Akuvox Smart Intercom

E21A

E21V

E21 Series Door Phone Admin Guide

About This Manual

Thank you for choosing Akuvox's E21A/V door phone. This manual is intended for end users who need to properly configure the door phone. This manual is applicable to 21.0.3.xx version, and it provides all functions' configurations of E21A/V. 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 E21A/V Series are outdoor-rated, SIP-compliant and hands-free Voice over IP (VoIP) Emergency Stations. It helps the emergency teams to coordinate their rescue missions with high efficiency.

E21A/V support two types: E21A(Audio) and E21V(Video).They are often used in public locations such as: parking facilities, college campuses, medical centers, and industrial parks etc.

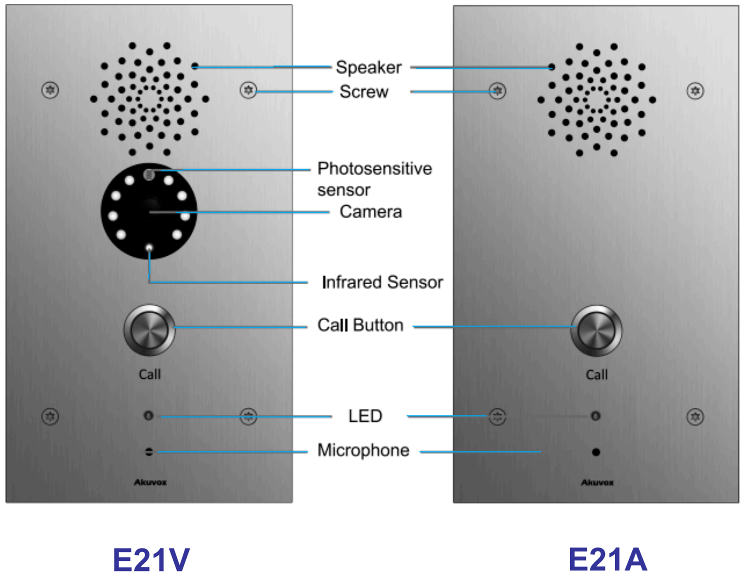

Figure 1.1 Product Description

1.2. Connector Introduction

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

(If Ethernet(POE) is used,do not plug in another power supply).

12V/GND: External power supply terminal.

Relay (NO/NC): Relay control terminal.(E21A/V supports 2 relays).

Input: Trigger signal input terminal.

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

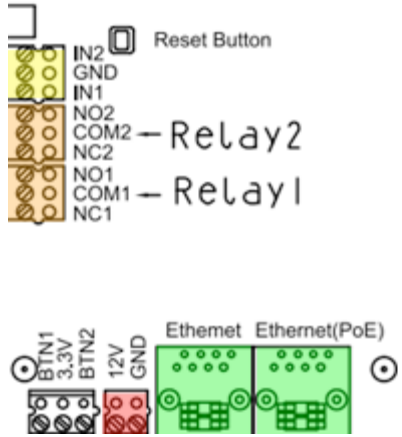

Figure 1.2-1 Connection introduction

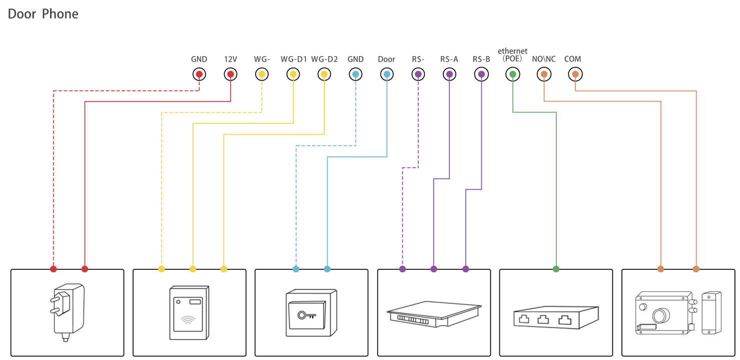

Figure 1.2-2 General interface

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 call out the predefined SIP number or IP address .Video call is only for E21V.
- During the talk,called party can press predefined DTMF code number to open the door.

2.2. Receive a Call

Auto Answer: E21A/V support auto answer by default. Incoming calls from door device (IP phone or indoor monitor) will be answered automatically.

2.3. 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 E21A/V starts up normally, hold the call button for several seconds after LED light turns blue,the voice system will be announced periodically.
- In announcement mode, the IP address will be announced periodically.
- Press the Call button again to exit the announcement mode.
- If not IP address has been obtained,E21A/V announces ‘IP 0.0.0.0’

3.1.2. Obtain IP Address

The Akuvox E21A/V series use DHCP IP address by default. If IP address is unknown, press and hold call button for a short period of time (about 5s) after LED light turns blue, E21A/V series will announce its IP continuously. Press once again to stop.

3.1.3. Access the Device Website

Open a Web Browser, access the corresponding IP address. Then, enter the default user name and password to login. The default administrator or user's User Name /Password are shown below:

Admin mode: admin/admin

User mode: user/user

Note: The recommended browser is Google Chrome

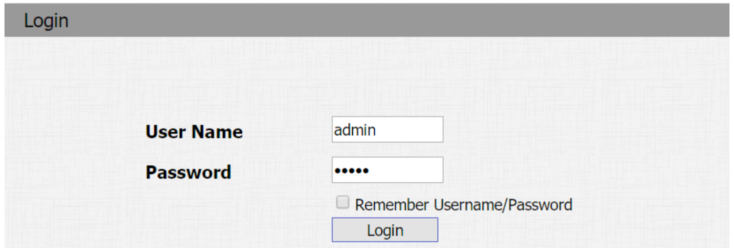

Figure 3.1.3 Access the device website

3.2. Password Modification

Go to **Security** - **Basic**, to modify the default website password - admin. Enter the original password and new password ,confirm the new one again.

Web Password Modify: change the password of login in website.

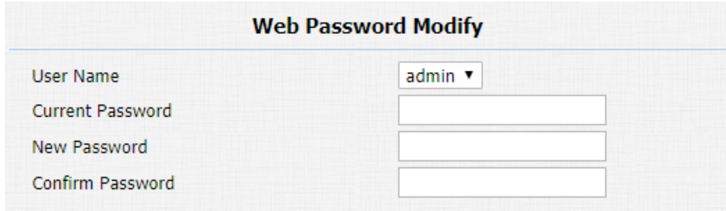

Figure 3.2 Password modification

3.3. Phone Configuration

3.3.1. Time/Lang

Time: Go to **Phone** - **Time/Lang**, to select local Time Zone for NTP server.

Lang: There are no language option for now. The web language is set to English by default.

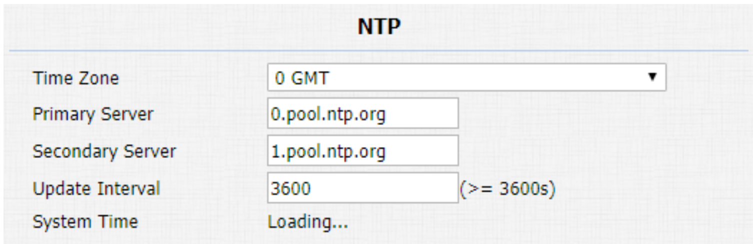

Figure 3.3.1 NTP

3.3.2. Network

3.3.2.1. IP Address Setting

Go to **Network - Basic**, choose to use DHCP or static IP way to obtain IP address.

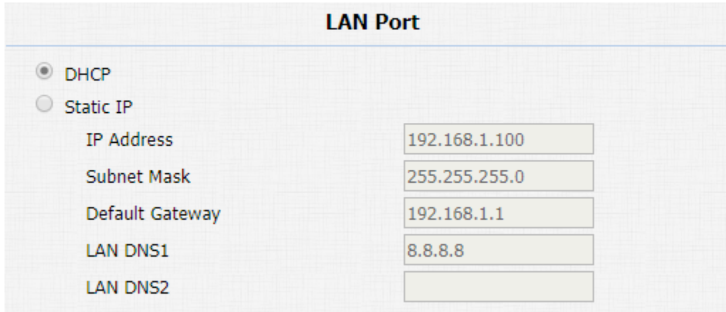

Figure 3.3.2.2 DHCP mode

3.3.2.2. DHCP Mode

In website, go to **Network** - **Basic**.

E21A/V uses DHCP mode by default which will get IP address, subnet mask, default gateway and DNS server address from DHCP server automatically.

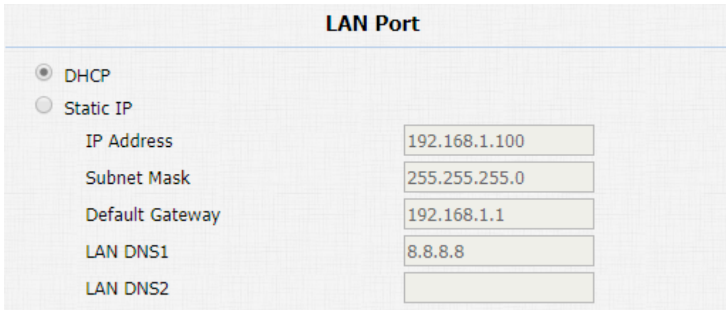

Figure 3.3.2.3 Static IP mode

3.3.2.3. Static IP Mode

In website, go to **Network** - **Basic**.

If select static IP, users should manually setup IP address, subnet mask, default gateway and DNS server address. The figure right shows static IP setting.

3.3.2.4. Local RTP

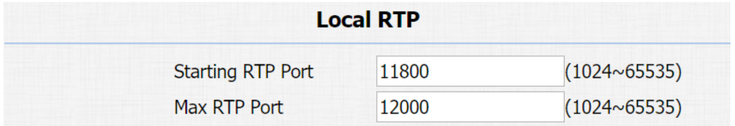

Figure 3.3.2.4 Local RTP

Go to **Network** - **Advanced** to configure.

Local RTP: To display and configure Local RTP settings.

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

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

3.3.2.5. SNMP

Figure 3.3.2.5 SNMP

Go to **Network** - **Advanced** to configure.

SNMP: 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, it could be an IP address or any valid URL domain name.

Note: SNMP is Internet-standard protocol for managing devices on IP networks.

3.3.2.6. VLAN

Go to **Network** - **Advanced** to configure.

VLAN: To display and configure VLAN settings.

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 administrator for specific VLAN settings in the networking environment.

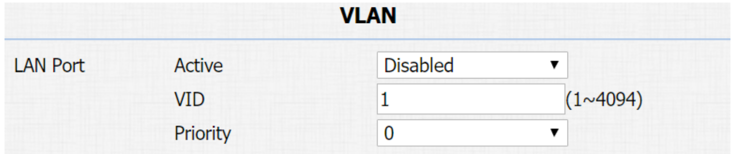

Figure 3.3.2.6 VLAN

3.3.2.7. TR069

Go to **Network** - **Advanced** to configure.

TR069: To display and configure TR069 settings.

Active: To enable or disable TR069 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.

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

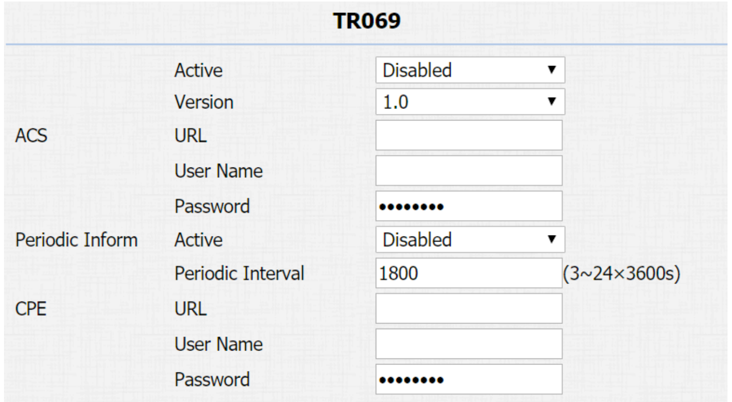

Figure 3.3.2.7 TR069

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.

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

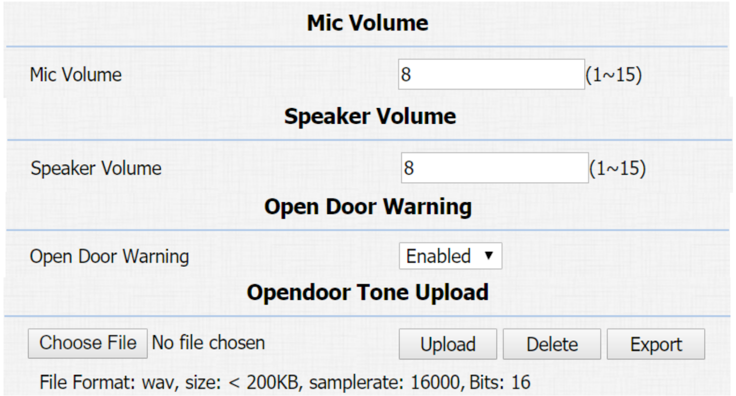

Figure 3.3.3 Sound

3.3.4. DND

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

DND: DND allows phones to ignore any incoming calls.

Return Code when DND: Determine what response code should be sent back to server when there is an incoming call if DND is on.

DND On Code: The code is used to turn on DND on server's side, if configured, door phones will send a SIP message to server to turn

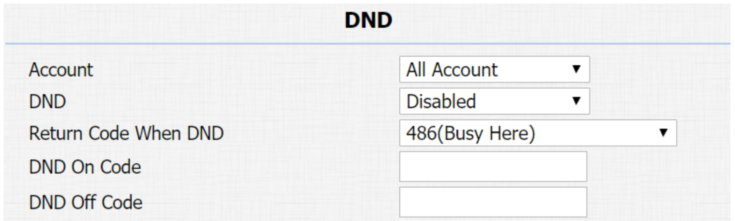

Figure 3.3.4 DND

on DND on server side if users press DND when DND is off.

DND Off Code: The code is used to turn off DND on server's side, if configured, door phones will send a SIP message to server to turn off DND on server side if users press DND when DND is on.

3.4. Intercom Call

3.4.1. Direct IP Call

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

Figure 3.4.1 Direct IP call

Configure push button with IP address, press push button to make direct ip call.

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

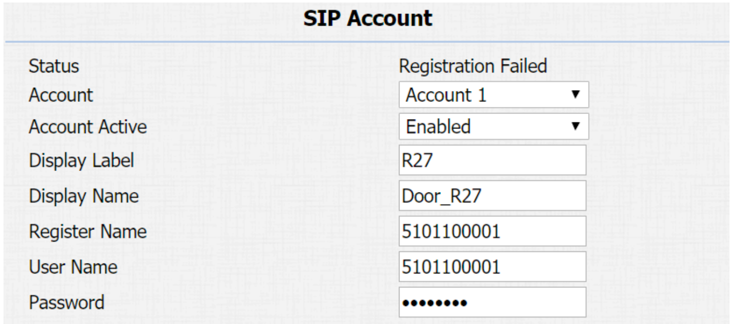

Figure 3.4.2.1 SIP account

Status: To display register result.

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

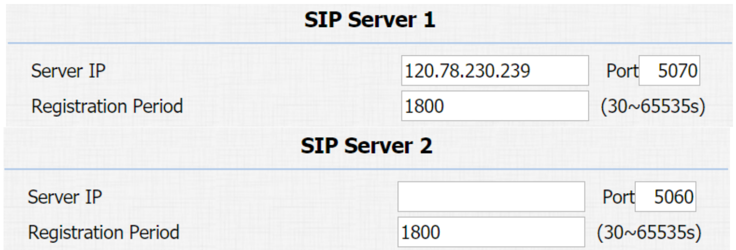

Figure 3.4.2.2 SIP server 1&2

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.

3.4.2.3. Account Selection

Go to **Intercom-Basic**:

Select Account: Select default account to make calls.

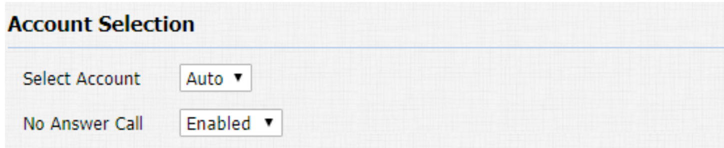

Figure 3.4.2.3 No answer call

3.4.2.4. Outbound Proxy Server

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

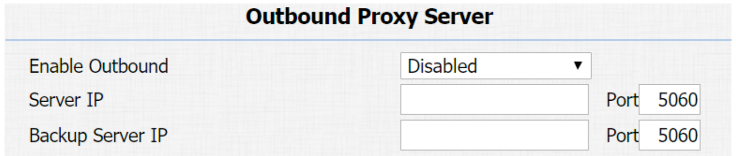

Figure 3.4.2.4 Outbound proxy server

3.4.2.5. 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.

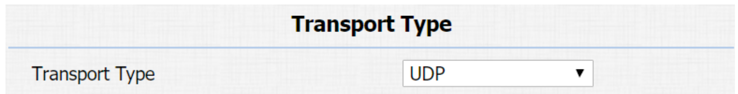

Figure 3.4.2.5 Transport type

3.4.2.6. NAT

To display and configure NAT settings.

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

Notes: By default, NAT is disabled.

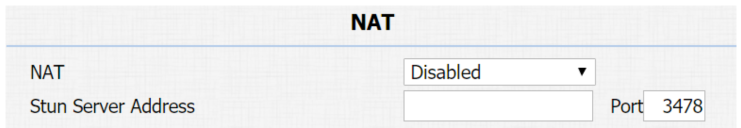

Figure 3.4.2.6 NAT

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

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

Figure 3.4.3-2 Auto answer 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.

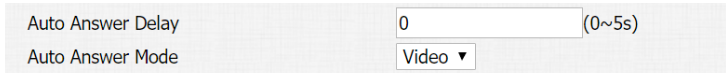

Figure 3.4.3-3 Auto answer options' parameters

3.4.4. Web Call

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

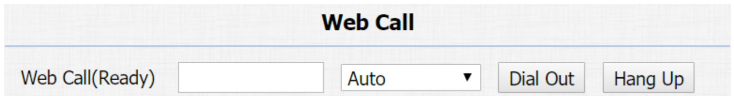

Figure 3.4.4 Web call

3.4.5. No Answer Call

No Answer Call: Enable or Disable 'No Answer Call ' feature.

If the call is not answered, the call will forward to 'No Answer Call'.

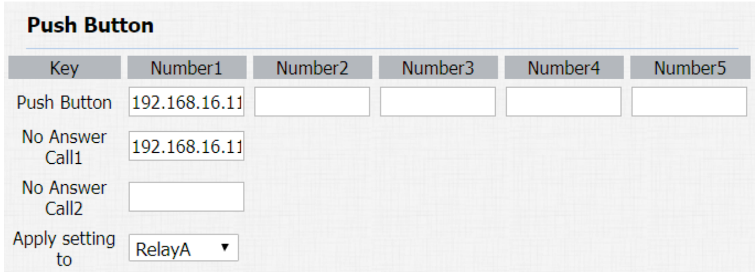

Figure 3.4.5 No answer call

3.4.6. Multicast

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

Paging Barge: Choose the multicast number, and the range is from 1 to 10.

Paging priority Active: Enable or disable the multicast.

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

Label: Input the label for each listening address.

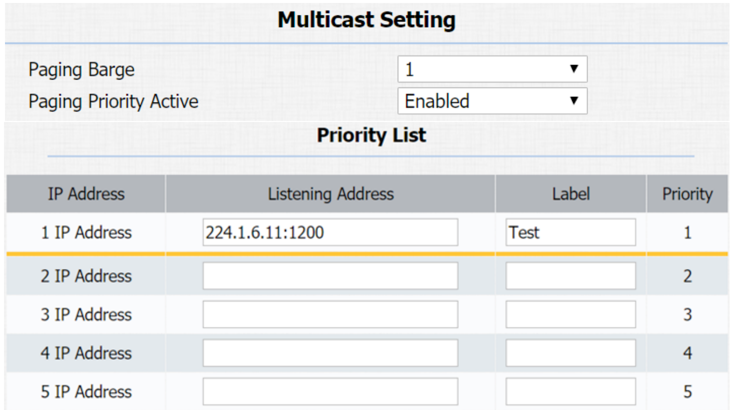

Figure 3.4.6 Multicast

3.5. Security

3.5.1. Live view(optional)

Go to **Intercom** - **Live Stream** to check the real-time video from E21V.

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

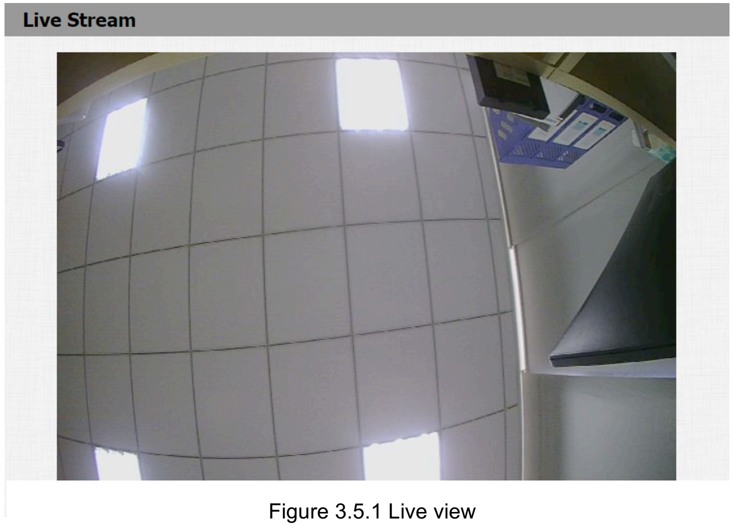

Figure 3.5.1 Live view

3.5.2. RTSP(optional)

E21V 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.

RTSP Stream: To enable RTSP video and select the video codec. E21V supports H.264 video codec by default.

H.264 Video Parameters: H.264 is a video stream compression standard. Different from H.263, it provides an approximately identical level of video stream quality but a half bit rate. This type of compression is sometimes called MPEG-4 part 10. To modify the resolution, framerate and bitrate of H.264.

MPEG4 Video Parameters: MPEG4 is one of the network video image compression standard. It supports the maximum compression ratio 4000:1. It is an important and common video function with great communication application integration ability and less core program space. To modify the resolution, framerate and bitrate of MPEG4.

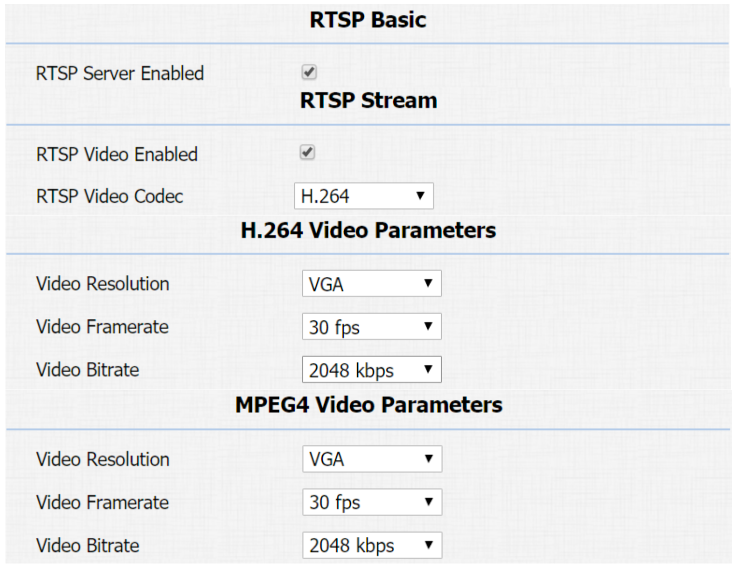

Figure 3.5.2 RTSP

3.5.3. ONVIF(optional)

E21V supports ONVIF protocol, which means E21V's camera can be searched by other devices, like NVR which supports ONVIF protocol as well.

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

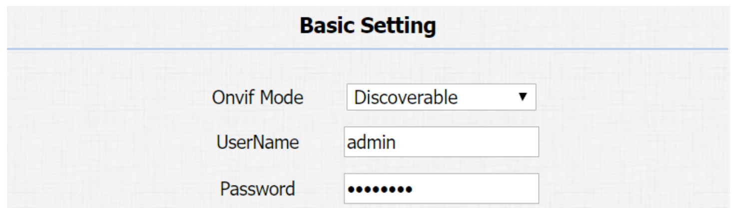

Figure 3.5.3 ONVIF

Switching ONVIF mode to “Undiscoverable,” and it means users must program ONVIF’s URL manually.

The ONVIF’s URL is:

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

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. NC stands for normally closed contact.

Relay ID: E21A/V 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.

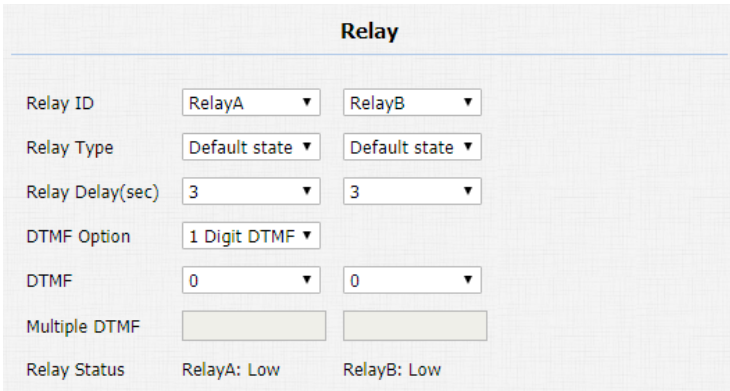

Figure 3.6.1 Relay

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

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

3.6.2. 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.

URL format:

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

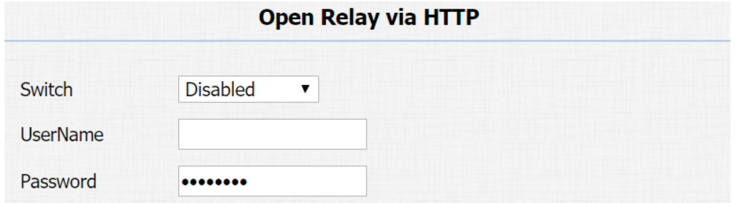

Figure 3.6 .2 Unlock via HTTP

3.6.3. Unlock via Exit Button

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

E21A/V supports three input triggers “Input A/B/C(DOOR A/B/C).”

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.

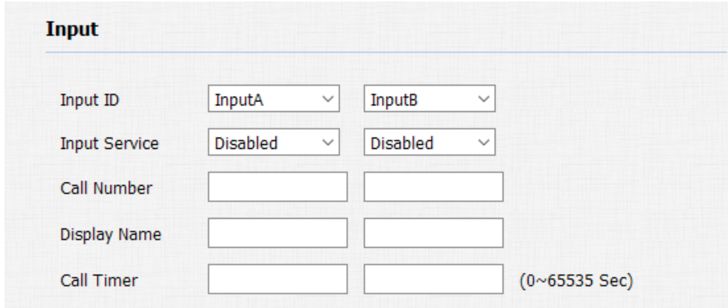

Figure 3.6.3 Unlock via Exit Button

3.7. Reboot

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

Figure 3.7 Reboot

3.8. Reset

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

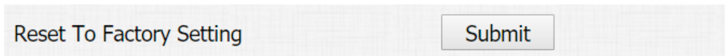

Figure 3.8 Reset

4. Advanced Features

4.1. Phone Configuration

4.1.1. LED

To set up the lighting mode.

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

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

Color Off: The default status if OFF.

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

Blink Mode: To set up the different blink mode frequency.

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 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.

Note: If you are using Direct IP call , the Max Call Time Range is 2 -30 minutes.

If you are using sip account, the time may be limited by the sip server.

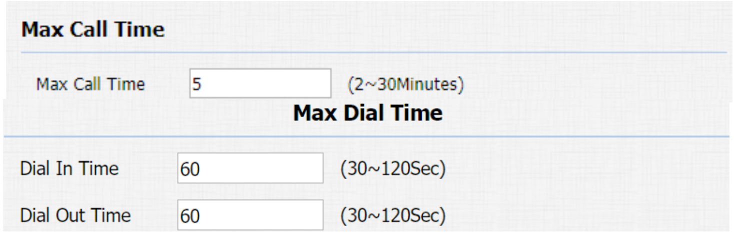

Figure 4.2.1 Call time related

4.2.2. Intercom

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

Intercom: Intercom allows users to establish a call directly with the

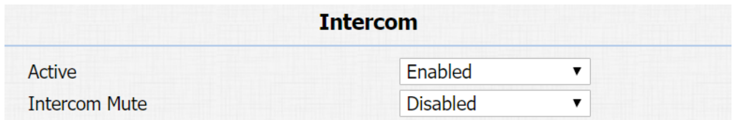

Figure 4.2.2 Intercom

callee.

Active: To enable or disable Intercom feature.

Intercom Mute: If enabled, once the call established, the callee will be muted.

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.

Provisional Response ACK: 100% reliability for all provisional messages, this means it will send ACK every time the IP phone receives a provisional SIP message from SIP server.

Register with user=phone: If enabled, IP phone will send user=phone within SIP message.

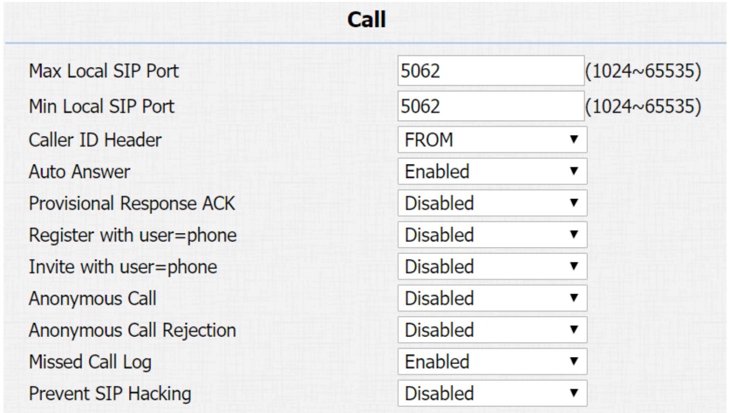

Figure 4.2.3 SIP call related

Anonymous Call: If enabled, E21A/V 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.

4.2.4. Codec

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

Sip Account: To choose which account to configure.

Audio Codec: E21A/V support 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:

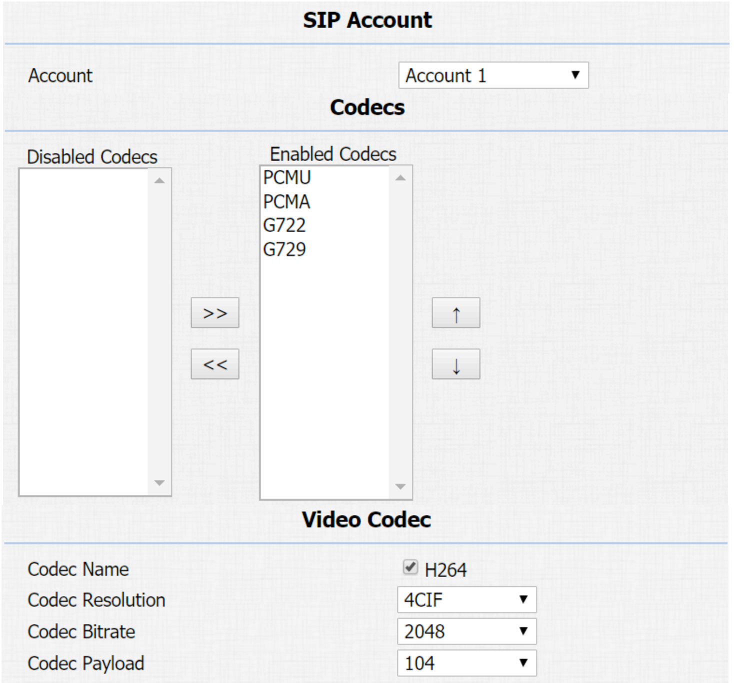

Figure 4.2.4-1 SIP call related codec

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

Video Codec: E21A/V support H.264 standard, which provides better video quality at substantially lower bit rates than previous standards.

Codec Resolution: E21A/V support four resolutions: QCIF, CIF, VGA, 4CIF and 720P.

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

Figure 4.2.4-2 Multicast related codec

4.2.5. Subscribe

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

MWI: Message waiting indicator which is used to indicate whether there is unread new voice message.

BLF Expire: BLF is short for busy lamp field which is used to monitor the designated extension status. The setting here is to negotiate with the server about expire time of BLF subscription.

ACD Expire: Automatic call distribution is often used in offices for customer service, such as call center. The setting here is to negotiate with the server about expire time of ACD subscription.

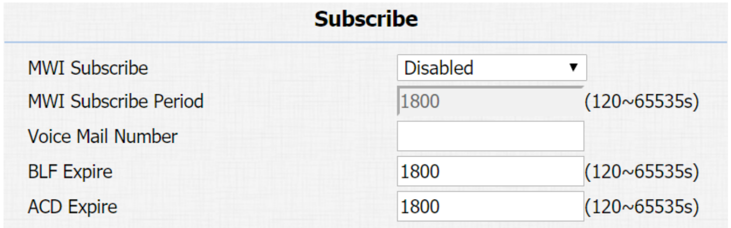

Figure 4.2.5 Subscribe

4.2.6. DTMF

Go to **Account** - **Advanced** to configure RTP audio video 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.

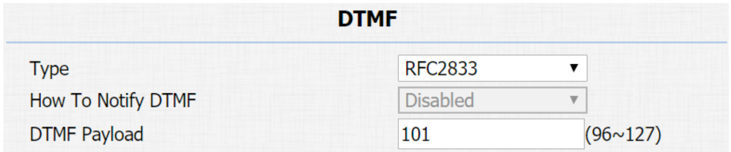

Figure 4.2.6 DTMF

4.2.7. Session Timer

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

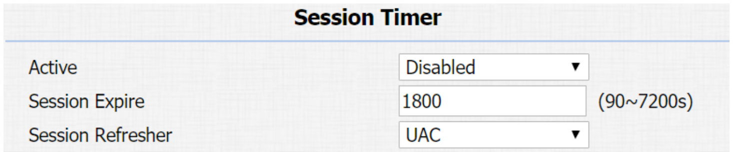

Figure 4.2.7 Session timer

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

4.2.8. BLF List

Go to **Account** - **Advanced** to configure to display or configure BLF list URI address.

BLF List URI: BLF List is short for busy lamp field list.

BLFList PickUp Code: To set the BLF pick up code.

BLFList BargeIn Code: To set the BLF barge in code.

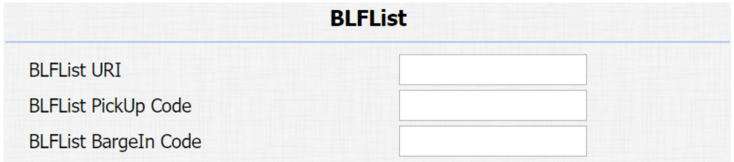

Figure 4.2.8 BLF list

4.2.9. Encryption

If enabled, voice will be encrypted.

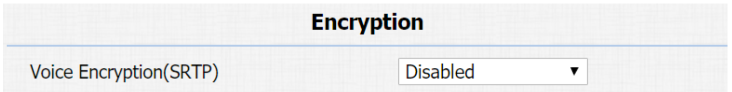

Figure 4.2.9 Encryption

4.2.10. NAT

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

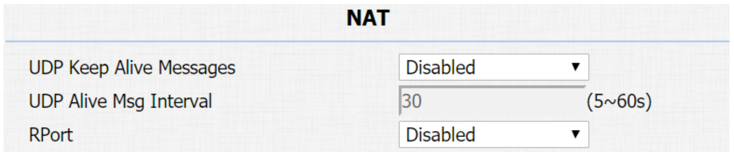

Figure 4.2.10 NAT

UDP Keep Alive message: If enabled, IP phone 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.

4.2.11. 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 not set by default, users can see the company name, model number and firmware version from PCAP.

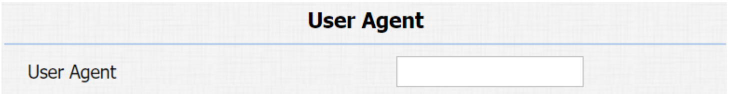

Figure 4.2.11 User Agent

4.3. Security

4.3.1. Action

E21A/V support to send notifications, snapshots(only for E21V) via email and ftp transfer method, or calls via sip call method, when trigger specific actions.

4.3.1.1. 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 namer of SMTP service (usually it is same with sender's email address).

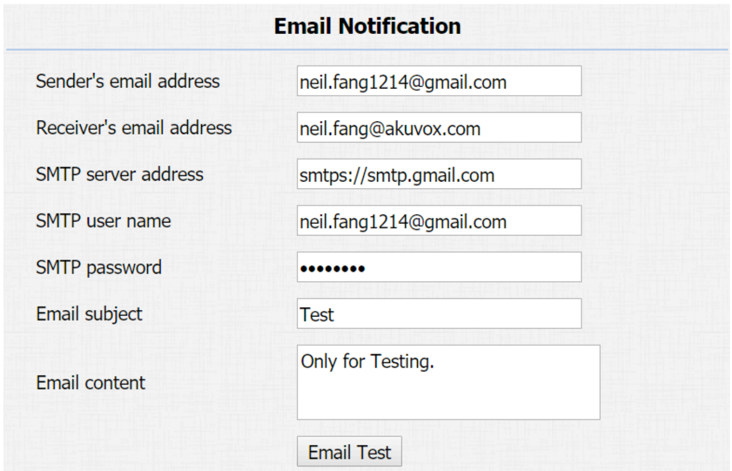

Figure 4.3.1.1-1 Action parameters

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 E21A/V.

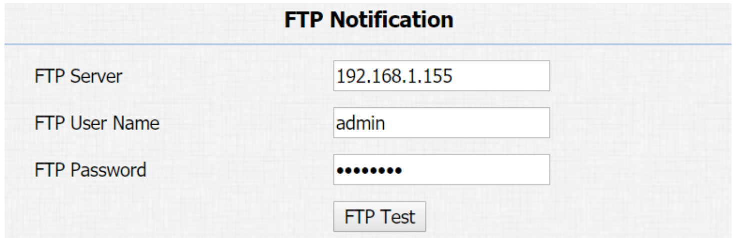

Figure 4.3.1.1-2 Action parameters

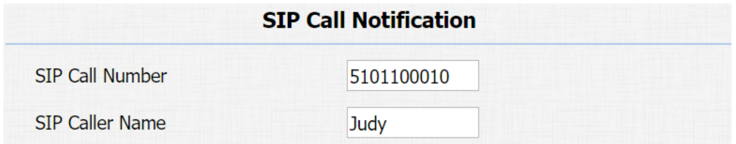

Figure 4.3.1.1-3 Action parameters

4.4. Upgrade

4.4.1. Web Upgrade

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

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

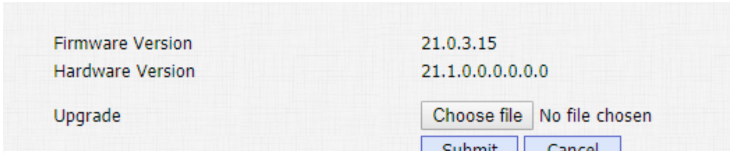

Figure 4.4.1 Web upgrade

4.4.2. Autop Upgrade

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

PNP

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).

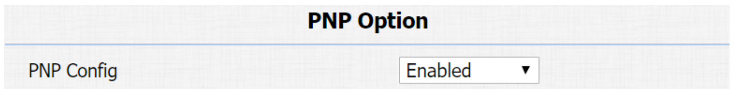

Figure 4.4.2-1 PNP

Manual Autop

Autop 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 user 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.

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.

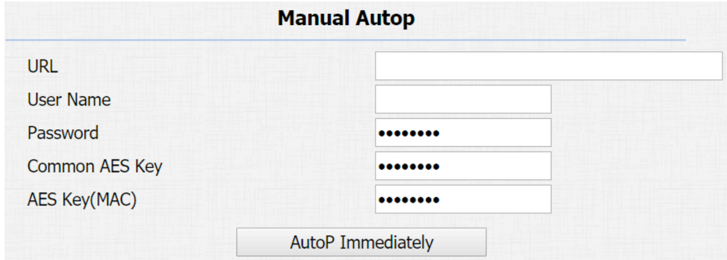

Figure 4.4.2-2 Manual auto provision

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.

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

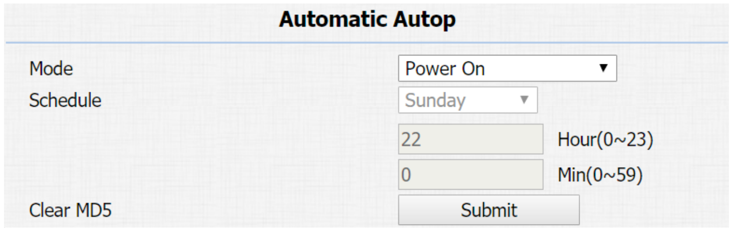

Figure 4.4.2-3 Automatic provision

4.4.3. Backup Config File

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

Export Autop Template: To export current config file.

Figure 4.4.3-1 Backup config file

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

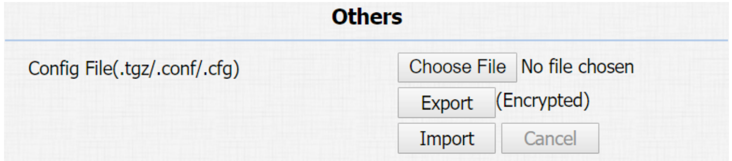

Figure 4.4.3-2 Backup config file

4.5. 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.

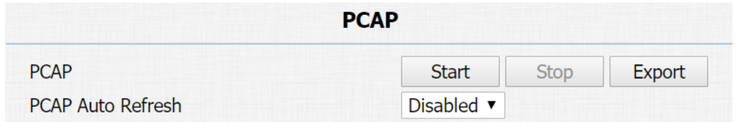

Figure 4.5 PCAP

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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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

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