E10 Series Door Phone Admin Guide

About this manual

Thank you for choosing Akuvox's E10 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 110.0.2.7.

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

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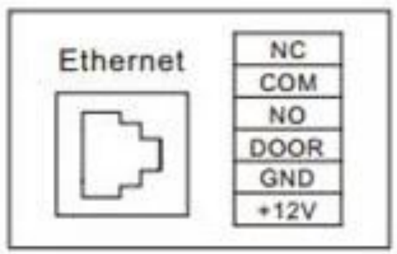
Content

1. Product Overview

1.1. Product Description

E10S/R is a smart SIP-based secondary entry phone. It can be connected with Akuvox indoor phone for unlock and monitor. It is more convenient and safe for residents to check the visitor identity through E10 series. They are often applicable in villas, apartments.

1.2. Connector Introduction



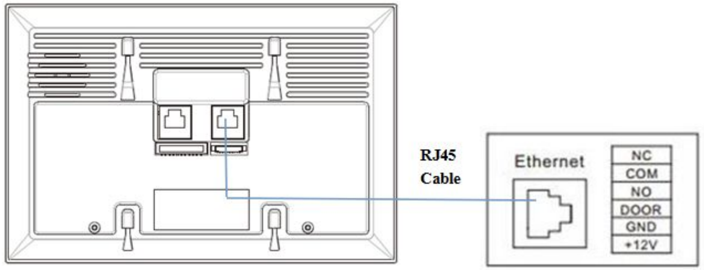
Ethernet(POE): Ethernet(POE) connector which can provide both power and network connection.

+12V/GND: External power supply terminal.

DOOR: Trigger signal input terminal.(Only E10R)

RelayA/B (NO/NC): Relay control terminal.(Only E10R)

2. Daily Use



2.1. Power On

E10 series can be powered on by Akuvox indoor monitor.Connecting E10 series with the PON port of indoor monitor. Also you can choose 12V/1A power adapter if you don’t use PON.

2.2. Making a Call

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

2.3. Receiving a Call

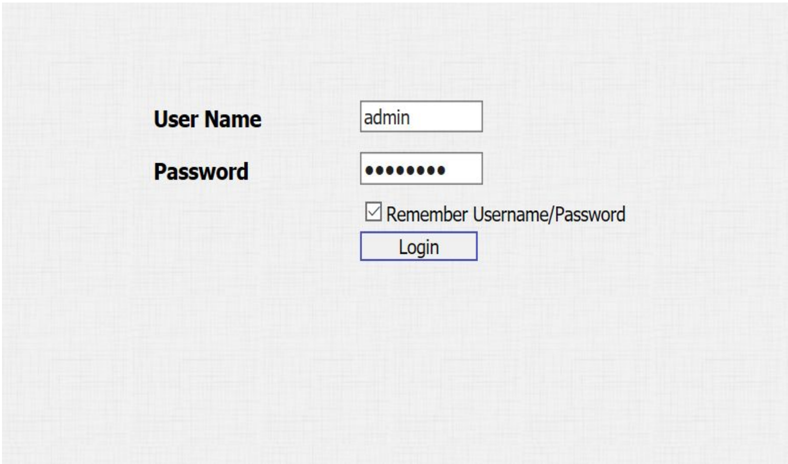
User can use IP phone or indoor monitor to call E10S/R and it will answer automatically by default. If user disable auto answer, pressing button to answer incoming call.

3. Configuration

3.1. Web Login

3.1.1. Obtaining IP Address

The Akuvox E10 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, E10 series will announce its IP continuously. Press once again to stop.



3.1.2. Access the Device Website

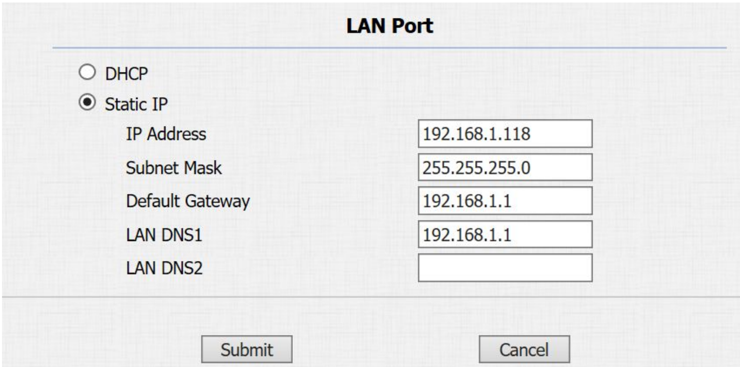
Open a Web Browser, access the corresponding IP address. Then, enter the default user name and password to login. The default administrator User Name and Password are shown below:

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

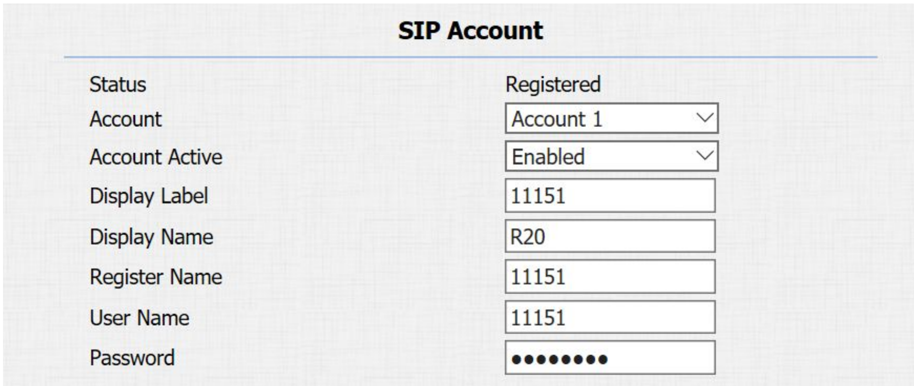
E10S/R uses DHCP by default, it will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically.

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.

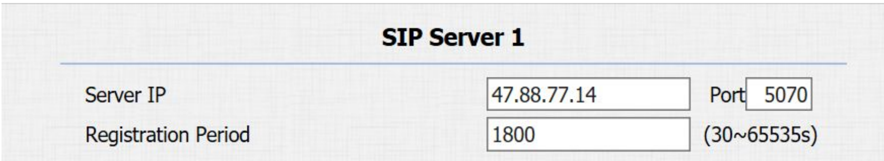
Register Name: To enter extension number you want and the number is allocated by SIP server.

User Name: To enter user name of the extension.

Password: To enter password for the extension

3.3.2. SIP Sever 1

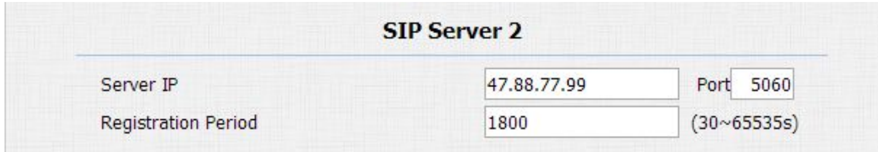
Server IP: To enter SIP server's IP address or URL



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

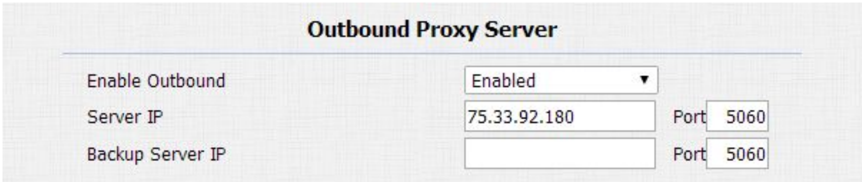
3.3.3. SIP Sever 2

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

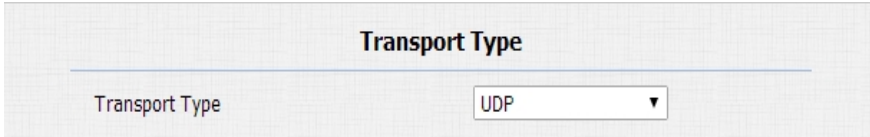
An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server.



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.

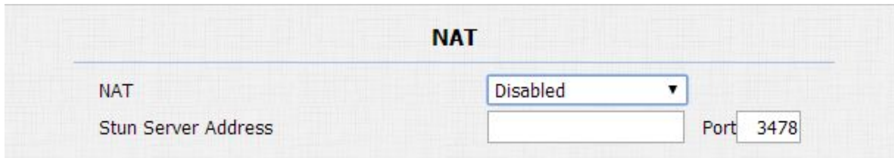


3.3.6. NAT

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.



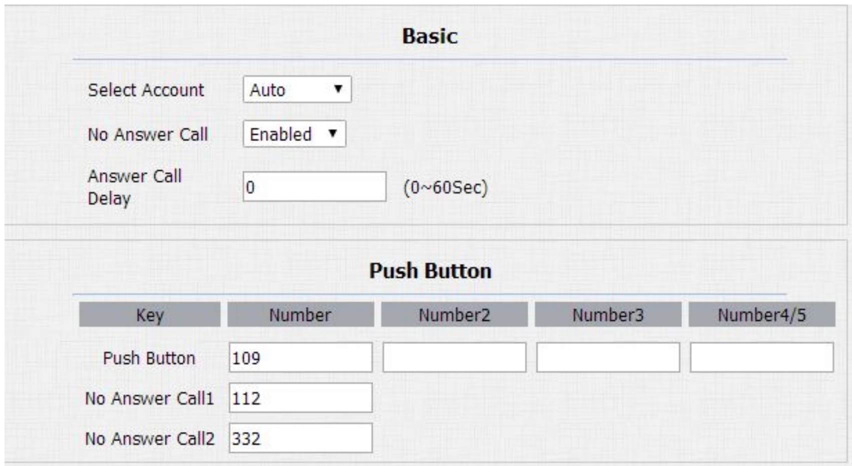
3.4. Call Setting

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, E10S/R will call predefined 'No Answer Call' number.

3.4.2. Push Button



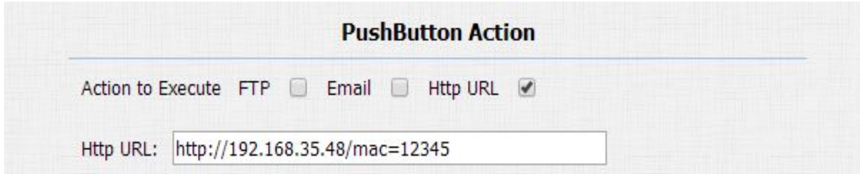
(1)Push Button: To configure the destination number or IP you want to contact with. Also you can call our four numbers at same time.

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

3.4.3. Push Button Action

Action to execute: To choose suitable way to receive message or snapshot when pushing button.

HTTP URL: If you tick HTTP URL, enter corresponding HTTP server IP address in the HTTP URL area.



3.4.4. Web Call

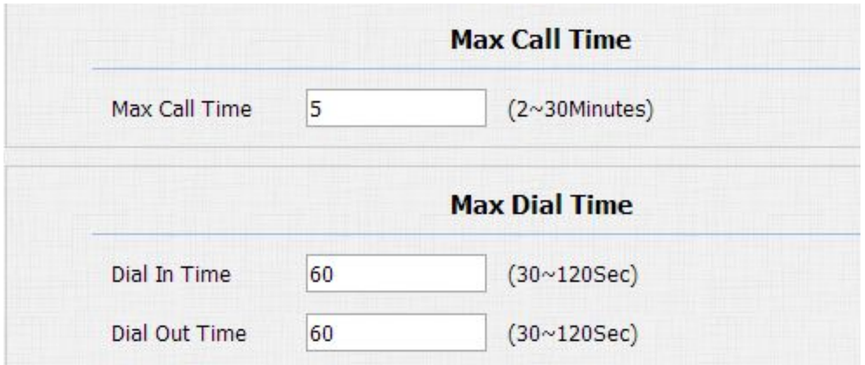
To dial out or answer incoming call from website.

3.4.5. Call&Dial Time

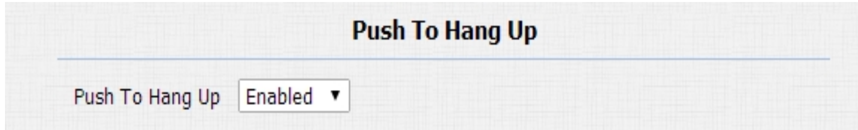
Max Call Time: To configure the max call time.

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.

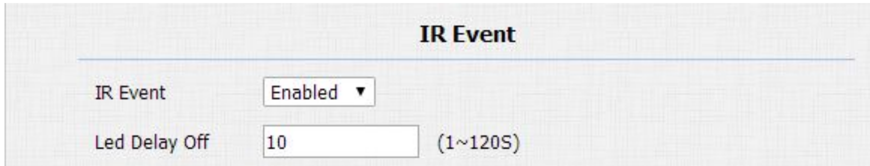


3.4.6. Push to Hang up



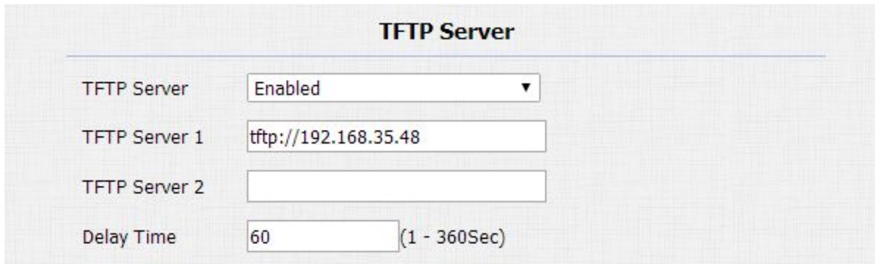
To enable or disable pushing button to hang up.

3.4.7. IR Event



When enable the IR event function, if someone close to the Infrared sensor, this action triggers the Key light led to turn on and the camera to capture picture. If someone stands outside the detective range of the Infrared sensor, the Key Light led will not turn on.

3.4.8. TFTP Server



This feature is specially for uploading the picture to SDMC.Once the device connected to SDMC, “TFTP server 1” will filled in IP of SDMC automatically, the captured picture will be sent to SDMC for storage.

3.5. Relay

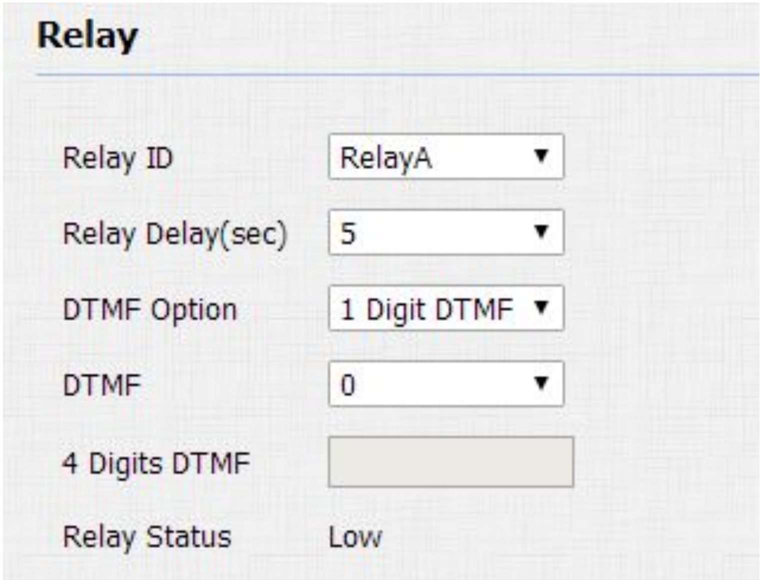
Relay ID: E10R supports one relay user can configure respectively.

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 .

3.6. Action

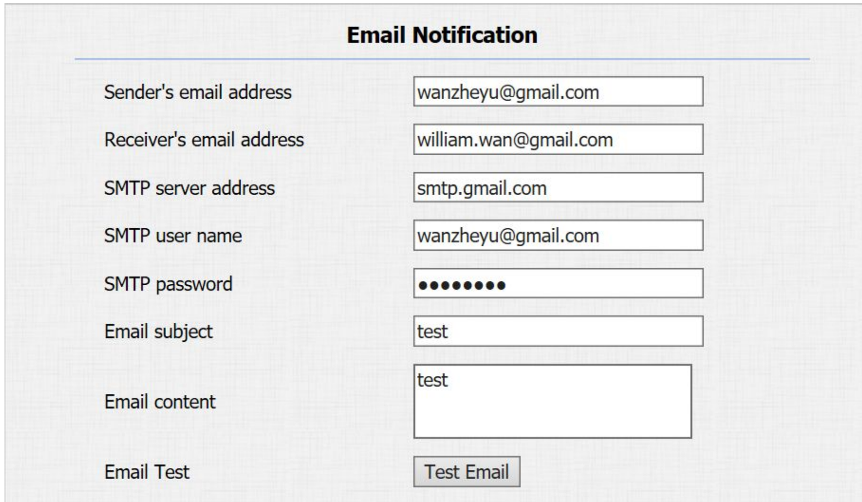
Go to Intercom->Action to set action receiver.

3.6.1. 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).

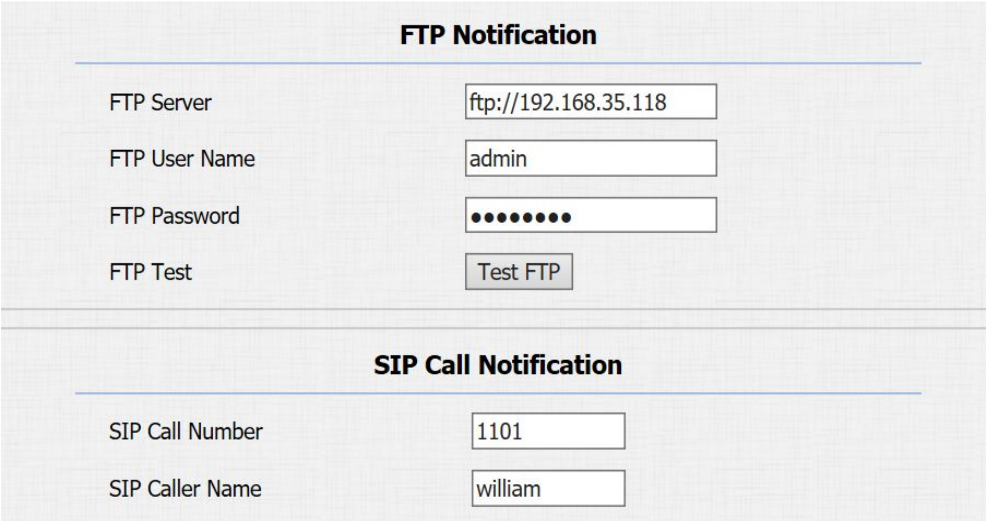
SMTP password: To configure password of SMTP service(usually it is same with the password of sender's email).

Email subject: To configure subject of email.

Email content: To configure content of email.

Email Test: To test whether email notification is available.

3.6.2. 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.

3.6.3. SIP Notification

SIP Call Number: To configure sip call number.

SIP Call Name: To configure display name of the callee.

4. Advance Setting

4.1. LED Setting

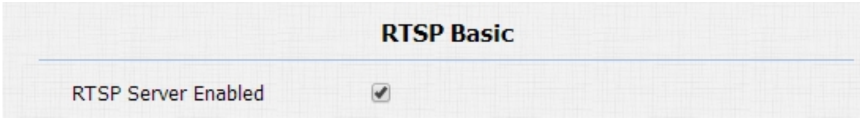
There are five LED statuses for E10S/R: NORMAL, OFFLINE, CALLING, TALKING and RECEIVING.

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

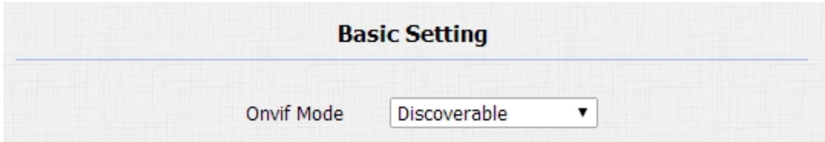
4.2. RTSP

E10S/R 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



4.3. Onvif

E10S/R supports ONVIF protocol, which means their 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.



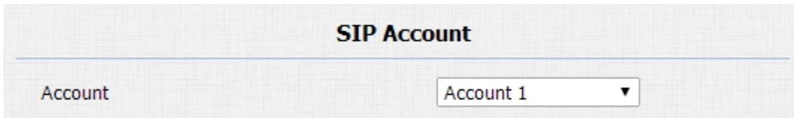
Switching Onvif Mode to undiscoverable means that User must program Onvif's URL manually.

The Onvif’s URL is:

http://IP_address:8090/onvif/device_service

4.4. Account-Advanced

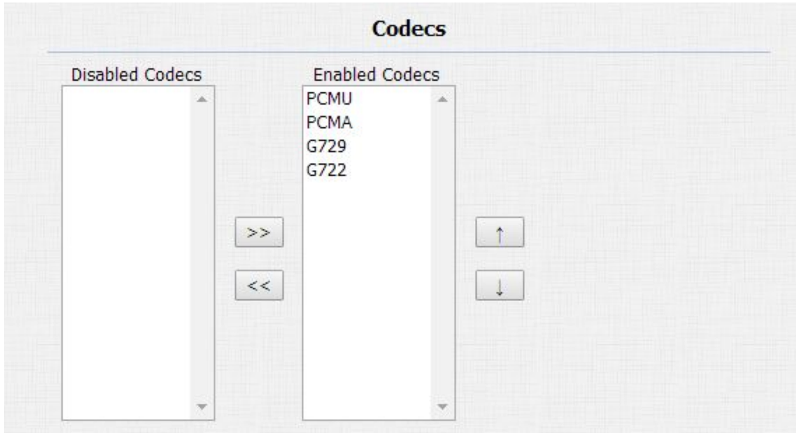
Go to Account->Advanced to configure advanced settings for account.



4.4.1. Audio Codec

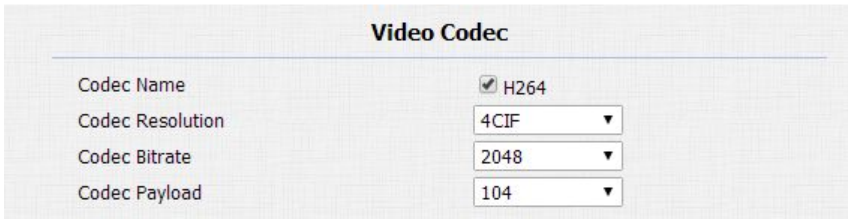
Sip Account: To choose which account to configure.

Audio Codec: E10S/R support four audio codec: PCMA, PCMU, G729, G722. Different audio codec requires different bandwidth, user can enable/disable them according to different network environment.



4.4.2. Video Codec

E10R/S supports H264 standard, which provides better video quality at substantially lower bit rates than previous standards.



Codec Resolution: E10R/S 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.

Subscribe

MWI: Message Waiting Indicator which is used to indicate whether there is unread new voice message.

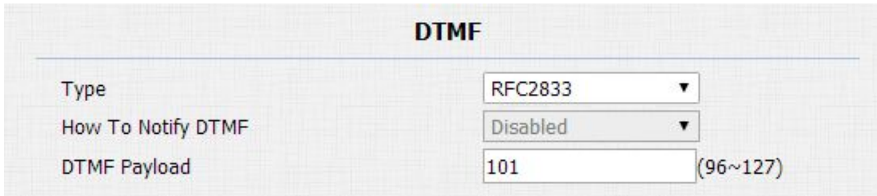
BLF: BLF is short for Busy Lamp Field which is used to monitor the designated extension status.

ACD: 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.

4.4.3. 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.

4.4.4. Call

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.

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.

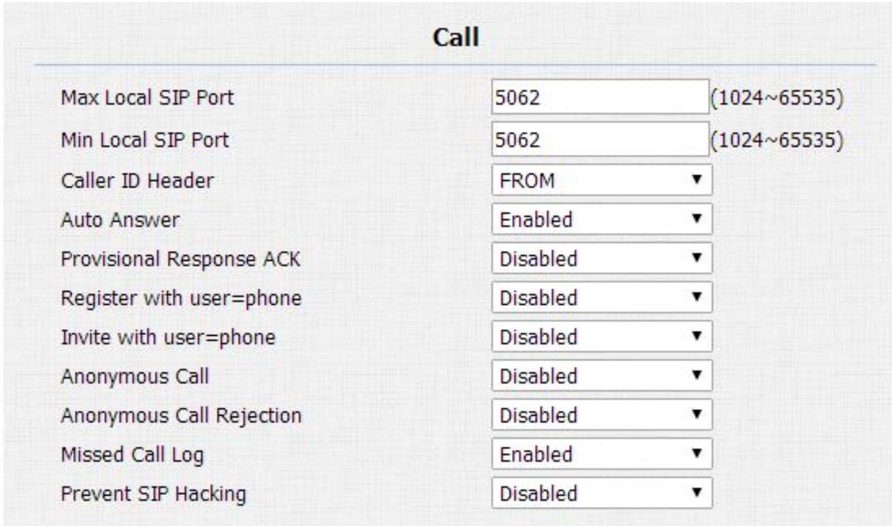
Register with user=phone: If enabled, IP phone will send user=phone within SIP message.

Anonymous Call: If enabled, R27A will lock its information when calling out.

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

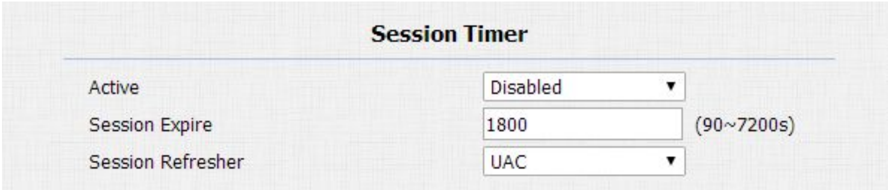
Missed Call Log: If enabled, any missed call will be recorded into call log.

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



4.4.5. 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.



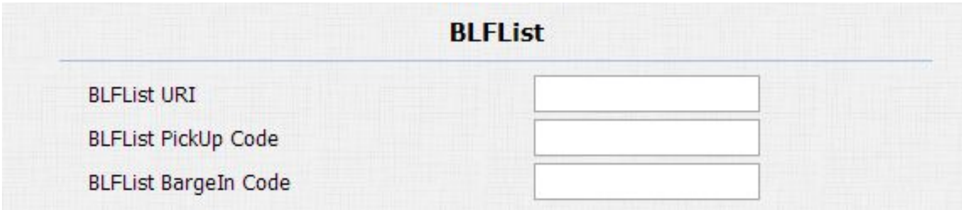
4.4.6. BLF List

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.



4.4.7. Encryption

If enabled, voice will be encrypted.

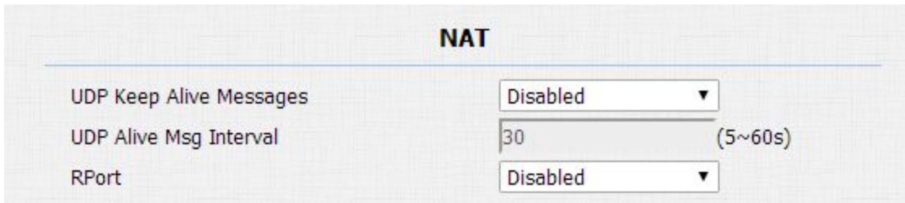
4.4.8. NAT

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.

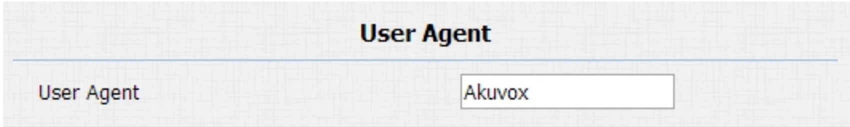
UDP Alive Msg Interval: Keepalive message interval.

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



4.4.9. 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.



4.5. Network-Advance

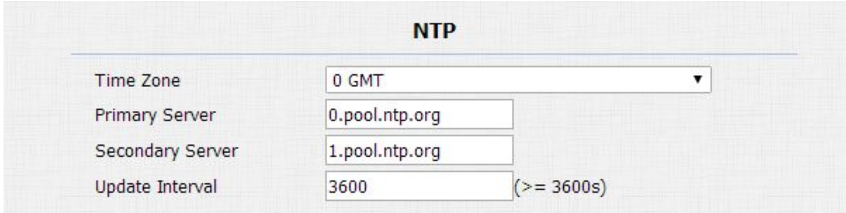
Local RTP: To display and configure Local RTP settings.

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

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

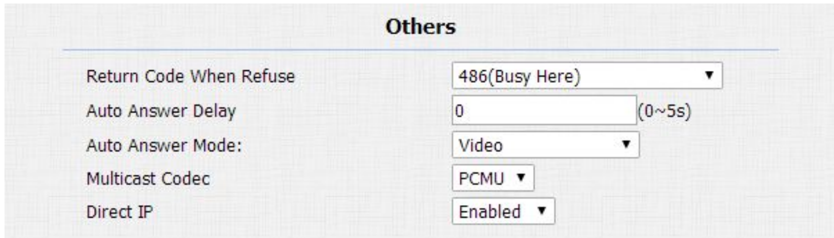
4.6. Time/Lang

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



4.7. Call Feature

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



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

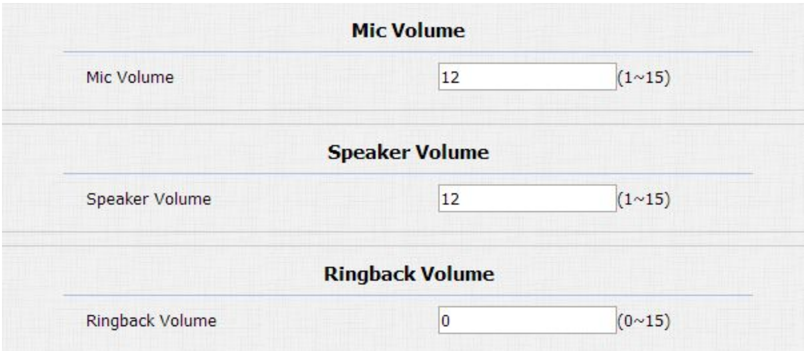
Auto Answer Delay: To configure answer delay when receiving a call.

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.

4.8. Voice



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

Mic Volume: To configure Microphone volume.

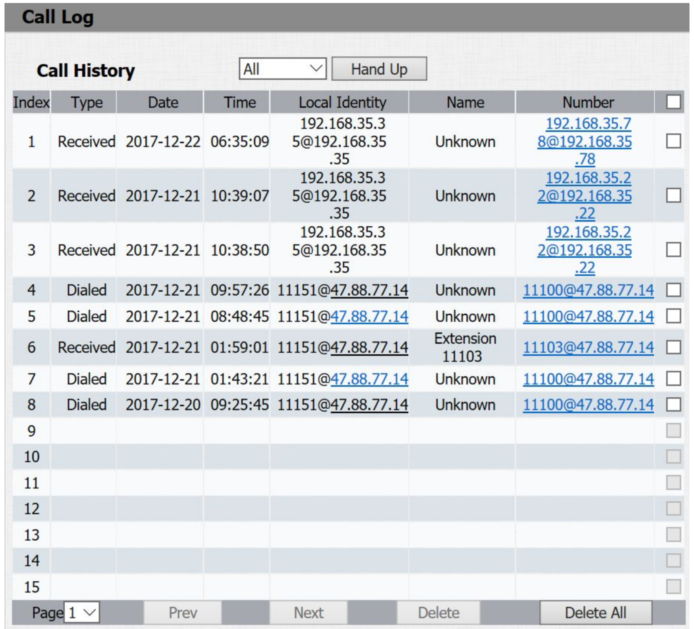
Speaker Volume: To configure Speaker volume.

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

4.9. Log

4.9.1. Call Log

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



Index	Type	Date	Time	Local Identity	Name	Number
1	Received	2017-12-22	06:35:09	192.168.35.35@192.168.35.35	Unknown	192.168.35.78@192.168.35.78
2	Received	2017-12-21	10:39:07	192.168.35.35@192.168.35.35	Unknown	192.168.35.22@192.168.35.22
3	Received	2017-12-21	10:38:50	192.168.35.35@192.168.35.35	Unknown	192.168.35.22@192.168.35.22
4	Dialed	2017-12-21	09:57:26	11151@47.88.77.14	Unknown	11100@47.88.77.14
5	Dialed	2017-12-21	08:48:45	11151@47.88.77.14	Unknown	11100@47.88.77.14
6	Received	2017-12-21	01:59:01	11151@47.88.77.14	Extension 11103	11103@47.88.77.14
7	Dialed	2017-12-21	01:43:21	11151@47.88.77.14	Unknown	11100@47.88.77.14
8	Dialed	2017-12-20	09:25:45	11151@47.88.77.14	Unknown	11100@47.88.77.14
9						
10						
11						
12						
13						
14						
15						

4.10. Upgrade

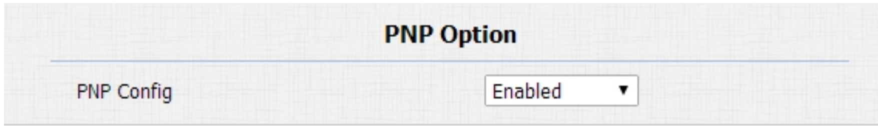
4.10.1. Upgrade-Basic

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

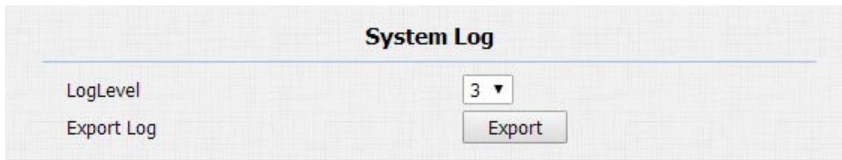
4.10.2. Upgrade-Advanced

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).



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.



4.11. Security-Basic

Go to Security->Basic, to modify password and session time.

4.11.1. Web Password Modify

To modify password of 'admin' or 'user' account.