

**Akuvox** Smart  
Intercom



## **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](http://www.akuvox.com) or feel free to contact us by

Sales email: [sales@akuvox.com](mailto:sales@akuvox.com)

Technical support email: [techsupport@akuvox.com](mailto:techsupport@akuvox.com)

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

**We highly appreciate your feedback about our products**

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# 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.

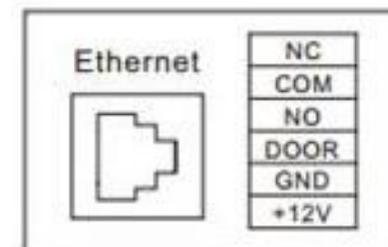


## 1.2. Connector Introduction

**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)

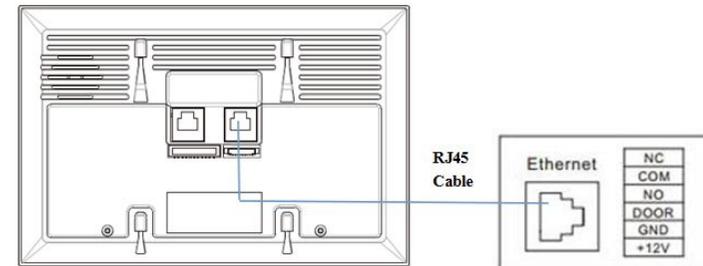
## 2. Daily Use

### 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.



The screenshot shows a web login form with the following elements:

- User Name:** A text input field containing the value "admin".
- Password:** A text input field where the password is masked with seven black dots.
- Remember Username/Password
- Login:** A button with a blue border and the text "Login".

### 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**

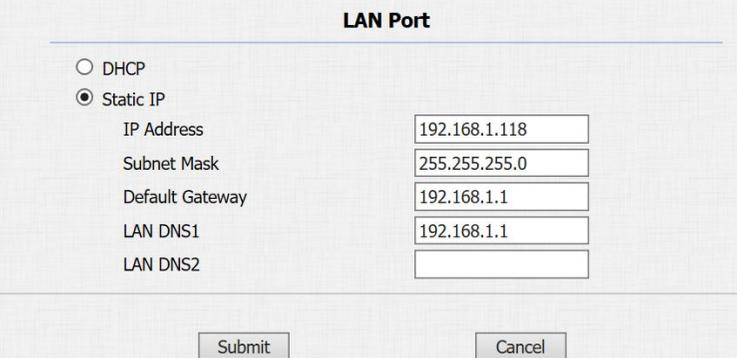
Password: **admin**

## 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.



The screenshot shows the 'LAN Port' configuration page. It has two radio buttons: 'DHCP' (unselected) and 'Static IP' (selected). Below the radio buttons are five input fields for static IP configuration: 'IP Address' (192.168.1.118), 'Subnet Mask' (255.255.255.0), 'Default Gateway' (192.168.1.1), 'LAN DNS1' (192.168.1.1), and 'LAN DNS2' (empty). At the bottom right, there are 'Submit' and 'Cancel' buttons.

LAN Port	
<input type="radio"/> DHCP	
<input checked="" type="radio"/> Static IP	
IP Address	192.168.1.118
Subnet Mask	255.255.255.0
Default Gateway	192.168.1.1
LAN DNS1	192.168.1.1
LAN DNS2	
<input type="button" value="Submit"/> <input type="button" value="Cancel"/>	

## 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.

SIP Account	
Status	Registered
Account	Account 1
Account Active	Enabled
Display Label	11151
Display Name	R20
Register Name	11151
User Name	11151
Password	••••••••

**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

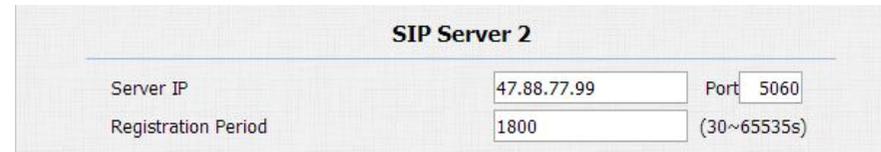
**Server IP:** To enter SIP server's IP address or URL

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

SIP Server 1		
Server IP	<input type="text" value="47.88.77.14"/>	Port <input type="text" value="5070"/>
Registration Period	<input type="text" value="1800"/>	(30~65535s)

### 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.

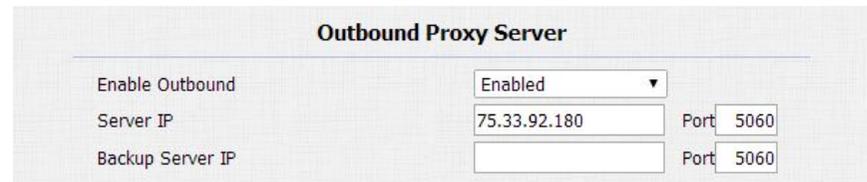


The screenshot shows the configuration for 'SIP Server 2'. It includes a 'Server IP' field with the value '47.88.77.99' and a 'Port' field with the value '5060'. Below these is a 'Registration Period' field with the value '1800' and a note '(30~65535s)'.

SIP Server 2	
Server IP	47.88.77.99
Port	5060
Registration Period	1800 (30~65535s)

### 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.



The screenshot shows the configuration for the 'Outbound Proxy Server'. It includes an 'Enable Outbound' dropdown menu set to 'Enabled', a 'Server IP' field with the value '75.33.92.180' and a 'Port' field with the value '5060'. Below these is a 'Backup Server IP' field and a 'Port' field with the value '5060'.

Outbound Proxy Server	
Enable Outbound	Enabled
Server IP	75.33.92.180
Port	5060
Backup Server IP	
Port	5060

### 3.3.5. Transport Type

To display and configure Transport type for SIP message

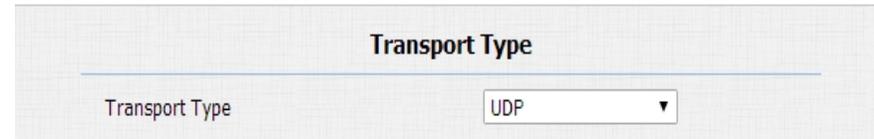
- UDP: UDP is an unreliable but very efficient transport layer protocol.
- TCP: Reliable but less-efficient transport layer protocol.
- TLS: Secured and Reliable transport layer protocol.
- DNS-SRV: A DNS RR for specifying the location of services.

### 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.



The screenshot shows a configuration panel titled "Transport Type". Below the title, there is a label "Transport Type" and a dropdown menu with "UDP" selected.



The screenshot shows a configuration panel titled "NAT". Below the title, there is a label "NAT" and a dropdown menu with "Disabled" selected. Below this, there is a label "Stun Server Address" followed by an empty text input field, and a label "Port" followed by a text input field containing "3478".

## 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.

The screenshot shows a configuration interface with two main sections: 'Basic' and 'Push Button'.

**Basic**

- Select Account: Auto (dropdown)
- No Answer Call: Enabled (dropdown)
- Answer Call Delay: 0 (input field) (0~60Sec)

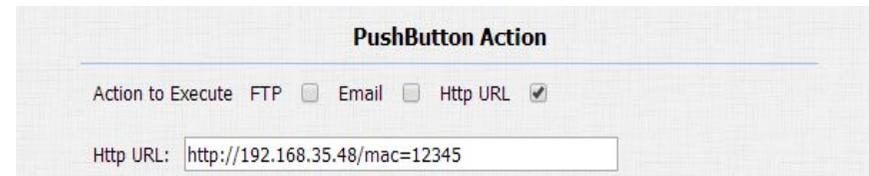
**Push Button**

Key	Number	Number2	Number3	Number4/5
Push Button	109			
No Answer Call1	112			
No Answer Call2	332			

### 3.4.3. Push Button Action

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

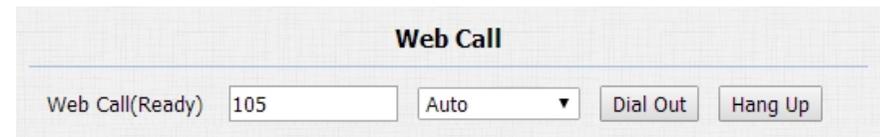
**HTTP URL:** If you tick HTTP URL, enter corresponding HTTP server IP address in the HTTP URL area.



The screenshot shows a configuration window titled "PushButton Action". It contains three radio buttons for "Action to Execute": "FTP" (unchecked), "Email" (unchecked), and "Http URL" (checked). Below these is a text input field labeled "Http URL:" containing the value "http://192.168.35.48/mac=12345".

### 3.4.4. Web Call

To dial out or answer incoming call from website.



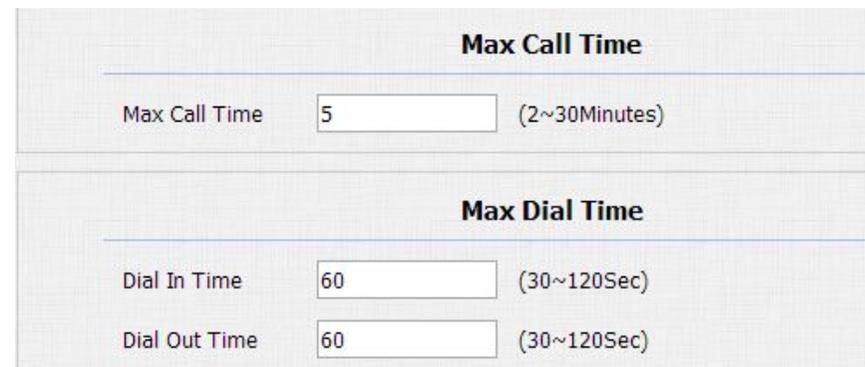
The screenshot shows a configuration window titled "Web Call". It features a "Web Call(Ready)" label, a text input field with the value "105", a dropdown menu set to "Auto", and two buttons: "Dial Out" and "Hang Up".

### 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.

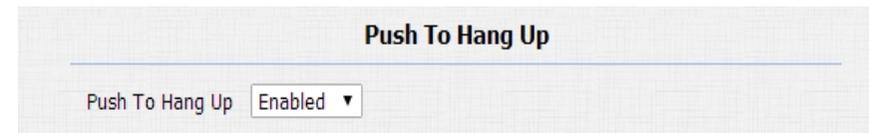
**Dial Out Time:** To configure the max no answer call time.



The screenshot shows two configuration windows. The top one is titled "Max Call Time" and has a "Max Call Time" input field with the value "5" and a range indicator "(2~30Minutes)". The bottom one is titled "Max Dial Time" and has two input fields: "Dial In Time" with the value "60" and range "(30~120Sec)", and "Dial Out Time" with the value "60" and range "(30~120Sec)".

### 3.4.6. Push to Hang up

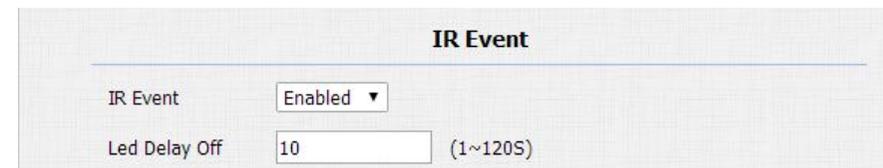
To enable or disable pushing button to hang up.



Push To Hang Up	
Push To Hang Up	Enabled ▼

### 3.4.7. IR Event

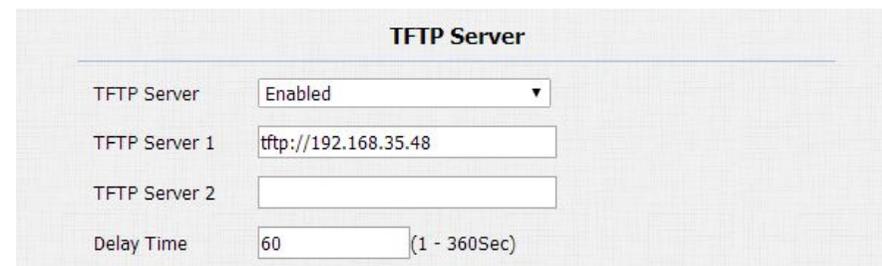
When enable the IR event function, if someone close to the Infrared sensor, this action triggers the Key light led to turn on and the camera to capture picture. If someone stands outside the detective range of the Infrared sensor, the Key Light led will not turn on.



IR Event	
IR Event	Enabled ▼
Led Delay Off	10 (1~120S)

### 3.4.8. TFTP Server

This feature is specially for uploading the picture to SDMC. Once the device connected to SDMC, "TFTP server 1"



TFTP Server	
TFTP Server	Enabled ▼
TFTP Server 1	tftp://192.168.35.48
TFTP Server 2	
Delay Time	60 (1 - 360Sec)

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.

The image shows a configuration window titled "Relay". It contains several settings:

- Relay ID: RelayA (dropdown menu)
- Relay Delay(sec): 5 (dropdown menu)
- DTMF Option: 1 Digit DTMF (dropdown menu)
- DTMF: 0 (dropdown menu)
- 4 Digits DTMF: (empty text input field)
- Relay Status: Low (text label)

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

## 3.6. Action

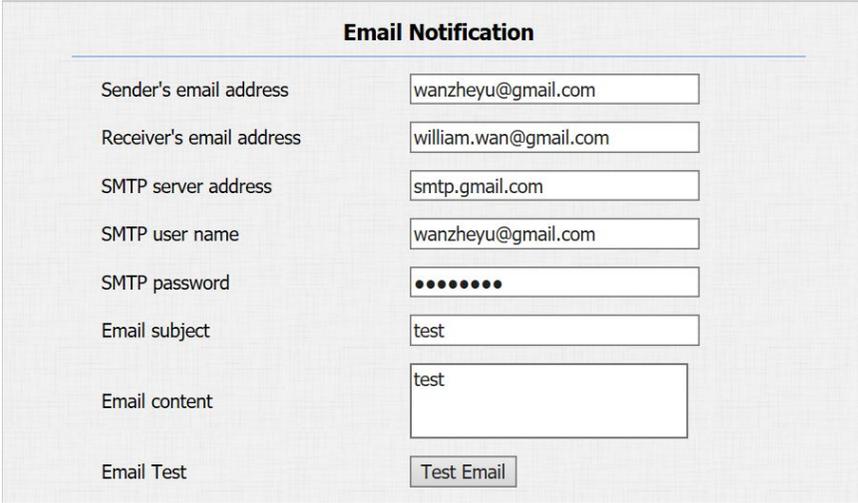
Go to Intercom->Action to set action receiver.

### 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.



The screenshot shows a web form titled "Email Notification" with the following fields and values:

Email Notification	
Sender's email address	wanzheyu@gmail.com
Receiver's email address	william.wan@gmail.com
SMTP server address	smtp.gmail.com
SMTP user name	wanzheyu@gmail.com
SMTP password	••••••••
Email subject	test
Email content	test
Email Test	Test Email

**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.

### 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.

The image shows a configuration interface with two sections: 'FTP Notification' and 'SIP Call Notification'. The 'FTP Notification' section includes fields for 'FTP Server' (ftp://192.168.35.118), 'FTP User Name' (admin), 'FTP Password' (masked with dots), and a 'Test FTP' button. The 'SIP Call Notification' section includes fields for 'SIP Call Number' (1101) and 'SIP Caller Name' (william).

FTP Notification	
FTP Server	<input type="text" value="ftp://192.168.35.118"/>
FTP User Name	<input type="text" value="admin"/>
FTP Password	<input type="password" value="••••••••"/>
FTP Test	<input type="button" value="Test FTP"/>

SIP Call Notification	
SIP Call Number	<input type="text" value="1101"/>
SIP Caller Name	<input type="text" value="william"/>

**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

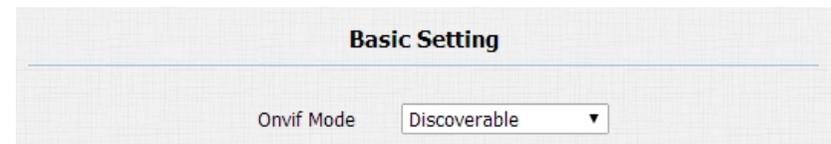
E10S/R supports RTSP stream, go to Intercom->RTSP, to enable or disable RTSP server. The URL for RTSP stream is:  
rtsp://IP\_address/live/ch00\_0



## 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

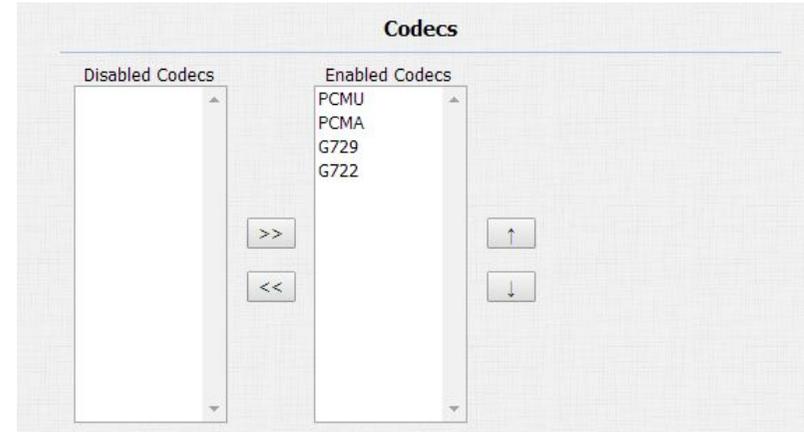
Go to Account->Advanced to configure advanced settings for account.



### 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.



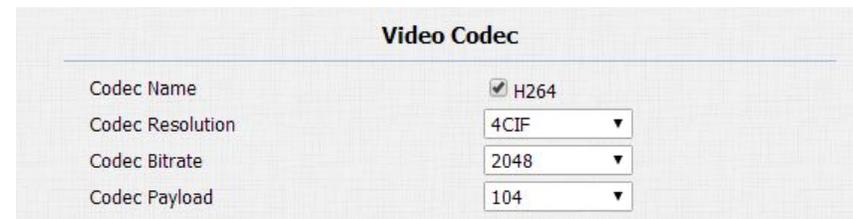
## 4.4.2. Video Codec

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.

Subscribe	
MWI Subscribe	Disabled ▼
MWI Subscribe Period	1800 (120~65535s)
Voice Mail Number	
BLF Expire	1800 (120~65535s)
ACD Expire	1800 (120~65535s)

DTMF	
Type	RFC2833 ▼
How To Notify DTMF	Disabled ▼
DTMF Payload	101 (96~127)

- 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.

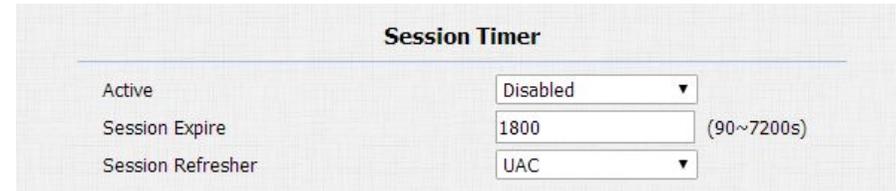
**Missed Call Log:** If enabled, any missed call will be recorded into call log.

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

Call		
Max Local SIP Port	<input type="text" value="5062"/>	(1024~65535)
Min Local SIP Port	<input type="text" value="5062"/>	(1024~65535)
Caller ID Header	<input type="text" value="FROM"/>	▼
Auto Answer	<input type="text" value="Enabled"/>	▼
Provisional Response ACK	<input type="text" value="Disabled"/>	▼
Register with user=phone	<input type="text" value="Disabled"/>	▼
Invite with user=phone	<input type="text" value="Disabled"/>	▼
Anonymous Call	<input type="text" value="Disabled"/>	▼
Anonymous Call Rejection	<input type="text" value="Disabled"/>	▼
Missed Call Log	<input type="text" value="Enabled"/>	▼
Prevent SIP Hacking	<input type="text" value="Disabled"/>	▼

## 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.



The screenshot shows a configuration panel titled "Session Timer". It contains three rows of settings:

Session Timer	
Active	Disabled ▼
Session Expire	1800 (90~7200s)
Session Refresher	UAC ▼

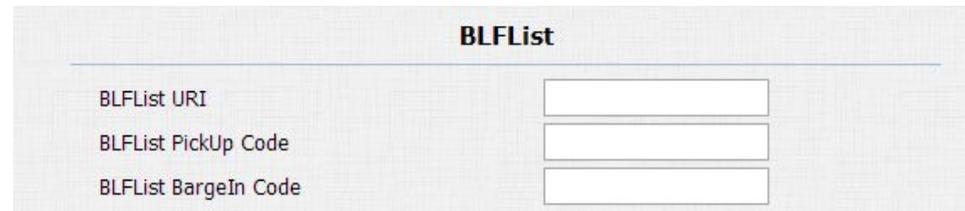
## 4.4.6. BLF List

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 Bargeln Code:** To set the BLF barge in code.



The screenshot shows a configuration panel titled "BLFList". It contains three rows of settings, each with a text input field:

BLFList	
BLFList URI	<input type="text"/>
BLFList Pickup Code	<input type="text"/>
BLFList BargeIn Code	<input type="text"/>

## 4.4.7. Encryption

If enabled, voice will be encrypted.

Encryption	
Voice Encryption(SRTP)	Disabled ▼

## 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.

NAT	
UDP Keep Alive Messages	Disabled ▼
UDP Alive Msg Interval	30 (5~60s)
RPort	Disabled ▼

## 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.

User Agent	
User Agent	<input type="text" value="Akuvox"/>

Local RTP	
Starting RTP Port	<input type="text" value="11800"/> (1024~65535)
Max RTP Port	<input type="text" value="12000"/> (1024~65535)

## 4.6. Time/Lang

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

NTP	
Time Zone	0 GMT
Primary Server	0.pool.ntp.org
Secondary Server	1.pool.ntp.org
Update Interval	3600 (>= 3600s)

## 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.

Others	
Return Code When Refuse	486(Busy Here)
Auto Answer Delay	0 (0~5s)
Auto Answer Mode:	Video
Multicast Codec	PCMU
Direct IP	Enabled

**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.

Mic Volume	
Mic Volume	<input type="text" value="12"/> (1~15)

Speaker Volume	
Speaker Volume	<input type="text" value="12"/> (1~15)

Ringback Volume	
Ringback Volume	<input type="text" value="0"/> (0~15)

## 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.

Call Log							
Call History							
Index	Type	Date	Time	Local Identity	Name	Number	
1	Received	2017-12-22	06:35:09	192.168.35.3 5@192.168.35.35	Unknown	<a href="#">192.168.35.78</a> <a href="#">8@192.168.35.22</a>	<input type="checkbox"/>
2	Received	2017-12-21	10:39:07	192.168.35.3 5@192.168.35.35	Unknown	<a href="#">192.168.35.22</a> <a href="#">2@192.168.35.22</a>	<input type="checkbox"/>
3	Received	2017-12-21	10:38:50	192.168.35.3 5@192.168.35.35	Unknown	<a href="#">192.168.35.22</a> <a href="#">2@192.168.35.22</a>	<input type="checkbox"/>
4	Dialed	2017-12-21	09:57:26	11151@47.88.77.14	Unknown	<a href="#">11100@47.88.77.14</a>	<input type="checkbox"/>
5	Dialed	2017-12-21	08:48:45	11151@47.88.77.14	Unknown	<a href="#">11100@47.88.77.14</a>	<input type="checkbox"/>
6	Received	2017-12-21	01:59:01	11151@47.88.77.14	Extension 11103	<a href="#">11103@47.88.77.14</a>	<input type="checkbox"/>
7	Dialed	2017-12-21	01:43:21	11151@47.88.77.14	Unknown	<a href="#">11100@47.88.77.14</a>	<input type="checkbox"/>
8	Dialed	2017-12-20	09:25:45	11151@47.88.77.14	Unknown	<a href="#">11100@47.88.77.14</a>	<input type="checkbox"/>
9							<input type="checkbox"/>
10							<input type="checkbox"/>
11							<input type="checkbox"/>
12							<input type="checkbox"/>
13							<input type="checkbox"/>
14							<input type="checkbox"/>
15							<input type="checkbox"/>

Page 1 | Prev | Next | Delete | Delete All

## 4.10. Upgrade

### 4.10.1. Upgrade-Basic

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

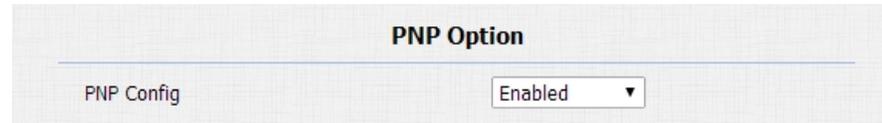
Firmware Version	110.0.2.7
Hardware Version	110.6.0.0.0.0.0
Upgrade	<input type="button" value="选择文件"/> 未选择任何文件 <input type="button" value="Submit"/> <input type="button" value="Cancel"/>
Reset To Factory Setting	<input type="button" value="Submit"/>
Reboot	<input type="button" value="Submit"/>

## 4.10.2. Upgrade-Advanced

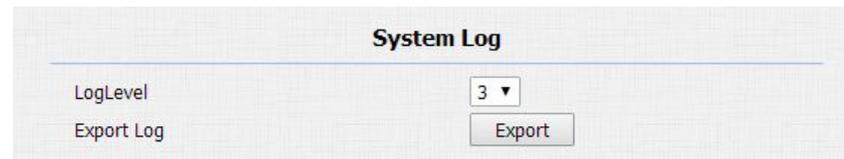
**PNP:** Plug and Play, once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get Auto Provisioning server's address.

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.



The image shows a configuration panel titled "PNP Option". Below the title, there is a label "PNP Config" and a dropdown menu currently set to "Enabled".



The image shows a configuration panel titled "System Log". Below the title, there are two rows: "LogLevel" with a dropdown menu set to "3", and "Export Log" with an "Export" button.

## 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.

#### Web Password Modify

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User Name	<input type="text" value="admin"/>
Current Password	<input type="password"/>
New Password	<input type="password"/>
Confirm Password	<input type="password"/>