



## **R27A Door Phone User Manual**

## About This Manual

Thank you for choosing Akuvox's products. In user manual, we provide all functions and configurations you want to know about R27A. Please verify the packaging content and network status before setting. This manual applies to firmware 26.0.2.170 or lower version.

Note: The old firmware may be a little different from 27.0.2.170 about some configuration. Please consult your administrator for more information.

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### FCC Caution:

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions : (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Note : This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

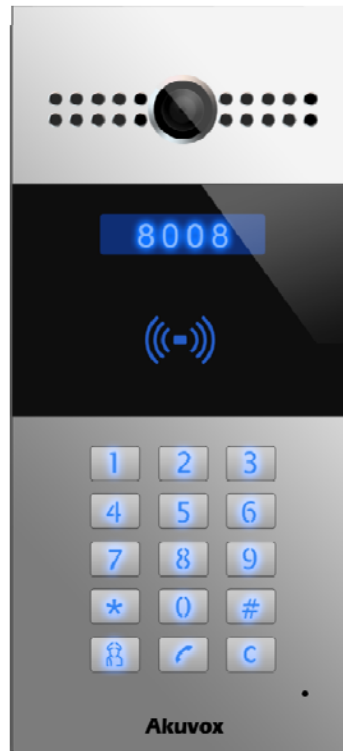
# Content

<b>1. Overview</b> .....	<b>1</b>
1.1. Product Description.....	1
1.2. Features.....	2
1.3. Keypad.....	4
<b>2. Configuration</b> .....	<b>5</b>
2.1. Administrator interface .....	5
2.1.1 System Information .....	5
2.1.2 Admin Settings .....	5
2.1.3 System Setting .....	6
2.2. User interface .....	7
2.2.1 Public Pin Modif .....	7
2.2.2 Add User Cards .....	7
2.2.3 Add Private Pin .....	7
<b>3. Basic Using</b> .....	<b>8</b>
3.1. Make a call .....	8
3.2. Receive a call .....	8
3.3. Unlock.....	8
<b>4. Web</b> .....	<b>9</b>
4.1. Obtain IP address .....	9
4.2. Login the web.....	9
4.3. Status.....	10
4.3.1 Basic .....	10
4.4. Intercom .....	11
4.4.1 Basic .....	11
4.4.2 Advanced.....	12
4.4.3 Relay .....	13
4.4.4 Input.....	15
4.4.5 Live Stream.....	16
4.4.6 RTSP.....	17
4.4.7 ONVIF .....	18
4.4.8 Motion.....	18
4.4.9 Card Setting(R27A only) .....	19
4.4.10 Action .....	20
4.5. Account .....	22
4.5.1 Basic .....	22
4.5.2 Advanced.....	24
4.6. Network.....	27
4.6.1 Basic .....	27

4.6.2 Advanced.....	28
4.7. Phone .....	29
4.7.1 Time/Language .....	29
4.7.2 Call Feature .....	30
4.7.3 Voice.....	31
4.7.4 Dial Plan.....	32
4.7.5 Multicast.....	33
4.7.6 Call log.....	34
4.7.7 Door log.....	35
4.8. Upgrade .....	35
4.8.1 Basic .....	35
4.8.2 Advanced.....	36
4.9. Security.....	38
4.9.1 Basic .....	38

# 1. Overview

## 1.1. Product Description



**R27A**

Akuvox R27A is a SIP-compliant, hands-free and video outdoor phone. It can be connected with your Akuvox IP Phone for remote unlock control and monitor. You can operate the indoor handset to communicate with visitors via voice and video, and unlock the door if you wish. Users can also use RF card to unlock the door( R27AC only). It's applicable in villas, office and so on.

## 1.2. Features

### ➤ **Highlight**

- Vandal resistant body, with a flush button
- Wild-angle camera:120°
- POE(IEEE802.3af, Power-over-Ethernet)
- Two-way audio communication over IP network with Echo cancel feature
- Complies with SIP Standard for easy integration in each SIP PBXes
- Complies with ONVIF standard for easy integration with any network surveillance system

### ➤ **Physical&Power**

- Body material: all-aluminum
- Camera: 3M pixels, automatic lighting
- Numeric keypad with extra buttons
- Infrared Sensor: Support
- Wiegand port: Support
- RF Card Reader:13.56MHz & 125kHz
- Output Relay: 3 output relays for door opener
- 802.3af Power-Over-Ethernet
- 12V DC connector(if not using POE)
- Water proof&Dust proof: IP65
- Installation: Flush-mounted & Wall-mounted
- Flush-mounted DIM:280x130x68mm
- Wall-mounted DIM:280x130x38mm

### ➤ **SIP Endpoint**

- SIP v1(RFC2543), SIP v2(RFC3261)
- Audio codecs: G.711a, G.711  $\mu$  , G.722, G.729
- Video codecs: H264
- Speech Quality: 7kHz Audio
- Echo Cancellation
- Voice Activation Detection
- Comfort Noise Generator

### ➤ **Video**

- Resolution: up to 720p
- Maximum image transfer rate:720p-30pfs
- High intensity IR LEDs for picture lighting during dark hours with internal light sensor
- Compatible with 3<sup>rd</sup>.Party.Video components,e.g.NVRs.

### ➤ **Door Entry Feature**





- Relay control individually by DTMF tones
- Camera permanently operational
- White Balance: Auto
- Auto-night mode with LED illumination
- Minimum illumination: 0.1LUX

### ➤ **Network Features**

- 1x10/100Mbps Ethernet Port

- Protocols support: IPv4, HTTP, HTTPS, FTP, SNMP, DNS, NTP, RTSP, RTP, TCP, UDP, ICMP, DHCP, ARP

### 1.3. Keypad

Key	Description
	Numeric Key
	Manage Center Key
	Dialing Key
	Delete Key



## ***2. Configuration***

### **2.1. Administrator interface**

Press \*2396# to enter administrator interface. Administrator interface provides some advanced permissions to administrators, including System Information, Admin Settings and System Settings.

#### **2.1.1 System Information**

Press 1 to enter System Information to check IP address, Mac address and Firmware version of the door phone.

#### **2.1.2 Admin Settings**

##### **2.1.2.1 Admin card setting**


###### **Add admin card**

Enter Admin Card Setting interface, and press 1 to quick add admin card. When you see “Please Swipe Admin Card...”, please place admin card in the RF card reader area. After the screen shows “An admin card is added +1”, it means adding successfully.


###### **Clean admin card data**

Enter Admin Card Setting interface, press 2 to delete the current admin card. When you see “Please Swipe Admin Card...”, and place the added admin card you want to delete in the RF card area. After the screen shows “An admin card is deleted”, it means deleting successfully.

### **2.1.2.2 Admin Code Setting**


Admin code is used to enter administrator interface. The default code is 2396. Enter Admin Code Setting to input 4 digit new admin codes, click Dial key  to save.

### **2.1.2.3 Service Code Setting**

Service Code Setting is used to enter user interface. The default code is 3888. Enter service code setting to input 4 digit new user codes, and click Dial key  to save.

## **2.1.3 System Setting**

### **2.1.3.1 Network settings**

Enter System Setting interface, and press 1 to enter Network setting. Select DHCP mode, door phone will access network automatically. Choose Static mode, users need to setup IP address, subnet mask and default gateway. Press Dial key  when you finish each step.

### **2.1.3.2 Station No.Settings**

Users can setup the device ID to limit the unlock permissions.

(This function can not be used now. Akuvox will perfect it in next version )


### **2.1.3.3 Restore default**

Enter System setting, and press 3 to enter restore interface. After you sure to make the device restore to factory setting, you can swipe your admin card or enter admin code, then the device will restore.

## 2.2. User interface

Press \*3888# to enter user interface. User interface includes Public Pin Modif, Add User Cards and Add Private Pin. These functions can only be accessed by administrator.


### 2.2.1 Public Pin Modif

The default public Pin is 33333333. Before you modify public Pin, users need to swipe admin card or enter admin code, then you can enter 8 digit new Public Pin, click Dial key  to save.

### 2.2.2 Add User Cards


User card is used to unlock. Before adding users card, users need to swipe admin card or enter admin code, then you will see “Please Swipe IC Card...”, place user card in the RF card reader . Then the screen will show “Add IC Card +1”, it means adding successfully.

### 2.2.3 Add Private Pin

Users can also use private pin code to unlock. Before adding private pin , users need to swipe admin card or enter admin code. Then enter a 8 digit private pin, and click Dial key  to save.

## ***3. Basic Using***

### **3.1. Make a call**

In the idle interface, press the account or IP address + Dial key  to make a call.

### **3.2. Receive a call**

R27X will auto answer the incoming call by default. If users disable auto answer function, they can press dial key to answer the incoming call.

### **3.3. Unlock**

**Unlock by Pin code:** Users can unlock the door by using predefined Public Pin or Private Pin. Press # + 8digit Pin Code + # to unlock, then you will hear “The door is now opened”. If users input the wrong Pin code, the screen will show “Incorrect Code”.

**Unlock by RF Card(Only R27A):** Place the predefined user card in RF card reader to unlock. Under normal conditions, the phone will announce “The door is now opened”. If the card has not been registered, the phone will show “ Unauthorized”.

**Unlock by DTMF Code:** During the talking, the president can press the predefined DTMF code to remote unlock the door. ( Please refer to chapter 4.4.4 about DTMF code setting). Then you will also hear “ The door is now opened”.

# 4. Web

## 4.1. Obtain IP address

The Akuvox R27A use DHCP IP by default. Press \*2396# to enter Administrator interface. Enter System Information to check the phone IP address.

## 4.2. Login the web

Open a Web Browser, and enter the corresponding IP address. Then, type the default user name and password as below to log in:

User name: admin

Password: admin

The screenshot shows a web browser window displaying the login page for the Akuvox R27A administrator interface. The page has a blue header and a yellow border. The main content area is white and contains a 'Login' form with the following elements:

- Login** (header)
- User Name** (label) and a text input field.
- Password** (label) and a text input field.
- Remember Username/Password (checkbox)
- Login** (button)

On the right side of the page, there is a 'Help' link with the text 'Login Page' below it.

## 4.3. Status

### 4.3.1 Basic

Status, including product information, network information and account information, can be viewed from Status -> Basic.

Status	
<b>Product Information</b>	
Model	R27-A
MAC Address	0C:11:05:05:63:AE
Firmware Version	27.0.2.170
Hardware Version	27.0.0.0.0.0.0
<b>Network Information</b>	
LAN Port Type	Static IP
LAN Link Status	Connected
LAN IP Address	192.168.35.2
LAN Subnet Mask	255.255.255.0
LAN Gateway	192.168.35.1
LAN DNS1	192.168.35.1
LAN DNS2	8.8.8.8
<b>Account Information</b>	
Account1	1009@192.168.35.250 Registered
Account2	None@None Disabled

Sections	Description
<b>Product Information</b>	To display the device's information such as Model name, MAC address (IP device's physical address), Firmware version and Hardware firmware.
<b>Network Information</b>	To display the device's Networking status(LAN Port),such as Port Type(which could be DHCP/Static/PPPoE), Link Status, IP Address, Subnet Mask, Gateway, Primary DNS server, Secondary DNS server, Primary NTP server and Secondary NTP server(NTP server is used to synchronize time from INTERNET automatically).
<b>Account Information</b>	To display device's Account information and Registration status (account username, registered server's address, Register result).

## 4.4. Intercom

### 4.4.1 Basic

Go to the path: Intercom-Basic

**Intercom-Basic**

**Public Key**

Key Switch  ▾

Send Key  ▾

Key Value  (4-8 digit number)

**Display Number**

Display Number  ▾

**Speed Dial**

Key	Number
Speed Dial	<input type="text"/>
Speed Dial2	<input type="text"/>
Speed Dial3	<input type="text"/>
Speed Dial4	<input type="text"/>

**Call Event**

Action to Execute  FTP  Email  Http URL

Http URL:

**Web Call**

Web Call(Ready)   ▾

**Max Call Time**

Max Call Time  (2~30Minutes)

**Max Dial Time**

Dial In Time  (30~120Sec)

Dial Out Time  (30~120Sec)

Sections	Description
<b>Public Key</b>	Public Key is used to unlock. <ul style="list-style-type: none"> <li>● Key Switch: Users can disable or enable this function.</li> <li>● Key Value: The default public Key is 33333333. Users can modify by yourself.</li> </ul>
<b>Display Number</b>	This function is used to hide or display the number when you operate in the phone. If you select “Disabled”, the phone will show “*” when you dial.
<b>Speed Dial</b>	This Feature is used to call out 4 numbers in the same time. After setup the number you need to call, press manage

	center key to call .
<b>Web Call</b>	To dial out or answer the phone from website.
<b>Call Event</b>	<p>This feature is similar with the Input event. Once users make a call , it will execute the action.</p> <p>It supports 3 types - FTP, Email, and HTTP</p> <p>To setup the FTP and Email in Action interface, the FTP server and Email will receive the capture picture when call out. If you choose HTTP mode, enter the URL format: http://http server IP address/any information (such as, http://192.168.35.48/mac=000 ).Then you will check this information which captures the network packet.</p>
<b>Max Call Time</b>	To configure the max call time.
<b>Max Dial Time</b>	<ul style="list-style-type: none"> <li>● Dial in Time: When other phone calls to R27A, if ring tone is over the Dial in Time without answer. The call will be hang up.</li> <li>● Dial out Time: When R27A call to the other party, if the ringtone is over the Dial out Time without answer. R27A will continue to call to no answer call number in order.</li> </ul>

#### 4.4.2 Advanced

**Intercom-Advanced**

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**AEC Setting**

AEC Level

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**Photoresistor**

Photoresistor Setting  -  (0~100)

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**Tamper Alarm**

Tamper Alarm  ▼

Gravity Sensor Threshold  (0~127)

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**Wiegand**

WiegandType  ▼

Sections	Description
<b>AEC Level</b>	AEC(Configurable Acoustic and Line Echo Cancelers) is used to adjust the echo effect during the communication. The default value is 700. Increase the level, the echo control is better.



<b>Photoresistor</b>	Photoresistor is used to sense the light intensity that R27A will auto enable infrared LED. Users can adjust the photosensitive value by yourselves.
<b>Tamper Alarm</b>	Enable the Tamper Alarm, if the gravity of R27A changes, the phone will alarm. The Threshold value is smaller, the faster the reaction of device.
<b>Wiegand</b>	Akuvox provides two Wiegand protocol. According to the corresponding wiegand access device to choose the suitable protocol.

### 4.4.3 Relay

Sections	Description
<b>Private Key</b>	<ul style="list-style-type: none"> <li>● Import or Export the Private Key template.</li> </ul>
<b>Relay</b>	<p>To configure some settings about unlock</p> <ul style="list-style-type: none"> <li>● Relay Select: R27A support 3 relays</li> <li>● Relay Type: Different locks use different relay types, default state or invert state. If you connect the Lock in NO connector, select default state. Otherwise using invert</li> </ul>

	<p>state.</p> <ul style="list-style-type: none"> <li>● Relay Delay(sec): Allows door remain “open” for certain period The range is from 1 to 10 seconds</li> <li>● DTMF Option: R27A support 1、 2、 3、 4 digits DTMF unlock code. Please select one type and enter the corresponding code.</li> <li>● DTMF: Setup 1 digit DTMF code for remote unlock</li> <li>● Multiple DTMF: Setup multiple digits DTMF code for remote unlock.</li> <li>● Status: the status will be changed by the relay state.</li> </ul>
<b>Open Relay via HTTP</b>	<p>Users can use a URL to remote unlock the door.</p> <ul style="list-style-type: none"> <li>● Switch: Enable this function. Disable by default.</li> <li>● Username &amp; password: Users can setup the username and password for HTTP unlock. Null by default</li> </ul> <p>URL format:<b>http://192.168.1.102/fcgi/do?action=OpenDoor&amp;UserName=&amp;Password=&amp;DoorNum=1</b></p>

## 4.4.4 Input

**Input**

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**Input A**

Input Service:

Trigger Option:

Action to execute: FTP  Email  Sip Call  HTTP

Http URL:

Action Delay:  (0~300 Sec)

Open Relay:

Door Status: DoorA: High

Light Status: LightA: Warning

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**Input B**

InputB Service:

Trigger Option:

Action to execute: FTP  Email  Sip Call  HTTP

Http URL:

Action Delay:  (0~300 Sec)

Open Relay:

Door Status: DoorB: High

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**Input C**

InputC Service:

Trigger Option:

Action to execute: FTP  Email  Sip Call  HTTP

Http URL:

Action Delay:  (0~300 Sec)

Open Relay:

Door Status: DoorC: High

Sections	Description
<b>Input</b>	<p>Input function is used to open the door from inside.</p> <ul style="list-style-type: none"> <li>● Trigger Option: According to different lock connection to choose different trigger mode. If users connect in normal open contact, select low. If you choose High, please connect in normal close contact.</li> <li>● Action to execute: Choose one or more ways to receive the action message.</li> <li>● Http URL: If you tick Http URL, then enter the Http server IP address in the HTTP URL area. When the Input is triggered, it will send Http message. URL format: http://http server IP address/any information (such as http://192.168.35.48/mac=000 ). Then you will check this</li> </ul>

	<p>information which captures the network packet.</p> <ul style="list-style-type: none"> <li>● Action Delay: Setup the action delay time. After the delay time, the phone will send to the action information in the corresponding way.</li> <li>● Open Relay: To choose a suitable relay for input connector.</li> </ul>
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#### 4.4.5 Live Stream



Sections	Description
Live Stream	To check the real-time video from R27A.

## 4.4.6 RTSP

**RTSP**

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**RTSP Basic**

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RTSP Server Enabled

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**RTSP Stream**

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RTSP Audio Enabled

RTSP Video Enabled

RTSP Video Codec H.264 ▼

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**H.264 Video Parameters**

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Video Resolution VGA ▼

Video Framerate 30 fps ▼

Video Bitrate 2048 kbps ▼

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**MPEG4 Video Parameters**

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Video Resolution VGA ▼

Video Framerate 30 fps ▼

Video Bitrate 2048 kbps ▼

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**MJPEG Video Parameters**

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Video Resolution VGA ▼

Video Framerate 30 fps ▼

Video Quality 90 ▼

Sections	Description
<b>RTSP Basic</b>	To active the RTSP function, then R27A can be monitored. RTSP stream format : rtsp:// device IP/live/ch00_0
<b>RTSP Stream</b>	To enable RTSP video and select the video codec. R27A supports H264 video codec.
<b>H.264 Video Parameters</b>	H264: 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
<b>MPEG4 Video Parameters</b>	MPEG4: It 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
<b>MJPEG Video Parameters</b>	MJPEG: called Motion Joint Photographic Experts Group. It is a video encoding format in which each image is compressed separately by JPEG. MJPEG compression can produce high

	<p>quality video image and has a flexible configuration in video definition and Compressed frames</p> <p>To modify the resolution, framerate and bitrate of MJPEG</p>
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#### 4.4.7 ONVIF

Sections	Description
<b>Basic Setting</b>	<p>To setup the ONVIF function parameters. It is used to connect with the corresponding ONVIF tool.</p> <ul style="list-style-type: none"> <li>● ONVIF Mode: Two modes - Discoverable and Non-discoverable. Discoverable by default. Only Discoverable mode, then Onvif software can search R27X</li> <li>● User Name: To modify the user name you need. Admin by default.</li> <li>● Password: To modify the password you want. Admin by default.</li> </ul> <p><b>Note:</b> User name and password is used for authentication.</p>

#### 4.4.8 Motion

Sections	Description
<b>Motion Detection</b>	<p>Motion detection is used to record the change of the surrounding environment.</p> <ul style="list-style-type: none"> <li>● Motion Detection Options: Enable to active this</li> </ul>

	<p>function.</p> <ul style="list-style-type: none"> <li>Action to execute: Select a suitable way to receive the motion detection information. (FTP,EMAIL,SIP Call setting please refer to chapter4.4.10)</li> </ul>
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#### 4.4.9 Card Setting(R27A only)

Sections	Description
<b>Import/Export Card Data</b>	To import or export the card data file. Only support .xml format.
<b>Card Status</b>	<ul style="list-style-type: none"> <li>Normal: Choose Normal mode when reading card.</li> <li>Card Issuing: Choose Card Issuing mode when writing card.</li> </ul>
<b>Card Event</b>	<p>This feature is similar with the Input event. Once users use card to unlock, it will execute the action.</p> <p>It supports 3 types - FTP, Email, HTTP</p> <p>To setup the FTP and Email in Action interface, the FTP server and Email will receive the capture picture when unlocking. If you choose HTTP mode, enter the URL format: http://http server IP address/any information (such as http://192.168.35.48/mac=000 ).Then you will check</p>

	this information which captures the network packet.
<b>Card Setting</b>	<ul style="list-style-type: none"> <li>● IC Key DoorNum: R27A can support to connect 3 relays Choose one and add the valid card for unlock.</li> <li>● IC Key Day: To choose the valid day for the card you added.</li> <li>● IC Key Time: Setup an accurate valid time for the card.</li> <li>● IC Key Name: To setup corresponding name for the card.</li> <li>● IC Key Code: Place the card in the R27A RF Card Read area, then click Obtain button. After R27A reads the card code, click Add, the card information will show in the Door Card Management list.</li> </ul>
<b>Door Card Management</b>	Valid card information will show in the list. Users can tick the current card information then delete one or all in the list.

#### 4.4.10 Action

**Action**

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**Email Notification**

Sender's email address

Receiver's email address

SMTP server address

SMTP user name

SMTP password

Email subject

Email content

Email Test

---

**FTP Notification**

FTP Server

FTP User Name

FTP Password

FTP Test

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**SIP Call Notification**

SIP Call Number

SIP Caller Name

Sections	Description
<b>Email Notification</b>	<ul style="list-style-type: none"> <li>● Sender Email Address: Input the sender email address</li> <li>● Receiver Email Address: Input the receiver email address</li> <li>● SMTP Server Address: Enter the SMTP server format</li> </ul>



	<ul style="list-style-type: none"> <li>● SMTP User name: Enter the SMTP</li> <li>● SMTP password: Enter the sender email password</li> <li>● Email Subject: Enter the subject name.</li> <li>● Email content: Enter the content name.</li> <li>● Email test: Click test to make sure the parameters you enter is right.</li> </ul>
<b>FTP Notification</b>	<ul style="list-style-type: none"> <li>● FTP Server: Enter the FTP server address.</li> <li>● FTP User Name: Enter the FTP server user name.</li> <li>● FTP Password: Enter the corresponding FTP server password.</li> <li>● FTP test: Click test to make sure the parameters you enter is right.</li> </ul>
<b>SIP Call Notification</b>	<p>When you enable SIP Call function of motion. Enter the number and name in the corresponding area. When the motion is triggered, the device will call out the number automatically.</p>

## 4.5. Account

### 4.5.1 Basic

**Account-Basic**

**SIP Account**

Status	Disabled
Account	Account 1 ▼
Account Active	Disabled ▼
Display Label	<input type="text"/>
Display Name	<input type="text"/>
Register Name	<input type="text"/>
User Name	<input type="text"/>
Password	*****

**SIP Server 1**

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

**SIP Server 2**

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

**Outbound Proxy Server**

Enable Outbound	Disabled ▼		
Server IP	<input type="text"/>	Port	<input type="text" value="5060"/>
Backup Server IP	<input type="text"/>	Port	<input type="text" value="5060"/>

**Transport Type**

Transport Type	UDP ▼
----------------	-------

**NAT**

NAT	Disabled ▼		
Stun Server Address	<input type="text"/>	Port	<input type="text" value="3478"/>

Sections	Description
<b>SIP Account</b>	To display and configure the specific Account settings. <ul style="list-style-type: none"> <li>● Status: To display register result.</li> <li>● Display Name: Which is sent to the other call party for display.</li> <li>● Register Name: Allocated by SIP server provider, used for authentication.</li> <li>● User Name: Allocated by your SIP server provide, used for authentication.</li> <li>● Password: Used for authorization.</li> </ul>
<b>SIP Server 1</b>	To display and configure Primary SIP server settings. <ul style="list-style-type: none"> <li>● Server IP: SIP server address, it could be an URL or IP address.</li> </ul>

	<ul style="list-style-type: none"> <li>● Registration Period: The registration will expire after Registration period, the IP phone will re-register automatically within registration period.</li> </ul>
<b>SIP Server 2</b>	<p>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.</p> <p><b>Note:</b> Secondary SIP server is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.</p>
<b>Outbound Proxy Server</b>	<p>To display and configure Outbound Proxy server settings. An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server.</p> <p><b>Note:</b> If configured, all SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.</p>
<b>Transport Type</b>	<p>To display and configure Transport type for SIP message</p> <ul style="list-style-type: none"> <li>● UDP: UDP is an unreliable but very efficient transport layer protocol.</li> <li>● TCP: Reliable but less-efficient transport layer protocol.</li> <li>● TLS: Secured and Reliable transport layer protocol.</li> <li>● DNS-SRV: A DNS RR for specifying the location of services.</li> </ul>
<b>NAT</b>	<p>To display and configure NAT(Net Address Translator) settings.</p> <ul style="list-style-type: none"> <li>● STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues.</li> </ul> <p><b>Note:</b> By default, NAT is disabled.</p>

## 4.5.2 Advanced

**Account-Advanced**

---

**SIP Account**

Account: Account 1

---

**Codecs**

Disabled Codecs <div style="border: 1px solid #ccc; height: 100px;"></div>	>>  <<	Enabled Codecs PCMU PCMA G722 G729 <div style="border: 1px solid #ccc; height: 100px;"></div>
---	--------------	--

↑  
↓

---

**Video Codec**

Codec Name	<input checked="" type="checkbox"/> H264
Codec Resolution	<span style="border: 1px solid #ccc; padding: 2px;">4CIF</span>
Codec Bitrate	<span style="border: 1px solid #ccc; padding: 2px;">2048</span>
Codec Payload	<span style="border: 1px solid #ccc; padding: 2px;">104</span>

---

**Subscribe**

MWI Subscribe	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
MWI Subscribe Period	<span style="border: 1px solid #ccc; padding: 2px;">1800</span> (120~65535s)
Voice Mail Number	<span style="border: 1px solid #ccc; padding: 2px;"></span>
BLF Expire	<span style="border: 1px solid #ccc; padding: 2px;">1800</span> (120~65535s)
ACD Expire	<span style="border: 1px solid #ccc; padding: 2px;">1800</span> (120~65535s)

---

**DTMF**

Type	<span style="border: 1px solid #ccc; padding: 2px;">RFC2833</span>
How To Notify DTMF	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
DTMF Payload	<span style="border: 1px solid #ccc; padding: 2px;">101</span> (96~127)

---

**Call**

Max Local SIP Port	<span style="border: 1px solid #ccc; padding: 2px;">5062</span> (1024~65535)
Min Local SIP Port	<span style="border: 1px solid #ccc; padding: 2px;">5062</span> (1024~65535)
Caller ID Header	<span style="border: 1px solid #ccc; padding: 2px;">FROM</span>
Auto Answer	<span style="border: 1px solid #ccc; padding: 2px;">Enabled</span>
Provisional Response ACK	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
Register with user=phone	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
Invite with user=phone	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
Anonymous Call	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
Anonymous Call Rejection	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
Missed Call Log	<span style="border: 1px solid #ccc; padding: 2px;">Enabled</span>
Prevent SIP Hacking	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>

---

**Session Timer**

Active	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
Session Expire	<span style="border: 1px solid #ccc; padding: 2px;">1800</span> (90~7200s)
Session Refresher	<span style="border: 1px solid #ccc; padding: 2px;">UAC</span>

---

**BLFList**

BLFList URI	<span style="border: 1px solid #ccc; padding: 2px;"></span>
BLFList Pickup Code	<span style="border: 1px solid #ccc; padding: 2px;"></span>
BLFList BargeIn Code	<span style="border: 1px solid #ccc; padding: 2px;"></span>

---

**Encryption**

Voice Encryption(SRTP)	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
------------------------	---

---

**NAT**

UDP Keep Alive Messages	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>
UDP Alive Msg Interval	<span style="border: 1px solid #ccc; padding: 2px;">30</span> (5~60s)
RPort	<span style="border: 1px solid #ccc; padding: 2px;">Disabled</span>

---

**User Agent**

User Agent	<span style="border: 1px solid #ccc; padding: 2px;"></span>
------------	---

Sections	Description
<b>SIP Account</b>	To display current Account settings or to select which account to display.
<b>Codecs</b>	To display and configure available/unavailable codecs list. Codec means coder-decoder which is used to transfer analog signal to digital signal or vice versa. Familiar codecs are PCMU(G711U), PCMA(G711A), G722 (wide-bandth codecs), G729 and so on.
<b>Video Codec</b>	To configure the video quality <ul style="list-style-type: none"> <li>● Codec Name: The default video codec is H264.</li> <li>● Codec Resolution: It can support QCIF, CIF, VGA, 4CIF, 720P.</li> <li>● Codec Bitrate: The lowest bitrate is 128, the highest bitrate is 2048.</li> <li>● Codec payload: From 90-119.</li> </ul>
<b>Subscribe</b>	To display and configure MWI, BLF, ACD subscription settings. <ul style="list-style-type: none"> <li>● MWI: Message Waiting Indicator which is used to indicate whether there is unread new voice message.</li> <li>● BLF: BLF is short for Busy Lamp Field which is used to monitor the designated extension status.</li> <li>● 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.</li> </ul>
<b>DTMF</b>	To display and configure DTMF settings. <ul style="list-style-type: none"> <li>● Type: Support Inband, Info, RFC2833 or their combination.</li> <li>● How To Notify DTMF: Only available when DTMF Type is Info.</li> <li>● DTMF Payload: To configure payload type for DTMF.</li> </ul> <p><b>Note:</b> By default, DTMF type is RFC2833 which is the standard. Type Inband uses inband frequency to indicate DTMF tone which is most used to be compatible to traditional telephone server. Type Info use SIP Info message to indicate DTMF message.</p>
<b>Call</b>	To display and configure call-related features. <ul style="list-style-type: none"> <li>● Max Local SIP Port: To configure maximum local sip port for designated account.</li> <li>● Min Local SIP Port: To configure minimum local sip port for designated account.</li> <li>● Caller ID Header: To configure which Caller ID format to fetch for displaying on Phone UI.</li> <li>● Auto Answer: If enabled, IP phone will be auto-answered</li> </ul>

	<p>when there is an incoming call for designated account.</p> <ul style="list-style-type: none"> <li>● Ringtones: Choose the ringtone for each account.</li> <li>● Provisioning 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.</li> <li>● User=phone: If enabled, IP phone will send user=phone within SIP message.</li> <li>● PTime: Interval time between two consecutive RTP packets.</li> <li>● Anonymous Call: If enabled, all outgoing call for the designated account will be anonymous number.</li> <li>● Anonymous Call Rejection: If enabled, all incoming anonymous-out call for the designated account will be rejected.</li> <li>● Is escape non Ascii character: To transfer the symbol to Ascii character.</li> <li>● Missed Call Log: To display the miss call log.</li> <li>● Prevent SIP Hacking: Enable to prevent SIP from hacking.</li> </ul>
<b>Session Timer</b>	<p>To display or configure session timer settings.</p> <ul style="list-style-type: none"> <li>● Active: To enable or disable this feature, if it is enabled, the ongoing call will be disconnected automatically once the session expired unless it's been refreshed by UAC or UAS.</li> <li>● Session Expire: Configure session expire time.</li> <li>● Session Refresher: It is used to configure who should be responded for refreshing a session.</li> </ul> <p><b>Note:</b> UAC means User Agent Client, here stands for IP phone. UAS means User Agent Server, here stands for SIP server.</p>
<b>BLF List</b>	<p>To display or configure BLF List URI address.</p> <ul style="list-style-type: none"> <li>● BLF List URI: BLF List is short for Busy Lamp Field List.</li> <li>● BLFList PickUp Code: To set the BLF pick up code.</li> <li>● BLFList BargelIn Code : To set the BLF barge in code.</li> </ul>
<b>Encryption</b>	<p>To enable or disabled SRTP feature.</p> <ul style="list-style-type: none"> <li>● Voice Encryption(SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.</li> </ul>
<b>NAT</b>	<p>To display NAT-related settings.</p> <ul style="list-style-type: none"> <li>● UDP Keep Alive message: If enabled, IP phone will send UDP keep-alive message periodically to router to keep NAT port alive.</li> <li>● UDP Alive Msg Interval: Keepalive message interval.</li> <li>● Rport: Remote Port, if enabled, it will add Remote Port</li> </ul>

	into outgoing SIP message for designated account.
<b>User Agent</b>	One can customize User Agent field in the SIP message; if user agent is set to specific value, user can see the information from PCAP. If user agent is not set by default, users can see the company name, model number and firmware version from PCAP

## 4.6. Network

### 4.6.1 Basic

**Network-Basic**

**LAN Port**

DHCP  
 Static IP

IP Address: 192.168.1.100  
 Subnet Mask: 255.255.255.0  
 Default Gateway: 192.168.1.1  
 LAN DNS1: 8.8.8.8  
 LAN DNS2:

Sections	Description
<b>LAN Port</b>	<p>To display and configure LAN Port settings.</p> <ul style="list-style-type: none"> <li>● DHCP: If selected, IP phone will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically.</li> <li>● Static IP: If selected, you have to set IP address, Subnet Mask, Default Gateway and DNS server manually.</li> </ul>

## 4.6.2 Advanced

Network-Advanced	
<b>Local RTP</b>	
Starting RTP Port	11800 (1024~65535)
Max RTP Port	12000 (1024~65535)
<b>SNMP</b>	
Active	Disabled
Port	(1024~65535)
Trusted IP	
<b>VLAN</b>	
LAN Port	Active: Disabled
	VID: 1 (1~4094)
	Priority: 0
<b>TR069</b>	
	Active: Disabled
	Version: 1.0
ACS	URL: <input type="text"/>
	User Name: <input type="text"/>
	Password: <input type="password"/>
Periodic Inform	Active: Disabled
	Periodic Interval: 1800 (3~24×3600s)
CPE	URL: <input type="text"/>
	User Name: <input type="text"/>
	Password: <input type="password"/>

Sections	Description
<b>Local RTP</b>	<p>To display and configure Local RTP settings.</p> <ul style="list-style-type: none"> <li>● Max RTP Port: Determine the maximum port that RTP stream can use.</li> <li>● Starting RTP Port: Determine the minimum port that RTP stream can use.</li> </ul>
<b>SNMP</b>	<p>To display and configure SNMP settings.</p> <ul style="list-style-type: none"> <li>● Active: To enable or disable SNMP feature.</li> <li>● Port: To configure SNMP server's port.</li> <li>● Trusted IP: To configure allowed SNMP server address, it could be an IP address or any valid URL domain name.</li> </ul> <p><b>Note:</b> SNMP (Simple Network Management Protocols) is Internet-standard protocol for managing devices on IP networks.</p>
<b>VLAN</b>	<p>To display and configure VLAN settings.</p> <ul style="list-style-type: none"> <li>● Active: To enable or disable VLAN feature for designated port.</li> <li>● VID: To configure VLAN ID for designated port.</li> <li>● Priority: To select VLAN priority for designated port.</li> </ul> <p><b>Note:</b> Please consult your administrator for specific VLAN settings in your networking environment.</p>



<p><b>TR069</b></p>	<p>To display and configure TR069 settings.</p> <ul style="list-style-type: none"> <li>● Active: To enable or disable TR069 feature.</li> <li>● Version: To select supported TR069 version (version 1.0 or 1.1).</li> <li>● ACS/CPE: ACS is short for Auto configuration servers as server side, CPE is short for Customer-premise equipment as client side devices.</li> <li>● URL: To configure URL address for ACS or CPE.</li> <li>● User name: To configure username for ACS or CPE.</li> <li>● Password: To configure Password for ACS or CPE.</li> <li>● Periodic Inform: To enable periodically inform.</li> <li>● Periodic Interval: To configure interval for periodic inform.</li> </ul> <p><b>Note:</b> TR-069(Technical Report 069) is a technical specification entitled CPE WAN Management Protocol (CWMP).It defines an application layer protocol for remote management of end-user devices.</p>
---------------------	---

## 4.7. Phone

### 4.7.1 Time/Language

The screenshot shows a configuration page titled "Time/Lang" with a sub-section for "NTP". The settings are as follows:

NTP	
Time Zone	0 GMT
Primary Server	0.pool.ntp.org
Secondary Server	1.pool.ntp.org
Update Interval	3600 (>= 3600s)
System Time	03:40:44

Sections	Description
<p><b>NTP</b></p>	<p>To configure NTP server related settings.</p> <ul style="list-style-type: none"> <li>● Time Zone: To select local Time Zone for NTP server.</li> <li>● Primary Server: To configure primary NTP server address.</li> <li>● Secondary Server: To configure secondary NTP server address, it takes effect if primary NTP server is unreachable.</li> <li>● Update interval: To configure interval between two consecutive NTP requests.</li> </ul> <p><b>Note:</b> NTP, Network Time Protocol is used to automatically</p>

	synchronized local time with INTERNET time, since NTP server only response GMT time, so that you need to specify the Time Zone for IP phone to decide the local time.
--	---

### 4.7.2 Call Feature

**Phone-Call Feature**

---

**Mode Phone**

Mode  Phone  Custom

---

**DND**

Account: All Account ▼

DND: Disabled ▼

Return Code When DND: 486(Busy Here) ▼

DND On Code:

DND Off Code:

---

**Intercom**

Active: Enabled ▼

Intercom Mute: Disabled ▼

---

**Others**

Return Code When Refuse: 486(Busy Here) ▼

Auto Answer Delay:  (0~5s)

Auto Answer Mode: Video ▼

Multicast Codec: PCMU ▼

Direct IP: Enabled ▼

Sections	Description
<b>Mode</b>	Mode: Select the desired mode.
<b>DND</b>	<p>DND (Do Not Disturb) allows IP phones to ignore any incoming calls.</p> <ul style="list-style-type: none"> <li>● Return Code when DND: Determine what response code should be sent back to server when there is an incoming call if DND on.</li> <li>● DND On Code: The Code is used to turn on DND on server's side, if configured, IP phone will send a SIP message to server to turn on DND on server side if you press DND when DND is off.</li> <li>● DND Off Code: The Code is used to turn off DND on server's side, if configured, IP phone will send a SIP message to server to turn off DND on server side if you press DND when DND is on.</li> </ul>
<b>Intercom</b>	<p>Intercom allows users to establish a call directly with the callee.</p> <ul style="list-style-type: none"> <li>● Active: To enable or disable Intercom feature.</li> </ul>

	<ul style="list-style-type: none"> <li>● Intercom Mute: If enabled, once the call established, the callee will be muted.</li> </ul>
<b>Others</b>	<ul style="list-style-type: none"> <li>● Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected.</li> <li>● Auto Answer Delay: To configure delay time before an incoming call is automatically answered.</li> <li>● Auto Answer Mode: To set video or audio mode for auto answer by default.</li> <li>● Direct IP: Direct IP call without SIP proxy.</li> </ul>

### 4.7.3 Voice

**Voice**

---

**Mic Volume**

Mic Volume

(1~15)

---

**Speaker Volume**

Speaker Volume

(1~15)

---

**Open Door Warning**

Open Door Warning

---

**RingBack Upload**

File Format: wav, size: < 200KB, samplerate: 16000, Bits: 16

---

**Opendoor Tone Upload**

File Format: wav, size: < 200KB, samplerate: 16000, Bits: 16

Sections	Description
<b>Mic Volume</b>	To configure Microphone volume, from 1-15. 8 by default.
<b>Speaker Volume</b>	To configure Speaker Volume, from 1-15,8 by default.
<b>Open Door Warning</b>	When the door is opened, users will hear that opendoor prompt voice. If you disable it, you won't hear the announcement.
<b>RingBack Upload</b>	During the calling, user will hear the ringback tone before the other party answer. User can upload the suitable RingBack

	Tone by yourselves. Please note the file format and size.
<b>Opendoor Tone Upload</b>	Choose a suitable opendoor warning tone to upload. Please not the file format and size.

#### 4.7.4 Dial Plan

**Dial Plan**

**Rules Management**

Browse... Import Export

Index	Account	Prefix	Replace	<input type="checkbox"/>
1				<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"/>

Page: 1 ▾ Add Edit Delete Prev Next

**Rules Modify >>**

Account:  Auto ▾  
Prefix:   
Replace:

Submit Cancel

Sections	Description
<b>Rules Management</b>	For easy management, users can export and import the replace rule file directly. (The export file format is .tgz, users need to unzip it, then check the .XML file. The Import format is .XML)
<b>Rules</b>	Allow user to select Replace rule to display or edit.
<b>Rules Modify</b>	Allow user to modify selected rules information, for replace rule, you can modify related accounts, prefix and replace. Such as: Account:1 Prefix: 100 Replace: 110 Then users dial 100 with account1, the phone will call out 110 actually.

## 4.7.5 Multicast

**Multicast**

**Multicast Setting**

---

Paging Barge Disabled ▼

Paging Priority Active Enabled ▼

**Priority List**

---

IP Address	Listening Address	Label	Priority
1 IP Address	<input type="text"/>	<input type="text"/>	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

Sections	Description
<b>Multicast Setting</b>	To display and configure the Multicast setting. <ul style="list-style-type: none"> <li>● Paging Barge: Choose the multicast number, the range is 1-10.</li> <li>● Paging priority Active: Enable o disable the multicast.</li> </ul>
<b>Priority List</b>	To setup the multicast parameters. <ul style="list-style-type: none"> <li>● Listening Address: Enter the IP address you need to listen.</li> <li>● Label: Input the label for each listening address.</li> </ul>

## 4.7.6 Call log

Call Log							
Call History							All ▼
Index	Type	Date	Time	Local Identity	Name	Number	<input type="checkbox"/>
1							<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

Sections	Description
<b>Call History</b>	<p>To display call history records.</p> <p>Available call history types are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls.</p> <p>Users can check the call history in detail. Tick the number to delete or delete all logs. R27A supports 100 call logs.</p>

## 4.7.7 Door log

The screenshot shows a web interface titled "Door Log". It features a table with the following columns: Index, Name, Code, Date, and Time. The table contains 15 rows, each with an index number from 1 to 15. To the right of each row is a small square checkbox. Below the table is a footer area containing a dropdown menu set to "1", and buttons for "Prev", "Next", "Delete", and "DeleteAll".

Sections	Description
Door Log	To display unlock history. This interface can only show the RF card unlock history now. Users can check the unlock information in detail. Users can delete one or all logs. The maximum door log is 500.

## 4.8. Upgrade

### 4.8.1 Basic

The screenshot shows a web interface titled "Upgrade-Basic". It displays the following information and controls:

- Firmware Version: 27.0.2.170
- Hardware Version: 27.0.0.0.0.0.0.0
- Upgrade: A text input field with a "Browse..." button, and "Submit" and "Cancel" buttons below it.
- Reset To Factory Setting: A "Submit" button.
- Reboot: A "Submit" button.

Sections	Description
----------	-------------

<b>Upgrade</b>	To select upgrading zip file from local or a remote server automatically. <b>Note:</b> Please make sure it's right file format for right model.
<b>Firmware version</b>	To display firmware version, firmware version starts with MODEL name.
<b>Hardware Version</b>	To display Hardware version.
<b>Reset to Factory Setting</b>	To enable you to reset IP phone's setting to factory settings.
<b>Reboot</b>	To reboot IP phone remotely from Web UI.

## 4.8.2 Advanced

**Upgrade-Advanced**

**PNP Option**

---

PNP Config Enabled ▼

**Manual Autop**

---

URL

User Name

Password

Common AES Key

AES Key(MAC)

**Automatic Autop**

---

Mode Power On ▼

Schedule Sunday ▼

Hour(0~23)

Min(0~59)

Clear MD5

Export Autop Template

**RebootSchedule**

---

Mode Disabled ▼

Schedule Every Day ▼

Hour(0~23)

**System Log**

---

LogLevel 3 ▼

Export Log



Sections	Description
<b>PNP Option</b>	<p>To display and configure PNP setting for Auto Provisioning.</p> <ul style="list-style-type: none"> <li>● 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.</li> </ul> <p>By default, this SIP message is sent to multicast address 224.0.1.75(PNP server address by standard).</p>
<b>Manual Autop</b>	<p>To display and configure manual update server's settings.</p> <ul style="list-style-type: none"> <li>● URL: Auto provisioning server address.</li> <li>● User name: Configure if server needs a username to access, otherwise left blank.</li> <li>● Password: Configure if server needs a password to access, otherwise left blank.</li> <li>● Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file.</li> <li>● AES Key (MAC): Used for IP phone to decipher MAC-oriented auto provisioning configuration file(for example, file name can be 0c1105888888.cfg if IP phone's MAC address is 0c1105888888).</li> </ul> <p><b>Note:</b> AES is one of many encryption, it should be configure only configure file is ciphered with AES, otherwise left blank.</p>
<b>Automatic Autop</b>	<p>To display and configure Auto Provisioning mode settings.</p> <p>This Auto Provisioning mode is actually self-explanatory.</p> <p>For example, mode "Power on" means IP phone will go to do Provisioning every time when it powers on.</p>
<b>System Log</b>	<p>To display system log level and export system log file.</p> <ul style="list-style-type: none"> <li>● 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.</li> <li>● Export Log: Click to export temporary system log file to local PC.</li> </ul>

## 4.9. Security

### 4.9.1 Basic

The screenshot shows a web interface for modifying a password. It is titled 'Security-Basic' and 'Web Password Modify'. The form contains the following elements:

- User Name:** A dropdown menu with 'admin' selected.
- Current Password:** A text input field.
- New Password:** A text input field.
- Confirm Password:** A text input field.

Sections	Description
<b>Web Password Modify</b>	To modify user's password. <ul style="list-style-type: none"><li>● Current Password: The current password you used.</li><li>● New Password: Input new password you intend to use.</li><li>● Confirm Password: Repeat the new password.</li></ul>