Akuvox Smart Intercom

E21V

E21A

E21 Series Emergency Station User Manual

About this manual

Thank you for choosing Akuvox's E21 series door phone. This manual is intended for end users, who need to use and configure the door phone. It provides an overview of the most essential functions and features of the product, whose firmware version is 21.0.2.52.

Contact us

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Content

1. Product Overview

1.1. Product Description

Akuvox E21 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. E21 supports 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

E21V

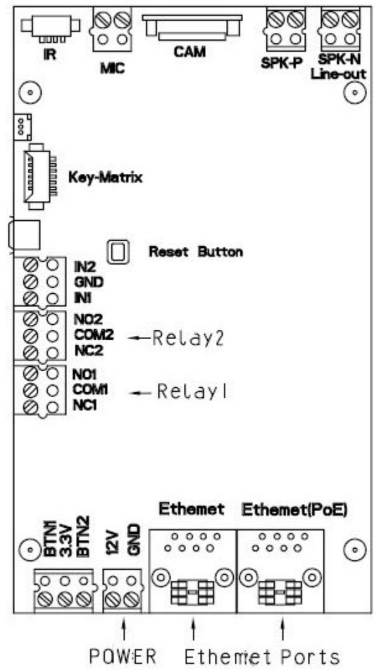
E21A

1.2. Connector Introduction

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

+12V/GND: External power supply terminal.

Relay (NO/NC): Relay control terminal.



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.

Note: Please refer to chapter 3.4.2 about push button setting.

E21V

E21A

2.2. Receiving a Call

User can use IP phone or indoor monitor to call E21V and E21A will answer it automatically by default. If user disable auto answer, pressing button to answer incoming call.

3. Configuration

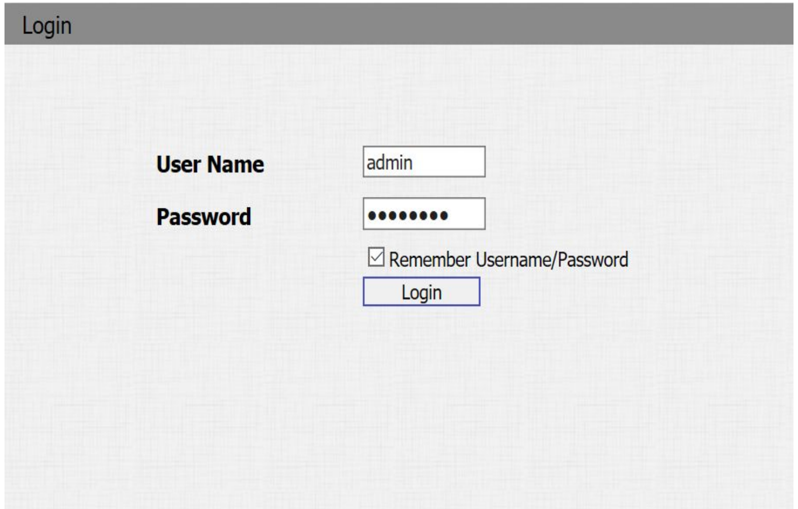
3.1. Web Login

3.1.1. Obtaining IP Address

The Akuvox E21 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, E21 series will announce its IP continuously. Press once again to stop.

3.1.2. 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 User Name and Password are shown below:



User name: **admin**

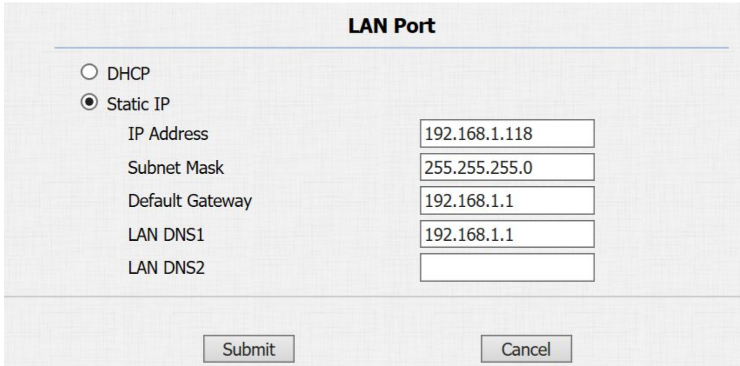
Password: **admin**

3.2. IP Address Setting

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

3.2.1. DHCP

E21 series uses DHCP by default, it will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically.



3.2.2. Static IP

If selected, you could manually set IP address, Subnet Mask, Default Gateway and DNS server.

3.3. Account

Go to Account->Basic to configure sip account and sip server.

3.3.1. 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.

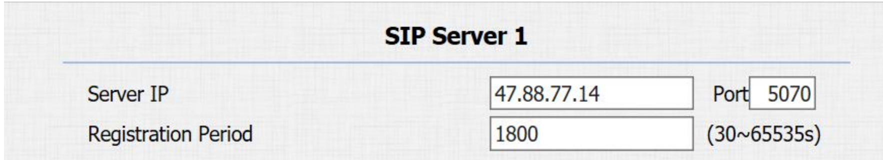
Register Name: To enter extension number you want and the number is allocated by SIP server.

User Name: To enter User Name of the extension.

Password: To enter Password for the extension.

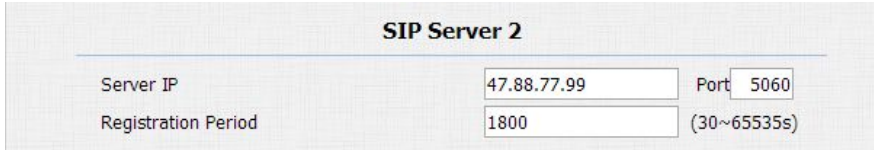
3.3.2. SIP Sever 1

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



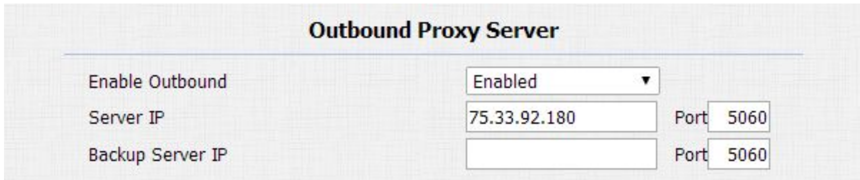
3.3.3. SIP Sever 2

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



3.3.4. Outbound Proxy Server

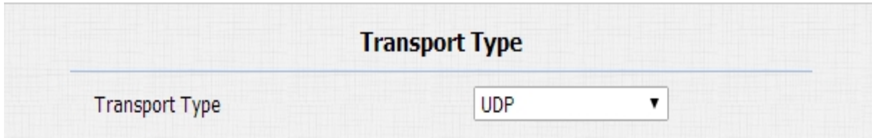
An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server.



3.3.5. 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: A DNS RR for specifying the location of services.

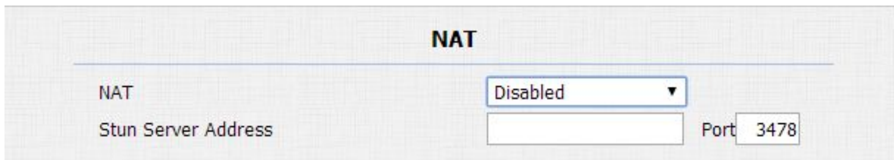


3.3.6. NAT

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

- STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues.

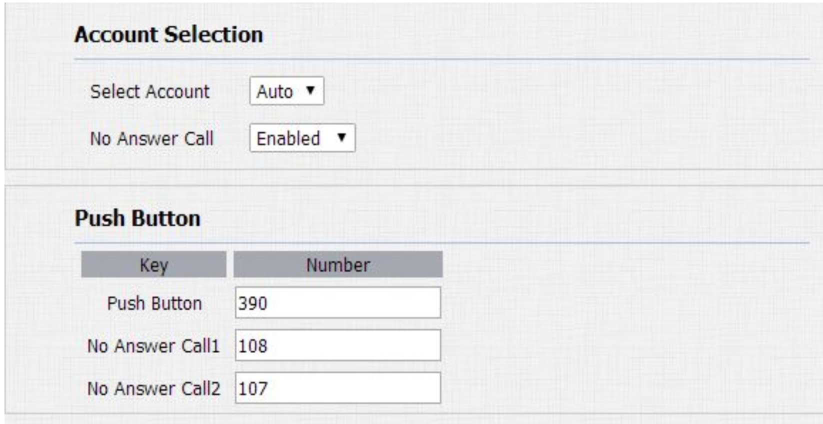
Note: By default, NAT is disabled.



3.4. Call Setting

Go to Intercom->Basic, to configure basic call setting.

3.4.1. No Answer Call



Enable it, if there is no answer from push button number over Answer Call Delay time, E21 series will call predefined 'No Answer Call' number.

3.4.2. Push Button

(1) **Push Button:** To configure the destination number or IP you want to contact with. Also you can call our four numbers at same time.

(2) **No Answer Call 1&2:** To setup one or two no answer call number.

3.4.3. Web Call

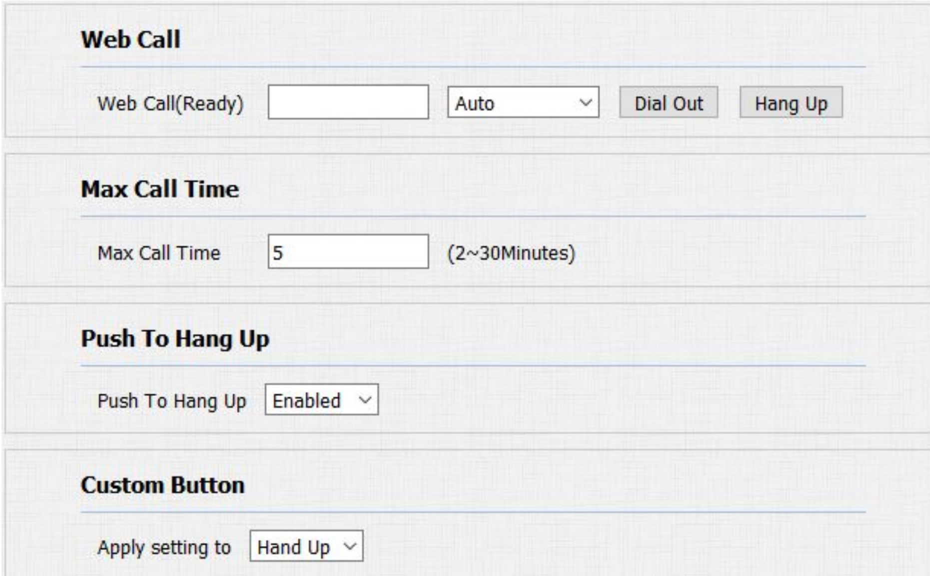
To dial out or answer incoming call from website.

3.4.4. Max Call Time

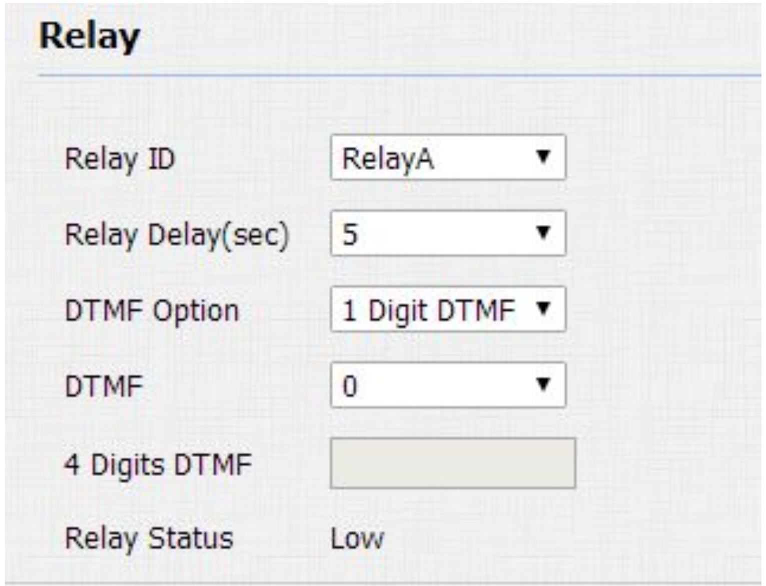
To configure the max call time.

3.4.5. Push to Hang up

To enable or disable pushing button to hang up.



3.5. Relay



Relay ID: E10R supports one relay user can configure respectively.

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

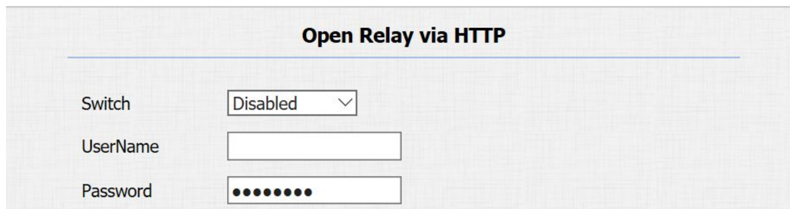
DTMF Option: To select digit of DTMF code, E10R supports maximum 4 digits DTMF code.

DTMF: To configure 1 digit DTMF code for remote unlock

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 .

3.5.1. Open Relay via HTTP



User can use a URL to remote unlock the door.

Switch: Enable this function. Disable by default.

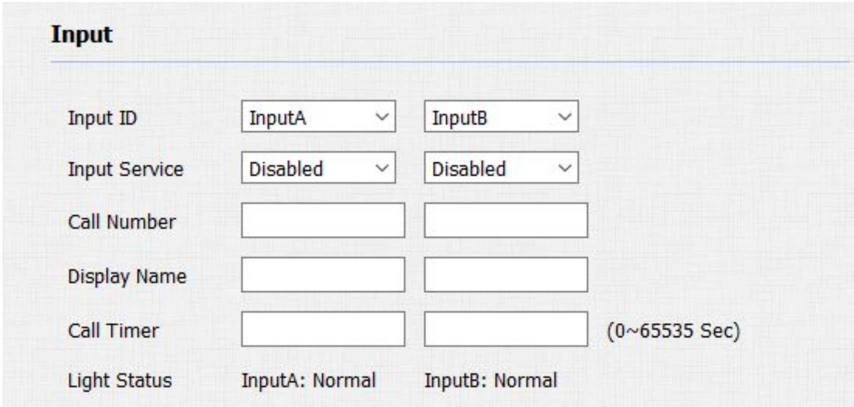
Username & Password: Users can setup the username and password for HTTP unlock.

URL format:

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

3.6. Input

E21 series supports two input triggers Input A/B(DOOR A/B), and go to Intercom->Input to configure.



Input Service: To enable or disable input trigger service.

4. Advance Setting

4.1. LED Setting

There are five LED statuses for E10S/R: NORMAL, OFFLINE, CALLING, TALKING and RECEIVING.

Go to Intercom->Led setting, to configure corresponding LED response.

State	Color Off	Color On	Blink Mode
NORMAL	OFF	Blue	Always On
OFFLINE	OFF	Red	2500/2500
CALLING	OFF	Blue	2500/2500
TALKING	OFF	Blue	500/500
RECEIVING	OFF	Blue	1500/1500

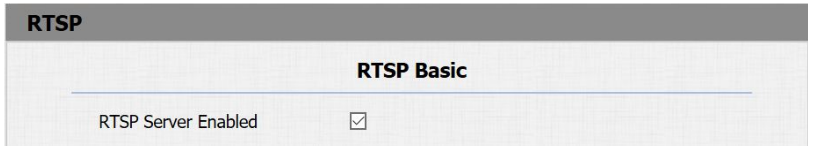
4.2. Live Stream(E21V Only)

Go to Intercom->Live Stream, check the real-time video from E21 series. In addition, user also can check the real-time picture via URL: http://IP_address:8080/picture.jpg.

4.3. RTSP(E21V Only)

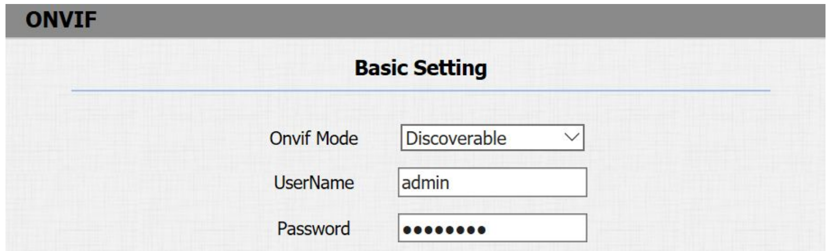
E21 series supports RTSP stream, go to Intercom->RTSP, to enable or disable RTSP server. The URL for RTSP stream is:

rtsp://IP_address/live/ch00_0



4.4. Onvif

E21 series supports ONVIF protocol, which means E21 series'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/password.



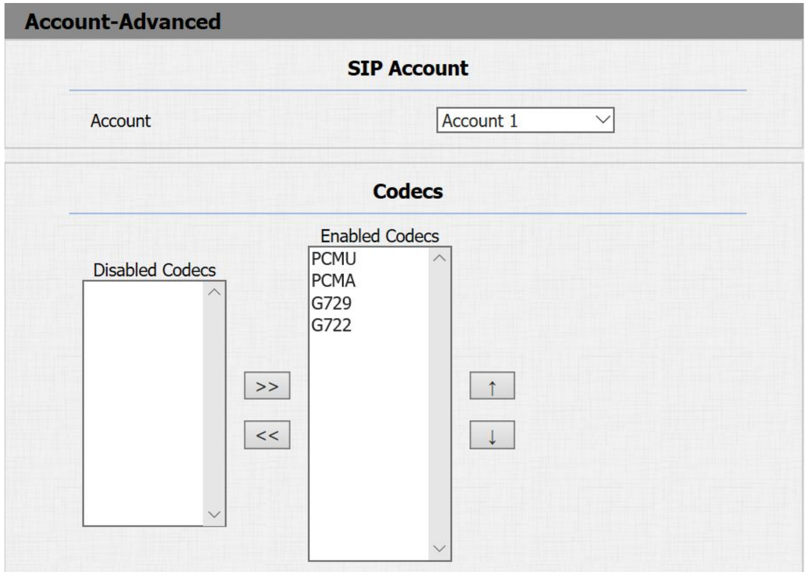
Switching Onvif Mode to undiscoverable means that User must program Onvif's URL manually.

The Onvif's URL is:

http://IP_address:8090/onvif/device_service

4.5. Account-Advanced

Go to Account->Advanced to configure advanced settings for account.



4.5.1. Audio Codec

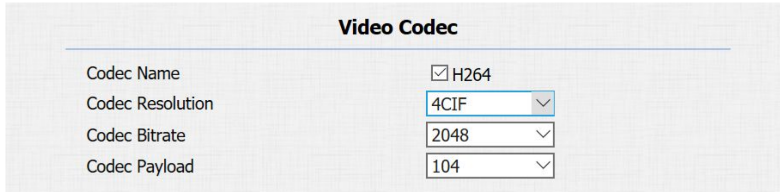
Sip Account: To choose which account to configure.

Audio Codec: R20 series support four audio codec: PCMA, PCMU, G729, G722. Different audio codec requires different bandwidth, user can enable/disable them according to different network environment.

Bandwidth consumption and sample rates.
PCMA: 64kbit/s 8kHz
PCMU: 64kbit/s 8kHz
G729: 8kbit/s 8kHz Least consumption
G722: 64kbit/s 16kHz Best quality

4.5.2. Video Codec

E21 series supports H264 standard, which provides better video quality at substantially lower bit rates than previous standards.

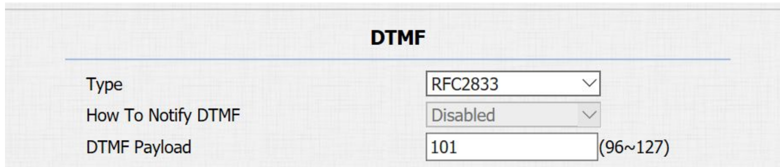


Codec Resolution: R20 series supports four resolutions: QCIF, CIF, VGA, 4CIF and 720P.

Codec Bitrate: To configure bit rates of video stream.

Codec Payload: To configure RTP audio video profile.

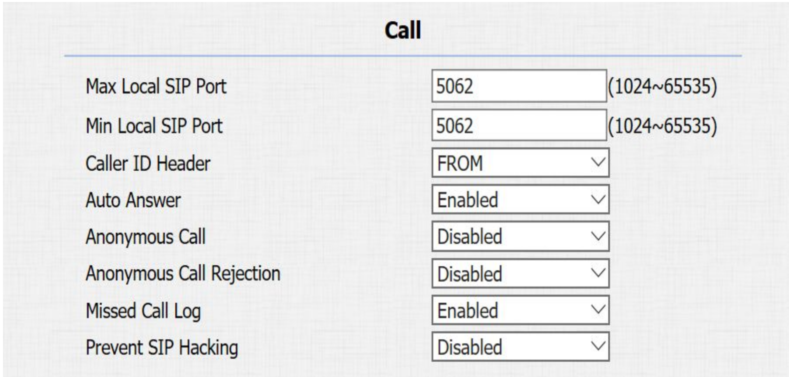
4.5.3. DTMF



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.

4.5.4. Call



To configure RTP audio video profile for DTMF and its payload type.

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 .

Auto Answer: If enabled, incoming call will be answered automatically.

Anonymous Call: If enabled, R20 series will block its information when calling out.

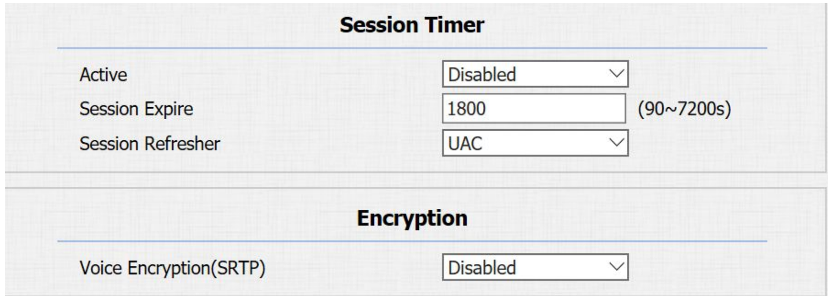
Anonymous Call Rejection: If enabled, calls who block their information will be screened out.

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.

4.5.5. 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.



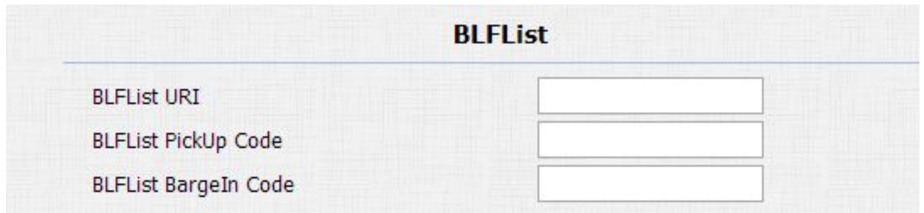
4.5.6. BLF List

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.



4.5.7. Encryption

If enabled, voice will be encrypted.

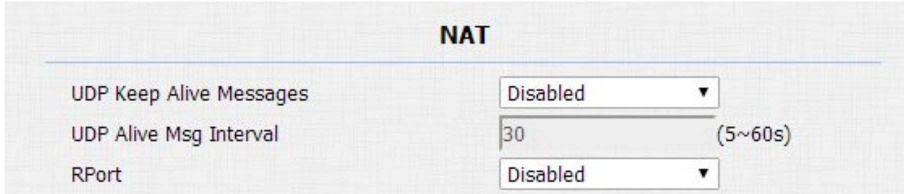
4.5.8. NAT

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.

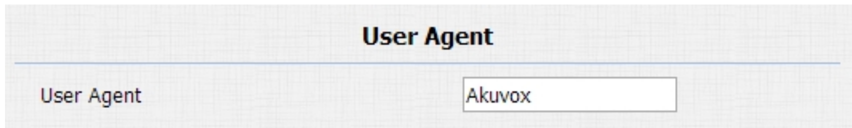
UDP Alive Msg Interval: Keepalive message interval.

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



4.5.9. User Agent

One can customize User Agent field in the SIP message; if user agent is set to specific value, user 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.



4.6. Network-Advance

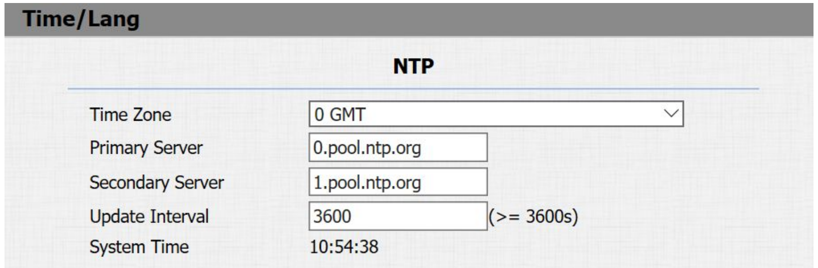
Local RTP: To display and configure Local RTP settings.

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

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

4.7. Time/Lang

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

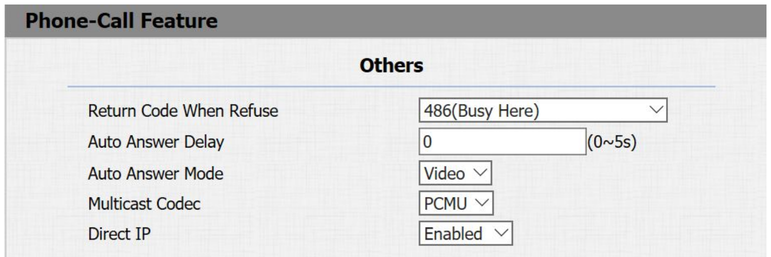


4.8. Call Feature

Go to Phone->Call Feature, to configure Phone-Call Feature.

Return Code When Refuse: To configure return sip status code.

Auto Answer Delay: To configure answer delay when receiving a call.



Auto Answer Mode: To choose Video or Audio mode for auto answer.

Multicast Codec: To configure video codec for multicast.

Direct IP: If disabled, incoming direct IP call will be blocked.

4.9. Voice

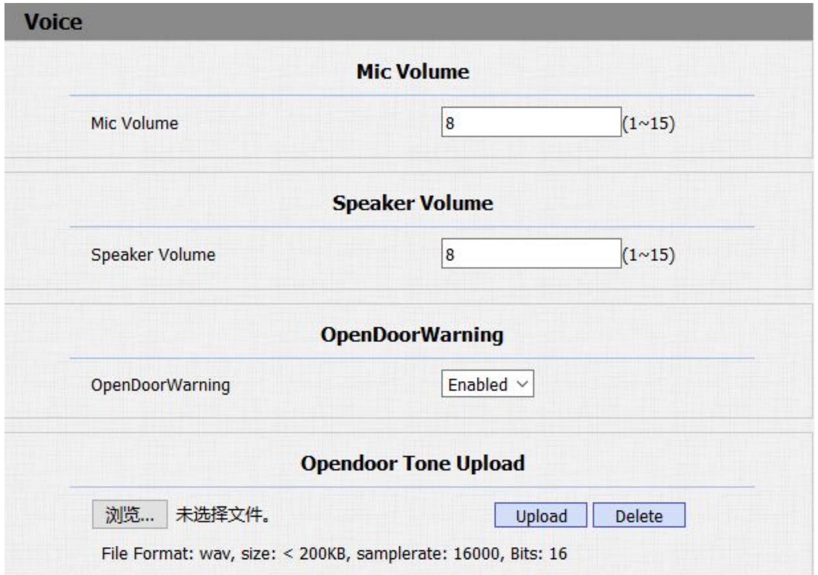
Go to Phone->Voice, to configure volume and upload tone file.

Mic Volume: To configure Microphone volume.

Speaker Volume: To configure Speaker volume.

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

Opendoor Tone Upload: To upload the Opendoor tone by yourself.



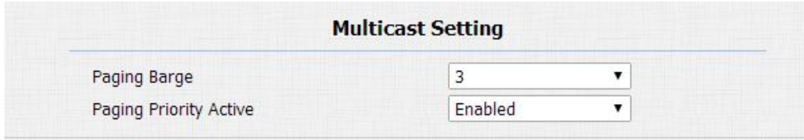
4.10. Multicast

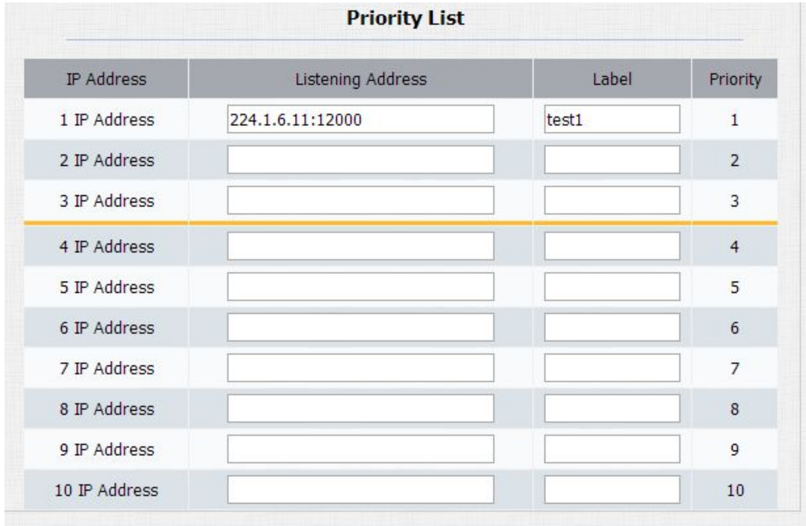
Paging Barge: Choose the multicast number, the range is 1-10.

Paging priority Active: Enable o disable the multicast.

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

Label: Input the label for each listening address.





4.11. Upgrade

4.11.1. Upgrade-Basic

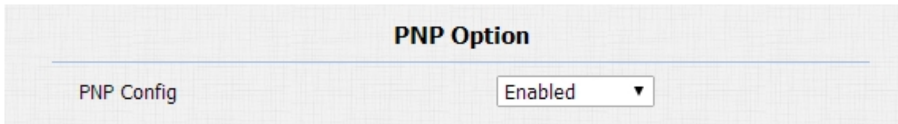
Go to Upgrade->Basic, user can upgrade firmware; Reset to factory setting and reboot.

4.11.2. Upgrade-Advanced

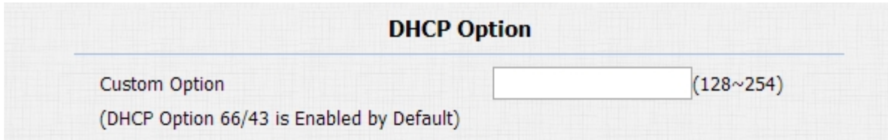
4.11.2.1. PNP

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).



4.11.2.2. DHCP Option

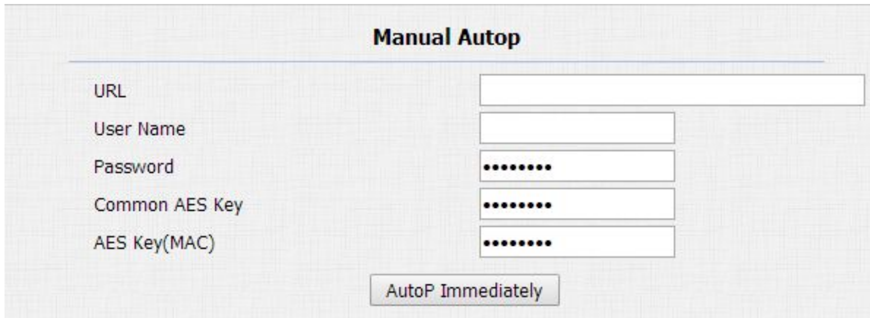


To display and configure DHCP setting for AutoP. Option 66/43 is enable by default. It can support Https,Http,Ftp,Tftp server.

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

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

4.11.2.3. Manual Autop



Autop (Auto-Provisioning) is a centralized and unified upgrade of IP telephone. It is a simple and time-saving configuration for IP 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.

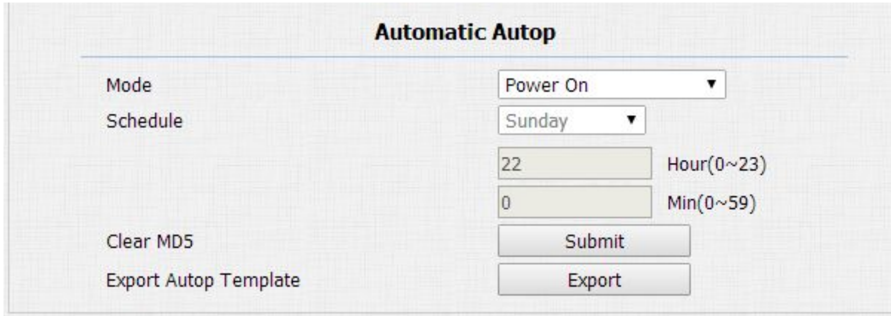
Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file.

AES Key (MAC): Used for IP phone to decipher MAC-oriented auto provisioning configuration file(for example, file name could be 0c1105888888.cfg if IP phone's MAC address is 0c1105888888).

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

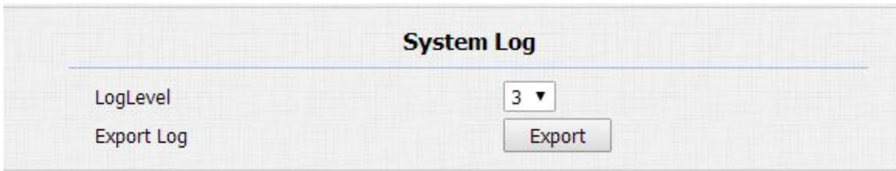
4.11.2.4. Automatic Autop

To display and configure Auto Provisioning mode settings. This Auto Provisioning mode is actually self-explanatory. For example, mode “Power on” means IP phone will go to do Provisioning every time it powers on.



4.11.2.5. System Log

System log: System log is used to debug, higher LogLevel means more specific system log will be recorded. When device failure occur, user can export System Log send to Akuvox techsupport and we would try our best to address the issue for you.



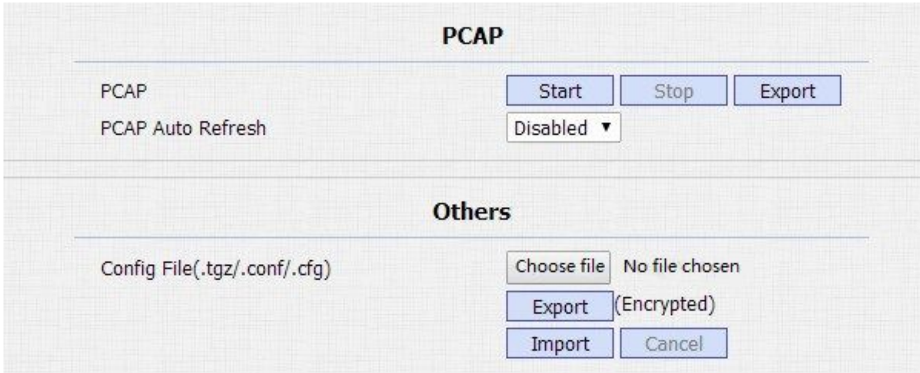
System log level: From level 0~7.The higher level means the more specific system log is saved to a temporary file. By default, it's level 3.

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

4.11.2.6. PCAP

PCAP: To capture packet which is useful for us to address issue.

Other: To export and import config file.



4.11.2.7. Other

To export current config file or import new config file.