R26X Door Phone User Manual

Content

Overview

1 Product Description

R26C

R26P

Akuvox R26X is a SIP-compliant handfree 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. User can also use RF card to unlock the door(Only R26C). 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.

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（Optional）
- 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.

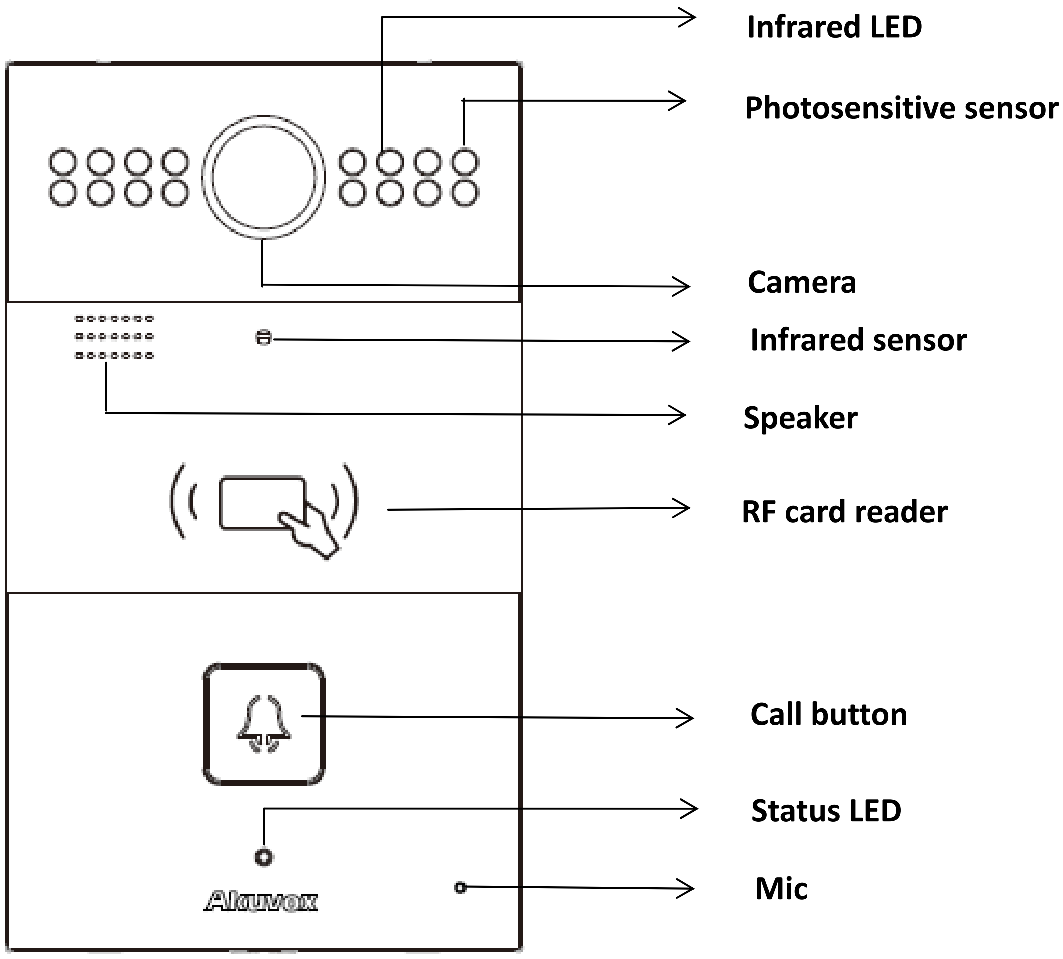
Door Entry Feature

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

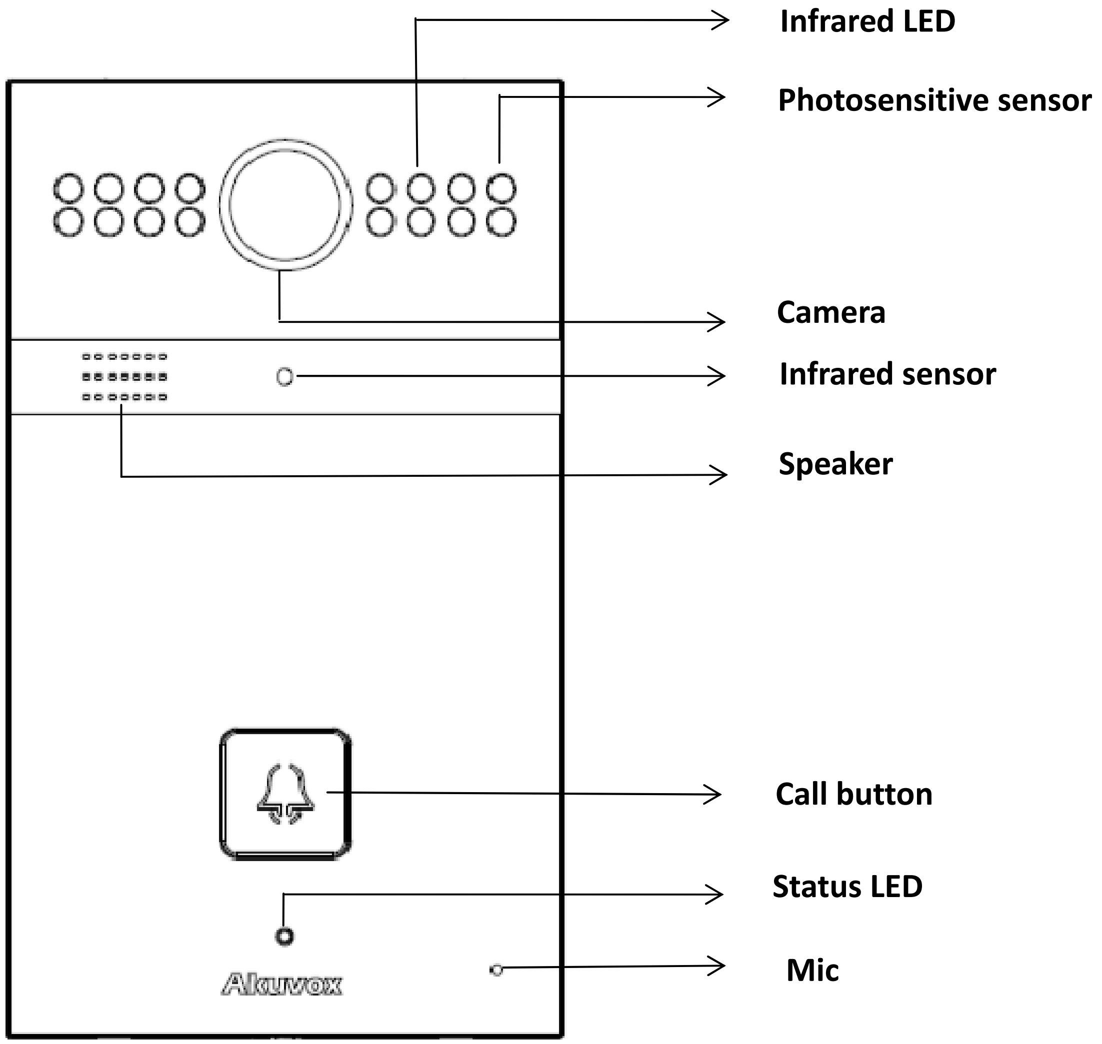
Network Features

- 1x10/100Mbps Ethernet Port
- Protocols support: IPv4, HTTP, HTTPS, FTP, SNMP, DNS, NTP, RTSP, RTP, TCP, UDP, ICMP, DHCP, ARP

3 Panel Description



R26C


Infrared LED
Photosensitive sensor
Camera
Infrared sensor
Speaker
Call button
Status LED
Mic
Akuvox

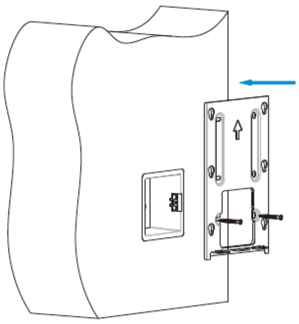
R26P

4 Unpacking

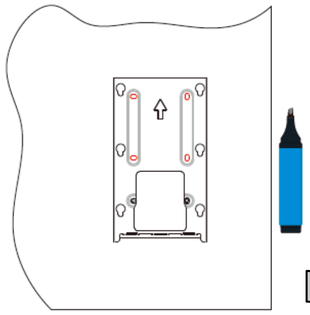
Name	Quantity
R26C/P	1
Back cover	1
Wall bracket	1
Cable buckle	1
M4X20 screw	2
ST4x20 screw	4
Screw fixing seat	4
M3X5 screw	4
M3X10 screw	1

5 Installation

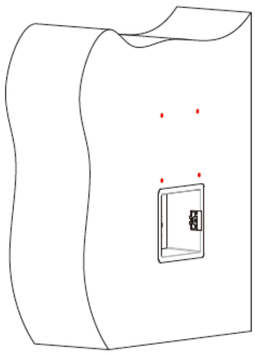
With 86 embedded box:



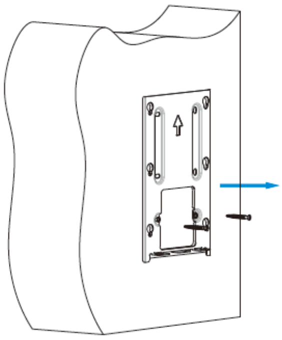
Use 2 M4X20 screws to fix the wall bracket and Embedded box.



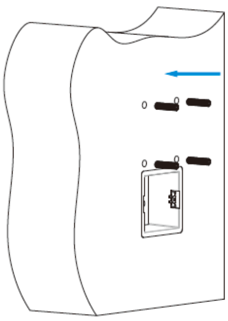
Use the pen to mark the hole location on the wall bracket



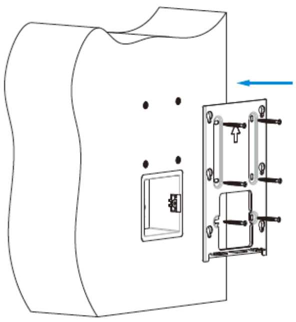
Use 6mm hand drill to make the red hole you marked before



Take down the M4 screws and remove the wall bracket

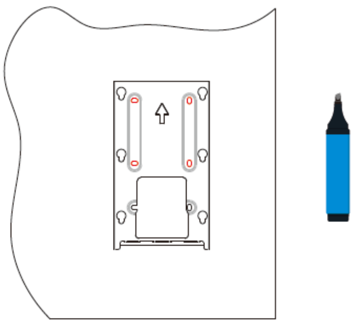


Insert the screw fixing seat into the hole

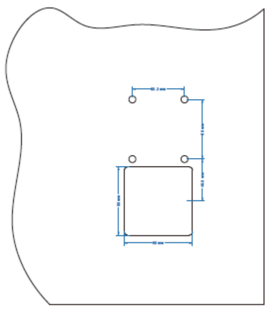


Use 4 St4 screws and 2 M4 screws to fix the wall bracket

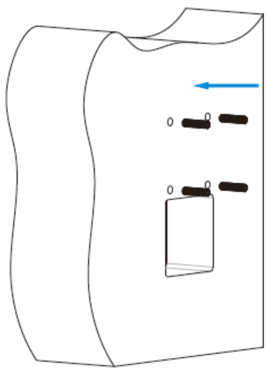
Without 86 embedded box, you can follow the below step:



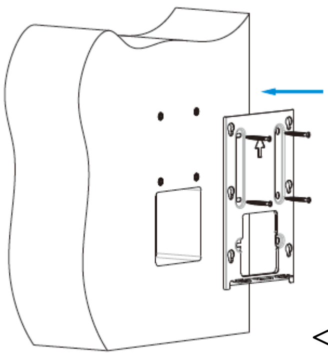
Mark the red hole of the bracket in the wall via using pen



Use 6mm hand drill to make the red hole you marked before ,and dig a 5cm depth square hole

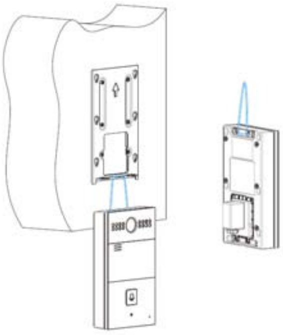


Insert the screw fixing seat into the hole

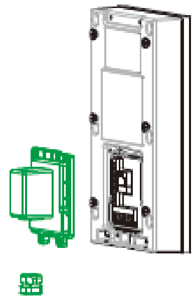


Use 4 ST4 screws to fix the wall bracket

Back cover installation:

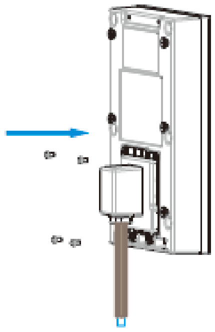


For convenient wiring, hang R26X in the bracket via using rope



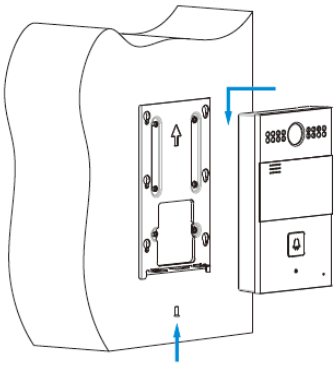
Take down the back cover ,cable buckle and sealing ring

Make cables go through all the components of the bottom hole of the back cover , connect with the contacts in the main board. Then put the cable buckle and sealing ring into the back cover.



Use 4 M3x5 screws to fix the back cover

Device mounting:



Place R26 into the wall bracket hole . Push down so that the device is fixed securely. Lock the the M3X10 screws at the bottom of device.

Configuration

1 Web login

1.1 Obtaining IP address

The Akuvox R26X uses Static IP by default, and the default IP address is 192.168.1.100.

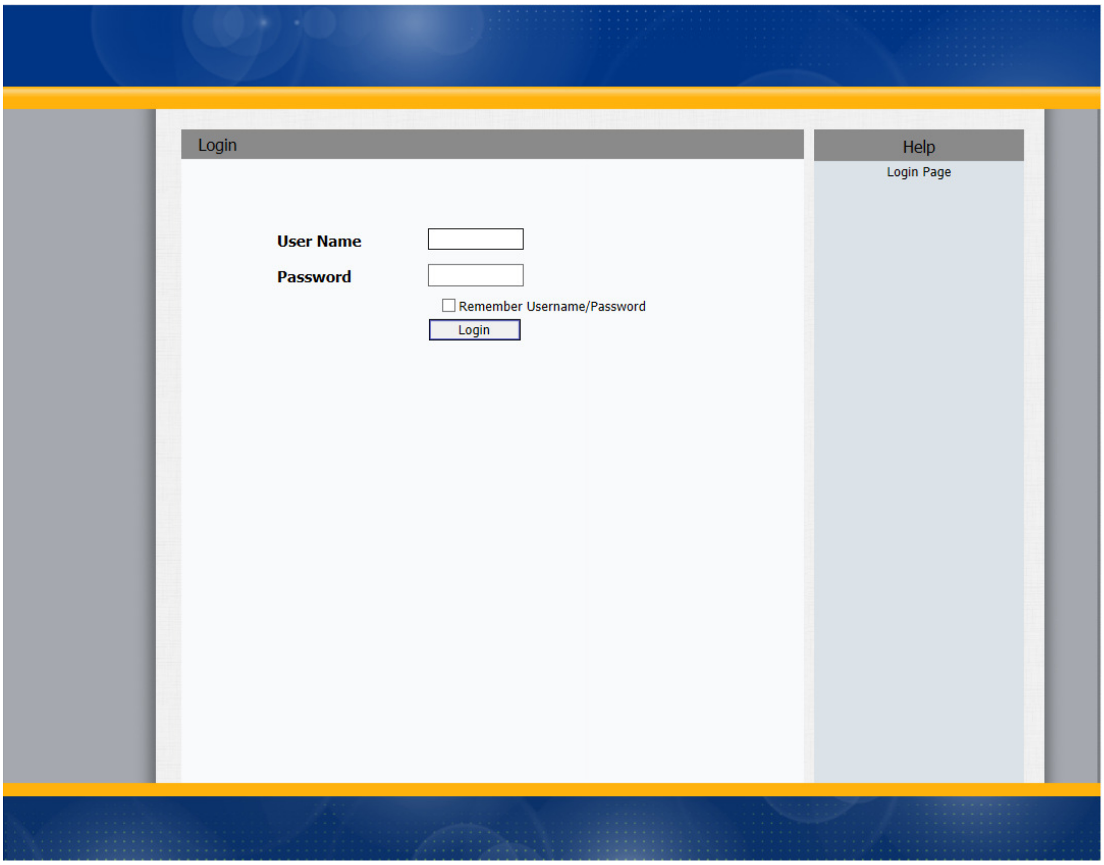
If the IP address is unknown, after power on, when you see the LED light turns Blue, press the call button about 5s, the phone will announce its IP.

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:

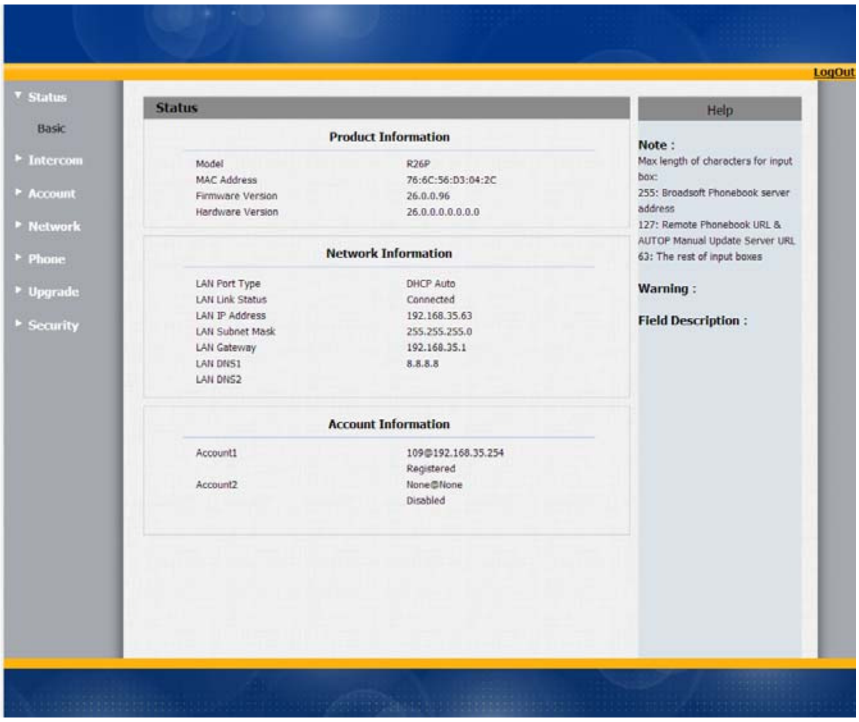
User name: admin

Password: admin


Login
Help
Login Page
User Name
Password
Remember Username/Password
Login

2 Status-Basic

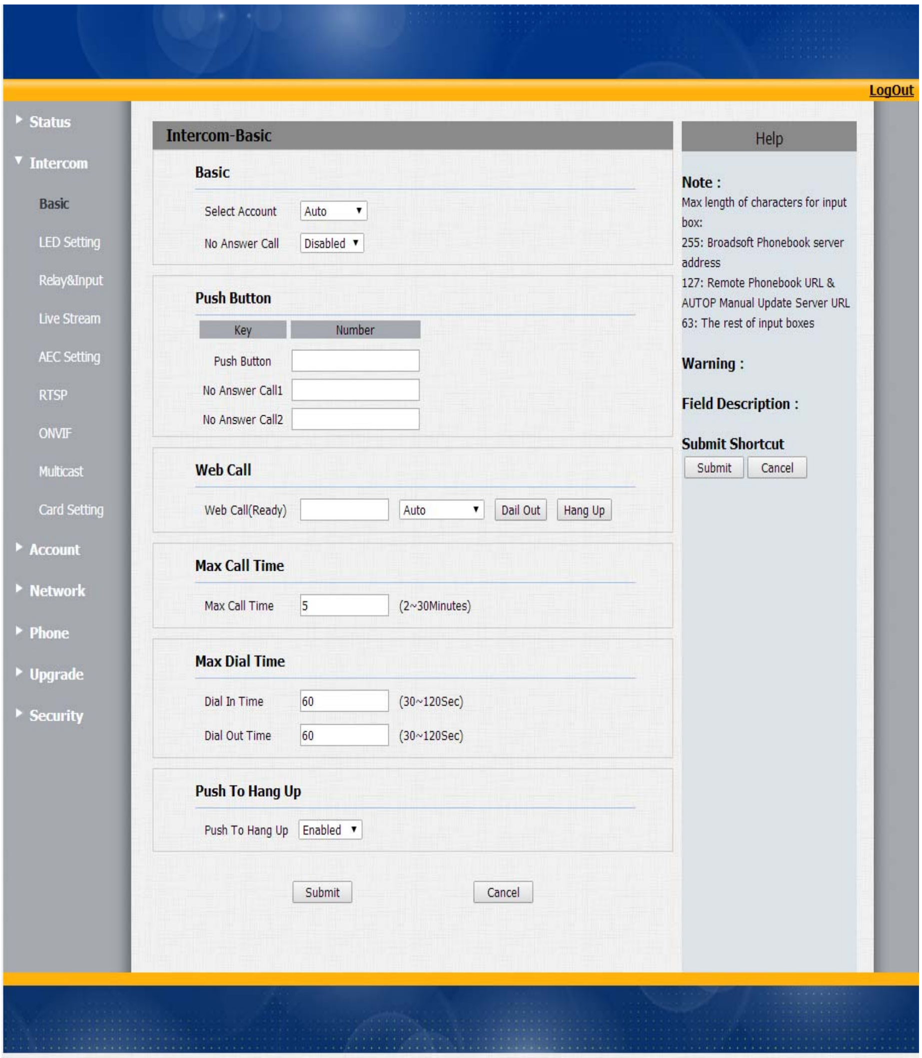
Status, including product information, network information and Account information, can be viewed from Status -> Basic.



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 firmware.
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 Intercom-Basic

Go to the path: Intercom-Basic

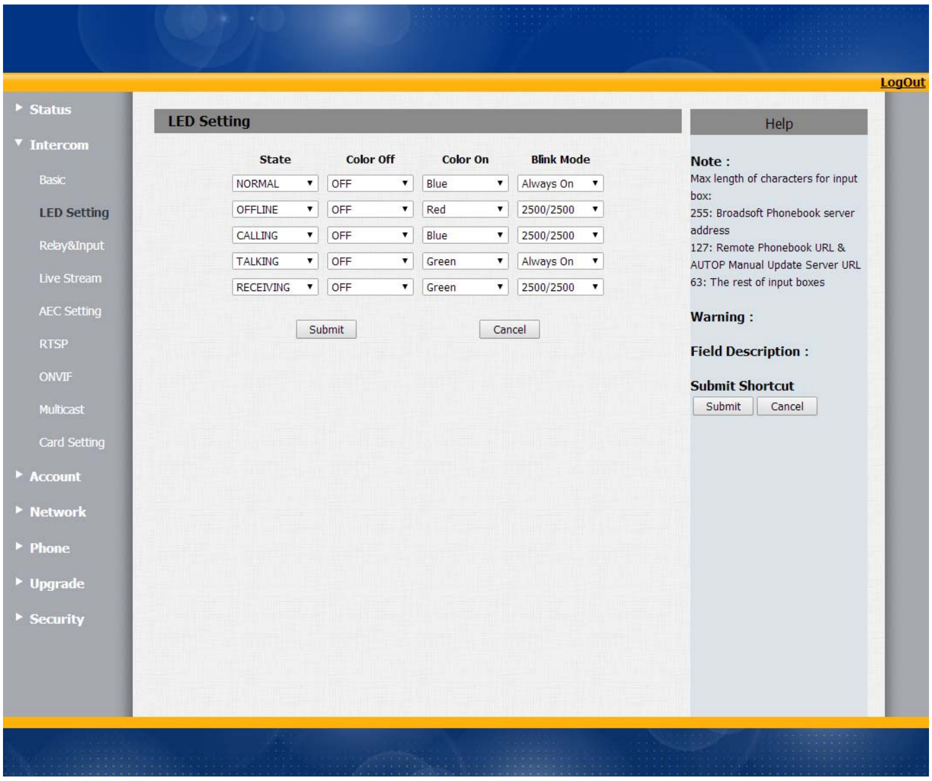


Sections	Description
Basic	● Select Account: R26 supports 2 accounts. You can choose one account or Auto mode for the following Intercom basic settings. ● No Answer Call：R26 will call to the No answer call number in order when the ringtone is time out without answer of the push button number. Disable 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 anser call numbers or one no answer call number.
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 R26, 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

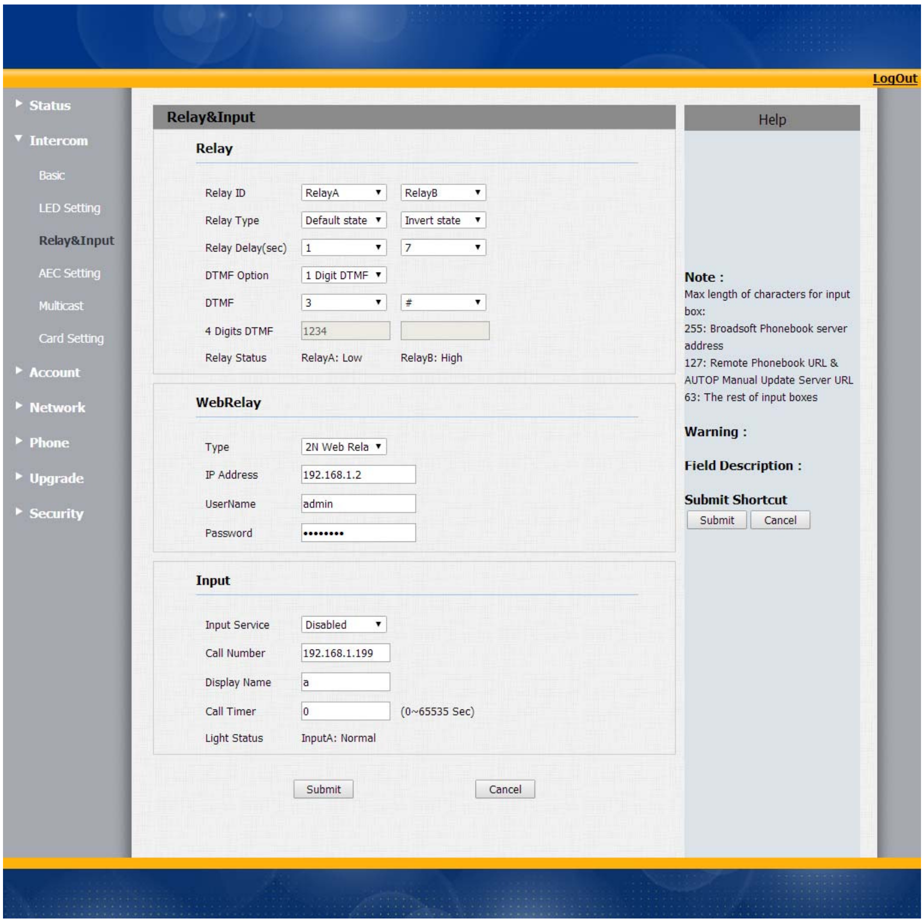
4 Intercom-LED Setting

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.

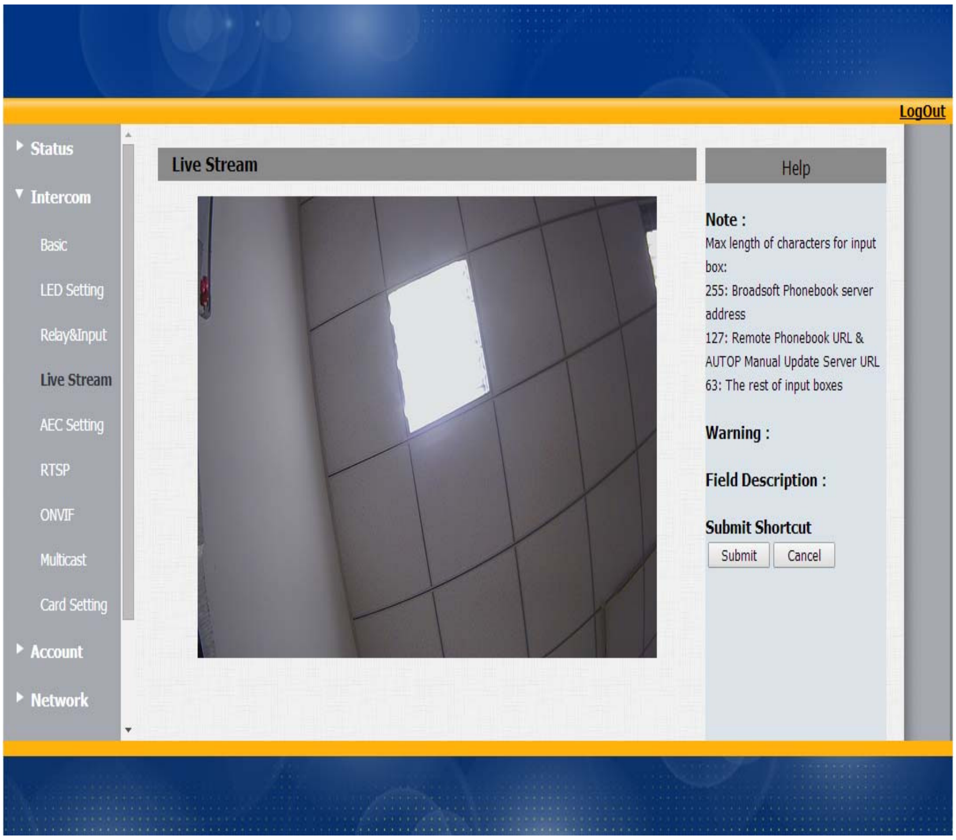
5 Intercom- Relay&Input



Sections	Description
Relay	To configure some settings about unlock ● Relay Select: R26 supports 2 relays ● Relay Type: Different locks use different relay types, positive or negative. If you connect the Lock in NO connector, select positive type. Otherwise using negative type. ● Relay Delay(sec): Allows door remain “open” for certain period The range is from 1 to 10 seconds ● DTMF Option: R26 supports 1digit or 4 digits DTMF unlock code. Please select one type and enter the corresponding code. ● DTMF: Setup 1 digit DTMF code for remote unlock ● 4 Digits DTMF : Setup 4 digits DTMF code for remote unlock. ● Status: Different relay types will show different status.
Web relay	R26 can support extra web relay. This function is more safety

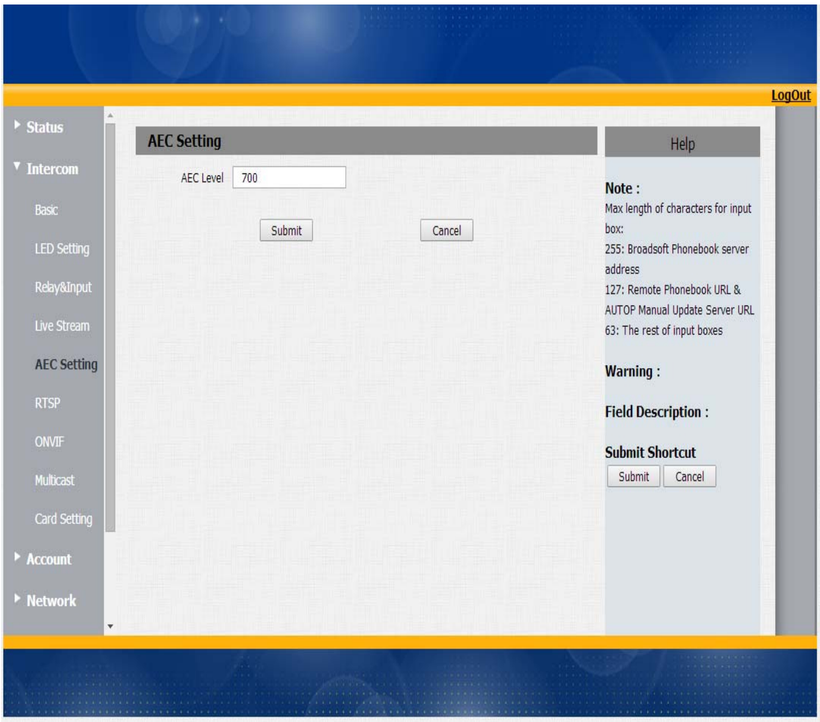
	to use DTMF code to remote unlock. ● Type: Connect web relay and choose the type. ● IP Address: Enter web relay IP address. ● User name: it is an authentication for connecting web relay ● password: it is an authentication for connecting web relay **Note:** Users can modify username and password in web relay website.
Input	There is a sensor that used to anti vandal in R26X. When R26X is broken by violent means. The sensor will be triggered, then management center will receive the alarm. ● Service: Enable by default ● Call Number: To setup management center number for alarm. ● Display Name: Which is sent to the other call party for displaying ● 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. ● Light Status: The status will change according to the sensor. Once the sensor is triggered , the status will show Warning. Normal by default.

6 Intercom-Live Stream



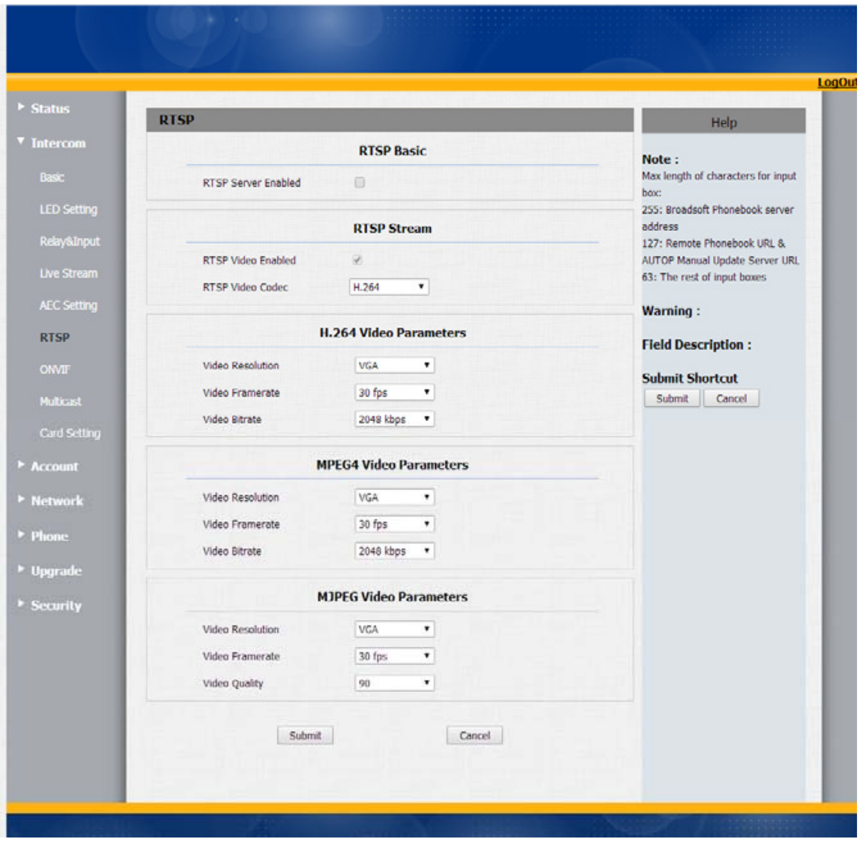
Sections	Description
Live Stream	To check the real-time video from R26.

7 Intercom-AEC Setting



Sections	Description
AEC Level	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.

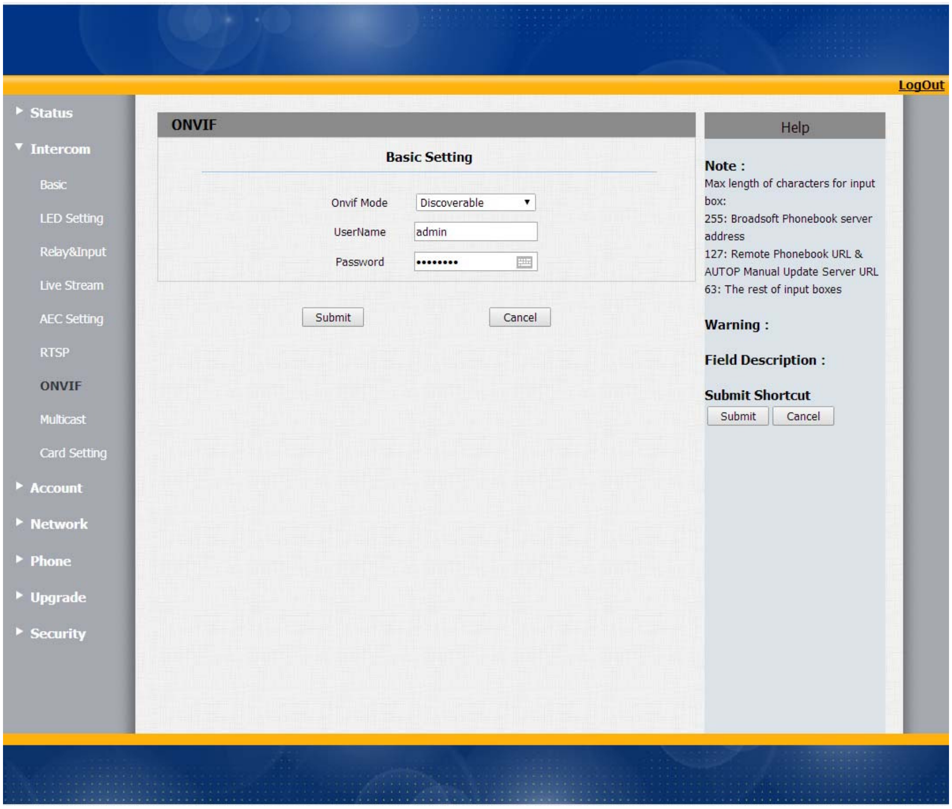
8 Intercom-RTSP



Sections	Description
RTSP Basic	To active the RTSP function, then R26 can be monitored.
RTSP Stream	To enabled RTSP video and select the video codec. R26 supports 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

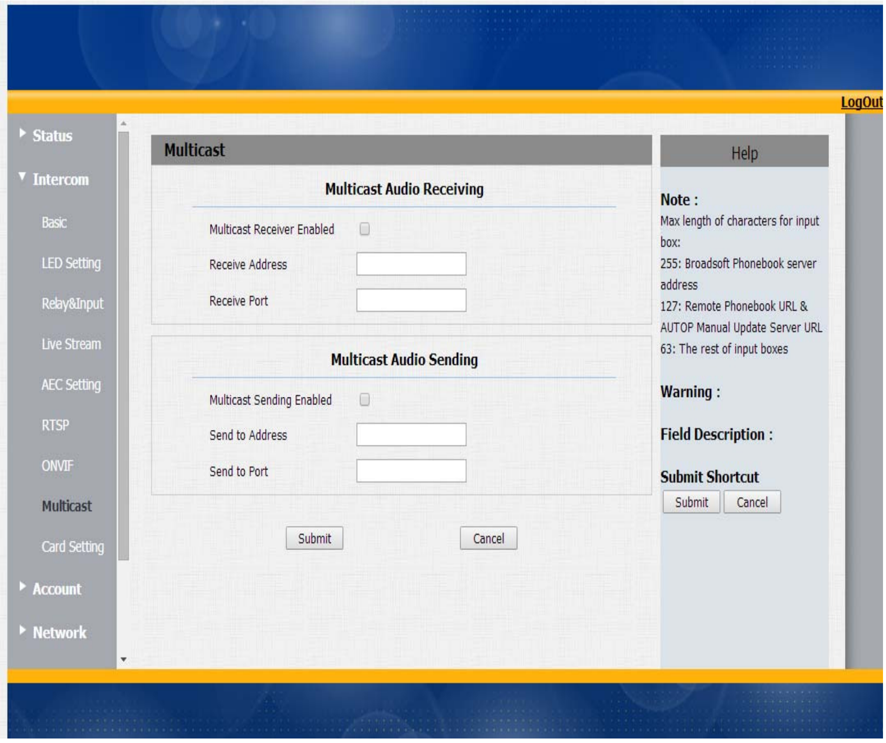
	quality video image and has a flexible configuration in video definition and Compressed frames To modify the resolution,framerate and bitrate of MJPEG

9 Intercom-Onvif



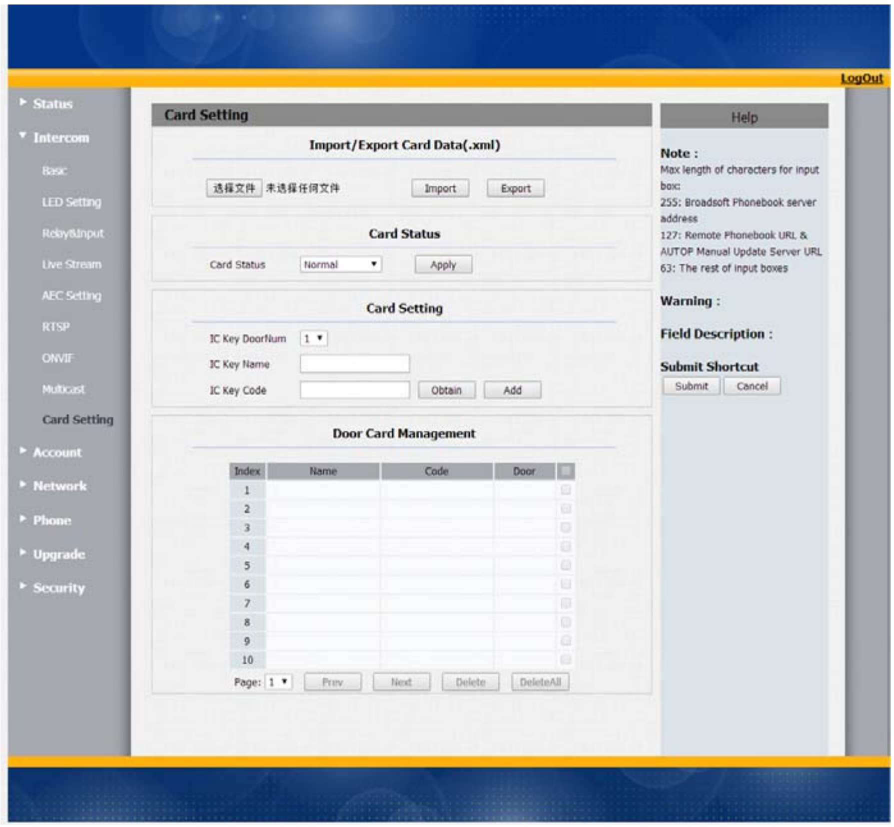
Sections	Description
Basic Setting	To setup the Onvif function parameters. It is used to connect with the corresponding Onvif tool. ● Onvif Mode: Two modes - Discoverable and Non-discoverable. Discoverable by default. Only Discoverable mode, then Onvif software can search R26. ● User Name: To modify 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.

10 Intercom-Multicast



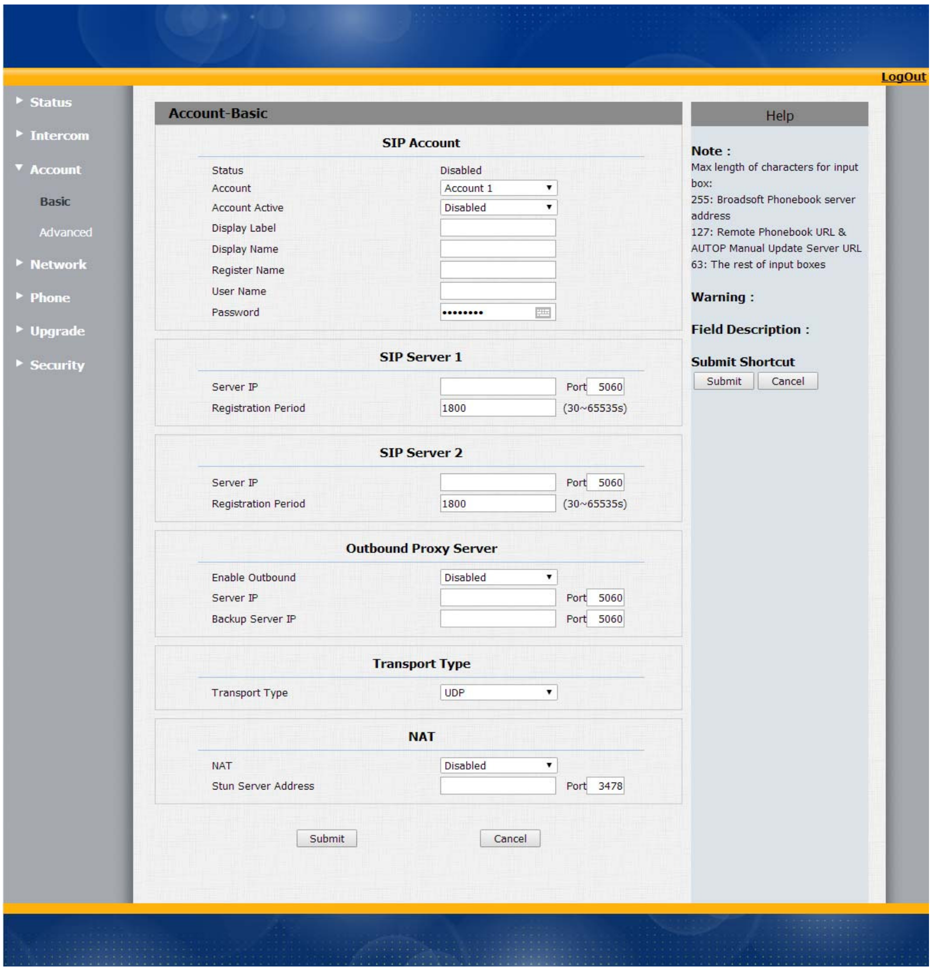
Sections	Description
Multicast Audio Receiving	To display and configure the Multicast setting. ● Multicast Receiver Enable: Enable receiver multicast function. ● Receiver address : Setup the multicast address. ● Receiver port : setup the multicast address port.
Multicast Audio Sending	To setup the multicast parameters. ● Multicast Sending Enable: Enable sender multicast function ● Send to Address: setup the multicast address. ● Send to port: setup the multicast address port.

11 Intercom-Card Setting(Optional)



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: R23X 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 R26 RF Card Read area, then click Abtain button. After R26 read 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.

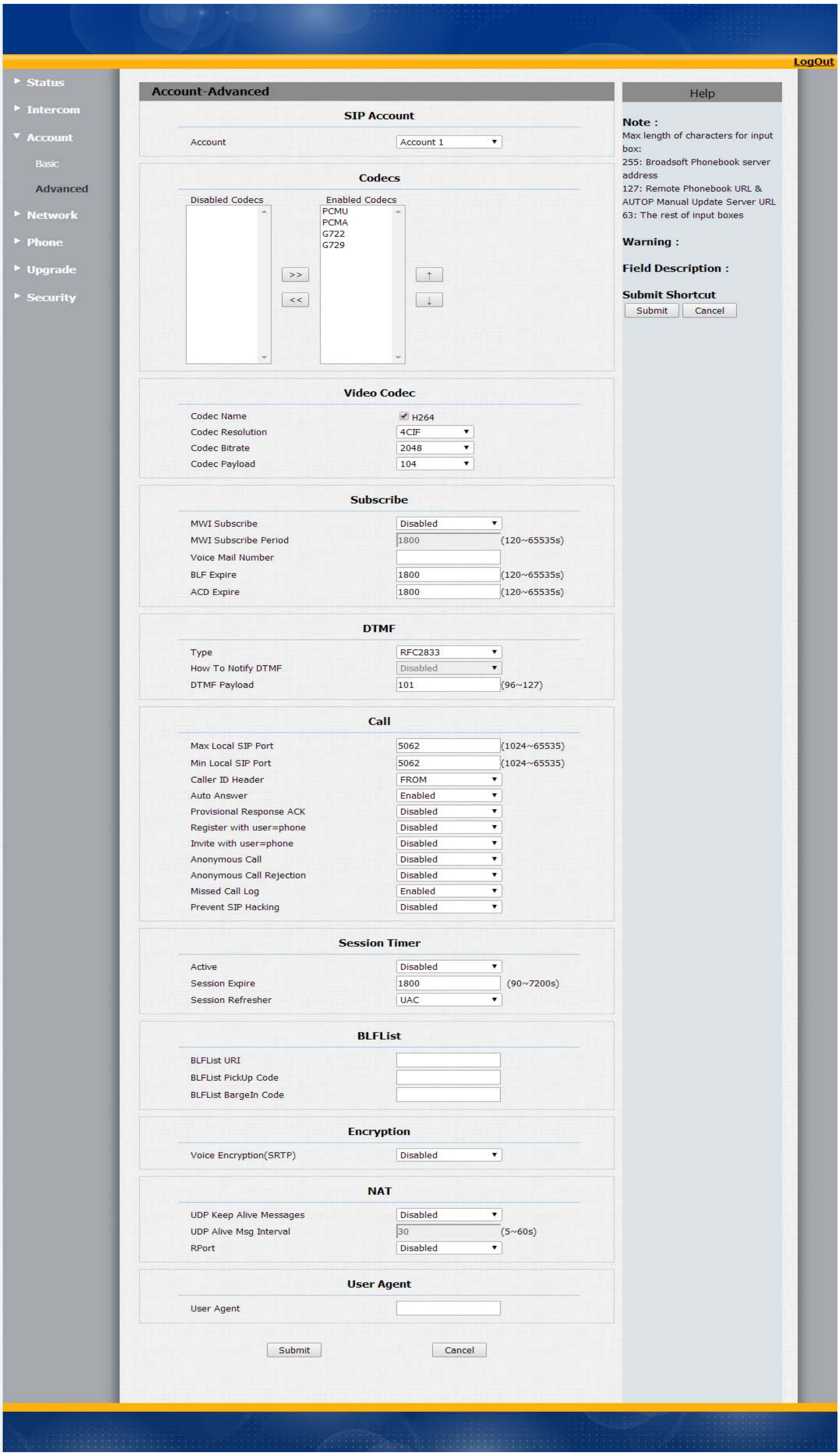
12 Account-Basic



Sections	Description
SIP Account	To display and configure the specific Account settings. ● Status: To display register result. ● 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.

13 Account-Advanced

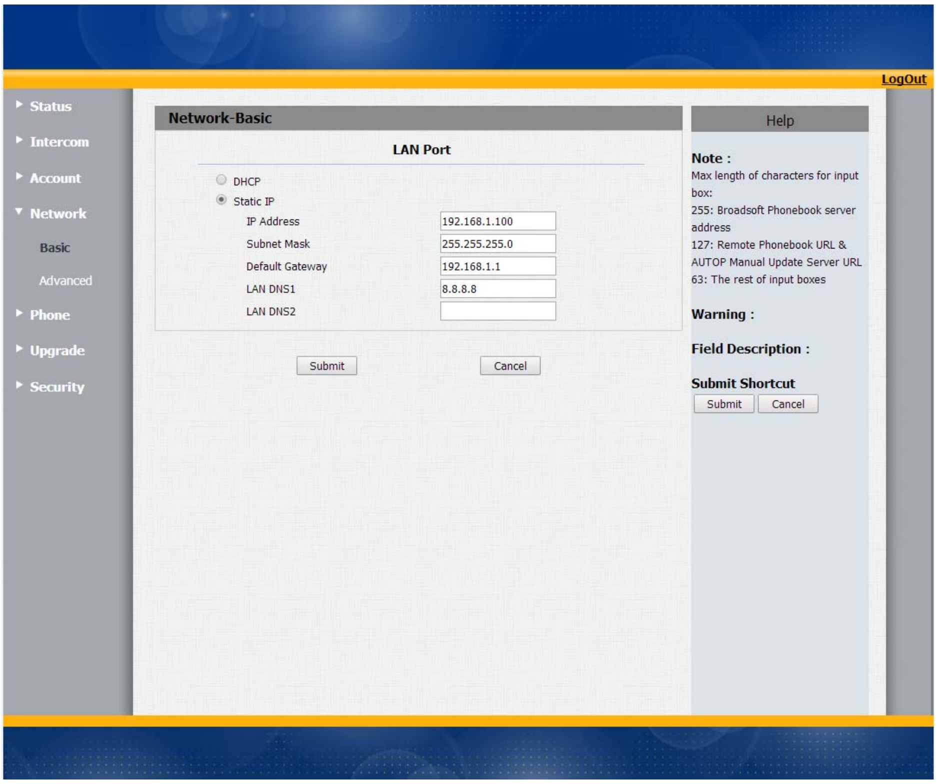


Sections	Description
SIP Account	To display current Account settings or to select which account to display.
Codecs	To display and configure available/unavailable codecs list. Codec means coder-decoder which is used to transfer analog signal to digital signal or vice versa. Familiar codecs are PCMU(G711U), 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. ● Is escape non Ascii character: To transfer the symbol to Ascii character. ● 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.
BLF List	To display or configure BLF List URI address. ● BLF List URI: BLF List is short for Busy Lamp Field List. ● BLFList PickUp Code: To set the BLF pick up code. ● BLFList BargeIn Code : To set the BLF barge in code.
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

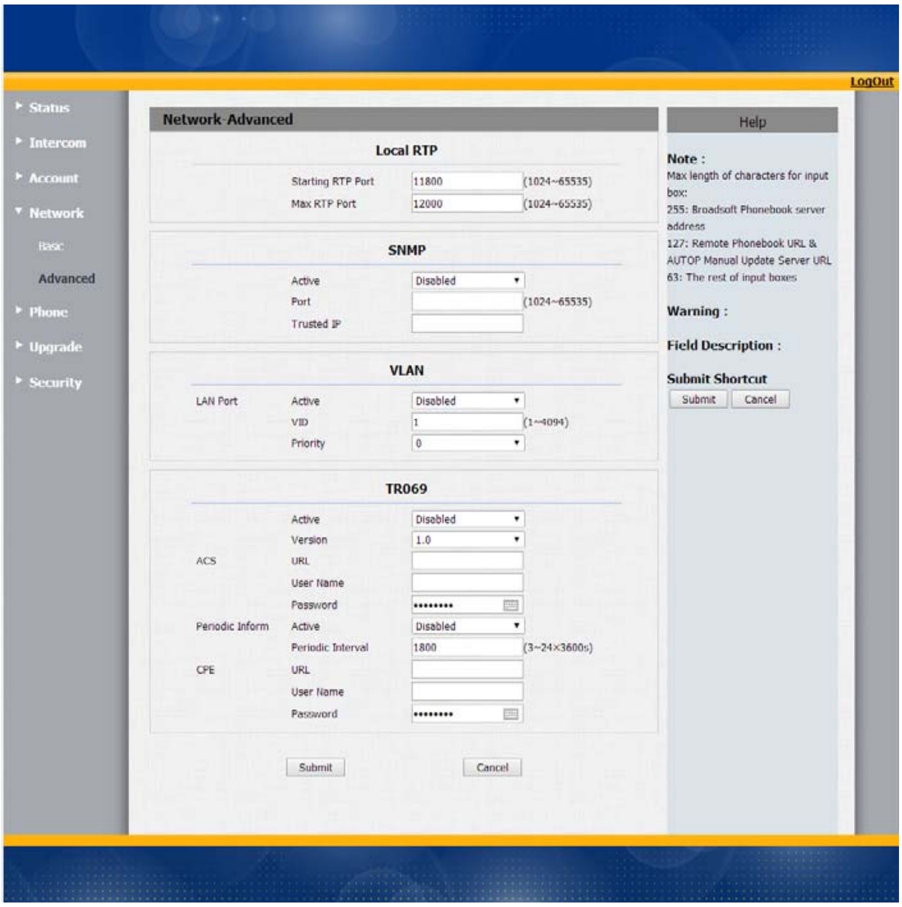
	into outgoing SIP message for designated account.
User Agent	One can customize User Agent field in the SIP message; If user agent is set to specific value, user could see the information from PCAP. If user agent is not set by default, user could see the company name, model number and firmware version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

14 Network-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.

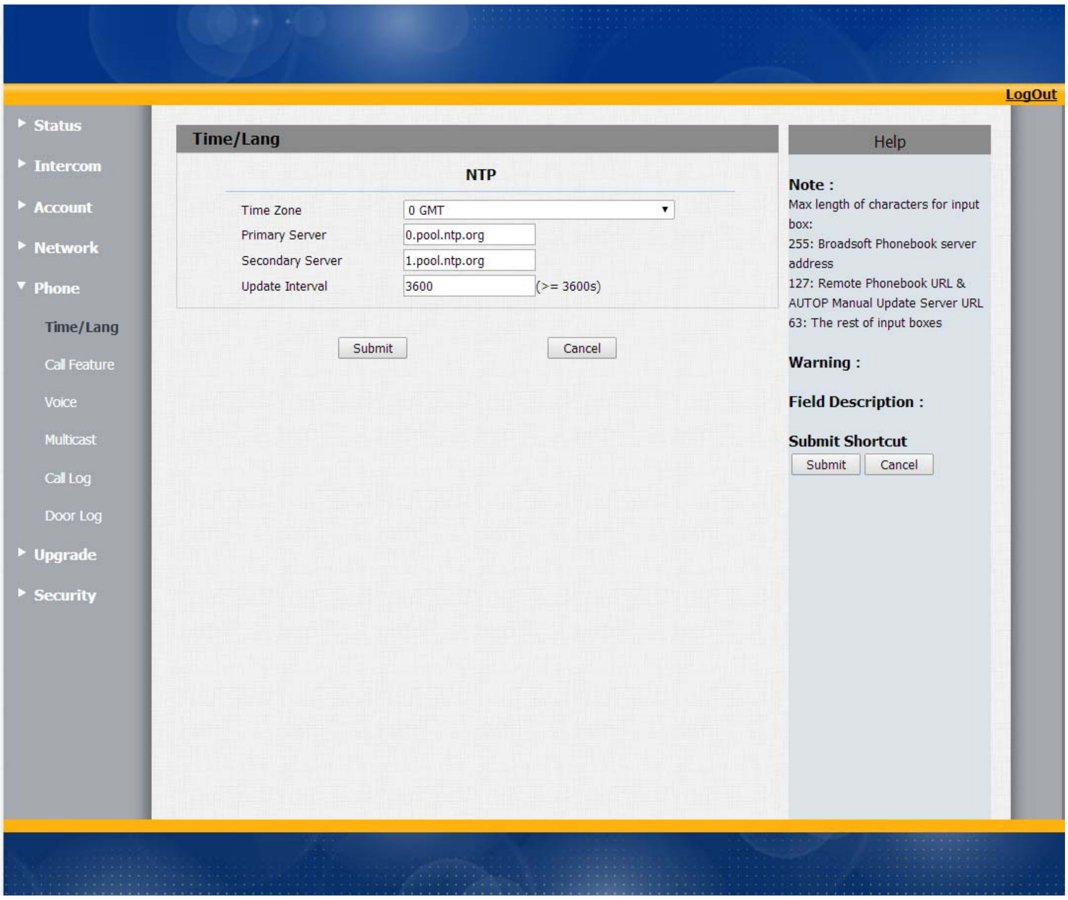
15 Network-Advanced



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.

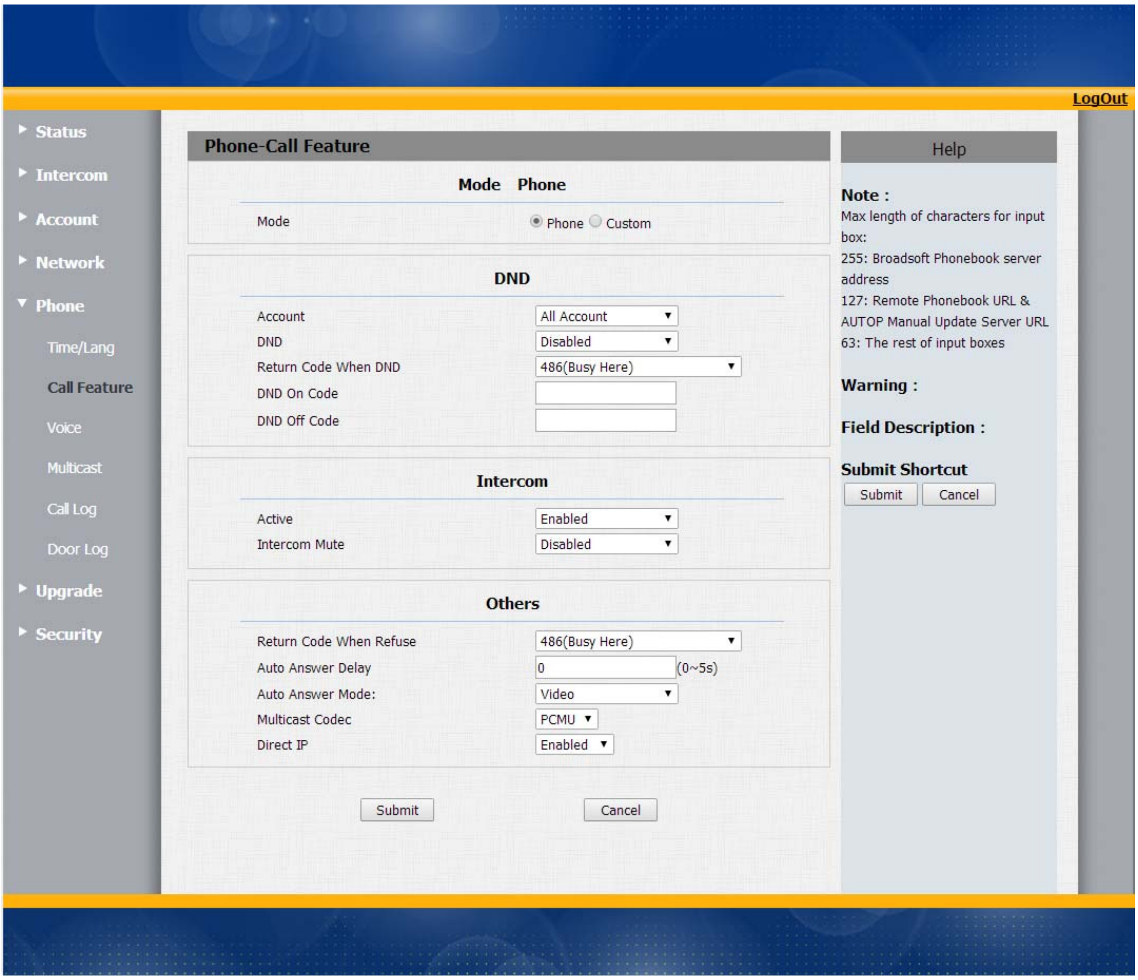
	Note: Please consult your administrator for specific VLAN settings in your networking environment.
TR069	To display and configure TR069 settings. ● Active: To enable or disable TR069 feature. ● Version: To select supported TR069 version (version 1.0 or 1.1). ● ACS/CPE: ACS is short for Auto configuration 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.

16 Phone-Time/Language



Sections	Description
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.

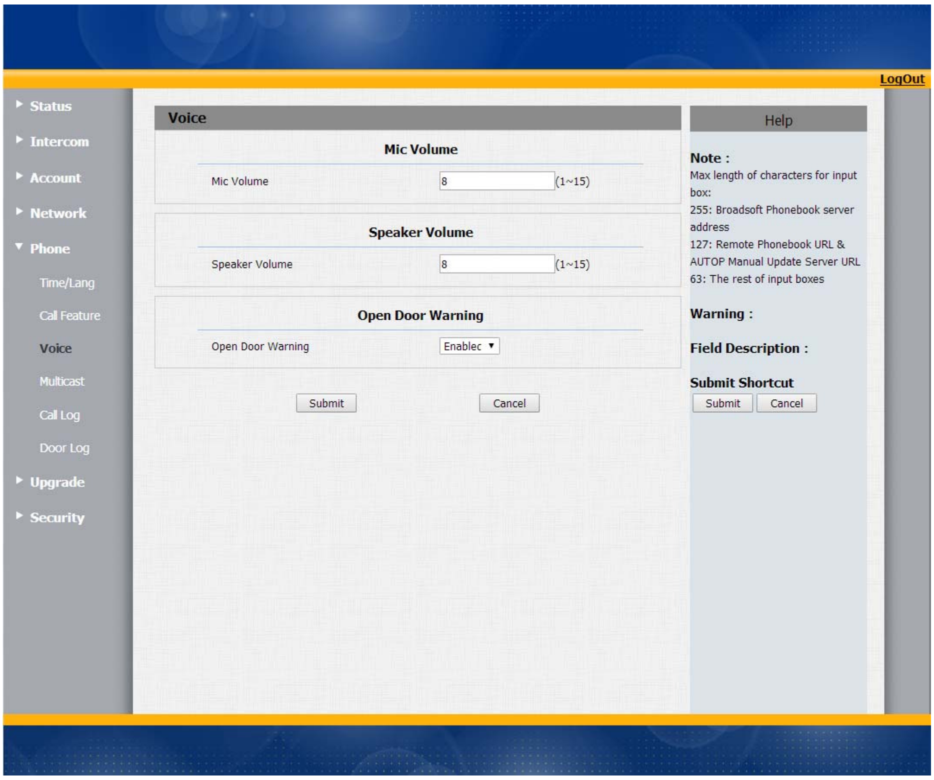
17 Phone-Call Feature



Sections	Description
Mode	● Mode: Select the desired mode.
DND	DND (Do Not Disturb) allows IP phones to ignore any incoming calls. ● Return Code when DND: Determine what response code should be sent back to server when there is an incoming call if DND on. ● DND On Code: The Code 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. ● DND Off Code: The Code 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.
Intercom	Intercom allows user to establish a call directly with the callee.

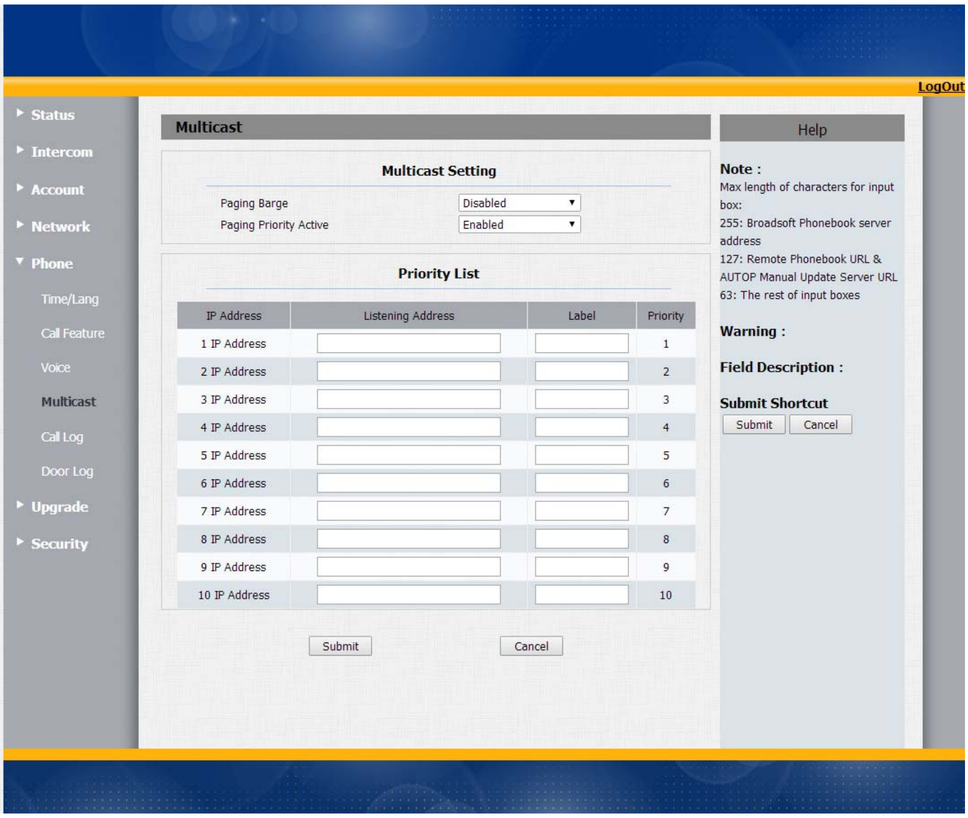
	● Active: To enable or disable Intercom feature. ● Intercom Mute: If enabled, once the call established, the callee will be muted.
Others	● Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected. ● Auto Answer Delay: To configure delay 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.

18 Phone-Voice



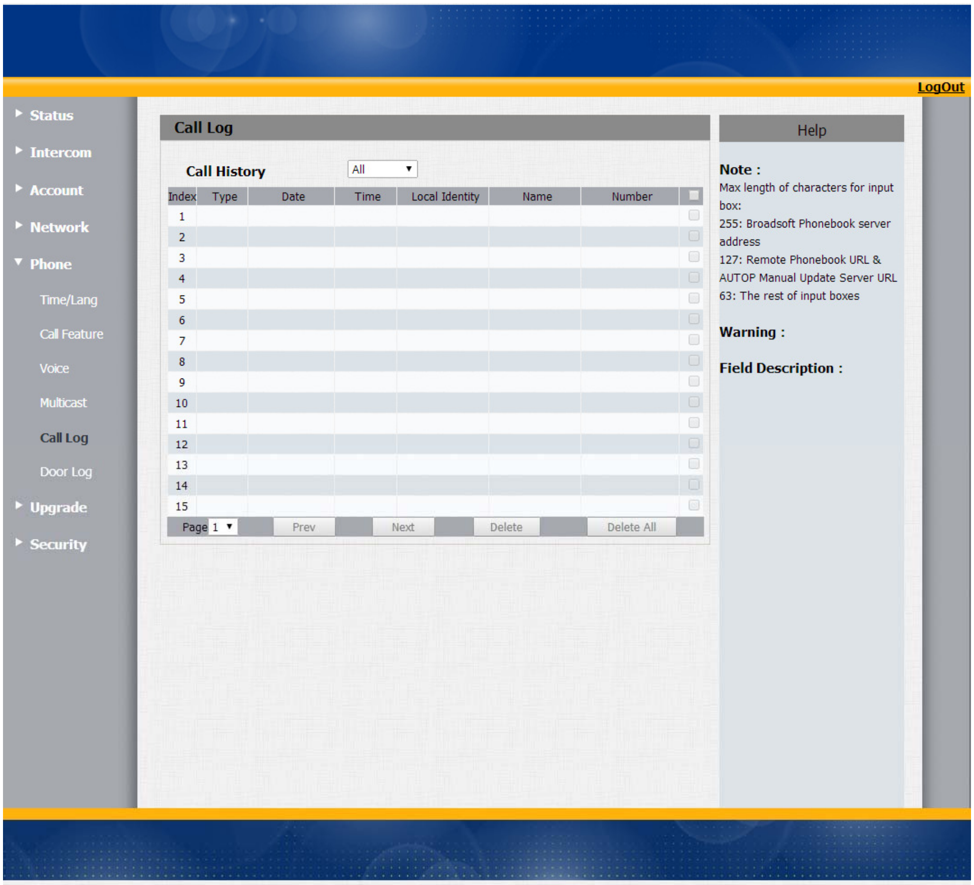
Sections	Description
Mic Volume	To configure Microphone volume , from 1-15. 8 by default.
Speaker Volume	To configure Speaker Volume,from 1-15,8 by default.
Open Door Warning	To configure door opening voice. Disable it, you won't hear the prompt voice when the door is opened.

19 Multicast



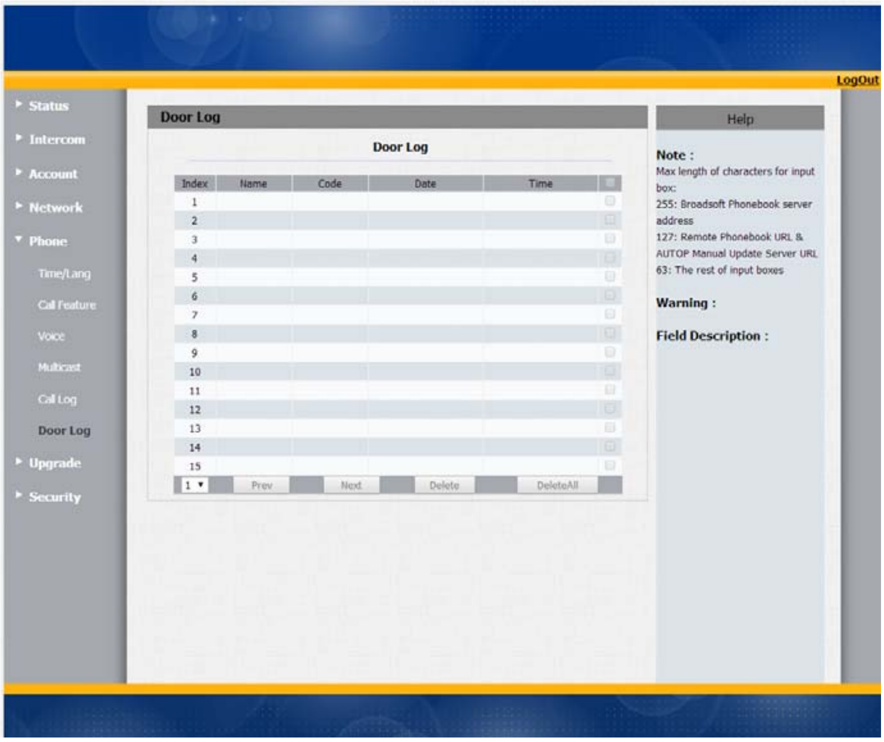
Sections	Description
Multicast Setting	To display and configure the Multicast setting. ● Paging Barge: Choose the multicast number ,the range is 1-10. ● Paging priority Active: Enable o disable the multicast.
Priority List	To setup the multicast parameters. ● Listening Address: Enter the IP address you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

20 Call Log



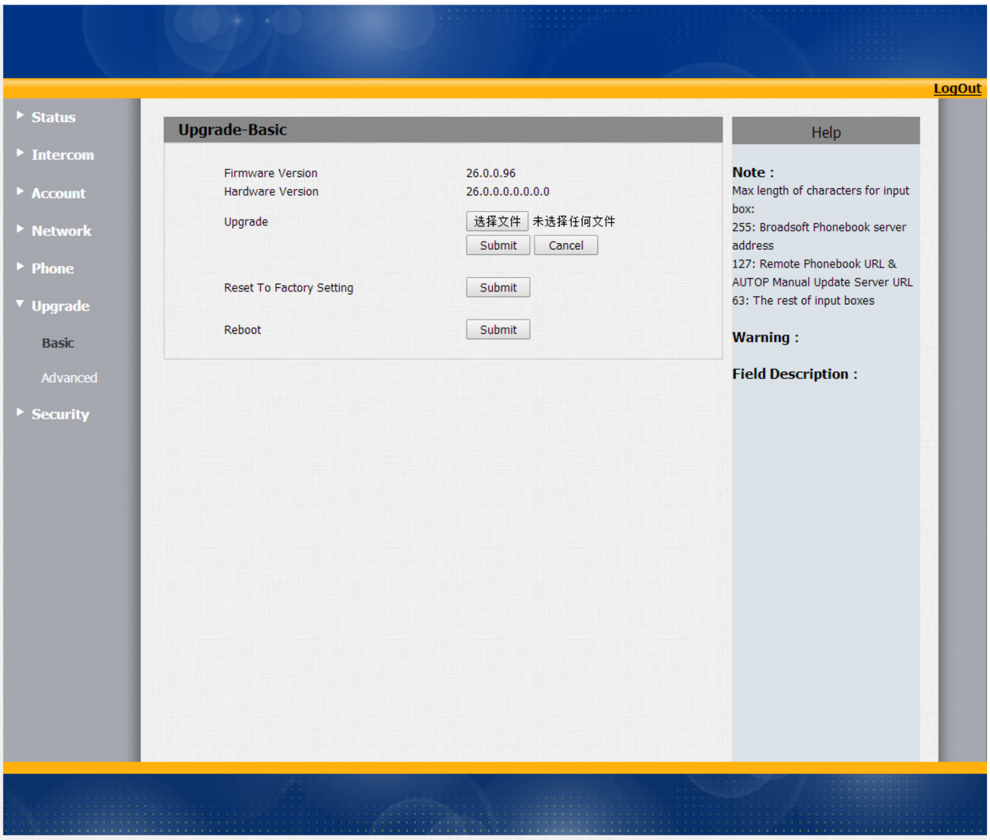
Sections	Description
Call History	To display call history records. Available call history types are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls. Users can check the call history in detail. Tick the number to delete or delete all logs. R26 supports 100 call logs.

21 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.

22 Upgrade-Basic



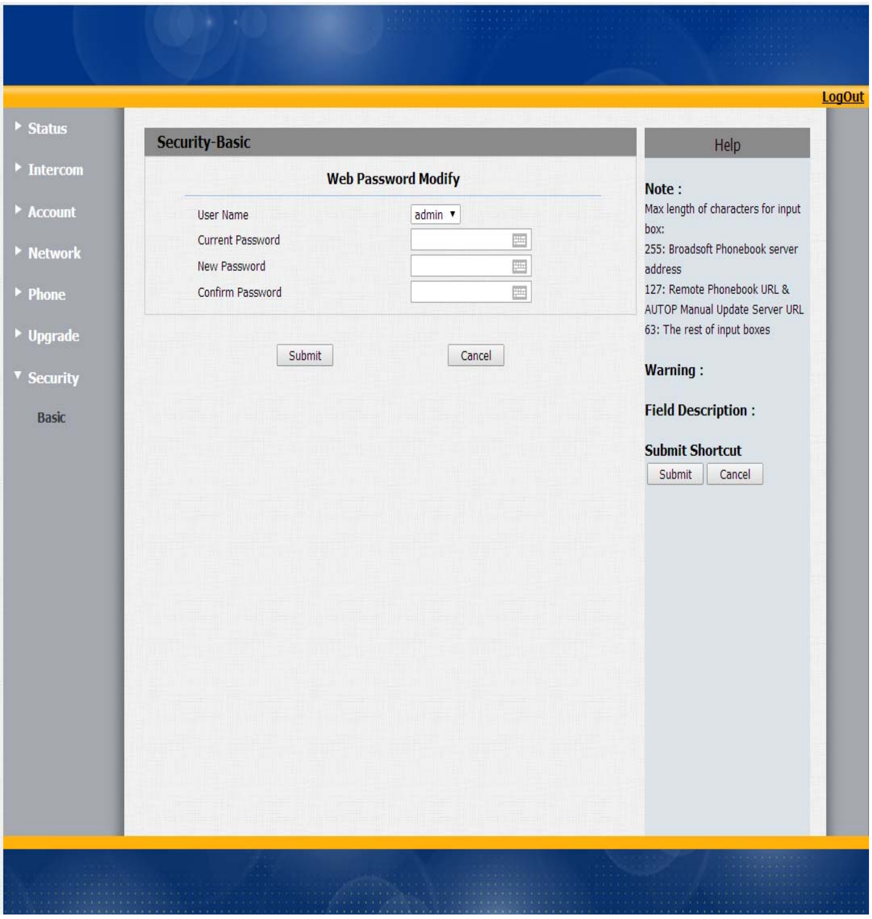
Sections	Description
Upgrade	To select upgrading zip file from local or a remote server automatically. **Note:** Please make sure it's right file format for right model.
Firmware version	To display firmware version, firmware version starts with MODEL name.
Hardware Version	To display Hardware version.
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.

23 Upgrade-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 Update Server	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.
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.

24 Security-Basic



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.