R23C

R23P

R23 Series Door Phone Admin Guide

About This Manual

Thank you for choosing Akuvox's R23 series door phone. This manual is intended for end users, who need to properly configure the door phone. This manual is applicable to 26.0.3.xx version, and it provides all functions' configurations of R23C/P. 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

1.1. Product Description 1

1.2. Connector Introduction 2

1.3. LED Status Information 3

2. Daily Use 4

2.1. Make a Call 4

2.2. Receive a Call 4

2.3. Unlock by RFID Card (Optional) 4

3. Basic Features 5

3.1. Access the Website Setting 5

3.1.1. Obtain IP Address 5

3.1.2. Access the Device Website 5

3.2. Password Modification 6

3.2.1. Modify the Web Password 6

3.2.2. Session Time Out 6

3.3. Phone Configuration 7
3.3.1. Time/Lang 7
3.3.2. Network 7
3.3.2.1. DHCP Mode 7
3.3.2.2. Static IP Mode 8
3.3.2.3. Local RTP 8
3.3.2.4. SNMP 9
3.3.2.5. VLAN 9
3.3.2.6. TR069 10
3.3.3. Sound 11
3.4. Intercom Call 12
3.4.1. Direct IP Call 12
3.4.2. SIP Call 13
3.4.2.1. SIP Account 13
3.4.2.2. SIP Server 1&2 14
3.4.2.3. Outbound Proxy Server 15

3.4.2.4. Transport Type 15
3.4.2.5. NAT 15
3.4.3. Auto Answer 16
3.4.4. Web Call 16
3.4.5. No Answer Call 17
3.4.6. Multicast 17
3.4.7. Push To Hang Up 17
3.5. Access Control 18
3.5.1. Relay 18
3.5.2. Card Setting (Optional) 19
3.5.3. Open Relay via HTTP 20
3.5.4. Unlock via Exit Button 21
3.6. Reboot 22
3.7. Reset 22
4. Advance Feature 22
4.1. Phone Configuration 22

4.1.1. LED 22
4.1.2. IR LED 24
4.1.3. RF Card Code Display Related 24
4.2. Intercom 25
4.2.1. Call Time Related 25
4.2.2. Return Code When Refuse 25
4.2.3. SIP Call Related 26
4.2.4. Codec 27
4.2.5. DTMF 28
4.2.6. Session Timer 28
4.2.7. Encryption 28
4.2.8. NAT 29
4.2.9. User Agent 29
4.3. Access Control 30
4.3.1. Web Relay 30
4.4. Security 31

4.4.1. Anti-alarm 31
4.4.2. Motion 32
4.4.3. Action 32
4.4.3.1. Action Parameters 33
4.4.3.2. No Answer Action 34
4.4.3.3. Push Button Action 34
4.4.3.4. Input Interface Triggered Action 35
4.4.3.5. Motion Triggered Action 36
4.4.3.6. Action URL 36
4.5. Upgrade 37
4.5.1. Web Upgrade 37
4.5.2. Autop Upgrade 37
4.5.3. Backup Config File 40
4.5.4. DHCP Option 40
4.6. Log 41
4.6.1. Call log 41

4.6.2. Door Log 41
4.6.3. System Log 41
4.6.4. PCAP 42

1. Product Overview

1.1. Product Description

Akuvox R23X is a SIP-compliant, hands-free one button audio outdoor phone. It can be connected with users Akuvox indoor monitors for remote access controlling and monitoring. Users can operate the indoor phone to communicate with visitors via voice, and use RFID cards to unlock the door (R23C only). It's applicable in villas, offices and so on.

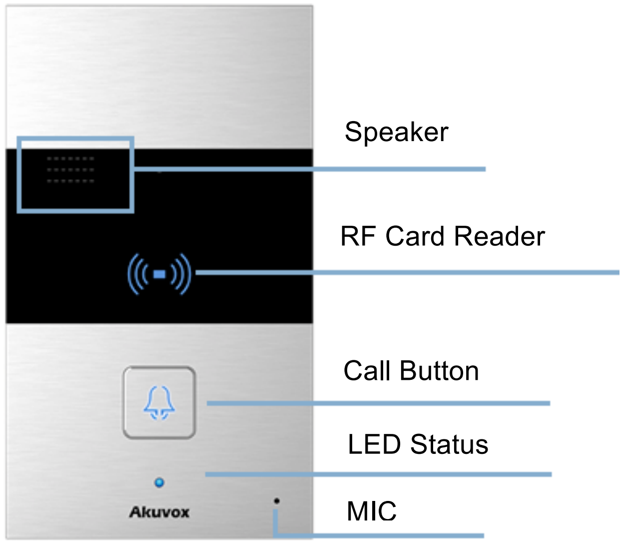


Figure 1.1 Product description

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 is not available.

RS485A/B: RS485 terminal.

DOORA/B: Trigger signal input terminal.

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

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

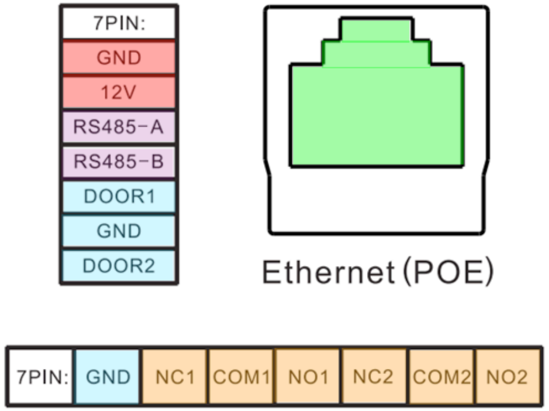


Figure 1.2-1 R23's interface

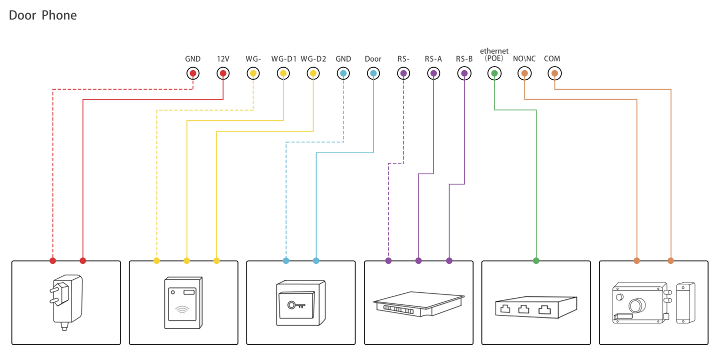


Figure 1.2-2 General interface

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. Make a Call

Press the call button to dial out the predefined number or IP address.

If LED turns green, it means the call has been answered.

2.2. Receive a Call

Users can use phone or indoor monitor to call R23X and R23X will answer it automatically by default. If auto answer function is disabled, pressing call button to answer incoming call.

2.3. Unlock by RFID Card (Optional)

Place the predefined RFID card on the card reader. The door phone will announce “the door is now opened” and unlock the door.

13.56MHz RF card is supported on R23C.

3. Basic Features

3.1. Access the Website Setting

3.1.1. Obtain IP Address

While R23X power up normally, hold the call button for several seconds after the statue LED turns blue and it will enter IP announcement mode. In announcement mode, the IP address will be announced periodically and “IP 0.0.0.0” would be announced if no IP address is obtained. 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

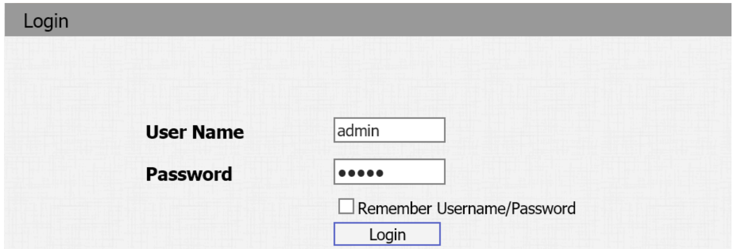


Figure 3.1.2 Access the device website

administrator user name and password are shown below**:**

User Name**: admin**

Password**: admin**

3.2. Password Modification

Go to **Security - Basic** to modify password and session time.

3.2.1. Modify the Web Password

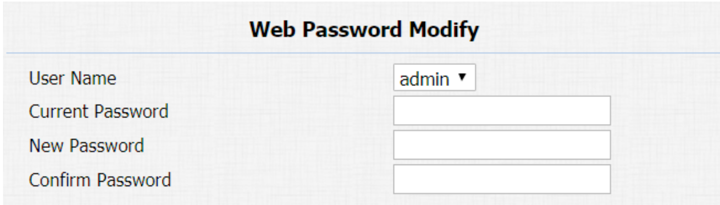


Figure 3.2.1 Modify the web password

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

3.2.2. Session Time Out

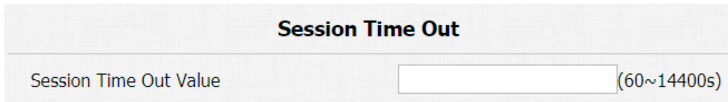


Figure 3.2.2 Session time out

To configure session time out value. Over the value, users need to login again to continue configuring.

3.3. Phone Configuration

3.3.1. Time/Lang

Go to **Phone - Time/Lang** to configure it.

Time Zone: To select local time zone for NTP server.

Primary Server: To configure primary NTP server address. ✦

Secondary Server: To configure secondary NTP server address, it takes effect if primary NTP server is unreachable. ✦

Update Interval: To configure interval between two consecutive NTP requests.

System Time: The current time of the phone.

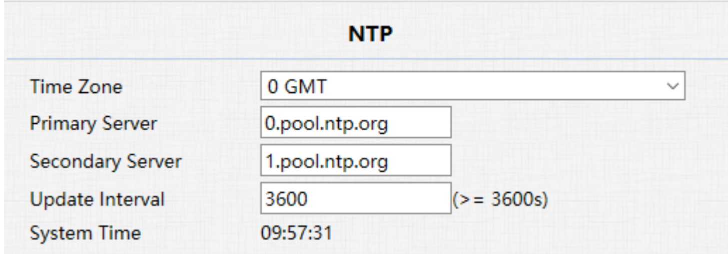


Figure 3.3.1 Time

3.3.2. Network

3.3.2.1. DHCP Mode

Go to **Network - Basic**.

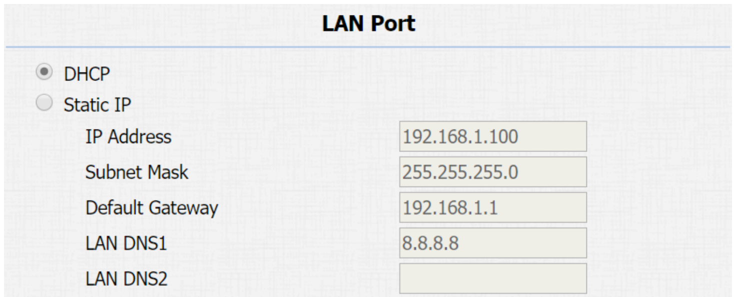


Figure 3.3.2.1 DHCP mode

R23X uses DHCP by default, and it will obtain IP address, subnet mask, default gateway and DNS server address from DHCP server automatically.

3.3.2.2. Static IP Mode

Go to **Network - Basic** to configure.

If selected, users could manually set IP address, subnet mask, default gateway and DNS server. The figure below shows static IP setting.

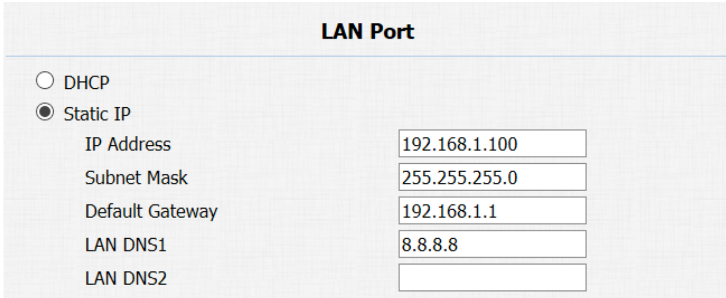


Figure 3.3.2.2 Static IP mode

3.3.2.3. Local RTP

Go to **Network - Advanced** to configure. To display and configure Local RTP settings.

Max RTP Port: Determine the maximum port that RTP stream can use.

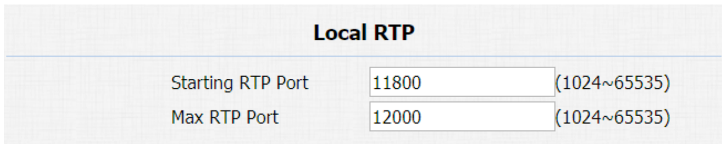


Figure 3.3.2.3 Local RTP

Starting RTP Port: Determine the minimum port that RTP stream can use.

3.3.2.4. SNMP

Go to **Network - Advanced** to configure. To display and configure SNMP settings. ✦

Active: To enable or disable SNMP feature. ✦

Port: To configure SNMP server's port. ✦

Trusted IP: To configure allowed SNMP server address, and it could be an IP address or any valid URL domain name.

Note: SNMP (Simple Network Management Protocols) is Internet-standard protocol for managing devices on IP networks.

Figure 3.3.2.4 SNMP

3.3.2.5. VLAN

Go to **Network - Advanced** to configure. To display and configure VLAN settings. ✦

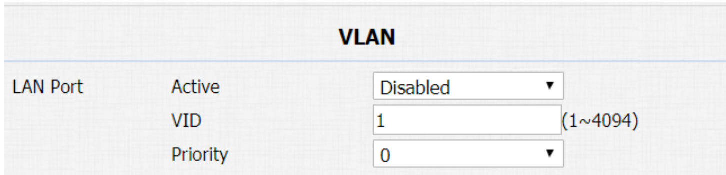


Figure 3.3.2.5 VLAN

Active: To enable or disable VLAN feature for designated port.

VID: To configure VLAN ID for designated port.

Priority: To select VLAN priority for designated port.

Note: Please consult users administrator for specific VLAN settings in your networking environment.

3.3.2.6. TR069

Go to **Network - Advanced** to configure. To display and configure TR069 settings.

Active: To enable or disableTR069 feature.

Version: To select supported TR069 version (version 1.0 or 1.1).

ACS/CPE: ACS is short for auto configuration servers as server side, CPE is short for customer-premise equipment as client side devices.

URL: To configure URL address for ACS or CPE.

User Name: To configure username for ACS or CPE.

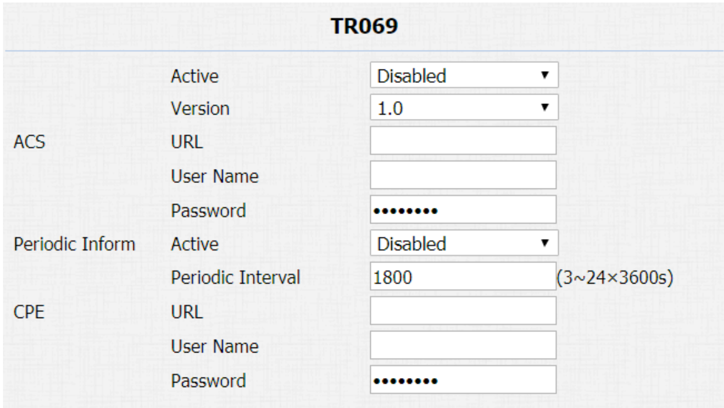


Figure 3.3.2.6 TR069

Password: To configure Password for ACS or CPE. ✦

Periodic Inform: To enable periodically inform. ✦

Periodic Interval: To configure interval for periodic inform.

Note: TR-069 (Technical Report 069) is a technical specification entitled CPE WAN Management Protocol (CWMP). It defines an application layer protocol for remote management of end-user devices.

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, users will not hear the prompt voice when the door is opened.

Figure 3.3.3-1 Sound

IP Announcement: To setup the IP announcement active time. Over the configured value, the phone will not announce the IP when users hold the button.

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

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

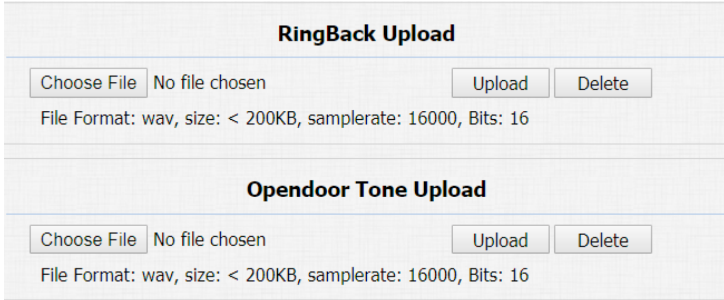


Figure 3.3.3-2 Sound

3.4. Intercom Call

3.4.1. Direct IP Call

Figure 3.4.1-1 Direct IP call

Without sip server, users can also use IP address to call each other, but this way is only suitable in the LAN.

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.

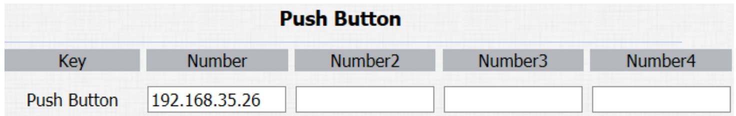


Figure 3.4.1.1 Push button

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

If you would like to call multiple numbers at the same time, divide them by semicolon.

Note: The push button number can also enter the SIP account.

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 phone first. Then press the push button to make SIP call.

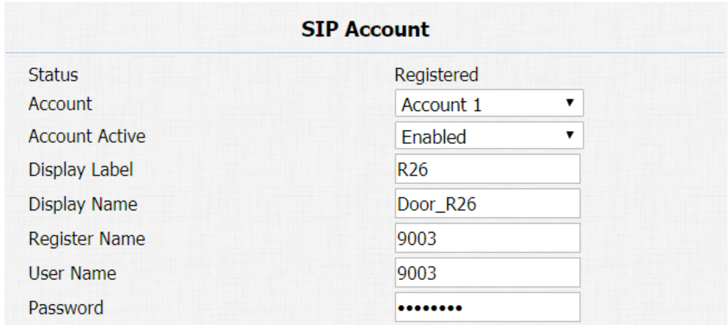


Figure 3.4.2.1 SIP account

3.4.2.1. SIP Account

Status: To display register result.

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 you 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.

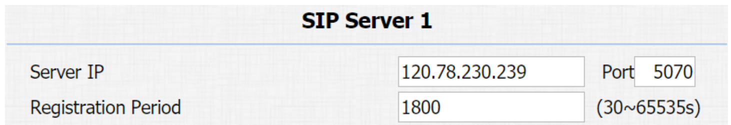


Figure 3.4.2.2-1 SIP server 1&2

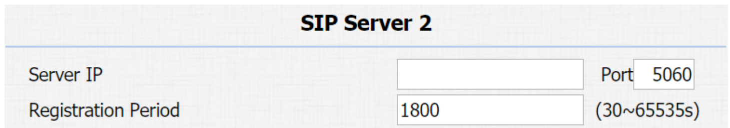


Figure 3.4.2.2-2 SIP server 1&2

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.

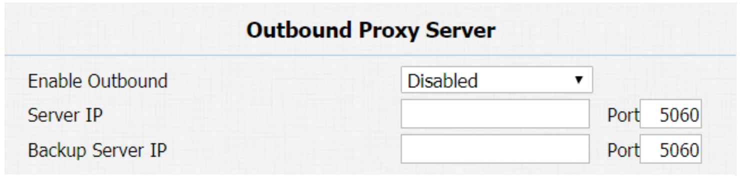


Figure 3.4.2.3 Outbound proxy server

3.4.2.4. Transport Type

To display and configure transport type for SIP message.

- 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.

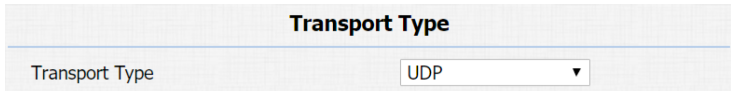


Figure 3.4.2.4 Transport type

3.4.2.5. NAT

To display and configure NAT (Net Address Translator) settings.

- STUN: Short for simple traversal of UDP over NATs, a solution to solve NAT issues.

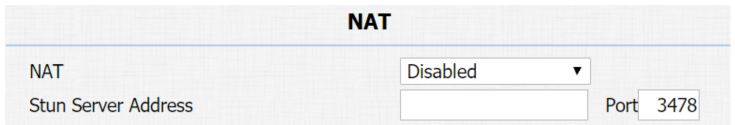


Figure 3.4.2.5 NAT

Note: By default, NAT is disabled.

3.4.3. Auto Answer

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

Figure 3.4.3-1 Auto answer

Go to **Phone - Call Feature** to enable auto answer feature for direct IP call without SIP proxy.

Figure 3.4.3-2 Auto answer

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

Auto Answer Mode: To set audio mode for auto answer by default. Then incoming call will be answered automatically.

Figure 3.4.3-3 Auto answer

3.4.4. Web Call

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

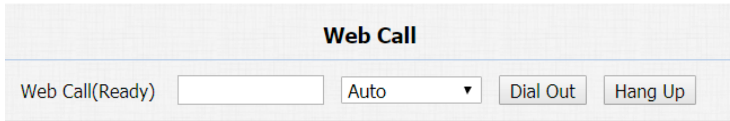


Figure 3.4.4 Web call

3.4.5. No Answer Call

Go to **Intercom - Basic** and enable the no answer call.

Go to **Intercom - Basic** and set the no answer call number.

Figure 3.4.5-1 No Answer call

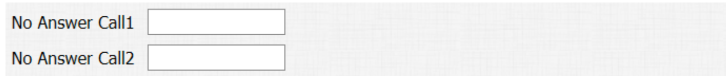


Figure 3.4.5-2 No answer call

3.4.6. Multicast

Go to **Intercom - Multicast** to configure.

Paging Barge: Choose the multicast number, the range is 1-10.

Paging priority Active: Enable to disable the multicast.

Listening Address: Enter the IP address users need to listen.

Label: Input the label for each listening address.

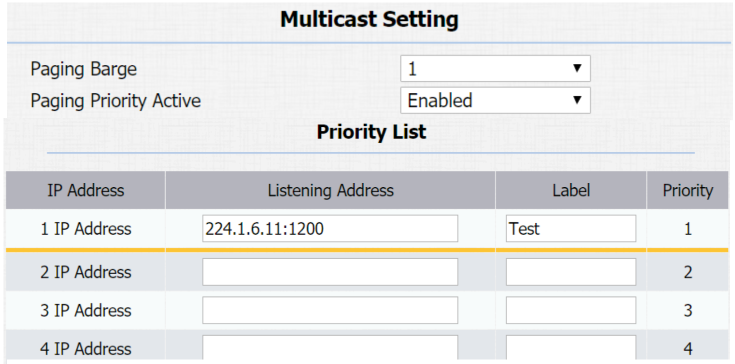


Figure 3.4.6 Multicast

3.4.7. Push To Hang Up

Go to **Intercom - Basic** to configure. To enable or disable pushing button to hang up.

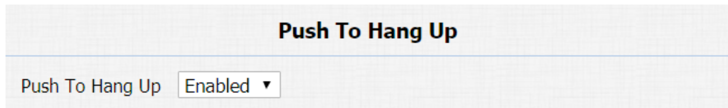


Figure 3.4.7 Push to hang up

3.5. Access Control

3.5.1. Relay

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

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: R23X supports two relays, users 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.

DTMF Option: To select digit of DTMF code, R23X supports maximum 4 digits DTMF code.

DTMF: To configure 1 digit DTMF code for remote unlock.

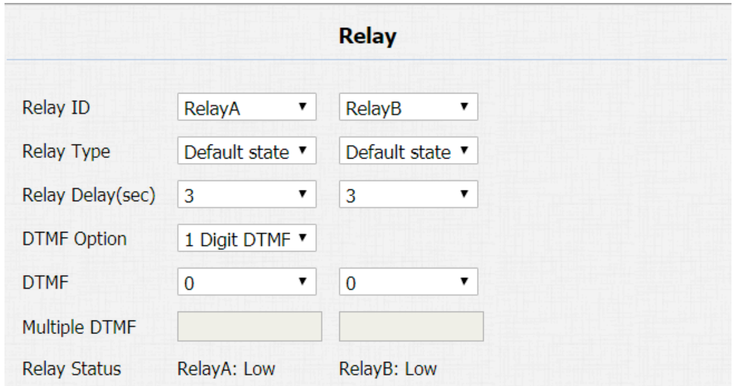


Figure 3.6.1 Relay

Multiple DTMF: To configure multiple digits DTMF code for remote unlock.

Relay Status: Low means that COM is connecting to NC while High means that COM is connecting to NO.

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

3.5.2. Card Setting (Optional)

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

Import/Export Card Data

R23X 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 template RFID cards data file.

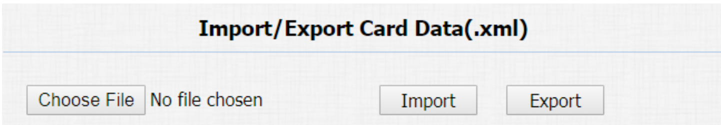


Figure 3.6.2-1 Card setting

Obtain and Add Card

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

Note: Users can use card to access only when card status has been switched to “Normal”.

Door Card Management

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

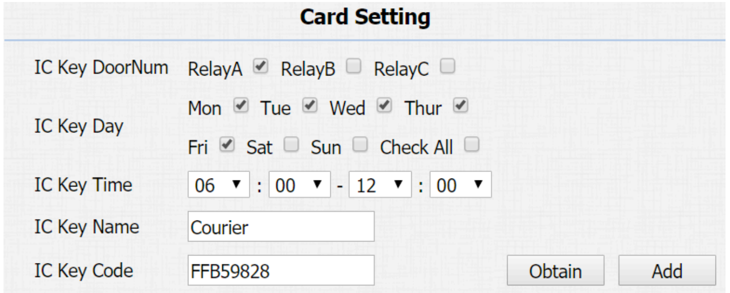


Figure 3.6.2-2 Card setting

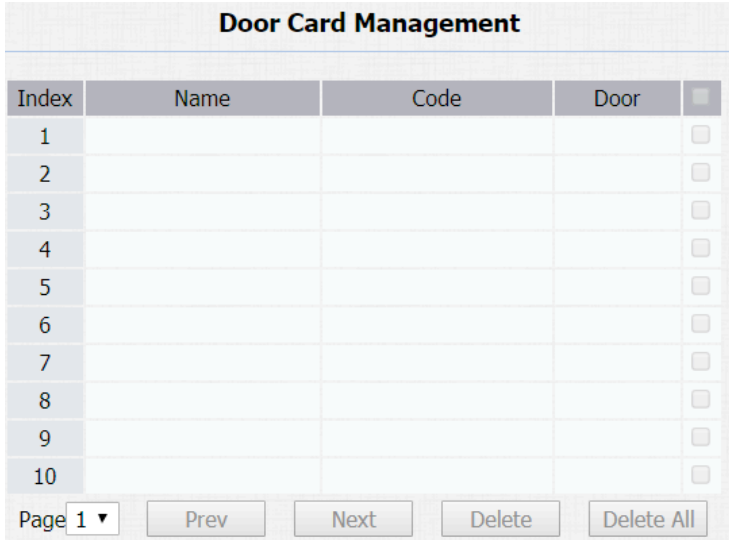


Figure 3.6.2-3 Card setting

3.5.3. Open Relay via HTTP

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.

URL format:

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

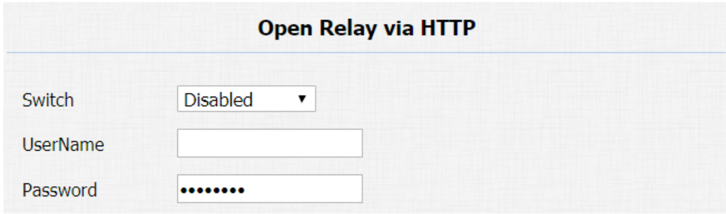


Figure 3.6.3 Open relay via HTTP

3.5.4. Unlock via Exit Button

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

R23X 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.

Action to execute: To choose which action to execute after the input terminal is triggered.

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

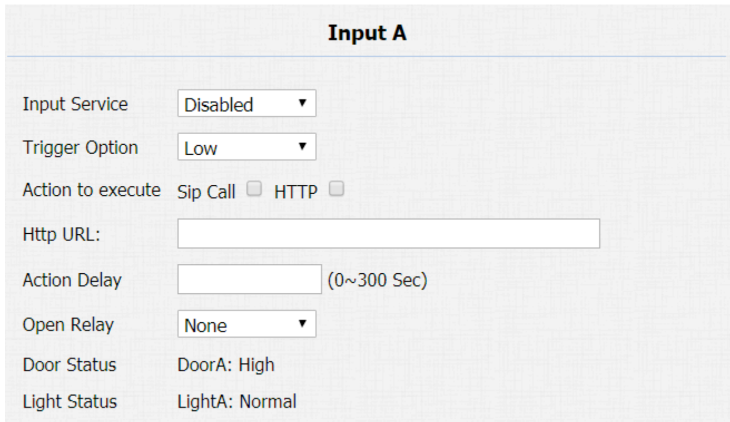


Figure 3.6.4-1 Unlock via exit button

Open Relay: To configure relay to open.

Door Status: To show the status of input signal.

3.6. Reboot

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

Figure 3.7 Reboot

3.7. Reset

Go to **Upgrade - Basic**, users can reset to factory setting.

Figure 3.8 Reset

4. Advance Feature

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.

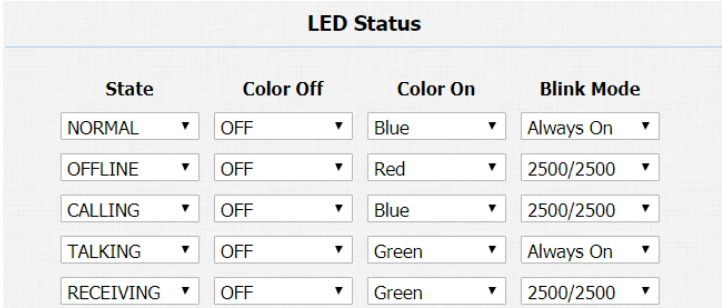
LED Control:

Use HTTP URL to remote control the LED status.

Http format:

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

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



LED Status

State	Color Off	Color On	Blink Mode
NORMAL	OFF	Blue	Always On
OFFLINE	OFF	Red	2500/2500
CALLING	OFF	Blue	2500/2500
TALKING	OFF	Green	Always On
RECEIVING	OFF	Green	2500/2500

Figure 4.1.1-1 LED

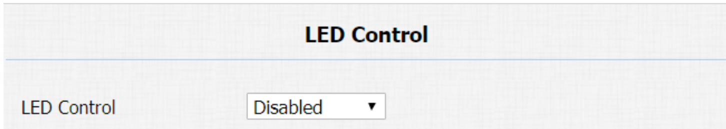


Figure 4.1.1-2 LED

4.1.2. IR LED

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

Photoresistor: The setting is for night vision, when the surrounding of R23X is very dark, infrared LED will turn on and R23X 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.

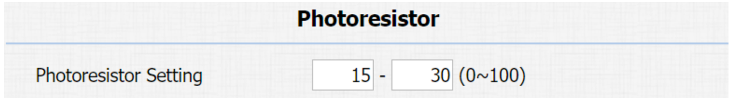


Figure 4.1.2 IR LED

4.1.3. RF Card Code Display Related

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

RFID Display Mode: To be compatible different card number formats. The default 8HN means hexadecimal.

Figure 4.1.3 RF card code display related

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.

Hang Up After Open Door: To set the time that hang up the call after open the door.

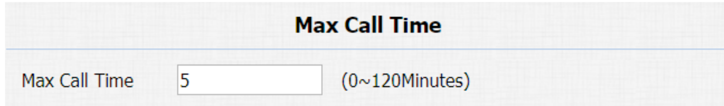


Figure 4.2.1-1 Call time related

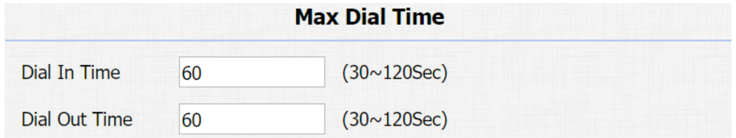


Figure 4.2.1-2 Call time related

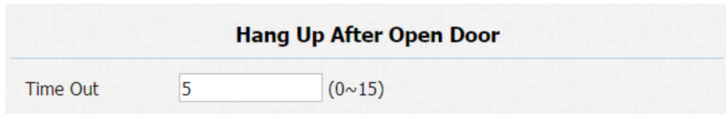


Figure 3.4.8 Hang up after open door

4.2.2. Return Code When Refuse

Go to **Phone - Call Feature - Others** 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, R23X 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 messages from hacking.

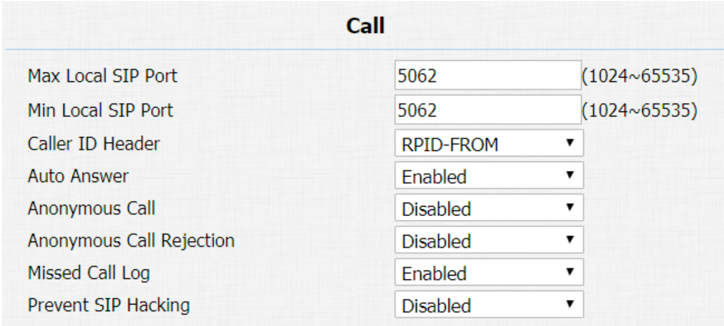


Figure 4.2.3-1 SIP call related

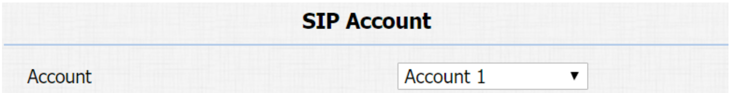


Figure 4.2.3-2 SIP call related

4.2.4. Codec

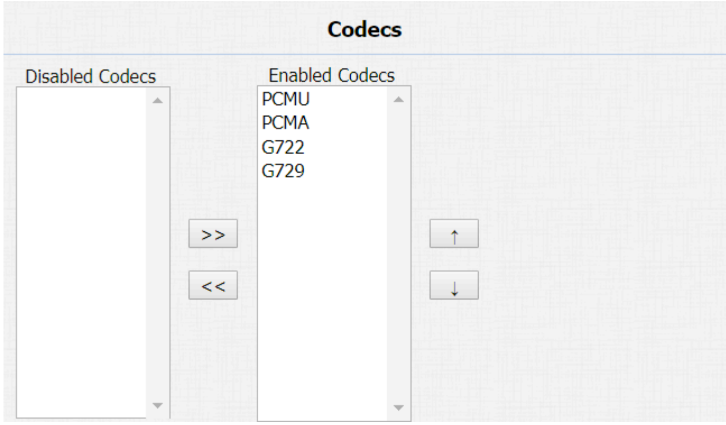


Figure 4.2.4-1 Codec

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

SIP Account: To choose which account to configure.

Audio Codec: R23X support four audio codec**:** PCMA, PCMU, G729, G722. Different audio codec requires 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

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

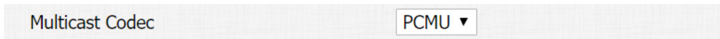


Figure 4.2.4-3 Codec

4.2.5. DTMF

Go to **Account - Advanced** to configure RTP audio profile for DTMF and its payload type.

Type: Support Inband, Info, RFC2833 or their combination.

How To Notify DTMF: Only available when DTMF type is Info.

DTMF Payload: To configure payload type for DTMF.

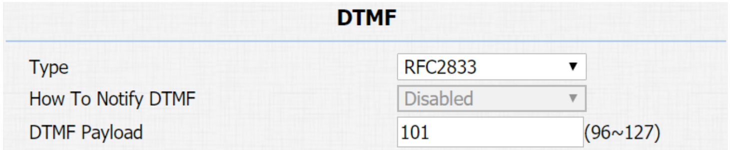


Figure 4.2.5 DTMF

4.2.6. Session Timer

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

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

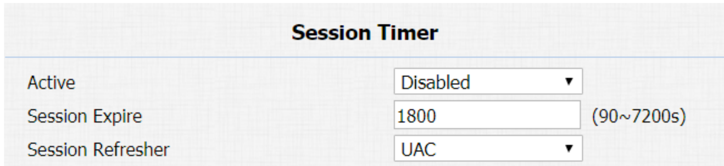


Figure 4.2.6 Session timer

4.2.7. Encryption

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

If enabled, voice will be encrypted.

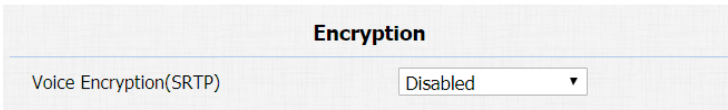


Figure 4.2.7 Encryption

4.2.8. NAT

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

UDP Keep Alive message: If enabled, R23X will send UDP keep-alive message periodically to router to keep NAT port alive.

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.

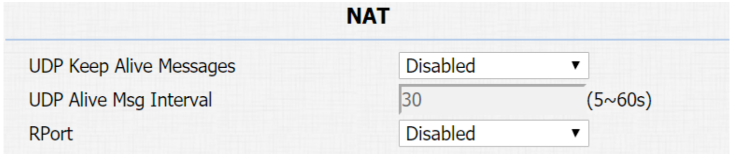


Figure 4.2.8 NAT

4.2.9. User Agent

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

To customize user agent field in the SIP message.

If users agent is set to specific value, users could see the information from network package. If user agent is not set by default, users could see the company name, model number and firmware version from network package.

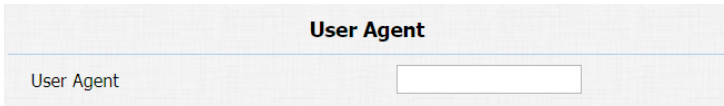


Figure 4.2.9 User agent

4.3. Access Control

4.3.1. Web Relay

R23X can support extra web relay which is connected with the door phone via network.

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

Type: Connect web relay and choose the type.

IP Address: Enter web relay's IP address.

UserName: 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 same as the local relay, the web relay will be open with local relay. But if there are different, the web relay is invalid.

Figure 4.3.1-1 Web relay

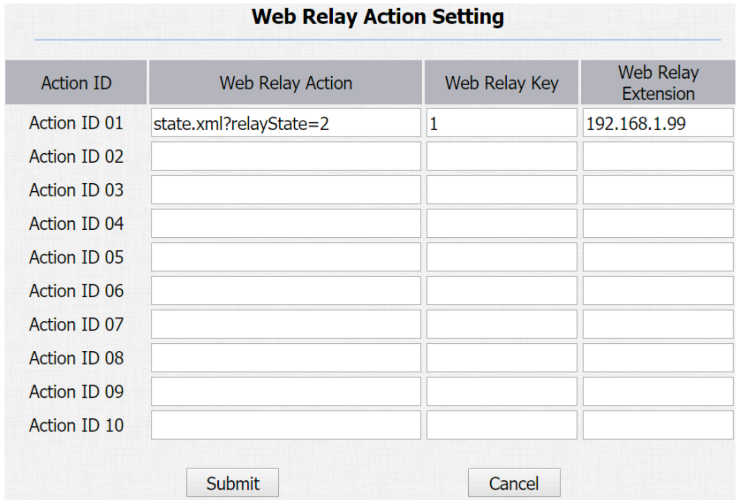


Figure 4.3.1-2 Web relay

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

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.

R23X integrates internal gravity sensor for the own security, and after enabling tamper alarm, if the gravity of R23X changes dramatically, the phone will alarm. Gravity sensor threshold stands for sensitivity of sensor.

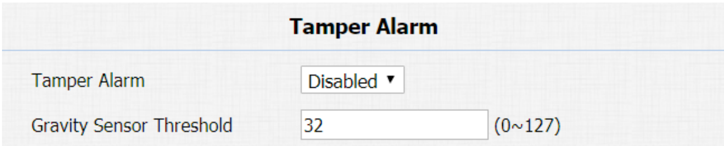


Figure 4.4.1 Anti-alarm

4.4.2. Motion

R23X supports motion detection, go to **Intercom - Motion** to configure detection parameter.

Motion Detection: To enable or disable motion detection

Motion Delay: To configure minium time gap between two snapshots.

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

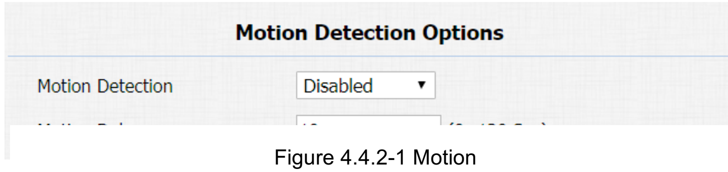


Figure 4.4.2-1 Motion

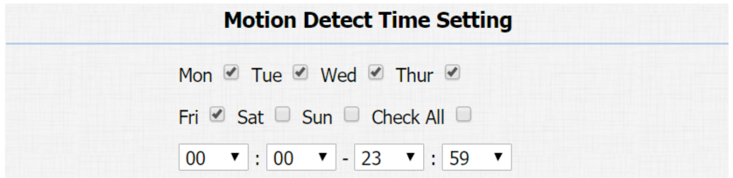


Figure 4.4.2-2 Motion

4.4.3. Action

R23X 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.

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

SMTP password: To configure password of SMTP service (usually it is 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.

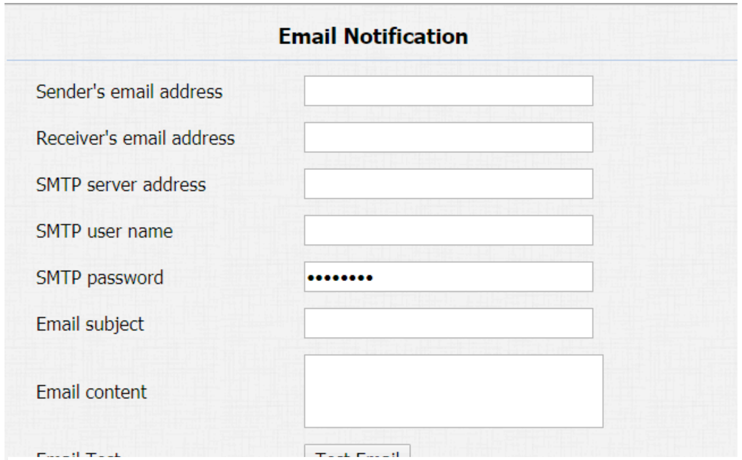


Figure 4.4.3.1-1 Action parameters

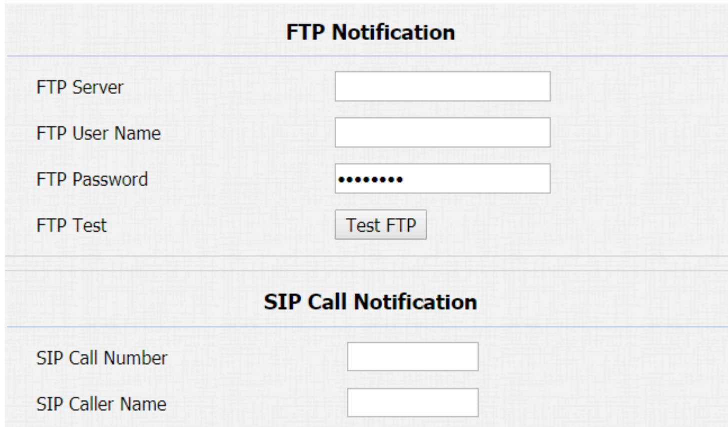


Figure 4.4.3.1-2 Action parameters

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 R23X.

4.4.3.2. No Answer Action

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

No Answer Action: For sending the notification to specified email if the call is not answered.

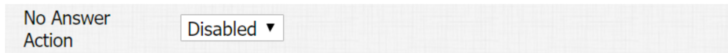


Figure 4.4.3.2 No answer action

4.4.3.3. Push Button Action

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

Enable this function, the device will record any changes of the surrounding environment then send the message or picture to the corresponding receiver.

Action to execute: Tick the suit the suitable way to receive the action message.

HTTP URL: If you tick HTTP URL, and then enter the HTTP server IP address in the HTTP URL area. When the device detects any changes, it will send HTTP network package.

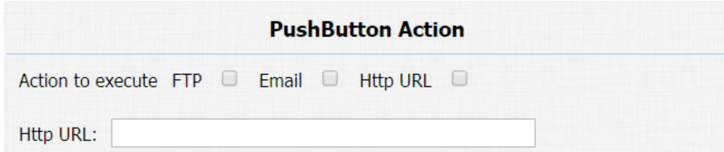


Figure 4.4.3.3 PushButton action

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.

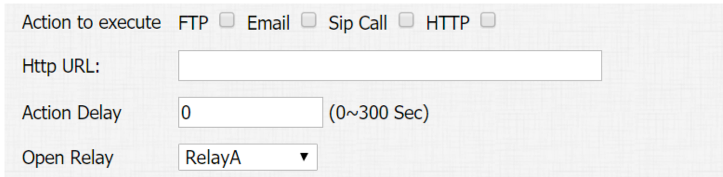


Figure 4.4.3.4 Input interface trigger action

Open Relay: To configure which relay to trigger.

4.4.3.5. Motion Triggered Action

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

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

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

SDMC Upload: Upload the capture to the SDMC.

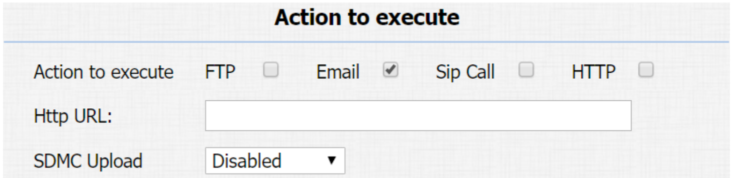


Figure 4.4.3.5 Motion trigger action

4.4.3.6. Action URL

Action URL can be triggered by some predefined incidents.

Go to **Phone - Action URL**, pick **Active** to be “Enabled”, pick to demand triggered incident, each “HTTP” request to have to including the key and value, use “=” to separate，each value staring with “$.” For example, “**Open Relay Success**” incident, input **http://server IP address/help.xml?mac=$mac**, when the relay of

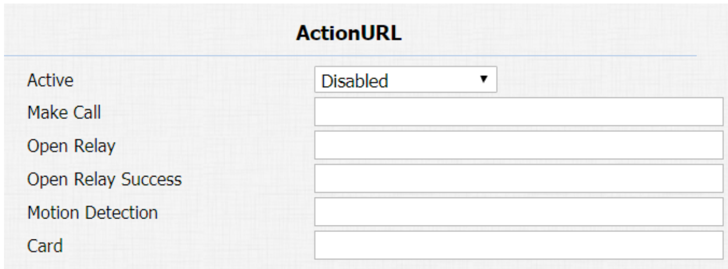


Figure 4.4.3.6 Action URL

R23X is triggered successfully, the phone will send a HTTP packet to the server, through the HTTP package to know the MAC of the phone.

4.5. Upgrade

4.5.1. Web Upgrade

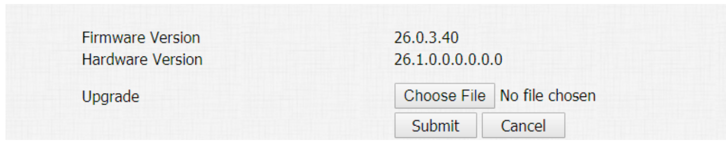


Figure 4.5.1 Web update

Go to **Upgrade - Basic**, users can upgrade firmware. Reset to factory setting and reboot.

Upgrade: Choose .rom firmware from the PC, and then click **Submit** to start update.

4.5.2. Autop Upgrade

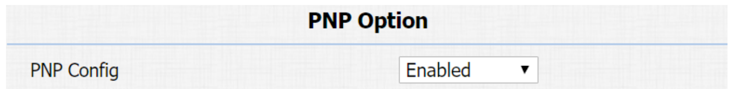


Figure 4.5.2-1 Autop update

Go to **Upgrade - Advanced** to configure automatically update server's settings.

PNP Option

Plug and Play, once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get auto provisioning server's address.

By default, this SIP message is sent to multicast address 224.0.1.75 (PNP server address by standard).

Manual Autop

Autop (Auto-Provisioning) is a centralized and unified upgrade of telephone. It is a simple and time-saving configuration for phone. It is mainly used by the device to download corresponding configuration document from the server using TFTP / FTP / HTTP / HTTPS network protocol. To achieve the purpose of updating the device configuration, making the users to change the phone configuration more easily. This is a typical C/S architecture upgrade mode, mainly by the terminal device or PBX server to initiate an upgrade request.

URL: Auto provisioning server address.

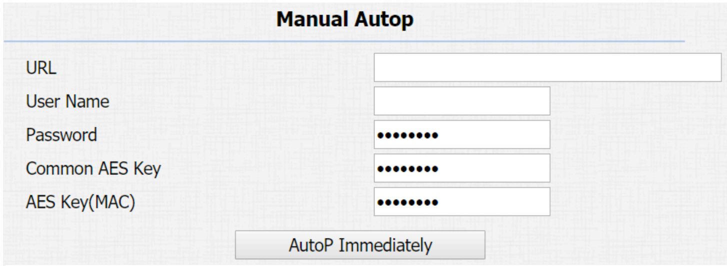


Figure 4.5.2-2 Autop update

User Name: Configure if server needs an username to access, otherwise left blank.

Password: Configure if server needs a password to access, otherwise left blank.

Common AES Key: Used for phone to decipher common auto provisioning configuration file.

AES Key (MAC): Used for phone to decipher MAC-oriented auto provisioning configuration file (for example, file name could be 0C1105888888.cfg if phone's MAC address is 0C1105888888).

Note: AES is one of many encryption, it should be configured only when configure file is ciphered with AES, otherwise left blank.

Automatic Autop

To display and configure auto provisioning mode settings.

This auto provisioning mode is actually self-explanatory.

For example, mode "Power on" means phone will go to do provisioning every time it powers on.

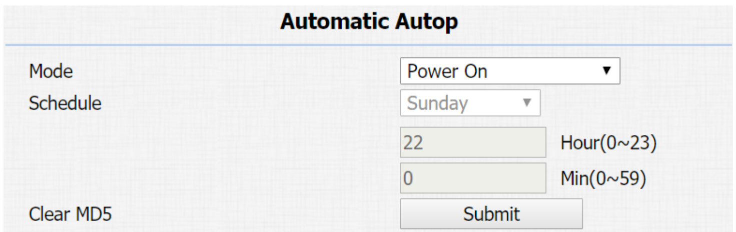


Figure 4.5.2-3 Autop update

Note: Please refer to the related feature guide from Akuvox forum.

4.5.3. Backup Config File

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

Export Autop Template: To export current config file.

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

Figure 4.5.3-1 Backup config file

Figure 4.5.3-2 Backup config file

4.5.4. DHCP Option

To display and configure DHCP setting for AutoP. Option 66/43 is enable by default. It can support HTTPS, HTTP, FTP, TFTP server.

Customer Option: Enter the server URL. Click “Submit” to save.

Note: To make DHCP autop URL works, the PNP should be disable.

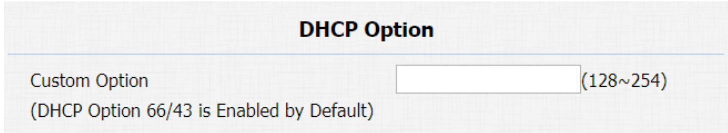
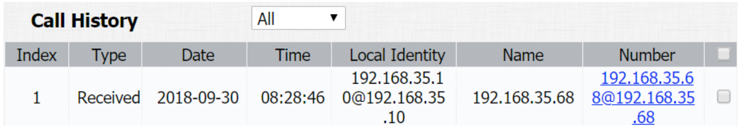


Figure 4.5.4 Backup config file



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	

Figure 4.6.1 Call log

4.6. Log

4.6.1. Call log

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

4.6.2. Door Log

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

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							

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

LogLevel	3 ▾
Export Log	Export

Figure 4.6.3 System log

System Log Level: From level from 0 to 7.The higher level means the more specific system log is saved to a temporary file. By default, it's level 3.

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

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.

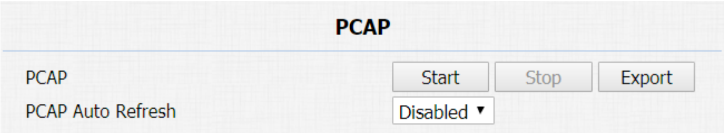


Figure 4.6.4 PCAP

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

Contact us

For more information about the product, please visit us at www.akuvox.com or feel free to contact us by

Sales email: sales@akuvox.com

Technical support email: techsupport@akuvox.com

Telephone: +86-592-2133061 ext.7694/8162

We highly appreciate your feedback about our products.