SPA2 SIP Gateway User Manual

Document VER	Firmware VER	Explanation	Time
V1.0	2.1.1.2545	Initial issue	20170518

Safety Notices

1. Please use the specified power adapter. If you need to use the power adapter provided by other manufacturers under special circumstances, please make sure that the voltage and current provided is in accordance with the requirements of this product, meanwhile, please use the safety certificated products, otherwise may cause fire or get an electric shock.
2. When using this product, please do not damage the power cord either by forcefully twist it, stretch pull, banding or put it under heavy pressure or between items, otherwise it may cause damage to the power cord, lead to fire or get an electric shock.
3. Before using, please confirm that the temperature and environment is humidity suitable for the product to work. (Move the product from air conditioning room to natural temperature, which may cause this product surface or internal components produce condense water vapor, please open power use it after waiting for this product is natural drying).
4. Please do not let non-technical staff to remove or repair. Improper repair may cause electric shock, fire, malfunction, etc. It would lead to injury accident or cause damage to your product.
5. Do not use fingers, pins, wire, other metal objects or foreign body into the vents and gaps. It may cause current through the metal or foreign body, which may even cause electric shock or injury accident. If any foreign body or objection falls into the product please stop using.
6. Please do not discard the packing bags or store in places where children could reach, if children trap his head with it, may cause nose and mouth blocked, and even lead to suffocation.
7. Please use this product with normal usage and operating, in bad posture for a long time to use this product may affect your health.
8. Please read the above safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

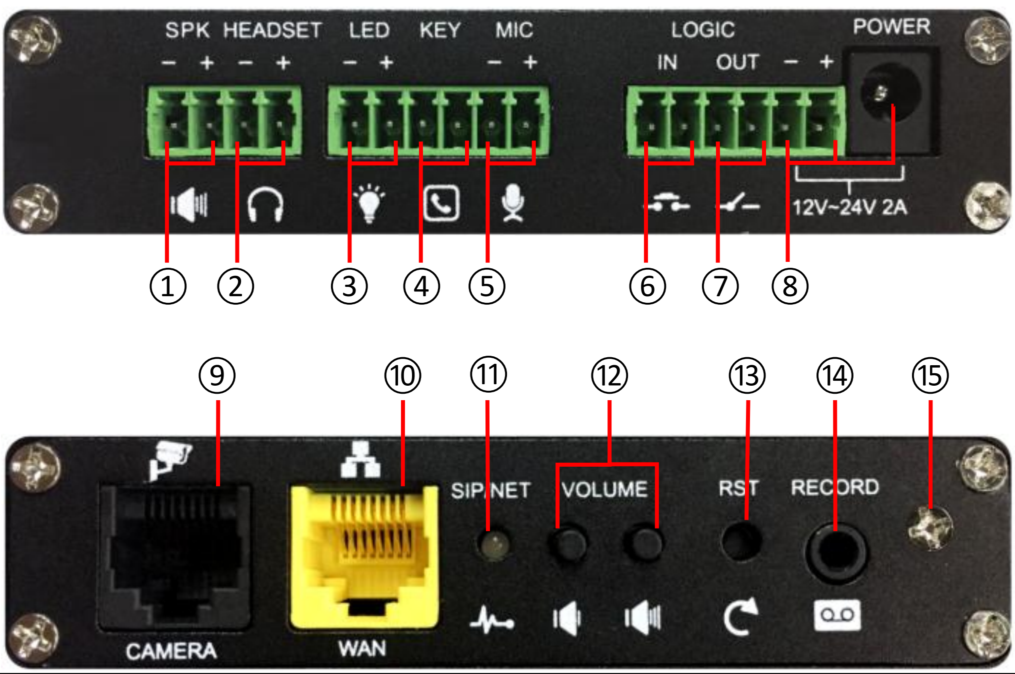
Directory

I. Product introduction

SPA2 is dedicated to the needs of industry users to develop a SIP audio and video intercom, voice transmission using standard IP / RTP protocol, video transmission using RTSP. It is well inherited Voptech phone stability, carrier-class sound quality advantages, and perfectly compatible with all current SIP-based mainstream IPPBX / softswitch / IMS platform, such as Asterisk, Broadsoft, 3CX, Elastix and so on. It is a variety of functional interface in one, call, radio, video, security, recording, to adapt to a variety of use of the environment, convenient and rapid deployment of equipment, is ideal for everyone.

1. Appearance of the product

2. Description



Label	Description	Label	Description	Label	Description
①	Speaker interface	②	Headset interface	③	LED interface
④	Function key interface	⑤	Microphone interface	⑥	Switch input interface
⑦	Switch output interface	⑧	Power input interface	⑨	Camera interface
⑩	Ethernet interface	⑪	Registration/Network LED	⑫	Volume control key

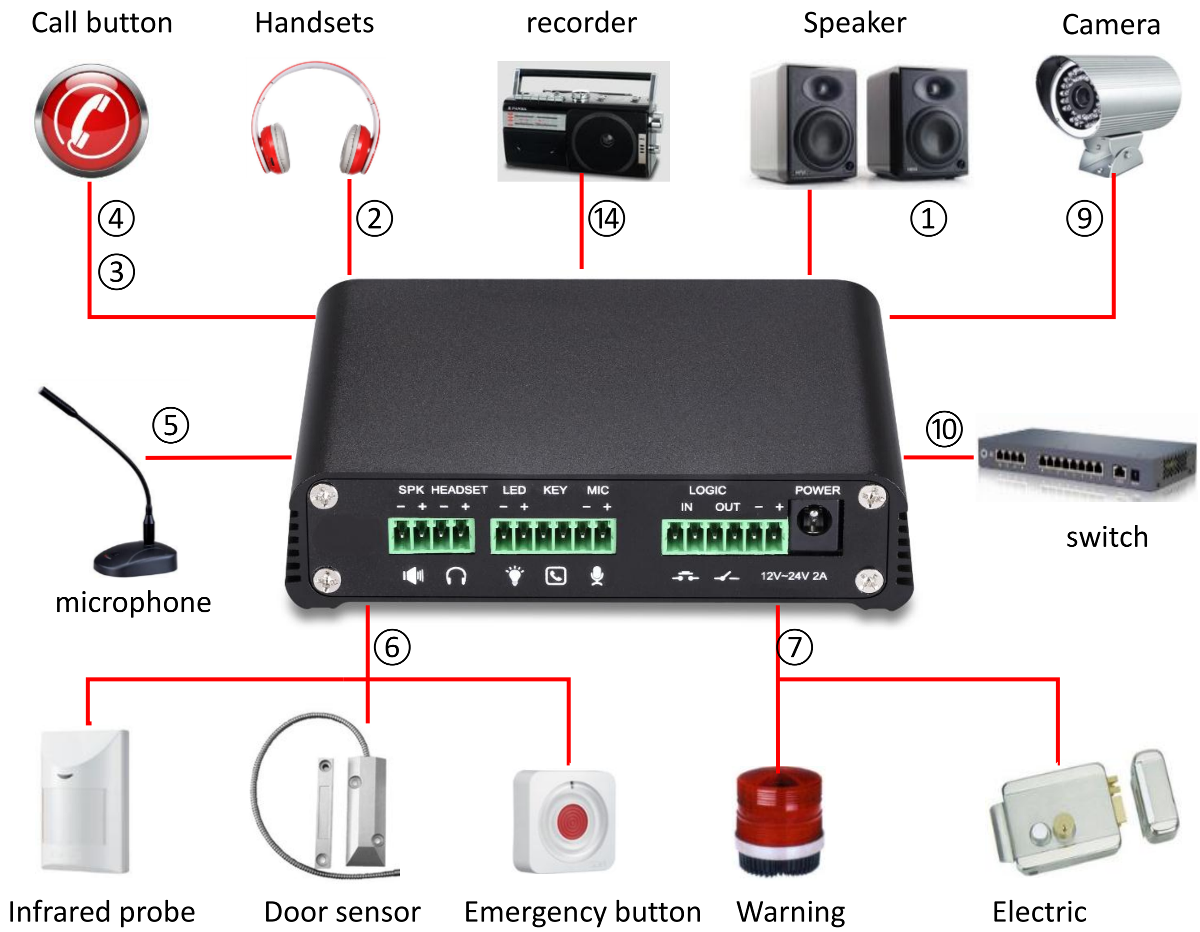
⑬	Restore factory key	⑭	Recording output interface	⑮	Grounding screw

II. Start Using

Before you start to use the equipment, please make the following installation.

1. Confirm the connection

Confirm whether the equipment of the power cord, network cable and the boot-up is normal. (Check the network state of light



Label	Explanation
① **Speaker interface**	according to the device input voltage adaptive output maximum power; 4Ω speaker, POE / 8W, 12V / 10W, 18V / 20W, 24V / 30W. The greater the horn impedance, the smaller the output power. Suggested wire diameter: 18AWG or larger diameter.
② **Headset interface**	speaker audio signal output, for external headset or active speakers.
③ **LED interface**	5V output, can be an external LED, indicating the network status, call status, registration status.
④ **Function key interface**	connection switch, you can log on page set the call number or IP address.

⑤ **Microphone interface**	the proposed use of electret condenser microphone, sensitivity: -38dB, bias voltage 2.2V. Microphone signal cable it is recommended to use a shielded cable and connect the shield cable to the grounding screw, improve anti-interference.
⑥ **Switch input interface**	Connect an infrared probe or emergency switch or Doorsensor and other switch components.
⑦ **Switch output interface**	corresponding to the short-circuit input interface, login device security page settings, you can control the alarm light, electric locks and other equipment; with the adjacent ⑧ power port connection for external equipment power supply.
⑧ **Power input interface**	12V ~ 24V 2A input, according to the input voltage to determine the maximum output power amplifier.
⑨ **Camera interface**	standard RJ45 interface, connect the original camera, the proposed use of five or five sub-network cable
⑩ **Ethernet interface**	WAN port, standard RJ45 interface, 10 / 100M adaptive, support POE input, it is recommended to use five or super five network cable.
⑪**Registration/Network LED**	indicates network status, call status, registration status. Fast flashing: network anomaly or SIP account exception. Slow flashing: during a call. Always bright: successful registration.
⑫**Volume control key**	standby to adjust the volume of the ringtone, call only adjust the call volume, broadcast only adjust the broadcast volume. Long press the volume down key to broadcast the IP address. Long press the volume plus key to switch the IP address acquisition mode (specific operation see below search door phone).
⑬**Restore factory key**	press and hold for 3 seconds to flash the device to restart and restore the factory settings.
⑭**Recording output interface**	the local microphone voice and call voice mixed output, suitable for computer and other equipment recording.
⑮**Grounding screw**	external parts metal housing needs to be connected to this interface to prevent static electricity and other interference caused the equipment to work abnormally.

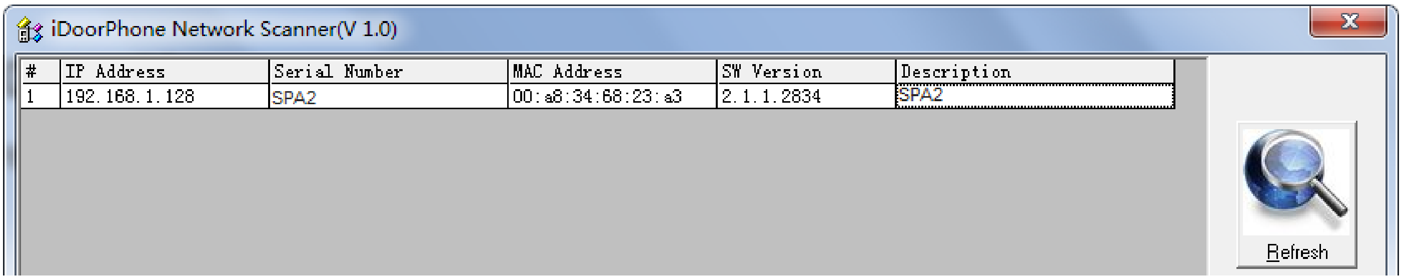
2. Quick Setting

The product provides a complete function and parameter setting. Users may need to have the network and SIP protocol knowledge to understand the meaning all parameters represent. In order to let equipment users, enjoy the high quality of voice service and low cost advantage brought by the device immediately, here we list some basic but necessary setting options in this section to let users know how to operate device without understanding such complex SIP protocols.

In prior to this step, please make sure your broadband Internet can be normally operated, and you must complete the connection of the network hardware. The product factory default network mode is static IP address 192.168.1.128.

- Press and hold volume down key for 3 seconds; the door phone would report the IP address by voice. Or you can also use the "iDoorPhoneNetworkScanner.exe" software to find the IP address of the device. (download address *www.Voptech.com*)

- Long press the volume plus key for 10 seconds, the speaker issued a rapid beep, and then quickly press the three volumes plus the key, beep stopped. Wait 10 seconds, successfully switch to dynamic IP after the system automatically voice broadcast IP address. Switching again will become a fixed IP address.
- Note: when the device is powered on, 30s waiting is needed for device running.
- Log on to the WEB device configuration.
- In a line configuration page, service account, user name, server address and other parameters are required for server address registration.



III. Basic operation

1. Answer a call

When a call comes in, the device would answer automatically. If you cancel auto answer feature and set auto answer time, you would hear the ring at the set time and the device would auto answer after configured timer.

2. Call

Configure Function key as hot key and then set up a number; after that you might press the Function key for making call to the configured extension(s).

3. End call

Enable Release（You can set Function key to Release）key for hanging up feature to end call.

4. Security linkage

- switch input interface received a door or emergency button and other sensor signals, the output port connected to the alarm lights or electric locks and other equipment will automatically respond to the server and send alarm information.
- The output port defaults to the call automatically triggers the response and supports call triggering with DTMF number triggering and short message triggering.

5. Video linkage

- Use another manufacturers camera please connect to the switch, the device ⑨ interface can only connect the original camera.

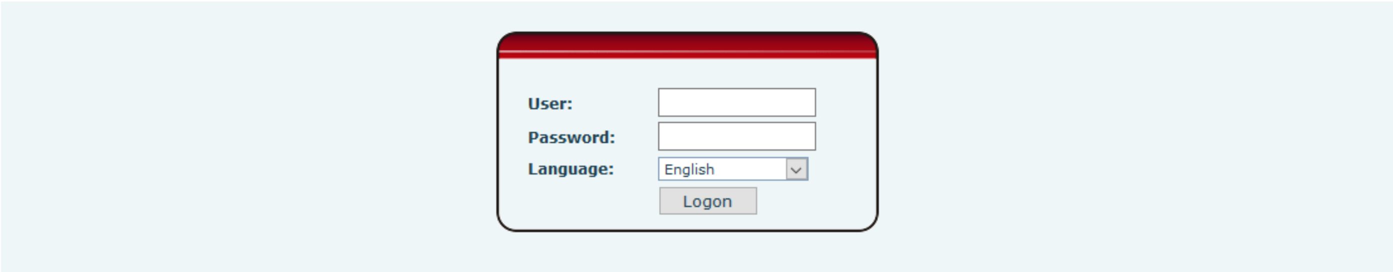
◆ Landing page configuration camera user name, password, port number and other information. For more information, please refer to the Video settings

IV. Page settings

1. Browser configuration

When the device and your computer are successfully connected to the network, you might enter the IP address of the device in the browser as http://xxx.xxx.xxx.xxx/ and you can see the login interface of the web page management.

Enter the user **admin** and password **admin** and click the Logon button to enter the settings screen.



2. Password Configuration

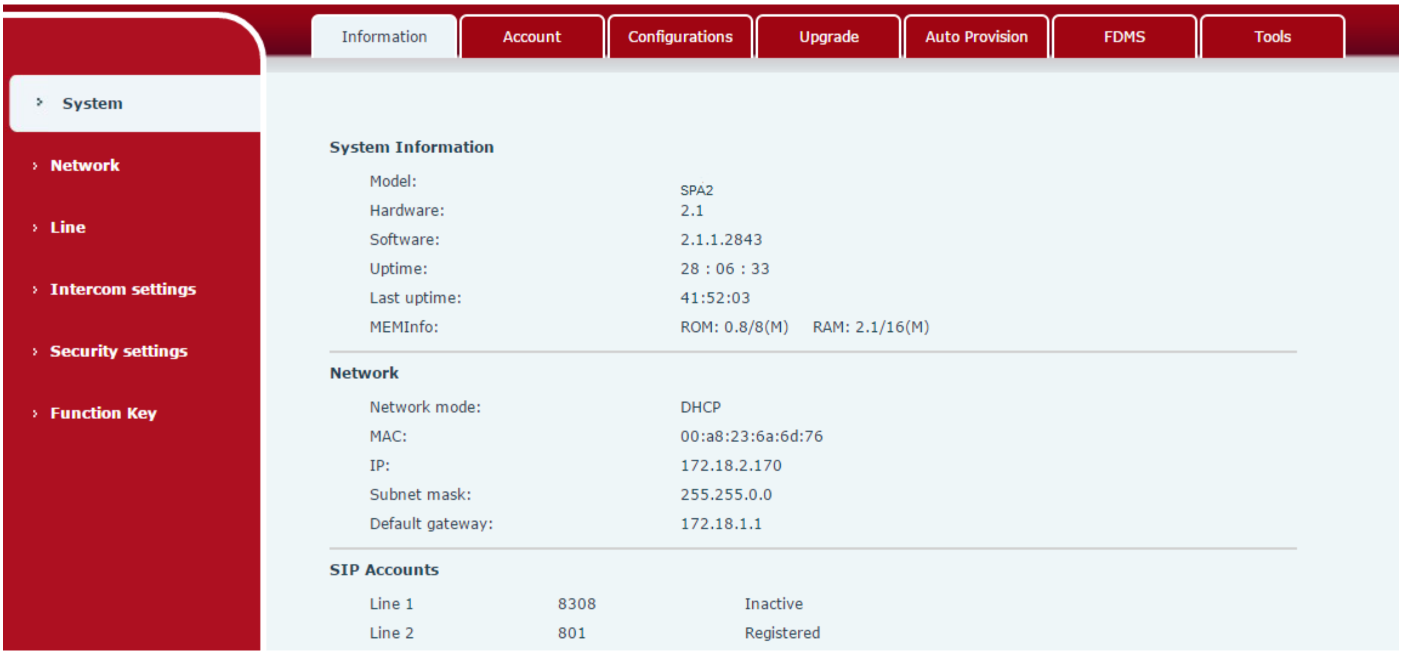
There are two levels of access: root level and general level. A user with root level can browse and set all configuration parameters, while a user with general level can set all configuration parameters except server parameters for SIP.

- General level: It is not be set by default, you can add the feature when you need
- User uses root level by default:
  - ◆ User name: admin
  - ◆ Password: admin

3. Configuration via WEB

(1) System

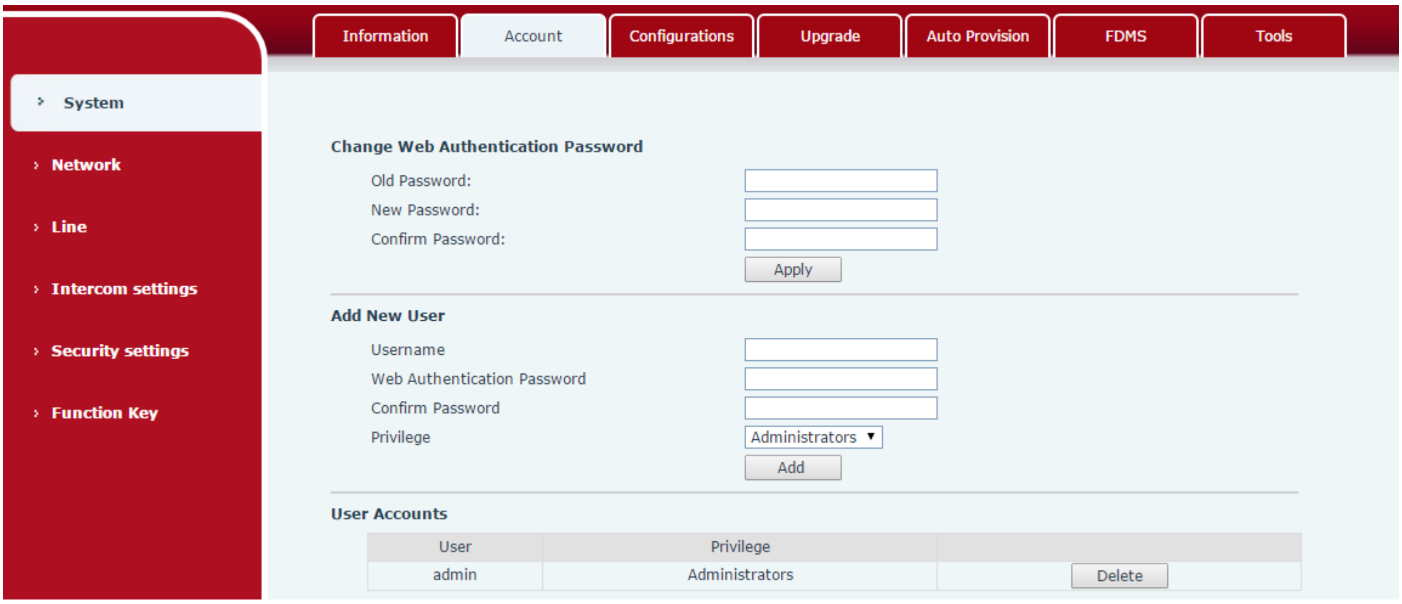
a) Information



Information	
Field Name	**Explanation**
System Information	Display equipment model, hardware version, software version, uptime, last uptime and MEMInfo.
Network	Shows the configuration information of WAN port, including connection mode of WAN port (Static, DHCP, PPPoE), MAC address, IP address of WAN port.
SIP Accounts	Shows the phone numbers and registration status of the 2 SIP LINES.

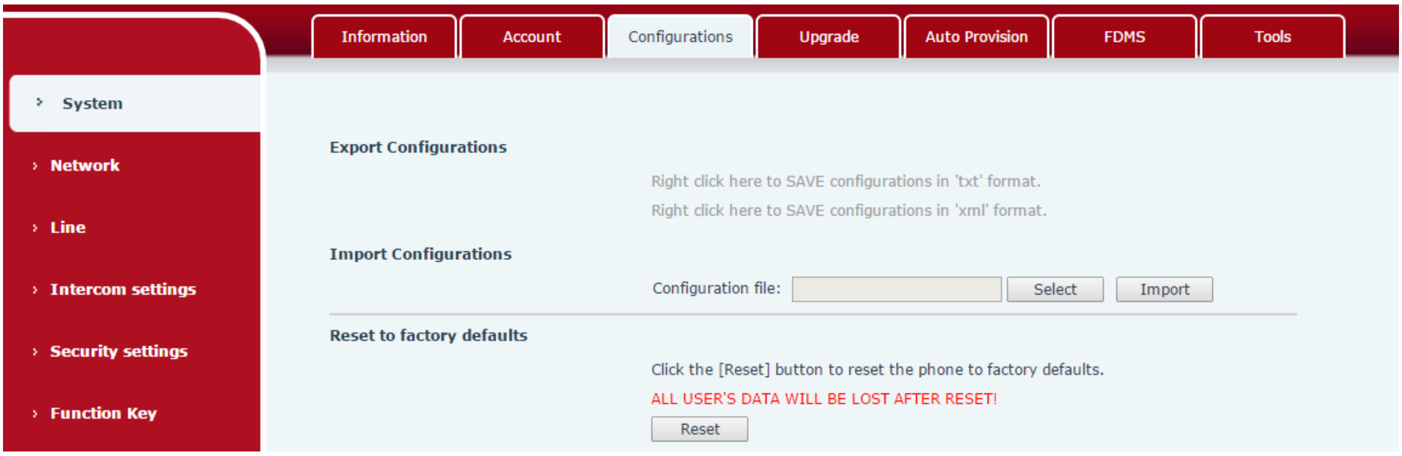
b) Account

Through this page, administrator can add or remove user accounts depend on their needs or modify existed user accounts permission.



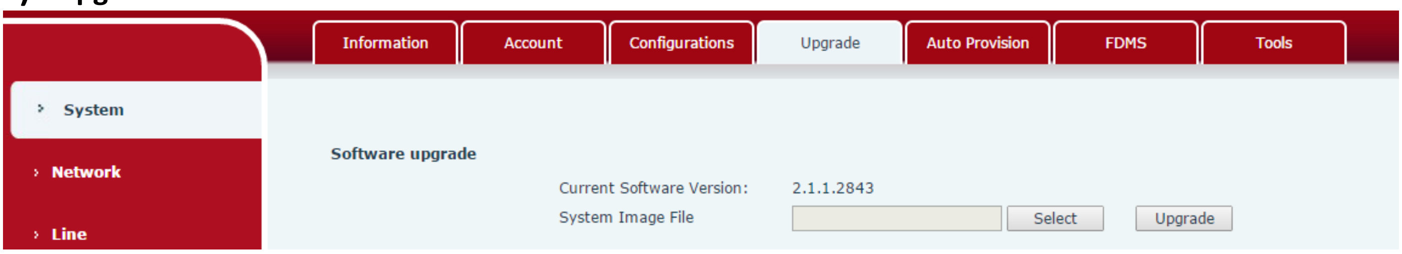
Account	
Field Name	**Explanation**
Change Web Authentication Password	
You can modify the login password of the account	
Add New User	
You can add new user	
User Accounts	
Show the existed user accounts' information	

c) Configurations



Configurations	
Field Name	**Explanation**
Export Configurations	Save the equipment configuration to a txt or xml file. Please right click on the choice and then choose "Save Link As."
Import Configurations	Find the config file, and press **Update** to load it to the equipment.
Reset to factory defaults	SPA2 would restore to factory default configuration and remove all configuration information.

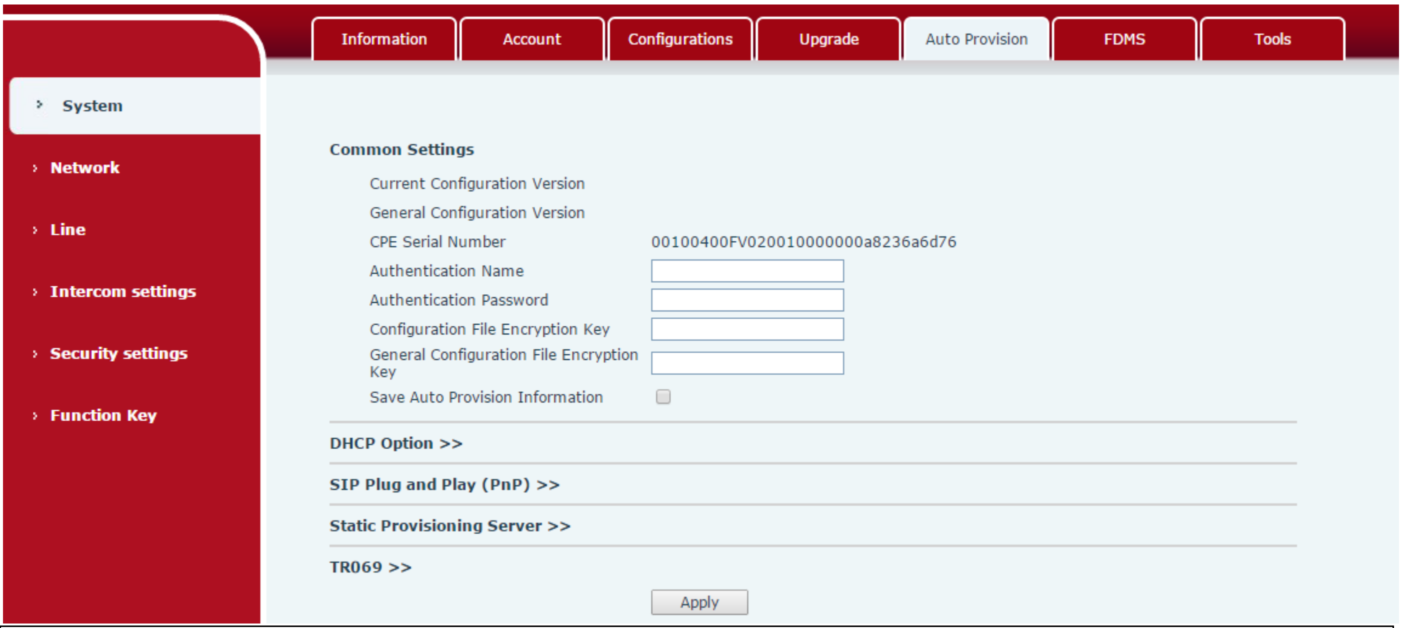
d) Upgrade



Upgrade	
Field Name	**Explanation**
Software upgrade	

Find the firmware, and press Update to load it to the equipment.

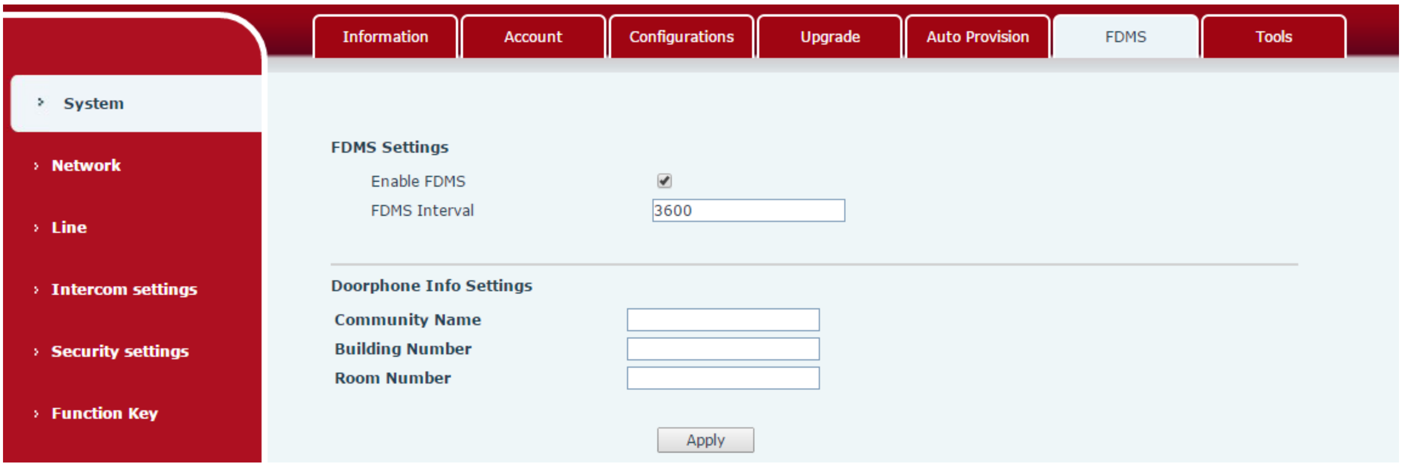
e) Auto Provision



Auto Provision	
Field Name	**Explanation**
Common Settings	
Current Configuration Version	Show the current config file's version. If the config file to be downloaded is higher than current version, the configuration would be upgraded. If the endpoints confirm the configuration by the Digest method, the configuration would not be upgraded unless it differs from the current configuration
General Configuration Version	Show the common config file's version. If the configuration to be downloaded and this configuration is the same, the auto provision would stop. If the endpoints confirm the configuration by the Digest method, the configuration would not be upgraded unless it differs from the current configuration.
CPE Serial Number	Serial number of the equipment
Authentication Name	Username for configuration server. It is used for FTP/HTTP/HTTPS. If this is blank, the phone would use anonymous access
Authentication Password	Password for configuration server. It is used for FTP/HTTP/HTTPS.
Configuration File Encryption Key	Encryption key for the configuration file
General Configuration File Encryption Key	Encryption key for common configuration file
Save Auto Provision Information	Save the auto provision username and password in the phone until the server url changed
DHCP Option	
Option Value	The equipment supports configuration from Option 43, Option 66, or a Custom DHCP option. It may also be disabled.

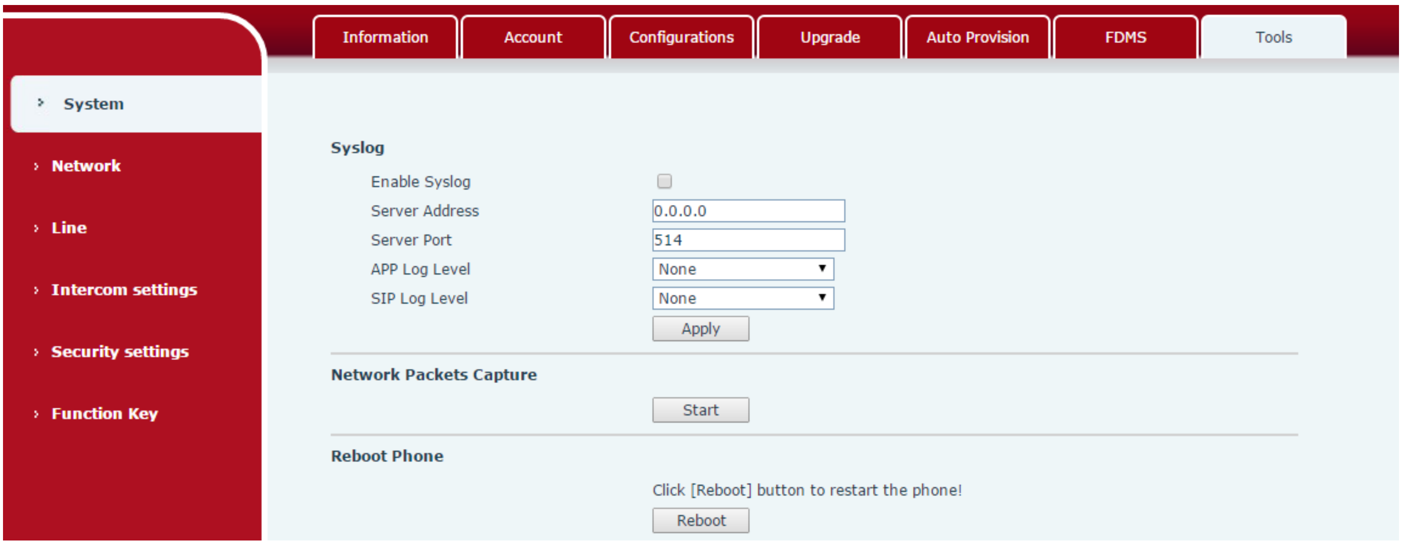
Custom Option Value	Custom option number. It must be from 128 to 254.
SIP Plug and Play (PnP)	
Enable SIP PnP	If it is enabled, the equipment would send SIP SUBSCRIBE messages to the server address when it boots up. Any SIP server compatible with that message would reply with a SIP NOTIFY message containing the Auto Provisioning Server URL where the phones can request their configuration.
Server Address	PnP Server Address
Server Port	PnP Server Port
Transportation Protocol	PnP Transfer protocol – UDP or TCP
Update Interval	Interval time for querying PnP server. Default is 1 hour.
Static Provisioning Server	
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be an IP address or domain name with subdirectory.
Configuration File Name	Specify configuration file name. The equipment would use its MAC ID as the config file name if this is blank.
Protocol Type	Specify the Protocol type FTP, TFTP or HTTP.
Update Interval	Specify the update interval time. Default is 1 hour.
Update Mode	1. Disable – not to update 2. Update after reboot – update only after reboot. 3. Update at time period – update at periodic update period
TR069	
Enable TR069	Enable/Disable TR069 configuration
ACS Server Type	Select Common or CTC ACS Server Type.
ACS Server URL	ACS Server URL.
ACS User	User name of ACS.
ACS Password	ACS Password.
TR069 Auto Login	Enable/Disable TR069 Auto Login.
INFORM Sending Period	Time between transmissions of “Inform”; the unit is second.

f) FDMS



FDMS Settings	
Enable FDMS	Enable/Disable FDMS configuration
FDMS Interval	The time to send sip Subscribe information to the FDMS server on a regular basis. Unit seconds
Doorphone Info Settings	
Community Name	The name of the community where the device is installed
Building Number	The name of the building where the equipment is installed
Room Number	The name of the room where the equipment is installed

g) Tools



Syslog is a protocol used to record log messages using a client/server mechanism. The Syslog server receives the messages from clients and classifies them based on priority and type. Then these messages would be written into a log by rules which the administrator has configured.

There are 8 levels of debug information.

Level 0: emergency; System is unusable. This is the highest debug info level.

Level 1: alert; Action must be taken immediately.

Level 2: critical; System is probably working incorrectly.

Level 3: error; System may not work correctly.

Level 4: warning; System may work correctly but needs attention.

Level 5: notice; It is normal but significant condition.

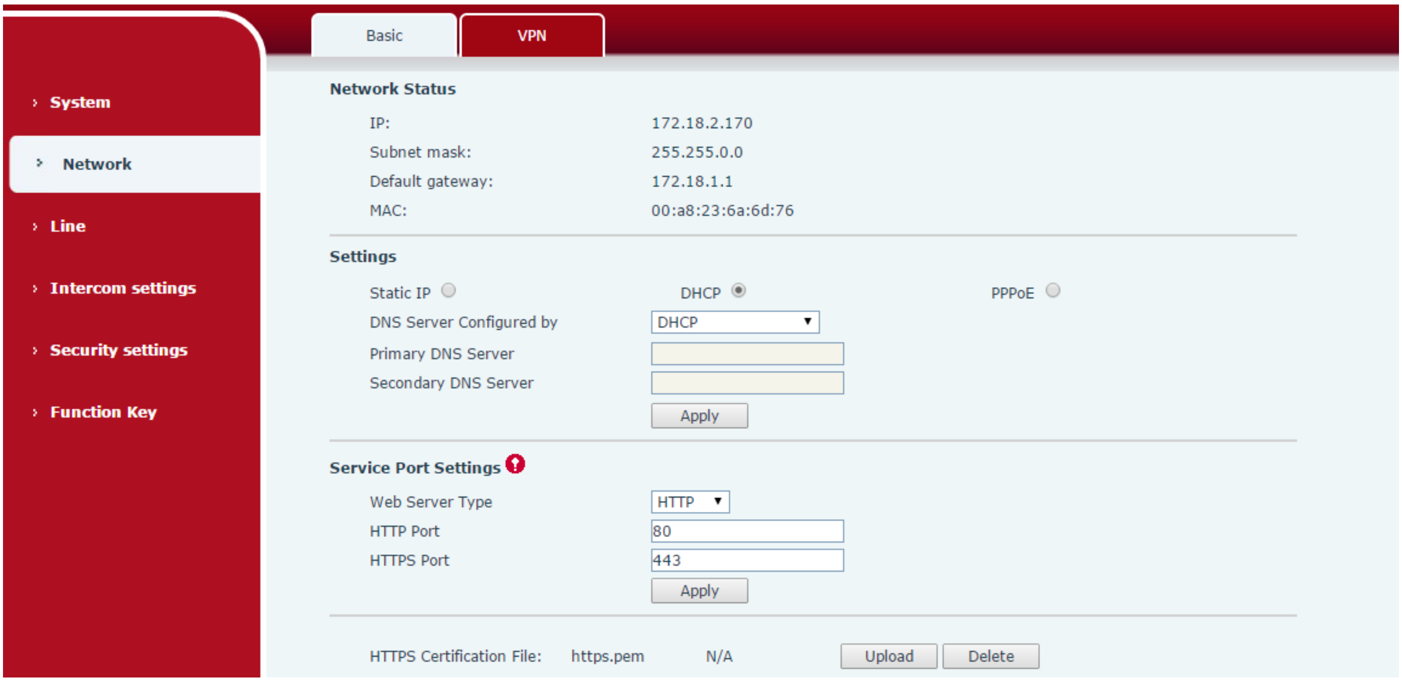
Level 6: Informational; It is normal daily messages.

Level 7: debug; Debug messages normally used by system designer. This level can only be displayed via telnet.

Tools	
Field Name	**Explanation**
Syslog	
Enable Syslog	Enable or disable system log.
Server Address	System log server IP address.
Server Port	System log server port.
APP Log Level	Set the level of APP log.
SIP Log Level	Set the level of SIP log.
Network Packets Capture	
Capture a packet stream from the equipment. This is normally used to troubleshoot problems.	
Reboot Phone	
Some configuration modifications require a reboot to become effective. Clicking the Reboot button would lead to reboot immediately. Note: Be sure to save the configuration before rebooting.	

(2) Network

a) Basic

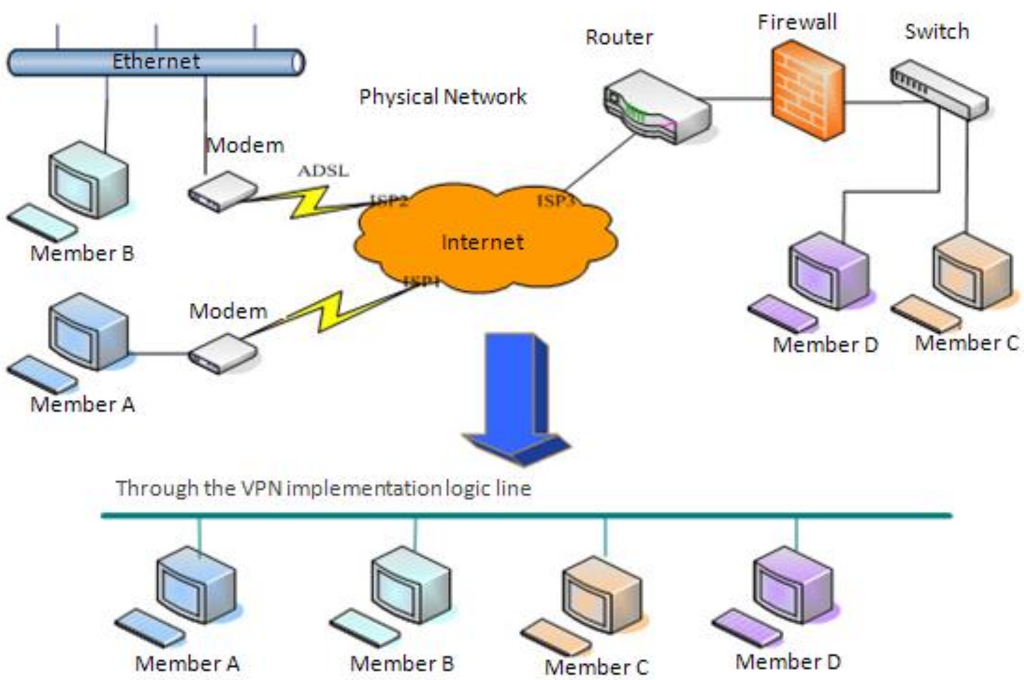


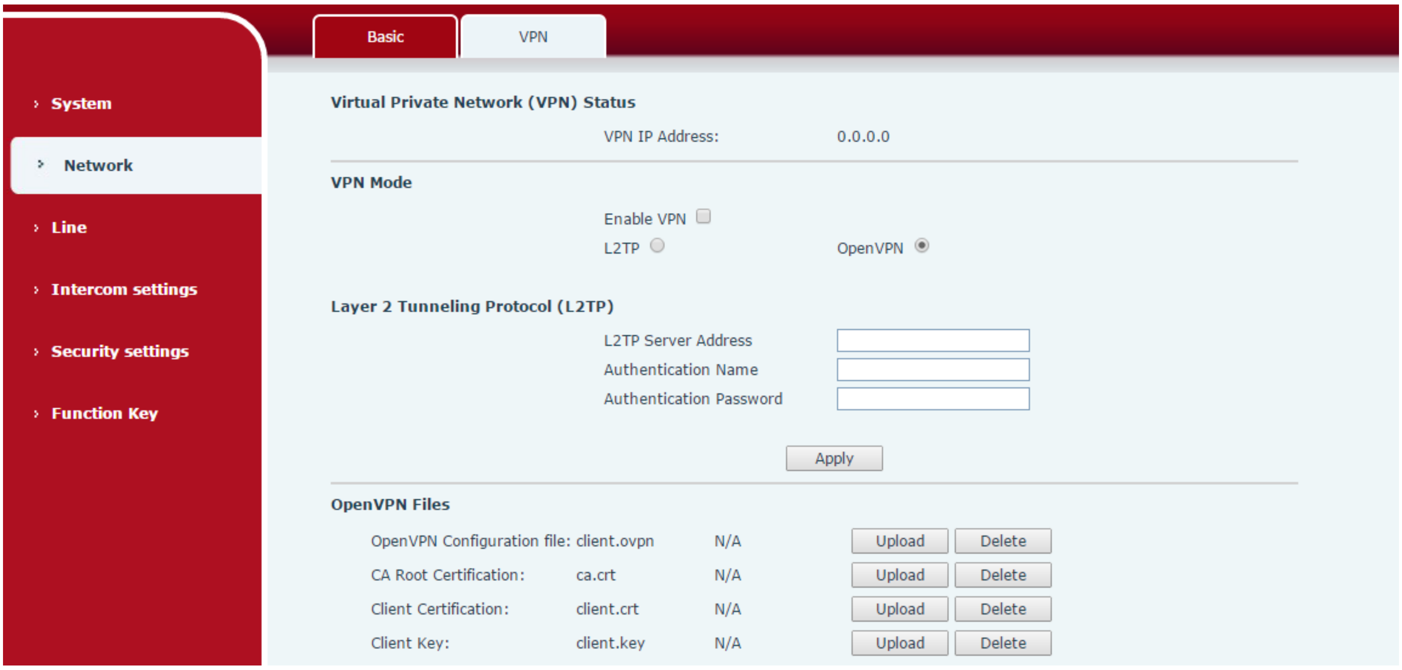
Field Name	**Explanation**
Network Status	
IP	The current IP address of the equipment
Subnet mask	The current Subnet Mask
Default gateway	The current Gateway IP address

MAC	The MAC address of the equipment
MAC Timestamp	Get the MAC address's time.
Settings	
Select the appropriate network mode. The equipment supports three network modes:	
Static IP	Network parameters must be entered manually and would 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 would appear. Enter values provided by the ISP.	
DNS Server Configured by	Select the Configured mode of the DNS Server.
Primary DNS Server	Enter the server address of the Primary DNS.
Secondary DNS Server	Enter the server address of the Secondary DNS.
After entering the new settings, click the **Apply** button. The equipment would save the new settings and apply them. If a new IP address was entered for the equipment, it must be used to login to the phone after clicking the **Apply** button.	

b) VPN

The device supports remote connection via VPN. It supports both Layer 2 Tunneling Protocol (L2TP) and OpenVPN protocol. This allows users at remote locations on the public network to make secure connections to local networks.



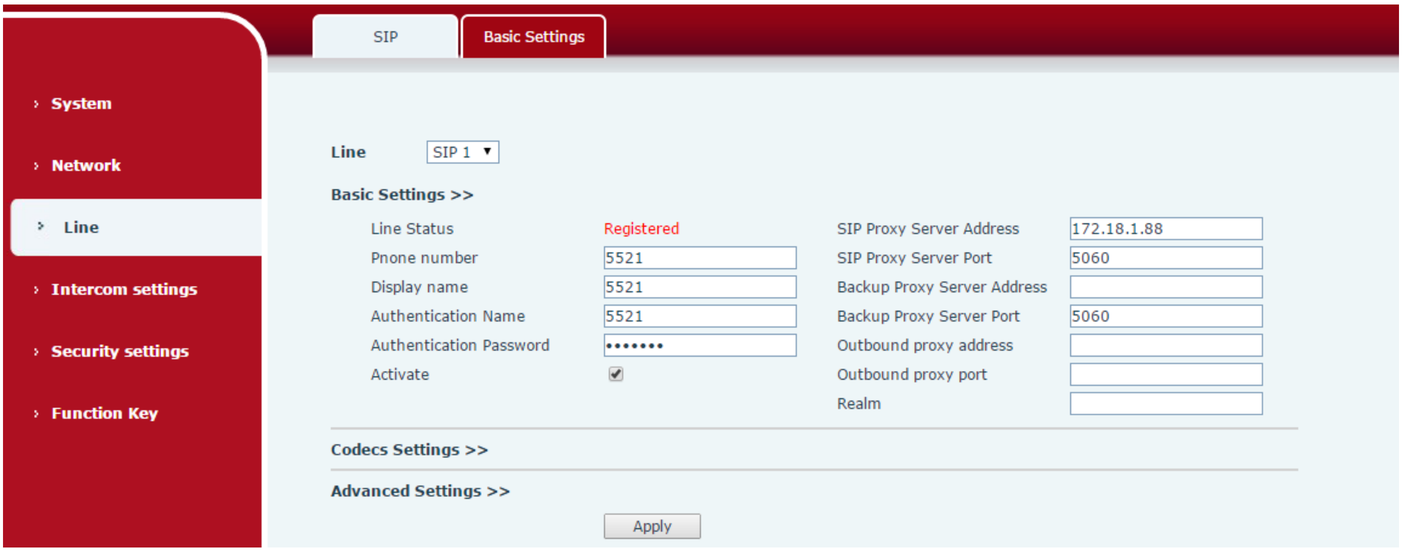


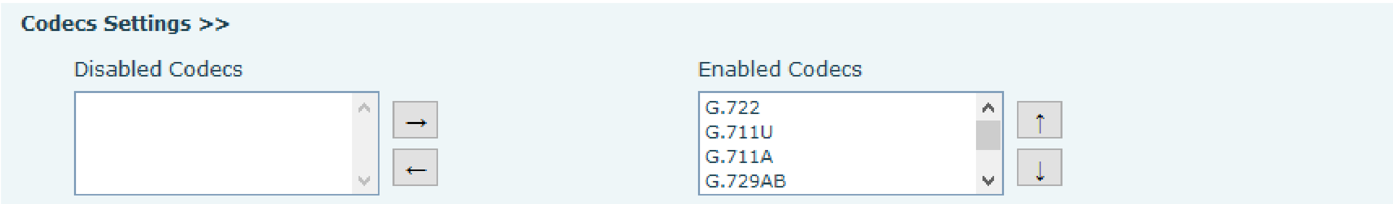
Field Name	Explanation
VPN IP Address	Shows the current VPN IP address.
VPN Mode	
Enable VPN	Enable/Disable VPN.
L2TP	Select Layer 2 Tunneling Protocol
OpenVPN	Select OpenVPN Protocol. (Only one protocol may be activated. After the selection is made, the configuration should be saved, and the phone be rebooted.)
Layer 2 Tunneling Protocol (L2TP)	
L2TP Server Address	Set VPN L2TP Server IP address.
Authentication Name	Set User Name access to VPN L2TP Server.
Authentication Password	Set Password access to VPN L2TP Server.
Open VPN Files	
Upload or delete Open VPN Certification Files	

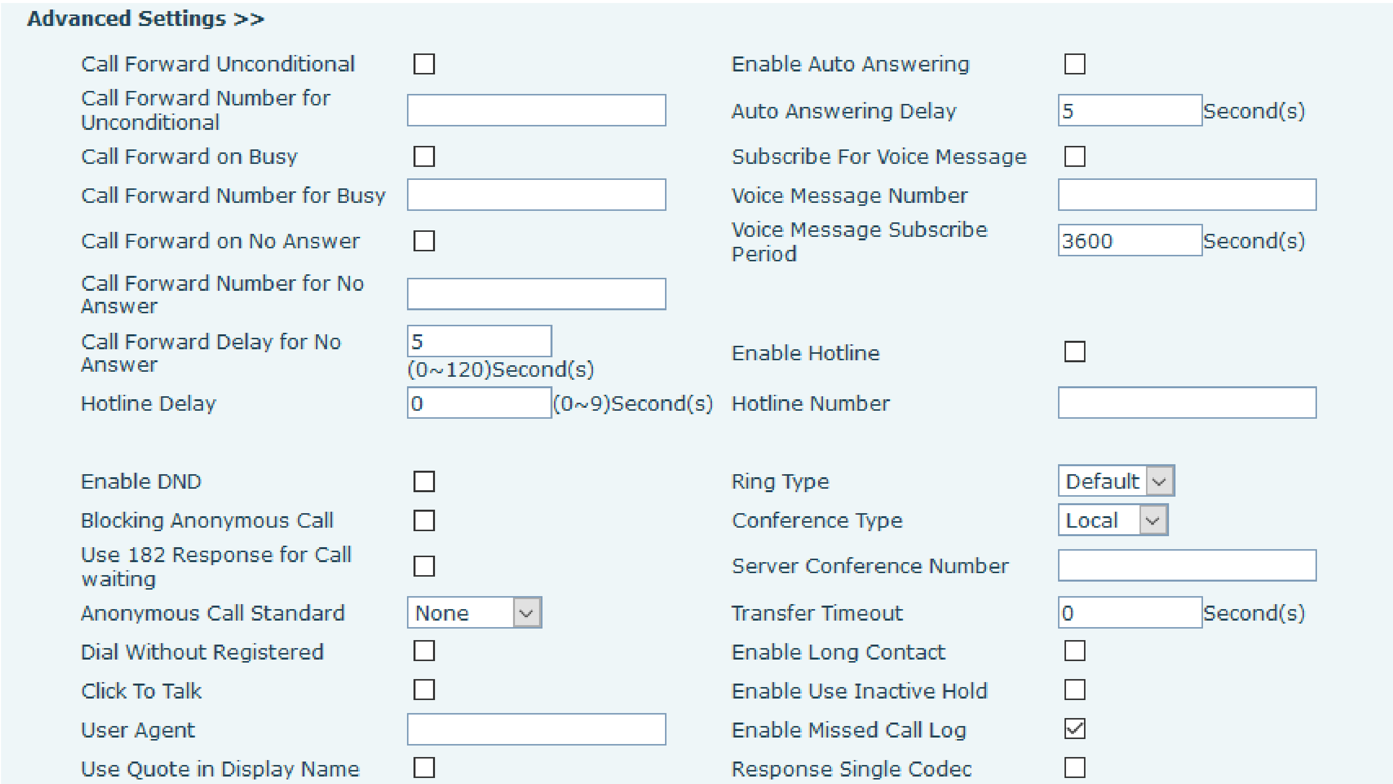
(3) Line

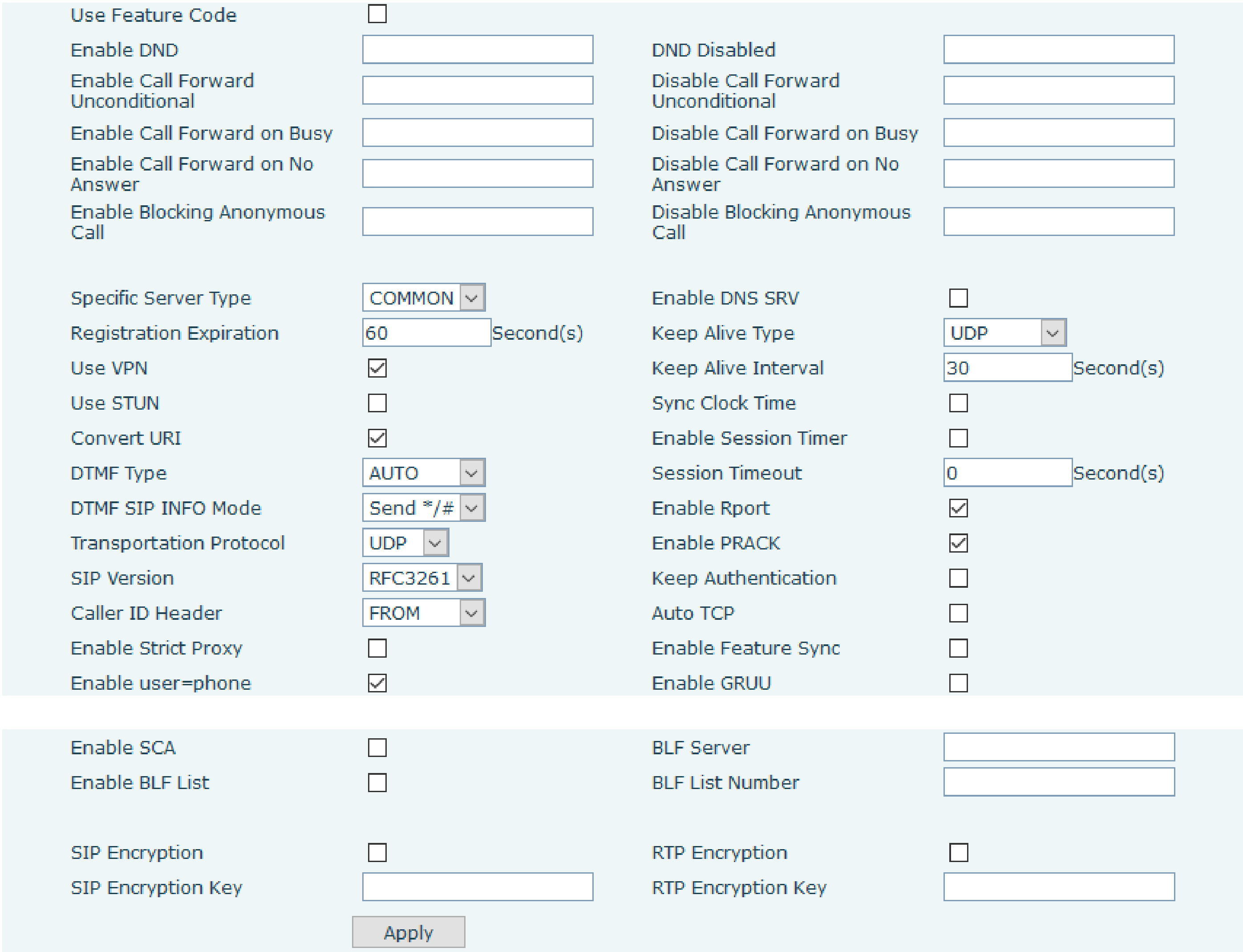
a) SIP

You can configure a SIP server on this page.


SIP
Basic Settings
System
Network
Line
Intercom settings
Security settings
Function Key
Line SIP 1
Basic Settings >>
Line Status Registered
Phone number 5521
Display name 5521
Authentication Name 5521
Authentication Password
Activate
SIP Proxy Server Address 172.18.1.88
SIP Proxy Server Port 5060
Backup Proxy Server Address
Backup Proxy Server Port 5060
Outbound proxy address
Outbound proxy port
Realm
Codecs Settings >>
Advanced Settings >>
Apply


Codecs Settings >>
Disabled Codecs
Enabled Codecs
G.722
G.711U
G.711A
G.729AB


Advanced Settings >>
Call Forward Unconditional
Call Forward Number for Unconditional
Call Forward on Busy
Call Forward Number for Busy
Call Forward on No Answer
Call Forward Number for No Answer
Call Forward Delay for No Answer 5 (0~120)Second(s)
Hotline Delay 0 (0~9)Second(s)
Enable DND
Blocking Anonymous Call
Use 182 Response for Call waiting
Anonymous Call Standard None
Dial Without Registered
Click To Talk
User Agent
Use Quote in Display Name
Enable Auto Answering
Auto Answering Delay 5 Second(s)
Subscribe For Voice Message
Voice Message Number
Voice Message Subscribe Period 3600 Second(s)
Enable Hotline
Hotline Number
Ring Type Default
Conference Type Local
Server Conference Number
Transfer Timeout 0 Second(s)
Enable Long Contact
Enable Use Inactive Hold
Enable Missed Call Log
Response Single Codec



SIP	
Field Name	**Explanation**
Basic Settings (Choose the SIP line to configured)	
Line Status	Display the current line status after page loading. To get the up to date line status, user must refresh the page manually.
Username	Enter the username of the service account.
Display name	Enter the display name to be sent in a call request.
Authentication Name	Enter the authentication name of the service account
Authentication Password	Enter the authentication password of the service account
Activate	Whether the service of the line should be activated
SIP Proxy Server Address	Enter the IP or FQDN address of the SIP proxy server
SIP Proxy Server Port	Enter the SIP proxy server port, default is 5060
Outbound proxy address	Enter the IP or FQDN address of outbound proxy server provided by the service provider
Outbound proxy port	Enter the outbound proxy port, default is 5060
Realm	Enter the SIP domain if it is needed by the service provider

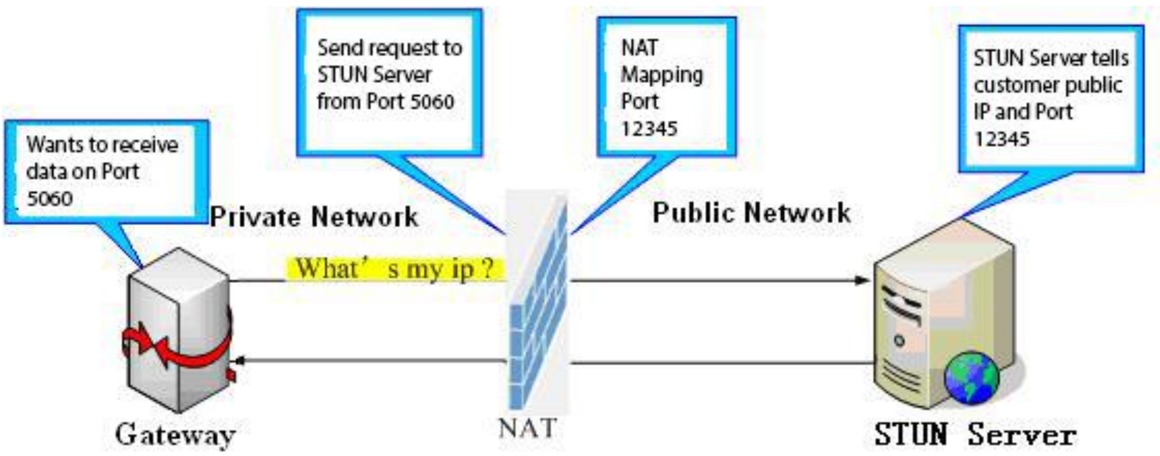
Codecs Settings	
Set the priority and availability of the codecs by adding or removing them from the list.	
Advanced Settings	
Call Forward Unconditional	Enable unconditional call forwarding, all incoming calls would be forwarded to the number specified in the next field
Call Forward Number for Unconditional	Set the number of unconditional call forwarding
Call Forward on Busy	Enable call forward on busy, when the phone is busy, any incoming call would be forwarded to the number specified in the next field
Call Forward Number for Busy	Set the number of call forwarding when the SPA2 is 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 would 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
Hotline Delay	Set the delay for hotline before the system automatically dial it
Enable Auto Answering	Enable auto-answering, the incoming calls would be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the system automatically answered answer it
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if you enable it , the device would 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 period of voice message notification subscription
Enable Hotline	Enable hotline configuration, the device would dial to the specific number immediately at audio channel opened by off-hook or turning on hands-free speaker or headphone
Hotline Number	Set the hotline dialing number
Enable DND	Enable Do-not-disturb, any incoming call on this line would be rejected automatically
Blocking Anonymous Call	Reject any incoming call without presenting caller ID
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Anonymous Call Standard	Set the standard to be used for anonymous call
Dial Without Registered	Set call out by proxy without registration

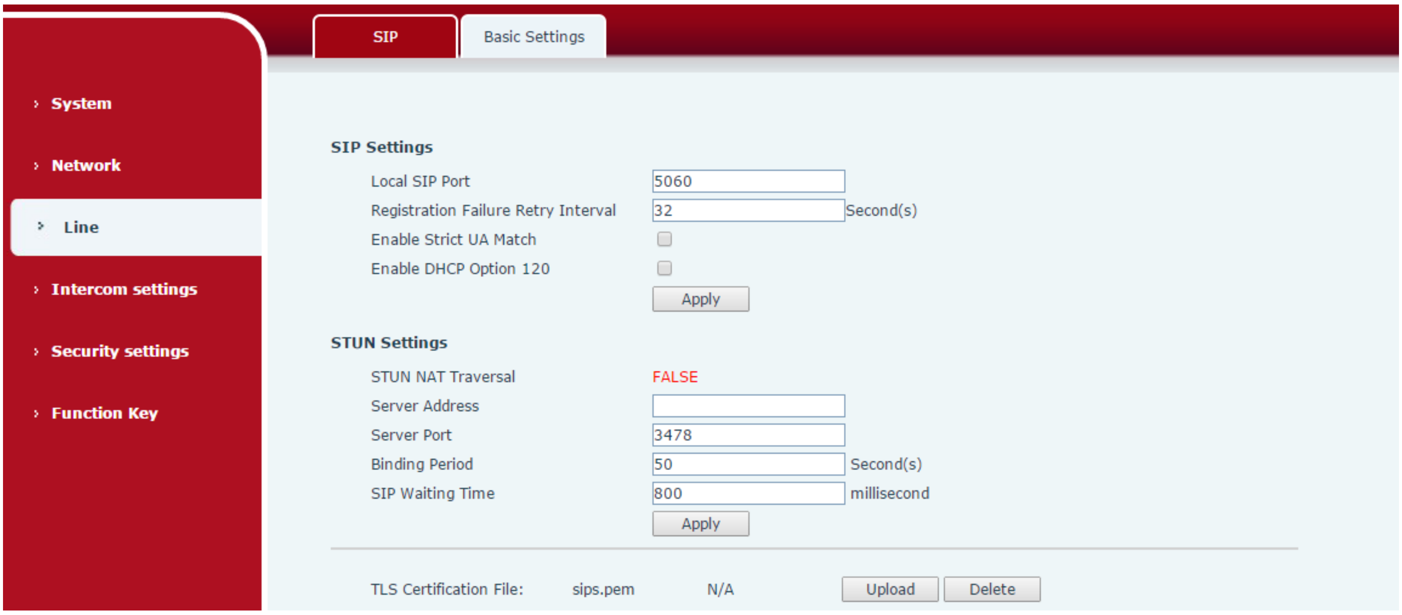
Click To Talk	Set Click To Talk
User Agent	Set the user agent, the default is Model with Software Version.
Use Quote in Display Name	Whether to add quote in display name
Ring Type	Set the ring tone type for the line
Conference Type	Set the type of call conference, Local=set up call conference by the device itself; SPA2 maximally 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 be Server
Transfer Timeout	Set the timeout of call transfer process
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Missed Call Log	If it is enabled, the phone would save missed calls into the call history record.
Response Single Codec	If it is enabled, the device would use single codec in response to an incoming call request
Use Feature Code	When this setting is enabled, the features in this section would not be handled by the device itself but by the server instead. In order to control the authorization of the features, the device would send feature code to the server by dialing the number specified in each feature code field.
Specific Server Type	Set the line to collaborate with specific server type
Registration Expiration	Set the SIP expiration period
Use VPN	Set the line to use VPN restrict route
Use STUN	Set the line to use STUN for NAT traversal
Convert URI	Convert not digit and alphabet characters to %hh hex code
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'
Transportation Protocol	Set the line to use TCP or UDP for SIP transmission
SIP Version	Set the SIP version
Caller ID Header	Set the Caller ID Header
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server, it would use the source IP address, not the address in via field.
Enable user=phone	Sets user=phone in SIP messages.
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
Enable BLF List	Enable/Disable BLF List
Enable DNS SRV	Set the line to use DNS SRV which would resolve the FQDN in proxy server into a service list
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
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session would be ended if there is not new session timer event updating received after

	the timeout period
Session Timeout	Set the session timer timeout period
Enable Rport	Set the line to add Rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
Keep Authentication	Keep the authentication parameters of previous authentication
Auto TCP	Using TCP protocol to guarantee usability of transport when SIP messages have more than 1500 bytes
Enable Feature Sync	Feature Sync with server
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
BLF Server	The registered server would 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 would be separated.
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
SIP Encryption	Enable SIP encryption such that SIP transmission would be encrypted
SIP Encryption Key	Set the pass phrase for SIP encryption
RTP Encryption	Enable RTP encryption such that RTP transmission would be encrypted
RTP Encryption Key	Set the pass phrase for RTP encryption

b) Basic Settings

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

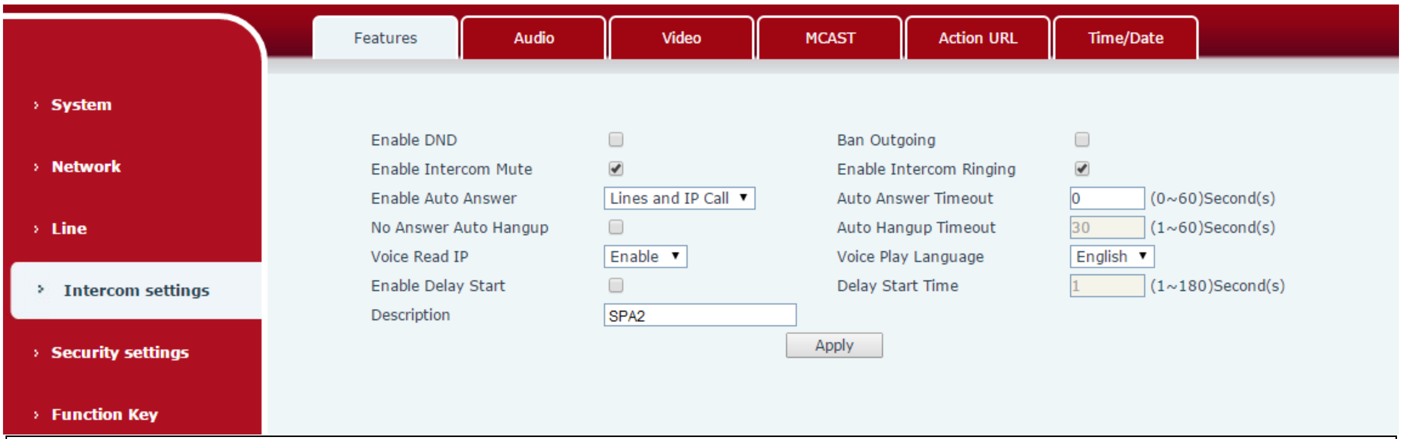




Basic Settings	
Field Name	**Explanation**
SIP Settings	
Local SIP Port	Set the local SIP port used to send/receive SIP messages.
Registration Failure Retry Interval	Set the retry interval of SIP registration when registration failed.
STUN Settings	
Server Address	STUN Server IP address
Server Port	STUN Server Port – Default is 3478.
Binding Period	STUN blinding period – STUN packets are sent once every this period to keep the NAT mapping active.
SIP Waiting Time	Waiting time for SIP. This would vary depending on the network.
SIP Line Using STUN(SIP1 or SIP2)	
Use STUN	Enable/Disable STUN on the selected line.
TLS Certification File	
Upload or delete the TLS certification file used for encrypting SIP transmission.	
Note: the SIP STUN is used to achieve the penetration of SIP NAT; it is a realization of service, when the equipment is configured the STUN server IP and port (usually the default is 3478), and selected “Use Stun SIP server”, you can make common SIP equipment achieve penetration.	

(4) Intercom settings

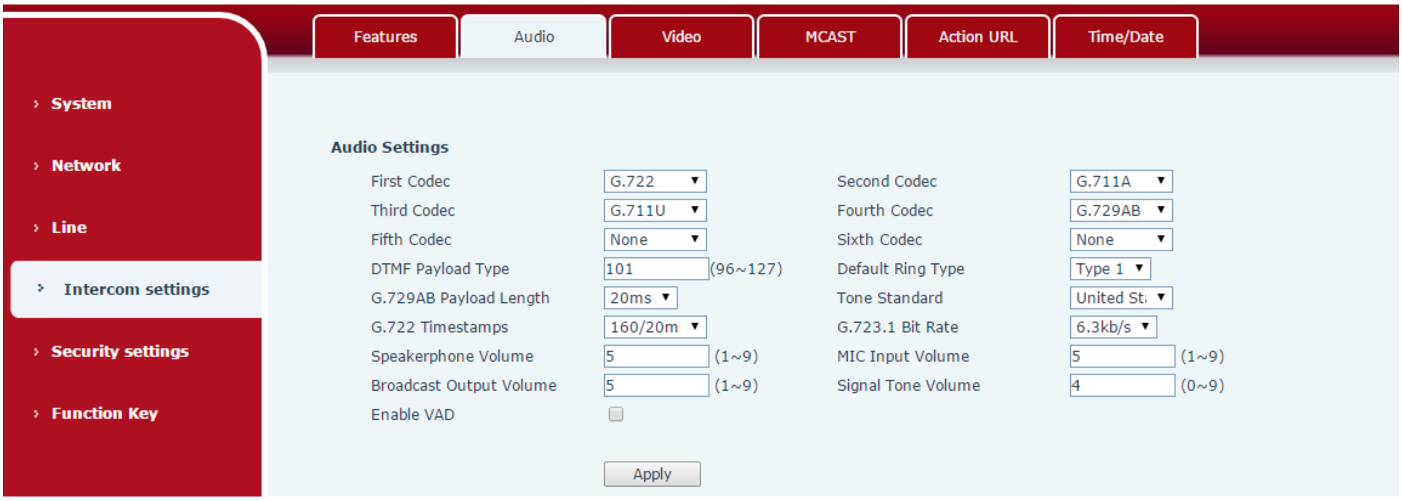
a) Features



Features	
Field Name	**Explanation**
Basic Settings	
Enable DND	DND feature can refuse all incoming calls for all SIP lines, or for individual SIP line. But the outgoing calls would not be affected
Ban Outgoing	If it is enabled, no outgoing calls can be made.
Enable Intercom Mute	If it is enabled, device would mute incoming calls during an intercom call.
Enable Intercom Ringing	If it is enabled, device would play intercom ring tone to alert that there is a new incoming call during an intercom call.
Enable Auto Answer	Enable Auto Answer function
Auto Answer Timeout	Set Auto Answer Timeout
No Answer Auto Hangup	Enable automatically hang up feature when there is no answer
Auto Hangup Timeout	Configuration in a set time, the device would automatically hang up when there is no answer
Voice Read IP	Enable or disable voice broadcast IP address
Voice Play Language	Set language of the voice prompt
Enable Delay Start	Enable or disable the start delay
Delay Start Time	Set start delay time
Description	Device description displayed on IP scanning tool software or FDMS.

b) Audio

This page configures audio parameters such as voice codec, speak volume, mic volume and ringer volume.

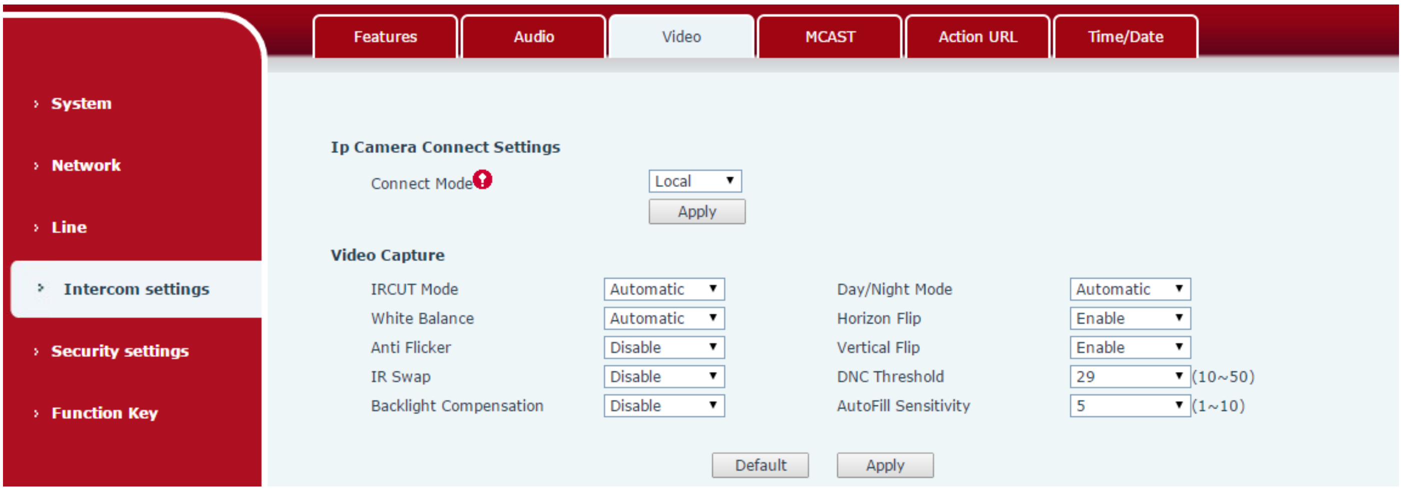


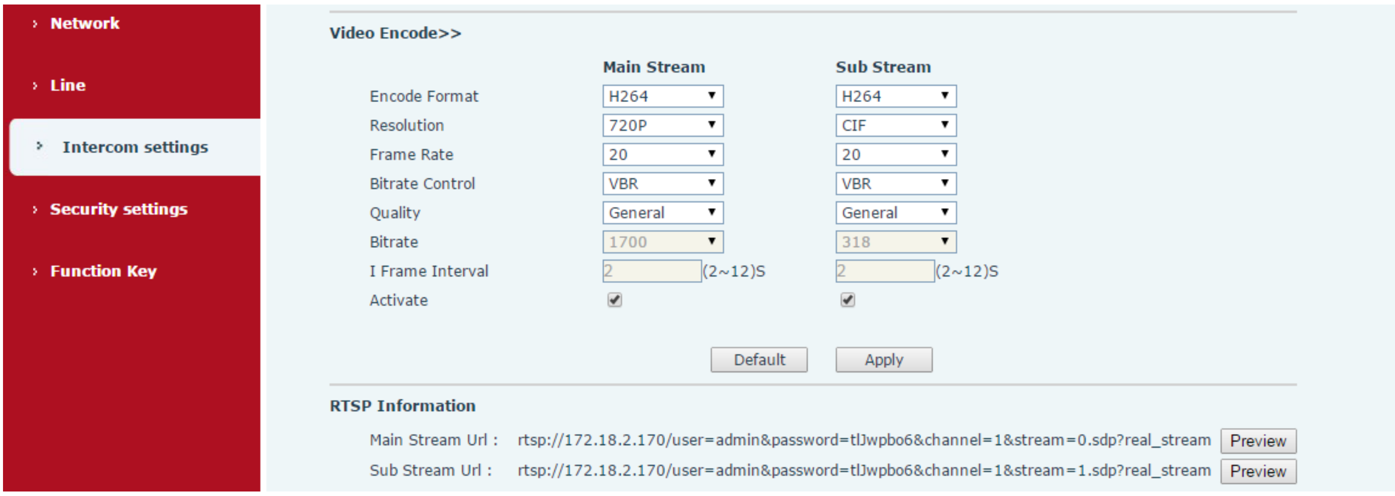
Audio Setting	
Field Name	**Explanation**
First Codec	The first codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB
Second Codec	The second codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None
Third Codec	The third codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None
Fourth Codec	The forth codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None
DTMF Payload Type	The RTP Payload type that indicates DTMF. Default is 101
Default Ring Type	Ring sound – there are 9 standard types and 3 user types.
G.729AB Payload Length	G.729AB Payload length – adjust from 10 – 60 msec.
Tone Standard	Configure tone standard area.
G.722 Timestamps	Choices are 160/20ms or 320/20ms.
G.723.1 Bit Rate	Choices are 5.3kb/s or 6.3kb/s.
Speakerphone Volume	Set the speaker call volume level.
MIC Input Volume	Set the MIC call volume level.
Broadcast Output Volume	Set the broadcast output volume level.
Signal Tone Volume	Set the audio signal output volume level.
Enable VAD	Enable or disable Voice Activity Detection (VAD). If VAD is enabled, G729 Payload length cannot be set greater than 20 msec.

c) Video

This page allows you to set the video capture and video encode.

Local Mode



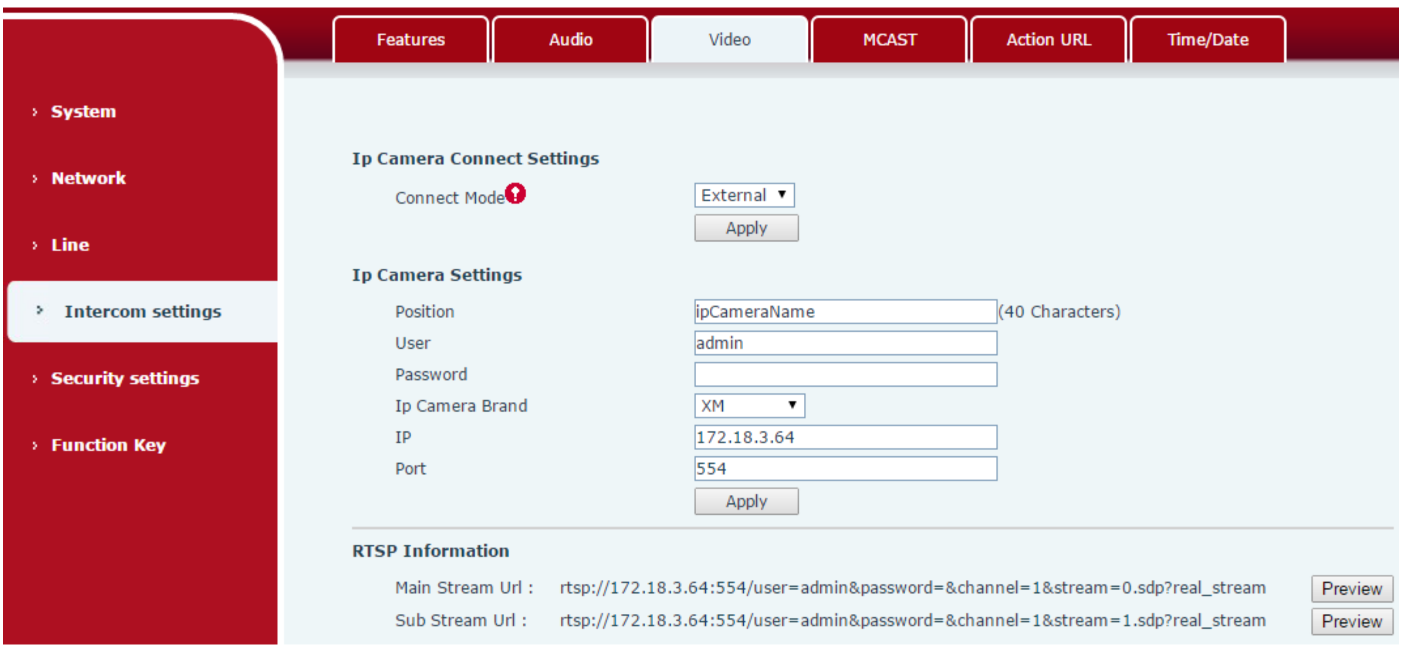


Camera Connect Settings	
Field Name	**Explanation**
Connect Mode	Local: Connect the original camera External: Connect to another manufacturers camera
Video Capture (Local Mode)	
IRCUT Mode	Auto: IRCUT switches according to the actual ambient light level of the camera Synchronization: The switching of the IRCUT is determined by the actual brightness of the IR lamp.
Day/Night Mode	Automatic: automatically switches according to the DNC Threshold and the brightness of the actual environment where the camera is located Day Mode: The camera's video screen is always colored, if there is IR-cut will be synchronized to switch. Night Mode: the camera's video screen is always black and white, if there is IR-cut will be synchronized switch.
White Balance	Automatic: Automatically adjusts according to the actual environment in which the camera is located. Outdoor: installed in the outdoor preferred. Indoor: installed in the room preferred.
Horizon Flip	The video is flipped horizontally
Anti Flicker	Enable the option. In a fluorescent environment can eliminate the video horizontal

	scroll
Vertical Flip	The video is flipped horizontally
IR Swap	IR-cut filter switch
DNC Threshold	In the Day / Night mode Auto option, the color switching black and white threshold is set
Backlight Compensation	In front of a very strong background light can see people or objects clearly
AutoFill Sensitivity	In the environment changes in light and shade, the higher the sensitivity the faster the video changes

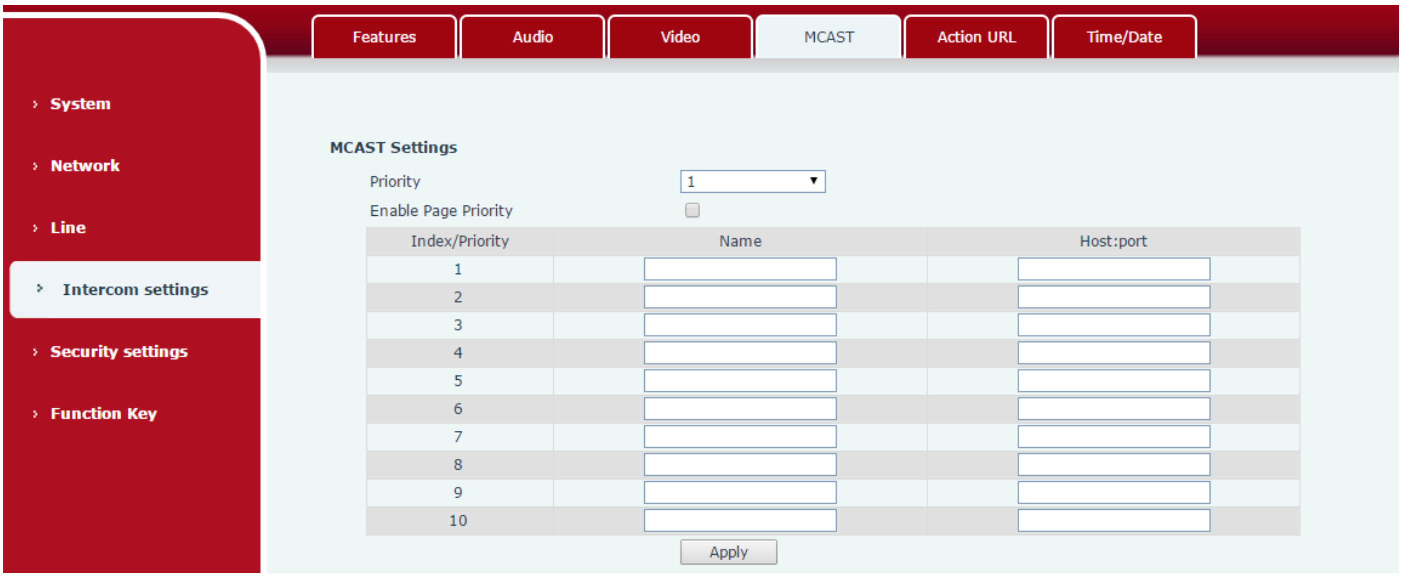
Video Encode(Local Mode)	
Field Name	**Explanation**
Encode Format	Only H.264 encoding format is supported
Resolution	Main stream: support 720P Sub-stream: you can select CIF (352 * 288), D1 (720 * 576)
Frame Rate	The larger the value is, the more coherent the video would be got; not recommend adjusted.
Bitrate Control	CBR: If the code rate (bandwidth) is insufficient, it is preferred. VBR: Image quality is preferred, not recommended.
Quality	Video quality adjustment, the better the quality needs to transfer faster
Bit rate	It is proportional to video file size, not recommend adjusted.
I Frame Interval	The greater the value is, the worse the video quality would be, otherwise the better video quality would be; not recommend adjusted.
Activate	When you selected it, the stream is enabled, otherwise disabled
Preview	copy and paste the main stream or sub-stream Url into the VLC player, or click [Preview] to display the current camera video.

External Mode



Connection mode	Select external, click [Apply], restart the device
IP Camera Settings (External Mode)	
Field Name	**Explanation**
User name	External camera login required account
Password	External camera login password required
Camera type	Select the camera manufacturers
IP address	IP address of the camera, please use the camera matching scan tool to obtain the IP address
Port	Camera port number
RTSP information	Click [Apply], the connection automatically shows the camera does not show the reverse
Preview	Copy and paste the main stream or sub-stream Url into the VLC player, or click [Preview] to display the current camera video

d) MCAST



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

MCAST Settings

Equipment can be set up to monitor up to 10 different multicast addresses, used to receive the multicast RTP stream sent by the multicast address.

Here are the ways to change equipment receiving multicast RTP stream processing mode in the web interface: set the ordinary priority and enable page priority.

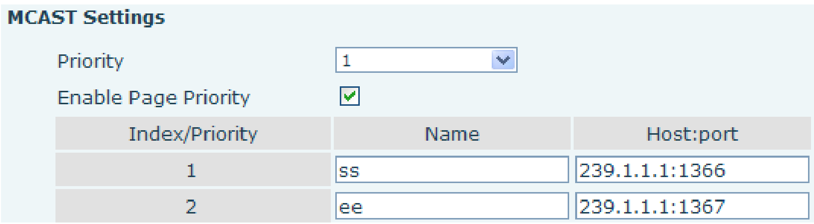
- Priority:

In the drop-down box to choose priority of ordinary calls, if the priority of the incoming streams of multicast RTP, lower precedence than the current common calls, device would automatically ignore the group RTP streams. If the priority of the incoming stream of multicast RTP is higher than the current common calls priority, device would automatically receive the group RTP streams, and keep the current common calls in maintained status. You can also choose to disable the function in the receiving threshold drop-down box, the device would automatically ignore all local network multicast RTP streams.

- The options are as follows:
  - ✧ 1-10: To definite the priority of the common calls, 1 is the top level while 10 is the lowest
  - ✧ Disable: ignore all incoming multicast RTP streams
  - ✧ Enable the page priority:

    Page priority determines the device how to deal with the new receiving multicast RTP streams when it is in multicast session currently. When Page priority switch is enabled, the device would automatically ignore the low priority multicast RTP streams but receive top-level priority multicast RTP streams and keep the current multicast session in maintained status; If it is not enabled, the device would automatically ignore all receiving multicast RTP streams.

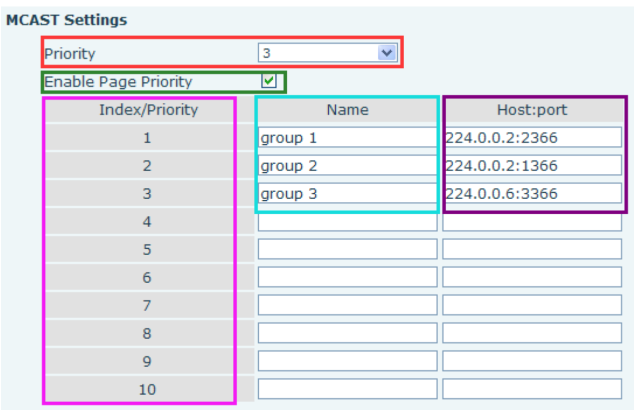
- Web Settings:



The multicast ss priority is higher than that of ee; ss has the highest priority.

Note: when you press the multicast key for multicast session, both multicast sender and receiver would beep.

Listener configuration



- **Blue part (name)**

"Group 1","Group 2" and "Group 3" are your setting monitoring multicast name.The group name would be displayed on the screen when you answer the multicast. If you have not set, the screen would display the IP: port directly.

- **Purple part (host: port)**

It is a set of addresses and ports to listen, separated by a colon.

- **Pink part (index / priority)**

Multicast is a sign of listening, but also the monitoring multicast priority. The smaller number refers to higher priority.

- **Red part (priority)**

It is the general call, non-multicast call priority. The smaller number refers to higher priority. The followings would explain how to use this option:

✧ The purpose of setting monitoring multicast "Group 1" or "Group 2" or "Group 3" is to launch a multicast call.

✧ All equipment has one or more common non multicast communication.

✧ When you set the priority as disabled, any level of multicast would not be answered , multicast call is rejected.

✧ when you set the priority as some value, only the multicast higher than the priority can come in. If you set the priority as 3, group 2 and group 3 would be rejected, for its priority level is equal to 3 and less than 3; multicast 1 priority is set up with 2, higher than ordinary call priority, device can answer the multicast message, at the same time, holding the other call.

- **Green part (Enable Page priority)**

Set whether to open multicast comparison function, multicast priority is pink part number. Following explains how to use:

✧ The purpose of setting monitoring multicast "group 1" or “group 3”is listening "group of 1" or “group 3”multicast call of multicast address.

✧ The device has a path or multi-path multicast calls, such as listening to "multicast information group 2".

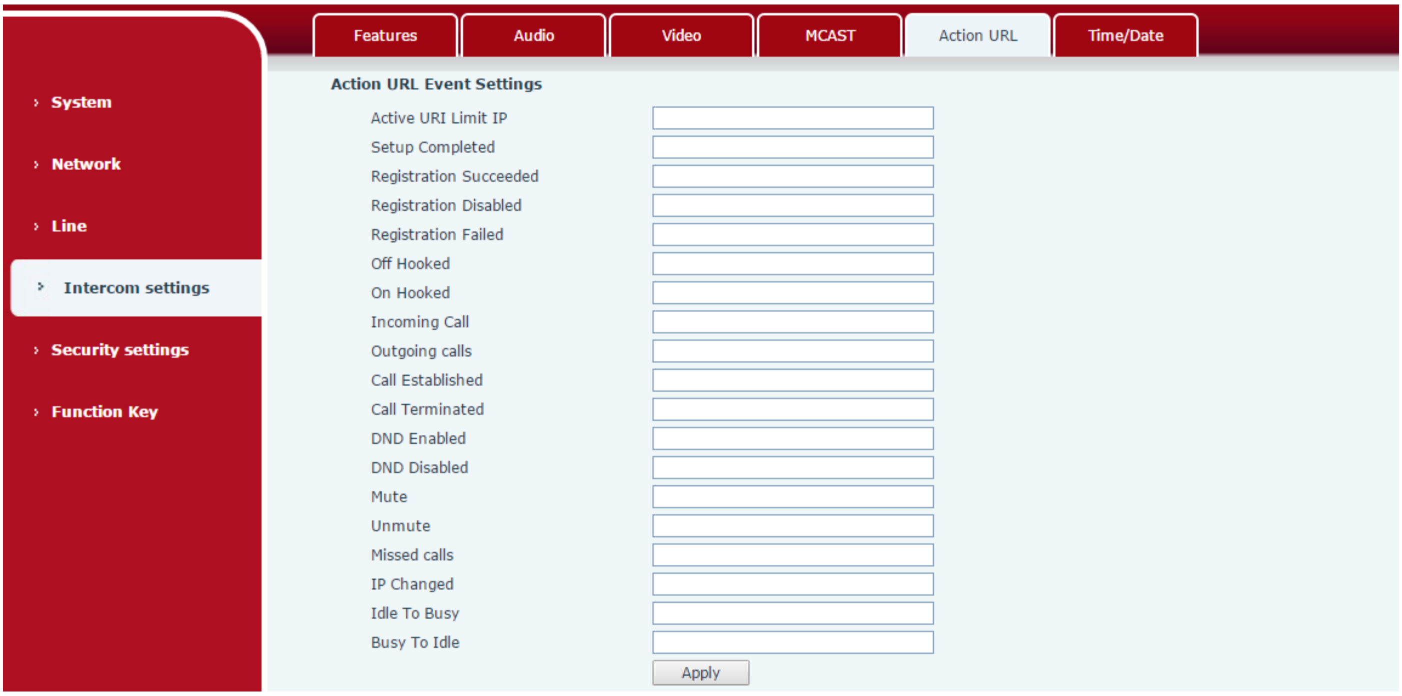
✧ If multicast is a new "group 1", and because the priority of group 1" is 2, higher than the current call priority 3 of “group 2”, so multicast call would come in.

✧ If multicast is a new "group 3", and because the priority of group 3" is 4, lower than the current call priority 3 of “group 2”, the device would listen to the “group 1” and maintain the "group 2".

Multicast service

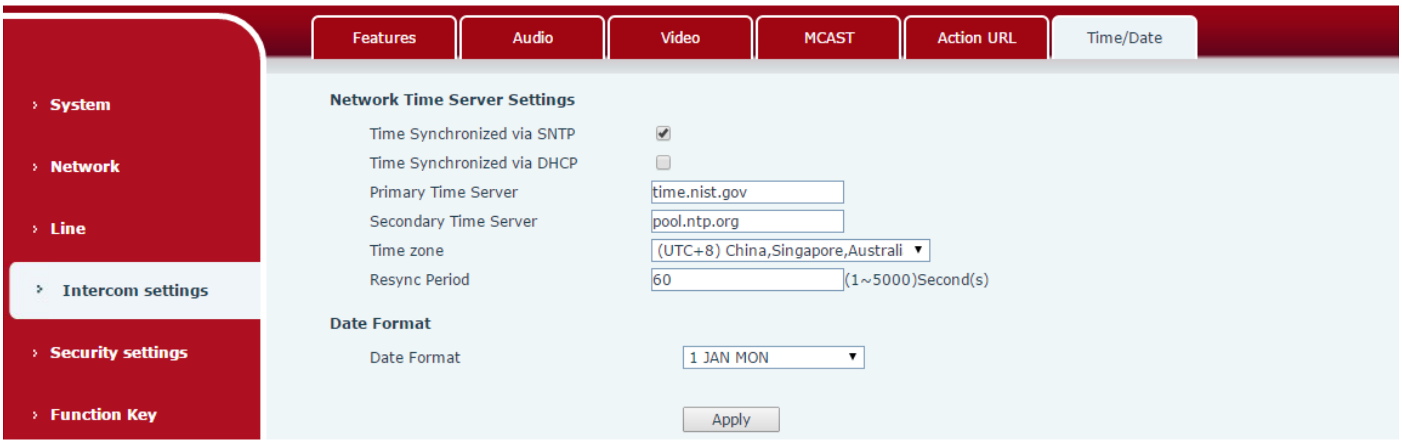
- **Send:** when you configure the item, pressing the corresponding key on the equipment shell, equipment would directly enter the Talking interface; the premise is to ensure no current multicast call and three-way conference, so the multicast can be established.
- **Monitor:** IP port and priority are configured to monitor the device, when the call is initiated by multicast and the call is successful; the device would directly enter the Talking interface.

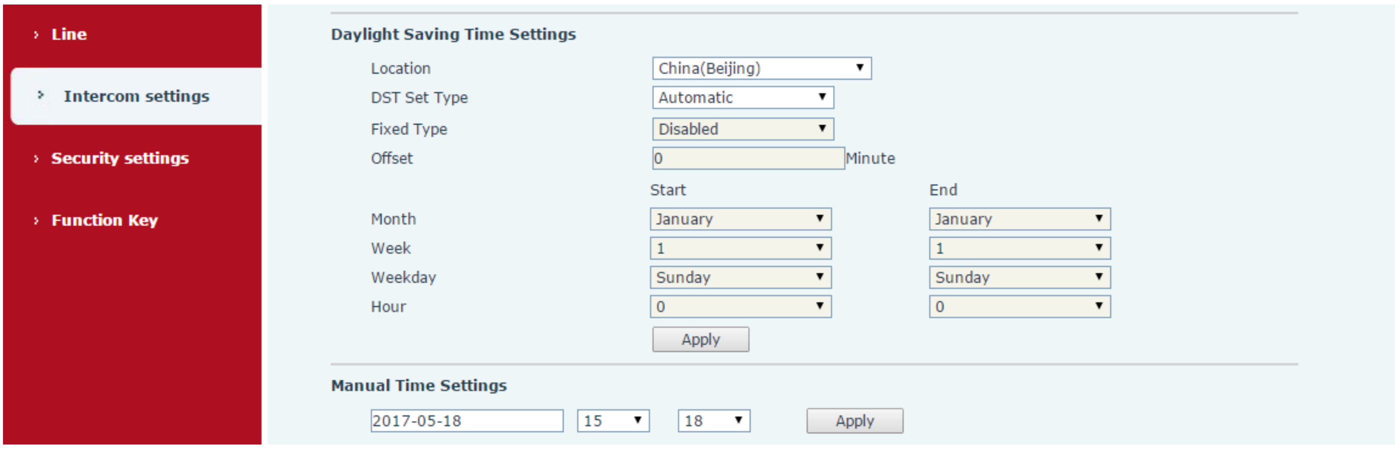
e) Action URL



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

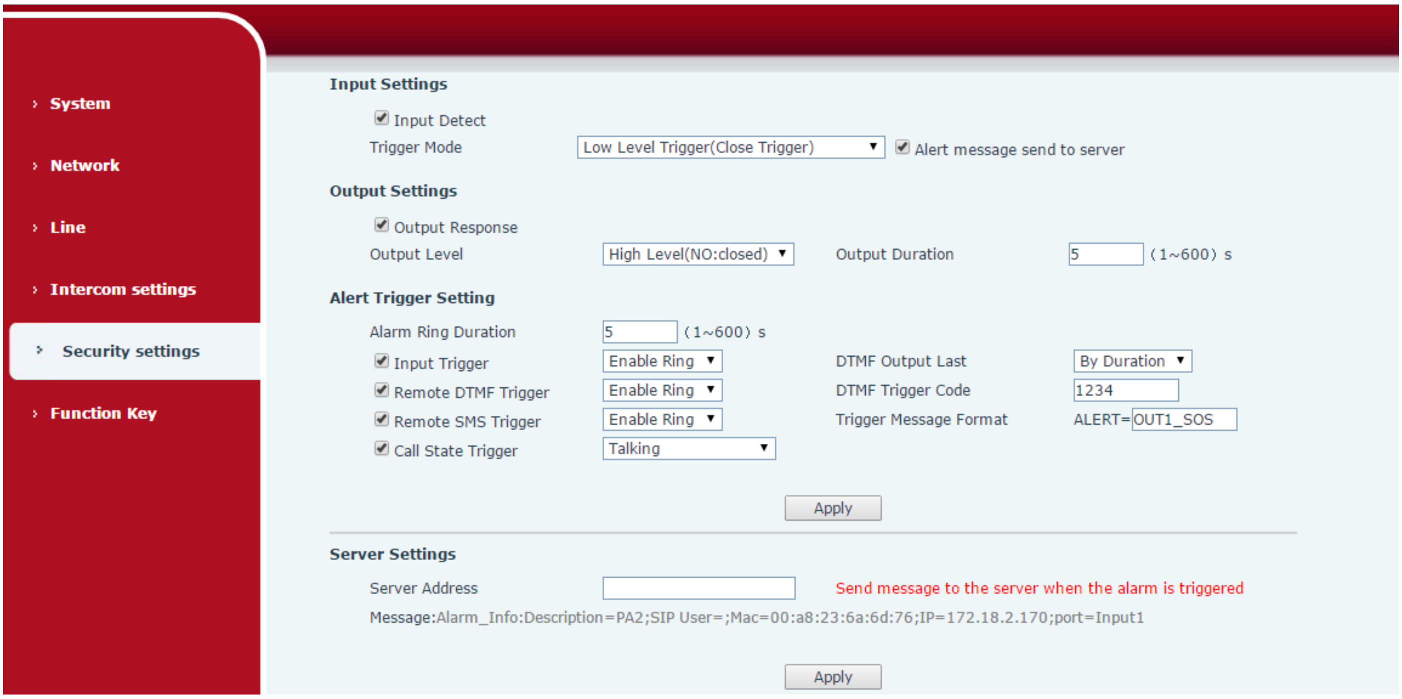
f) Time/Date





Time/Date	
Field Name	**Explanation**
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Primary Time Server	Set primary time server address
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device would 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
Date Format	
12-hour clock	Set the time display in 12-hour mode
Date Format	Select the time/date display format
Daylight Saving Time Settings	
Location	Select the user's time zone according to specific area
DST Set Type	Select automatic DST according to the preset rules of DST, or you can manually input rules
Offset	The DST offset time
Month Start	The DST start month
Week Start	The DST start week
Weekday Start	The DST start weekday
Hour Start	The DST start hour
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Manual Time Settings	
The time might be set manually, needed user to disable SNTP service first.	

(5) Security settings

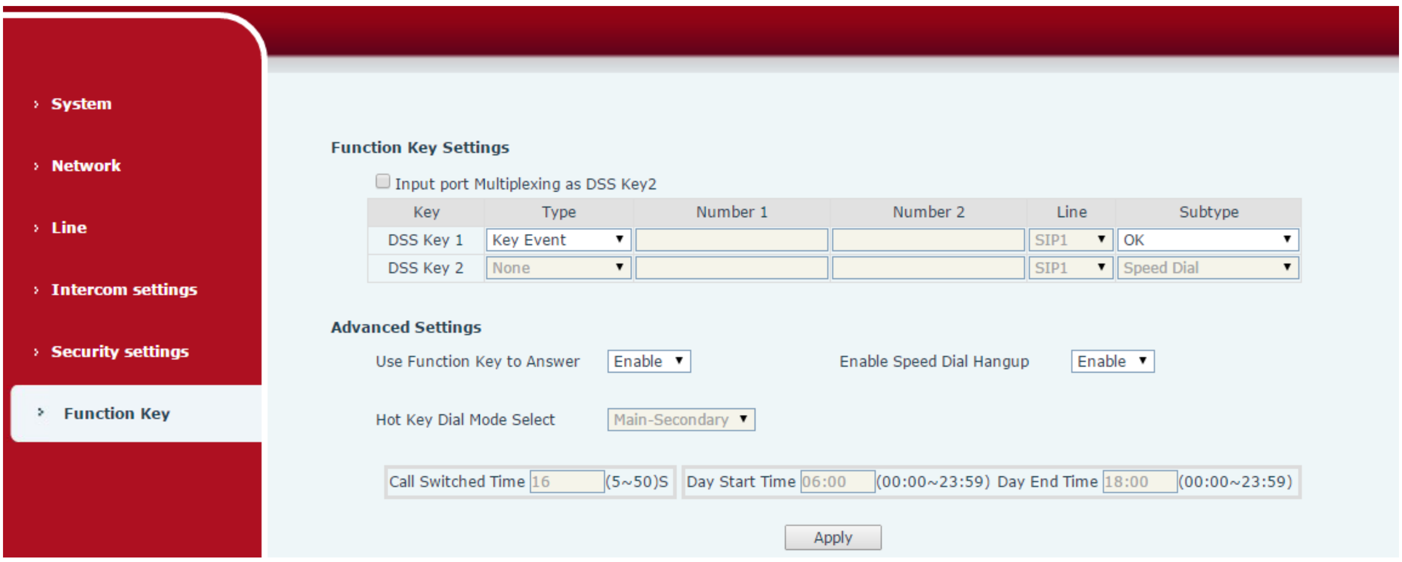


Security Settings	
Field Name	**Field Name**
Input settings	
Input Detect	Enable or disable Input Detect
Trigger Mode	When choosing the low level trigger (Closed Trigger), detect the input port (low level) closed trigger.
	When choosing the high level trigger (Disconnected Trigger), detect the input port (high level) disconnected trigger.
Alert message send to server	Enable or disable Input port send message to server
Output Settings	
Output Response	Enable or disable Output Response
Output Level	When choosing the low level (NO: open), when meet the trigger condition, trigger the NO port disconnected.
	When choosing the high level (NO: closed), when meet the trigger condition, trigger the NO port close.
Output Duration	The port changes the duration. The default is 5 seconds.
Alert Trigger Setting	
Alarm Ring Duration	duration of alarm ring
Input Trigger	When the input port meet to trigger condition, the output port will trigger (The Port

	level time change, By < Output Duration > control). You can choose to enable or disable the ringtone	
DTMF Output Last	By duration	The Port level time change, By < Output Duration > control
	By Calling State	By call state control, after the end of the call, port to return the default state
Remote DTMF Trigger	Received the terminal equipment to send the DTMF password, if correct, which triggers the corresponding output port. You can choose to enable or disable the ringtone	
DTMF Trigger Code	During the call, receive the terminal equipment to send the DTMF password, if correct, which triggers the corresponding output port. The default is 1234.	
Remote SMS Trigger	Enable or disable Remote SMS Trigger. You can choose to enable or disable the ringtone	
Trigger Message Format	In the remote device or server to send instructions to ALERT=[instructions], if correct, which triggers the corresponding output port	
Call State Trigger	The port output continuous time synchronization and trigger state changes, including the trigger conditions. Four models, such as: call trigger output port, will be in a call state to continue to respond) 1, Taking; 2, Taking and ringing; 3, ringing; 4, Call.	
Server Settings		
Server Address	Configure remote response server address (including remote response server address and tamper alarm server address). When the input port is triggered will send a short message to the server, the message format is as follows: Alarm Info: Description=SPA2; SIP User=;Mac=00:a8:34:68:23:d1;IP=172.18.2.243;port=Input1	

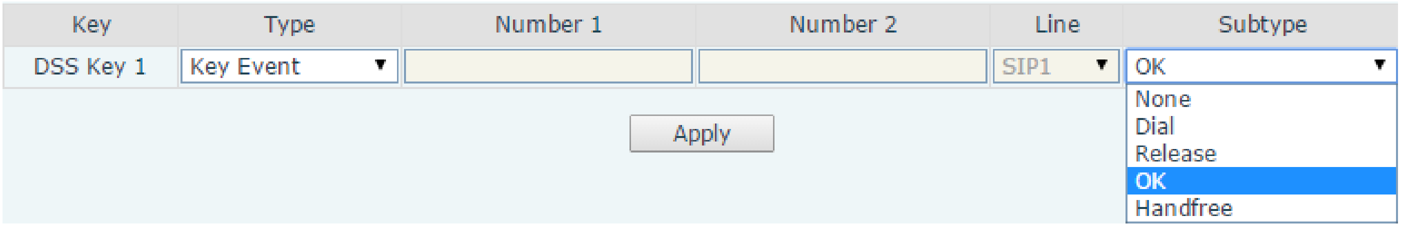
(6) Function Key

a) Function Key Settings



➢ Key Event

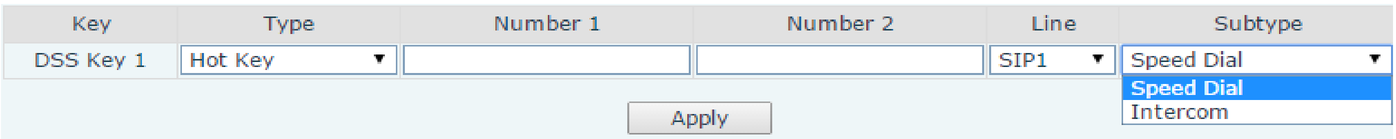
You might set up the key type with the Key Event.



Type	Subtype	Usage
Key Event	None	No responding
	Dial	Dialing function
	Release	Delete password input, cancel dialing input and end call
	OK	identification key
	Handfree	The hand-free key (with hooking dial, hanging up functions)

➢ Hot Key

You might enter the phone number in the input box. When you press the shortcut key, equipment would dial preset telephone number. This button can also be used to set the IP address: you can press the shortcut key to directly make a IP call.

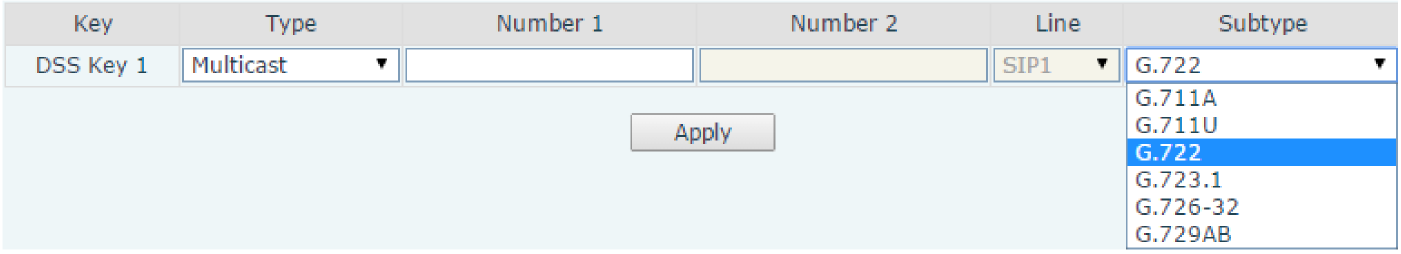


Type	Number	Line	Subtype	Usage
Hot Key	Fill the called party's SIP account or IP address	The SIP account corresponding lines	Speed Dial	Using Speed Dial mode together with Enable Speed Dial Hangup [Enable], can define whether this call is allowed to be hung up by re-pressing the speed dial key.
			Intercom	In Intercom mode, if the caller's IP phone supports Intercom feature, the device can automatically answer the Intercom calls

➢ Multicast

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

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



Type	Number	Subtype	Usage
Multicast	Set the host IP address and port number; they must be separated by a colon	G.711A	Narrowband speech coding (4Khz)
		G.711U	
		G.722	Wideband speech coding (7Khz)
		G.723.1	Narrowband speech coding (4Khz)
		G.726-32	
		G.729AB	

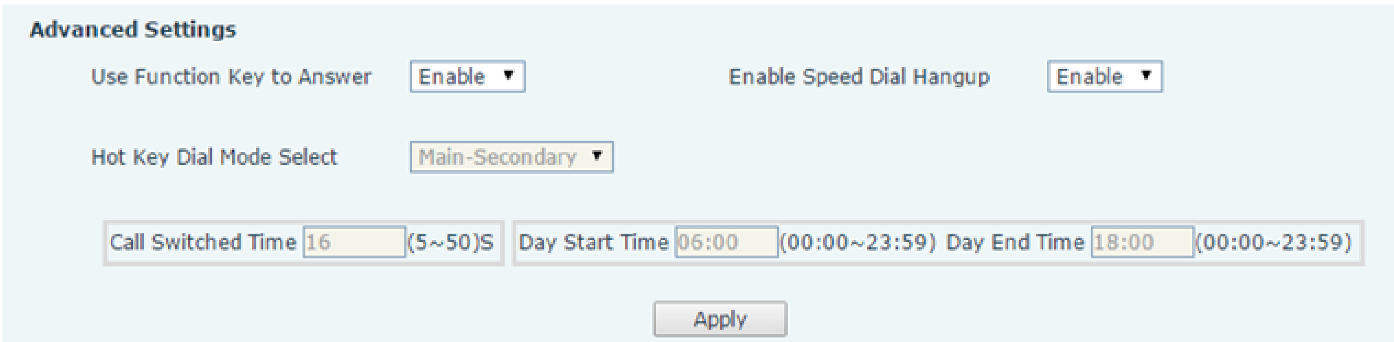
- ◆ operation mechanism

You can define the Function Key configuration with multicast address, port and used codec. The device can configure via WEB to monitor the multicast address and port. When the device make a multicast, all devices monitoring the address can receive the multicast data.

- ◆ calling configuration

If the device is in calls, or it is three-way conference, or initiated multicast communication, the device would not be able to launch a new multicast call.

b) Advanced Settings



Advanced Settings	
Field Name	**Field Name**
Input Port Multiplexing 2	Enable or disable input port reuse for DSS key 2.
Use Function Key to Answer	Enable or disable DSS Key answer
Enable Speed Dial key Hang up	Enable or disable the DSS Key to hang up
Hot Key Dial Mode Select	Number 1 Transfer Call Number 2 Mode Select. <Primary /Secondary>mode allow system to call primary extension first, if there is no answer, system would cancel the call and then call secondary extension automatically. <Day/Night>mode allow system to check the calling time is belong to day time or night time, and then system decides to call the number 1 or number

	2 automatically. Users just press speed dial key once.
Call Switched Time	Set number 1 to transfer call number 2 time, default 16 seconds
Day Start Time	The start time of the day when you select<Day/Night>mode.
Day End Time	The end time of the day when you select <Day/Night>mode.

V. Appendix

1. Technical parameters

Communication protocol		SIP 2.0(RFC-3261)
Main chipset		Broadcom
Speech flow	**Protocols**	RTP
	Decoding	G.729、G.723、G.711、G.722、G.726
WAN		10/100BASE-TX s Auto-MDIX, RJ-45
MIC(Optional)		electret condenser microphone Sensitivity: -38dB, bias voltage 2.2V
Speaker(Optional)		4Ω / 3W, internal magnetic horn, diameter: 57mm
Camera(Optional)		1/4 "color CMOS, 1 megapixel, wide angle
Power supply mode		12～24V 2A DC or PoE
PoE		PoE 802.3af (Class 3 - 6.49~12.95W)
Cables		CAT5 or better
Shell Material		Aluminum alloy
Working temperature		-20°C to 60°C
Storage temperature		-40°C to 70°C
Installation way		Embedded or desktop
External size		113x83x28mm
Package size		138x108x77mm
Equipment weight		250g

2. Basic Functions

- 2 SIP lines
- PoE Enabled
- External power supply
- Supports two lines RTSP

- Voice broadcast IP address
- button or remote volume adjustment
- switch signal input and output
- Talkback recording output
- Default auto answer
- Dynamic multicast function
- Support for Function key interface
- Support monophonic active speakers