Akuvox

R26X 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 R26X. Please verify the packaging content and network status before setting. This manual applies to firmware 26.0.2.31 or lower version.

Note: The old firmware may be a little different from 26.0.2.31 about some configuration. Please consult your administrator for more information.

Content

1. Product Overview

1.1. Product description

R26C

R26P

Akuvox R26X 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 unlock the door if you wish. User can also use RF card to unlock the door(R26C only). It’s applicable in villas, office and so on.

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.

Specific Absorption Rate (SAR) information

SAR tests are conducted using standard operating positions accepted by the FCC with the device transmitting at its highest certified power level in all tested frequency bands, although the SAR is determined at the highest certified power level, the actual SAR level of the device while operating can be well below the maximum value. Before a new product is a available for sale to the public, it must be tested and certified to the FCC that it does not exceed the exposure limit established by the FCC, tests for each phone are performed in positions and locations as required by the FCC. For headset, this part has been tested and meets the FCC RF exposure guidelines when used with an accessory designated for this product or when used with an accessory that contains no metal.

For baseband, this equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment .This equipment should be installed and operated with minimum distance 20cm between the radiator& your body.

1.2. Features

- **Highlight**

- Vandal resistant body, with a flush button
- Wild-angle camera : 90°
- 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

- **Physical&Power**

- Body material: all-aluminum
- Camera: 3M pixels, automatic lighting
- Button: 1 call button
- Infrared Sensor
- RF Card Reader:13.56MHz Supported（R26C only）
- Output Relay: 2 output relays for door opener
- 802.3af Power-Over-Ethernet
- 12V DC connector(if not using POE)
- Power consumption: less than 12w
- Water proof&Dust proof: IP65
- Installation: Wall-mounted
- Dimension: 190x110x35mm

- **SIP Endpoint**

- SIP v1(RFC2543), SIP v2(RFC3261)
- Audio codecs: G.711a, G.711μ, G.722, G.729
- Video codecs: H263,H264

- Speech Quality: 7kHz Audio
- Echo Cancellation
- Voice Activation Detection
- Comfort Noise Generator

Video

- Resolution: up to 1080p
- Maximum image transfer rate:1080p-30pfs
- High intensity white LEDs for picture lighting during dark hours with internal light sensor
- Compatible with 3rd.Party.Video components,e.g.NVRs.

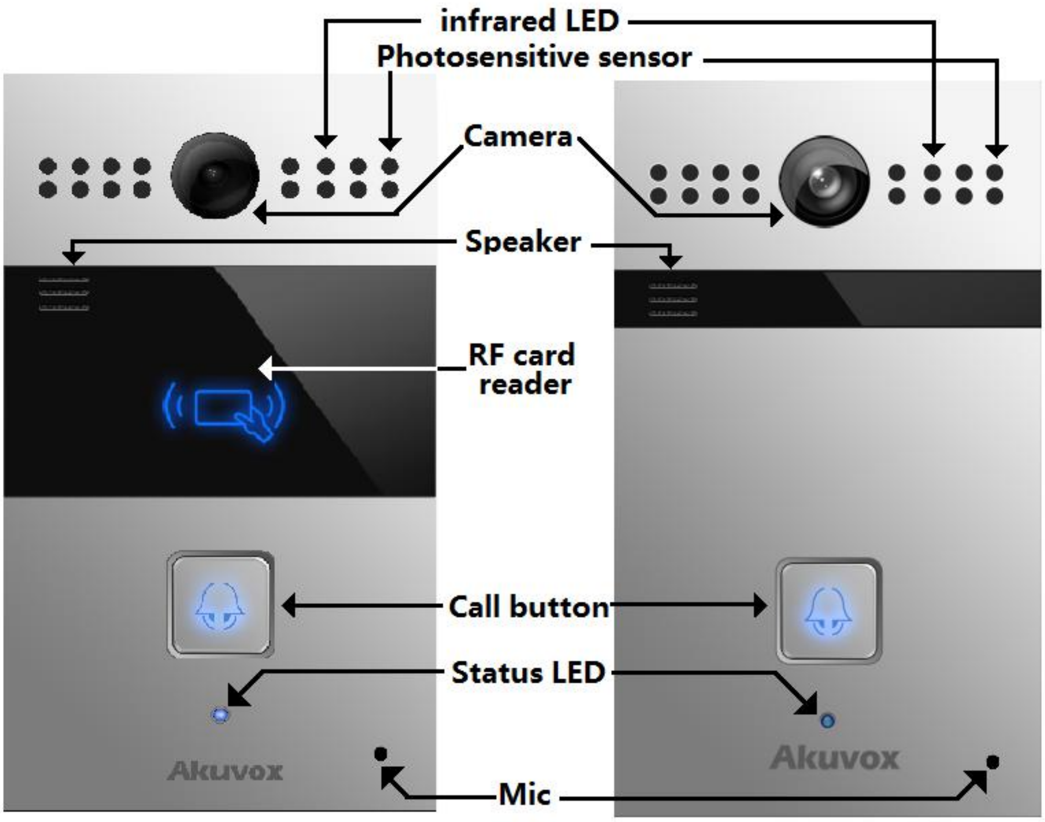
Door Entry Feature

- 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. Panel Description



R26C R26P

1.4. Unpacking

Name	Quantity
R26C/P	1
Back cover	1
Wall bracket	1
Cable buckle	1
M4X20 screw	2
ST4x20 screw	4
Screw fixing seat	4
M3X5 screw	4
M3X10 screw	1
Screwdriver	1

2. Basic Using

2.1. Make a Call

Press the call button to call out the predefined number or IP address. During the talking, the LED will show green. User can refer to chapter 3.2.1 about one button setting.

2.2. Receive a Call

R26X will auto answer the incoming call by default. If user disable Auto Answer function, when you see the flash green LED, Please press the call button to answer the call.

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”. R26X support 13.56MHz RF card. Please to chapter 3.2.8 about Card Setting.

3. Configuration

3.1. Web login

3.1.1. Obtaining IP address

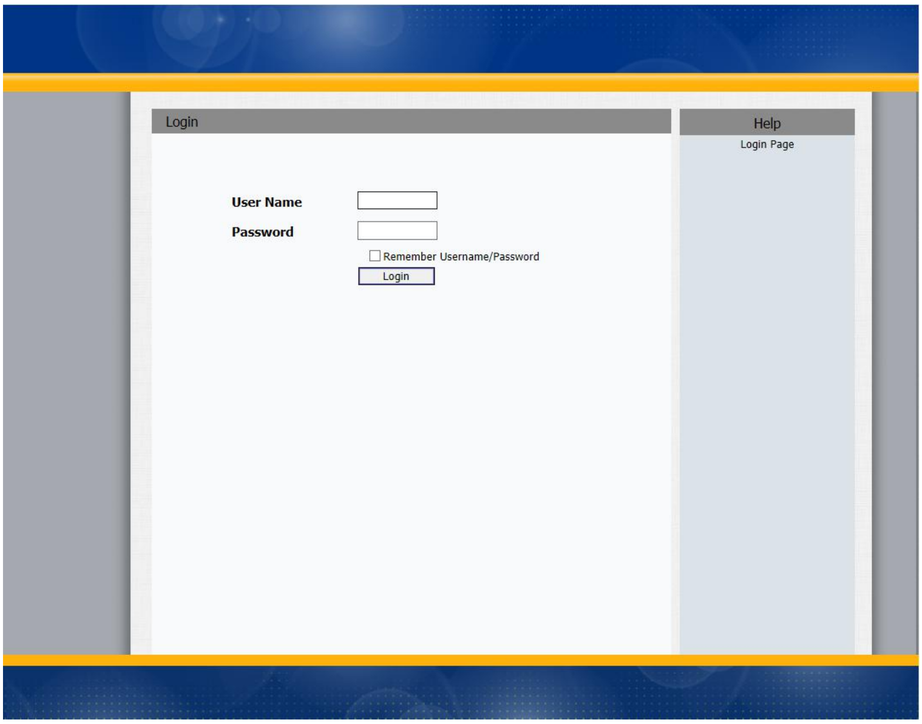
The Akuvox R26X uses DHCP IP address by default. If the IP address is unknown, press and hold the call button for a short period of time(about 5s) after LED light turns blue, the phone will announce its IP.

3.1.2. Login the web

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

User name: admin

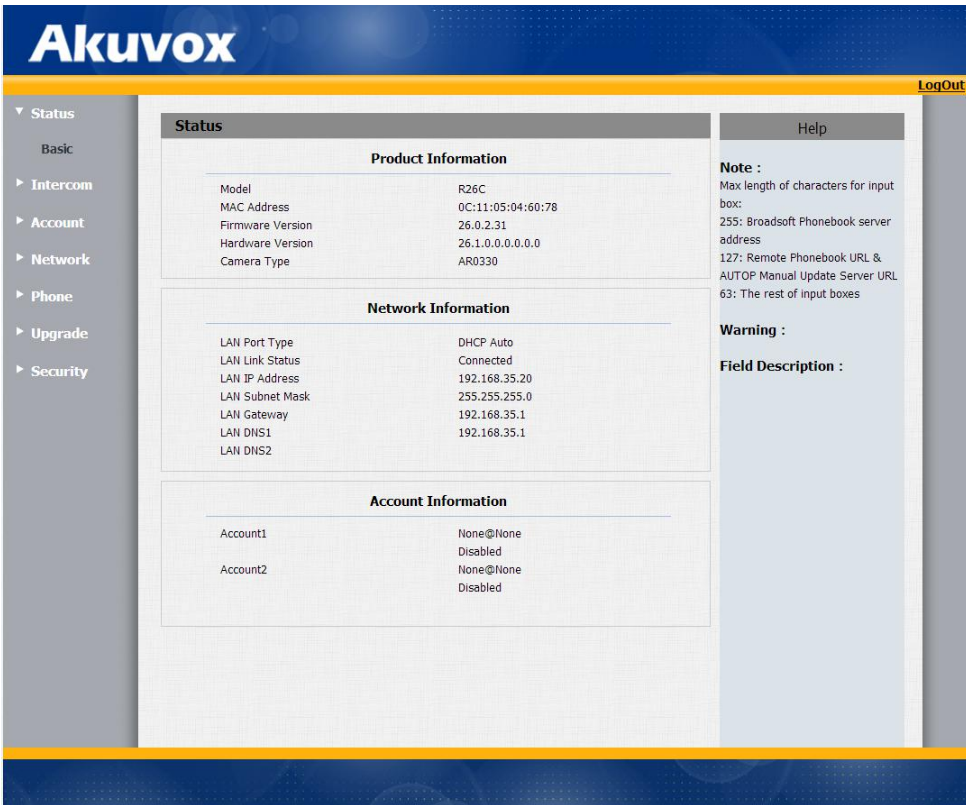
Password: admin



3.2. Status

3.2.1. Basic

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

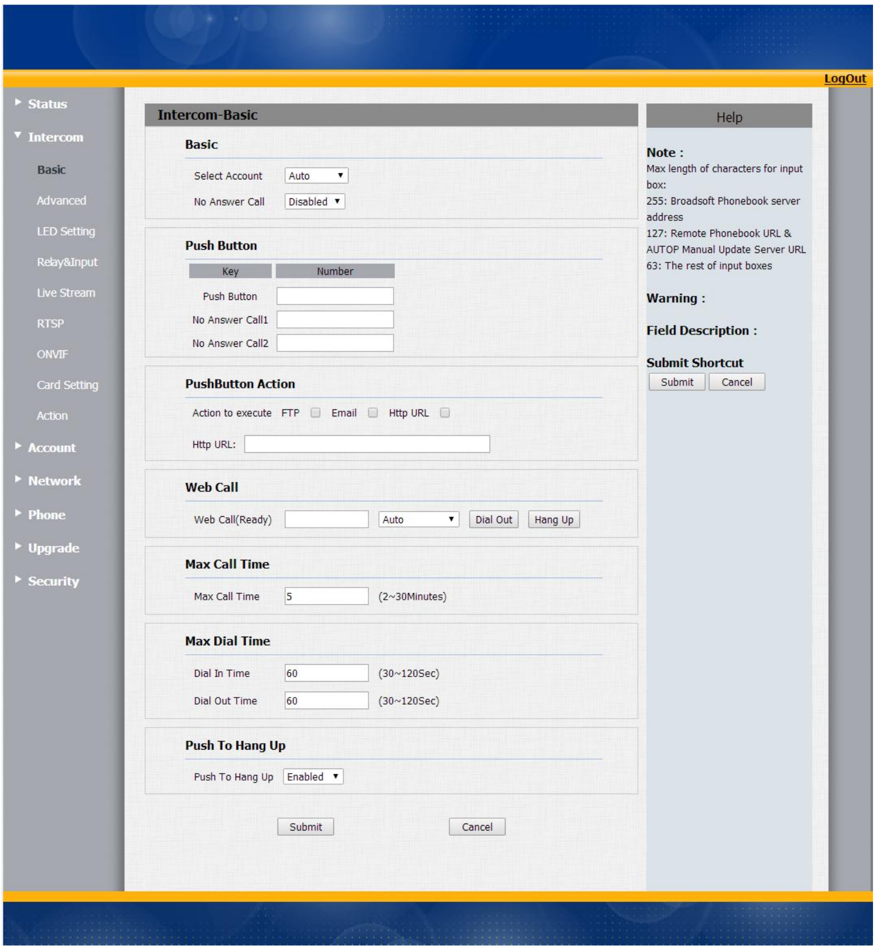


Sections	Description
Product Information	To display the device's information such as Model name, MAC address (IP device's physical address), Firmware version and Hardware version.
Network Information	To display the device's Networking status(LAN Port),such as Port Type(which could be DHCP/Static/PPPoE), 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).
Account Information	To display device's Account information and Registration status (account username, registered server's address, Register result).

3.3. Intercom

3.3.1. Basic

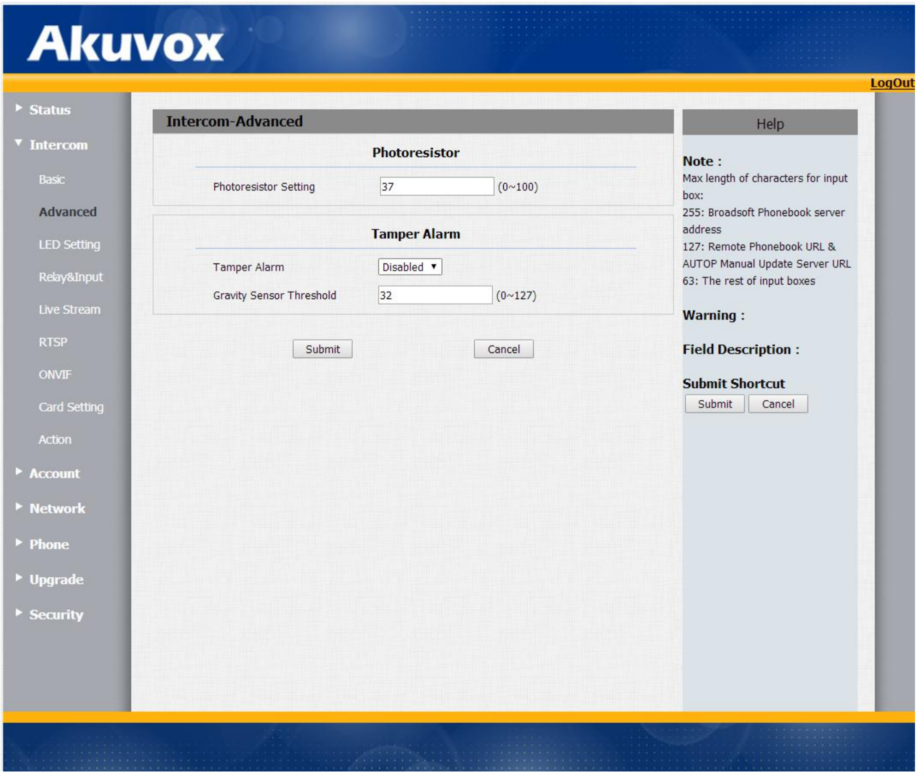
Go to the path: Intercom-Basic



Sections	Description
Basic	● Select Account: R26X supports 2 accounts. You can choose one account or Auto mode for the following Intercom basic settings. ● No Answer Call： R26X will call to the No answer call number in order when the ringtone is time out without answer of the push button number. Disable by default.
Push Button	● Push Button: To configure the destination number or IP you want to contact with. ● No Answer Call 1&2: To setup two no answer call numbers or one no answer call number.
Push Button Action	Enable this function, the device will record any changes of the surrounding environment then send the message or

	picture to the corresponding receiver. ● Action to execute : Tick the suit the suitable way to receive the action message. ● HTTP URL: If you tick Http URL, then enter the Http server IP address in the HTTP URL area. When the device detects any changes, it will send Http network package.
Web Call	To dial out or answer the phone from website.
Max Call Time	To configure the max call time
Max Dial Time	● Dial in Time: When other phone calls to R26X, if ring tone is over the Dial in Time without answer. The call will be hang up. ● Dial out Time: When R26 calls to the other party, if the ringtone is over the Dial out Time without answer. R26 will continue calls to no answer call number in order.
Push to Hang up	To enable or disable the Push to Hang up function

3.3.2. Advanced

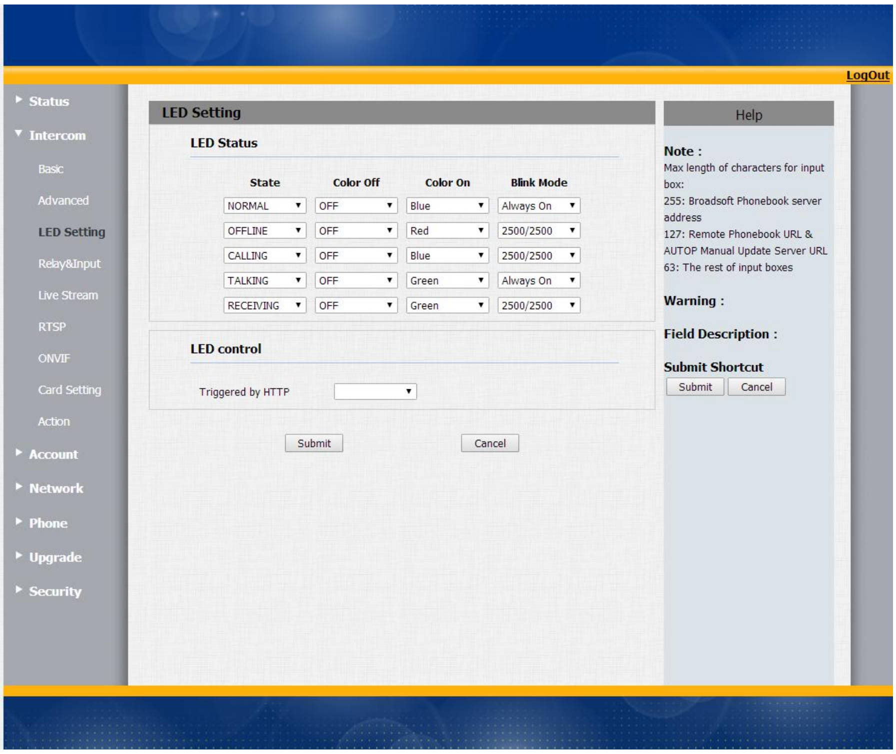


Sections	Description
Photoresistor	Photoresistor is used to sense the light intensity that R26X

	will auto enable infrared LED. Users can adjust the photoresistor value manually. The value is smaller, the infrared LED is more sensitive.
Tamper Alarm	Enable the Tamper Alarm, if the gravity of R26 changes, the phone will alarm. The Threshold value is smaller, the faster the reaction of device.

3.3.3. LED Setting

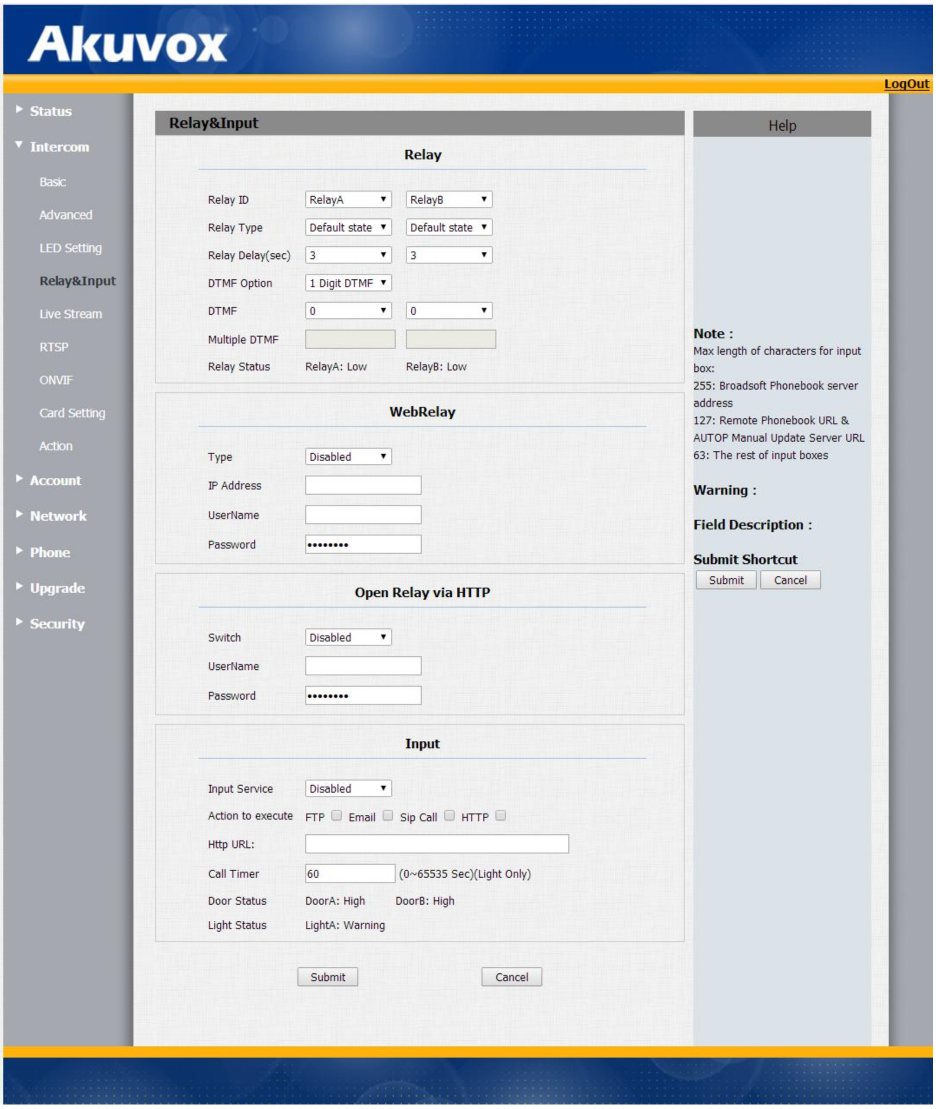
To setup the LED lighting mode.



Sections	Description
State	There is five states: Normal, Offline, Calling, Talking and Receiving.
Color Off	The default status is OFF.
Color On	It can support three color: Red, Green, Blue.
Blink Mode	To setup the different blink frequency.
LED control	Use Http URL to remote control the LED status. **Http format:** **http://PhoneIP/fcgi/do?action=LedAction&State=1&Color=1&Mode=2500 Status: 1=Idle ;2=OffLine ; 3=Calling ; 4=Talking ; 5=Receivin ; Color: 1=Green ; 2=Bule ; 3=Red; Mode:0=Always**

	on;1=Always Off 500/1000/1500/2000/25000/3000

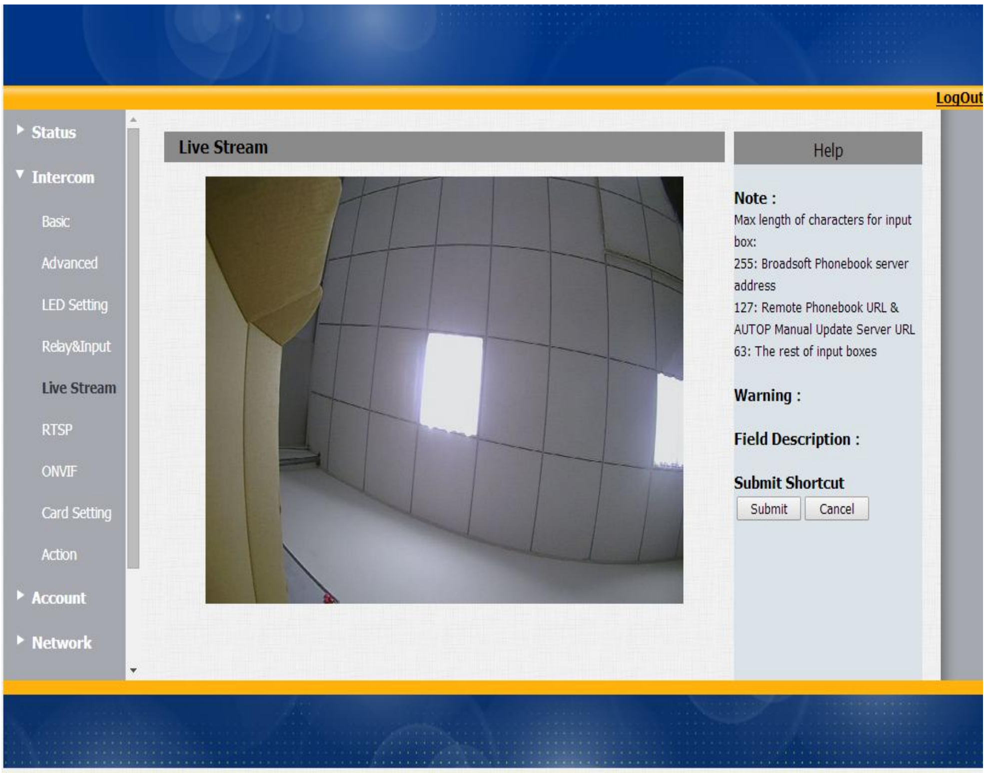
3.3.4. Relay&Input



Sections	Description
Relay	To configure some settings about unlock ● Relay Select: R26X support 2 relays. ● 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 state.

	• Relay Delay(sec): Allows door remain “open” for certain period The range is from 1 to 10 seconds • DTMF Option: R26X support 1/2/3/4 digits DTMF unlock code. Please select one type and enter the corresponding code. • DTMF: Setup 1 digit DTMF code for remote unlock • Multiple DTMF : Setup 2/3/4 digits DTMF code for remote unlock. • Status: The status will be changed by the relay state.
Web relay	R26X 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. • 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.
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. Null by default URL format:**http://192.168.1.102/fcgi/do?action=OpenDoor&UserName=&Password=&DoorNum=1**
Input	This function is used to trigger the action by door 1,2 contacts. Once the door contacts status is changing , it will execute the corresponding action. • Service: Enable by default. • Action to execute: Choose one or more ways to receive the alarm message. • Http URL: If you tick Http URL ,then enter the Http server IP address in the HTTP URL area. When the alarm is triggered, it will send Http message. • Call Timer: The interval for calling. For instant , the Call timer is 5sec, if you hang up the calling in the third second, the calling will auto call out after 2sec. • Door Status: Show the status when the door is opened form inside. • Light Status: The status will change according to the sensor. Once the sensor is triggered , the status will show Warning. Normal by default.

3.3.5. Live Stream



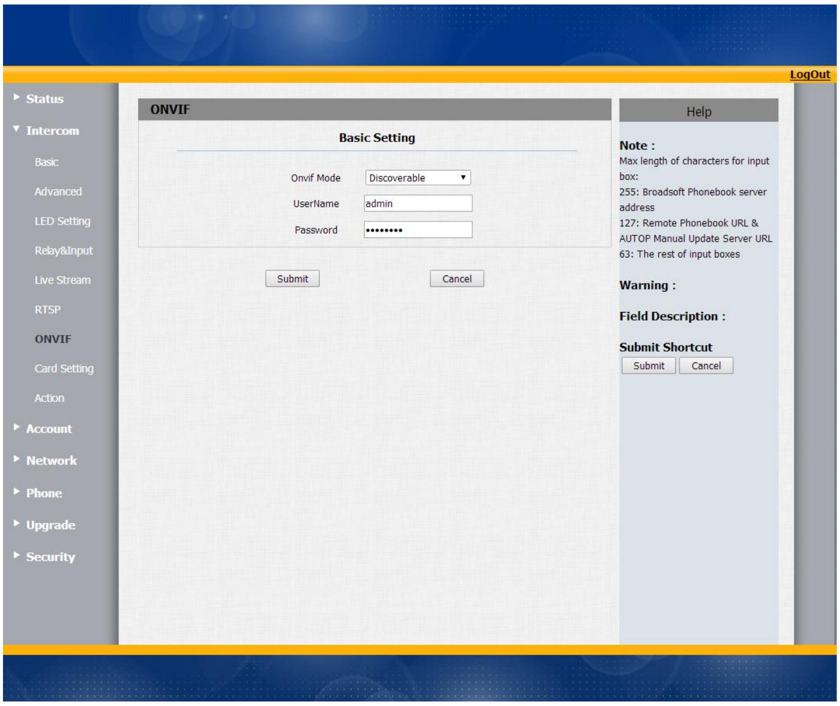
Sections	Description
Live Stream	To check the real-time video from R26X.

3.3.6. RTSP

Sections	Description
RTSP Basic	To active the RTSP function, then R26X can be monitored.
RTSP Stream	To enabled RTSP video and select the video codec. R26X support H264,H263 video codec. H264 by default.
H.264 Video Parameters	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
MPEG4 Video Parameters	MPEG4: it is one of the network video image Compression standard. It supports the maximum Compression ratio 4000:1. It is an important and commom 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: called Motion Joint Photographic Experts Group. It is a video encoding format.in which each image is compressed

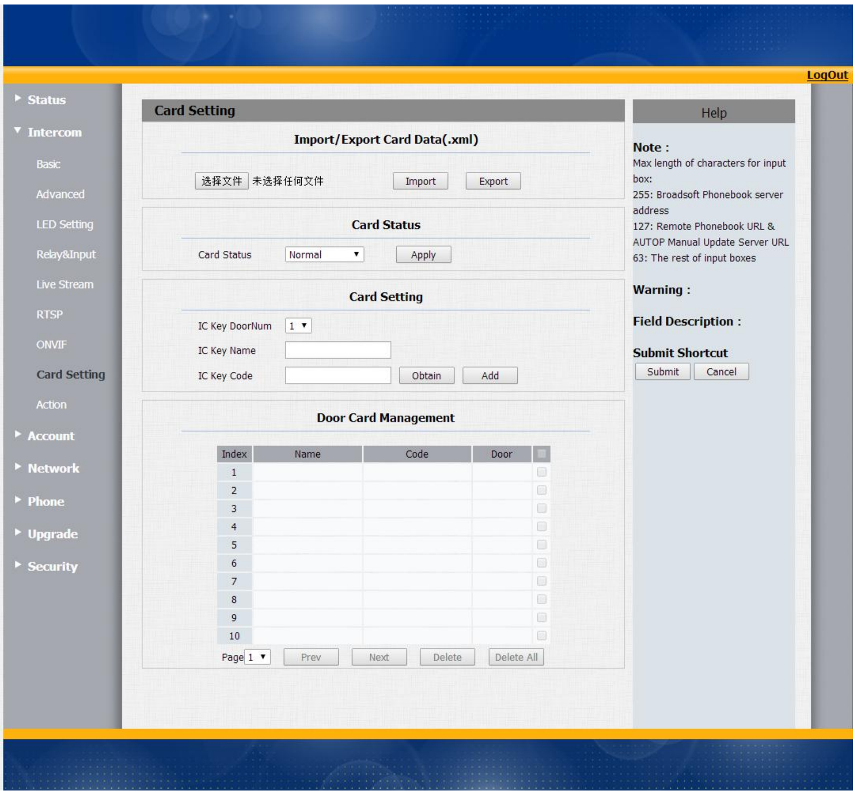
	separately by JPEG.MJPEG compression can produce high quality video image and has a flexible configuration in video definition and Compressed frames To modify the resolution,framerate and bitrate of MJPEG

3.3.7. ONVIF



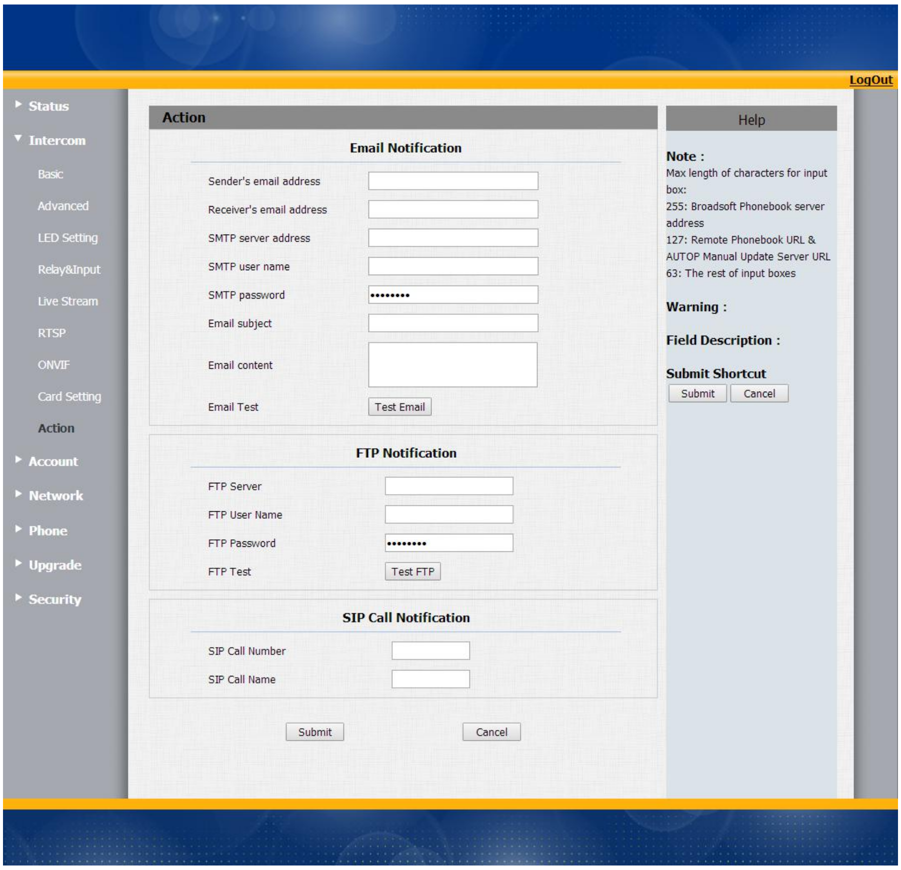
Sections	Description
Basic Setting	To setup the ONVIF function parameters. It is used to connect with the corresponding ONVIF tool. ● ONVIF Mode: To modify the state of ONVIF. R26X can only be searched in Discoverable mode which is set by default. ● User Name: To modify the user name you need. Admin by default. ● Password: To modify the password you want. Admin by default. **Note:** User name and password is used for authentication.

3.3.8. Card Setting(R26C only)



Sections	Description
Import/Export Card Data	To Import or Export the card data file. Only support .xml format.
Card Status	● Normal: Choose Normal mode when reading card. ● Card Issuing: Choose Card Issuing mode when writing card.
Card Setting	● IC Key DoorNum: R26X can support to connect 2 doors. Choose one and add the valid card for unlock. ● IC Key Name: To setup corresponding name for the card. ● IC Key Code: Place the card in the R26C RF Card Read area, then click Obtain button. After R26C reads the card code, click Add, the card information will show in the Door Card Management list.
Door Card Management	Valid card information will show in the list. Users can tick the current card information then delete one or all in the list.

3.3.9. Action

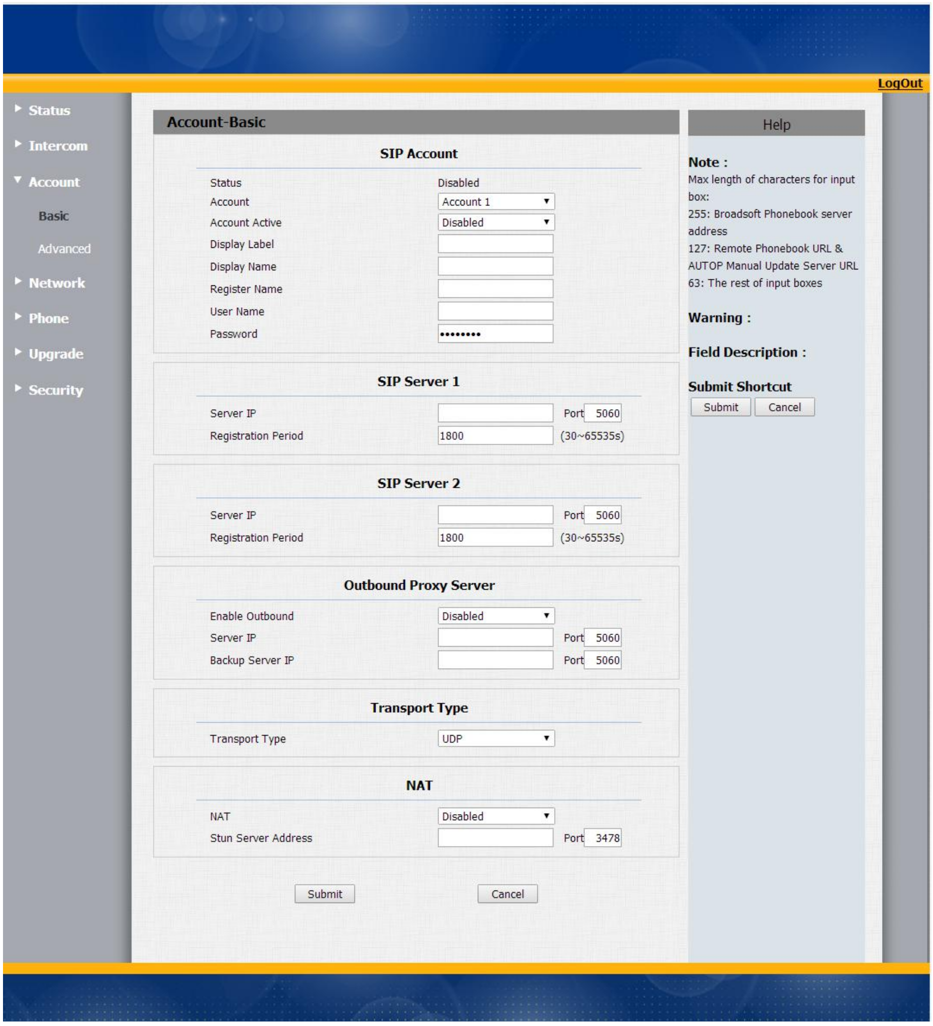


Sections	Description
Email Notification	● Sender Email Address: Input the sender email address. ● Receiver Email Address: Input the receiver email address. ● SMTP Server Address: Enter the SMTP server format. ● SMTP User name: Enter the SMTP. ● SMTP password: Enter the sender email password. ● Email Subject: Enter the subject name. ● Email content: Enter the content name. ● Email test: Click test to make sure the parameters you enter is right.
FTP Notification	● FTP Server: Enter the FTP server address. ● FTP User Name: Enter the FTP server user name. ● FTP Password: Enter the corresponding FTP server password. ● FTP test: Click test to make sure the parameters you enter is right.

SIP Call Notification	When you enable SIP Call function of alarm. Enter the number and name in the corresponding area. When the alarm is triggered, the device will call out the number automatically.

3.4. Account

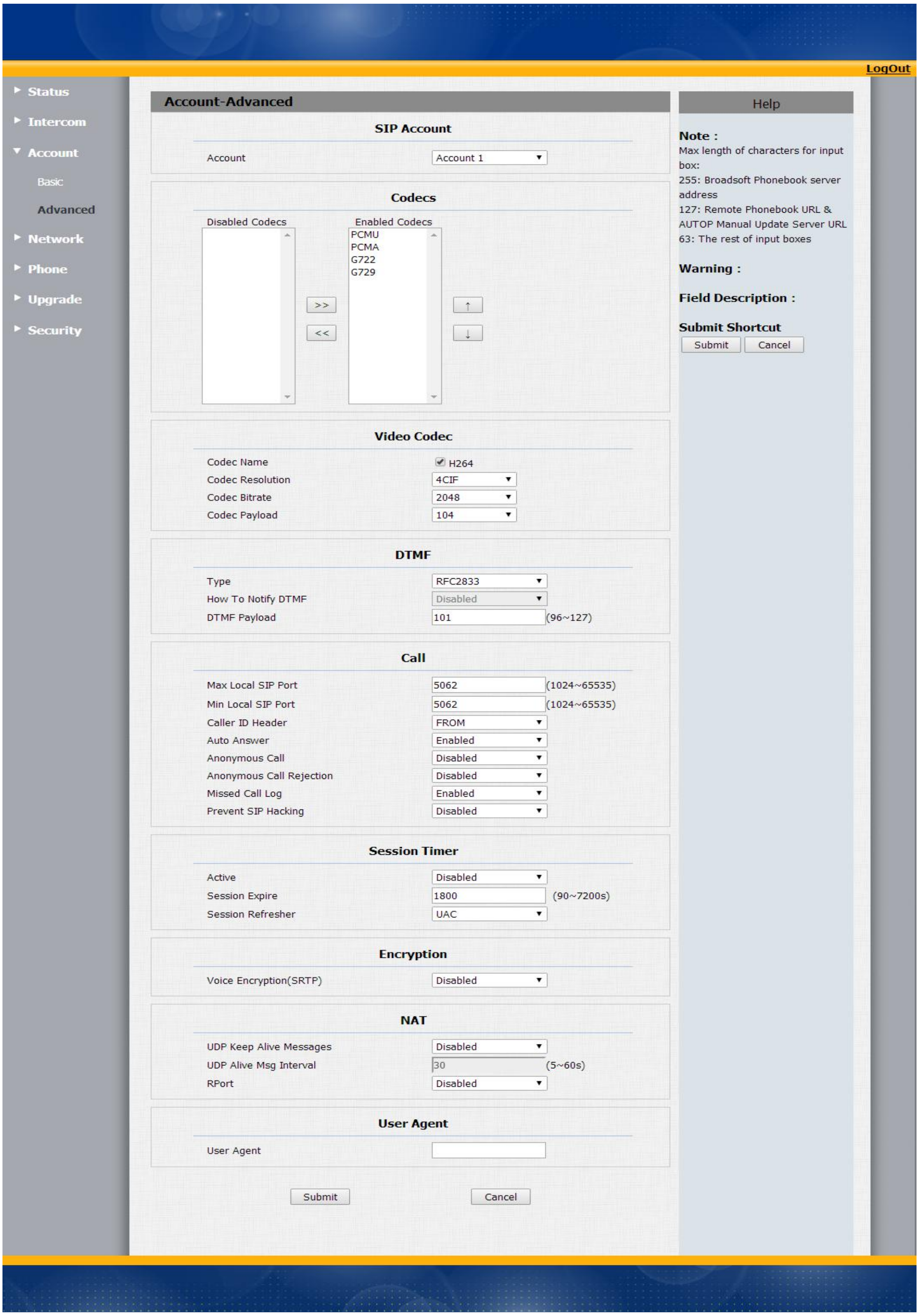
3.4.1. Basic



Sections	Description
SIP Account	To display and configure the specific Account settings. ● Status: To display register result. ● Account: Choose which account you need to register. ● Account Active: Enable it to active the account. ● Display Label: Which is displayed on the phone's LCD screen.

	● Display Name: Which is sent to the other call party for display. ● Register Name: Allocated by SIP server provider, used for authentication. ● User Name: Allocated by your SIP server provide, used for authentication. ● Password: Used for authorization.
SIP Server 1	To display and configure Primary SIP server settings. ● Server IP: SIP server address, it could be an URL or IP address. ● Registration Period: The registration will expire after Registration period, the IP phone will re-register automatically within registration period.
SIP Server 2	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. **Note**: Secondary SIP server is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.
Outbound Proxy Server	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. **Note**: If configured, all SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.
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.
NAT	To display and configure NAT(Net Address Translator) settings. ● STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues. **Note**: By default, NAT is disabled.

3.4.2. Advanced



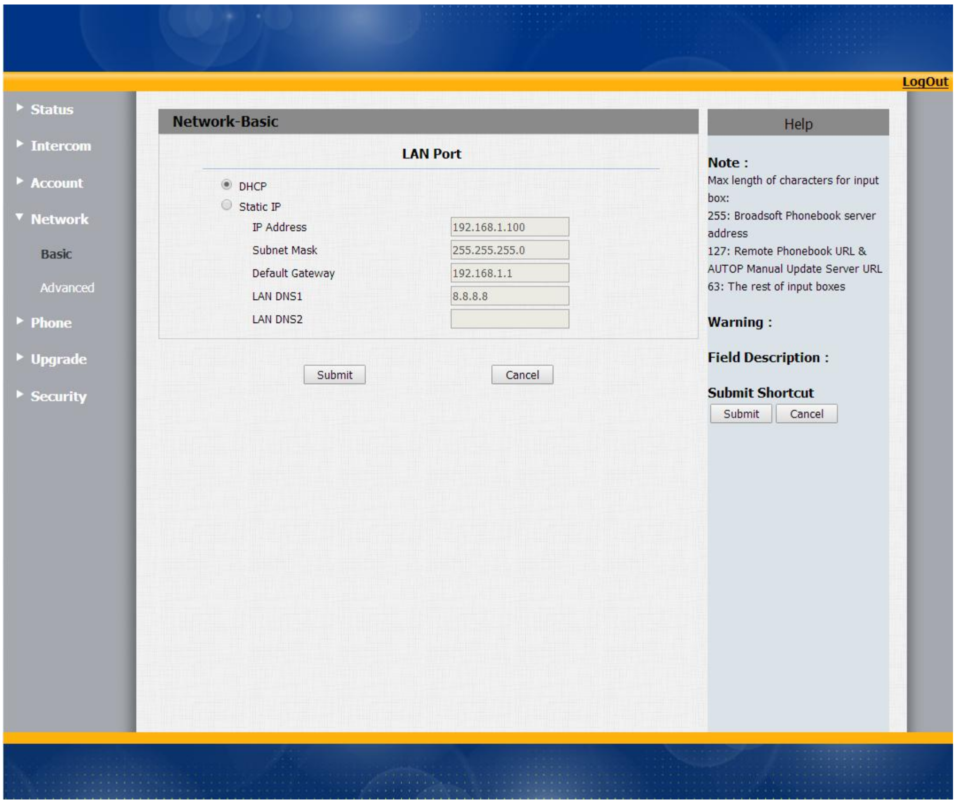
Sections	Description
SIP Account	To display current Account settings or to select which account to display.
Codecs	To display and configure available/unavailable codecs list. Codec means coder-decoder which is used to transfer analog signal to digital signal or vice versa. Familiar codecs are PCMU(G711U), PCMA(G711A), G722 (wide-bandth codecs), G729 and so on.
Video Codec	To configure the video quality. ● Codec Name: The default video codec is H264. ● Codec Resolution: It can support QCIF, CIF, VGA, 4CIF, 720P. ● Codec Bitrate: The lowest bitrate is 128, the highest bitrate is 2048. ● Codec payload: From 90-119.
Subscribe	To display and configure MWI, BLF, ACD subscription settings. ● MWI: Message Waiting Indicator which is used to indicate whether there is unread new voice message. ● BLF: BLF is short for Busy Lamp Field which is used to monitor the designated extension status. ● ACD: Automatic Call Distribution is often used in offices for customer service, such as call center. The setting here is to negotiate with the server about expire time of ACD subscription.
DTMF	To display and configure DTMF settings. ● 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. **Note**: 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.
Call	To display and configure call-related features. ● Max Local SIP Port: To configure maximum local sip port for designated account. ● Min Local SIP Port: To configure minimum local sip port for designated account. ● Caller ID Header: To configure which Caller ID format to fetch for displaying on Phone UI. ● Auto Answer: If enabled, IP phone will be

	auto-answered when there is an incoming call for designated account. ● Ringtones: Choose the ringtone for each account. ● 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. ● User=phone: If enabled, IP phone will send user=phone within SIP message. ● PTime: Interval time between two consecutive RTP packets. ● Anonymous Call: If enabled, all outgoing call for the designated account will be anonymous number. ● Anonymous Call Rejection: If enabled, all incoming anonymous-out call for the designated account will be rejected. ● Missed Call Log: To display the miss call log. ● Prevent SIP Hacking: Enable to prevent SIP from hacking.
Session Timer	To display or configure session timer settings. ● Active: To enable or disable this feature, If enable, the on going call will be disconnected automatically once the session expired unless it's been refreshed by UAC or UAS. ● Session Expire: Configure session expire time. ● Session Refresher: To configure who should be response for refreshing a session. **Note**: UAC means User Agent Client, here stands for IP phone. UAS means User Agent Server, here stands for SIP server.
Encryption	To enable or disabled SRTP feature. ● Voice Encryption(SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.
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.
User Agent	One can customize User Agent field in the SIP message; If user agent is set to specific value, user could see the information from PCAP. If user agent is not set by default, user could see the company name, model number and

	firmware version from PCAP

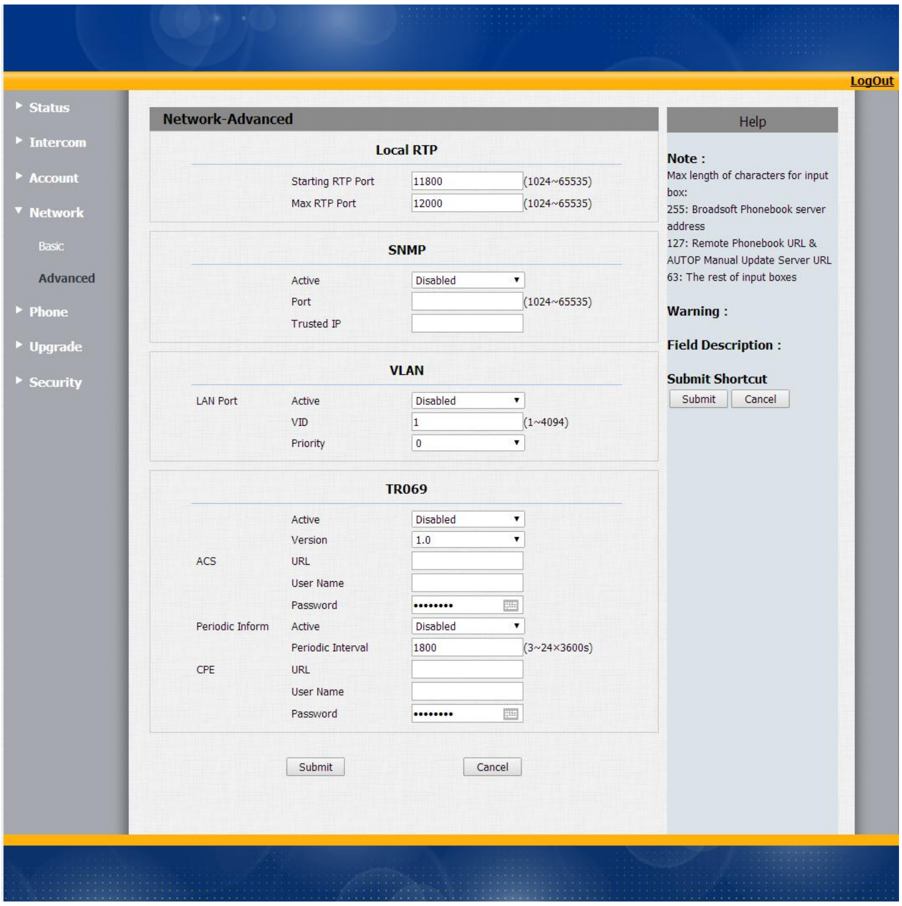
3.5. Network

3.5.1. Basic



Sections	Description
LAN Port	To display and configure LAN Port settings. ● DHCP: If selected, IP phone will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically. ● Static IP: If selected, you have to set IP address, Subnet Mask, Default Gateway and DNS server manually.

3.5.2. Advanced

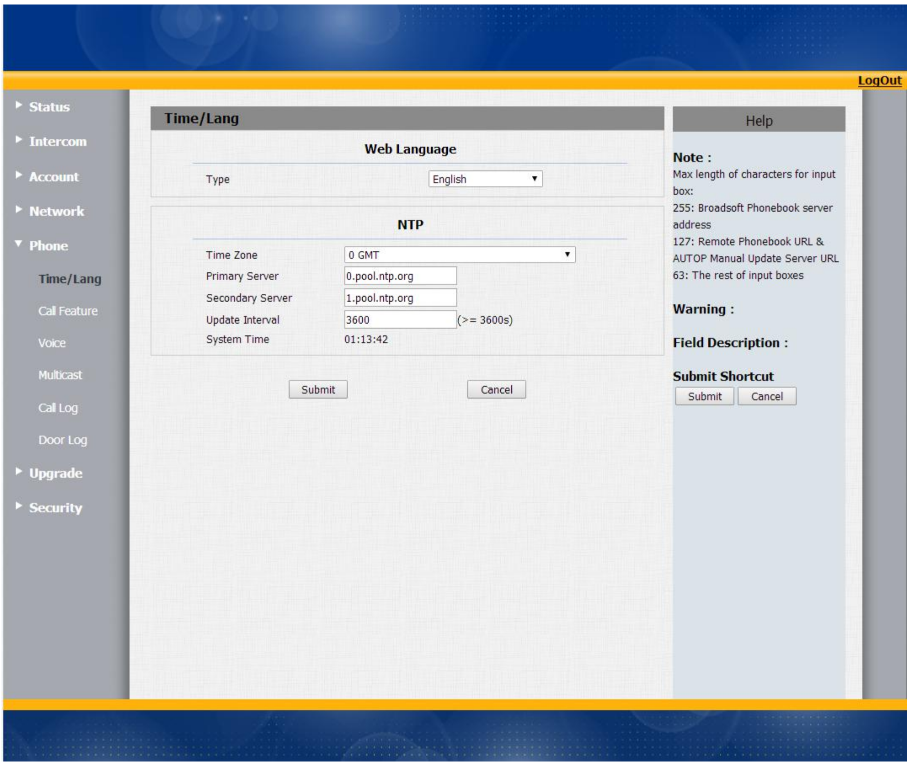


Sections	Description
Local RTP	To display and configure Local RTP settings. ● Max RTP Port: Determine the maximum port that RTP stream can use. ● Starting RTP Port: Determine the minimum port that RTP stream can use.
SNMP	To display and configure SNMP settings. ● Active: To enable or disable SNMP feature. ● Port: To configure SNMP server's port. ● Trusted IP: To configure allowed SNMP server address, it could be an IP address or any valid URL domain name. **Note**: SNMP (Simple Network Management Protocols) is Internet-standard protocol for managing devices on IP networks.
VLAN	To display and configure VLAN settings. ● Active: To enable or disable VLAN feature for designated port. ● VID: To configure VLAN ID for designated port.

	● Priority: To select VLAN priority for designated port. **Note**: Please consult your administrator for specific VLAN settings in your networking environment.
TR069	To display and configure TR069 settings. ● Active: To enable or disable TR069 feature. ● Version: To select supported TR069 version (version 1.0 or 1.1). ● ACS/CPE: ACS is short for Auto configuration servers as server side, CPE is short for Customer-premise equipment as client side devices. ● URL: To configure URL address for ACS or CPE. ● User name: To configure username for ACS or CPE. ● Password: To configure Password for ACS or CPE. ● Periodic Inform: To enable periodically inform. ● Periodic Interval: To configure interval for periodic inform. **Note**: 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.

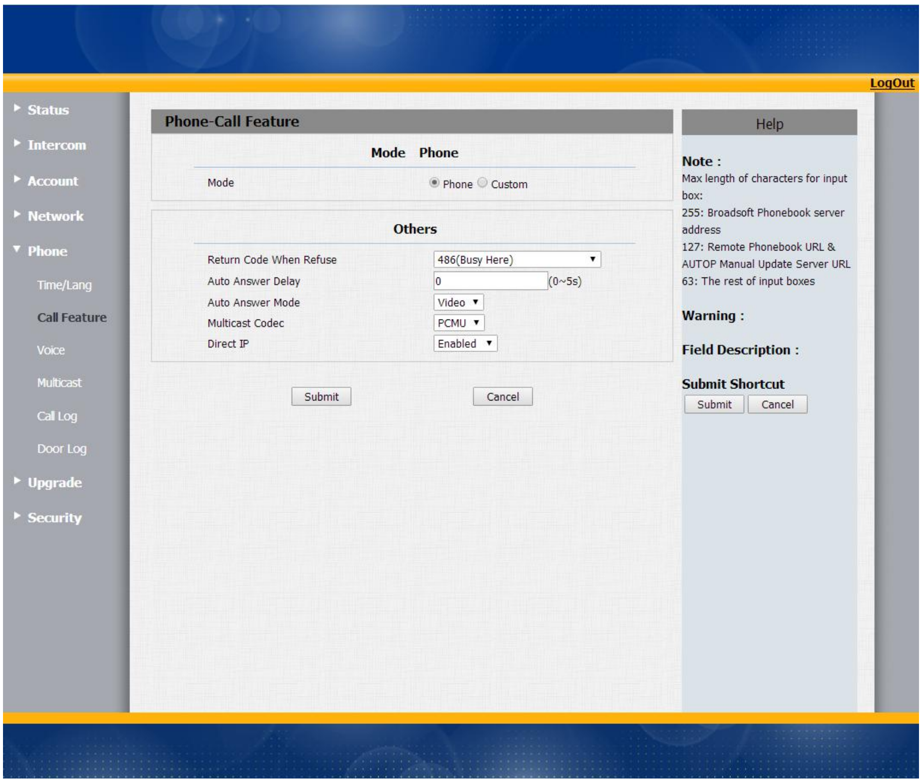
3.6. Phone

3.6.1. Time/Language



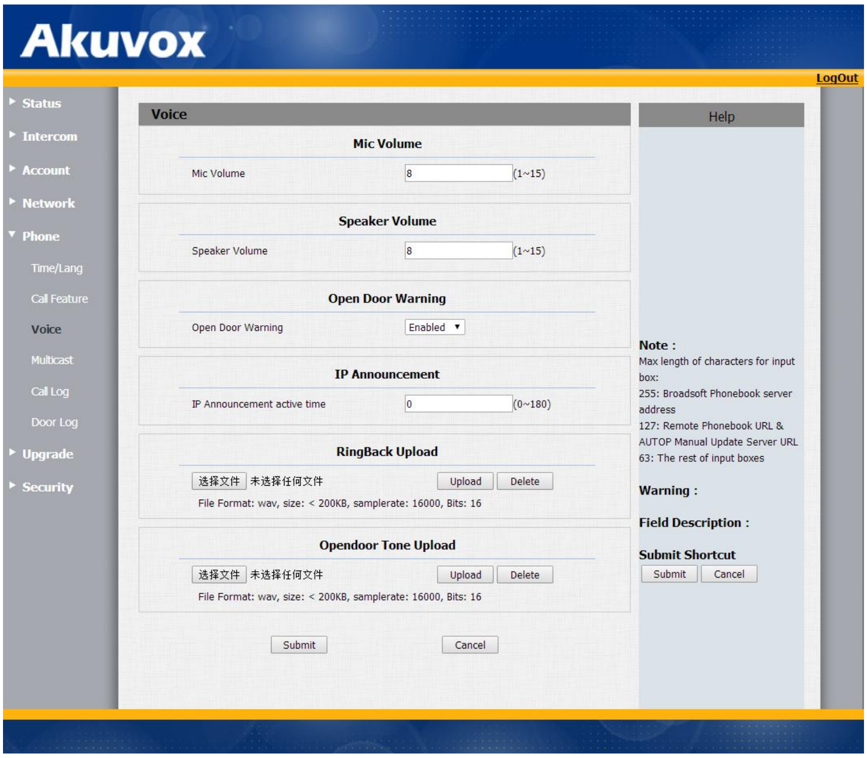
Sections	Description
Web Language	Choose the suitable web language you need. English by default.
NTP	To configure NTP server related settings. ● Time Zone: To select local Time Zone for NTP server. ● Primary Server: To configure primary NTP server address. ● Secondary Server: To configure secondary NTP server address, it takes effect if primary NTP server is unreachable. ● Update interval: To configure interval between two consecutive NTP requests. **Note**: NTP, Network Time Protocol is used to automatically 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.

3.6.2. Call Feature



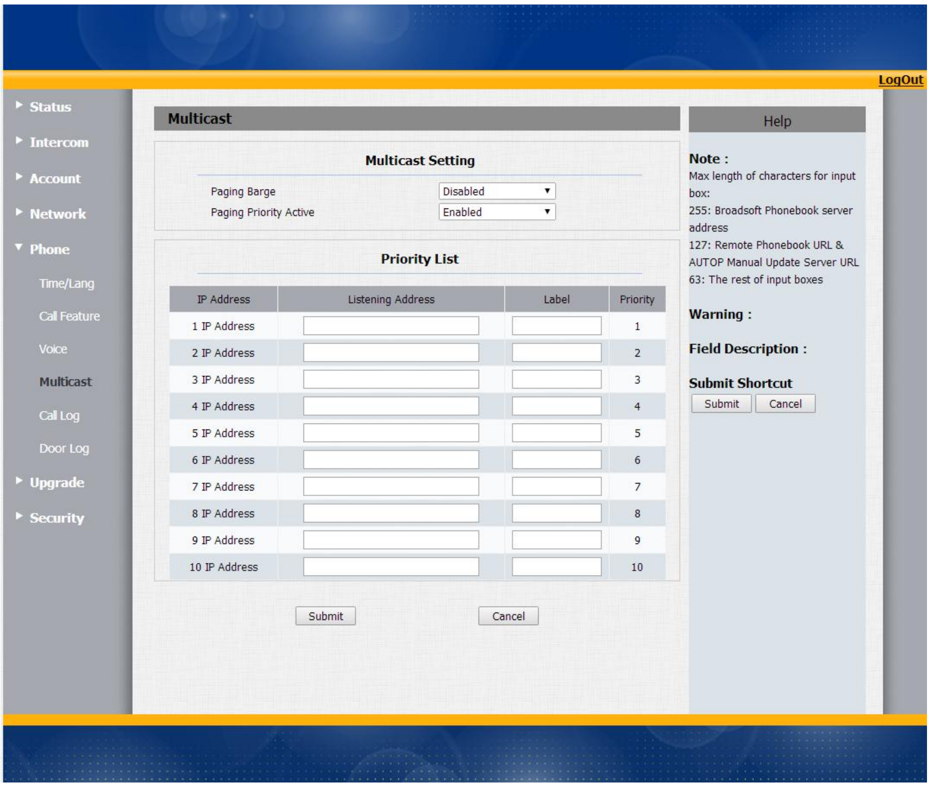
Sections	Description
Mode	● Mode: Select the desired mode.
Others	● Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected. ● Auto Answer Delay: To configure delay time before an incoming call is automatically answered. ● Auto Answer Mode: To set video or audio mode for auto answer by default. ● Direct IP: Direct IP call without SIP proxy.

3.6.3. Voice



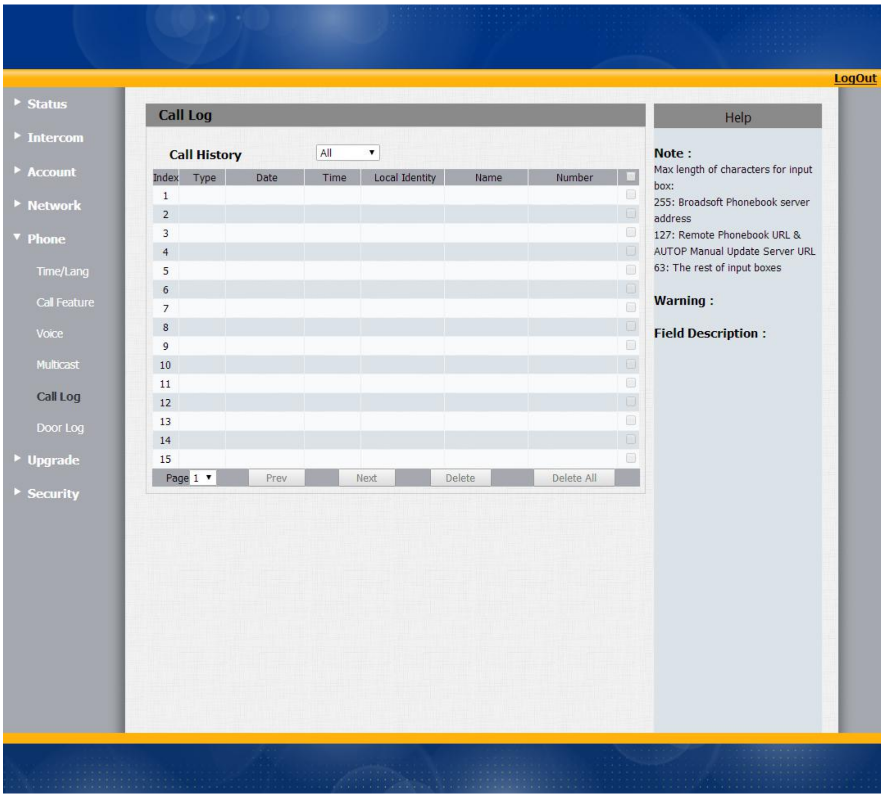
Sections	Description
Mic Volume	To configure Microphone volume , from 1-15. 8 by default.
Speaker Volume	To configure Speaker Volume,from 1-15,8 by default.
Open Door Warning	To configure door opening voice. Disable it, you won't hear the prompt voice when the door is opened.
IP Announcement	To setup the IP Announcement active time. Over the configured value, the phone will not announce the IP when you hold the button.
RingBack Upload	Users can upload the ring back tone if you need.
Opendoor Tone Upload	User can upload the Opendoor tone by yourself.

3.6.4. Multicast



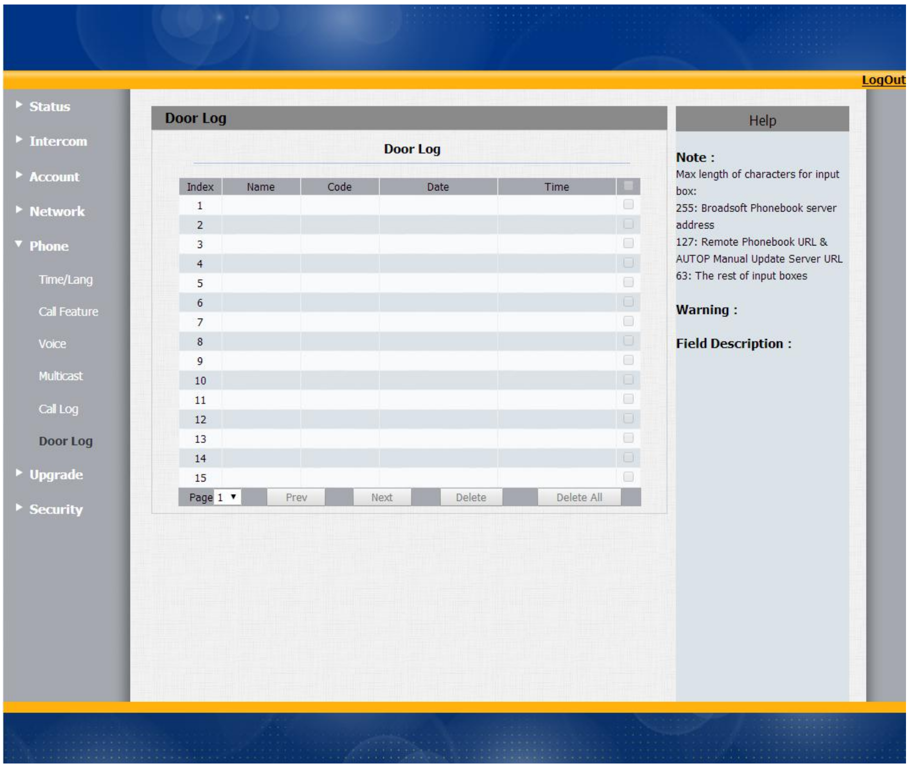
Sections	Description
Multicast Setting	To display and configure the Multicast setting. ● Paging Barge: Choose the multicast number ,the range is 1-10. ● Paging priority Active: Enable o disable the multicast.
Priority List	To setup the multicast parameters. ● Listening Address: Enter the IP address you need to listen ● Label： Input the label for each listening address

3.6.5. Call Log



Sections	Description
Call History	To display call history records. Available call history types are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls.Users can check the call history in detail. Tick the number to delete or delete all logs. R26 supports 100 call logs.

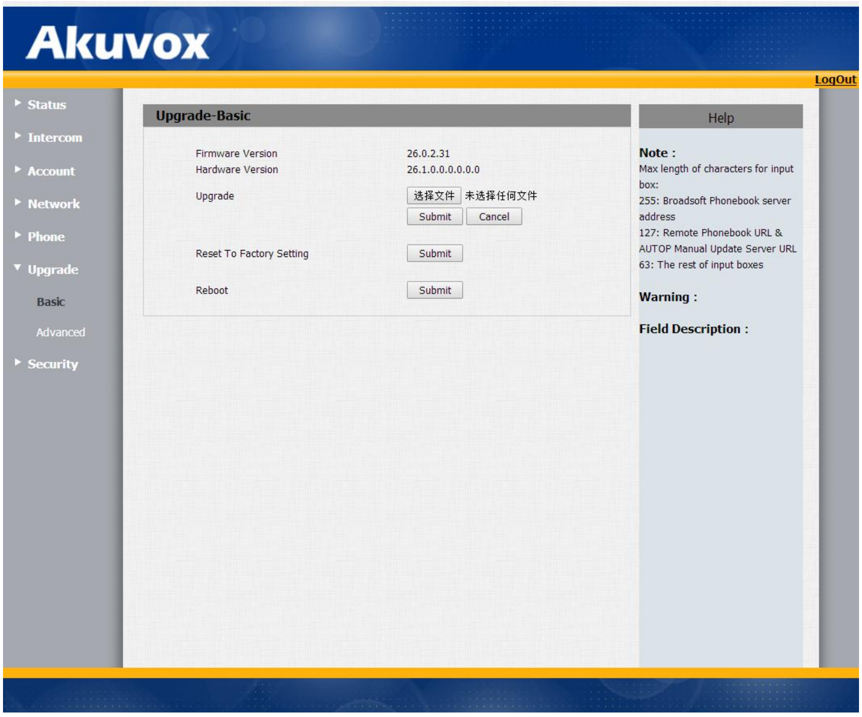
3.6.6. Door Log



Sections	Description
Door Log	To display unlock history Users can check the unlock information in detail. User can delete one or all logs. The maximum door log is 500.

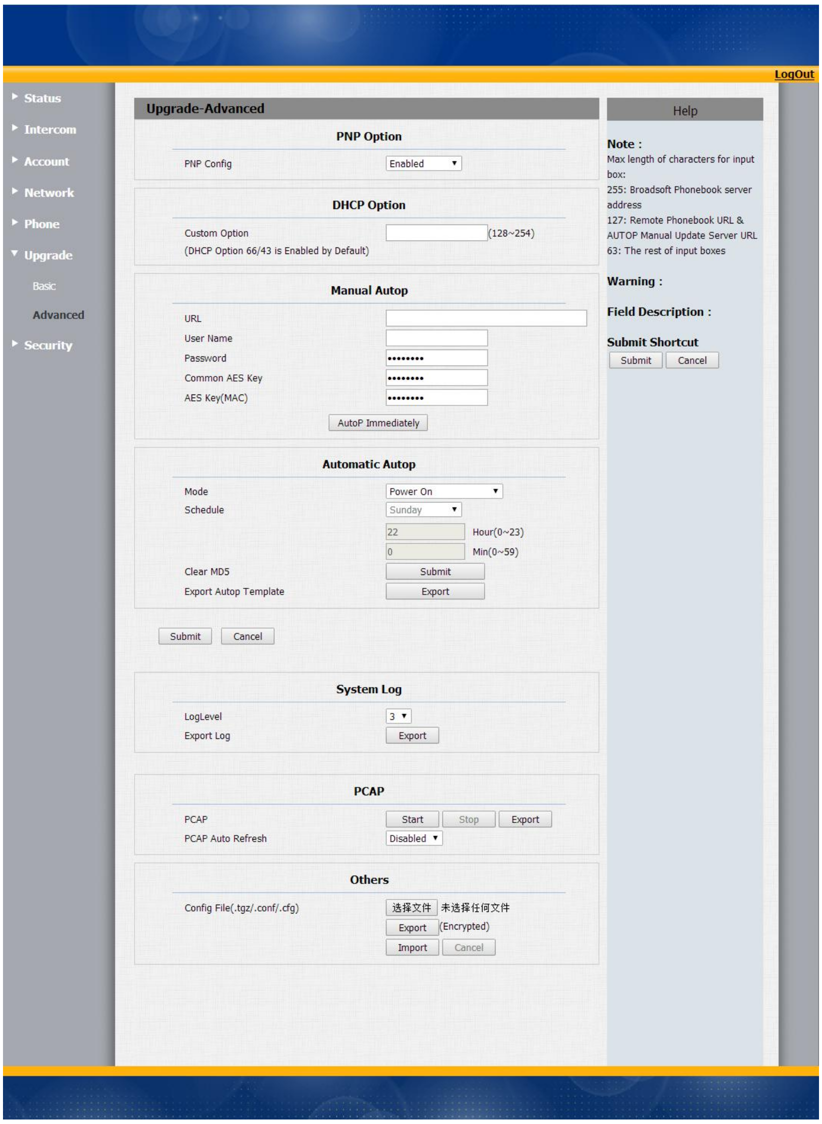
3.7. Upgrade

3.7.1. Basic



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

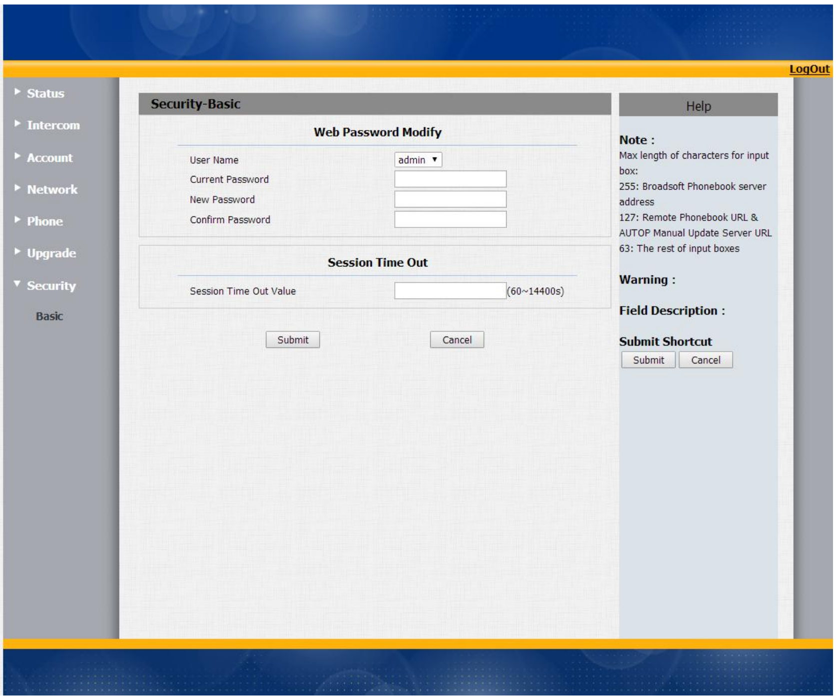
3.7.2. Advanced



Sections	Description
PNP Option	To display and configure PNP setting for Auto Provisioning. ● 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).
DHCP Option	To display and configure custom DHCP option. ● DHCP option: If configured, IP Phone will use designated DHCP option to get Auto Provisioning server's address via DHCP. This setting require DHCP server to support corresponding option.

Manual Autop	To display and configure manual update server's settings. ● URL: Auto provisioning server address. ● User name: Configure if server needs an username to access, otherwise left blank. ● Password: Configure if server needs a password to access, otherwise left blank. ● Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file. ● AES Key (MAC): Used for IP phone to decipher MAC-oriented auto provisioning configuration file(for example, file name could be 0c1105888888.cfg if IP phone's MAC address is 0c1105888888). **Note:** AES is one of many encryption, it should be configure only configure file is ciphered with AES, otherwise left blank.
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.
System Log	To display system log level and export system log file. ● 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.
PCAP	To start,stop packets capturing or to export captured Packet file. ● Start:To start capturing all the packets file sent or received from IP phone. ● Stop:To stop capturing packets. **Note**:IP phone will save captured packets file to a temporary file,this file maximum size is 1M(mega bytes), and will top capturing once reaching this maximum size.

3.8. Security



Sections	Description
Web Password Modify	To modify user's password. ● Current Password: The current password you used. ● New Password: Input new password you intend to use. ● Confirm Password: Repeat the new password.
Session Time Out	Over the Time out value, users need to login again.