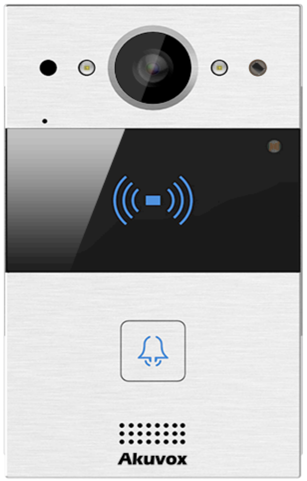
Akuvox Smart Intercom



R20A Door Phone Admin Guide

About This Manual

Thank you for choosing Akuvox's R20A door phone. This manual is intended for end users, who need to properly configure the door phone. This manual is applicable to 20.0.1.2xx version, and it provides an overview of the most essential functions and features of the product. Please visit Akuvox forum or consult technical support for any new information or latest firmware.

Note: Please refer to universal abbreviation form in the end of manual when meet any abbreviation letter.

Content

1. Product Overview

1.1. Product Description

Akuvox R20A is a SIP-compliant, hands-free one button video outdoor phone. It can be connected with Akuvox indoor monitors for remote access controlling and monitoring. Users can communicate with visitors via audio and video calls, and unlock the door if they need. Users can also use RFID cards to unlock the door. It is applicable in villas, offices and so on.

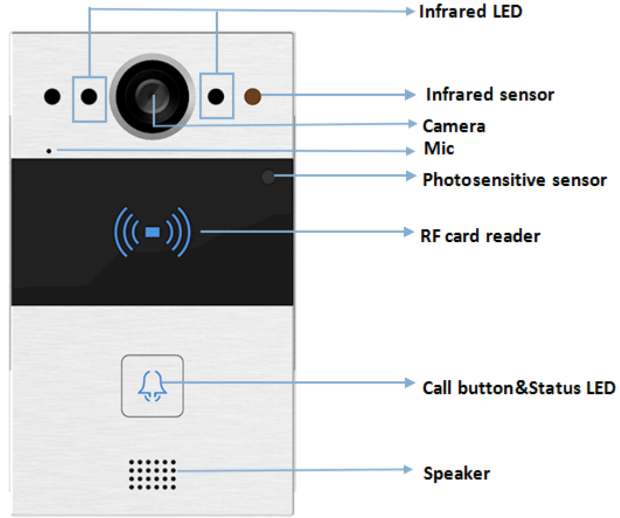


Figure 1.1 Product Description

1.2. Connector Introduction

Ethernet(POE): Ethernet (POE) connector which can provide both power and network connection.

12V/GND: External power supply terminal if POE connector is not available.

RS485-A/B: RS485 terminal.

DOORA/B: Trigger signal input terminal.

RelayA/B (NO/COM/NC): Relay control terminal.

Note: The general door phone interface diagram is only for reference.

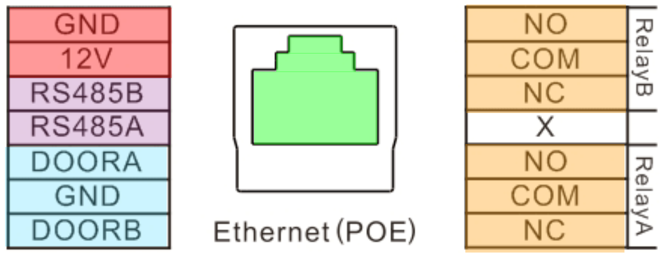


Figure 1.2-1 Connector Interface

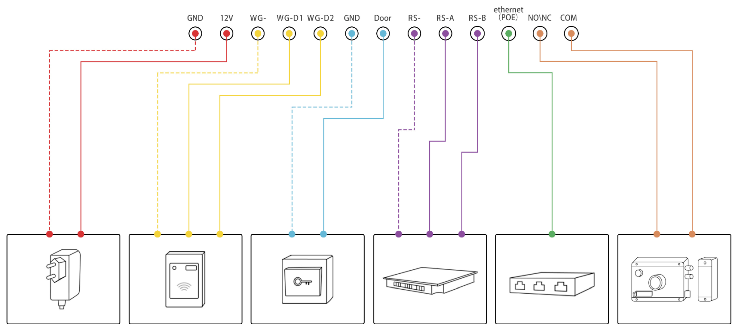


Figure 1.2-2 General interface

1.3. LED Status Information

LED Status		**Description**
Blue	Always on	Normal status
	Flashing	Calling
Red	Flashing	Network is unavailable
Green	Always on	Talking on a call
	Flashing	Receiving a call
Pink	Flashing	Upgrading

2. Daily Use

2.1. Making a Call

Press the call button to call out the predefined number or IP address and if LED turns green, it means the call has been answered.

2.2. Receiving a Call

Users can use IP phone or indoor monitor to call R20A and R20A will answer it automatically by default. If auto answer is disabled, pressing call button to answer the incoming call.

2.3. Unlock

2.3.1. Unlock by RF Card

Place the predefined user cards in RFID card reader to unlock. Under normal conditions, R20A will announce “The door is now opened”. Both 13.56MHz and 125KHz RFID cards are supported on R20A.

2.3.2. Unlock by DTMF Codes

Users can press the predefined DTMF code from an answer unit to remotely unlock the door during the call. Users will also hear “The door is now opened.”

3. Basic Features

3.1. Access the website setting

3.1.1. IP Announcement

While R20A starts up normally, hold the call button for several seconds after the Status LED turns blue, voice system will enter IP announcement mode. In IP announcement mode, the IP address will be announced periodically and “IP 0.0.0.0” would be announced if no IP address is gained. Press Call Button again to quit the announcement mode.

3.1.2. Access the device website

Open a web browser, and access the corresponding IP address. Enter the default user name and password to login. The default administrator’s user name and password are shown as below:

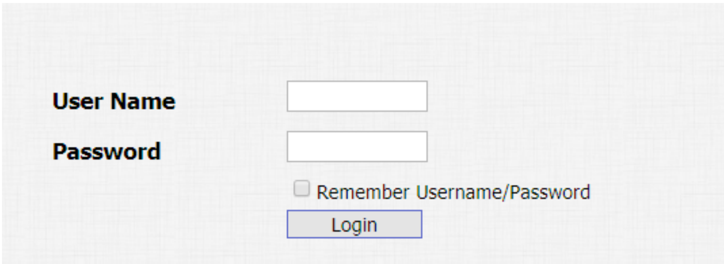


Figure 3.1.2 Access the device website

User Name**: admin**

Password**: admin**

Note: The recommended browser is Google Chrome.

3.2. Password Modification

3.2.1. Modify the web password

Go to **Security** - **Basic** to modify password for webpage.

To modify password for “admin” or “user” account.

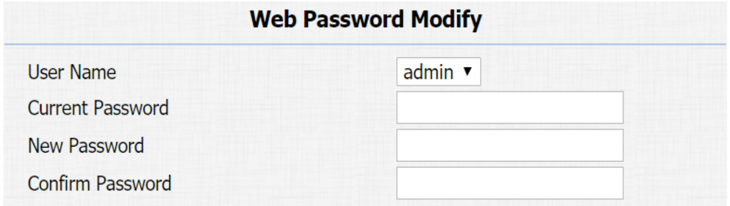


Figure 3.2.1 Modify the web password

3.3. Phone Configuration

3.3.1. Language

Go to **Phone** - **Time/Lang** to select language for webpage.

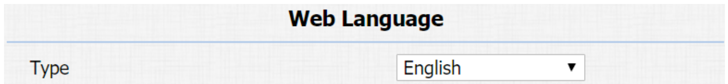


Figure 3.3.1 Language

3.3.2. Network Setting

Go to **Network - Basic**, dynamically or statically to obtain address.

3.3.2.1. DHCP

R20A uses DHCP by default, it will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically.

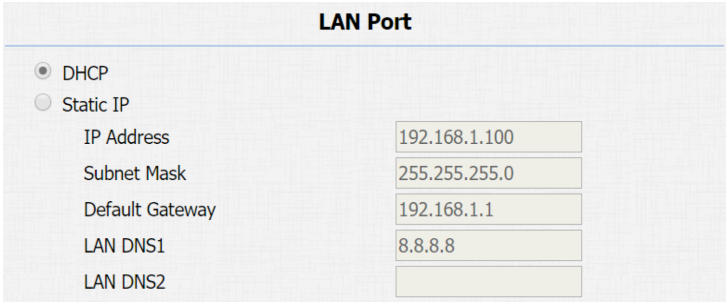


Figure 3.3.2.1 DHCP mode

3.3.2.2. Static IP

If selected, you could manually set IP address, Subnet Mask, Default Gateway and DNS server. The figure 3.3.2.2 shows static IP setting.

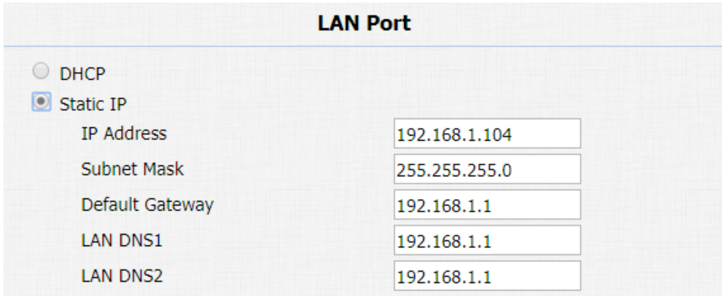


Figure 3.3.2.2 Static IP mode

3.3.3. Sound

Go to **Phone** - **Voice** to configure volume and upload tone file.

Mic Volume: To configure microphone volume.

Speaker Volume: To configure speaker volume.

Open Door Warning: Disable it, and users will not hear the prompt voice when the door is opened.

IP Announcement: To configure the valid time when IP Announcement is available and the loop time of IP Announcement.

RingBack Upload: To upload the ring back tone by users themselves.

Opendoor Tone Upload: To upload the opendoor tone by users themselves.

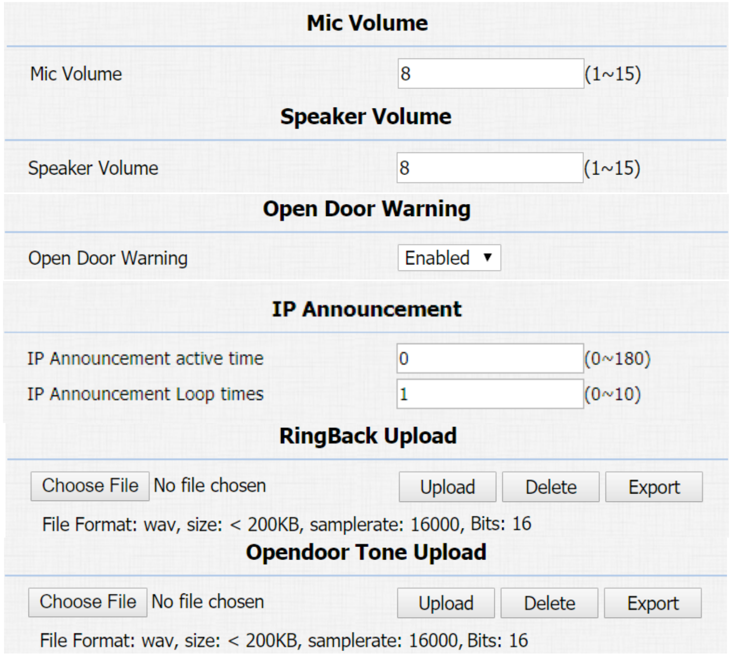


Figure 3.3.3 Sound

3.4. Intercom Call

3.4.1. Direct IP Call

Figure 3.4.1-1 Direct IP call

Go to **Phone** - **Call Feature** to enable the direct IP call for door phones first.

Then, go to **Intercom - Basic** to configure the IP address of the destination(E.g. IP address 192.168.1.100). It supports up to 8 lines simultaneously.

After all, press the push button to make direct IP call.

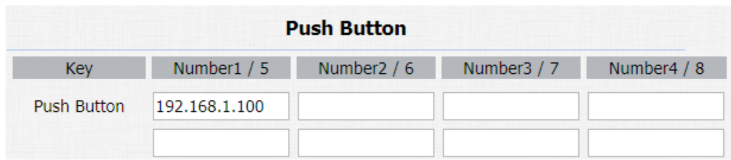


Figure 3.4.1-2 Push Button Number

3.4.2. SIP Call

SIP calls which use SIP numbers to make or receive calls should be supported by SIP server. Users need to register accounts and fill SIP feature parameters before using it.

Go to **Account** - **Basic** to configure SIP account and SIP server for door phones first.

3.4.2.1. SIP Account

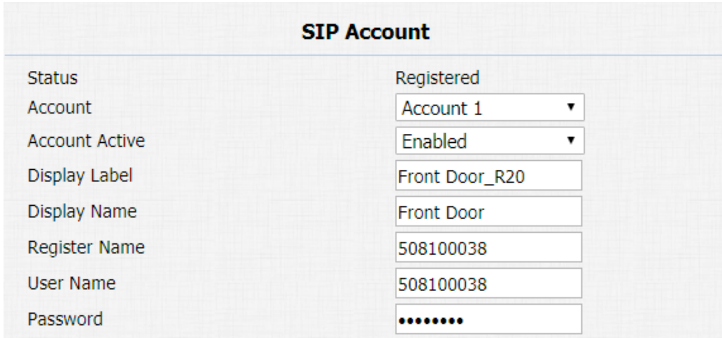


Figure 3.4.2.1 SIP account

Status: To display register result.

Account: To switch the account to be configured. R20A supports 2 SIP accounts.

Account Active: To enable the account, it is disabled by default.

Display Label: To configure label displayed on the phone's LCD screen.

Display Name: To configure name sent to the other call party for displaying.

Register Name: To enter extension number which users want and the number is allocated by SIP server.

User Name: To enter user name of the extension.

Password: To enter password for the extension.

3.4.2.2. SIP Server 1&2

Server IP 1: To enter SIP server's IP address or URL.

Server IP 2: To display and configure secondary SIP server settings. This is for redundancy, if registering to primary SIP server fails, the phone will go to secondary SIP server for registering.

Registration Period: The registration will expire after registration period, the phone will re-register automatically within registration period.

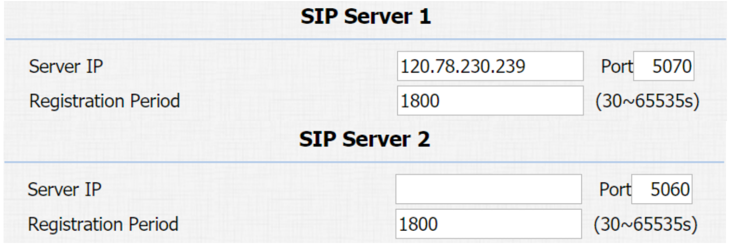


Figure 3.4.2.2 SIP server 1&2

3.4.2.3. Outbound Proxy Server

An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server.

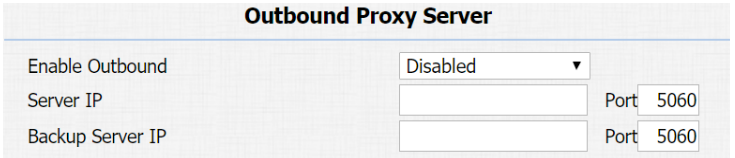


Figure 3.4.2.3 Outbound proxy server

3.4.2.4. Transport Type

To display and configure transport type for SIP message.

There are 4 transport types in total.

- **UDP:** UDP is an unreliable but very efficient transport layer protocol.
- **TCP:** Reliable but less-efficient transport layer protocol.
- **TLS:** Secured and reliable transport layer protocol.
- **DNS-SRV:** DNS record for specifying the location of services.

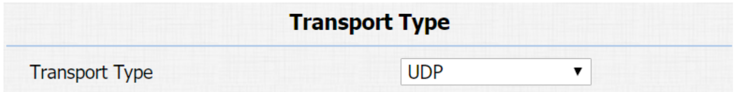


Figure 3.4.2.4 Transport type

3.4.2.5. NAT

To display and configure NAT settings.

- **STUN:** Short for session traversal utilities for NAT, a solution to solve NAT issues.

Note: By default, NAT is disabled.

After all, press the push button to make direct IP call.

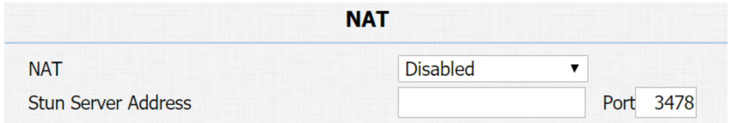


Figure 3.4.2.5 NAT

3.4.3. .Auto Answer

Go to **Account** - **Advanced** to enable auto answer feature for SIP calls.

Figure 3.4.3-1 Auto answer for sip calls

Figure 3.4.3-2 Auto answer for direct IP calls

Go to **Phone** - **Call Feature** to enable auto answer feature for direct IP calls.

Auto Answer Delay: To configure delay time before an incoming call is automatically answered.

Auto Answer Mode: To set video or audio mode for auto answer by default.

Then incoming calls will be answered automatically.

Figure 3.4.3-3 Auto answer options' parameters

3.4.4. Web Call

Go to **Intercom** - **Basic** to dial out or answer incoming calls from website.

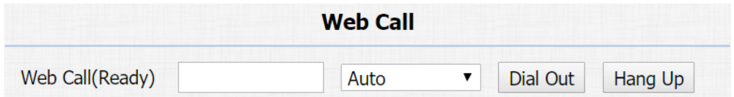


Figure 3.4.4 Web call

3.4.5. No Answer Call

Go to **Intercom - Basic** to configure.

No Answer Call: If enabled, R20A will call to No Answer Call1 and No Answer Call2 in sequence automatically when push button call is not answered over timeout(30s by default).

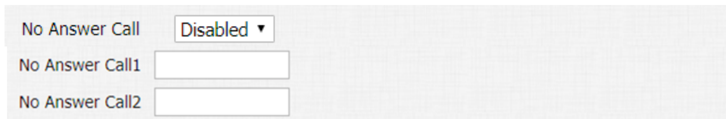


Figure 3.4.5- No Answer Call

3.5. Security

3.5.1. Live view

Go to **Intercom** - **Live Stream** to check the real-time video from R20A.

In addition, user also can check the real-time picture via URL**:** **http://IP_address:8080/picture.jpg**.

Figure 3.5.1 Live view

3.5.2. RTSP

R20A supports RTSP stream, go to **Intercom** - **RTSP** to enable or disable RTSP server. The URL for RTSP stream is**:** **rtsp://IP_address/live/ch00_0**.

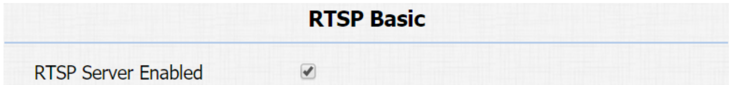


Figure 3.5.2 RTSP

3.5.3. Onvif

R20A supports ONVIF protocol, which means R20A's camera can be searched by other devices, like NVR, which supports ONVIF protocol as well.

Go to **Intercom** - **ONVIF** to configure ONVIF Mode and its username and password.

Switching ONVIF Mode to Undiscoverable means that User must program ONVIF's URL manually.

The ONVIF's URL is**:**

http://IP_address:8090/onvif/device_service.

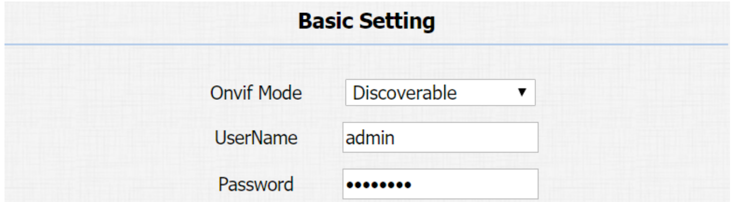


Figure 3.5.3 ONVIF

3.6. Access Control

3.6.1. Relay

Go to **Intercom** - **Relay** to configure relay settings.

There are three terminals of relay**:** NO, NC and COM. NO stands for normally open contact while NC stands for normally closed contact.

Relay ID: R20A supports two relays, user can configure them respectively.

Relay Type: Default state means NC and COM are normally closed, while Invert state means NC and COM are normally opened.

Relay Delay: To configure the duration of opened relay. Over the value, the relay would be closed again.

Relay Status: While the relay is triggered, the statues will be switched. When COM connects to NC, the status is Low.

Note: Relay operates a switch and does not deliver power, so user should prepare power adapter for external devices which connects to relay.

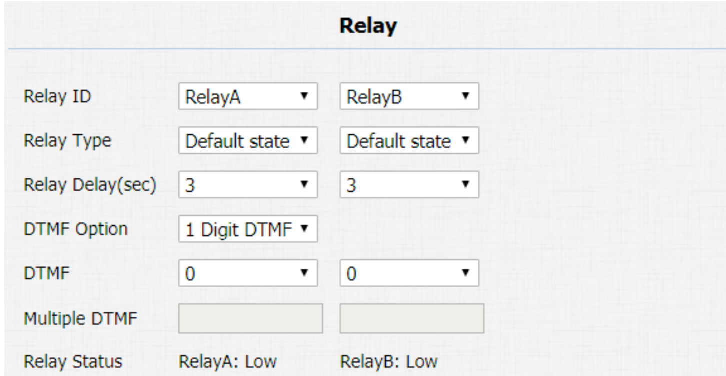


Figure 3.6.1 Relay

3.6.2. Unlock via DTMF code

Users can press the predefined DTMF code from an answer unit to remotely unlock the door during the call. Users will also hear "The door is now opened.”

Go to **Intercom** - **Relay** to configure DTMF code parameters.

DTMF Option: To select digit of DTMF code, R20A support maximum 4 digits DTMF code.

DTMF&Multiple DTMF: To configure DTMF code for remote unlocking.

3.6.3. Unlock via RF Card(Optional)

Go to **Intercom** - **Card setting** setting to manage card access system.

Import/Export Card Data

R20A supports import or export the card data file, which is convenient for administrator to deal with a large number of cards. The maximum card data file is 200K which is around 500 cards.

Note: Please consult administrator for the .xml format RFID cards template file.

Obtain and Add Card

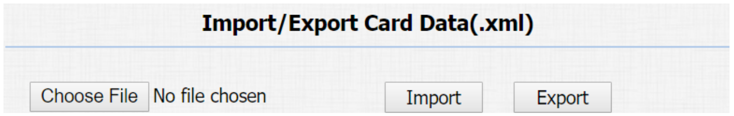


Figure 3.6.3-1 Import/Export Card Data

- Switch card status to “Card Issuing” and click “Apply;”
- Place card on the card reader area and click “Obtain;”
- Name card, choose which door users want to open and the valid day and time;
- Click “Add” to add it into list.

Valid card information will be shown in the list. Administrator could delete one card’s access permission or empty all the list.

Note: Remember to set Card Status back to “Normal” after adding cards.

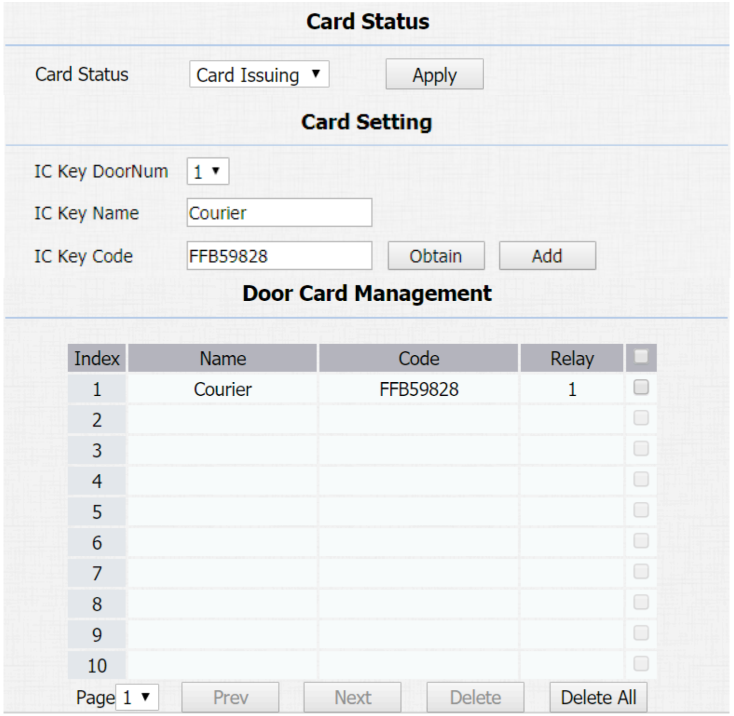


Figure 3.6.3-2 RFID cards in website

3.6.4. Unlock via HTTP command

Users can use a URL to remote unlock the door.

Go to **Intercom** - **Relay** to configure.

Switch: Enable this function. Disable by default.

UserName&Password: Users can setup the username and password for HTTP unlock.

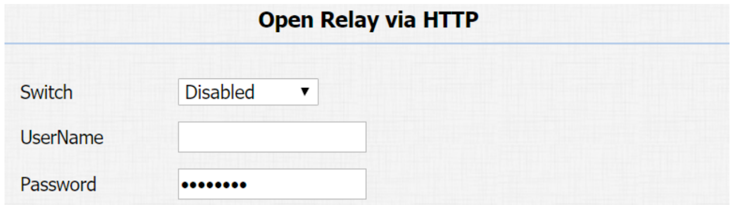


Figure 3.6.4 Unlock via HTTP command

URL format:

http://IP_address/fcgi/do?action=OpenDoor&UserName=&Password=&DoorNum=1.

3.6.5. Unlock via Exit Button

Go to **Intercom** - **Input** to configure input settings.

R20A supports two input triggers “Input A/B(DOOR A/B).”

Input Service: To enable or disable input trigger service.

Trigger Option: To choose open circuit trigger or closed circuit trigger. “Low” means that connection between door terminal and GND is closed, while “High” means the connection is opened.

Door status: To show the status of input signal.

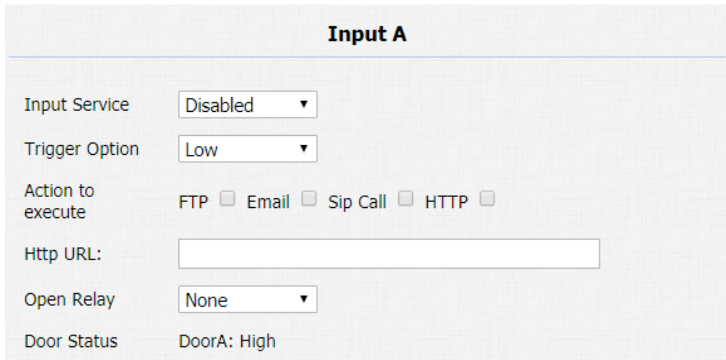


Figure 3.6.5 Unlock via exit button

3.7. Reboot

Go to **Upgrade** - **Basic**, users can reboot the phone.

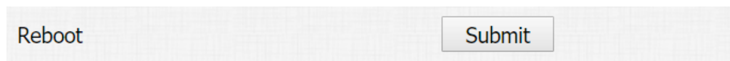


Figure 3.7 Reboot

3.8. Reset

Go to **Upgrade** - **Basic**, user can reset the phone to factory settings.

Figure 3.8 Reset in website

4. Advanced Features

4.1. Phone Configuration

4.1.1. LED

Go to **Intercom - LED Setting** to configure the LED status.

To setup the LED lighting mode.

State: There is five states**:** Normal, Offline, Calling, Talking and Receiving.

Color Off: The default status is OFF.

Color On: It can support three color**:** Red,Green,Blue.

Blink Mode: To setup the different blink frequency.

LED Control:

Use Http URL to remote control the LED status.

Http format:

http://PhoneIP/fcgi/do?action=LedAction&State=1&Color=1&Mode=2500

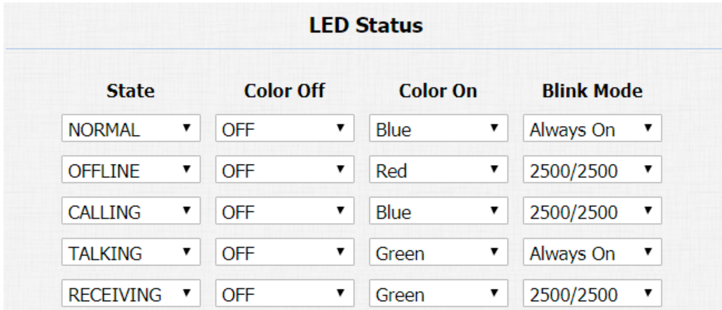


Figure 4.1.1-1 LED

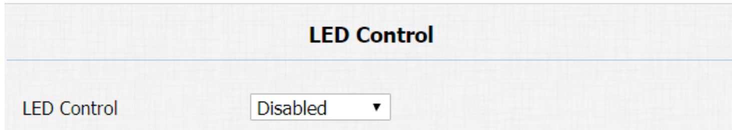


Figure 4.1.1-2 LED

Status: 1=Idle; 2=OffLine; 3=Calling; 4=Talking; 5=Receiving; Color: 1=Green; 2=Blue; 3=Red; Mode: 0=Always On; 1=Always Off; 500/1000/1500/2000/25000/3000

4.1.2. IR LED

Go to **Intercom** - **Advanced** to configure.

Photoresistor: The setting is for night vision, when the surrounding of R20A is very dark, infrared LED will turn on and R20A will turn to night mode.

Photoresistor value relates to light intensity and larger value means that light intensity is smaller.

Users can configure the upper and lower bound and when photoresistor value is larger than upper bound, infrared LED will turn on. As contrast, when photoresistor value is smaller than lower bound, infrared LED will turn off and device turns to normal mode.

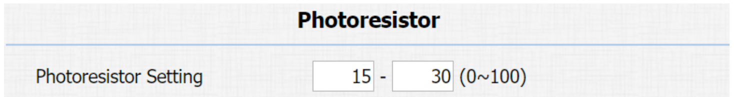


Figure 4.1.2 IR LED

4.2. Intercom

4.2.1. Call Time Related

Go to **Intercom** - **Basic** to configure.

Max Call Time: To configure the max call time.

Dial In Time: To configure the max incoming dial time, available when auto answer is disabled.

Dial Out Time: To configure the max no answer call time.

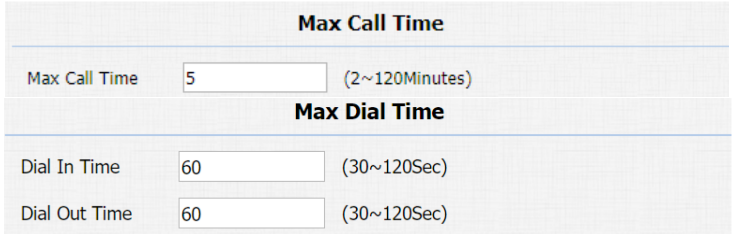


Figure 4.2.1 Call time related

4.2.2. Return Code When Refuse

Go to **Phone** - **Call Feature** to configure.

Return Code When Refuse: Allows users to assign specific code as return code to SIP server when an incoming call is rejected.

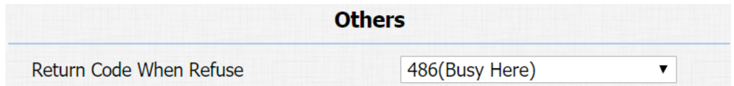


Figure 4.2.2 Return code when refuse

4.2.3. Sip Call Related

Go to **Account** - **Advanced** to configure the SIP call related.

Max Local SIP Port: To configure maximum local SIP port for designated SIP account.

Min Local SIP Port: To configure maximum local SIP port for designated SIP account.

Caller ID Header: To choose Caller ID Header format.

Anonymous Call: If enabled, R20A will block its information when calling out.

Anonymous Call Rejection: If enabled, calls who block their information will be screened out.

Missed Call Log: If enabled, any missed call will be recorded into call log.

Prevent Hacking: If enabled, it will prevent SIP message from hacking.

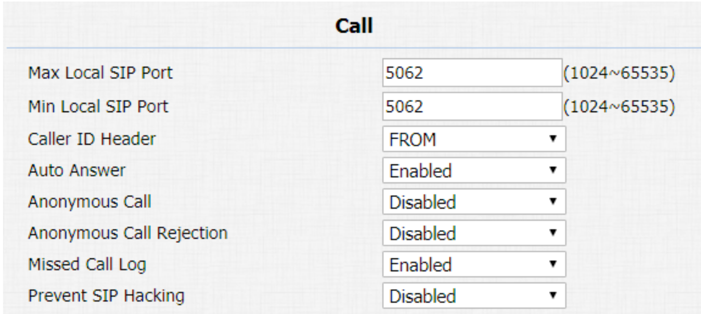


Figure 4.2.3 SIP call related

4.2.4. Codec

Go to **Account** - **Advanced** to configure SIP call related codec.

Sip Account: To choose which account to configure.

Audio Codec: R20A supports four audio codecs: PCMA, PCMU, G729, G722. Different audio codecs require different bandwidth, users can enable/disable them according to different network environment.

Note: Bandwidth consumption and sample rates are as below:

Codec	Bandwidth	Sample Rates
PCMA	64kbit/s	8kHz
PCMU	64kbit/s	8kHz
G729	8kbit/s	8kHz
G722	64kbit/s	16kHz

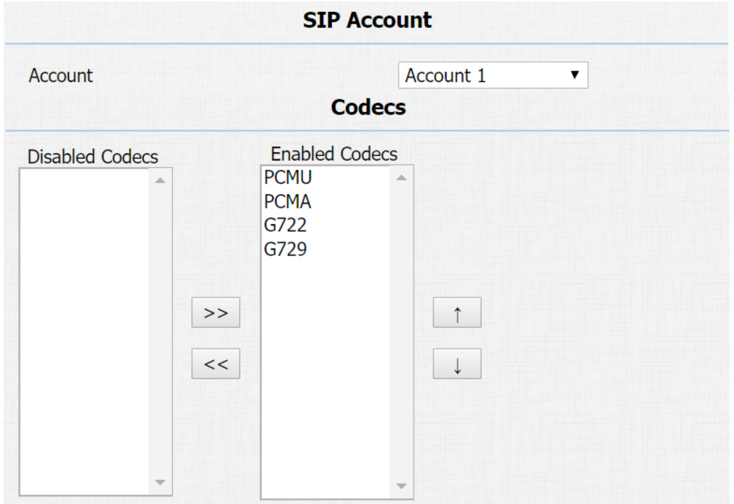


Figure 4.2.4-1 SIP call related codec

Video Codec: R20A supports H.264 standard, which provides better video quality at substantially lower bit rates than previous standards.

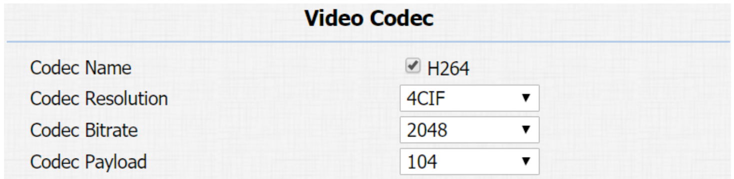


Figure 4.2.4-2 Video codec setting

Figure 4.2.4-2 Multicast related codec

Codec Resolution: R20A supports four resolutions: QCIF, CIF, VGA, 4CIF and 720P.

Codec Bitrate: To configure bit rates of video stream.

Codec Payload: To configure RTP audio video profile.

Multicast codec： Go to **Phone** - **Call Feature** to configure multicast related codec.

4.2.5. Session Timer

Go to **Account** - **Advanced** to configure.

If enabled, the on going call will be disconnected automatically once the session expired unless it's been refreshed by UAC or UAS.

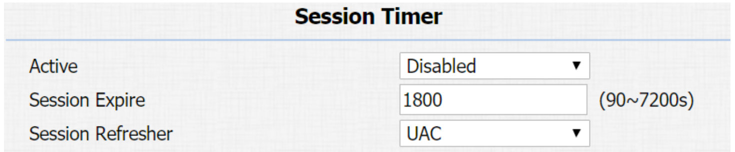


Figure 4.2.5 Session timer

4.2.6. Encryption

Go to the path **Account - Advanced** If enabled, voice will be encrypted.

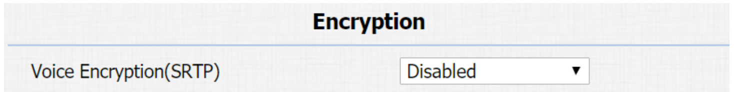


Figure 4.2.6 Encryption

4.2.7. NAT

Go to **Account** - **Advanced** to display NAT related settings.

UDP Keep Alive message: If enabled, IP phone will send UDP keep-alive message periodically to router to keep NAT port alive.

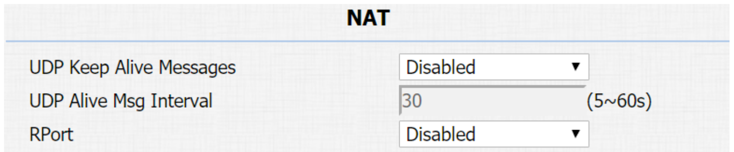


Figure 4.2.7 NAT

UDP Alive Msg Interval: Keep alive message interval.

Rport: Remote port, if enabled, it will add remote port into outgoing SIP message for designated account.

4.2.8. User Agent

Go to **Account** - **Advanced** to configure. One can customize user agent field in the SIP message. if user agent is set to specific value, users can see the information from PCAP. If user agent is blank, by default, users can see the company name "Akuvox", model number and firmware version from PCAP.

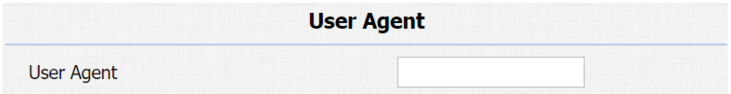


Figure 4.2.8 User Agent

4.3. Access Control

4.3.1. Web Relay

R20A supports extra web relay.

Go to **Phone** - **WebRelay** to configure.

Type: Connect web relay and choose the type.

IP Address: Enter web relay IP address.

User Name: It is an authentication for connecting web relay.

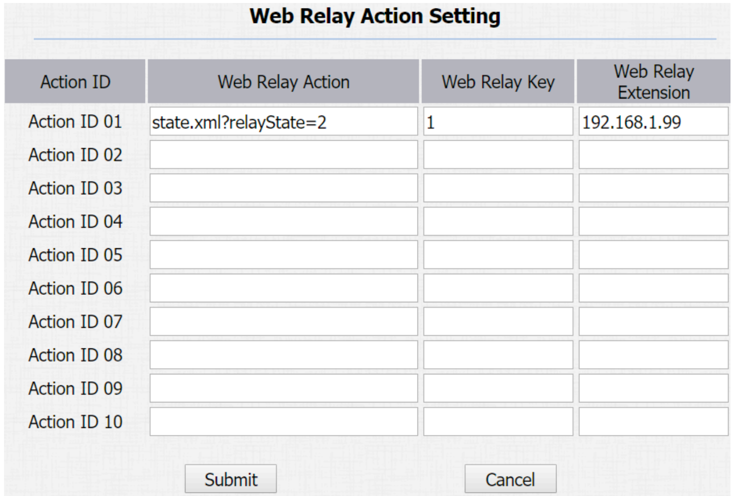
Password: It is an authentication for connecting web relay.

Web Relay Action: Web relay action is used to trigger the web relay. The action URL is provided by web relay vendor.

Web Relay Key: If the DTMF keys are same with the local relay, the web relay will be open with local relay. But if there are different, the web relay is invalid.

Web Relay Extension: The web relay can only receive the DTMF signal from the corresponding extension number.

Figure 4.3.1-1 Web relay



Web Relay Action Setting

Action ID	Web Relay Action	Web Relay Key	Web Relay Extension
Action ID 01	state.xml?relayState=2	1	192.168.1.99
Action ID 02			
Action ID 03			
Action ID 04			
Action ID 05			
Action ID 06			
Action ID 07			
Action ID 08			
Action ID 09			
Action ID 10			

Submit Cancel

Figure 4.3.1-2 Web relay action settings

Note: Users can modify username and password in web relay website.

4.4. Security

4.4.1. Anti-alarm

Go to **Intercom** - **Advanced** to configure.

Tamper Alarm: R20A integrates internal gravity sensor for the own security, and after enabling tamper alarm, if the gravity of R20A changes dramatically, the phone will alarm. Gravity sensor threshold stands for sensitivity of sensor.

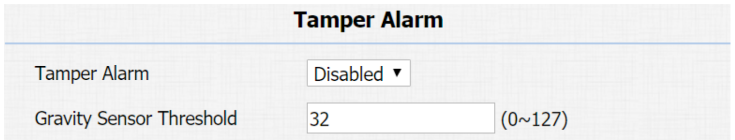


Figure 4.4.1 Anti-alarm

4.4.2. Motion

R20A supports motion detection, go to **Intercom** - **Motion** to configure detection parameter.

Motion Detection: To enable or disable Motion Detection.

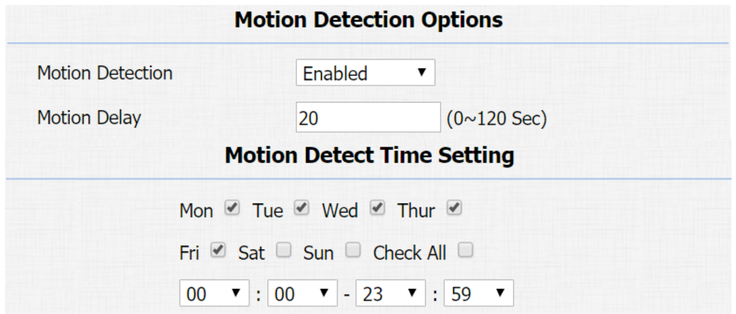


Figure 4.4.2 Motion

Motion Delay: To configure minimum time gap between two snapshot.

Motion Detect Time Setting: To make Motion Detect Time for a whole week.

4.4.3. Action

R20A supports to send notifications, snapshots via email and ftp transfer method, or calls via sip call method, when trigger specific actions.

4.4.3.1. Action Parameters

Go to **Intercom** - **Action** to set action receiver.

Email Notification

Sender's email address: To configure email address of sender.

Receiver's email address: To configure email address of receiver.

SMTP server address: To configure SMTP server address of sender.

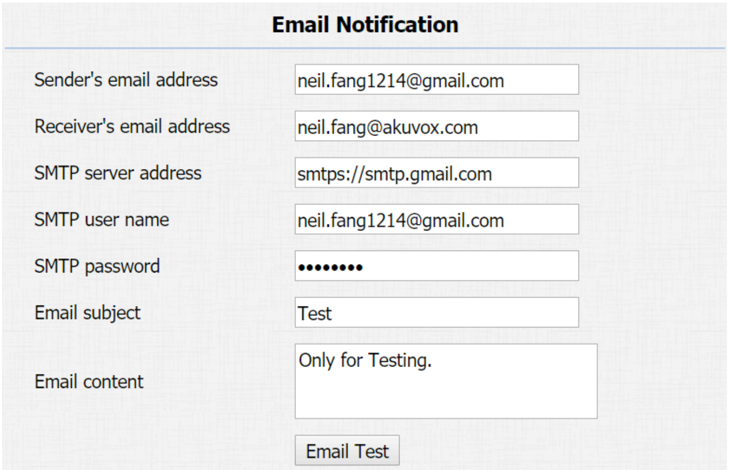


Figure 4.4.3.1-1 Email notification parameters

SMTP user name: To configure user namer of SMTP service (usually it is same with sender's email address).

SMTP password: To configure password of SMTP service (usually it is the same with the password of sender's email).

Email subject: To configure subject of email.

Email content: To configure content of email.

Email Test: To test whether email notification is available.

FTP Notification

FTP Server: To configure URL of FTP server.

FTP User Name: To configure user name of FTP server.

FTP Password: To configure password of FTP server.

FTP Test: To test whether FTP notification is available.

SIP Notification

SIP Call Number: To configure sip call number.

SIP Call Name: To configure display name of R20A.

Three specific actions which will be triggered on R20A**:**

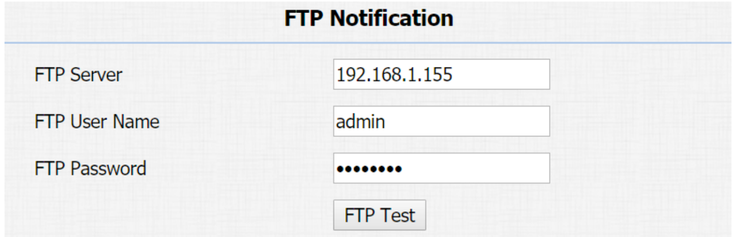


Figure 4.4.3.1-2 FTP notification parameters

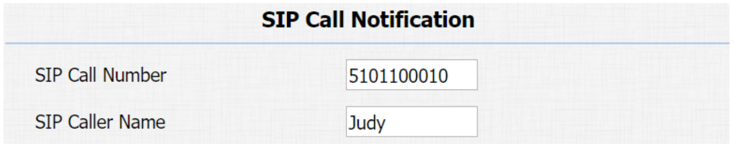


Figure 4.4.3.1-3 SIP call notification parameters

4.4.3.2. Pushbutton Action

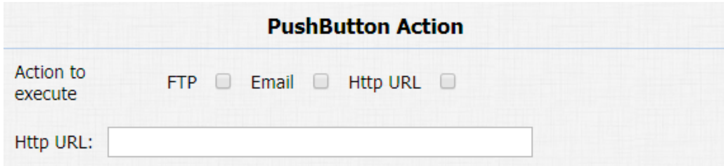


Figure 4.4.3.2 Pushbutton Action

Go to **Intercom** - **Basic** to configure.

Action to execute: To choose suitable way to receive message or snapshot when dialing out.

HTTP URL: If you choose HTTP mode, enter the URL format: **http://http server IP address/any information**.

4.4.3.3. Motion Triggered Action

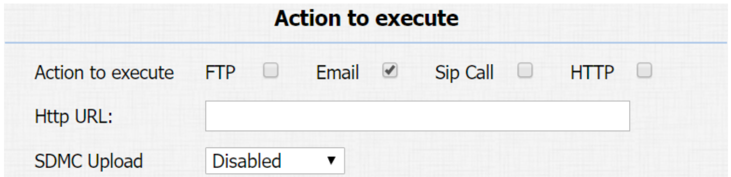


Figure 4.4.3.3 Motion triggered action

Go to **Intercom** - **Motion** to configure.

Action to execute: To choose which action to execute after triggering.

4.4.3.4. Input Interface Triggered Action

Go to **Intercom** - **Input** to configure.

Action to execute: To choose which action to execute after triggering.

Http URL: To configure URL, if HTTP action is chosen.

Action Delay: To configure after how long to execute to send out notifications and trigger relay.

Open relay: To configure which relay to trigger.

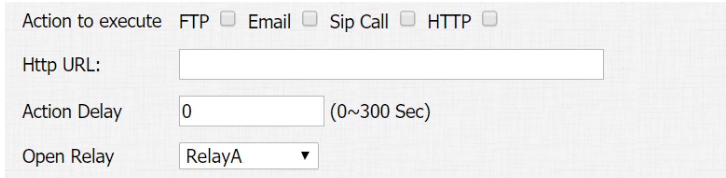


Figure 4.4.3.4 Input interface triggered action

4.5. Upgrade

4.5.1. Web Upgrade

Go to **Upgrade - Basic** to do web upgrade.

Upgrade: Choose .rom firmware from your PC, then click “Submit” to update.

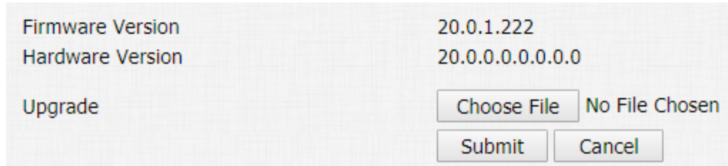


Figure 4.5.1 Web upgrade

4.5.2. Backup config file

Go to **Upgrade** - **Advanced** to backup the config file.

Export Config File: To export current config file.

Others: To export current config file (Encrypted) or import new config file.

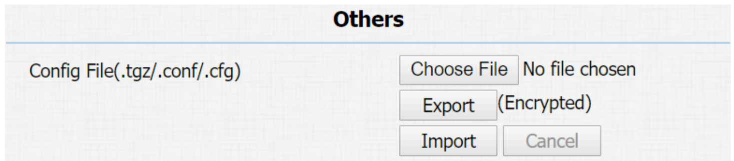


Figure 4.5.2 Backup config file

4.6. Log

4.6.1. Call Log

Go to **Phone - Call Log**, users can see a list of call which have dialed, received or missed. And users can delete calls from list.

Call History All

Index	Type	Date	Time	Local Identity	Name	Number	
1	Received	2018-09-30	08:28:46	192.168.35.10@192.168.35.10	192.168.35.68	192.168.35.68@192.168.35.68	
2	Received	2018-09-30	08:26:40	192.168.35.10@192.168.35.10	192.168.35.68	192.168.35.68@192.168.35.68	

Figure 4.6.1 Call log

4.6.2. Door Log

Go to **Phone - Door Log**, users can see a list of door log which records card information and date.

Door Log

Index	Name	Code	Type	Date	Time	Status	
1	Courier	FFB59828	Card	2018-09-30	10:49:19	Failed	
2	unKnown	1FEDBA28	Card	2018-09-30	10:49:16	Failed	
3	Courier	FFB59828	Card	2018-09-30	10:49:09	Failed	
4							
5							
6							
7							
8							
9							
10							
11							
12							
13							
14							
15							

Page 1 Prev Next Delete Delete All

Figure 4.6.2 Door log

.

4.6.3. System Log

Go to **Upgrade** - **Advanced** to configure system log level and export system log file.

System log level: From level 0 to 7.The higher level means the more specific system log is saved to a temporary file. It's level 3 by default.

Export Log: Click to export temporary system log file to local PC.

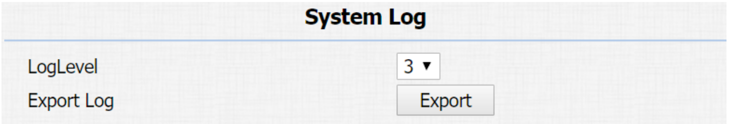


Figure 4.6.3 System log

4.6.4. PCAP

Go to **Upgrade** - **Advanced** to start, stop packets capturing or to export captured packet file.

Start: To start capturing all the packets file sent or received from phone.

Stop: To stop capturing packets.

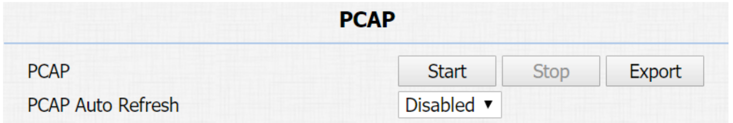


Figure 4.6.4 PCAP

Abbreviations

ACS: Auto Configuration Server

Auto: Automatically

AEC: Configurable Acoustic and Line Echo Cancelers

ACD: Automatic Call Distribution

Autop: Automatical Provisioning

AES: Advanced Encryption Standard

BLF: Busy Lamp Field

COM: Common

CPE: Customer Premise Equipment

CWMP: CPE WAN Management Protocol

DTMF: Dual Tone Multi-Frequency

DHCP: Dynamic Host Configuration Protocol

DNS: Domain Name System

DND: Do Not Disturb

DNS-SRV: Service record in the Domain Name System

FTP: File Transfer Protocol

GND: Ground

HTTP: Hypertext Transfer Protocol

HTTPS: Hypertext Transfer Protocol Secure

IP: Internet Protocol

ID: Identification

IR: Infrared

LCD: Liquid Crystal Display

LED: Light Emitting Diode

MAX: Maximum

POE: Power Over Ethernet

PCMA: Pulse Code Modulation A-Law

PCMU: Pulse Code Modulation μ-Law

PCAP: Packet Capture

PNP: Plug and Play

RFID: Radio Frequency Identification

RTP: Real-time Transport Protocol

RTSP: Real Time Streaming Protocol

MPEG: Moving Picture Experts Group

MWI: Message Waiting Indicator

NO: Normal Opened

NC: Normal Connected

NTP: Network Time Protocol

NAT: Network Address Translation

NVR: Network Video Recorder

ONVIF: Open Network Video Interface Forum

SIP: Session Initiation Protocol

SNMP: Simple Network Management Protocol

STUN: Session Traversal Utilities for NAT

SNMP: Simple Mail Transfer Protocol

SDMC: SIP Devices Management Center

TR069: Technical Report069

TCP: Transmission Control Protocol

TLS: Transport Layer Security

TFTP: Trivial File Transfer Protocol

UDP: User Datagram Protocol

URL: Uniform Resource Locator

VLAN: Virtual Local Area Network

WG: Wiegand

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