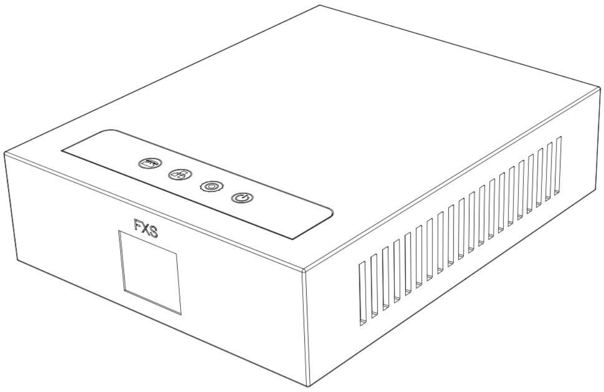


GA10&GA11
User Manual

Software Version:2.4.0

Release Date：2021/10/25

Directory

1 Picture

2 Table

3 Safety Instruction

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

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

4 Overview

4.1 Overview

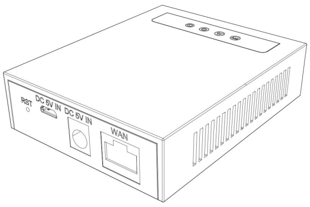
GA10&GA11 is a SIP analog gateway (ATA) with a single FXS port, featuring high-definition voice, 2 SIP accounts, three-way conference calls, etc.; through the convenient use of VoIP equipment to transfer analog calls to SIP calls, users are supported to call IP phones with ordinary analog phones. GA10&GA11 provides individual, enterprise, and operator user ports with a convenient, stable, and cost-effective communication solution to realize the deployment of VoIP.

5 Install Guide

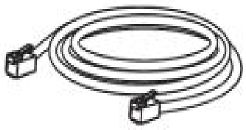
5.1 Use external Power Adapter

Please use the specified power adapter to eature the normal operation of the device.

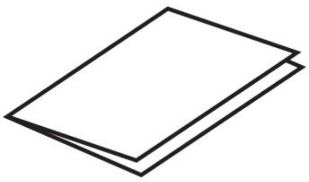
5.2 Packing Contents



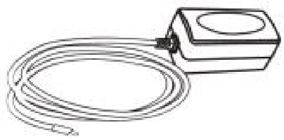
ATA Gateway



Ethernet Cable



Quick Installation Guide



Power Adapter

5.3 Interface description

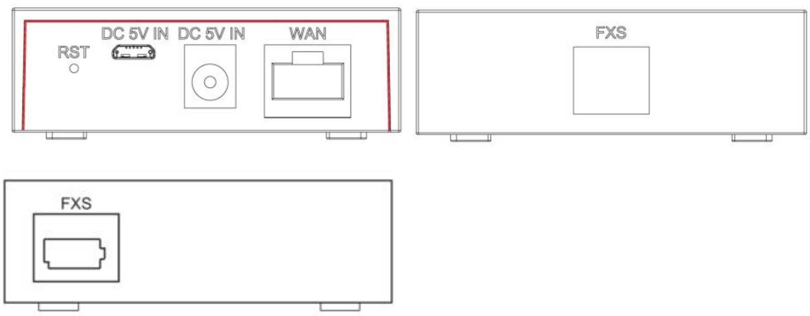


Table 1- Interface

Port	Icon	Function
Power	DC 5V IN	Connecting to a power source to the micro-USB or DC Jack

Network	WAN	Connecting to the network.
FXS	FXS	Use RJ11 telephone cable to connect to an analog phone. (GA10). Use BT plug telephone cable to connect to an analog phone. (GA11).
Reset	RST	Restore Default button. When the device is working properly, if you press this button (6 seconds) with a sharp object (such as a pencil) until the LED fast twinkling. Restore function will take effect after you release it.

5.4 Device Connection

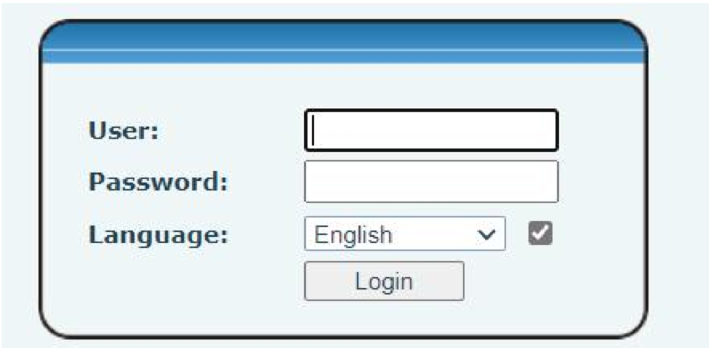
Please refer to the following steps to connect your device:

- Insert a standard RJ11/ BT plug telephone cable into the FXS port and connect the other end of the telephone cable to a standard touch-tone analog telephone.
- Insert the ethernet cable into the WAN port of the device and connect the other end of the ethernet cable to an uplink port (a router or a modem, etc.)
- Insert the power adapter into the device and connect it to a wall outlet.
- The power and ethernet LEDs will be solidly lit when the device and network is ready for use.

6 Introduction to the User

6.1 Web Management

Phone can be configured and managed on the web page of the phone. The user needs to enter the IP address of the phone in the browser at first and open the web page of the phone.



Picture 1- web login

Users must correctly enter the user name and password to log in to the web page. The default user name and password are "admin". For the specific details of the operation page, please refer to page 9 Web configurations.

6.2 Network Settings

The factory default of the device is DHCP mode. When the device is successfully connected according to 5.4 Connecting the device, you can obtain or modify the IP address of the device by connecting to an analog phone.

Support 3 ways to access the Internet:

DHCP

Static IP address

PPPoE

6.2.1 Read the IP

Dial "#*111" on the phone, the device will announce the IP address by voice

6.2.2 Set Static IP

The following describes how to set a static IP address for the device.

IP address: 192.168.10.123

Subnet mask: 255.255.255.0

Device address: 192.168.10.1

Primary DNS address: 8.8.8.8

Secondary DNS address: 114*114*114*114

Step 1. Dial #*50 on the phone, input the static IP address 192*168*10*123 to be set after hearing "Please input", and press # to end after inputting. After finishing, you can hear the prompt of successful setting.

Step 2. Dial #*51 on the phone, input the device to be set 192*168*10*1 after hearing "Please input", press # to end after inputting, and you can hear the prompt of successful setting after finishing;

Step 3. Dial #*52 on the phone, input the DNS 8*8*8*8*114*114*114*114 to be set after hearing "Please input", and press # to end after inputting, and you can Hear the prompt of successful setting;

Step 4. Dial #*53 on the phone, input the subnet mask to be set 255*255*255*0 after hearing "Please input", press # to end after inputting, and you can hear the setting successful hint;

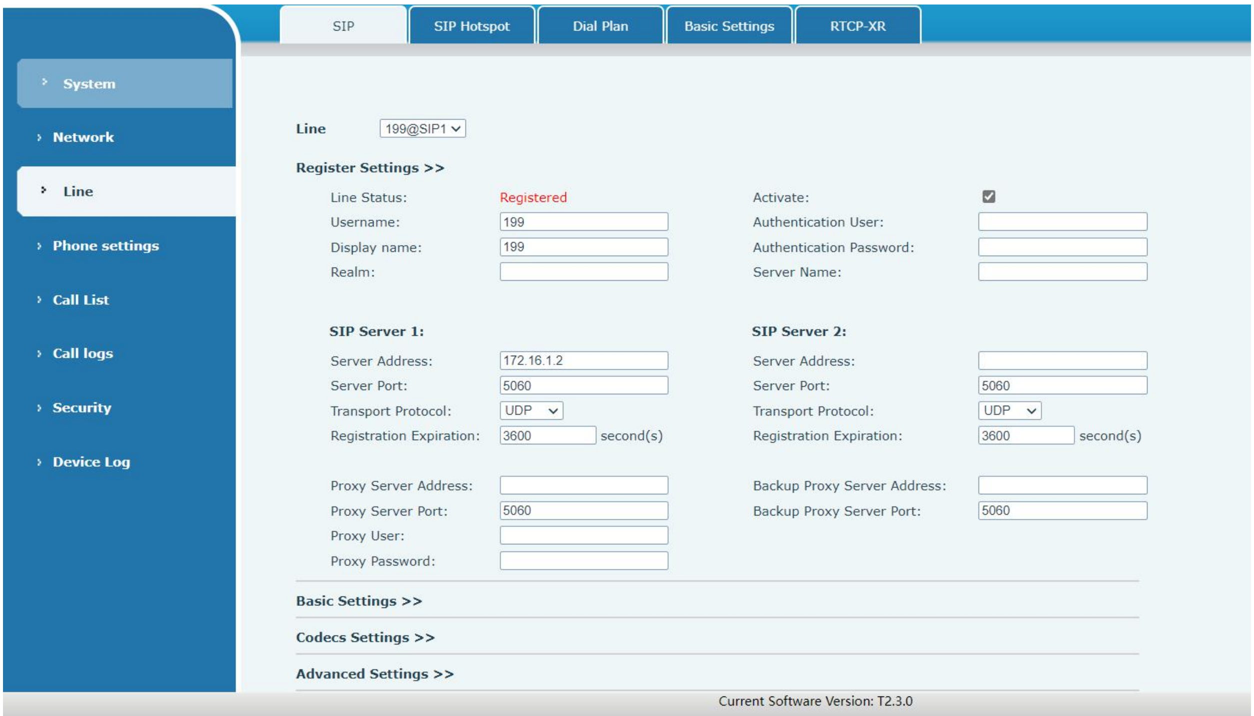
Step 5. Dial #*100 on the phone, switch to static mode, and use the static IP address set

6.3 SIP Configurations

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

The user can conduct line configuration on the interface of the phone or the webpage, and input the corresponding information at the registered address, registered user name, registered password and SIP user, display name and registered port respectively, which are provided by the SIP server administrator.

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



Picture 2- Line

6.4 LED

Table 2- LED

Type	LED light	LED status
Power light	Steady red	Connect the power supply
	Light off	Not connected to power supply
Network indicator light	Steady green	Connect to normal
	Green light flash	Network anomaly
	Light off	Unconnected network cable
FXS port indicator	Steady green	Calling/alerting/talking
	Green light flash	MWI
	Light off	Standby
Registration indicator light	Steady green	Registration success
	Green light flash	Registration failed
	Light off	Unregistered

6.5 Voice Menu

Table 3- Voice Menu

Effect	Instruction	Describe
Start Reboot	#****	Reboot the device
Enable Static Mode	#*100	Modify the network mode to gorgeous mode
Enable DHCP Mode	#*101	Modify the network mode to dynamic mode
Enable PPPOE Mode	#*102	Modify the network mode to PPPOE mode
Read WAN IP	#*111	Read IP address
Read Phone Number	#*222	Read the registered phone number
Set WAN IP Addr	#*50	Set the IP address of the WAN port
Set WAN Gateway	#*51	Set up WAN port equipment
Set WAN DNS	#*52	Set the DNS of the WAN port
Set WAN Subnet Mask	#*53	Set the subnet mask of the WAN port
Disabled Call Forward	#*90	Use as a forward function
Busy Call Forward	#*91	Allow forward on busy, when the device is busy, forward to the set number
No Ans Call Forward	#*92	Busy forwarding is allowed, when there is no one on the device, it will be forwarded to the set number
Always Call Forward	#*93	Enable unconditional forwarding, the device always forwards to the set number
Voice Mail	#*86	Broadcast the number of unread messages
Call Back	#*87	Call the last number called
Redial	#*88	Call a missed number
DND ON	#*94	Enable Do Not Disturb, all numbers will not be able to call in after activation
DND OFF	#*95	Turn off Do Not Disturb
Blind Transfer	#*27	Blind transfer operation
Attended Transfer	#*28	Attendance transfer operation
Conference (Conf)	#*29	Perform meeting operations

7 Basic Function

7.1 Making Phone Call

Step 1. Off hook the analog phone
Step 2. Enter the number and wait 5 seconds
Step 3. Enter the number and press #

7.2 Call Hold

Step 1. Off hook the analog phone
Step 2. Enter the number and press #
Step 3. Press the flash button of the analog phone to hold the call. Press again to release the call, If the analog phone does not have the Flash button, you can use switch hook.

7.3 Call Transfer

7.3.1 Blind transfer

Step 1. Set up a call between user A and user B
Step 2. User A presses the Flash key or switch hook to hold the call between user A and User B
Step 3. Enter #*27 + C and press # to dial user C's number
Step 4. User A hangs up, user B sets up a call with user C

7.3.2 Attended transfer

Step 1. Set up a call between user A and user B
Step 2. User A presses the Flash key or switch hook to hold the call between user A and User B
Step 3. Enter #*28 + C and press # to dial user C's number
Step 4. User C picks up, user A sets up a call with user C
Step 5. User A presses the Flash key or switch hook, user B sets up a call with user C

7.4 Conference

The device supports conference. Suppose user A and user B are on a call, and user A invites user C to join the conference. The following describes how to implement the function of conference.

Start a conference

Step 1. User A makes a call with user B.
Step 2. User A presses the Flash key or lightly presses the hook of the phone to hold the

call between A and B.
Step 3. Enter #*29 + C, and press # to dial user C's number.
Step 4. User C picks up the phone.
Step 5. User A presses Flash key or lightly presses the hook of the phone. At this time, users A, B, and C establish a conference call.

Cancel the conference

If user C is invited to the conference after answering the call, user A can press Flash key or lightly press the hook to remove user C from the conference.

End the conference

During the conference, if one party hangs up the call (not including user A), the other two users can continue the conversation.
If user A hangs up the phone, the conference ends and returns to standby status.

8 Advance Function

8.1 IP Direct

Step 1. Off hook the Analog phone
Step 2. Input the dialed the IP address,such as: 192*168*10*111
Step 3. Press "#" button to call out

8.2 Redial

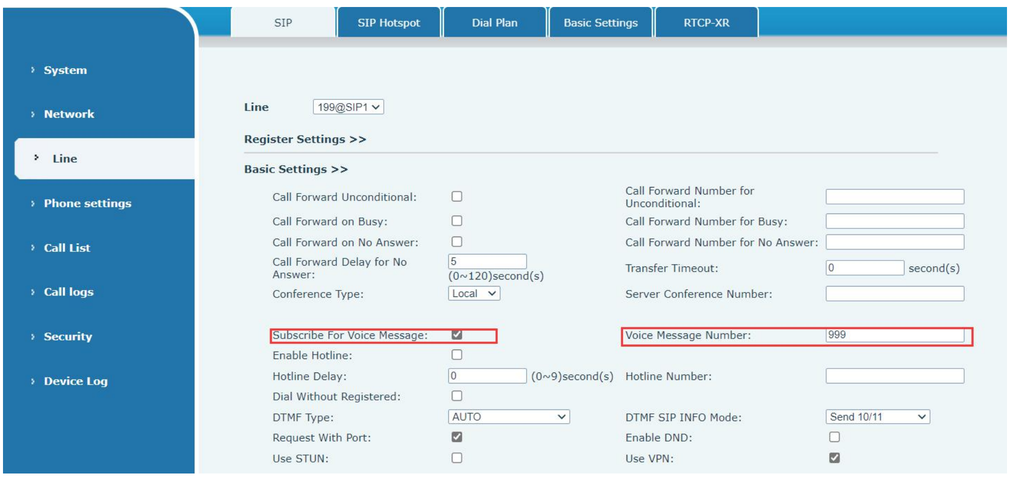
Redial the last dialed number
Step 1. Off hook the Analog phone
Step 2. Input"#*88"
Step 3. Press"#" button to call out

8.3 Call Back

Dial the last unanswered number
Step 1. Off hook the Analog phone
Step 2. Input"#*87"
Step 3. Press "#" button to call out

8.4 Voice Mail

Enable to Subscribe for Voice Message in the Web GUI page and set the Voice Message Number. When there is a Voice message, the indicator of the FXS interface flashes green quickly.



Picture 3- Voice mail

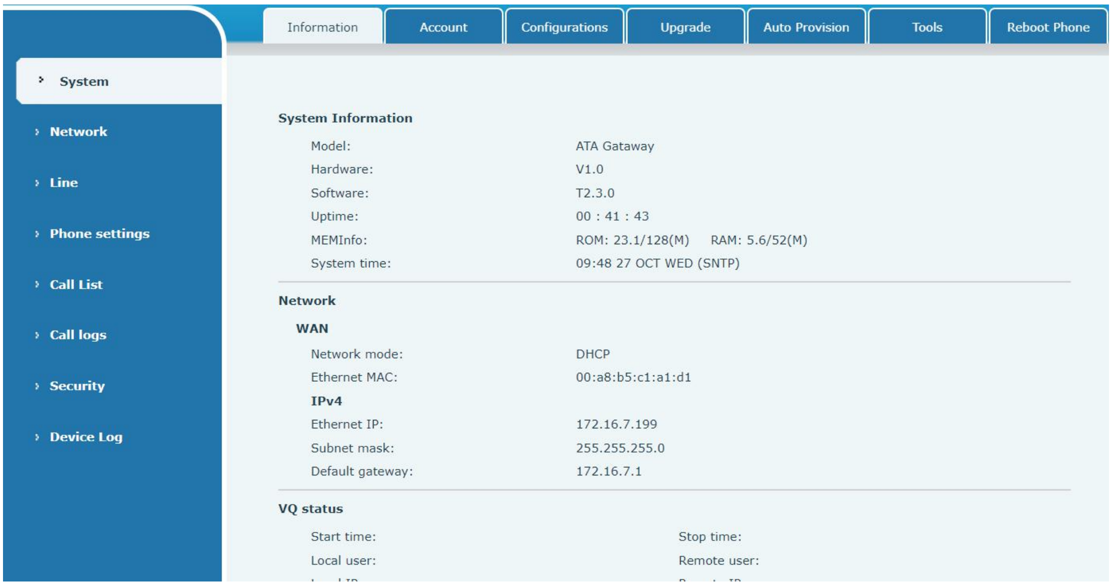
Off hook the analog phone and input the preset Voice Message Number to listen to the Voice Message according to the prompt
Off hook the Analog phone and Press "#*86" to check if there is a Voice Message

9 Web Configurations

9.1 Web Page Authentication

The user can log into the web page of the phone to manage the user's phone information and operate the phone. Users must provide the correct user name and password to log in.

9.2 System >> Information



Picture 4- Information

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

- Model
- Hardware
- Software
- Uptime
- MEMInfo
- System time

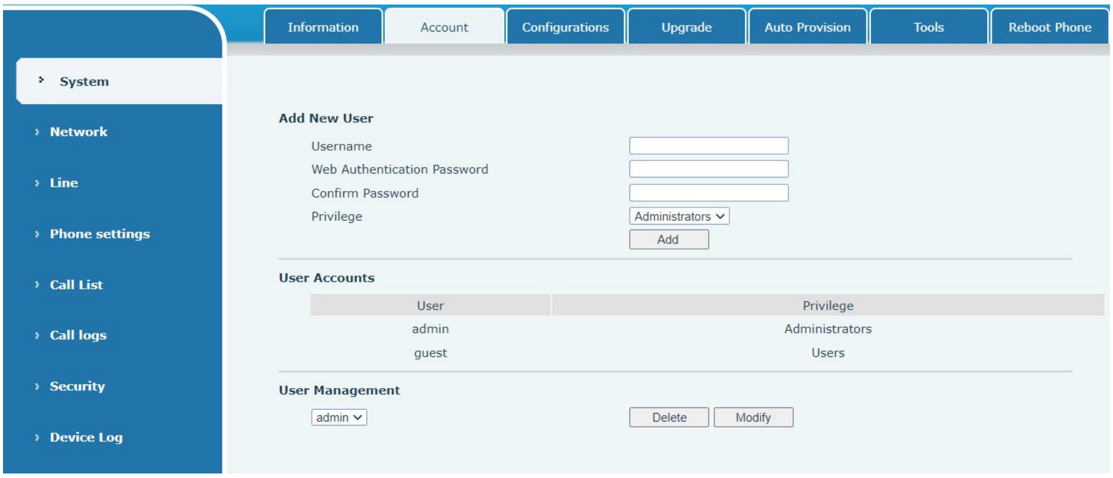
And summarization of network status,

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

Besides, summarization of SIP account status,

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

9.3 System >> Account

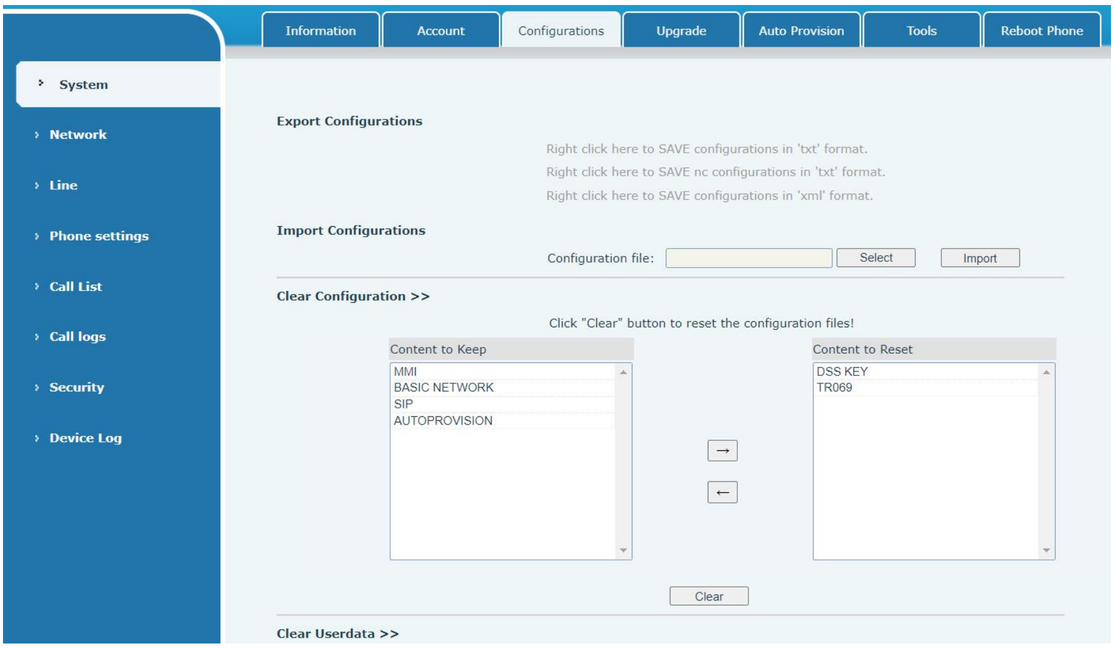


Picture 5- WEB Account

On this page the user can change the password for the login page.

Users with administrator rights can also add or delete users, manage users, and set permissions and passwords for new users.

9.4 System >> Configurations



Picture 6- System Setting

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

■ **Clear Configurations**

Select the module in the configuration file to clear.

SIP: account configuration.

AUTOPROVISION: automatically upgrades the configuration

TR069:TR069 related configuration

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

Clear Tables

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

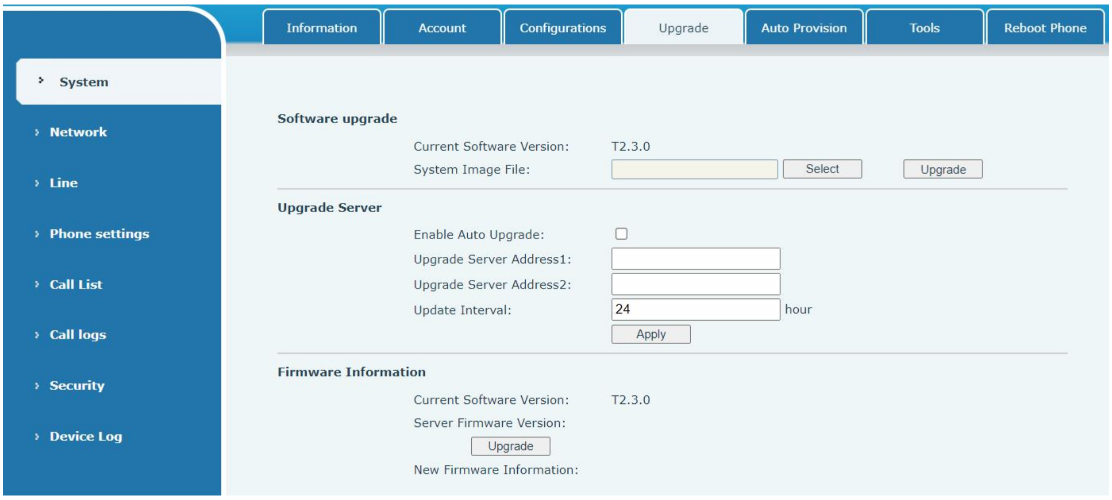
Clear ETC

Select the etc to be cleared, all selected by default

■ **Reset Phone**

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

9.5 System >> Upgrade



Picture 7- Upgrade

Table 4 - Firmware upgrade

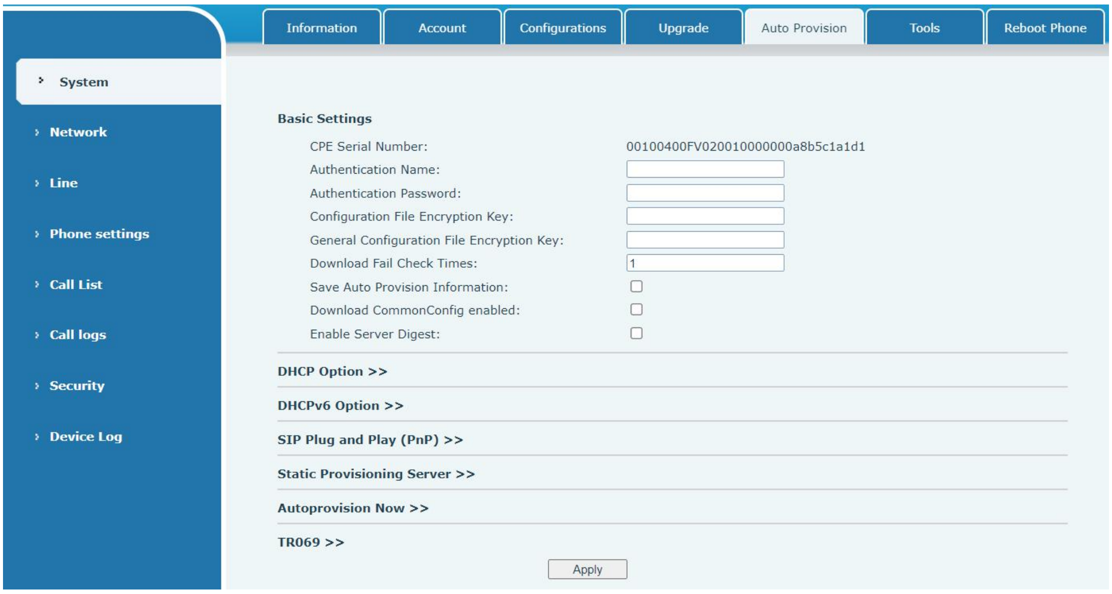
Parameter	**Description**
Upgrade server	
Enable Auto Upgrade	Enable automatic upgrade. If there is a new version txt and new software firmware on the server, phone will show a prompt upgrade message after Update Interval.
Upgrade Server Address1	Set available upgrade server address.
Upgrade Server Address2	Set available upgrade server address.
Update Interval	Set Update Interval.
Firmware Information	
Current Software Version	It will show Current Software Version.

Server Firmware Version	It will show Server Firmware Version.
[Upgrade] button	If there is a new version txt and new software firmware on the server, the page will display version information and upgrade button will become available; Click [Upgrade] button to upgrade the new firmware.
New version description information	When there is a corresponding TXT file and version on the server side, the TXT and version information will be displayed under the new version description information.

- The file requested from the server is a TXT file called vendor_model_hw10.txt.Hw followed by the hardware version number, it will be written as hw10 if no difference on hardware. All Spaces in the filename are replaced by underline.
- The URL requested by the phone is HTTP:// server address/vendor_Model_hw10 .txt： The new version and the requested file should be placed in the download directory of the HTTP server, as shown in the figure:
- TXT file format must be UTF-8
- vendor_model_hw10.TXT The file format is as follows：
  Version=1.6.3 #Firmware
  Firmware=xxx/xxx.z #URL，Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers.
  BuildTime=2018.09.11 20:00
  Info=TXT|XML

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

9.6 System >> Auto Provision



Picture 8- Auto provision settings

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

PNP>DHCP>TR069> Static Provisioning

Transferring protocol: FTP、 TFTP、 HTTP、 HTTPS

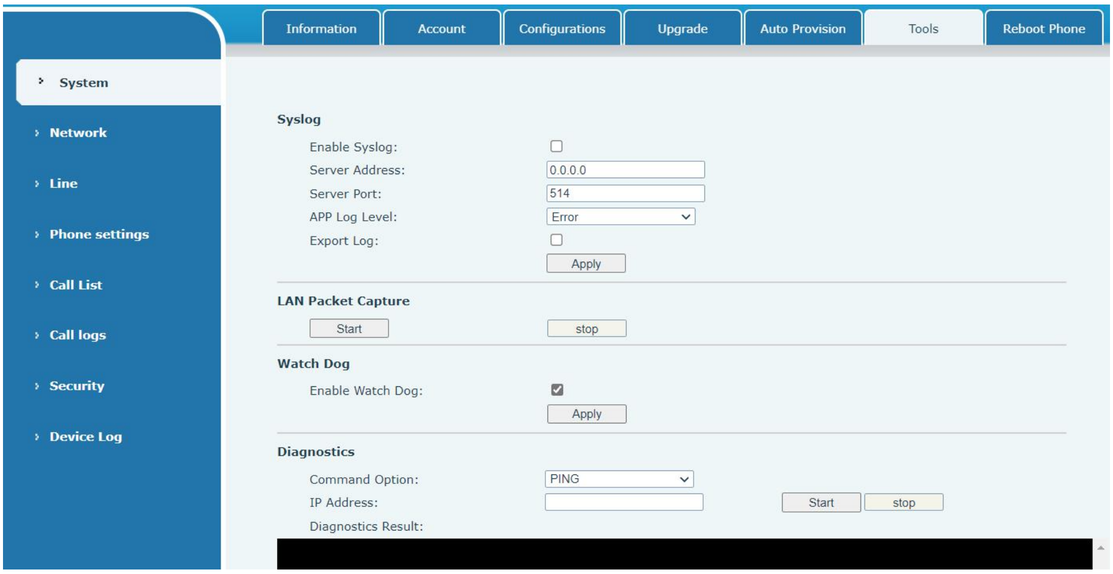
Table 5- Auto Provision

Parameters	Description
Basic settings	
CPE Serial Number	Display the device SN
Authentication Name	The user name of provision server
Authentication Password	The password of provision server
Configuration File Encryption Key	If the device configuration file is encrypted , user should add the encryption key here
General Configuration File Encryption Key	If the common configuration file is encrypted, user should add the encryption key here
Download Fail Check Times	If the download is failed, phone will retry with the configured times.
Save Auto Provision Information	Save the HTTP/HTTPS/FTP user name and password. If the provision URL is kept, the information will be kept.
Download Common Config enabled	Whether phone will download the common configuration file.
DHCP Option	

Option Value	Configure DHCP option, DHCP option supports DHCP custom option | DHCP option 66 | DHCP option 43, 3 methods to get the provision URL. The default is Option 66.
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be same as server define.
Enable DHCP Option 120	Use Option120 to get the SIP server address from DHCP server.
DHCPv6 Option	
Option Value	Configure DHCPv6 option, DHCPv6 option supports custom option | option 66 | option 43, 3 methods to get the provision URL. The default is Disable.
DHCP Option Vlan(128-254)	Custom Option value is allowed from 128 to 254. The option value must be same as server define.
SIP Plug and Play (PnP)	
Enable SIP PnP	Whether enable PnP or not. If PnP is enabled, phone will send a SIP SUBSCRIBE message with broadcast method. Any server can support the feature will respond and send a Notify with URL to phone. Phone could get the configuration file with the URL.
Server Address	Broadcast address. As default, it is 224.0.0.0.
Server Port	PnP port
Transport Protocol	PnP protocol, TCP or UDP.
Update Interval	PnP message interval.
Static Provisioning Server	
Server Address	Provisioning server address. Support both IP address and domain address.
Configuration File Name	The configuration file name. If it is empty, phone will request the common file and device file which is named as its MAC address. The file name could be a common name, $mac.cfg, $input.cfg. The file format supports CFG/TXT/XML.
Protocol Type	Transferring protocol type ，supports FTP、TFTP、HTTP and HTTPS
Update Interval	Configuration file update interval time. As default it is 1, means phone will check the update every 1 hour.
Update Mode	Provision Mode. 1. Disabled. 2. Update after reboot. 3. Update after interval.
Static Provisioning Server	
TR069	

Enable TR069	Enable TR069 after selection
ACS Server Type	There are 2 options Serve type, common and CTC.
ACS Server URL	ACS server address
ACS User	ACS server username (up to is 59 character)
ACS Password	ACS server password (up to is 59 character)
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.

9.7 System >> Tools

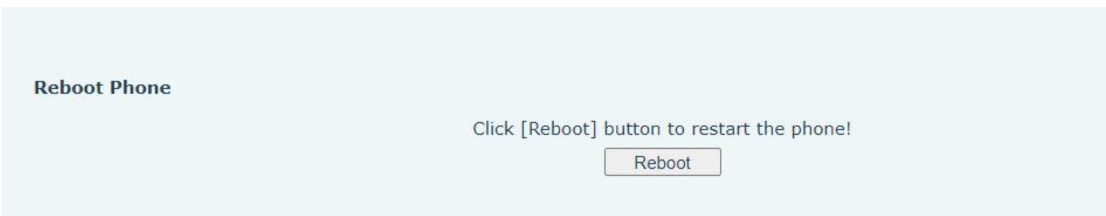


Picture 9- Tools

Tools provided in this page help users to identify issues at trouble shooting. Please refer to 10 Trouble Shooting for more detail.

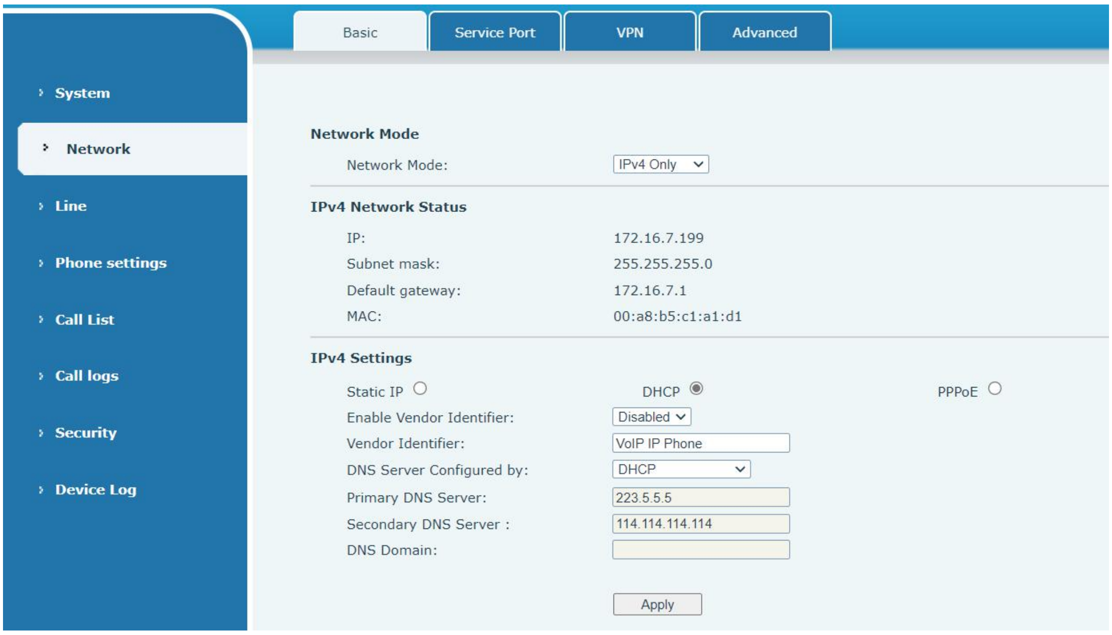
9.8 System >> Reboot Phone

This page can restart the phone.



Picture 10- Reboot Phone

9.9 Network >> Basic



Picture 11- Network Basic Setting

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

Table 6- Network Basic Setting

Parameter	Description
Network Status	
IP	The current IP address of the equipment
Subnet mask	The current Subnet Mask
Default gateway	The current Gateway IP address
MAC	The MAC address of the equipment
MAC Time stamp	Display the time when the device gets the MAC address
Settings	
Select the appropriate network mode. The equipment supports three network modes:	
Static IP	Network parameters must be entered manually and will not change. All parameters are provided by the ISP.
DHCP	Network parameters are provided automatically by a DHCP server.
PPPoE	Account and Password must be input manually. These are provided by your ISP.
If Static IP is chosen, the screen below will appear. Enter values provided by the ISP.	
DNS Server	Select the Configured mode of the DNS Server.

Configured by	
Primary DNS Server	Enter the server address of the Primary DNS.
Secondary DNS Server	Enter the server address of the Secondary DNS.

Attention :

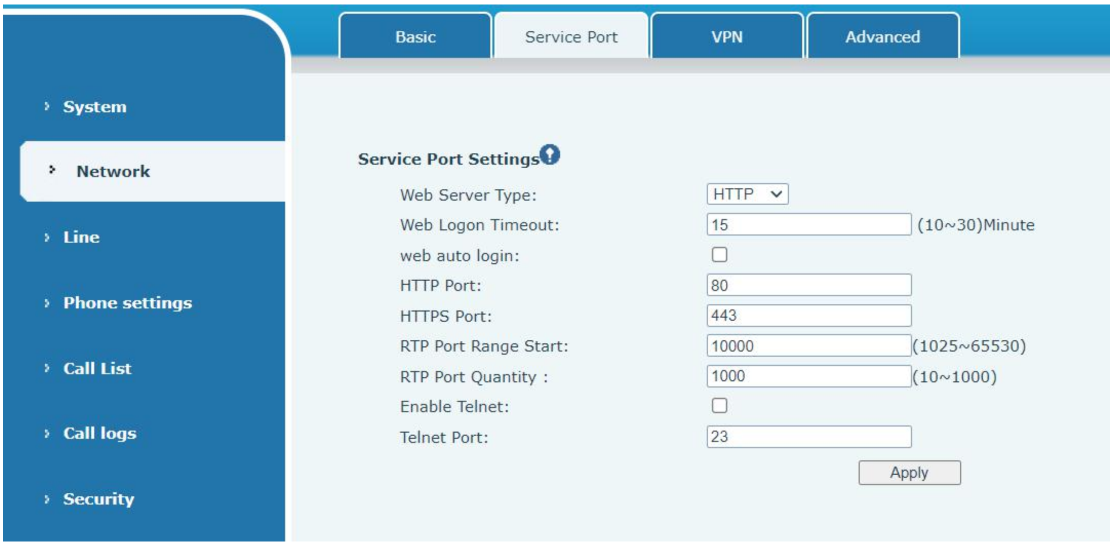
1) After setting the parameters, click【Apply】to take effect.

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

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

9.10Network >> Service Port

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

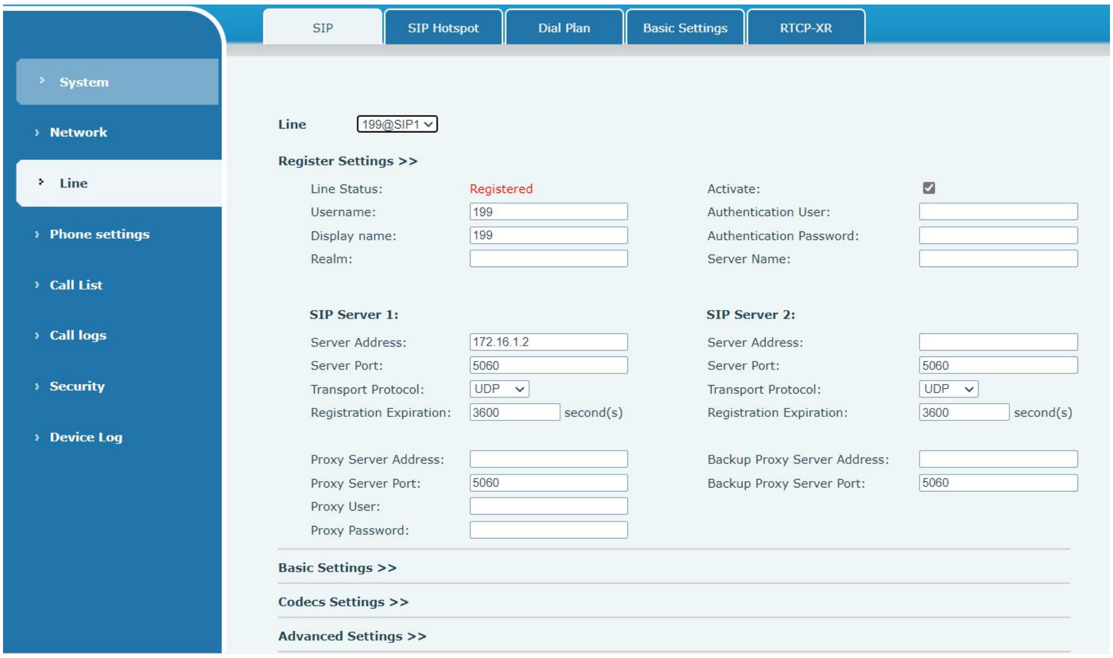


Picture 12- Service port setting interface

Table 7 - Service port

Parameter	Description
Web Server Type	Reboot to take effect after settings. Optionally, the web page login is HTTP/HTTPS.
Web Logon Timeout	Default as 15 minutes, the timeout will automatically exit the login page, need to login again.
Web auto login	After the timeout does not need to enter a user name password, will automatically login to the web page.
HTTP Port	The default is 80. If you want system security, you can set ports other than 80. Such as :8080, webpage login: HTTP://ip:8080
HTTPS Port	The default is 443, the same as the HTTP port.
RTP Port Range Start	The value range is 1025 to 65535. The value of RTP port starts from the initial value set. For each call, the value of voice and video port is added 2.
RTP Port Quantity	Number of calls.

9.11Line >> SIP



Picture 13- SIP

Configure the Line service configuration on this page.

Table 8- SIP

Parameters	Description
Register Settings	

Line Status	Display the current line status at page loading. To get the up-to-date line status, user must refresh the page manually.
Activate	Whether the service of the line is activated
Username	Enter the username of the service account.
Authentication User	Enter the authentication user of the service account
Display Name	Enter the display name to be sent in a call request.
Authentication Password	Enter the authentication password of the service account
Realm	Enter the SIP domain if requested 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 transport line using TCP or UDP or TLS.
Registration Expiration	Set SIP expiration date.
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 transport line using TCP or UDP or TLS.
Registration Expiration	Set SIP expiration date.
SIP Proxy Server Address	Enter the IP or FQDN address of the SIP proxy server.
Proxy Server Port	Enter the SIP proxy server port, default is 5060.
Proxy User	Enter the SIP proxy user.
Proxy Password	Enter the SIP proxy password.
Backup Proxy Server Address	Enter the IP or FQDN address of the backup proxy server.
Backup Proxy Server Port	Enter the backup proxy server port, default is 5060.
Basic Settings	
Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it

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

Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.
DTMF Type	Set the DTMF type to be used for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
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	A Register message is used to periodically detect the time interval for the availability of the main Proxy.
Signal Failback	Multiple proxy cases, whether to allow the invite/register request to also execute failback.
Signal Retry Counts	The number of attempts that the SIP Request considers proxy unavailable under multiple proxy scenarios.
Codecs Settings	Set the priority and availability of the codecs by adding or remove them from the list.
Video Codecs	Select video code to preview video.
Advanced Settings	
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field.
Enable DND	Set the feature code to dial to the server
Disable DND	Set the feature code to dial to the server
Enable Call Forward Unconditional	Set the feature code to dial to the server
Disable Call Forward Unconditional	Set the feature code to dial to the server
Enable Call Forward on Busy	Set the feature code to dial to the server
Disable Call Forward on Busy	Set the feature code to dial to the server
Enable Call Forward on	Set the feature code to dial to the server

No Answer	
Disable Call Forward on No Answer	Set the feature code to dial to the server
Enable Blocking Anonymous Call	Set the feature code to dial to the server
Disable Blocking Anonymous Call	Set the feature code to dial to the server
Call Waiting On Code	Set the feature code to dial to the server
Call Waiting Off Code	Set the feature code to dial to the server
Send Anonymous on Code	Set the feature code to dial to the server
Send Anonymous Off Code	Set the feature code to dial to the server
SIP Encryption	Enable SIP encryption such that SIP transmission will be encrypted
RTP Encryption	Enable RTP encryption such that 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 not new session timer event update received after the timeout period
Session Timeout	Set the session timer timeout period
Enable BLF List	Enable/Disable BLF List
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Response Single Codec	If setting enabled, the device will use single codec in response to an incoming call request
BLF Server	The registered server will receive the subscription package from ordinary application of BLF phone. Please enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened
Keep Alive Interval	Set the keep alive packet transmitting interval
Keep Authentication	Keep the authentication parameters from previous authentication
Blocking Anonymous Call	Reject any incoming call without presenting caller ID
User Agent	Set the user agent, the default is Model with Software Version.

Specific Server Type	Set the line to collaborate with specific server type
SIP Version	Set the SIP version
Anonymous Call Standard	Set the standard to be used for anonymous
Local Port	Set the local port
Ring Type	Set the ring tone type for the line
Enable user=phone	Sets user=phone in SIP messages.
Use Tel Call	Set use tel call
Auto TCP	Using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes
Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
DNS Mode	Select DNS mode, A, SRV, NAPTR
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server， it will use the source IP address, not the address in via field.
Convert URI	Convert not digit and alphabet characters to %hh hex code
Use Quote in Display Name	Whether to add quote in display name, i.e. “VoIP” vs VoIP
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
Sync Clock Time	Time Sync with server
Enable Inactive Hold	With the post-call hold capture package enabled, you can see that in the INVITE package, SDP is inactive.
Caller ID Header	Set the Caller ID Header
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Enable Feature Sync	Feature Sync with server
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
CallPark Number	Set the CallPark number.
Server Expire	Set the timeout to use the server.
TLS Version	Choose TLS Version.
uaCSTA Number	Set uaCSTA Number.
Enable Click to Talk	With the use of special server, click to call out directly after enabling.
Enable Chgport	Whether port updates are enabled.

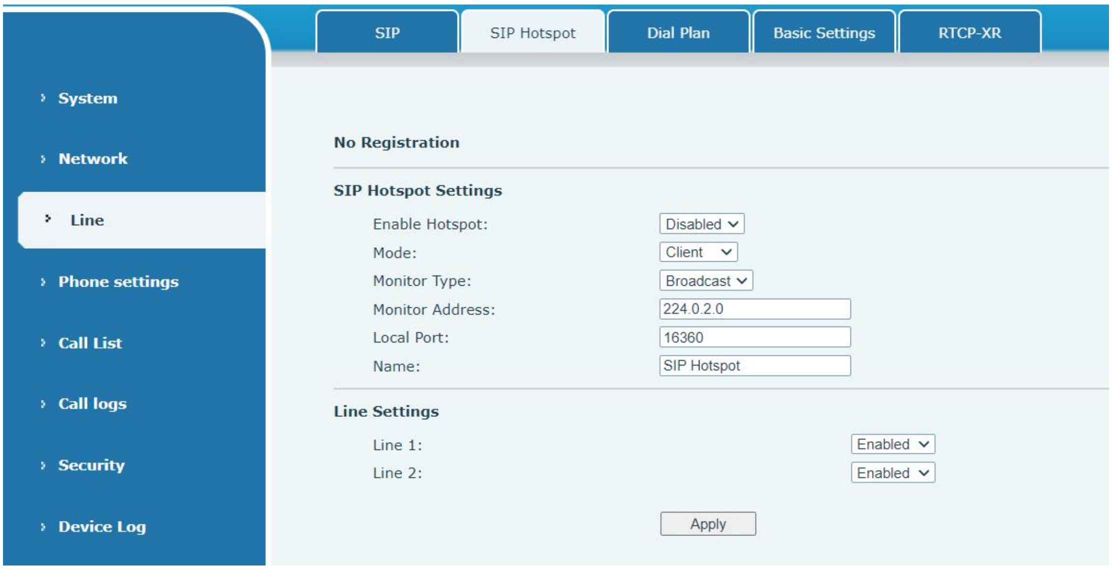
VQ Name	Open the VQ name for VQ RTCP-XR.
VQ Server	Open VQ server address for VQ RTCP-XR.
VQ Port	Open VQ port for VQ RTCP-XR.
VQ HTTP/HTTPS Server	Enable VQ server selection for VQ RTCP-XR.
Flash mode	Chose Flash mode， normal or SIP info.
Flash Info Content-Type	Set the SIP info content type.
Flash Info Content-Body	Set the SIP info content body.
PickUp Number	Set the scramble number when the Pickup is enabled.
JoinCall Number	Set JoinCall Number.
Intercom Number	Set Intercom Number.
Unregister On Boot	Whether to enable logout function.
Enable MAC Header	Whether to open the registration of SIP package with user agent with MAC or not.
Enable Register MAC Header	Whether to open the registration is user agent with MAC or not.
BLF Dialog Strict Match	Whether to enable accurate matching of BLF sessions.
PTime(ms)	Set whether to bring ptime field, default no.
SIP Global Settings	
Strict Branch	Set up to strictly match the Branch field.
Enable Group	Set open group.
Enable RFC4475	Set to enable RFC4475.
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Set the registration failure retry time.
Local SIP Port	Modify the phone SIP port.
Enable uaCSTA	Set to enable the uaCSTA function.

9.12Line >> SIP Hotspot

SIP hotspot is a simple but practical function. With simple configurations, the SIP hotspot function can implement group ringing. SIP accounts can be expanded.

Set a phone as a SIP hotspot and other phones (B and C) as SIP hotspot clients. When somebody calls phone A, phone A, B, and C all ring. When any phone answers the call, other phones stop ringing. The call can be answered by only one phone. When B or C initiates a call, the SIP number registered by phone A is the calling number.

To set a SIP hotspot, register at least one SIP account.

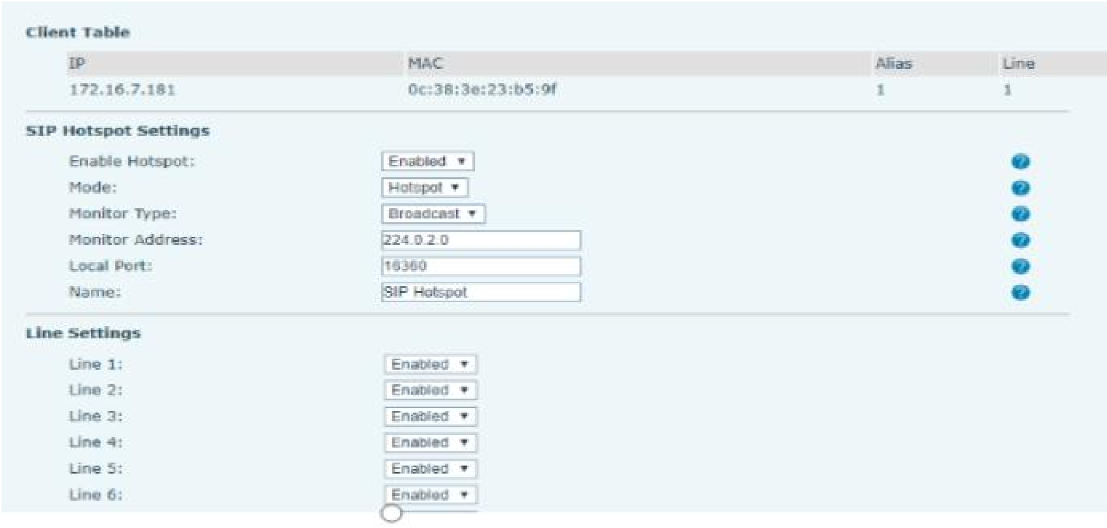


Picture 14 - SIP Hotspot

Table 9- SIP Hotspot

Parameters	**Description**
Device Table	If your phone is set to "SIP hotspot server", Device Table will display as Client Device Table which connected to your phone. If your phone is set to "SIP hotspot client", Device Table will display as Server Device Table which you can connect to.
SIP hotspot	
Enable hotspot	Set it to be Enable to enable the feature.
Mode	Choose hotspot, phone will be a "SIP hotspot server"; Choose Client, phone will be a "SIP hotspot Client"
Monitor Type	Either the Multicast or Broadcast is ok. If you want to limit the broadcast packets, you'd better use broadcast. But, if client choose broadcast, the SIP hotspot phone must be broadcast.
Monitor Address	The address of broadcast, hotspot server and hotspot client must be same.
Remote Port	Type the Remote port number.

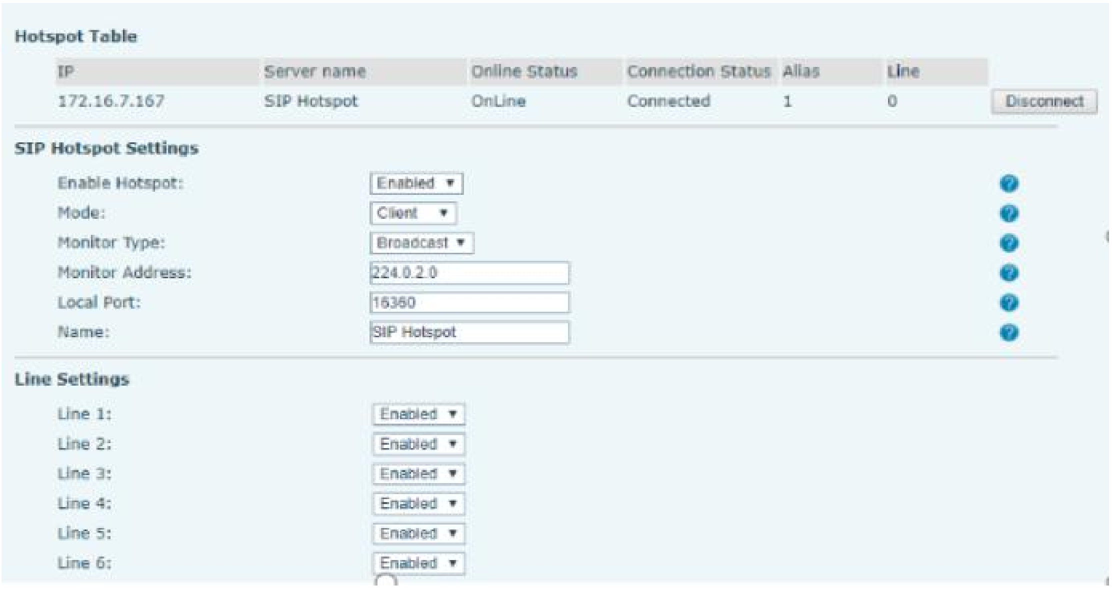
Configure SIP hotspot server:



Picture 15- SIP hotspot server

Configure SIP hotspot client:

As a SIP hotspot client, no SIP account needs to be set. The Phone set will automatically obtain and be configured a SIP account. On the SIP Hotspot tab page, set Mode to Client. The values of other options are the same as those of the hotspot.



Picture 16- SIP hotspot client

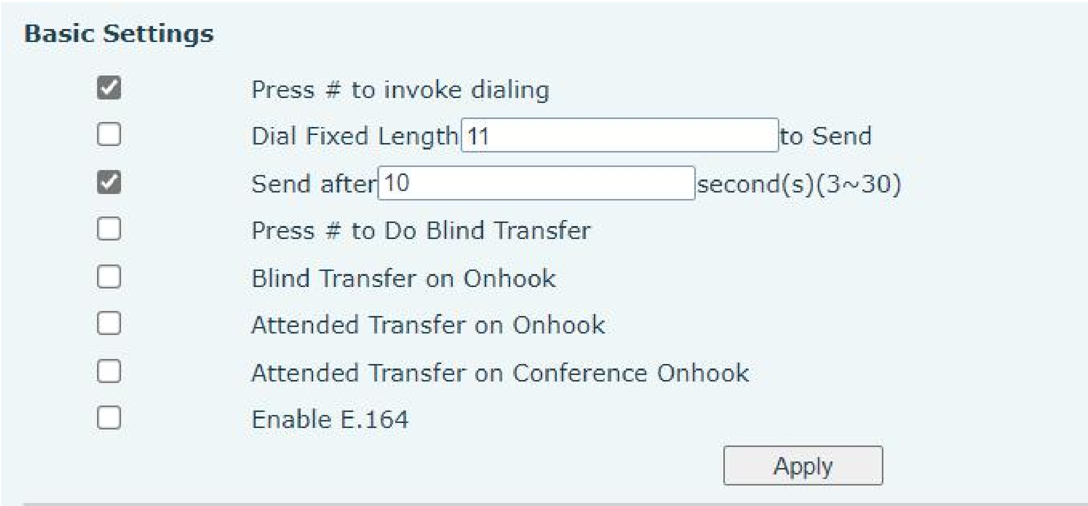
As the hotspot server, the default extension number is 0. When the phone is used as the client, the extension number is increased from 1, you can view the extension number through the [**SIP Hotspot**] page.

Call extension number:

The hotspot server and the client can dial each other through the extension number.

For example, extension 1 dials extension 0.

9.13Line >> Dial Plan



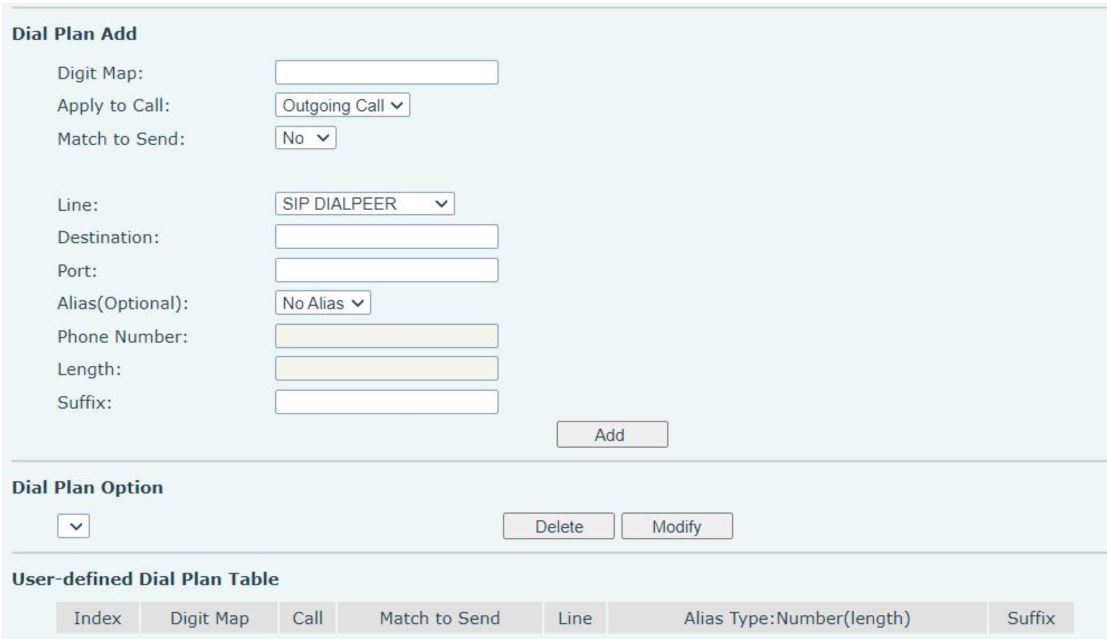
Picture 17- Dial Plan

Table 10- Phone 7 dialing methods

Parameters	**Description**
Press # to invoke dialing	The user dials the other party's number and then adds the # number to dial out;
Dial Fixed Length	The number entered by the user is automatically dialed out when it reaches a fixed length
Timeout dial	The system dials automatically after timeout
Press # to Do Blind Transfer	The user enters the number to be transferred and then presses the "#" key to transfer the current call to a third party
Blind Transfer on Onhook	After the user enters the number, hang up the handle or turn off the hands-free function to transfer the current call to a third party.
Attended Transfer on Onhook	Hang up the handle or press the hands-free button to realize the function of attention -transfer, which can transfer the current call to a third party.

Attended Transfer on Conference Onhook	During a three-way call, hang up the handle and the remaining two parties remain on the call.
Enable E.164	Please refer to e. 164 standard specification

Add dialing rules:



Picture 18- Custom setting of dial - up rules

Table 11 - Dial - up rule configuration table

Parameters	Description
Dial rule	There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules. In prefix matching, only part of the number is entered followed by T. The mapping with then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.
Note: Two different special characters are used. ■ x -- Matches any single digit that is dialed. ■ [ ] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.	
Destination	Set Destination address. This is for IP direct.

<table>
<tr><td>Port</td><td>Set the Signal port, and the default is 5060 for SIP.</td></tr>
<tr><td>Alias</td><td>Set the Alias. This is the text to be added, replaced or deleted. It is an optional item.</td></tr>
<tr><td colspan="2">Note: There are four types of aliases.
■ all: xxx – xxx will replace the phone number.
■ add: xxx – xxx will be dialed before any phone number.
■ del –The characters will be deleted from the phone number.
■ rep: xxx – xxx will be substituted for the specified characters.</td></tr>
<tr><td>Suffix</td><td>Characters to be added at the end of the phone number. It is an optional item.</td></tr>
<tr><td>Length</td><td>Set the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. It is an optional item.</td></tr>
</table>

This feature allows the user to create rules to make dialing easier. There are several different options for dialing rules. The examples below will show how this can be used.

Example 1: All Substitution -- Assume that it is desired to place a direct IP call to IP address 172.168.2.208. Using this feature, 123 can be substituted for 172.168.2.208.

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media
1	"123"	Out	No	SIP DIALPEER(172.16.1.15:5560)			Default

Picture 19 - Dial rules table (1)

Example 2: Partial Substitution -- To dial a long-distance call to Beijing requires dialing area code 010 before the local phone number. Using this feature 1 can be substituted for 010. For example, to call 62213123 would only require dialing 162213123 instead of 01062213123.

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media
1	"1T"	Out	No	Fanvil@SIP1	rep:010(1)		Default

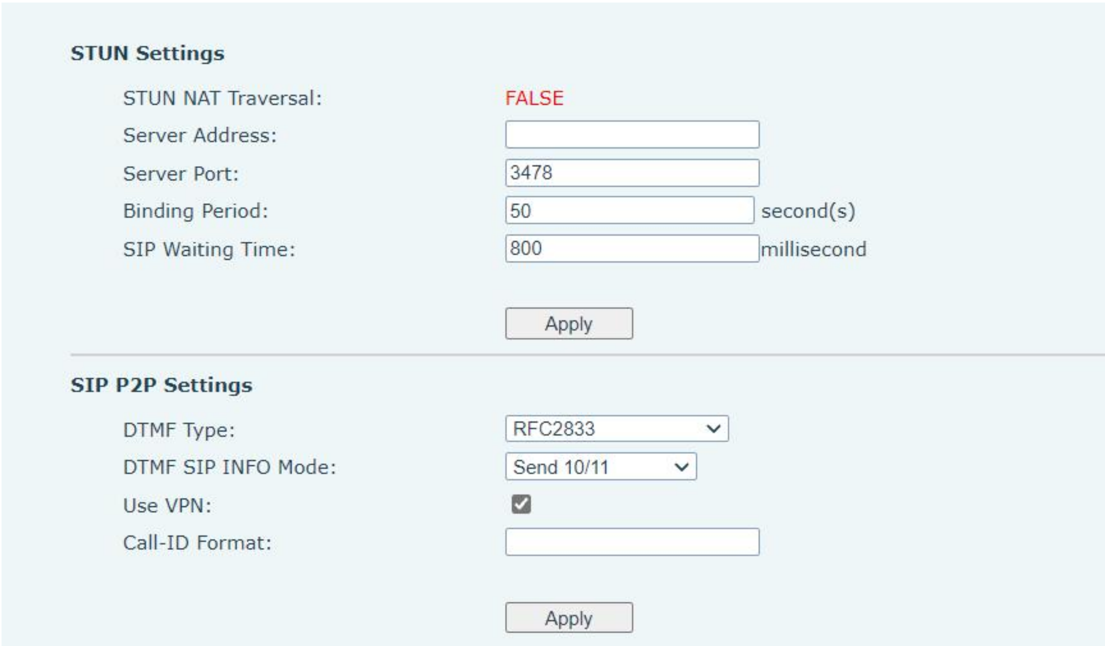
Picture 20 - Dial rules table (2)

Example 3: Addition -- Two examples are shown. In the first case, it is assumed that 0 must be dialed before any 11-digit number beginning with 13. In the second case, it is assumed that 0 must be dialed before any 11-digit number beginning with 135, 136, 137, 138, or 139. Two different special characters are used.

x -- Matches any single digit that is dialed.

[] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.

9.14Line >> Basic Settings



Picture 21- Basic Settings

Set up the register global configuration.

Table 12- Set the line global configuration on the web page

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

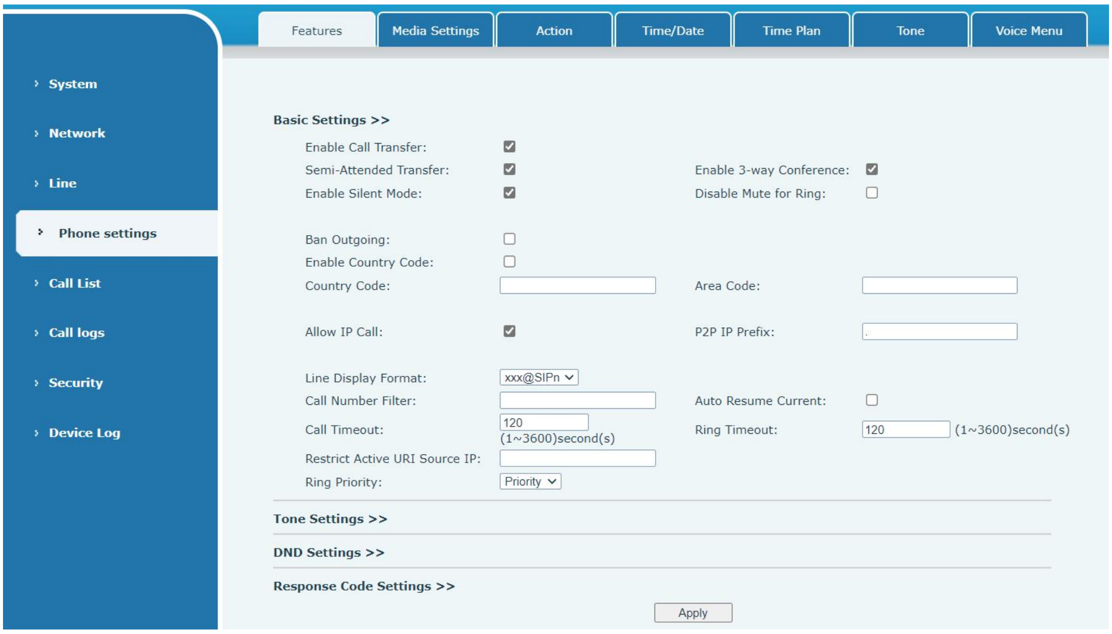
9.15Line >> RTCP-XR

The RTCP-XR mode is based on THE RTP Control Extended Report (RFC3611). It sends RTCP-XR packets to evaluate network packet loss, delay, and voice quality.

Table 13- set RTCP-XR

Parameters	Description
VQ RTCP-XR Settings	
VQ RTCP-XR Session Report	Whether to enable sending VQ reports in session mode
VQ RTCP-XR Interval Report	Whether to enable sending VQ reports in Interval mode
Period for Interval Report (5~99)	The interval at which VQ reports are periodically sent
Warning threshold for Moslq(15~40)	When the Moslq value x10 is lower than the threshold, a warning message is generated
Critical threshold for Moslq(15~40)	When the CALCULATED Moslq value x10 is lower than the threshold, a critical report is generated
Warning Threshold for Delay (10~2000)	When the One-way delay is greater than the threshold, the IP phone generates a warning report
Critical Threshold for Delay (10~2000)	When the One-way delay is greater than the threshold, the IP phone generates a critical report
Display Report Options on web	Whether to display the VQ report data for the last call through a web page

9.16Phone settings >> Features



Picture 22- Features

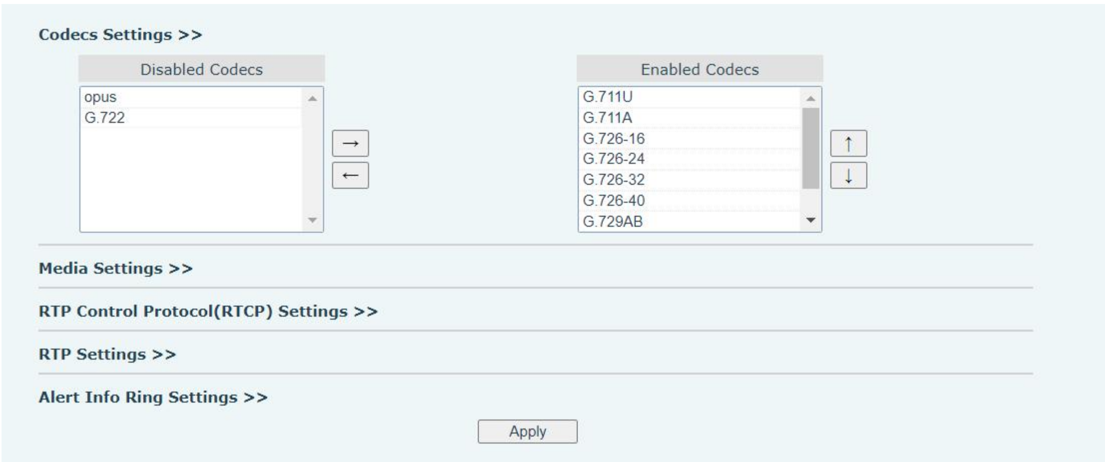
Configuration phone features

Table 14- Common device function Settings on the web page

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
Enable Call Transfer	Enable Call Transfer.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it
Enable 3-Way Conference	Enable 3-way conference by selecting it
Enable Silent Mode	When enabled, the phone is muted, there is no ringing when calls, you can use the volume keys and mute key to unmute.
Disable Mute for Ring	When it is enabled, you can't mute the phone
Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out any number.
Enable Country Code	Whether the country code is enabled.
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	Prefix a point-to-point IP call.
Caller Name Priority	Change caller ID display priority.
Emergency Call Number	
Search path	Select the search path.
LDAP Search	Select from with one LDAP for search
Emergency Call Number	Configure the Emergency Call Number. Despite the keyboard is locked, you can dial the emergency call number
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP address. More details please refer to this link
Tone Settings	
Enable Holding Tone	When turned on, a tone plays when the call is held
Enable Call Waiting Tone	When turned on, a tone plays when call waiting
Play Dialing DTMF Tone	Play DTMF tone on the device when user pressed a phone digit at dialing, default enabled.
DND Settings	
DND Option	Select to take effect on the line or on the phone or close.
Enable DND Timer	Enable DND Timer, If enabled, the DND is automatically

	turned on from the start time to the off time.
DND Start Time	Set DND Start Time
DND End Time	Set DND End Time
Intercom Settings	
Enable Intercom	When intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after specific delay.
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers the intercom call during a call. If the current call is intercom call, the phone will reject the second intercom call
Response Code Settings	
DND Response Code	Set the SIP response code on call rejection on DND
Busy Response Code	Set the SIP response code on line busy
Reject Response Code	Set the SIP response code on call rejection

9.17Phone settings >> Media Settings



Picture 23- Media Settings

Change voice Settings.

Table 15- Audio Settings

Parameter	**Description**

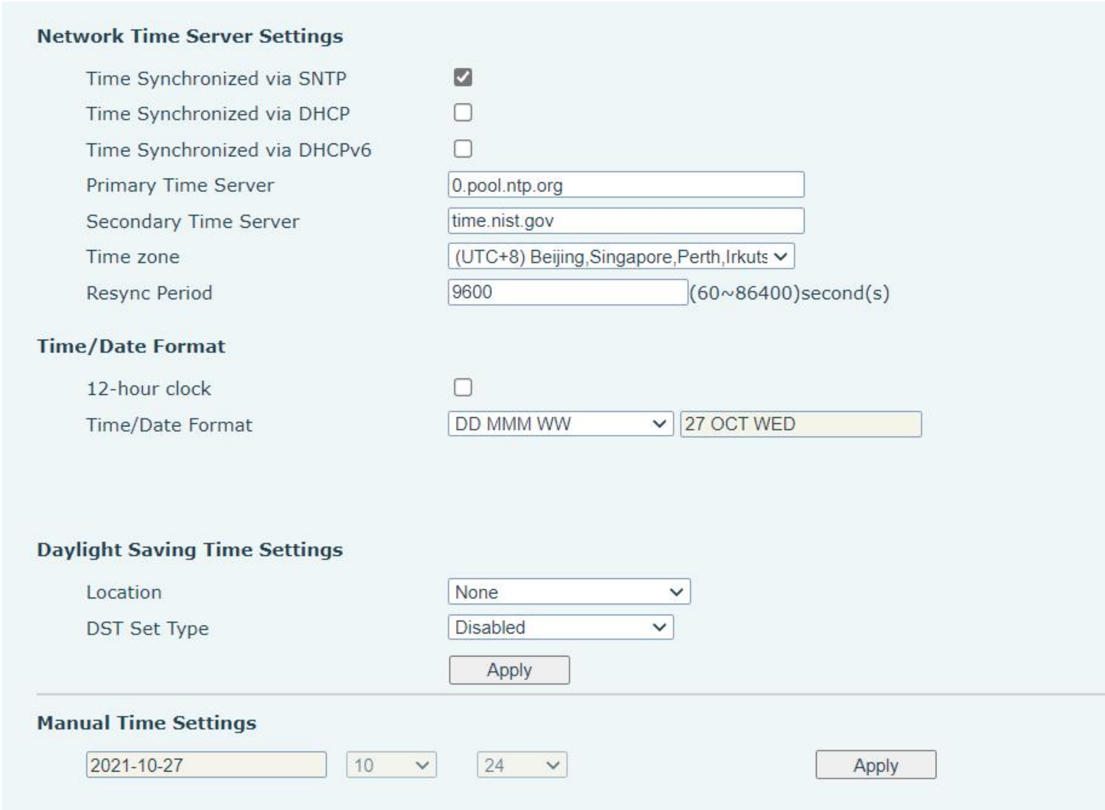
Codecs Settings	Select enable or disable voice encodec: G.711A/U,G.722,G.729, G.726-16,G726-24,G726-32,G.726-40, ILBC,opus
Media Setting	
Receive Volume	Set the call volume to 1~9
G.723.1 Bit Rate	The value can be 5.3 KB /s or 6.3 KB /s
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
Send Volume	Set the call volume to 1~9
Opus playload type	Set Opus load type, range 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
Onhook Time	Configure a minimum response time, which defaults to 800ms
Enable Hookflash	Whether to turn on the plug spring to generate Flash
Onhook Min Time	Configure a minimum response time, which defaults to 200ms
Caller ID Mode	Support bellcore、ETSI..
SLIC Impedance	Impedance settings
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 after 30s

9.18Phone settings >> Action

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

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

9.19Phone settings >> Time/Date



Picture 24- Time/Date

The user can configure the time Settings of the phone on this page.

Table 16- Time/Date

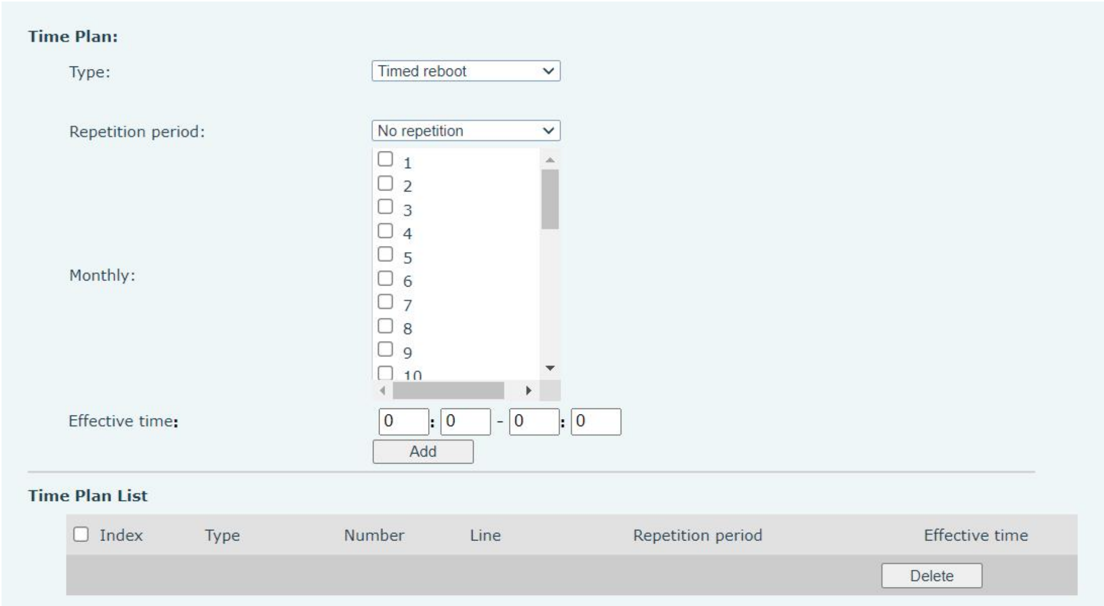
Parameters	**Description**
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Time Synchronized via DHCPv6	Enable time-sync through DHCPv6 protocol
Primary Time Server	Set primary time server 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 of re-synchronization with time server
12-Hour Clock	Set the time display in 12-hour mode
Date Format	Select the time/date display format

Daylight Saving Time Settings	
Local	Choose your local, phone will set daylight saving time automatically based on the local
DST Set Type	Choose DST Set Type, if Manual, you need to set the start time and end time.
Fixed Type	Daylight saving time rules are based on specific dates or relative rule dates for conversion. Display in read-only mode in automatic mode.
Offset	The offset minutes when DST started
Month Start	The DST start month
Week Start	The DST start week
Weekday Start	The DST start weekday
Hour Start	The DST start hour
Minute Start	The DST start minute
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Minute End	The DST end minute
Manual Time Settings	You can set your time manually

9.20Intercom Settings >> Time plan

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

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

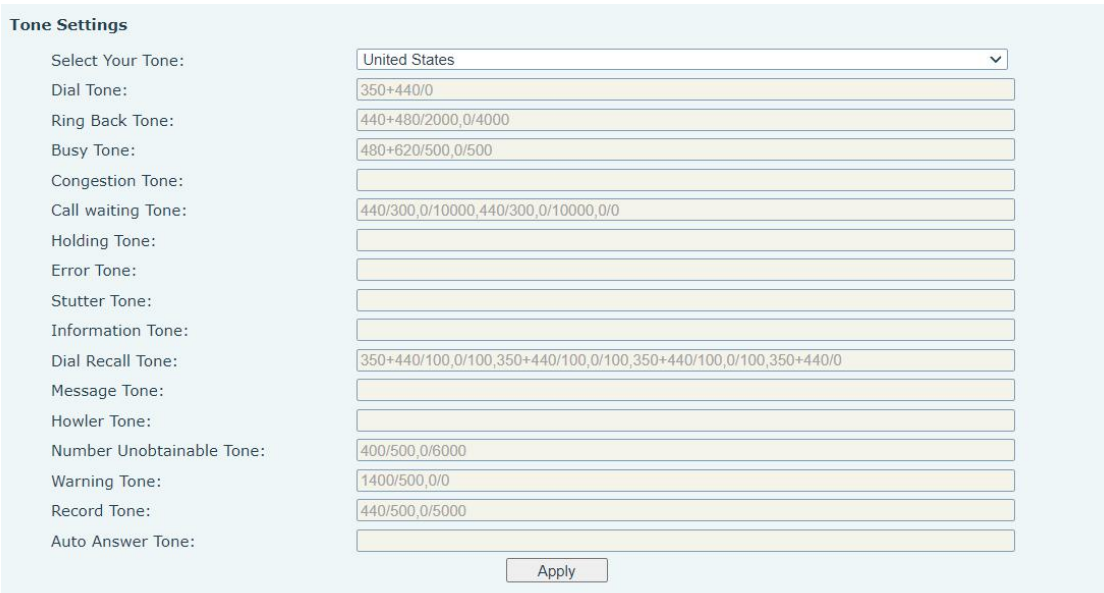


Picture 25- Time Plan

9.21Phone settings >> Tone

This page allows users to configure a phone prompt.

You can either select the country area or customize the area. If the area is selected, it will bring out the following information directly. If you choose to customize the area, you can modify the button tone, call back tone and other information.



Picture 26- Tone

9.22Phone settings >> Voice Menu

Supports customized voice menus

Picture 27- Voice Menu

9.23Call List >> Call List

■ Restricted Incoming Calls：

It is similar like a blacklist. Add the number to the blacklist, and the user will no longer receive calls from the stored number until the user removes it from the list.

Users can add specific Numbers to the blacklist or add specific prefixes to the blacklist to block calls with all Numbers with this prefix.

■ Allowed Incoming Calls：

When DND is enabled, the incoming call number can still be called.

■ Restricted Outgoing Calls：

Adds a number that restricts outgoing calls and cannot be called until the number is removed from the table.

9.24Call List >> Web Dial

Use web pages for call, reply, and hang up operations.

9.25Call Logs

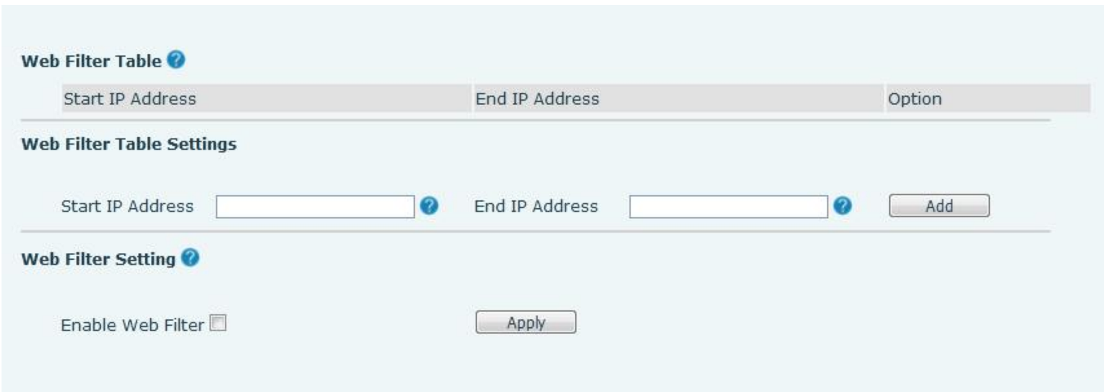
The user can browse the complete call record in this page. The call record can be sorted by time, call number, contact name or line, and the call record can be screened by call record type (incoming call, outgoing call, missed call, forward call).

The user can also save the number in the call record to his/her phone book or add it to the blacklist/whitelist.

Users can also dial the web page by clicking on the number in the call log.
Users can also download call records conditionally and save them locally.

9.26Security >> Web Filter

The user can set up a configuration management phone that allows only machines with a certain network segment IP access.



Picture 28 - Web Filter settings

Picture 29 - Web Filter Table

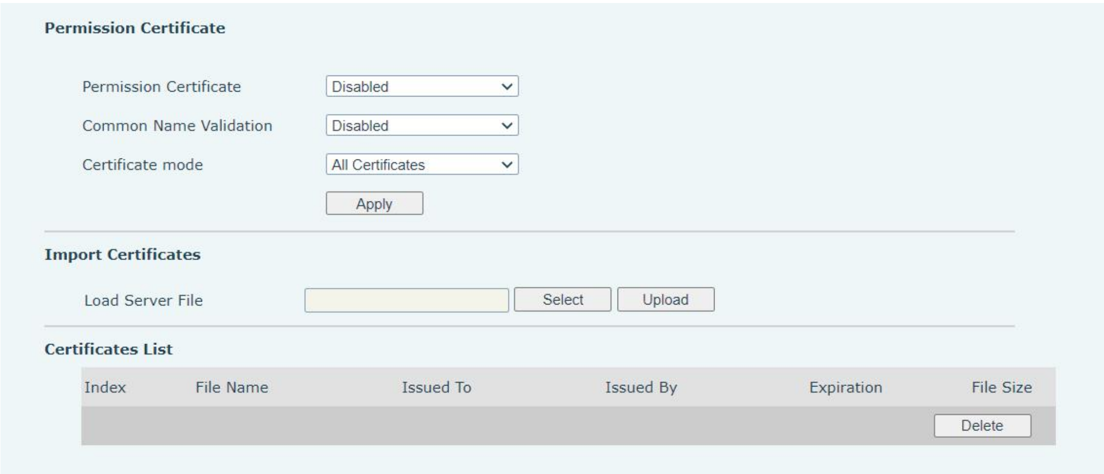
Add and remove IP segments that are accessible; Configure the starting IP address within the start IP, end the IP address within the end IP, and click [**Add**] to submit to take effect. A large network segment can be set, or it can be divided into several network segments to add. When deleting, select the initial IP of the network segment to be deleted from the drop-down menu, and then click [**Delete**] to take effect.

Enable web page filtering: configure enable/disable web page access filtering; Click the "apply" button to take effect.

Note: if the device you are accessing is in the same network segment as the phone, please do not configure the filter segment of the web page to be outside your own network segment, otherwise you will not be able to log in the web page.

9.27Security >> Trust Certificates

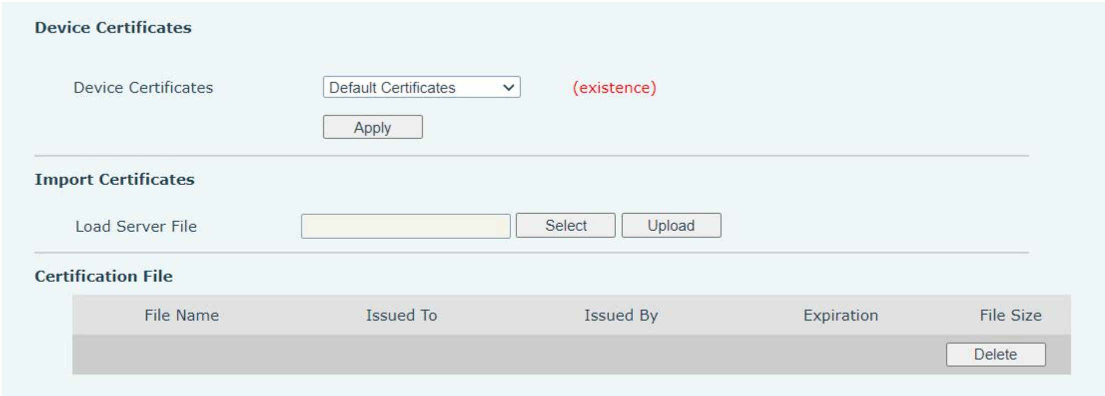
Set whether to open license certificate and general name validation, select certificate module.
You can upload and delete uploaded certificates.



Picture 30- Trust Certificates

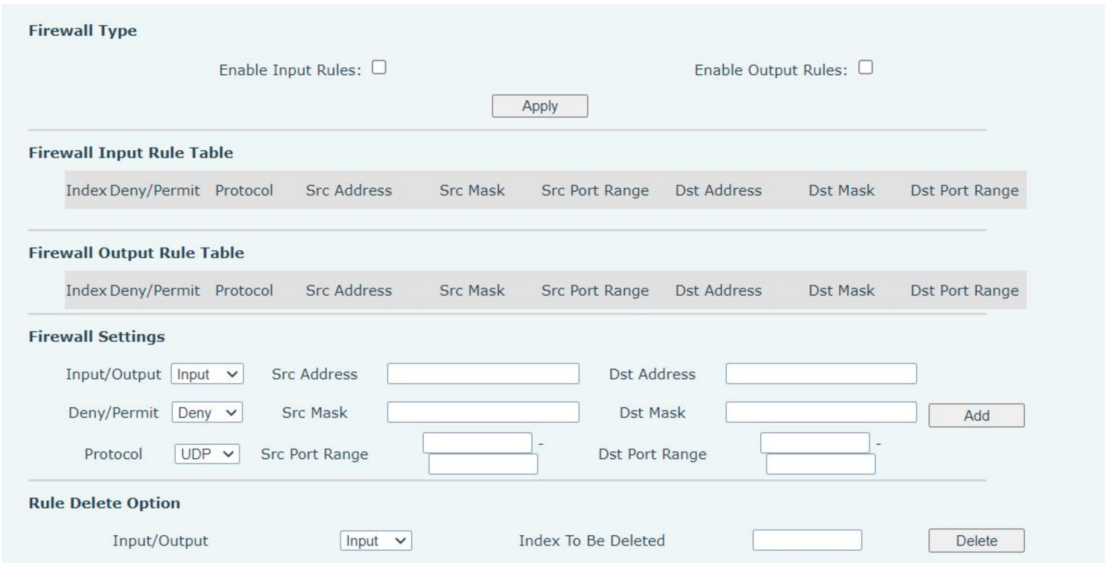
9.28Security >> Device Certificates

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



Picture 31- Device Certificates

9.29Security >> Firewall



Picture 32- Firewall

Through this page can set whether to enable the input, output firewall, at the same time can set the firewall input and output rules, using these Settings can prevent some malicious network access, or restrict internal users access to some resources of the external network, improve security.

Firewall rule set is a simple firewall module. This feature supports two types of rules: input rules and output rules. Each rule is assigned an ordinal number, allowing up to 10 for each rule.

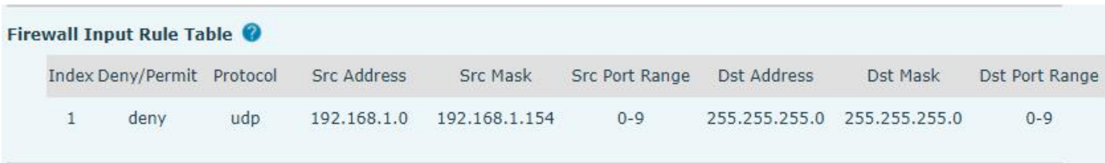
Considering the complexity of firewall Settings, the following is an example to illustrate:

Table 17 - Network Firewall

Parameter	**Description**
Enable Input Rules	Indicates that the input rule application is enabled.
Enable Output Rules	Indicates that the output rule application is enabled.
Input/Output	To select whether the currently added rule is an input or

	output rule.
Deny/Permit	To select whether the current rule configuration is disabled or allowed;
Protocol	There are four types of filtering protocols: TCP	UDP	ICMP	IP.
Src Port Range	Filter port range
Src Address	Source address can be host address, network address, or all addresses 0.0.0.0; It can also be a network address like *.*.*.0, such as: 192.168.1.0.
Dst Address	The destination address can be either the specific IP address or the full address 0.0.0.0; It can also be a network address like *.*.*.0, such as: 192.168.1.0.
Src Mask	Is the source address mask. When configured as 255.255.255.255, it means that the host is specific. When set as 255.255.255.0, it means that a network segment is filtered.
Dst Mask	Is the destination address mask. When configured as 255.255.255.255, it means the specific host. When set as 255.255.255.0, it means that a network segment is filtered.

After setting, click [**Add**] and a new item will be added in the firewall input rule, as shown in the figure below:



Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range
1	deny	udp	192.168.1.0	192.168.1.154	0-9	255.255.255.0	255.255.255.0	0-9

Picture 33- Firewall Input rule table

Then select and click the button [**Apply**].

In this way, when the device is running: ping 192.168.1.118, the packet cannot be sent to 192.168.1.118 because the output rule is forbidden. However, other IP of the ping 192.168.1.0 network segment can still receive the response packet from the destination host normally.

Picture 34 - Delete firewall rules

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

9.30Device Log >> Device Log

You can grab the device log, and when you encounter an abnormal problem, you can send the log to the technician to locate the problem. See 10.4 Get log information.

10 Trouble Shooting

When the phone is not in normal use, the user can try the following methods to restore normal operation of the phone or collect relevant information and send a problem report to technical support mailbox.

10.1Get Device System Information

Users can get information by web the [**System**] >> [**Information**] option in the phone. The following information will be provided:
The network information
Equipment information (model, software and hardware version), etc.

10.2Reset Device to Factory Default

Reset Device to Factory Default will erase all user's configuration, preference, database and profiles on the device and restore the device back to the state as factory default.

When the captain presses the reset button, all the indicators blink for 6 seconds. After release, the device will return to the factory default state.

10.3Network Packets Capture

Sometimes it is helpful to dump the network packets of the device for issue identification. To get the packets dump of the device, user needs to log in the device web portal, open page [**System**] >> [**Tools**] and click [**Start**] in "LAN Packets Capture" section. A pop-up message will be prompt to ask user to save the capture file. User then should perform relevant operations such as activate/deactivate line or making phone calls and click [**Stop**] button in the web page when operation finished. The network packets of the device during the period have been dumped to the saved file.
User may examine the packets with a packet analyzer or send it to support mailbox.

10.4Get Log Information

Log information is helpful when encountering an exception problem. In order to get the log information of the phone, the user can log in the phone web page, open the page [**Device log**], click the [**Start**] button, follow the steps of the problem until the problem appears, and then click the [**End**] button, [**Save**] to local analysis or send the log to the technician to locate the problem.

10.5Common Trouble Cases

Table 18- Trouble Cases

Trouble Case	**Solution**
Device could not boot up	1. The device is connected through a power adapter, please use the correct power supply
Device could not register to a service provider	1. Please check whether the device is connected to the network 2. If the network connection is good, please check your line configuration again. If all configurations are correct, contact your service provider for support, or follow the instructions in "10.3 Network Data Capture" to get a registered network packet and send it to the support mailbox to help analyze the problem