

# i31S SIP Video DoorPhone User Manual V1.0







Document VER	Firmware VER	Explanation	Time
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# **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 will 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.



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### I Product introduction

i31S voice access is a full digital network door phone, with its core part adopts mature VoIP solution (Broadcom chip), stable and reliable performance, hands-free adopting digital full-duplex mode, voice loud and clear, generous appearance, solid durable, easy for installation, comfortable keypad and low power consumption.

i31S voice access supports entrance guard control, voice intercom, ID card and keypad remote to open the door.

### 1. Appearance of the product





# 2. Description

Buttons and icons	Description	Function	
	Numeric keyboard	Input password to open the door or to call.	
	programmable keys	Can be set to a variety of functions, in order to meet the needs of different occasions	
CARD OSS	induction zone	RFID induction area	
	Camera	Video signal acquisition and transmission	



h	Lock Status	Door unlocking: On
		Door locking: Off
		Standby: Off
*{	Call status	Call Holding: Blink with 1s
		Calls: On
Δ	Ring status	Standby: Off
		Ringing: On
		Network error: Blink with 1s
all	Network/SIP	Network running: Off
	Registration	Registration failed: Blink with 3s
		Registration succeeded: On

# **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, electric lock control line connection and the boot-up is normal. (Check the network state of light)

### 1) Power, Electric Lock, Indoor switch port

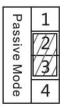
Voice access the power supply ways: 12v/DC or POE.

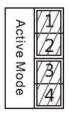
			CN7			
1	2	3	4	5	6	7
+12V	VSS	NC	СОМ	NO	S_IN	S_OUT
12V 1A/DC		Elec	tric-lock sw	vitch	Indoor	switch



# 2) Driving mode of electric-lock(Default in Passive mode)







Jumper in passive mode

Jumper in active mode



[Note] When the device is in active mode, it can drive 12V/650mA switch output maximum, to which a standard electric-lock or another compatible electrical appliance can be connected.

- When using the active mode, it is 12V DC in output.
- When using the passive mode, output is short control (normally open mode or normally close mode).

### 3) Wiring instructions

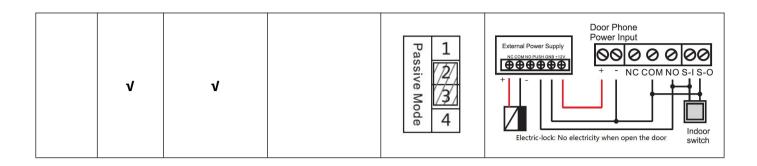
NO: Normally Open Contact.

• COM: Common Contact.

NC: Normally Close Contact.

<b>Driving Mode</b>		Electric lock			
Active	Passive	No electricity when open	When the power to open	Jumper port	Connections
V		٧		Active Mode	12V OO
V			V	Active Mode	12V OO
	٧	V		Passive Mode  4	Power Supply 12V/2A + - NC COM NO S-I S-O Indoor switch
	٧		V	Passive Mode  4	Power Supply 12V/2A + - NC COM NO S-I S-O Indoor switch



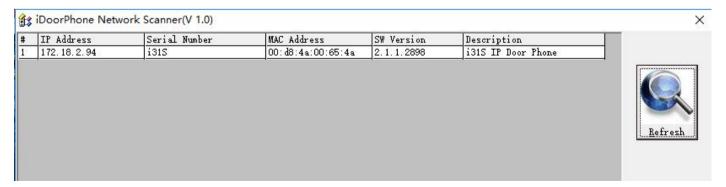


### 2. Quick Setting

The product provides a complete function and parameter setting. Users may need to have the network and SIP protocol knowledge to understand the meaning represented by all parameters. 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 compulsory setting options in this section to let users know how to operate without understanding such complex SIP protocols.

In prior to this step, please make sure your broadband Internet online can be normal operated, and complete the connection of the network hardware. The product factory default network mode is DHCP. Thus, only connect equipment with DHCP network environment that network can be automatically connected.

- Press and hold "#" key for 3 seconds and the door phone will report the IP address by voice, or use the "iDoorPhoneNetworkScanner.exe" software to find the IP address of the device.
- Note: when power on, 30s waiting is needed for device running.
- Log on to the WEB device configuration.
- In a Line page configuration service account, user name, parameters that are required for server address register.
- You can set DSS key in the Function key page.
- You can set Door Phone parameters in the Webpage (EGS Setting-> Features).





### **III** Basic operation

#### 1. Answer a call

When a call comes in, the device will answer automatically. If you cancel auto answer feature and set auto answer time, you will hear the bell ring at the set time and the device will auto answer after a timeout.

#### 2. Call

Configure shortcut key as hot key and setup a number, then press shortcut key can call the configured number.

#### 3. End call

Enable Release key hang up to end call.

### 4. Open the door operation

Through the following seven ways to open the door:

- 1) Input password on the keyboard to open the door.
- 2) Access to call the owner and the owner enter the remote password to open the door.
- 3) Owner/other equipment call the access control and enter the access code to open the door. (access code should be included in the list of access configuration, and enable for remote calls to open the door)
- 4) Swipe the RFID cards to open the door.
- 5) By means of indoor switch to open the door.
- 6) Private access code to open the door.

Enable for local authentication, and set private access code. Input the access code directly under standby mode to open the door. In this way, the door log will record corresponding card number and user name.

7) Active URL control command to open the door.

URL is "http://user:pwd@host/cgi-bin/ConfigManApp.com?key=F\_LOCK&code=openCode"

- a. User and pwd is Web the user name and password.
- b. "openCode" is the remote control code to open the door.

Example: "http://admin:admin@172.18.3.25/cgi-bin/ConfigManApp.com?key=\*"

If access code is input correctly, the device will play sirens sound to prompt access control and the remote user, while input error by low-frequency short chirp.

Password input successfully followed by high-frequency sirens sound, while input error is followed by



high-frequency short chirp.

When door has been opened, the device will play sirens sound to prompt.

# **IV Page settings**

### 1. Browser configuration

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

Enter the user name and password 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 access can browse and set all configuration parameters, while a user with general level can set all configuration parameters except server parameters for SIP.

- Default user with general level: The default is not set, are free to add.
- Default user with root level:

User name: adminPassword: admin



# 3. Configuration via WEB

# (1) System

### a) Information



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



### b) Account

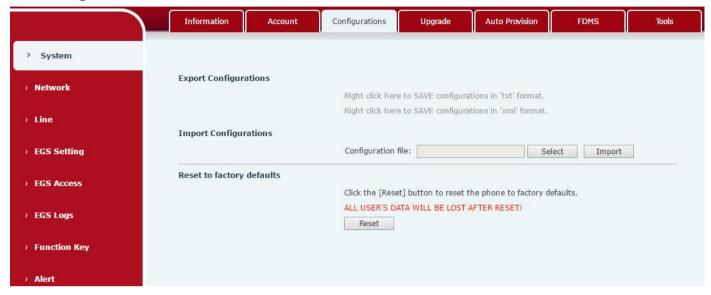
Through this page, user can add or remove users depends on their needs and can modify existing user permission.



Account	Account		
Field Name	Explanation		
Change Web Au	thentication Password		
You Can modify	the login password to the account		
Add New User			
You can add new user			
User Accounts			
Show the existing user information			

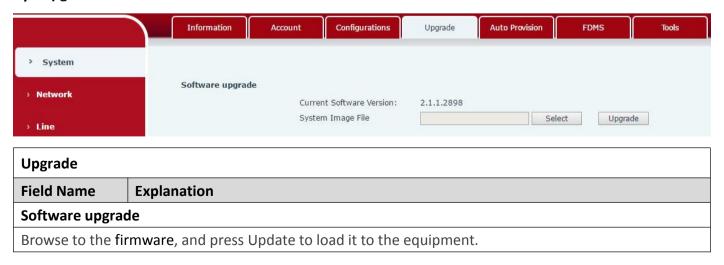


### c) Configurations



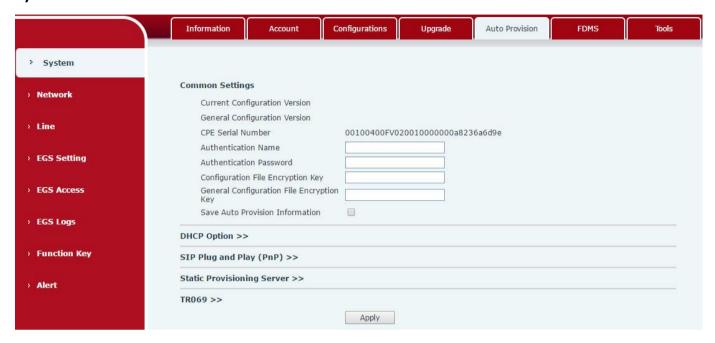
Configurations	
Field Name	Explanation
Export	Save the equipment configuration to a txt or xml file. Please note to Right click on
Configurations	the choice and then choose "Save Link As."
Import	Drawes to the config file and proce Undate to lead it to the equipment
Configurations	Browse to the config file, and press Update to load it to the equipment.
Reset to factory	This will receive factors default and remain all configuration information
defaults	This will restore factory default and remove all configuration information.

#### d) Upgrade





### e) Auto Provision



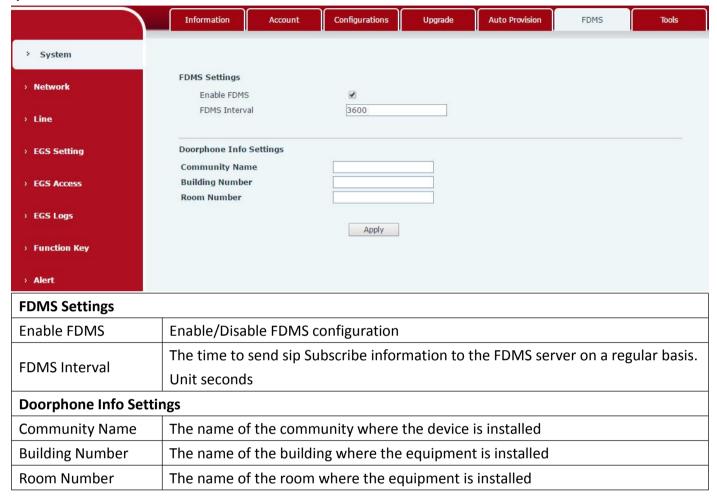
<b>Auto Provision</b>	
Field Name	Explanation
<b>Common Settings</b>	
Current Configuration Version	Show the current config file's version. If the version of configuration downloaded is higher than this, the configuration will be upgraded. If the endpoints confirm the configuration by the Digest method, the configuration will not be upgraded unless it differs from the current configuration
General Configuration Version	Show the common config file's version. If the configuration downloaded and this configuration is the same, the auto provision will stop. If the endpoints confirm the configuration by the Digest method, the configuration will not be upgraded unless it differs from the current configuration.
CPE Serial Number	Serial number of the equipment
Authentication Name	Username for configuration server. Used for FTP/HTTP/HTTPS. If this is blank the phone will use anonymous
Authentication Password	Password for configuration server. Used for FTP/HTTP/HTTPS.
Configuration File Encryption Key	Encryption key for the configuration file
General Configuration File Encryption Key	Encryption key for common configuration file



Save Auto Provision	Save the auto provision username and password in the phone until the server url		
Information	changes		
DHCP Option			
Option Value	The equipment supports configuration from Option 43, Option 66, or a Custom		
Option value	DHCP option. It may also be disabled.		
Custom Option	Custom option number. Must be from 128 to 254.		
Value	Custom option number. Must be from 128 to 254.		
SIP Plug and Play (Pnf	P)		
	If this is enabled, the equipment will send SIP SUBSCRIBE messages to a multicast		
Enable SIP PnP	address when it boots up. Any SIP server understanding that message will reply		
Eliable SIP PIIP	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	Dep Transfer protectal LIDD or TCD		
Protocol	PnP Transfer protocol – UDP or TCP		
Update Interval	Interval time for querying PnP server. Default is 1 hour.		
Static Provisioning Se	rver		
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be an IP		
Server Address	address or Domain name with subdirectory.		
Configuration File	Specify configuration file name. The equipment will use its MAC ID as the config		
Name	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.		
	1. Disable – no update		
Update Mode	2. Update after reboot – update only after reboot.		
	3. Update at time interval – update at periodic update interval		
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 for ACS.		
ACS Password	ACS Password.		
TR069 Auto Login	Enable/Disable TR069 Auto Login.		
INFORM Sending	Time between transmissions of "Inform" Unit is seconds.		
Period			



#### f)FDMS



### f) 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 will 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 the normal but significant condition.

Level 6: Informational; It is the 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 Continue	

#### **Network Packets Capture**

Capture a packet stream from the equipment. This is normally used to troubleshoot problems.

#### **Reboot Phone**

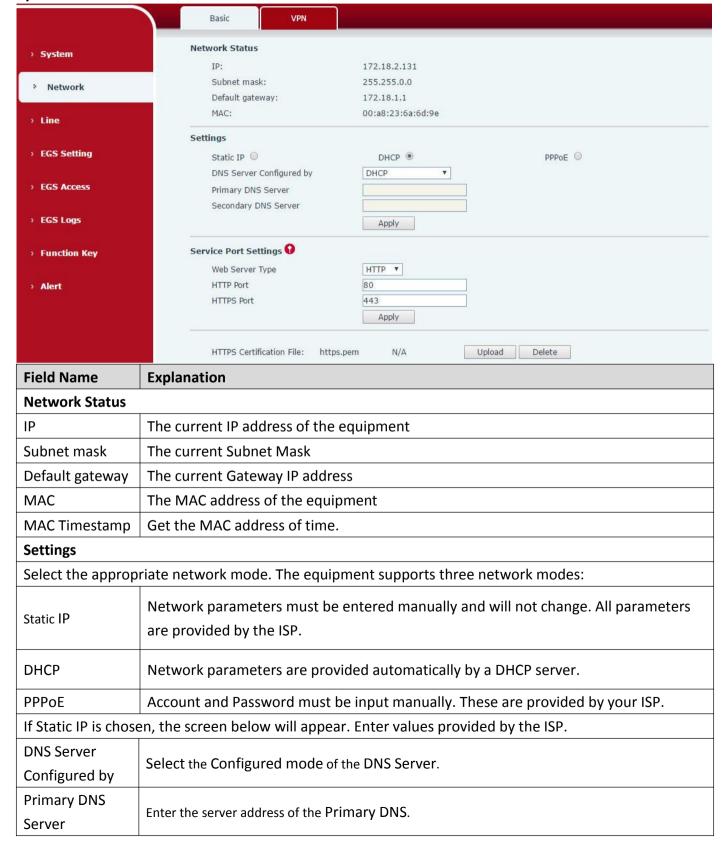
Some configuration modifications require a reboot to become effective. Clicking the Reboot button will lead to reboot immediately.

Note: Be sure to save the configuration before rebooting.



### (2) Network

#### a) Basic





Secondary DNS Server	Enter the server address of the Secondary DNS.
-------------------------	--

After entering the new settings, click the APPLY button. The equipment will 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.

Service Port Settings		
Web Server Type	Specify Web Server Type – HTTP or HTTPS	
	Port for web browser access. Default value is 80. To enhance security, change this	
HTTP Port	from the default. Setting this port to 0 will disable HTTP access.	
	Example: The IP address is 192.168.1.70 and the port value is 8090, the accessing	
	address is http://192.168.1.70:8090.	
	Port for HTTPS access. Before using https, an https authentication certification must	
HTTPS Port	be downloaded into the equipment.	
	Default value is 443. To enhance security, change this from the default.	

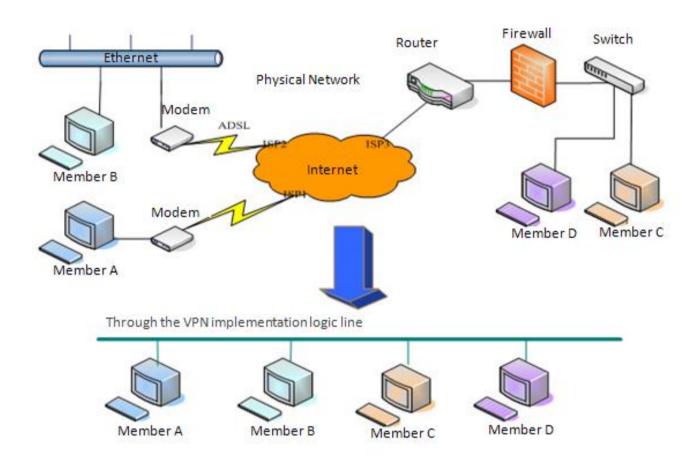
#### Note:

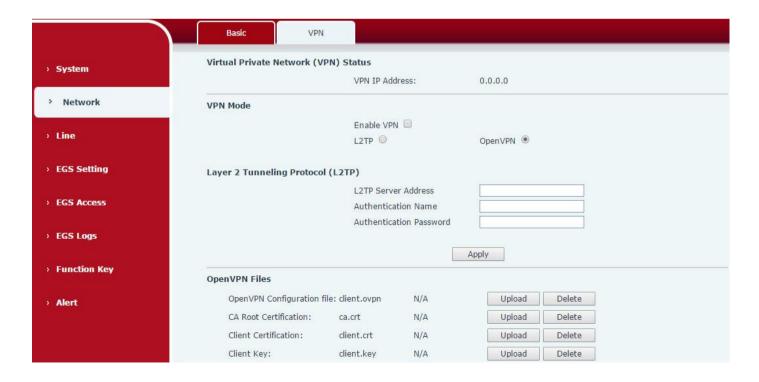
- 1) Any changes made on this page require a reboot to become active.
- 2) It is suggested that changes to HTTP Port be values greater than 1024. Values less than 1024 are reserved.
- 3) If the HTTP port is set to 0, HTTP service will be disabled.

#### 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.







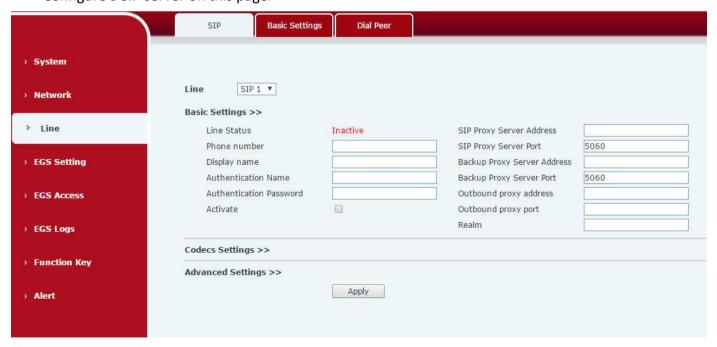


Field Name	Explanation	
VPN IP Address	Shows the current VPN IP address.	
VPN Mode		
Enable VPN	Enable/Disable VPN.	
L2TP	Select Layer 2 Tunneling Protocol	
Onon\/DN	Select OpenVPN Protocol. (Only one protocol may be activated. After the selection	
OpenVPN	is made, the configuration should be saved and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

# (3) Line

### a) SIP

Configure a SIP server on this page.





odecs Settings >>			
Disabled Codecs		Enabled Codecs	
	→ ←	G.722 G.711U G.711A G.729AB	1
dvanced Settings >>			
Subscribe For Voice Message Voice Message Number Voice Message Subscribe Period	3600 Second(s)		
Enable DND  Blocking Anonymous Call  Use 182 Response for Call waiting  Anonymous Call Standard  Dial Without Registered  Click To Talk  User Agent  Response Single Codec	None v	Ring Type Conference Type Server Conference Number Transfer Timeout Enable Long Contact Enable Use Inactive Hold Use Quote in Display Name	Default ▼  Local ▼  O Second(s)
Use Feature Code Enable DND Enable Blocking Anonymous Call		DND Disabled Disable Blocking Anonymous Call	
Specific Server Type Registration Expiration Use VPN Use STUN Convert URI DTMF Type DTMF SIP INFO Mode Transportation Protocol Local Port SIP Version Caller ID Header Enable Strict Proxy Enable user=phone Enable SCA Enable BLF List	COMMON ▼  60 Second(s)   AUTO ▼  Send */# ▼  UDP ▼  5060  RFC3261 ▼  PAI-RPID-I ▼	Enable DNS SRV Keep Alive Type Keep Alive Interval Sync Clock Time Enable Session Timer Session Timeout Enable Rport Enable PRACK Auto Change Port Keep Authentication Auto TCP Enable Feature Sync Enable GRUU BLF Server BLF List Number	UDP ▼ 30 Second(s)  0 Second(s)  1 Second(s)
SIP Encryption SIP Encryption Key	Apply	RTP Encryption RTP Encryption Key	



SIP	
Field Name	Explanation
Basic Settings (Choose the	e SIP line to configured)
Line Status	Display the current line status at page loading. To get the up to date line status,
Line Status	user has to 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
Outhound provided dross	Enter the IP or FQDN address of outbound proxy server provided by the service
Outbound proxy address	provider
Outbound proxy port	Enter the outbound proxy port, default is 5060
Realm	Enter the SIP domain if requested by the service provider
<b>Codecs Settings</b>	
Set the priority and availa	bility of the codecs by adding or remove them from the list.
Advanced Settings	
Subscribe For Voice	Enable the device to subscribe a voice message waiting notification, if enabled,
Message	the device will receive notification from the server if there is voice message
iviessage	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 DND	Enable Do-not-disturb, any incoming call to this line will 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
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.



Response Single Codec	If setting enabled, the device will use single codec in response to an incoming call request
Ring Type	Set the ring tone type for the line
0 //- ·	Set the type of call conference, Local=set up call conference by the device itself,
Conference Type	maximum supports two remote parties, Server=set up call conference by dialing
7,	to a conference room on the server
Server Conference Number	Set the conference room number when conference type is set to be Server
Transfer Timeout	Set the timeout of call transfer process
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Use Quote in Display Name	Whether to add quote in display name
	When this setting is enabled, the features in this section will not be handled by
Use Feature Code	the device itself but by the server instead. In order to control the enabling of the
Ose reature code	features, the device will 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 interval
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
Local Port	Set the Local Port
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 will 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
	Set the line to use DNS SRV which will resolve the FQDN in proxy server into a
Enable DNS SRV	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

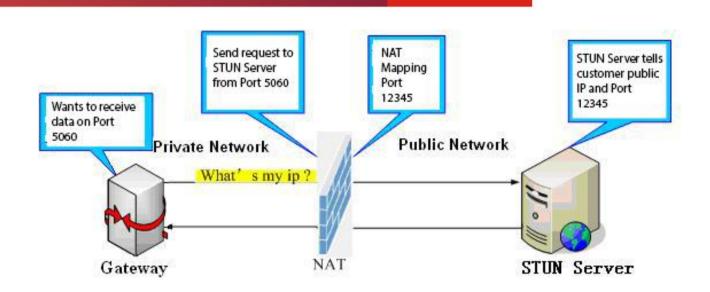


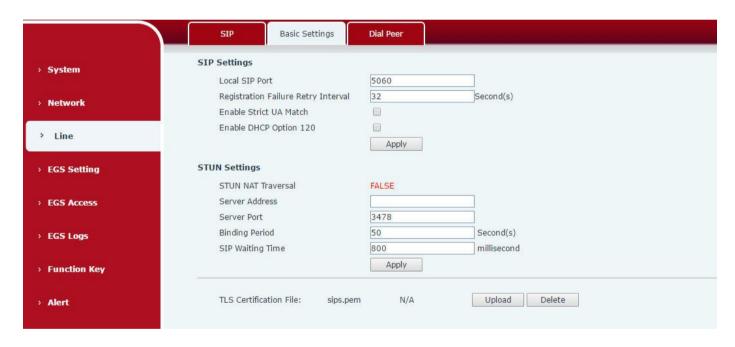
	Set the line to enable call ending by session timer refreshment. The call session
Enable Session Timer	will be ended if there is not new session timer event update received after the
	timeout period
Session Timeout	Set the session timer timeout period
Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
Enable DNS SRV	Set the line to use DNS SRV which will resolve the FQDN in proxy server into a
Eliable Div2 2KA	service list
Auto Change Port	Enable/Disable Auto Change Port
Keep Authentication	Keep the authentication parameters from previous authentication
Auto TCD	Using TCP protocol to guarantee usability of transport for SIP messages above
Auto TCP	1500 bytes
Enable Feature Sync	Feature Sycn with server
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
	The registered server will receive the subscription package from ordinary
BLF Server	application of BLF phone.
BLI Server	Please enter the BLF server, if the sever does not support subscription package,
	the registered server and subscription server will be separated.
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists
bei eist Number	are supported.
SIP Encryption	Enable SIP encryption such that SIP transmission will be encrypted
SIP Encryption Key	Set the pass phrase for SIP encryption
RTP Encryption	Enable RTP encryption such that RTP transmission will be encrypted
RTP Encryption Key	Set the pass phrase for RTP encryption

### 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	Set the retry interval of SID DECISTRATION when registration foiled	
Retry Interval	Set the retry interval of SIP REGISTRATION when registration failed.	
Enable Strict UA	Enable or disable Strict UA Match	
Match	Enable of disable Strict OA Match	
STUN Settings		
Server Address	STUN Server IP address	
Server Port	STUN Server Port – Default is 3478.	



SIP Waiting Time	Waiting time for SIP. This will vary depending on the network.
CID Mailing Time	Matter than for CID This ill an advantage on the college
Billullig Periou	mapping active.
Binding Period	STUN blinding period – STUN packets are sent at this interval to keep the NAT

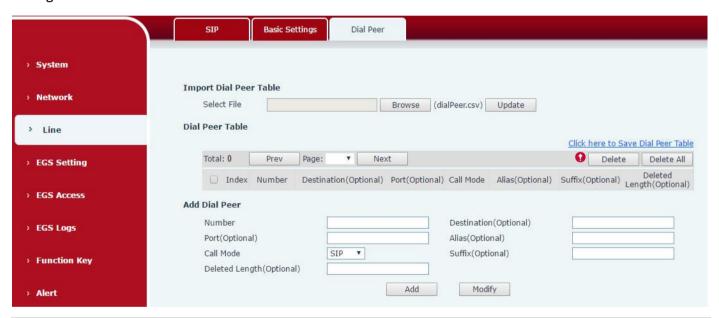
#### **TLS Certification File**

Upload or delete the TLS certification file used for encrypted SIP transmission.

Note: the SIP STUN is used to achieve the SIP penetration of NAT, is the realization of a service, when the equipment configuration of the STUN server IP and port (usually the default is 3478), and select the Use Stun SIP server, the use of NAT equipment to achieve penetration.

#### C) Dial Peer

Configure the Dial Peer to make the device call more flexible.



Import Dial peer Table		
Field Name	Explanation	
Select File	Select an existing dialing rule file. The file type must be a .CSV	
Add Dial Peer		
	In order to add an outgoing call number, the outgoing call number can be divided	
	into two types: one is the exact match, and after the exact match, if the number is	
	exactly the same as the user dialing the called number, the device will use the IP	
	address of this number mapping or (This is the area code prefix function of the	
Number	PSTN). If the number matches the N-bit (prefix number length) of the called	
	number, the device uses the IP address or configuration mapped to this number.	
	Make a call. Configuration prefix matching needs to be followed by a prefix	
	number to match the exact match number; the longest support of 30 bits; also	
	supports the use of x format and range of numbers.	



	Configure the destination address and, if configured as a point-to-point call, write
Destination	the peer IP address directly. Can also be set to domain name, by the device DNS
	server to resolve the specific IP address. If it is not configured, the IP address is
	0.0.0.0. This is an optional configuration item
Port	Configure the signaling port of the other party. This is an optional configuration
	item. The default is 5060
Alias	Configure aliases, this is an optional item: the replacement number used when
	the prefix is prefixed, and no alias when configured

Note: aliases are divided into four types and must be combined with the replacement length:

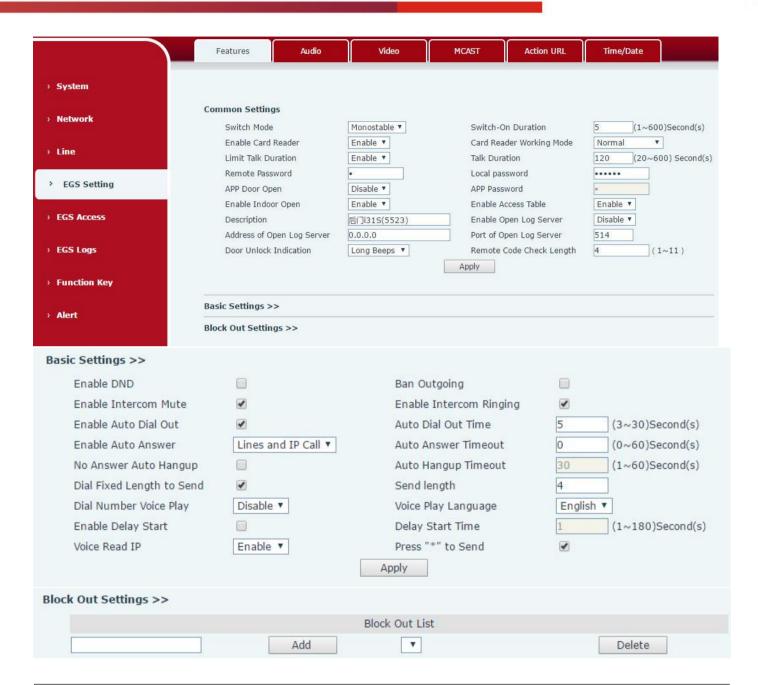
- 1) add: xxx, add xxx before the number. This can help users save dialing length;
- 2) all: xxx, all replaced by xxx; can achieve speed dial, such as user configuration dial-up 1, then by configuring all: number to change the actual call out the number;
- 3) del, delete the number before the n bit, n by the replacement length set;
- 4) rep: xxx, the number n before the number is replaced by xxx, n is set by the replacement length. For example, if the user wants to dial the PSTN (010-62281493) through the floor service provided by the VoIP operator, and the actual call should be 010-62281493, then we can configure the called number 9T, then rep: 010, and then delete the length Set to 1. Then all users call the 9 at the beginning of the phone will be replaced with 010 + number sent. To facilitate the user to call the habit of thinking mode;

Call Mode	Configuration selection of different signaling protocols, SIP;
Suffix	Configure the suffix, this is optional configuration items: that is, after the dial-up
	number to add this suffix, no configuration shows no suffix;
Deleted Length	Configure the replacement / delete length, the number entered by the user is
	replaced / deleted by this length; this is an optional configuration item;

### (4) EGS Setting

### a) Features





Features	
Field Name	Explanation
Common Settings	
Switch Mode	Monostable: there is only one fixed action status for door unlocking.
	Bistable: there are two actions and statuses, door unlocking and door locking.
	Each action might be triggered and changed to the other status. After
	changed, the status would be kept.
	Initial Value is Monostable
Switch-On Duration	Door unlocking time for Monostable mode only. If the time is up, the door
	would be locked automatically. Initial Value is 5 seconds.



Enable Card Reader	Enable or disable card reader for RFID cards.
	Set ID card stats:
	Normal: This is the work mode, after the slot card can to open the door.
Card Reader Working	Card Issuing: This is the issuing mode, after the slot card can to add ID cards.
Mode	Card Revoking: This is the revoking mode, after the slot card can to delete ID
	cards.
Limit Talk Duration	If enabled, calls would be forced ended after talking time is up.
Talk Duration	The call will be ended automatically when time up. Initial Value is 120 seconds
Remote Password	Remote door unlocking password. Initial Value is "*".
Local recovered	Local door unlocking password via keypad, the default password length is 4.
Local password	Initial Value is "6789".
APP Door Open	Enable or disable the APP Door Open
APP password	APP door unlocking password. Initial Value is "*".
Enable Indoor Open	Enable or disable to use indoor switch to unlock the door.
	Enable Access Table: enter <access code=""> for opening door during calls.</access>
Enable Access Table	Disable Access Table: enter <remote password=""> for opening door during calls.</remote>
	Default Enable.
Description	Device description displayed on IP scanning tool software. Initial Value is "i31S
Description	IP Door Phone".
Enable Open Log	Enable or disable to connect with log comer
Server	Enable or disable to connect with log server
Address of Open Log	Log conver address(ID or domain name)
Server	Log server address(IP or domain name)
Port of Open Log	Log conver port (0.65525) Initial Value is 514
Server	Log server port (0-65535), Initial Value is 514.
Door Unlock Indication	Indication tone for door unlocked. There are 3 type of tone: silent/short
Door Officer malcation	beeps/long beeps.
Remote Code Check	The remote access code length would be restricted with it. If the input access
	code length is matched with it, system would check it immediately. Initial
Length	Value is 4.
Basic Settings	
Enable DND	DND might be disabled phone for all SIP lines, or line for SIP individually. But
Ellable DND	the outgoing calls will not be affected
Ban Outgoing	If enabled, no outgoing calls can be made.
Enable Intercom Mute	If enabled, mutes incoming calls during an intercom call.
Enable Intercom	If anabled place intercoming tone to place to an intercoming
Ringing	If enabled, plays intercom ring tone to alert to an intercom call.



Enable Auto Dial Out	Enable Auto Dial Out
Auto Dial Out Time	Set Auto Dial Out Time
Enable Auto Answer	Enable Auto Answer function
Auto Answer Timeout	Set Auto Answer Timeout
No Answer Auto Hangup	Enable automatically hang up when no answer
Auto Hangup Timeout	Configuration in a set time, automatically hang up when no answer
Dial Fixed Length to Send	Enable or disable dial fixed length to send.
Send length	The number will be sent to the server after the specified numbers of digits are dialed.
Dial Number Voice Play	Configuration Open / Close Dial Number Voice Play
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
Voice Read IP	Enable or disable voice broadcast IP address
Press "*" to Send	Enable or disable the Press "*" to Send, Initial Value is enable
Block Out Settings	

#### Block Out Settlings

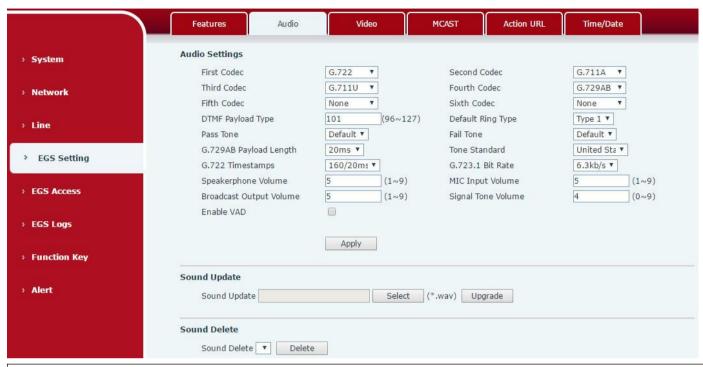
Add or delete blocked numbers – enter the prefix of numbers which should not be dialed by the phone. For example, if 001 is entered, the phone would not dial any number beginning with 001.

X and x are wildcards which match single digit. For example, if 4xxx or 4XXX is entered, the phone would not dial any 4 digits numbers beginning with 4. It would dial numbers beginning with 4 which are longer or shorter than 4 digits.

#### 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	C 720AB Bouload Longth Adjusts from 10 COmfor
Length	G.729AB Payload Length – Adjusts 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 calls the volume level.
MIC Input Volume	Set the MIC calls the volume level.
Broadcast Output	Set the breedeast the output valume level
Volume	Set the broadcast the output volume level.
Signal Tone Volume	Set the audio signal the 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.



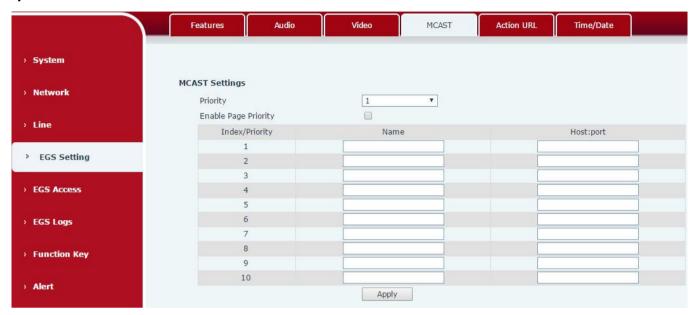
Video		
Field Name	Explanation	
Video Capture		
	Auto: IRCUT switches according to the actual ambient light level of the camera	
IRCUT Mode	Synchronization: The switching of the IRCUT is determined by the actual brightness of	
	the IR lamp.	
	Automatic: automatically switches according to the DNC Threshold and the brightness	
	of the actual environment where the camera is located	
Day/Night Mode	Day Mode: The camera's video screen is always colored, if there is IR-cut will be	
Day/Night widde	synchronized to switch.	
	Night Mode: the camera's video screen is always black and white, if there is IR-cut will	
	be synchronized switch.	
	Automatic: Automatically adjusts according to the actual environment in which the	
White Balance	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	



In the Day / Night mode Auto option, the color switching black and white threshold is set	
In front of a very strong background light can see people or objects clearly	
video changes	
Enable or disable Fill Light	
Only H.264 encoding format is supported	
Main stream: support 720P	
Sub-stream: you can select CIF (352 * 288), D1 (720 * 576)	
The larger the value is, the more coherent the video would be got; not recommend	
adjusted.	
CBR: If the code rate (bandwidth) is insufficient, it is preferred.	
VBR: Image quality is preferred, not recommended.	
Video quality adjustment, the better the quality needs to transfer faster	
It is proportional to video file size, not recommend adjusted.	
The greater the value is, the worse the video quality would be, otherwise the better	
video quality would be; not recommend adjusted.	
When you selected it, the stream is enabled, otherwise disabled	
RTSP Information	
Access the main address of RTSP	
Access the child address of RTSP	



#### 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, monitor and play 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 the priority, if the priority of the incoming flows of multicast RTP, lower precedence than the current common calls, device will automatically ignore the group RTP stream. If the priority of the incoming flow of multicast RTP is higher than the current common calls priority, device will automatically receive the group RTP stream, and keep the current common calls in state. You can also choose to disable in the receiving threshold drop-down box, the device will automatically ignore all local network multicast RTP stream.

- 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 stream
  - ♦ Enable the page priority:

Page priority determines the device how to deal with the new receiving multicast RTP stream when it is in multicast session currently. When Page priority switch is enabled, the device will automatically ignore the low priority multicast RTP stream but receive top-level priority multicast RTP



stream, and keep the current multicast session in state; If it is not enabled, the device will automatically ignore all receiving multicast RTP stream.

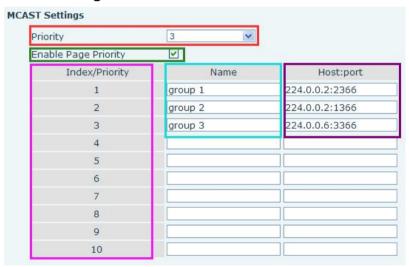
### Web Settings:

ST Settings		
Priority	1	
Enable Page Priority	€	
Index/Priority	Name	Host:port
1	SS	239.1.1.1:1366
2	ee	239.1.1.1:1367

The multicast SS priority is higher than that of EE, which is the highest priority.

Note: when pressing the multicast key for multicast session, both multicast sender and receiver will beep.

### Listener configuration



### Blue part (name)

"Group 1", "Group 2" and "Group 3" are your setting monitoring multicast name. The group name will be displayed on the screen when you answer the multicast. If you have not set, the screen will 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 high priority. The followings will explain how to use this option:



- ♦ The purpose of setting monitoring multicast "Group 1" or "Group 2" or "Group 3" launched a multicast call.
- ♦ All equipment has one or more common non multicast communication.
- ♦ When you set the Priority for the disable, multicast any level will not answer, multicast call is rejected.
- when you set the Priority to a value, only higher than the priority of multicast can come in, if you set the Priority is 3, group 2 and group 3 for priority level equal to 3 and less than 3 were rejected, 1 priority is 2 higher than ordinary call priority device can answer the multicast message at the same time, keep the hold the other call.

### Green part (Enable Page priority)

Set whether to open more priority is the priority of multicast, multicast is pink part number. Explain how to use:

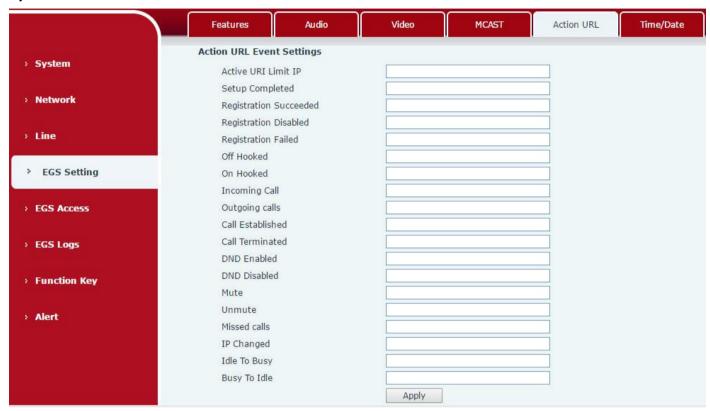
- ♦ The purpose of setting monitoring multicast "group 1" or "3" set up listening "group of 1" or "3" multicast address multicast call.
- ♦ All equipment has been a path or multi-path multicast phone, such as listening to "multicast information group 2".
- ♦ If multicast is a new "group of 1", because "the priority group 1" is 2, higher than the current call "priority group 2" 3, so multicast call will can come in.
- ♦ If multicast is a new "group of 3", because "the priority group 3" is 4, lower than the current call "priority group 2" 3, "1" will listen to the equipment and maintain the "group of 2".

#### **Multicast service**

- **Send:** when configured ok, our key press shell on the corresponding equipment, equipment directly into the Talking interface, the premise is to ensure no current multicast call and 3-way of the case, the multicast can be established.
- Lmonitor: IP port and priority configuration monitoring device, when the call is initiated and incoming
  multicast, directly into the Talking interface equipment.



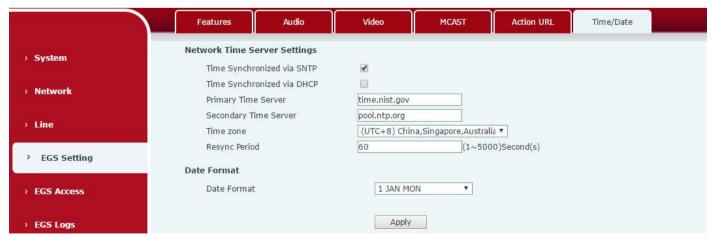
# e) Action URL



### **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





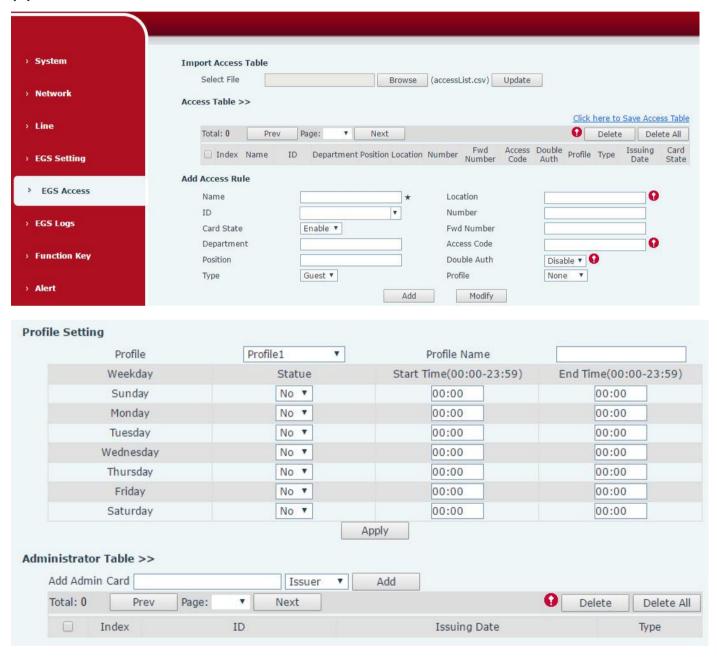


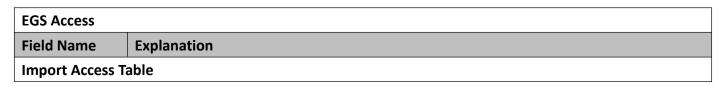
Time/Date			
Field Name	Explanation		
Network Time Server S	Network Time Server Settings		
Time Synchronized via SNTP	Enable time-sync through SNTP protocol		
Time Synchronized via DHCP	Enable time-sync through DHCP protocol		
Primary Time Server	Set primary time server address		
Secondary Time	Set secondary time server address, when primary server is not reachable, the device will try		
Server	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			
Date Format	Select the time/date display format		
Daylight Saving Time So	ettings		
Location	Select the user's time zone specific area		
DST Set Type	Select automatic DST according to the preset rules of DST, or the manually input rules		
Offset	The DST offset time		
Month Start	The DST start month		
Week Start	The DST start week		
Weekday Start	The DST start weekday		
Hour Start	The DST start hour		
Month End	The DST end month		
Week End	The DST end week		
Weekday End	The DST end weekday		



Hour End	The DST end hour		
Manual Time Settings			
The time set by hand, need to disable SNTP service first.			
Daylight Saving Time Settings			

# (5) EGS Access







Click the <Browse> to choose to import remote access list file (access List.csv) and then clicking <Update> can batch import remote access rule.

### **Access Table**

According to entrance guard access rules have been added, you can choose single or multiple rules on this list to delete operation.

tills list to delete o	peration	
Add Access Rule		
Name(necessary)	User name	
Location	Virtual extension number, used to make position call instead of real number.	
Location	It might be taken with unit number, or room number.	
ID.	RFID card number. You can manually fill in the first 10 digits of the card number or	
ID	select the existing card number	
Number	User phone number	
Card State	Enable or disable holder's RFID card	
Fwd Number	Call forwarding number when above phone number is unavailable.	
Department	Card holder's department	
	1/ When the door phone answers the call from the corresponding < Phone Num>	
	user, then the <phone num=""> user can input the access code via keypad to unlock the</phone>	
Access Code	door remotely.	
	2/ The user's private password should be input via keypad for local door unlocking.	
	The private password format is <b>Location</b> * <b>Access Code</b> .	
Position	Card holder's position	
Davida Avith	When the feature is enabled, private password inputting and RFID reading must be	
Double Auth	matched simultaneously for door unlocking.	
Type	Host: the door phone would answer all call automatically.	
Type	Guest: the door phone would ring for incoming call, if the auto answer is disabled.	
Drofile	It is valid for user access rules (including RFID, access code, etc) within corresponding	
Profile	time section. If NONE is selected, the feature would be taken effect all day.	
<b>Profile Setting</b>		
Profile	There are 4 sections for time profile configuration	
Profile Name	The name of profile to help administrator to remember the time definition	
Ctatus	If it is yes, the time profile would be taken effect. Other time sections not included in	
Status	the profiles would not allow users to open door	
Start Time	The start time of section	
End Time	The end time of section	
Administrator Tab	le	
Add Admin Card	You should input the top 10 digits of RFID card numbers. for example, 0004111806,	
Add Admin Card	selected the type of admin card , click <add>.</add>	



Type: Issuer and revocation

When entrance guard is in normal state, swipe card (issuing card) would make entrance guard into the issuing state, and then you can swipe a new card, which the card would be added into the database; when you swipe the issuing card again after cards added done, entrance guard would return to normal state. Delete card operation is the same with issuing card.

The device can support up to 10 admin cards, 1000 copies of ordinary cards.

Note: in the issuing state, swiping deleted card is invalid.

Shows the ID, Issuing Date and Type of admin card		
Delete Clicking < Delete > would delete the admin card list of the selected ID cards.		
Delete All Click < Delete All>, to delete all admin card lists.		

## (6) EGS Logs

According to open event log, can record up to 20W open event, after more than cover the old records.

Click here to Save Logs
Right click on the links to select save target as the door log can export CSV format.

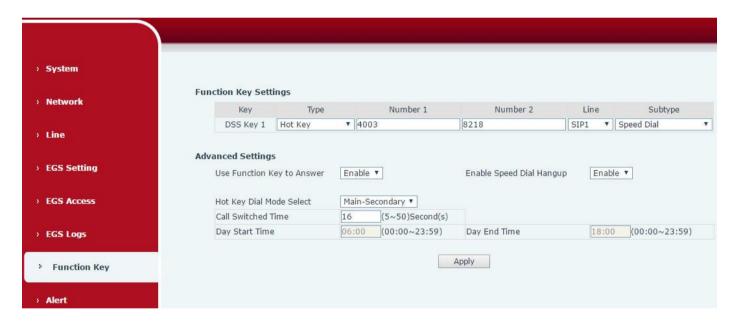


Field Name	Explanation		
Door Open Log	Door Open Log		
Result	Show the results of the open the door ( Succeeded or Failed)		
Time	The time of opening door.		
Access Name	If the door was opened by swipe card or remote unlocking door, the device would display remote access name.		
Access ID	1. If the opening door method is swiping card, it wound display the card number		



	2. If the opening door way is remote access, it wound display the remote extension's
	number.
	3. If the opening door way is local access, there is no display information.
	Open type: 1. Local, 2. Remote, 3. Brush card (Temporary Card, Valid Card and Illegal
	Card).
_	Note: there are three kinds of brushing card feedback results.
Type	1. Temporary Card (only added ) the card number, without adding other rules )
	2. Valid Card (added access rules)
	3. Illegal Card (Did not add information)

# (7) Function Key



# > Key Event

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



Туре	Subtype	Usage	
	None	No responding	
	Dial	Dialing function	
Key Event	Release	Delete password input, cancel dialing input and end call	
	ОК	identification key	



### 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.



Туре	Number	Line	Subtype	Usage
		The SIP account	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.
	account or IP address	ng lines	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 DSS Key multicast web configuration for calling party is as follow:



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



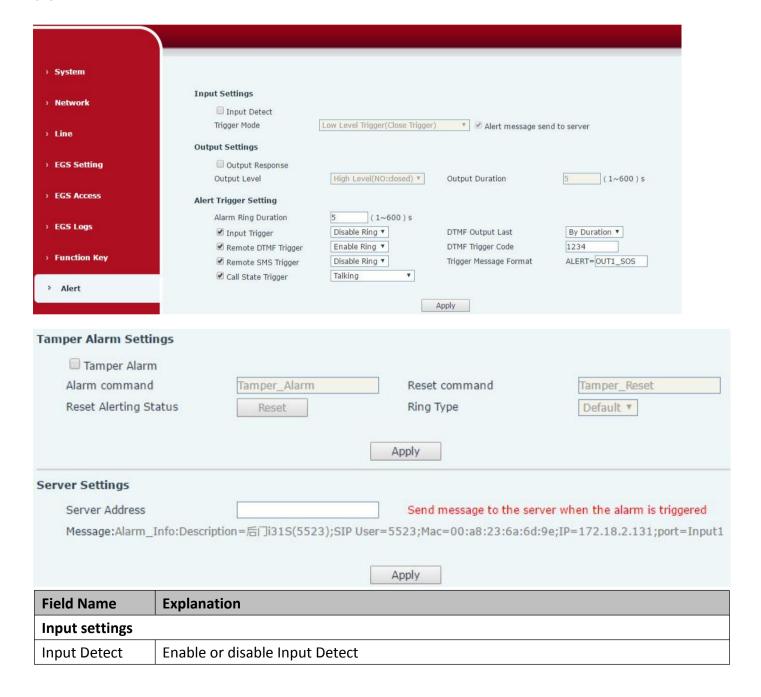
### ♦ operation mechanism

You can define the DSS 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.

### 

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.

## (8) Alert





		ng the low level trigger (closed trigger), detect the input port 1 (low	
Trigger Mode	level) closed t	50	
	When choosing the high level trigger (disconnected trigger), detect the input port 1		
	(high level) di	sconnected trigger.	
Alert message	Set the Alert	message send to server	
send to server			
Output Settings			
Output Response	Enable or disa	able Output Response	
	When choosi	ng the low level trigger (NO: normally open), when meet the trigger	
Output Lovel	condition, trig	gger the NO port disconnected.	
Output Level	When choosi	ng the high level trigger (NO: normally close), when meet the trigger	
	condition, trig	gger the NO port close.	
Output	Changes in no	ort, the duration of. The default is 5 seconds.	
Duration	Changes in pe	ore, the daration on the deladic is a seconds.	
Alert Trigger Sett	ing		
Alarm Ring	Set the Alarm	Ring Duration. The default is 5 seconds.	
Duration	Set the Alarm	Thing Duration. The default is 5 seconds.	
Trigger Mode: "In	iput trigger", "I	Remote DTMF trigger", "Remote SMS trigger", "Call state trigger".	
Call status triggering: there are four triggering modes of Talking / Talking and Ringing / Ringing / Calling			
Input trigger	When the input port meet to trigger condition, the output port will trigger(The Po		
	level time cha	ange, By < Output Duration > control)	
		Received the terminal equipment to send the DTMF password, if	
	By duration	correct, which triggers the corresponding output port (The Port level	
Remote DTMF		time change, By < Output Duration > control)	
trigger		During the call, receive the terminal equipment to send the DTMF	
886.	By Calling	password, if correct, which triggers the corresponding output port (The	
	State	Port level time change, (By call state control, after the end of the call,	
		port to return the default state)	
Remote SMS	In the remote device or server to send instructions to ALERT=[instructions], if correct,		
trigger	which triggers the corresponding output port		
Call state	When the emergency call button to trigger the equipment shell, which triggers the		
trigger	corresponding output port(after the end of the call, port to return the default state)		
Trigger Message	Send instructions on remote devices or servers, ALERT=[set instructions], if correct,		
Format	trigger the corresponding port output.		
Tamper Alarm Se	ttings		
Tamper Alarm	When the selection is enabled, the tamper detection enabled		



Alarm	When detected someone tampering the equipment, will be sent alarm to the	
command	corresponding server	
Docat command	When the equipment receives the command of reset from server, the equipment will	
Reset command	stop alarm	
Reset Alerting	Directly stop the alarm from equipment in the Webpage	
Status		
Ring Type	Set the Ring Type	



# ${\tt V}$ Appendix

# 1. Technical parameters

Communication protocol		SIP 2.0(RFC-3261)	
Main chipset		Broadcom	
Keys	DSS Key	1 (Stainless steel)	
	Numeric keyboard	Support	
Audio	MIC	1	
	Speaker	3W/4Ω	
	Volume control	Adjustable	
	Full duplex speakerphone	Support (AEC)	
Speech	Protocols	RTP	
flow	Decoding	G.729、G.723、G.711、G.722、G.726	
Ports	Active Switched	12V/650mA DC	
	Output		
	WAN	10/100BASE-TX s Auto-MDIX, RJ-45	
Camera		1/4 "color CMOS, 1 megapixel, wide angle	
RFID/IC card reader(optional)		EM4100 (125Khz)Standard configuration	
Ki ib/ic care	a reader(optional)	MIFARE One(13.56Mhz)Custom-made	
Power supply mode		12V / 1A DC or PoE	
PoE		PoE 802.3af (Class 3 - 6.49~12.95W)	
Cables		CAT5 or better	
Shell Material		Cast aluminium panel, Cast aluminium back shell	
Working temperature		-40°C to 70°C	
Working humidity		10% - 95%	
Storage temperature		-40°C to 70°C	
Installation way		Wall mounted or In-wall	
Dimension		Wall mounted: 223*130*74mm	
		In-wall: 270*150*61mm	
Package size		310x175x115mm	
Equipment weight		1500g	
Gross weight		1800g	



### 2. Basic functions

- 2 SIP lines
- PoE Enabled
- Full-duplex speakerphone (HF)
- Numeric keypad (Dial pad or Password input)
- Intelligent DSS Keys (Speed Dial/intercom etc)
- Wall mounted / In-wall
- Integrated RFID Card reader
- 1 indoor switch interface
- 1 electric lock relay
- Anti-tamper switch
- External power supply
- Door phone: call, password, RFID card, indoor switch
- Protection level: IP65, IK10, CE/FCC

# 3. Schematic diagram





# **VI** Other instructions

## 1. Open door modes

### Local

### 1) Local Password

- ♦ Set <Local Password> (the default is "6789") via DOOR PHONE\DOOR PHONE as above.
- ♦ Use the device's keypad to input password and "#" key, then the door will be unlocked.

### 2) Private access code

- ♦ Set <Add Access Rule\Access Code> and enable local authentication.
- ♦ Use the device's keypad to input access code and "#" key, then the door will be unlocked.

### Remote

#### 1) Visitors call to owner

- ❖ Visitors call to owner via position speed dial or phone number. (When set the speed dial key, can press it to call direct.)
- ♦ The owner answers the call, with pressing the "\*" key to unlock the door for visitors.

### 2) Owner calls to visitors

- ♦ Owner calls to visitors via SIP phone.
- ♦ SIP door phone answers the call automatically.
- ♦ Owner use keypad to input corresponding <Access codes> to unlock the door.

#### Slot cards

♦ Use pre assigned RFID cards to unlock the door, by touching RFID area of device.

### Indoor switch

Press indoor switch, which is installed and connected with device, to unlock the door.



### 2. Management of card

### Add Administrator

There are 2 types of Administrator cards: issuer used for adding cards, revocation used for deleting cards.



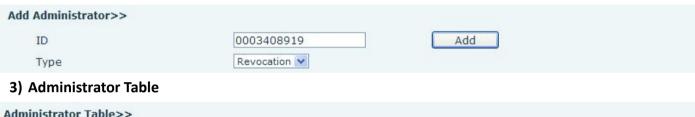
### 1) Add<Issuer admin card >

Input a card's ID, selected <Issuer> in the types and Clicked <Add>, you can add Issuer admin card.



### 2) Add<Revocation admin card>

Input a card's ID, selected <Revocation> in the types and Clicked <Add>, you can add Revocation admin card.



Administrator Table>>		
ID	Date	Туре
0003476384	JAN 01 02:09:04	Issuer
0003408919	JAN 01 02:09:29	Revocation

### Delete Administrator

Select the admin card of need to delete, click < Delete >.



### Add user cards

Method 1: used to add cards for starters typically

1) In web page < EGS Setting\Card Reader Working Mode> option, select <Card Issuing> function.



- 2) Click <Apply>, Card Reader would be entered the issuing status.
- 3) Use new card to touch card reader induction area, and then you might hear the confirmed indication tone from the device. Repeat step 3 to add more cards.
- 4) In web page <EGS Setting\Card Reader Working Mode > option, select <normal> function.





- 5) Click <Apply>, Card Reader would be back to the Normal status.
- 6) The issuing records can be found from the Access table list.



### Method 2: used to add cards for professionals

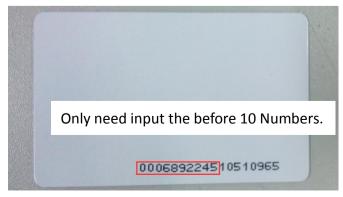
- 1) Use <lssuer admin card> to touch card reader induction area, and it would be entered issuing card status.
- 2) Use new card to touch card reader induction area, and you might hear the confirmed indication tone from the device. Repeat step 2 to add more cards.
- 3) Use <Issuer admin card> to touch card reader induction area again, it would be back to normal working status.

#### Methods 3: use to add few cards

1) Input cards number in <EGS Setting\Add Access Rule\ID> page, and then click <Add>.



Note: you can also use the USB card reader connected with PC to get cards ID automatically.





#### Delete user cards

Method 1: used to batch delete cards for starters.

1) In web page <EGS Setting\Card Reader Working Mode> option, select <Card revoking>.



- 2) Click <Apply>, Card Reader would be entered the revoking status.
- 3) Use card to touch card reader induction area, and you might hear the card reader confirmed indication tone. Repeat step 3 to delete more cards.
- 4) In web page <EGS Setting\Card Reader Working Mode >option, select <normal>.



5) Click <Apply>, Card Reader would be back to the Normal status.

Method 2: used to batch add cards for intermediates.

- 1) Use < Revocation admin card> to touch card reader induction area, and it would be entered revoking card status.
- 2) Use the cards you want to delete from system, to touch card reader induction area, and you might hear the card reader confirmed indication tone. Repeat step 2 to delete cards.
- 3) Use <Revocation admin card> to touch card reader induction area, and it would be back to card read only status.

Method 3: use to batch delete cards or delete few cards.

1) In web page<EGS Access\Access Table>select the card ID and then click <Delete>.

**Note:** If you click <Delete All>, system will delete all the ID cards.

