

Akuvox Smart
Intercom



R26 Series Door Phone Admin Guide

About this manual

Thank you for choosing Akuvox's R26 series door phone. This manual is intended for end users, who need to properly configure the door phone. It provides all functions and configuration of R26 series, the information detailed in this user manual applicable to firmware version 26.0.2.57.rom or lower version.

- Please verify the packaging content and network status before setting.
- The old firmware may be a little different from 26.0.2.57.rom about some configuration. Please consult your administrator for more information.

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We highly appreciate your feedback about our products

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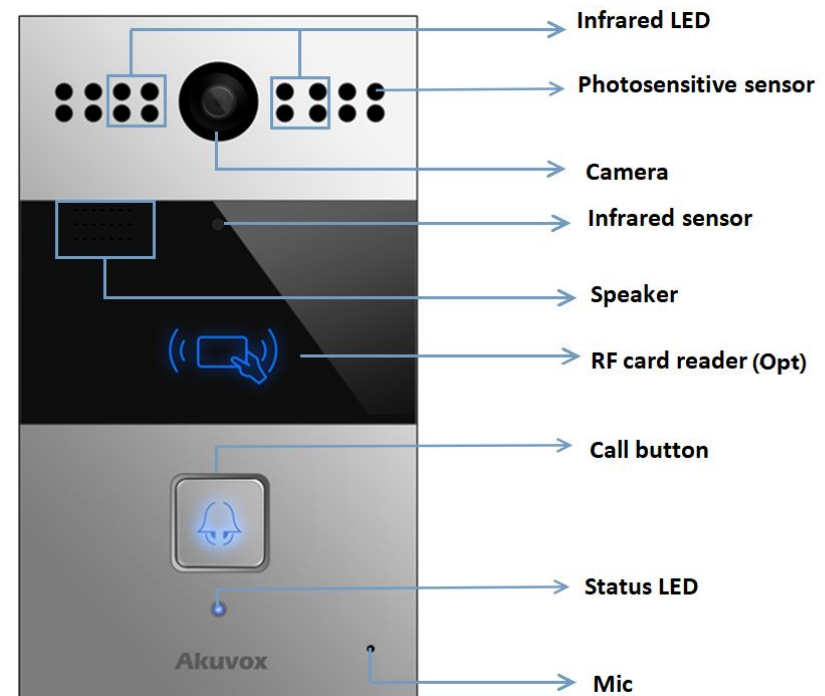
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1. Product Overview

1.1. Product Description

Akuvox R26 series is a SIP-compliant, hands-free one button video outdoor phone. It can be connected with your Akuvox IP Phone for remote unlock control and monitor. Users can operate the indoor phone to communicate with visitors via voice and video, and use RF card to unlock the door (R26C only). It's applicable in villas, office and so on.



1.2. Daily Use

1.2.1. 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.

1.2.2. Receiving a Call

User can use IP phone or indoor monitor to call R26 series and R26 series will answer it automatically by default. If user disable auto answer, pressing button to answer incoming call.

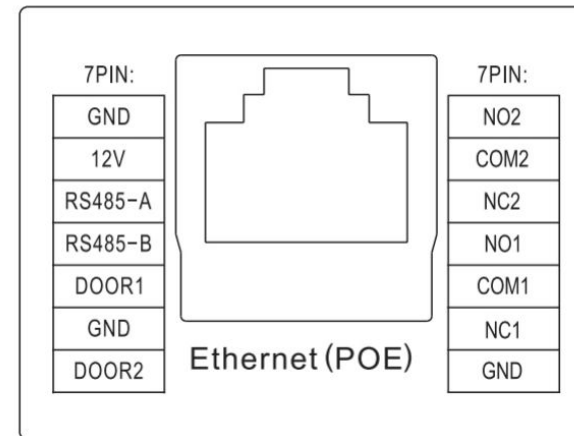
1.2.3. Unlock by RF Card (R26C only)

Place the predefined RF card on the card sensor area. The door phone will announce 'the door is now opened' and open the door. 13.56MHz RF card is supported by R26C.

1.3. Connector Introduction

Connector	
Ethernet(POE)	Ethernet(POE) connector which can provide both power and network connection.
12V/GND	External power supply terminal if POE is not available.
RS485A/B	RS485 terminal for automation system control(e.g. Elevator control).
DOORA/B	Trigger signal input terminal(e.g. Press indoor button to open relay).
RelayA/B	NO/NC Relay control terminal.

Notes: 12V/1A DC from LPS(Power cord≤ 3m) or POE.



2. Basic Setting

2.1. Getting Started

2.1.1. IP Announcement

While R26 series starts up normally, hold the call button for several seconds after the Status LED turns blue, voice system will enter IP announcement mode. In announcement mode, the IP address will be announced periodically and “IP 0.0.0.0” would be announced if no IP address is gained. Press Call Button again to quit the announcement mode.

2.1.2. 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**

2.2. Network Setting

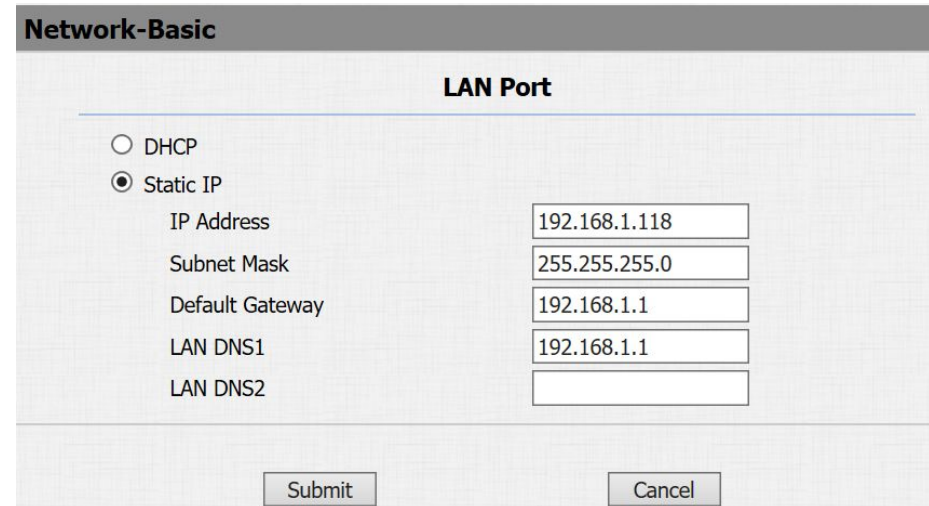
Go to Network->Basic, dynamically or statically to obtain address.

2.2.1. DHCP

R26 series uses DHCP by default, it will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically.

2.2.2. Static IP

If selected, you could manually set IP address, Subnet Mask, Default Gateway and DNS server. The figure below shows static IP setting.



The screenshot shows a configuration window titled "Network-Basic" with a sub-section "LAN Port". Under "LAN Port", there are two radio buttons: "DHCP" (unselected) and "Static IP" (selected). Below the "Static IP" option, there are five input fields for network 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
LAN DNS2	

At the bottom of the window, there are two buttons: "Submit" and "Cancel".

2.3. Account

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

2.3.1. 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.

2.3.2. SIP Server 1

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

The screenshot shows the 'Account-Basic' configuration page. It is divided into two main sections: 'SIP Account' and 'SIP Server 1'. The 'SIP Account' section includes fields for Status (Registered), Account (Account 1), Account Active (Enabled), Display Label (11151), Display Name (R20), Register Name (11151), User Name (11151), and Password (masked with dots). The 'SIP Server 1' section includes fields for Server IP (47.88.77.14), Port (5070), and Registration Period (1800, with a note '(30~65535s)').

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

SIP Server 1			
Server IP	47.88.77.14	Port	5070
Registration Period	1800		(30~65535s)

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

2.3.3. SIP Server 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.

2.3.4. Outbound Proxy Server

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

SIP Server 2		
Server IP	<input type="text" value="44.88.77.15"/>	Port <input type="text" value="5060"/>
Registration Period	<input type="text" value="1800"/>	(30~65535s)

Outbound Proxy Server		
Enable Outbound	<input type="text" value="Enabled"/>	
Server IP	<input type="text" value="75.33.92.180"/>	Port <input type="text" value="5060"/>
Backup Server IP	<input type="text"/>	Port <input type="text" value="5060"/>

2.3.5. Transport Type

To display and configure Transport type for SIP message

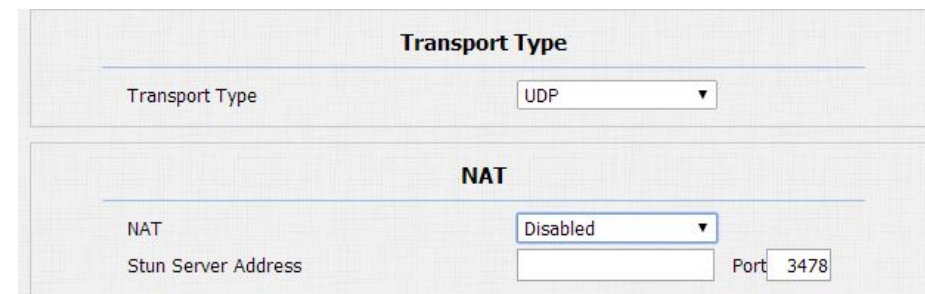
- UDP: UDP is an unreliable but very efficient transport layer protocol.
- TCP: Reliable but less-efficient transport layer protocol.
- TLS: Secured and Reliable transport layer protocol.
- DNS-SRV: A DNS RR for specifying the location of services.

2.3.6. 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 displays two configuration sections. The first section, titled "Transport Type", contains a dropdown menu labeled "Transport Type" with "UDP" selected. The second section, titled "NAT", contains a dropdown menu labeled "NAT" with "Disabled" selected, a text input field for "Stun Server Address" which is empty, and a "Port" field with the value "3478".

2.4. Call Setting

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

2.4.1. No Answer Call

Enable it, if there is no answer from push button number over 60s (default value), R26 series will call predefined 'No Answer Call' number.

2.4.2. Push Button

Push Button: To configure the destination number or IP you want to contact with. If you would like to call multiple numbers at the same time, divide them by semicolon.

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

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

Basic Section:

- Select Account: Auto (dropdown)
- No Answer Call: Disabled (dropdown)
- No Answer Action: Disabled (dropdown)

Push Button Section:

Key	Number	Number2	Number3	Number4
Push Button	132	444	235	124
No Answer Call1				
No Answer Call2				

2.4.3. 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.

2.4.4. Web Call

To dial out or answer incoming call from website.

2.4.5. 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 a configuration interface with five sections, each separated by a horizontal line:

- PushButton Action:** Contains radio buttons for "FTP", "Email", and "Http URL". Below is a text input field labeled "Http URL:".
- Web Call:** Contains a text input field for "Web Call(Ready)", a dropdown menu set to "Auto", and two buttons labeled "Dial Out" and "Hang Up".
- Max Call Time:** Contains a text input field for "Max Call Time" with the value "5" and a range indicator "(2~120Minutes)".
- Max Dial Time:** Contains two text input fields: "Dial In Time" with value "60" and "Dial Out Time" with value "60", both with range indicators "(30~120Sec)".
- Push To Hang Up:** Contains a dropdown menu for "Push To Hang Up" set to "Enabled".

2.4.6. Push to Hang up

To enable or disable pushing button to hang up.

2.5. Action

Go to Intercom->Action to set action receiver.

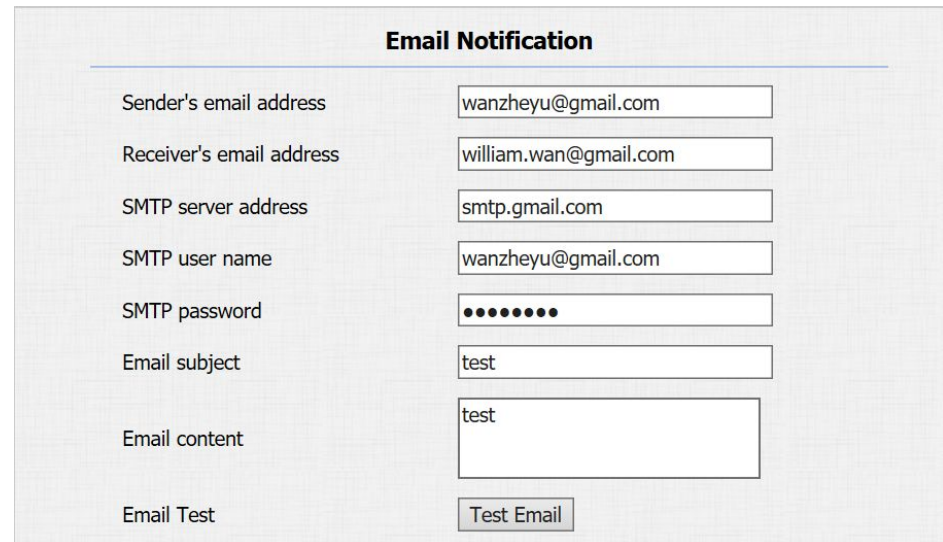
2.5.1. Email Notification

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



The screenshot shows a configuration form titled "Email Notification". It contains several input fields and a button:

Email Notification	
Sender's email address	<input type="text" value="wanzheyu@gmail.com"/>
Receiver's email address	<input type="text" value="william.wan@gmail.com"/>
SMTP server address	<input type="text" value="smtp.gmail.com"/>
SMTP user name	<input type="text" value="wanzheyu@gmail.com"/>
SMTP password	<input type="password" value="••••••••"/>
Email subject	<input type="text" value="test"/>
Email content	<input type="text" value="test"/>
Email Test	<input type="button" value="Test Email"/>

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.

2.5.2. 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.

2.5.3. SIP Notification

SIP Call Number: To configure sip call number.

SIP Call Name: To configure display name of R26 series.

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"/>

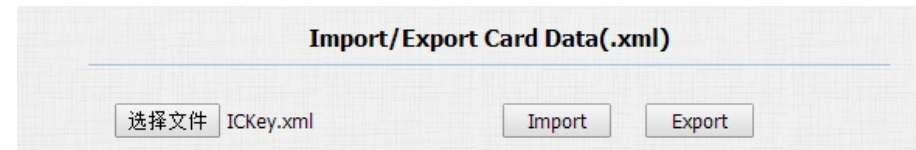
SIP Call Notification	
SIP Call Number	<input type="text" value="1101"/>
SIP Caller Name	<input type="text" value="william"/>

2.6. Card Setting (R26C only)

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

2.6.1. Import/Export Card Data

R26 series supports import or export the card data file, which is convenient for administrator to deal with a large number of cards. The maximum RF card is 500.



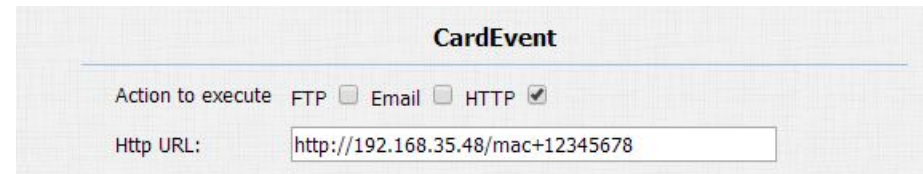
2.6.2. CardEvent

Once users make a call , it will execute the action.

It supports 3 types - FTP, Email, and HTTP.

To setup the FTP and Email in Action interface, the FTP server and Email will receive the capture picture when reading card. If you choose HTTP mode, enter the URL information.

URL format: http://http server IP address/any information.



2.6.3. Obtain and Add Card

- ① Switch card status to 'Card Issuing' and click 'Apply' ;
- ② Place card on the card reader area and click 'Obtain' ;
- ③ Name card and choose which door you want to open ;
- ④ Click 'Add' to add it into list .

Notes: User can use card to access only when card status has been switched to 'Normal'.

2.6.4. Door Card Management

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

Notes: Remember to set Card Status back to Normal after adding the cards.

Card Status

Card Status: Card Issuing [Apply]

Card Setting

IC Key DoorNum: RelayA RelayB RelayC

IC Key Day: Mon Tue Wed Thur Fri Sat Sun Check All

IC Key Time: 00 : 00 - 23 : 00

IC Key Name: TEST1

IC Key Code: A6F20C85 [Obtain] [Add]

Door Card Management

Index	Name	Code	Relay	
1	TEST1	A6F20C85	1	<input type="checkbox"/>
2	TEST2	00013131	1	<input type="checkbox"/>
3				<input type="checkbox"/>
4				<input type="checkbox"/>
5				<input type="checkbox"/>
6				<input type="checkbox"/>
7				<input type="checkbox"/>
8				<input type="checkbox"/>
9				<input type="checkbox"/>
10				<input type="checkbox"/>

Page: 1 [Prev] [Next] [Delete] [DeleteAll]

2.7. Relay Setting

Go to Intercom->Relay, to configure relay.

2.7.1. Relay

There are three terminal of relay: NO, NC and COM. NO stands for normally open contact while NC stands for normally closed contact.

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

Relay ID: R26 series supports two relays, user can configure them respectively.

Relay Type: Default state means NC and COM are normally closed, while Invert state means NC and COM are normally opened.

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

DTMF Option: To select digit of DTMF code, R26 series 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 .

2.7.2. 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.

Relay		
Relay ID	RelayA ▾	RelayB ▾
Relay Type	Default state ▾	Default state ▾
Relay Delay(sec)	3 ▾	3 ▾
DTMF Option	1 Digit DTMF ▾	
DTMF	# ▾	0 ▾
Multiple DTMF		
Relay Status	RelayA: Low	RelayB: Low

Open Relay via HTTP	
Switch	Disabled ▾
UserName	<input type="text"/>
Password	<input type="password"/>

URL format:

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

2.8. Input

R26 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.

Trigger Option: To choose open circuit trigger or closed circuit trigger. Low means that connection between Door terminal and GND is closed, while High means the connection is opened.

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

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

Open relay: To configure relay to open

The screenshot displays the 'Input' configuration page, which is divided into two sections: 'Input A' and 'Input B'. Each section contains the following fields:

- Input Service:** A dropdown menu currently set to 'Disabled'.
- Trigger Option:** A dropdown menu currently set to 'Low'.
- Action to execute:** A row of four checkboxes labeled 'FTP', 'Email', 'Sip Call', and 'HTTP', all of which are currently unchecked.
- Http URL:** An empty text input field.
- Open Relay:** A dropdown menu currently set to 'None'.
- Door Status:** A text label indicating the status, 'DoorA: High' for Input A and 'DoorB: High' for Input B.

Door status: To show the status of input signal.

3. Advance Setting

3.1. Intercom-Advanced

Photoresistor: The setting is for night vision, when the surrounding of R26 series is very dark, infrared LED will turn on and R26 will turn to night mode. Photoresistor value relates to light intensity and larger value means that light intensity is smaller. User can configure the upper and lower bound and when photoresistor value is larger than upper bound, infrared LED will turn on. As contrast, when photoresistor value is smaller than lower bound, infrared LED will turn off and device turns to normal mode.

Tamper Alarm: R26 series integrates internal gravity sensor for the own security, and after enabling Tamper Alarm, if the

The screenshot displays three distinct configuration sections for the R26 series device:

- Photoresistor:** This section contains a 'Photoresistor Setting' field with two input boxes. The first box contains the value '5' and the second box contains '37', with a range '(0~100)' indicated to the right.
- Tamper Alarm:** This section includes a 'Tamper Alarm' dropdown menu currently set to 'Disabled' and a 'Gravity Sensor Threshold' input field containing the value '32', with a range '(0~127)' indicated to the right.
- RFID:** This section features an 'RFID Display Mode' dropdown menu currently set to '8HN'.

gravity of R26 series changes dramatically, the phone will alarm. Gravity Sensor Threshold stands for sensitivity of sensor.

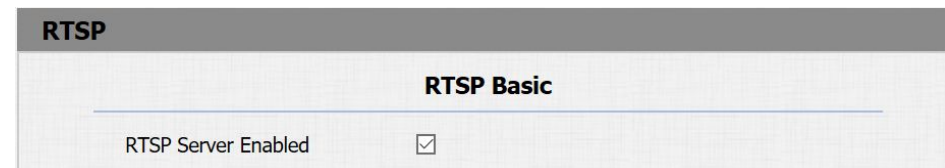
RFID: To be compatible different card number formats in different systems. The default 8HIN means hexadecimal.

3.2. Live Stream

Go to Intercom->Live Stream, check the real-time video from R26 series. In addition, user also can check the real-time picture via URL: http://IP_address:8080/picture.jpg

3.3. RTSP

R26 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.



RTSP Stream: To enable RTSP video and select the video codec. R26C/P supports H264 video codec.

H.264 Video Parameters: H264 is a video stream compression standard. Different from H263, 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 H264

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.

MJPEG Video Parameters: MJPEG is one of the network video image Compression standard. It supports the maximum Compression ratio 4000:1. It is an important and common

The image shows a configuration interface for RTSP Stream and Video Parameters. It is divided into three sections: RTSP Stream, H.264 Video Parameters, and MJPEG Video Parameters. Each section contains several settings with checkboxes or dropdown menus.

RTSP Stream	
RTSP Audio Enabled	<input type="checkbox"/>
RTSP Video Enabled	<input checked="" type="checkbox"/>
RTSP Video Codec	H.264

H.264 Video Parameters	
Video Resolution	VGA
Video Framerate	30 fps
Video Bitrate	2048 kbps

MPEG4 Video Parameters	
Video Resolution	VGA
Video Framerate	30 fps
Video Bitrate	2048 kbps

MJPEG Video Parameters	
Video Resolution	VGA
Video Framerate	30 fps
Video Quality	90

video function with great communication application integration ability and less core program space.

To modify the resolution, framerate and bitrate of MPEG4.

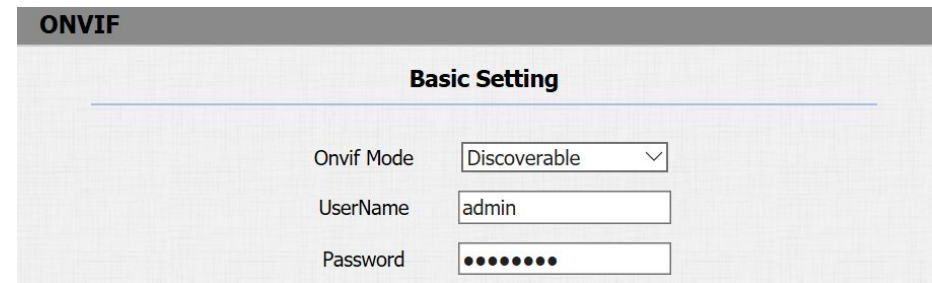
3.4. Onvif

R26 series supports ONVIF protocol, which means R26 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



The screenshot shows a web interface for configuring ONVIF settings. The page has a dark grey header with the text "ONVIF" in white. Below the header, the title "Basic Setting" is centered. The configuration area contains three fields: "Onvif Mode" with a dropdown menu set to "Discoverable", "UserName" with a text input field containing "admin", and "Password" with a text input field containing ten black dots.

3.5. Motion

R26 series supports motion detection, go to Intercom->Motion to configure detection parameter.

Motion Detection: To enable or disable Motion Detection

Motion Delay: To configure minimum time gap between two snapshot.

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

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

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

Motion Detection

Motion Detection Options

Motion Detection

Motion Delay (0~120 Sec)

Action to execute

Action to execute FTP Email Sip Call HTTP

Http URL:

Motion Detect Time Setting

Mon Tue Wed Thur

Fri Sat Sun Check All

: - :

3.6. Account-Advanced

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

3.6.1. Audio Codec

Sip Account: To choose which account to configure.

Audio Codec: R26 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

The screenshot shows the 'Account-Advanced' configuration page for a SIP Account. The 'SIP Account' section has a dropdown menu for 'Account' set to 'Account 1'. The 'Codecs' section features two lists: 'Disabled Codecs' (currently empty) and 'Enabled Codecs' (containing PCMU, PCMA, G729, and G722). Navigation buttons include '>>' and '<<' between the lists, and up/down arrows on the right side of the 'Enabled Codecs' list.

3.6.2. Video Codec

R26 series supports H264 standard, which provides better video quality at substantially lower bit rates than previous standards.

Codec Resolution: R26 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.

3.6.3. 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.

Video Codec	
Codec Name	<input checked="" type="checkbox"/> H264
Codec Resolution	4CIF ▼
Codec Bitrate	2048 ▼
Codec Payload	104 ▼

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

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.

3.6.4. DTMF

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

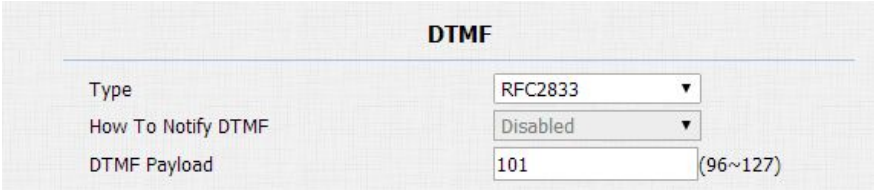
Type: Support Inband, Info, RFC2833 or their combination.

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

DTMF Payload: To configure payload type for DTMF.

3.6.5. Call

Max Local SIP Port: To configure maximum local sip port for designated SIP account.



The image shows a configuration window titled "DTMF". It contains three settings:

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

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, R26 series will block its information when calling out.

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

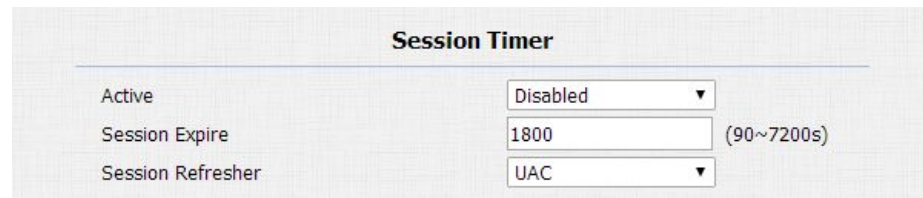
Missed Call Log: If enabled, any missed call will be recorded into call log.

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

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

3.6.6. 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 the 'Session Timer' configuration panel. It contains three settings: 'Active' is set to 'Disabled' via a dropdown menu; 'Session Expire' is set to '1800' in a text input field, with a range '(90~7200s)' indicated to the right; 'Session Refresher' is set to 'UAC' via a dropdown menu.

3.6.7. Encryption

If enabled, voice will be encrypted.



The screenshot shows two configuration panels. The top panel is 'Encryption', with 'Voice Encryption(SRTP)' set to 'Disabled' via a dropdown menu. The bottom panel is 'NAT', with three settings: 'UDP Keep Alive Messages' is set to 'Disabled' via a dropdown menu; 'UDP Alive Msg Interval' is set to '30' in a text input field, with a range '(5~60s)' indicated to the right; 'RPort' is set to 'Disabled' via a dropdown menu.

3.6.8. 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.

3.6.9. User agent

To customize User Agent field in the SIP message; If user agent is set to specific value, user could see the information from network package. If user agent is not set by default, user could see the company name, model number and firmware version from network package.

3.7. Time/Lang

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

User Agent	
User Agent	Akuvox

Time/Lang	
NTP	
Time Zone	0 GMT
Primary Server	0.pool.ntp.org
Secondary Server	1.pool.ntp.org
Update Interval	3600 (>= 3600s)
System Time	10:54:38

3.8. 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.

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

3.9. Voice

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

Mic Volume:To configure Microphone volume.

Speaker Volume:To configure Speaker volume.

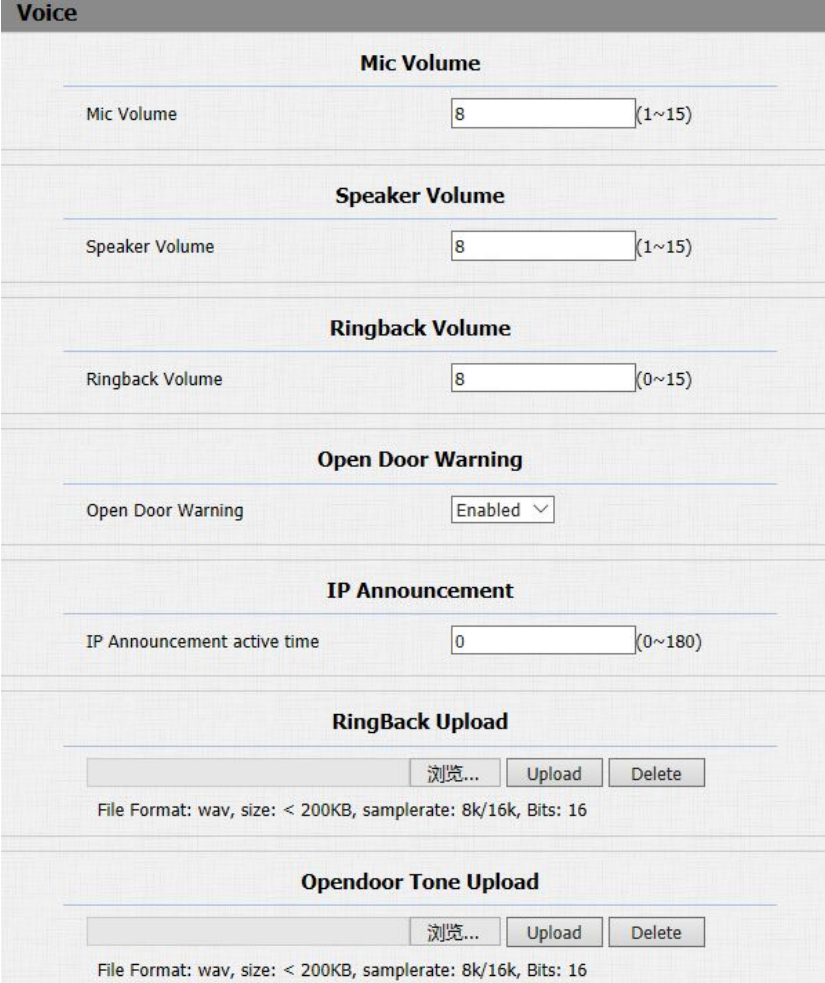
Ringback Volume:To configure Speaker volume.

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

IP Announcement:To setup the IP Announcement active time. Over the configured value, the phone will not announce its IP address, even you hold the button.

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

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



Voice

Mic Volume

Mic Volume (1~15)

Speaker Volume

Speaker Volume (1~15)

Ringback Volume

Ringback Volume (0~15)

Open Door Warning

Open Door Warning

IP Announcement

IP Announcement active time (0~180)

RingBack Upload

File Format: wav, size: < 200KB, samplerate: 8k/16k, Bits: 16

Opendoor Tone Upload

File Format: wav, size: < 200KB, samplerate: 8k/16k, Bits: 16

3.10. 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.

Multicast Setting

Paging Barge

Paging Priority Active

Priority List

IP Address	Listening Address	Label	Priority
1 IP Address	<input type="text" value="224.1.6.11:12000"/>	<input type="text" value="test1"/>	1
2 IP Address	<input type="text"/>	<input type="text"/>	2
3 IP Address	<input type="text"/>	<input type="text"/>	3
4 IP Address	<input type="text"/>	<input type="text"/>	4
5 IP Address	<input type="text"/>	<input type="text"/>	5
6 IP Address	<input type="text"/>	<input type="text"/>	6
7 IP Address	<input type="text"/>	<input type="text"/>	7
8 IP Address	<input type="text"/>	<input type="text"/>	8
9 IP Address	<input type="text"/>	<input type="text"/>	9
10 IP Address	<input type="text"/>	<input type="text"/>	10

3.11. Log

3.11.1. Call Log

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

3.11.2. Door Log

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

3.12. Webrelay

R26C/P can support extra web relay. This function is more safety to use DTMF code to remote unlock.

Type: Connect web relay and choose the type.

The image shows two screenshots of a web interface. The top screenshot is titled 'Call Log' and displays a 'Call History' table with columns for Index, Type, Date, Time, Local Identity, Name, and Number. It lists several call records, including received and dialed calls, with associated phone numbers and extensions. The bottom screenshot is titled 'Door Log' and displays a table with columns for Index, Name, Code, Date, and Time. It shows a single record for a door log entry with the name 'William' and code '57FAC741'.

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	192.168.35.78@192.168.35.78	<input type="checkbox"/>
2	Received	2017-12-21	10:39:07	192.168.35.3 5@192.168.35.35	Unknown	192.168.35.22@192.168.35.22	<input type="checkbox"/>
3	Received	2017-12-21	10:38:50	192.168.35.3 5@192.168.35.35	Unknown	192.168.35.22@192.168.35.22	<input type="checkbox"/>
4	Dialed	2017-12-21	09:57:26	11151@47.88.77.14	Unknown	11100@47.88.77.14	<input type="checkbox"/>
5	Dialed	2017-12-21	08:48:45	11151@47.88.77.14	Unknown	11100@47.88.77.14	<input type="checkbox"/>
6	Received	2017-12-21	01:59:01	11151@47.88.77.14	Extension 11103	11103@47.88.77.14	<input type="checkbox"/>
7	Dialed	2017-12-21	01:43:21	11151@47.88.77.14	Unknown	11100@47.88.77.14	<input type="checkbox"/>
8	Dialed	2017-12-20	09:25:45	11151@47.88.77.14	Unknown	11100@47.88.77.14	<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

Index	Name	Code	Date	Time	
1	William	57FAC741	2017-12-22	10:30:34	<input type="checkbox"/>
2					<input type="checkbox"/>
3					<input type="checkbox"/>
4					<input type="checkbox"/>
5					<input type="checkbox"/>
6					<input type="checkbox"/>
7					<input type="checkbox"/>
8					<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

IP Address: Enter web relay IP address.

User name: it is an authentication for connecting web relay

password: it is an authentication for connecting web relay

Note: Users can modify username and password in web relay website.

Web Relay Action: Web Relay Action is used to trigger the web relay . The action URL is provided by web relay vendor

Web Relay Key: If the DTMF keys same as the local relay, the web relay will be open with local relay. But if there are different, the web relay is invalid.

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

The screenshot shows a web interface with two main sections: "Web Relay" and "Web Relay Action Setting".

Web Relay Section:

- Type: 2N WebRelay (dropdown menu)
- IP Address: 192.168.1.2 (text input)
- UserName: (empty text input)
- Password: (empty text input with a yellow background)

Web Relay Action Setting Section:

Action ID	Web Relay Action	Web Relay Key	Web Relay Extension
Action ID 01	state.xml?relayState=2	1	
Action ID 02	state.xml?relayState=2	3	
Action ID 03	state.xml?relayState=2	#	192.18.1.168
Action ID 04	state.xml?relayState=2	12	
Action ID 05	state.xml?relayState=2	123	
Action ID 06	state.xml?relayState=2	1234	
Action ID 07			
Action ID 08			
Action ID 09			
Action ID 10			

3.13. Upgrade-Basic

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

Upgrade: Choose .rom firmware from your PC, then click Submit to start update.

Reset To Factory Setting: Directly click Submit to reset R26C/P. Use this function with caution. All configuration will be removed.

Reboot: Click to reboot.

3.14. Upgrade-Advanced

To display and configure manual update server's settings.



The screenshot displays a web interface for the 'Upgrade-Basic' section. It features a table with the following information:

Firmware Version	26.0.3.32
Hardware Version	26.1.0.0.0.0.0.0
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"/>

3.14.1. 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.

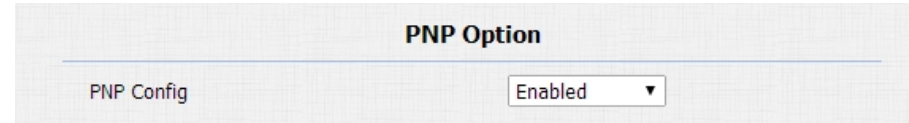
By default, this SIP message is sent to multicast address 224.0.1.75(PNP server address by standard).

3.14.2. DHCP Option

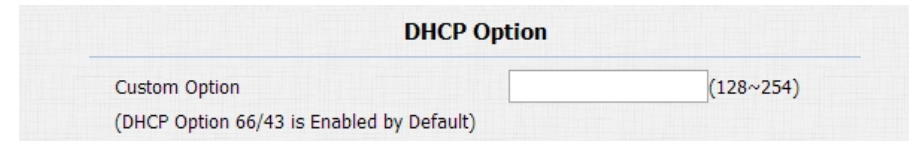
To display and configure DHCP setting for AutoP. Option 66/43 is enable by default. It can support Https,Http,Ftp,Tftp server.

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

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



The screenshot 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 screenshot shows a configuration panel titled "DHCP Option". Below the title, there is a label "Custom Option" followed by an empty text input field and the text "(128~254)". Below the input field, there is a note: "(DHCP Option 66/43 is Enabled by Default)".

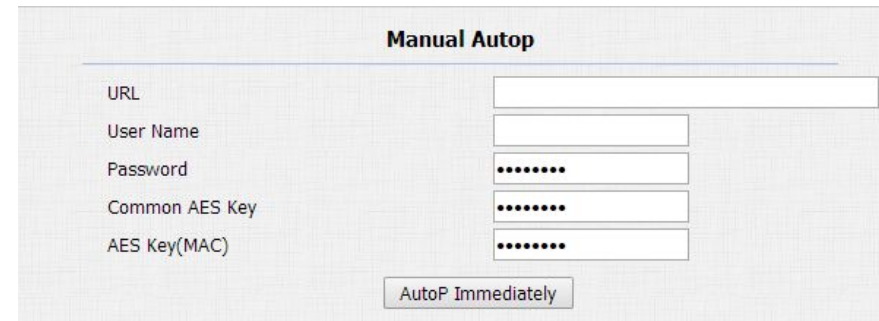
3.14.3. Manual Autop

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.



The screenshot shows a web interface titled "Manual Autop". It contains five input fields for configuration: "URL", "User Name", "Password", "Common AES Key", and "AES Key(MAC)". The "Password", "Common AES Key", and "AES Key(MAC)" fields are masked with dots. Below the input fields is a button labeled "AutoP Immediately".

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.

3.14.4. 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.

Automatic Autop

Mode	Power On
Schedule	Sunday
	22 Hour(0~23)
	0 Min(0~59)
Clear MD5	Submit
Export Autop Template	Export

3.14.5. 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

System Log

LogLevel	3
Export Log	Export

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.

3.14.6. PCAP

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

3.14.7. Others

To export current config file or import new config file.

3.15. Security-Basic

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

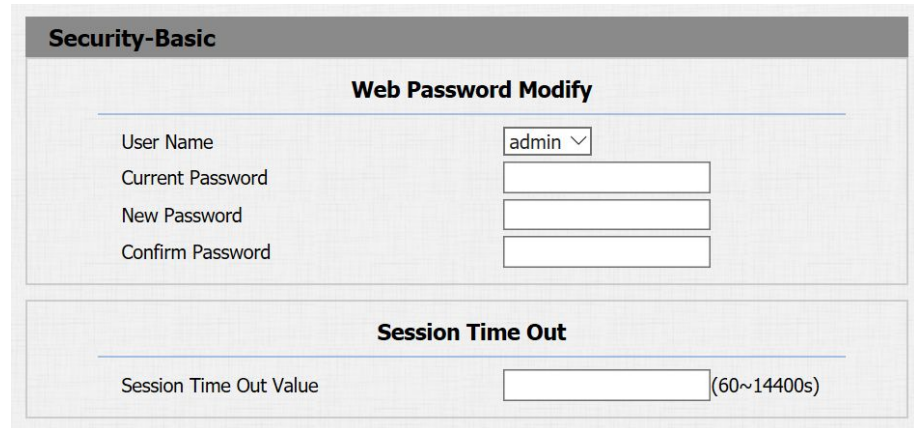


3.15.1. Web Password Modify

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

3.15.2. Session time out

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



Security-Basic

Web Password Modify

User Name

Current Password

New Password

Confirm Password

Session Time Out

Session Time Out Value (60~14400s)