

Akuvox

E21 Emergency Station User Manual

Content

Production Overview.....	3
1 Production description.....	3
2 Features.....	3
3 Panel Description.....	5
4 Installation.....	6
Configuration.....	8
1 Web Login.....	8
1.1 Obtaining the IP address.....	8
1.2 Login the web.....	8
2 Status.....	9
3 Intrecom.....	11
3.1 Basic.....	11
3.2 LED Settings.....	12
3.3 Relay&Input.....	12
3.4 Live Stream(Optional).....	14
3.5 AEC Setting.....	15
3.6 RTSP(optional).....	16
3.7 Onvif(optional).....	17
3.7 Multicast.....	18
4 Account.....	19
4.1 Account->Basic.....	19
4.2 Account-> Advanced.....	20
5 Network.....	24
5.1 Network-> Basic.....	24
5.2 Network-> Advanced.....	25
6 Phone.....	27
6.1 Time/Language.....	27
6.2 Call Feature.....	28
6.3 Voice.....	29
6.4 Multicast.....	30
7 Upgrade.....	30
7.1 Basic Upgrade.....	30
7.2 Advanced upgrade.....	31
8 Security.....	33

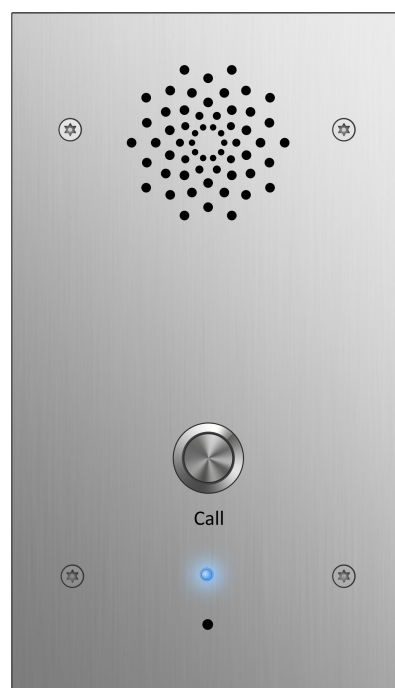
Production Overview

1 Production description

Akuvox E21 Series are outdoor-rated, SIP-compliant hands-free Voice over IP (VoIP) Emergency Stations. It makes the emergency teams to coordinate their rescue missions with high efficiency. E21 supports two types: E21A(Audio) and E21V(Video). They are often used in locations such as: parking facilities, college campuses, medical centers, and industrial parks.



E21V



E21A

2 Features

➤ Key Features

- One panic button input for emergency intercom;
- Two-way audio communication over IP networks with Echo Cancel feature;

- PoE (IEEE802.3af, Power-over-Ethernet);
- Camera resolution of 3M pixel;(E21V Only);
- MPEG-4/MJPEG compression; (E21V only);
- Complies with SIP standard for easy integration in every SIP capable PBXes: CUCM, Avaya, Asterisk, Digium, etc;

➤ **Physical Features**

- Body material: 316 grade stainless steel
- Camera: 3M pixels (E21V only)
- Resolution: up to 1080P(E21V only)
- Button: 1 panic button; 1 reset button (on board)
- Microphone: 1 integrated microphone, IP67
- Speaker: 1W, IP66
- Input Relay: 2 input relays for alarm
- Output Relay: 2 output relays for door opener
- Call Indication: 1 RGB LED (colors: red, green, blue)
- 12V DC input
- Power consumption: less than 12W
- Water-proof & Dust-proof: IP65
- Installation: Flush-mounted, Fit in Clipsal 164/4 back box
- Dimension: PCB - 74x140mm, With flush mount kit - 210x120x61mm

➤ **Phone Features**

- Web support multi-language
- Auto-answer
- Volume control
- Direct IP call without SIP proxy
- Auto-Provision

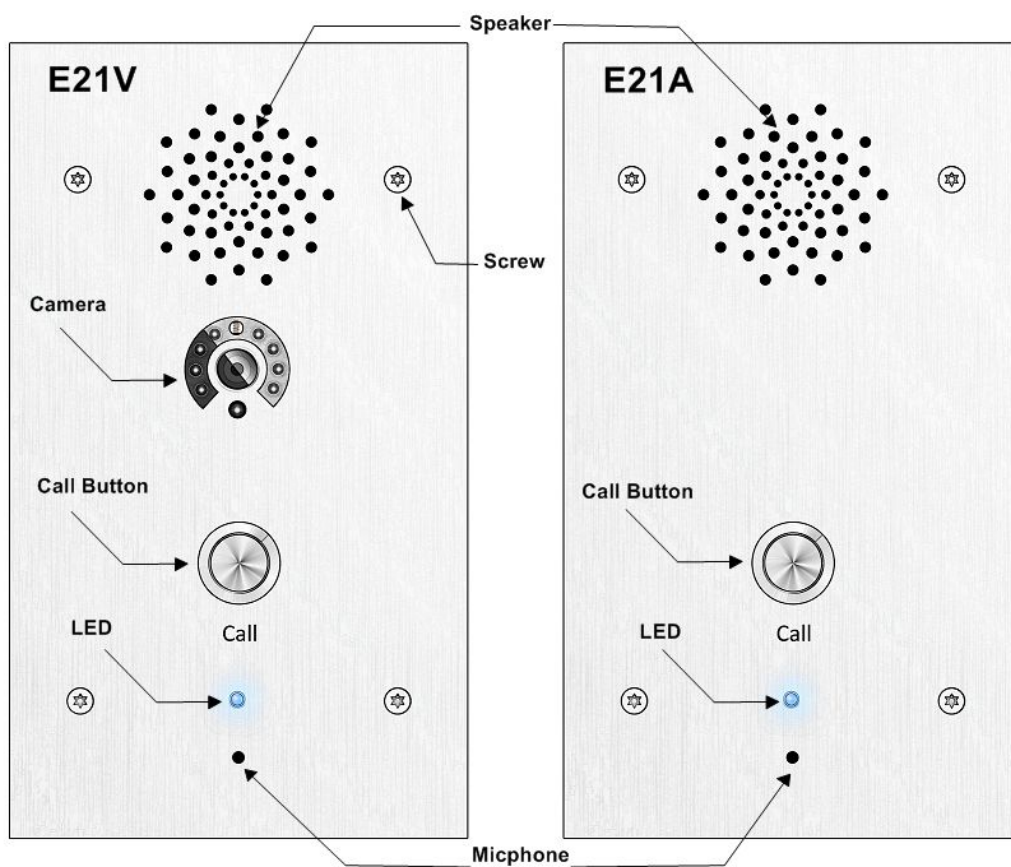
➤ **Network Features**

- 2x10/100Mbps Ethernet Port
- Security: Password Protection, IP address filtering, SIP over TLS, HTTPS encryption, user access log
- Protocols support: IPv4, HTTP, HTTPS, FTP, SNMP, DNS, NTP, RTSP, RTP, TCP, UDP, ICMP, DHCP, ARP

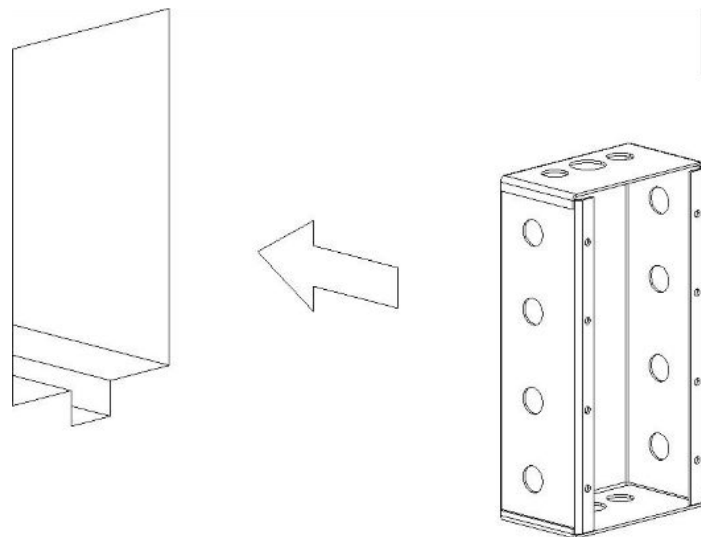
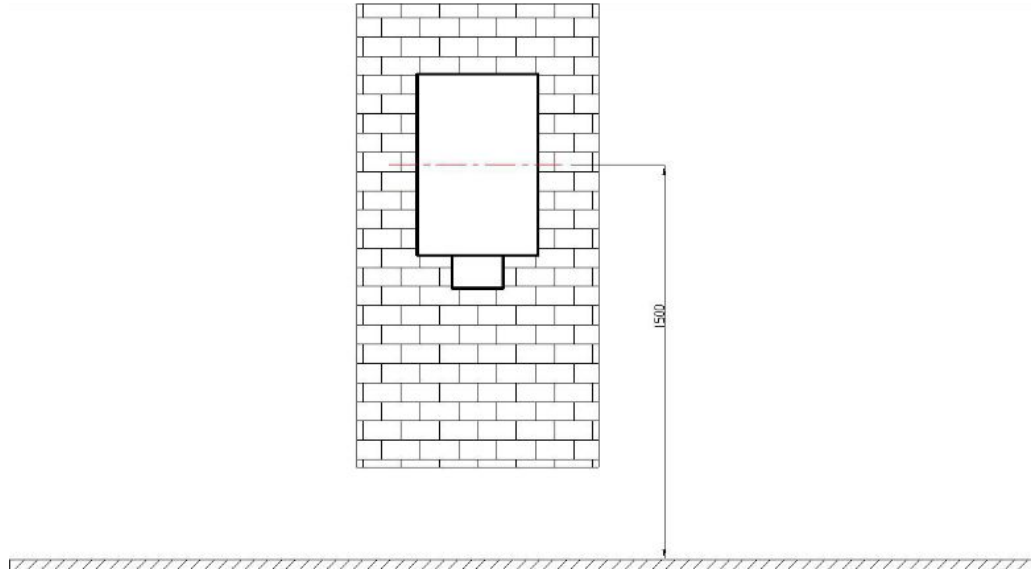
➤ SIP Features

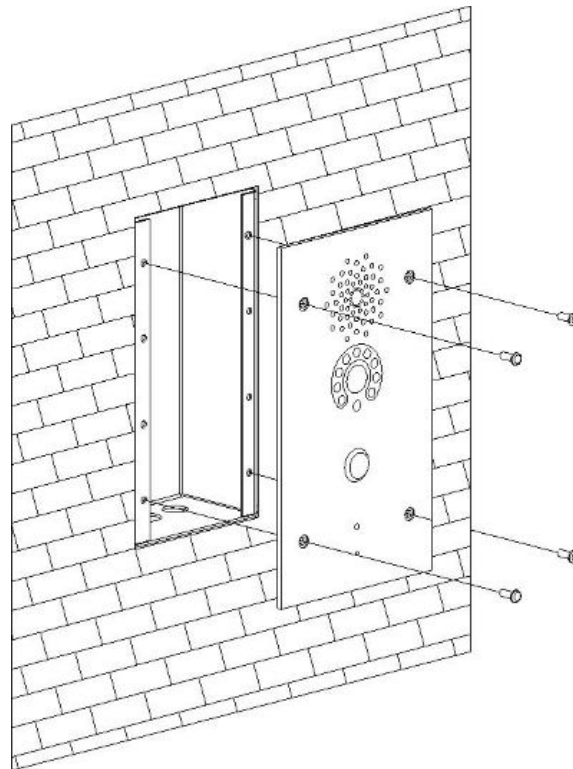
- SIP v1(RFC2543), SIP v2(RFC3261)
- Audio codecs: G.711a, G.711μ, G.722, G.729
- Video codecs: MPEG-4/MJEG (E21V only)
- Speech Quality: 7kHz Audio
- Echo Cancellation
- Voice Activation Detection
- Comfort Noise Generator

3 Panel Description



4 Installation





Installation step:

1. Use coment to fix the back cover in the wall(installation height about 1500mm)
2. Place E21 panel into the back cover.
3. Use screws to fix the panel.

Configuration

1 Web Login

1.1 Obtaining the IP address

The Akuvox E21 uses Static IP by default, and the default IP address is 192.168.1.100.

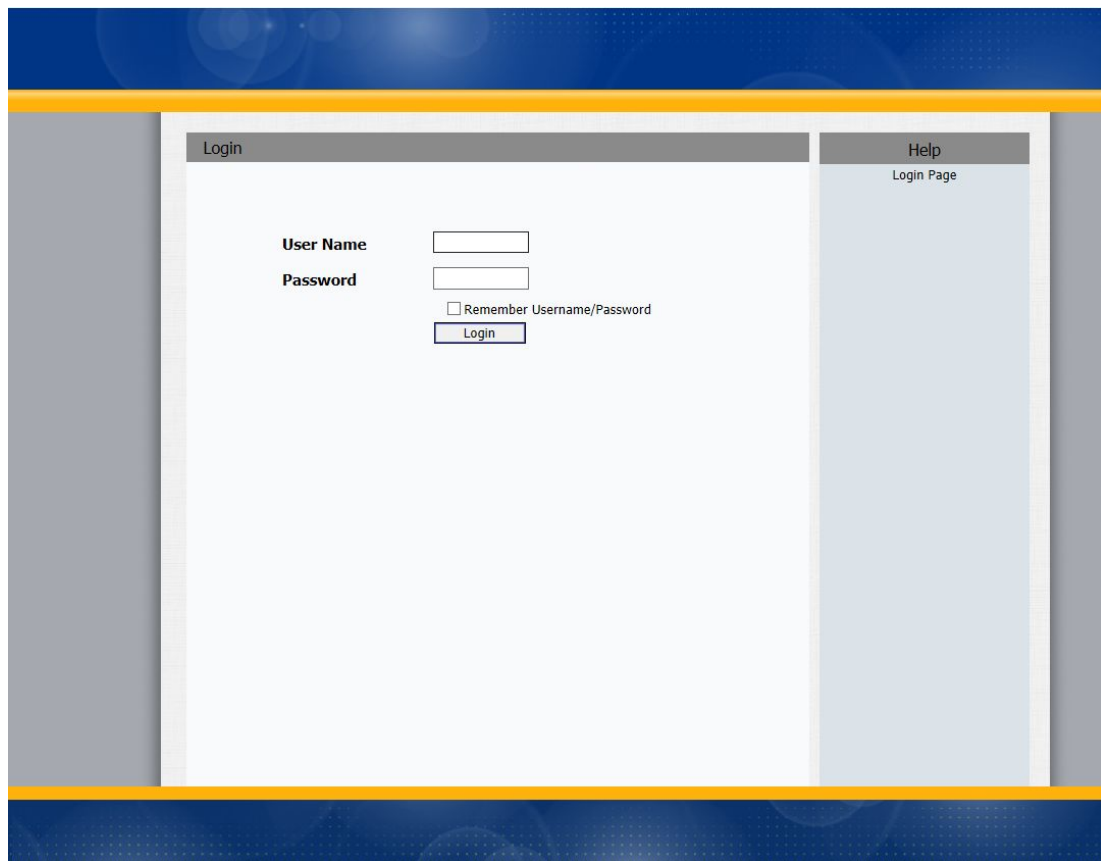
If the IP address is unknown, press the call button when LED light turns blue, after a short period of time (about 5s), the phone will announce its IP.

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



2 Status

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

Status

Product Information

Model	E21P
MAC Address	C4:09:38:D2:A1:B1
Firmware Version	21.0.0.122
Hardware Version	21.0.0.0.0.0.0

Network Information

LAN Port Type	DHCP Auto
LAN Link Status	Connected
LAN IP Address	192.168.35.14
LAN Subnet Mask	255.255.255.0
LAN Gateway	192.168.35.1
LAN DNS1	8.8.8.8
LAN DNS2	

Account Information

Account1	None@None Disabled
Account2	None@None Disabled

Help

Note :
Max length of characters for input box:
255: Broadsoft Phonebook server address
127: Remote Phonebook URL & AUTOP Manual Update Server URL
63: The rest of input boxes

Warning :

Field Description :

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 Intrecom

3.1 Basic

The screenshot displays the 'Intercom-Basic' configuration page. On the left is a sidebar with navigation items: Status, Intercom (Basic, LED Setting, Relay&Input, Live Stream, AEC Setting, RTSP, ONVIF, Multicast), Account, Network, Phone, Upgrade, and Security. The main content area is titled 'Intercom-Basic' and contains several sections:

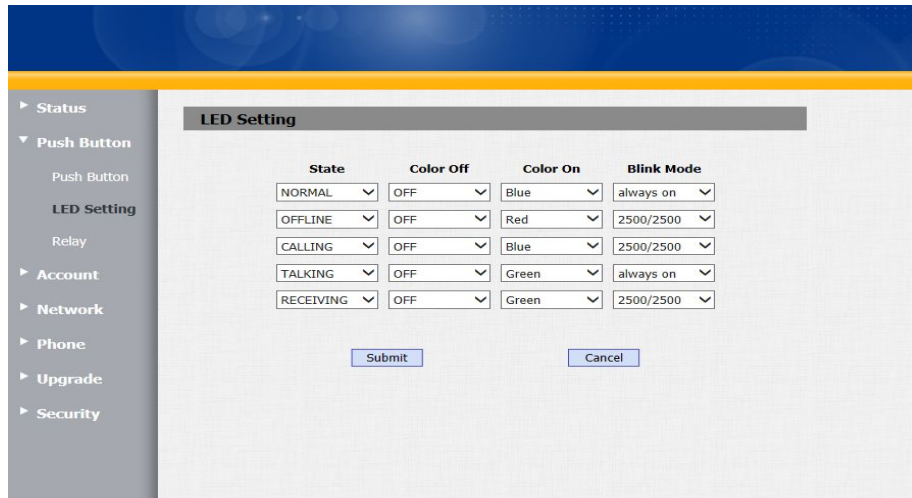
- Account Selection:** A 'Select Account' dropdown menu set to 'Auto'.
- Push Button:** A table with columns 'Key' and 'Number'. Below it is a 'Push Button' input field.
- Web Call:** A 'Web Call(Ready)' input field, an 'Auto' dropdown, and buttons for 'Dial Out' and 'Hang Up'.
- Max Call Time:** A 'Max Call Time' input field set to '5' with a note '(2~30Minutes)'.
- Push To Hang Up:** A 'Push To Hang Up' dropdown menu set to 'Enabled'.

At the bottom of the main content area are 'Submit' and 'Cancel' buttons. On the right side, there is a 'Help' section with a 'Note' (regarding character limits and server addresses), a 'Warning', and a 'Submit Shortcut' with 'Submit' and 'Cancel' buttons.

Sections	Description
Basic	<ul style="list-style-type: none"> Select Account: E21 supports 2 accounts. You can choose one account or Auto mode for the following Intercom basic settings.
Push Button	<ul style="list-style-type: none"> Push Button: To configure the destination number or IP you want to contact with.
Web Call	To dial out or answer the phone from website.
Max Call Time	To configure the max call time
Max Dial Time	<ul style="list-style-type: none"> Dial in Time: When other phone calls to E21, if ring tone is over the Dial in Time without answer. The call will be hang up. Dial out Time: When E21 calls to the other party, if the ringtone is over the Dial out Time without answer. E21 will continue calls to no answer call number in order.
Push to Hang up	To enable or disable the Push to Hang up function

3.2 LED Settings

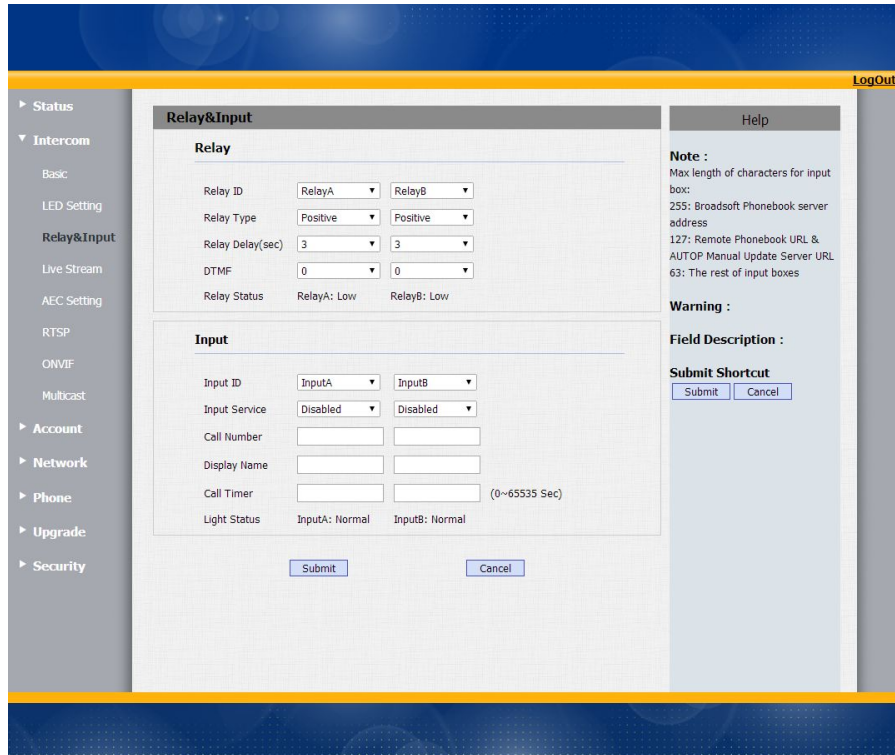
To configure the different LED blink mode of different states.



Sections	Description
States	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.

3.3 Relay&Input

To configure unlock and alarm setting. Go to the path: Push Button-> Relay&Input.



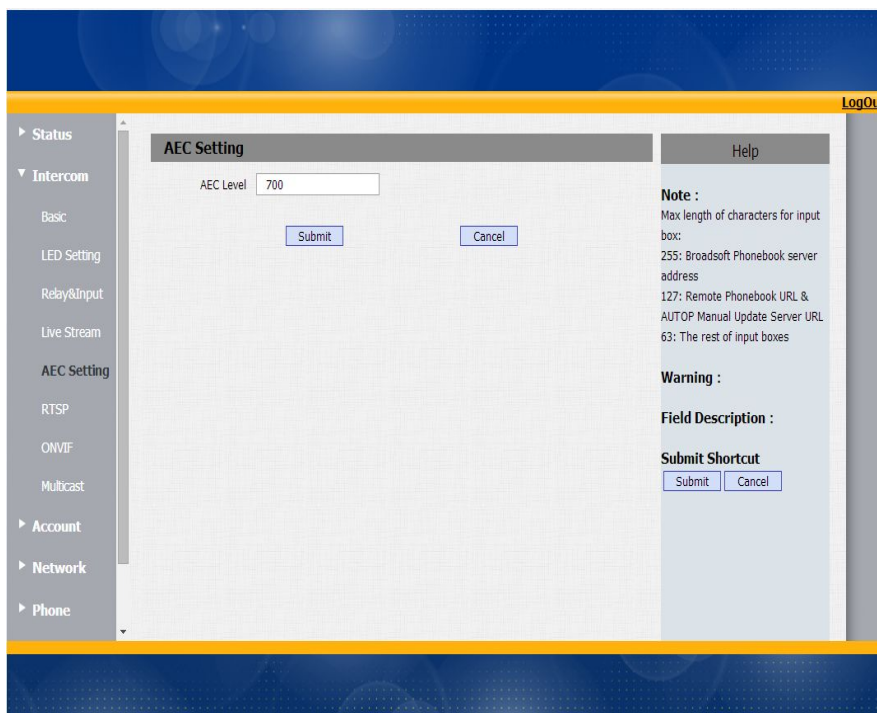
Sections	Description
Relay	<p>To configure some settings about unlock</p> <ul style="list-style-type: none"> ● Relay Select: E21 support 2 relays ● Relay Type: Different locks use different relay types. ● Delay(s): Allows door remain “open” for certain period The range is from 1 to 5 seconds ● DTMF: Setup DTMF code for remote unlock ● Status: Different relay type will show different status.
Input	<p>There is a sensor that used to anti vandal in E21. When E21 is broken by violent means. The sensor will be triggered, then management center will receive the alarm.</p> <ul style="list-style-type: none"> ● Input ID: E21 supports 2 opticalcouplers. Once the opticalcoupler is triggered, it will alarm when this function is enabled. ● Input Service: Disable by default ● Call Number: To setup management center number for alarm. ● Display Name: Which is sent to the other call party for displaying

3.4 Live Stream(Optional)

The screenshot displays a web-based configuration interface. On the left is a vertical sidebar menu with the following items: Status, Intercom (expanded), Basic, LED Setting, Relay&Input, Live Stream (highlighted), AEC Setting, RTSP, ONVIF, Multicast, Account, Network, and Phone. The main content area is split into two panels. The left panel, titled 'Live Stream', shows a real-time video feed of a room with a curved desk and a grid ceiling with several bright lights. The right panel, titled 'Help', contains the following text: **Note :** Max length of characters for input box: 255: Broadsoft Phonebook server address, 127: Remote Phonebook URL & AUTOP Manual Update Server URL, 63: The rest of input boxes. **Warning :** **Field Description :** **Submit Shortcut** with two buttons: 'Submit' and 'Cancel'. A 'LogOut' link is visible in the top right corner of the interface.

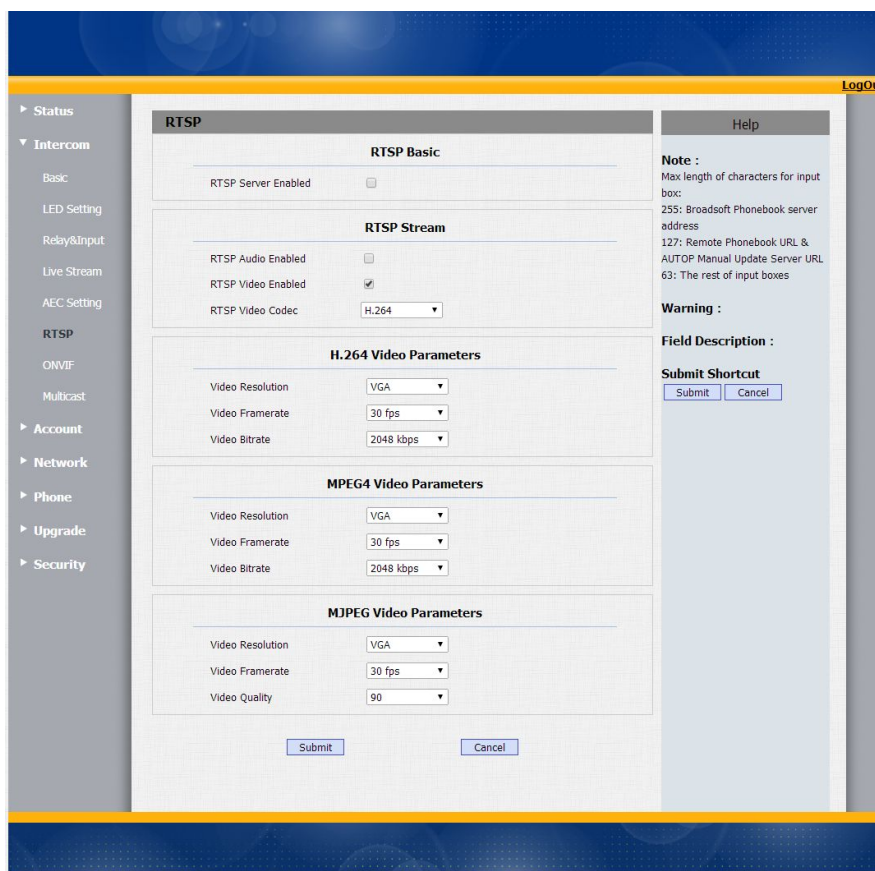
Sections	Description
Live Stream	To check the real-time video from E21V.

3.5 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.

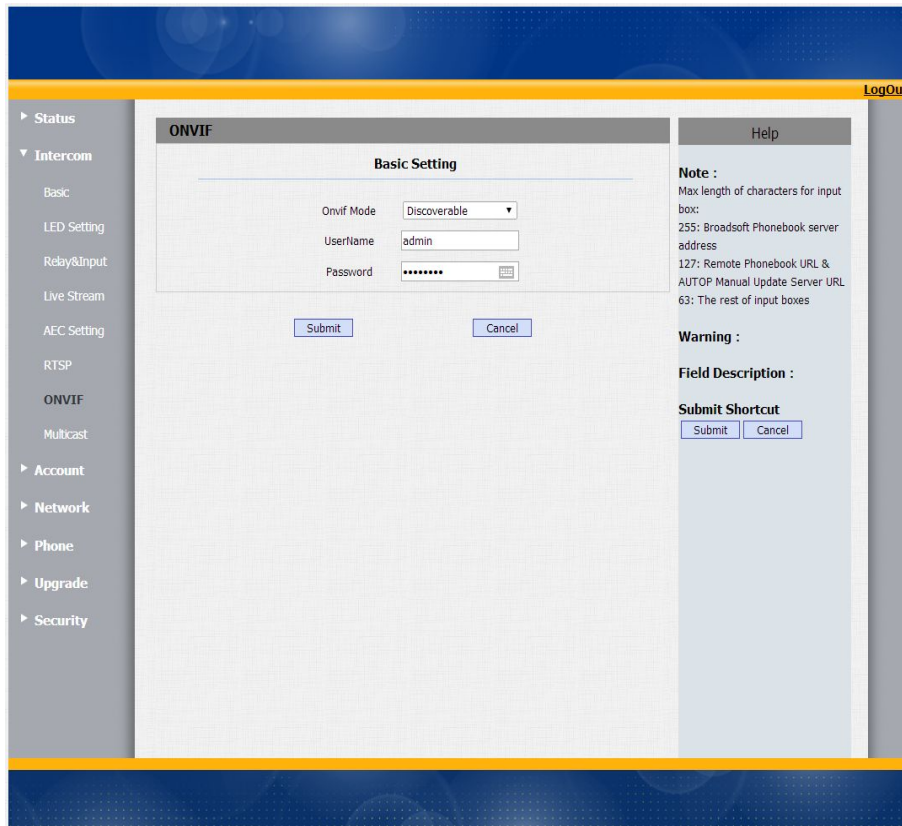
3.6 RTSP(optional)



Sections	Description
RTSP Basic	To active the RTSP function, then E21 can be monitored.
RTSP Stream	To enabled RTSP video and select the video codec. E21 supports H264,H263 video codec. H264 by default.
H.264 Video Parameters	<p>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.</p> <p>To modify the resolution,framerate and bitrate of H264</p>
MPEG4 Video Parameters	<p>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.</p> <p>To modify the resolution,framerate and bitrate of MPEG4</p>
MJPEG Video Parameters	MJPEG: called Motion Joint Photographic Experts Group. It is a video encoding format.in which each image is compressed

	<p>separately by JPEG.MJPEG compression can produce high 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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3.7 Onvif(optional)



Sections	Description
Basic Setting	<p>To setup the Onvif function parameters. It is used to connect with the corresponding Onvif tool.</p> <ul style="list-style-type: none"> ● Onvif Mode: Two modes - Discoverable and Non-discoverable. Discoverable by default. Only Discoverable mode, then Onvif software can search E21. ● User Name: To modify the user name you need. Admin by default. ● Password: To modify the password you want. Admin by default.

3.7 Multicast

The screenshot displays a web-based configuration interface for Multicast settings. On the left is a navigation menu with categories: Status, Intercom (Basic, LED Setting, Relay&Input, Live Stream, AEC Setting, RTSP, ONVIF, Multicast), Account, Network, and Phone. The main content area is titled 'Multicast' and is divided into two sections: 'Multicast Audio Receiving' and 'Multicast Audio Sending'. The 'Multicast Audio Receiving' section includes a checkbox for 'Multicast Receiver Enabled', a text input field for 'Receive Address', and another text input field for 'Receive Port'. The 'Multicast Audio Sending' section includes a checkbox for 'Multicast Sending Enabled', a text input field for 'Send to Address', and another text input field for 'Send to Port'. At the bottom of the main area are 'Submit' and 'Cancel' buttons. On the right side, there is a 'Help' section containing a 'Note' (regarding character limits and server addresses), a 'Warning', and a 'Field Description'. Below the help section are 'Submit Shortcut' buttons for 'Submit' and 'Cancel'.

Sections	Description
Multicast Audio Receiving	<p>To display and configure the Multicast setting.</p> <ul style="list-style-type: none"> ● Multicast Receiver Enable: Enable receiver multicast function. ● Receiver address : Setup the multicast address. ● Receiver port : setup the multicast address port.
Multicast Audio Sending	<p>To setup the multicast parameters.</p> <ul style="list-style-type: none"> ● Multicast Sending Enable: Enable sender multicast function ● Send to Address: setup the multicast address. ● Send to port: setup the multicast address port.

4 Account

4.1 Account->Basic

To configure sip account, go to the path: Account->Basic

The screenshot shows the 'Account-Basic' configuration page. The main content area is divided into several sections:

- SIP Account:** Includes fields for Status (Registered), Account (Account 1), Account Active (Enabled), Display Label (2003), Display Name (2003), Register Name (2003), User Name (2003), and Password (masked).
- SIP Server 1:** Includes Server IP (192.168.35.230), Port (5060), and Registration Period (1800).
- SIP Server 2:** Includes Server IP and Port (5060), and Registration Period (1800).
- Outbound Proxy Server:** Includes Enable Outbound (Disabled), Server IP, Port (5060), Backup Server IP, and Port (5060).
- Transport Type:** Includes Transport Type (UDP).
- NAT:** Includes NAT (Disabled) and Stun Server Address, Port (3478).

The right sidebar contains a 'Help' section with a 'Note' (Max length of characters for input box: 255: Broadsoft Phonebook server address; 127: Remote Phonebook URL & AUTOP Manual Update Server URL; 63: The rest of input boxes), a 'Warning', and a 'Field Description'. Below this is a 'Submit Shortcut' section with 'Submit' and 'Cancel' buttons. The main form also has 'Submit' and 'Cancel' buttons at the bottom.

Sections	Description
SIP Account	<p>To display and configure the specific Account settings.</p> <ul style="list-style-type: none"> ● Status: To display register result. ● Display Name: Which is sent to the other call party for displaying. ● Register Name: Allocated by SIP server provider, used for authentication. ● User Name: Allocated by your SIP server provide, used for authentication.

	<ul style="list-style-type: none"> ● Password: Used for authorization.
SIP Server 1	<p>To display and configure Primary SIP server settings.</p> <ul style="list-style-type: none"> ● 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	<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>Note: Secondary SIP server is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.</p>
Outbound Proxy Server	<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>Note: If configured, all SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.</p>
Transport Type	<p>To display and configure Transport type for SIP message</p> <ul style="list-style-type: none"> ● 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	<p>To display and configure NAT(Net Address Translator) settings.</p> <ul style="list-style-type: none"> ● STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues. <p>Note: By default, NAT is disabled.</p>

4.2 Account-> Advanced

For advance account settings, go to the path: Account -> Advanced.

- ▶ Status
- ▶ Intercom
- ▼ Account
 - Basic
 - Advanced**
- ▶ Network
- ▶ Phone
- ▶ Upgrade
- ▶ Security

Account-Advanced

SIP Account

Account

Codecs

Disabled Codecs	Enabled Codecs
	PCMU PCMA G722 G729
<input type="button" value=""/> >>	<input type="button" value="↑"/>
<input type="button" value=""/> <<	<input type="button" value="↓"/>

Video Codec

Codec Name H264
 Codec Resolution
 Codec Bitrate
 Codec Payload

Subscribe

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

DTMF

Type
 How To Notify DTMF
 DTMF Payload (96~127)

Call

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

Session Timer

Active
 Session Expire (90~7200s)
 Session Refresher

BLFList

BLFList URI
 BLFList Pickup Code
 BLFList BargeIn Code

Encryption

Voice Encryption(SRTP)

NAT

UDP Keep Alive Messages
 UDP Alive Msg Interval (5~60s)
 RPort

User Agent

User Agent

Help

Note :
 Max length of characters for input box:
 255: Broadsoft Phonebook server address
 127: Remote Phonebook URL & AUTOP Manual Update Server URL
 63: The rest of input boxes

Warning :

Field Description :

Submit Shortcut

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 (wid-bandth codecs), G729 and so on.
Video Codec(optional)	To configure the video quality <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● 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. <p>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.</p>
Call	To display and configure call-related features. <ul style="list-style-type: none"> ● 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

	<p>auto-answered when there is an incoming call for designated account.</p> <ul style="list-style-type: none"> ● 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.
<p>Session Timer</p>	<p>To display or configure session timer settings.</p> <ul style="list-style-type: none"> ● Active: To enable or disable this feature, If enable, the ongoing 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. <p>Note: UAC means User Agent Client, here stands for IP phone. UAS means User Agent Server, here stands for SIP server.</p>
<p>BLF List</p>	<p>To display or configure BLF List URI address.</p> <ul style="list-style-type: none"> ● BLF List URI: BLF List is short for Busy Lamp Field List. ● BLFList Pickup Code: To set the BLF pick up code. ● BLFList Bargain Code : To set the BLF barge in code.
<p>Encryption</p>	<p>To enable or disabled SRTP feature.</p> <ul style="list-style-type: none"> ● Voice Encryption(SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.
<p>NAT</p>	<p>To display NAT-related settings.</p> <ul style="list-style-type: none"> ● 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.

	<ul style="list-style-type: none"> ● 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

5 Network

5.1 Network-> Basic

To configure the basic network settings, Go to the path: Network -> Basic.

The static IP is set as default, and its IP address is 192.168.1.100.

The screenshot displays the 'Network-Basic' configuration page. The 'LAN Port' section is active, showing two options: 'DHCP' (unselected) and 'Static IP' (selected). Under 'Static IP', there are five input fields: 'IP Address' (192.168.35.119), 'Subnet Mask' (255.255.255.0), 'Default Gateway' (192.168.35.1), 'LAN DNS1' (8.8.8.8), and 'LAN DNS2' (empty). 'Submit' and 'Cancel' buttons are located below the fields. On the right, a 'Help' sidebar contains a 'Note' about character limits, a 'Warning' about field descriptions, and 'Submit Shortcut' buttons for 'Submit' and 'Cancel'.

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

	<ul style="list-style-type: none"> ● Static IP: If selected, you have to set IP address, Subnet Mask, Default Gateway and DNS server manually.
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5.2 Network-> Advanced

For advanced settings, go to the path: Network -> Advanced.

The screenshot shows the 'Network-Advanced' configuration page. It features a sidebar on the left with navigation options: Status, Push Button, Account, Network (Basic, Advanced), Phone, Upgrade, and Security. The main content area is titled 'Network-Advanced' and is divided into three sections: 'Local RTP', 'SNMP', and 'TR069'. Each section contains configuration fields for various parameters. The 'Local RTP' section includes 'Max RTP Port' (12000) and 'Starting RTP Port' (11800). The 'SNMP' section includes 'Active' (Disabled), 'Port', and 'Trusted IP'. The 'TR069' section includes 'Active' (Disabled), 'Version' (1.0), 'ACS' (URL, User Name, Password), 'Periodic Inform' (Active, Periodic Interval), and 'CPE' (URL, User Name, Password). A right sidebar contains a 'Help' section with a 'Note' and a 'Warning', and a 'Submit Shortcut' section with 'Submit' and 'Cancel' buttons. A 'LogOut' link is visible in the top right corner.

Sections	Description
Local RTP	To display and configure Local RTP settings. <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● Active: To enable or disable SNMP feature.

	<ul style="list-style-type: none"> ● 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. <p>Note: SNMP (Simple Network Management Protocols) is Internet-standard protocol for managing devices on IP networks.</p>
TR069	<p>To display and configure TR069 settings.</p> <ul style="list-style-type: none"> ● 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. <p>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.</p>

6 Phone

6.1 Time/Language

Go to the path: Phone-> Time/Language

The screenshot shows a web interface for configuring NTP settings. The page is titled "Time/Lang" and "NTP". It features a sidebar on the left with navigation options: Status, Intercom, Account, Network, Phone (expanded), Time/Lang (selected), Call Feature, Voice, Multicast, Upgrade, and Security. The main content area contains the following fields:

- Time Zone: 0 GMT (dropdown menu)
- Primary Server: 0.pool.ntp.org (text input)
- Secondary Server: 1.pool.ntp.org (text input)
- Update Interval: 3600 (text input, with a note ">= 3600s")

There are "Submit" and "Cancel" buttons at the bottom of the form. To the right of the form is a "Help" section with the following text:

Note :
Max length of characters for input box:
255: Broadsoft Phonebook server address
127: Remote Phonebook URL & AUTOP Manual Update Server URL
63: The rest of input boxes

Warning :

Field Description :

Submit Shortcut
Submit Cancel

Sections	Description
NTP	<p>To configure NTP server related settings.</p> <ul style="list-style-type: none">● 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. <p>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.</p>

6.2 Call Feature

Go to the path: Phone->Call Feature

Sections	Description
Mode	<p>To enable or disable feature key sync.</p> <ul style="list-style-type: none"> ● Feature Key Sync: To enable or disable feature key sync. ● Mode: Select the desired mode.
DND	<p>DND (Do Not Disturb) allows IP phones to ignore any incoming calls.</p> <ul style="list-style-type: none"> ● 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	<p>Intercom allows user to establish a call directly with the callee.</p> <ul style="list-style-type: none"> ● Active: To enable or disable Intercom feature. ● Intercom Mute: If enabled, once the call established, the callee will be muted.
Others	<ul style="list-style-type: none"> ● 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.

6.3 Voice

Go to the path: Phone->Voice

Sections	Description
Mic Volume	To configure Microphone volume
Speaker Volume	To configure Speaker Volume

6.4 Multicast

Multicast

Multicast Setting

Paging Barge: Disabled

Paging Priority Active: Enabled

Priority List

IP Address	Listening Address	Label	Priority
1 IP Address			1
2 IP Address			2
3 IP Address			3
4 IP Address			4
5 IP Address			5
6 IP Address			6
7 IP Address			7
8 IP Address			8
9 IP Address			9
10 IP Address			10

Help

Note :
Max length of characters for input box:
255: Broadsoft Phonebook server address
127: Remote Phonebook URL & AUTOP Manual Update Server URL
63: The rest of input boxes

Warning :

Field Description :

Submit Shortcut
Submit Cancel

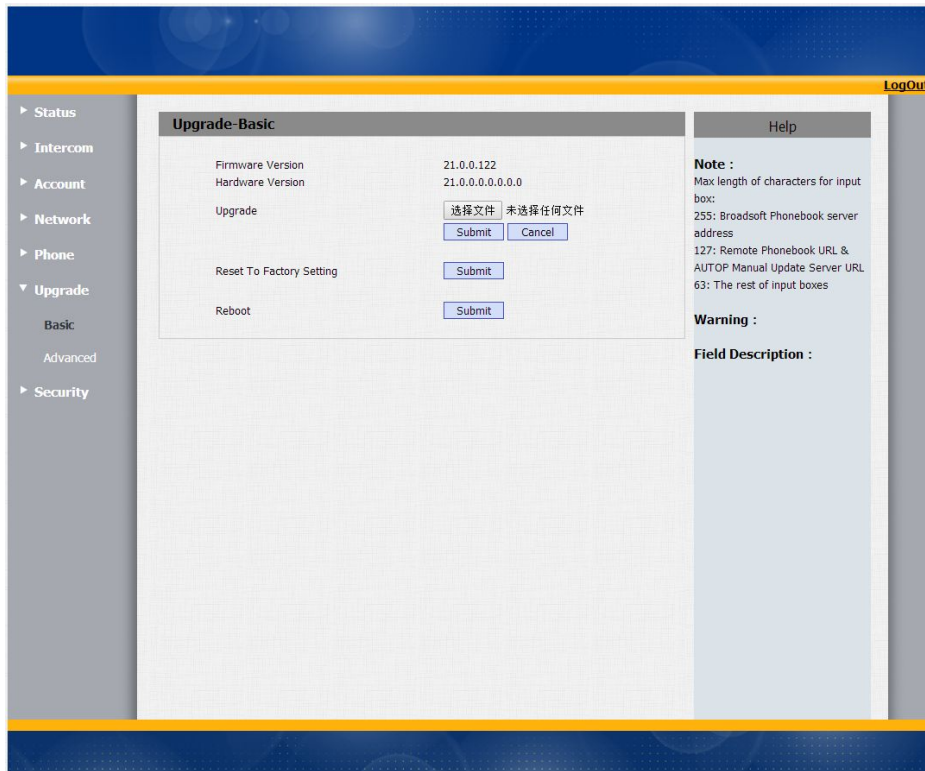
Submit Cancel

Sections	Description
Multicast Setting	To display and configure the Multicast setting. <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● Listening Address: Enter the IP address you need to listen ● Label: Input the label for each listening address

7 Upgrade

7.1 Basic Upgrade

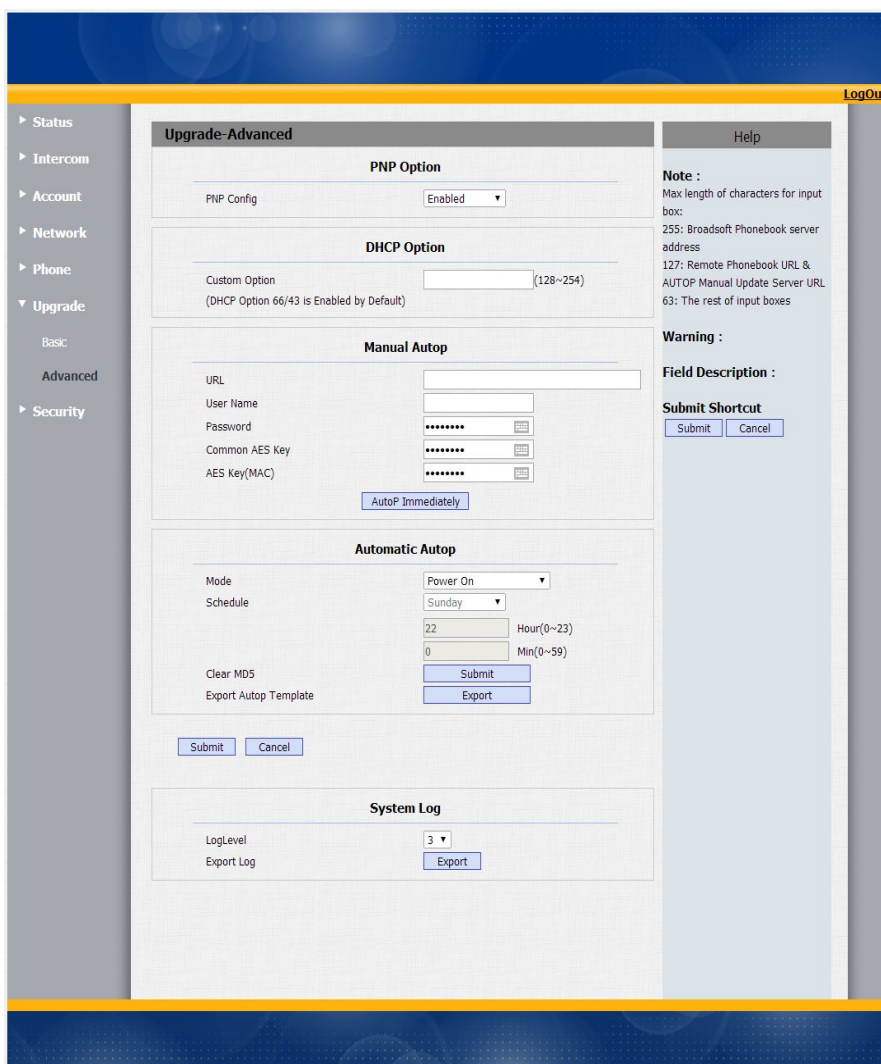
To upgrade your device, go to the path: Upgrade > Basic.



Sections	Description
Upgrade	To select upgrading rom 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.

7.2 Advanced upgrade

To do the advanced upgrade for your device, go to the path: Upgrade -> Advanced.



Sections	Description
PNP Option	<p>To display and configure PNP setting for Auto Provisioning.</p> <ul style="list-style-type: none"> ● 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. <p>By default, this SIP message is sent to multicast address 224.0.1.75(PNP server address by standard).</p>
Manual Autop	<p>To display and configure manual update server's settings.</p> <ul style="list-style-type: none"> ● 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

	<p>example, file name could be 0c1105888888.cfg if IP phone's MAC address is 0c1105888888).</p> <p>Note: AES is one of many encryption, it should be configure only configure filed is ciphered with AES, otherwise left blank.</p>
Automatic AutoP	<p>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 Provisioning every time it powers on.</p>
System Log	<p>To display system log level and export system log file.</p> <ul style="list-style-type: none"> ● 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.

8 Security

To modify web password, go to the path: Security-> Basic

The screenshot shows a web interface for modifying a password. On the left is a navigation menu with 'Security' expanded to 'Basic'. The main content area is titled 'Security-Basic' and contains a 'Web Password Modify' form. The form has a 'User Name' dropdown set to 'admin', and three password input fields: 'Current Password', 'New Password', and 'Confirm Password'. Below the form are 'Submit' and 'Cancel' buttons. On the right side, there is a 'Help' section with a 'Note' (Max length of characters for input box: 255: Broadsoft Phonebook server address, 127: Remote Phonebook URL & AUTOP Manual Update Server URL, 63: The rest of input boxes), a 'Warning', and a 'Field Description' section with its own 'Submit' and 'Cancel' buttons. A 'LogOut' link is in the top right corner.

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

- Confirm Password: Repeat the new password.

Note: For now, IP phone can only support user admin.