >> Softkey, and change screen to ringing. Remove the None key and add the Video key. This will provide you with a new softkey only for answering video calls.

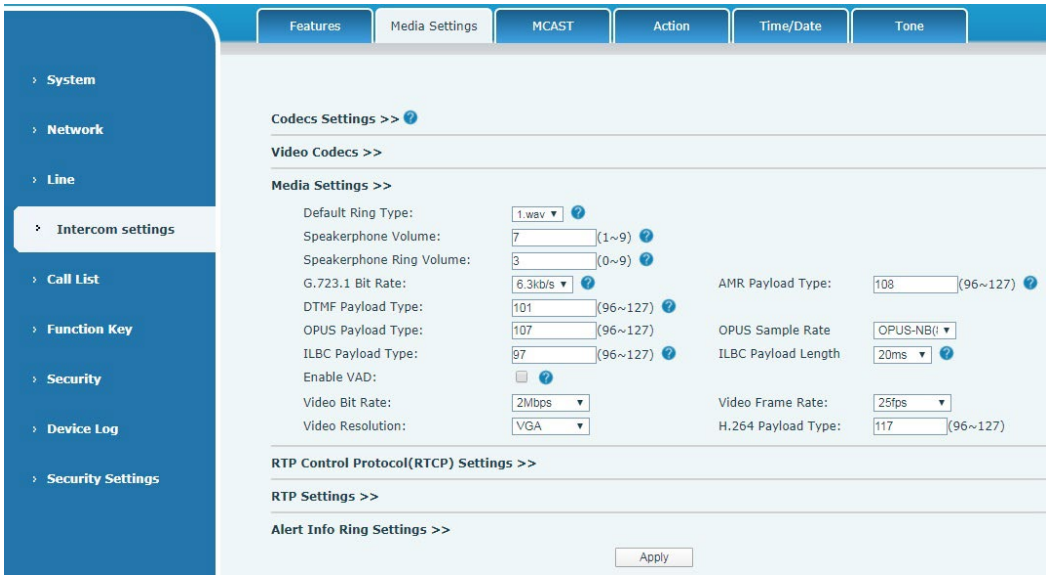


3: On the same page under DSS Soft key settings create a DTMF key that utilizes your video intercoms DTMF open code. Then modify the screen type to talking and add that key to the soft keys that will appear under that screen.



Now these keys will streamline the use of the video intercoms on the eSIP platform as soft keys will be utilized vs a DSS or PFK key.

Intercom Setting >> Audio



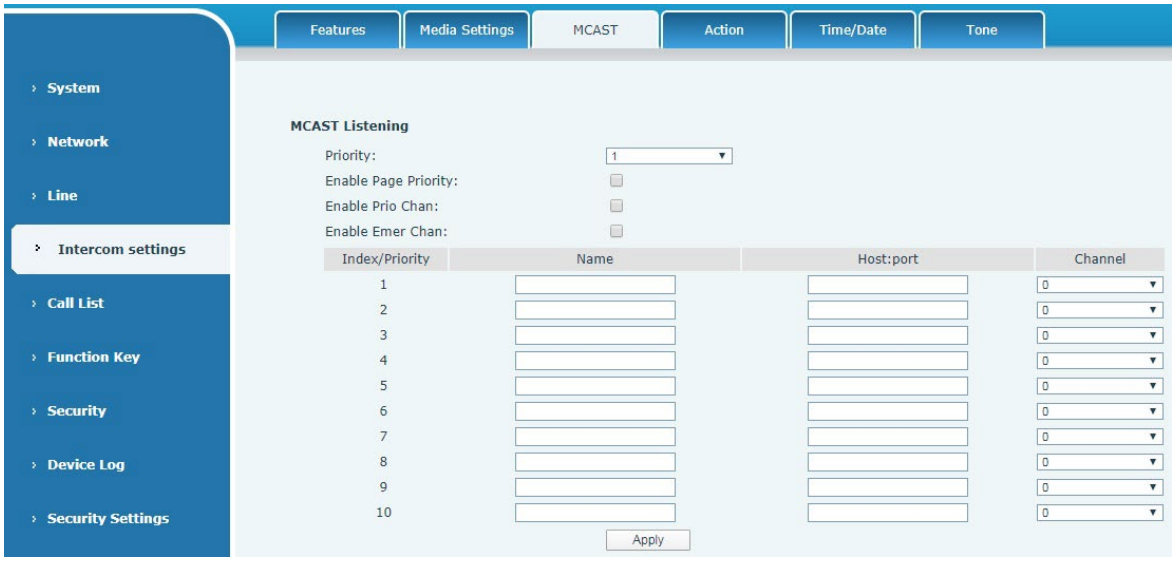
Parameter	Description
Codecs Settings	Enable or disable voice codecs: G.711A/U, G.722, G.729AB, iLBC, opus.
Media Settings	
Default Ring Type	Configure default ringtones. If no special ringtone is set, the default ringtone will be used.
Speakerphone Volume	Set the speaker volume, value can be 1~9.
Speakerphone 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 payload 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 range is 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.

Intercom Setting >> MCAST

It is easy and convenient to use the multicast function to send notice to each member of the multicast group by setting the multicast key on the device and sending a multicast RTP stream to a pre-configured multicast address. By configuring monitoring multicast address on the device, the device will receive multicast from the configured monitoring multicast address.

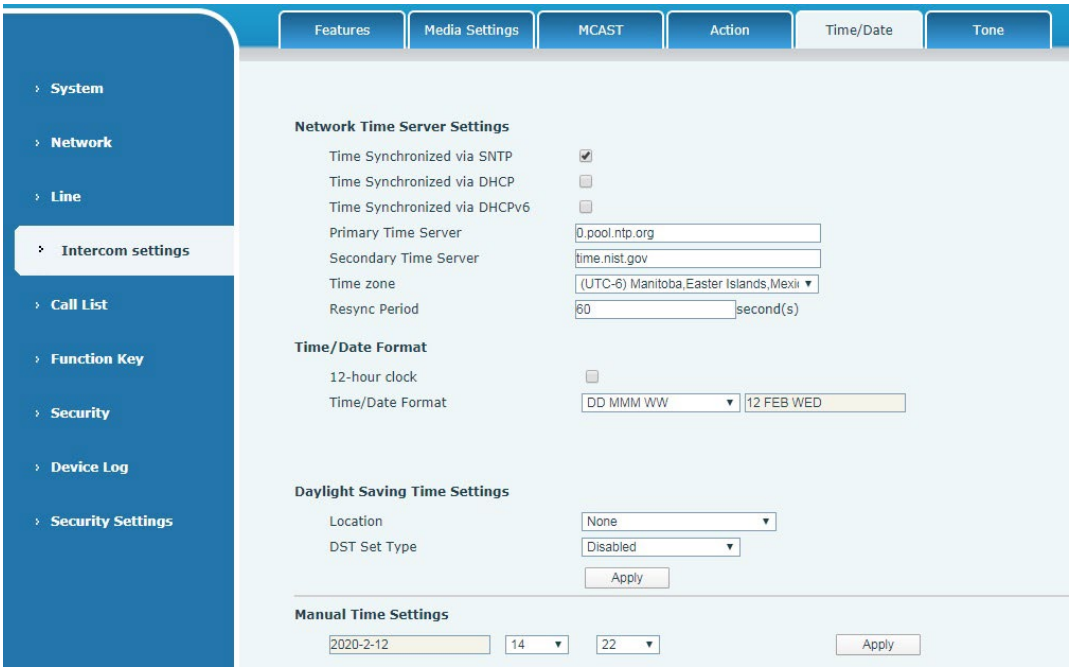


Intercom Setting >> Action

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

Intercom Setting >> Time/Date

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

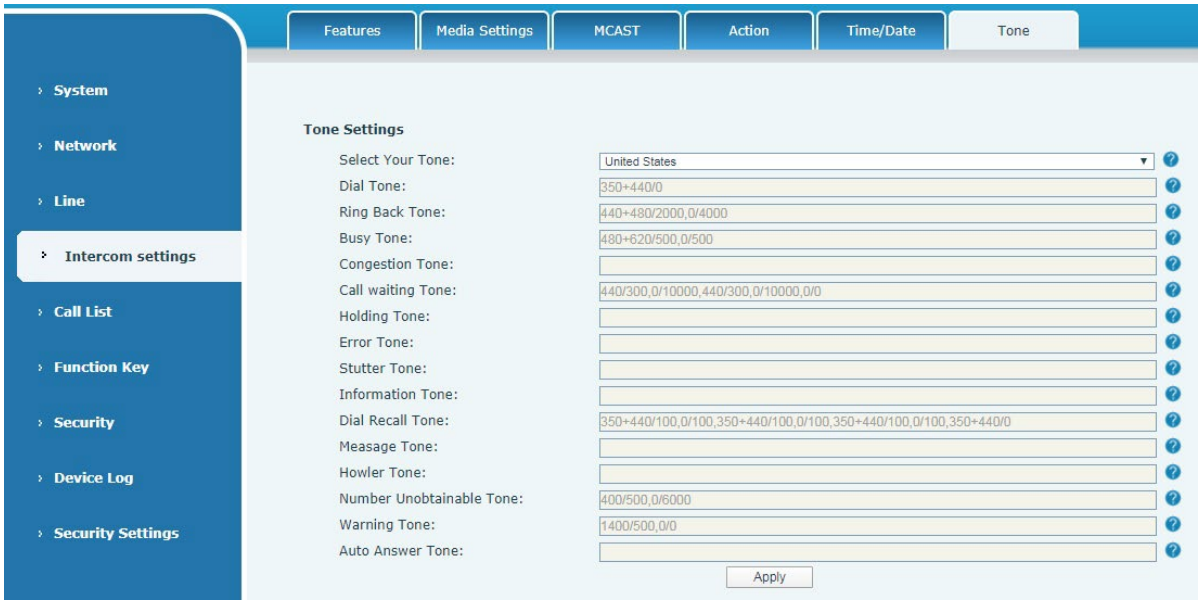


Parameter	Description
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol.
Time Synchronized via DHCP	Enable time-sync through DHCP protocol.
Time Synchronized via DHCPv6	Enable time-sync through DHCPv6 protocol.
Primary Time Server	Set primary time server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

Intercom settings >> Tone

User can set device's tone in this page.
You can select the corresponding country and use the default settings, or select custom and set the tone manually.



Call List >> Call List

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

Restricted Incoming Calls:

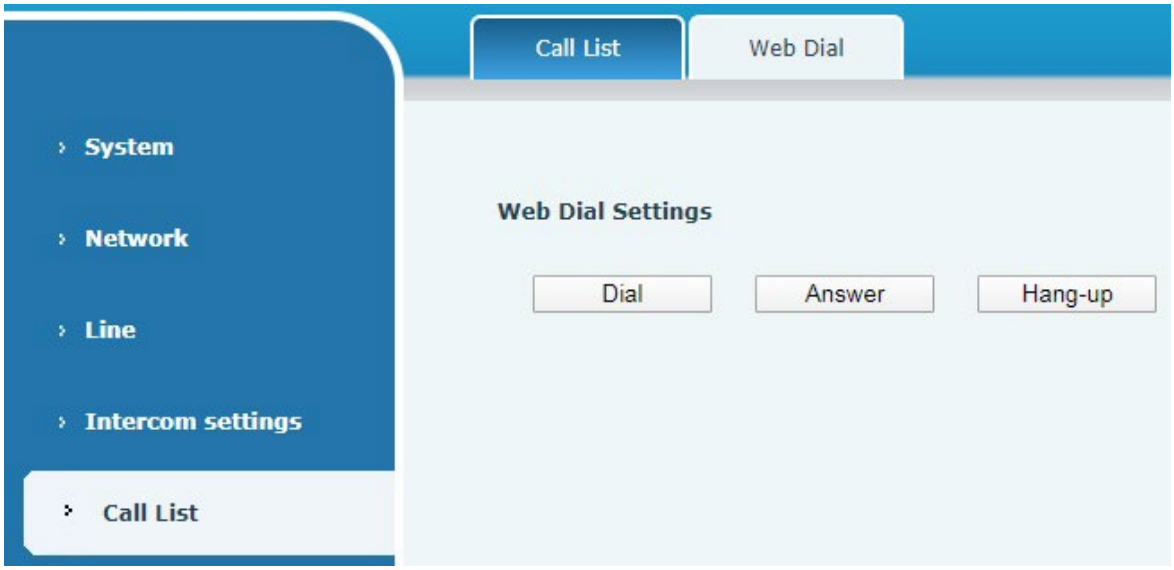
The function is same with blacklist. Add the numbers in restricted incoming calls list, the device will reject all the calls from these blacklist numbers, unless user deletes the numbers from the list.
User can add both numbers and prefix in the restricted incoming calls list, the AC10v series device will reject all the calls from the blacklist numbers or calls from numbers with blacklist prefix.

Restricted Outgoing Calls:

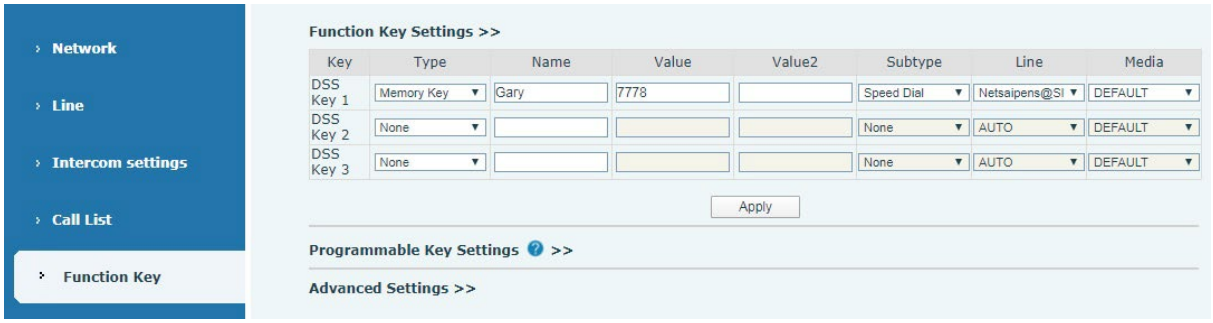
Add numbers to restricted outgoing calls list, the device will end the call when user dial these numbers, unless user removes the numbers from the list.

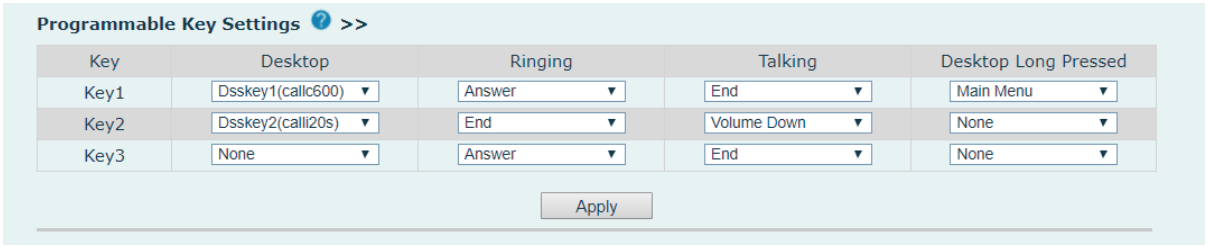
Web Dial

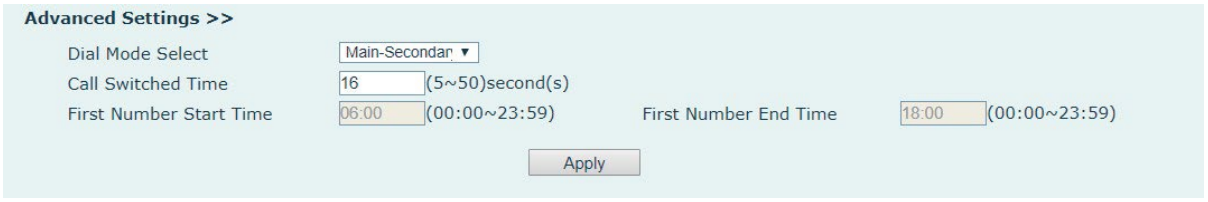
In this page, user can make calls, answer the calls or hang up the calls.



Function Key







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 makes the operator or secretary 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 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 settings.
Desktop Long pressed	None: None. Main Menu: Long press the key to make device go to command mode. For details, see **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 sends 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.

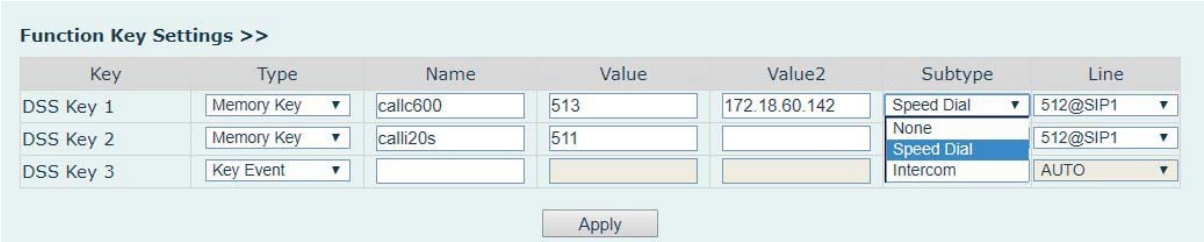
Key Event

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

Type	Subtype	Usage
Key Event	None	Disabled
	Handfree	Handfree

Memory Key

When the speed dial key is set as Memory Key, the device would dial a 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.



Type	Value	Line	Subtype	Usage
Memory Key	Enter number to call when key is pressed	Select the desired SIP line	Speed Dial	Set speed dial, press the key to call out to the number.
			Intercom	In Intercom mode, if the caller's IP phone supports Intercom feature, the device can automatically answer the Intercom calls

Multicast

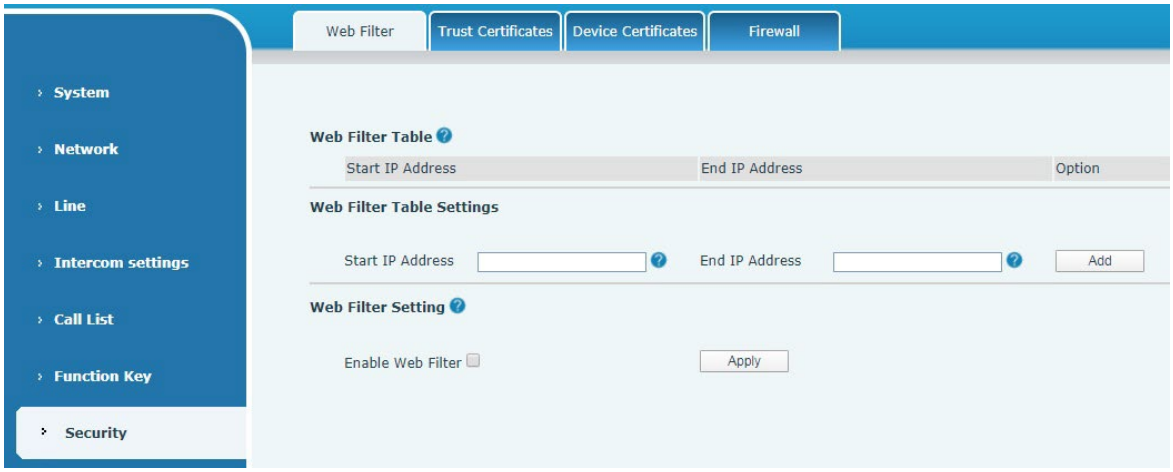
Multicast function is to deliver voice streams to configured multicast address; all equipment monitoring the multicast address can receive and play the broadcast.

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

Type	Value	*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

Security >> Web Filter

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

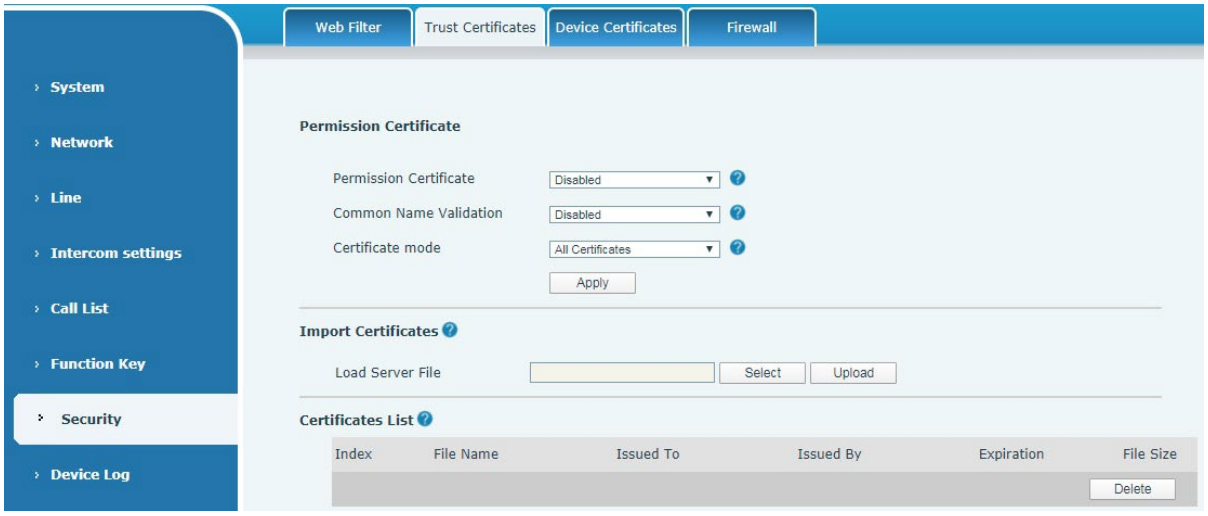


Add or delete the allowed IP address segment. Input start IP address in Start IP address field, and input end IP address in End IP address field and click Apply to save the settings. Click Delete to remove the corresponding IP address segment.
Enable Web Filter: Enable or disable web filter, select it and click apply to save the settings.

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

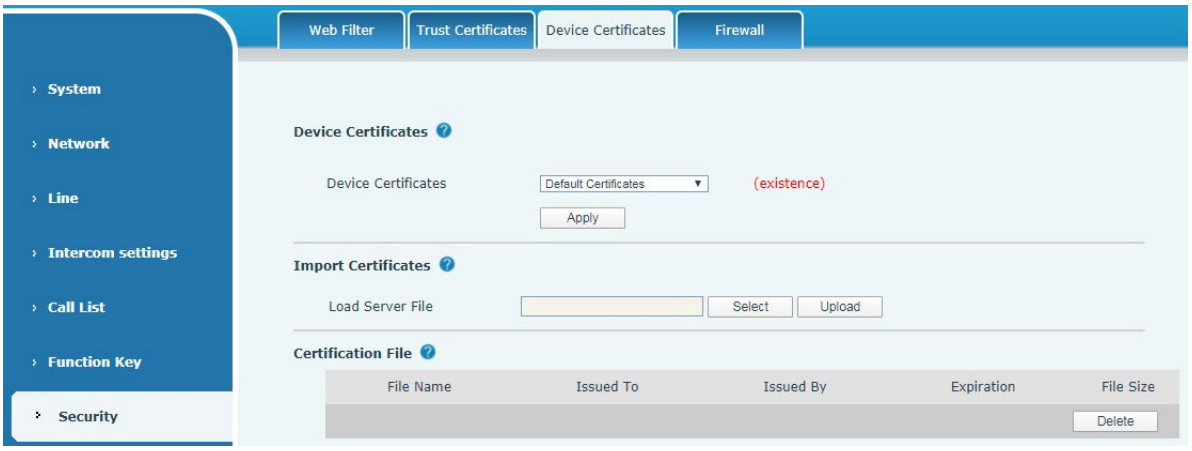
Security >> Trusted Certificates

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

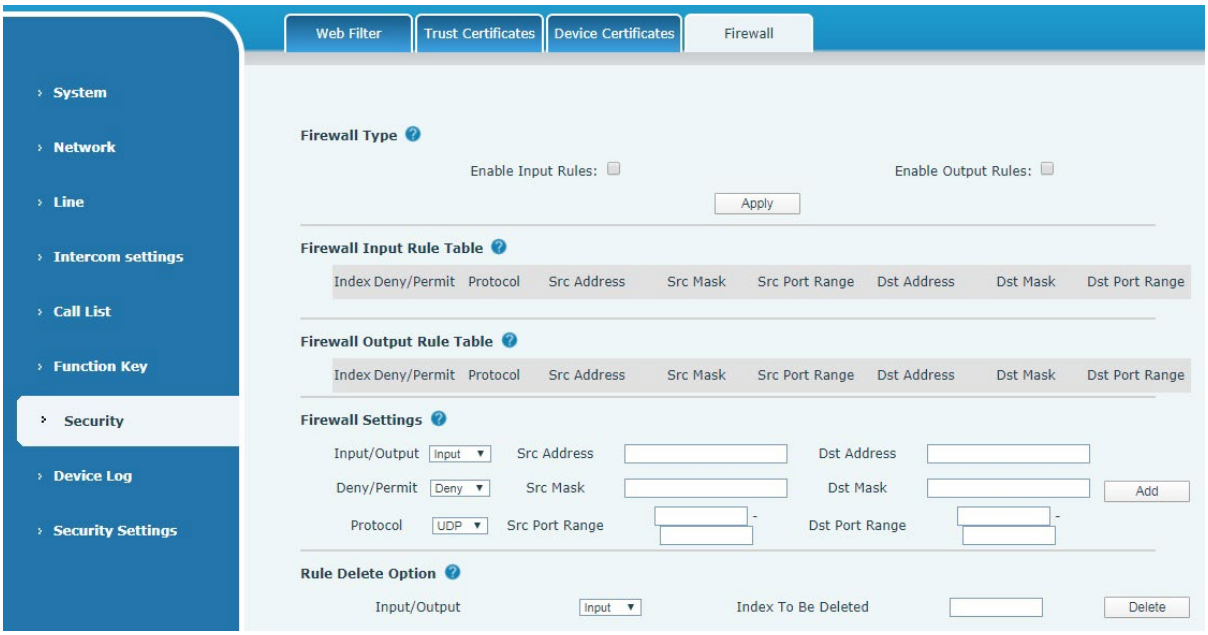


Security >> Device Certificates

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



Security >> 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 security.

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 deny rule or permit 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 configures destination mask to be 255.255.255.255, the destination IP should be one specific 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.

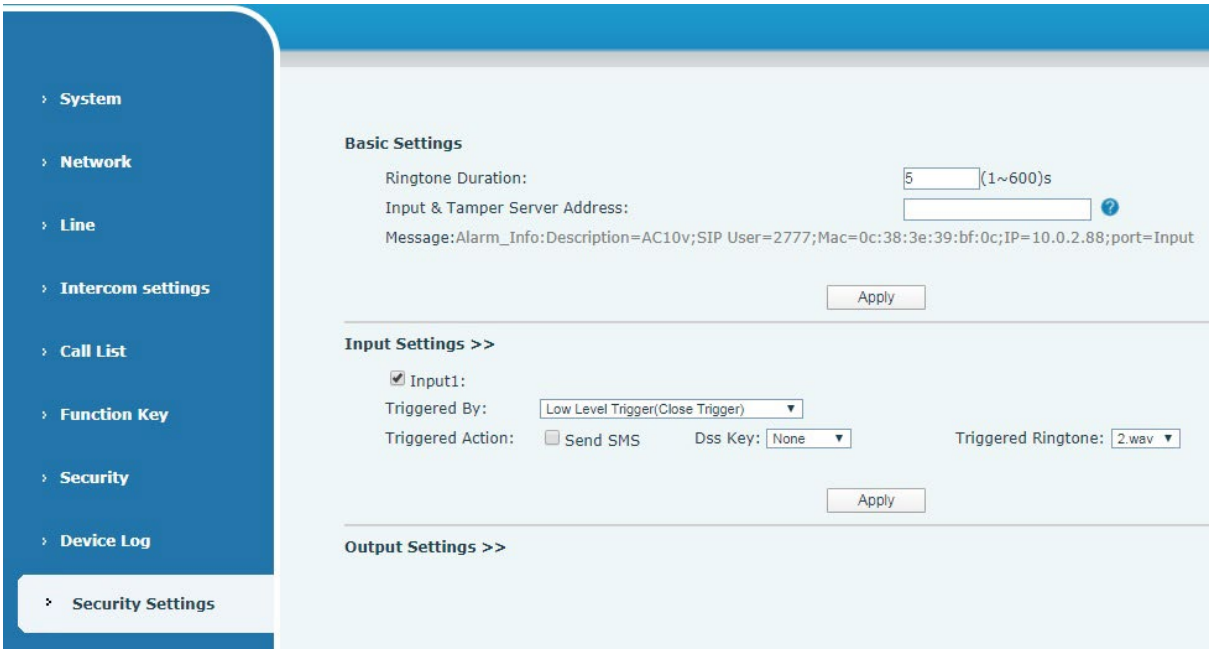
Input the parameters and click Add. The new rules will be added to the firewall list, for example:

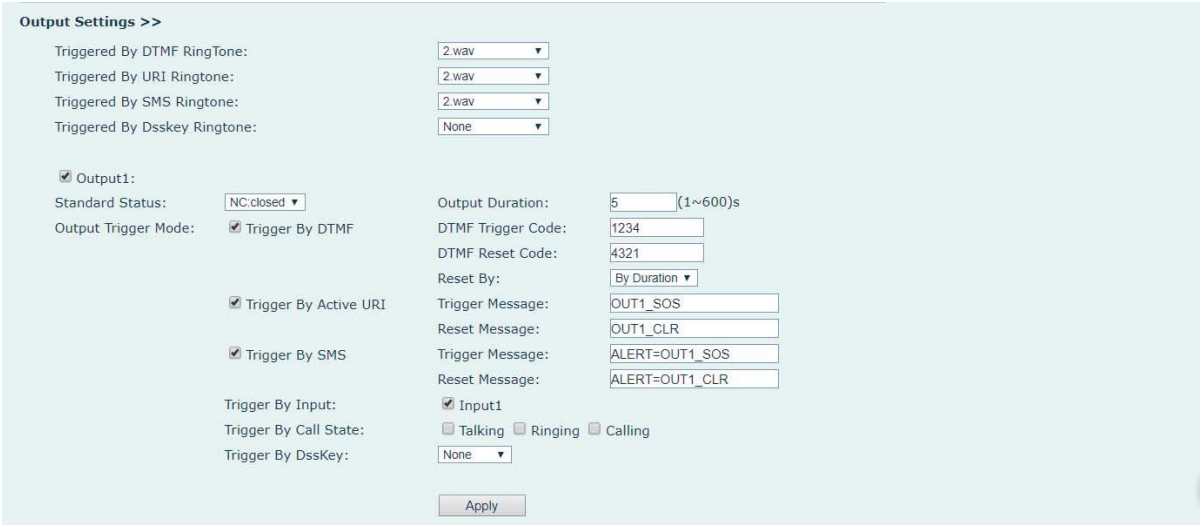
Select Input/Output rule, and enter the index of the rule in “Index To Be Deleted” option, click delete and the corresponding rule will be removed.

Device Log

In this page, user can get the device's logs. When device works abnormally, user can get logs for diagnosing the problem. For details see **Get Log Information**.

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=AC10v;SIP User=;Mac=0c:38:3e:3a:06:65;IP=;port=Input .
Input settings	
Input1	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	Perform one of the folling. **Send SMS:** Set the alert message send to server. **Dss Key** is pressed. The device will perform selected Dss Key configuration. Default value is none. **Triggered Ringtone:** Select triggered ring tone.

Output Settings	
Output1	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	(NO: normally open) is triggered when condition is met.
	(NC: normally close), is trigger when condition is met.
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 Active URI	Enable or disable trigger by URI. User can send commands from remote device or server to the device. If the command is correct, the device will trigger corresponding output port.
Trigger Message	Input trigger message for trigger by URI mode.
Reset Message	Input reset message for trigger by URI mode.
Trigger by SMS	Enable or disable trigger by SMS. User can send ALERT command to device. If the command is correct, the device will trigger corresponding output port.
Trigger Message	Input trigger message for trigger by SMS mode.
Reset Message	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 the dsskey is selected, the output port will be triggered.

Trouble Shooting

When the device does not work properly, user can try the following methods to restore the device to normal operation or collect relevant information for diagnostics.

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.

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.

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.

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 when capture is complete. Network packets are saved in a file.

Get Log Information

Log information is helpful when encountering a problem. In order to get the log, the user can log in to 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.

Device will not boot up

1. Check power connection and confirm the power adapter or PoE switch.
2. If the device goes to POST Mode (the LED flashes slowly), it means the device is damaged. Please contact ESI technical support for assistance.

Device will not register to a service provider

1. Check network cable connection and confirm the device is connected to internet.
2. If the network connection is good, check your SIP line configuration again. If all configurations are correct, please contact your service provider for support or a registration network packet capture according to the instructions in "**Network Data Capture**" for analysis.