E10S Secondary Entry Phone

Content

1 Production Overview

1.1 Production Description

E10S is a smart SIP-based secondary entry phone. It can be connected with Akuvox indoor phone for unlock and monitor. It is more convenient and safe for residents to check the visitor identity through E10S. E10S is often applicable in villas , apartments.

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.

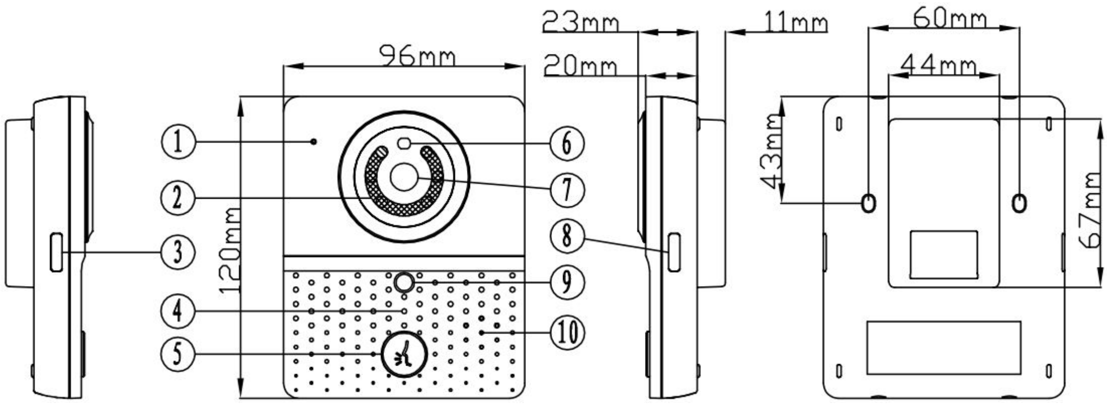
—Consult the dealer or an experienced radio/TV technician for help.

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.

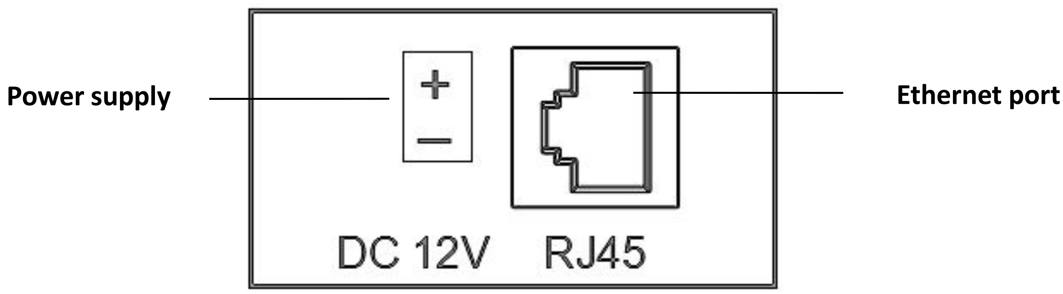
For baseband, this equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment .This equipment should be installed and operated with minimum distance 20cm between the radiator& your body.

1.2 Dimension



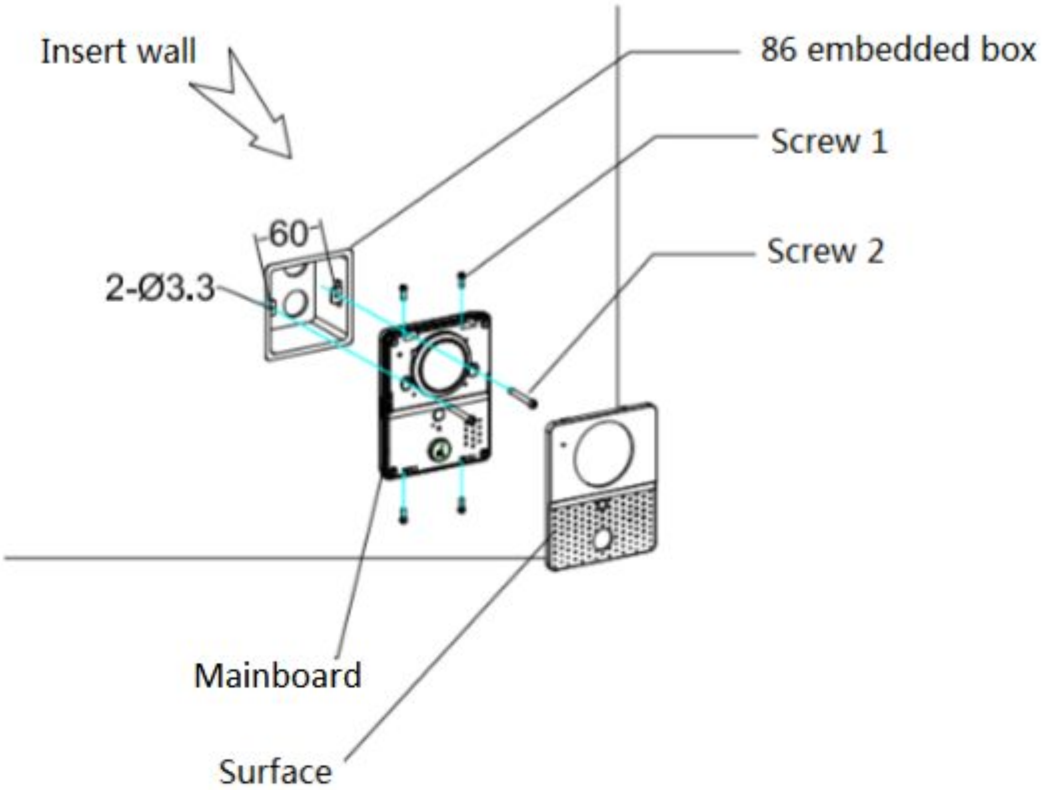
① Mic
② LED
③ Key Light 1
④ Indicator Light
⑤ Button
⑥ Preformed hole
⑦ Camera
⑧ Key Light 2
⑨ Infrared sensor
⑩ Loudspeaker

1.3 Connector



Note: Akuvox IT81 indoor phone can also provide power for E10S through Ethernet port.

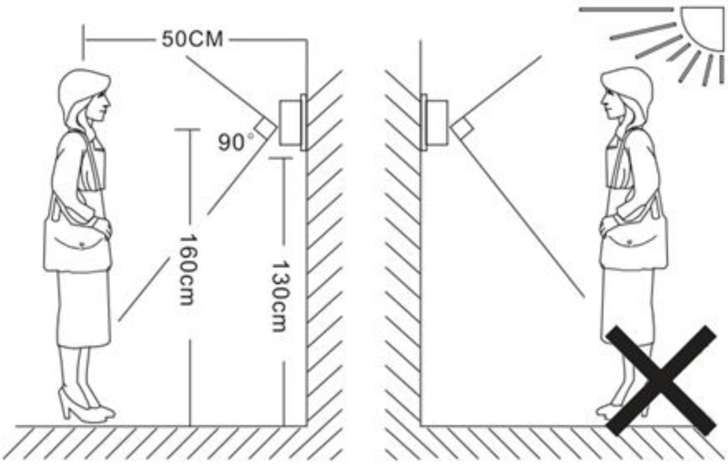
1.4 Installation



1. Use screws 2 to fix the mainboard into the 86 embedded box of the wall.(the screw of length: 10mm，diameter：4mm)
2. Use screw 1(M2.5X5) Lock the surface at the mainboard.

Warning :

1. To protect the product from crashing, knocking or shaking.
2. Please don't place the E10S under the sun, high temperature, snow or chemical corrosion or dust.
3. Please install the E10S in a good visual level. (about 160cm)
4. Please cut down the power if you find the product is not working properly.
5. If the stairway phone is broken, you should cut down the power immediately and check for fault. Otherwise call the customer service manager to help.
6. Please protect the IC card from water and broken, antimagnetic.


50CM
90°
160cm
130cm

2 Basic Function

2.1 Make a call

Once you setup push button number(please refer to chapter Intercom- Basic) , press it to call the indoor phone. After the call is answered, the LED and Key light will turn up. Resident can pickup the call in video or audio mode .

2.2 Monitor

Users can press Monitor button in indoor phone to get the live video from E10S any time.

3 Configuration

3.1 Web login

3.1.1 Obtaining IP address

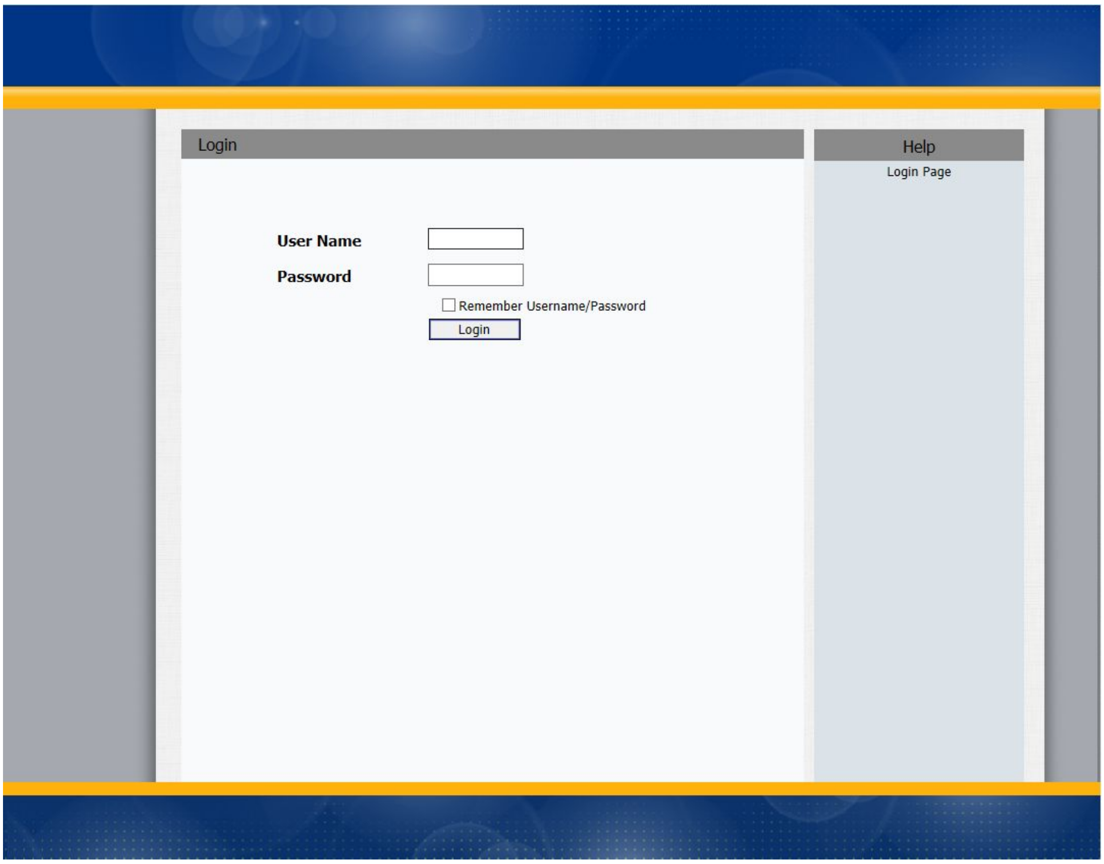
Hold the call button about 5s, the phone will announce its IP. Press again to stop.

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:

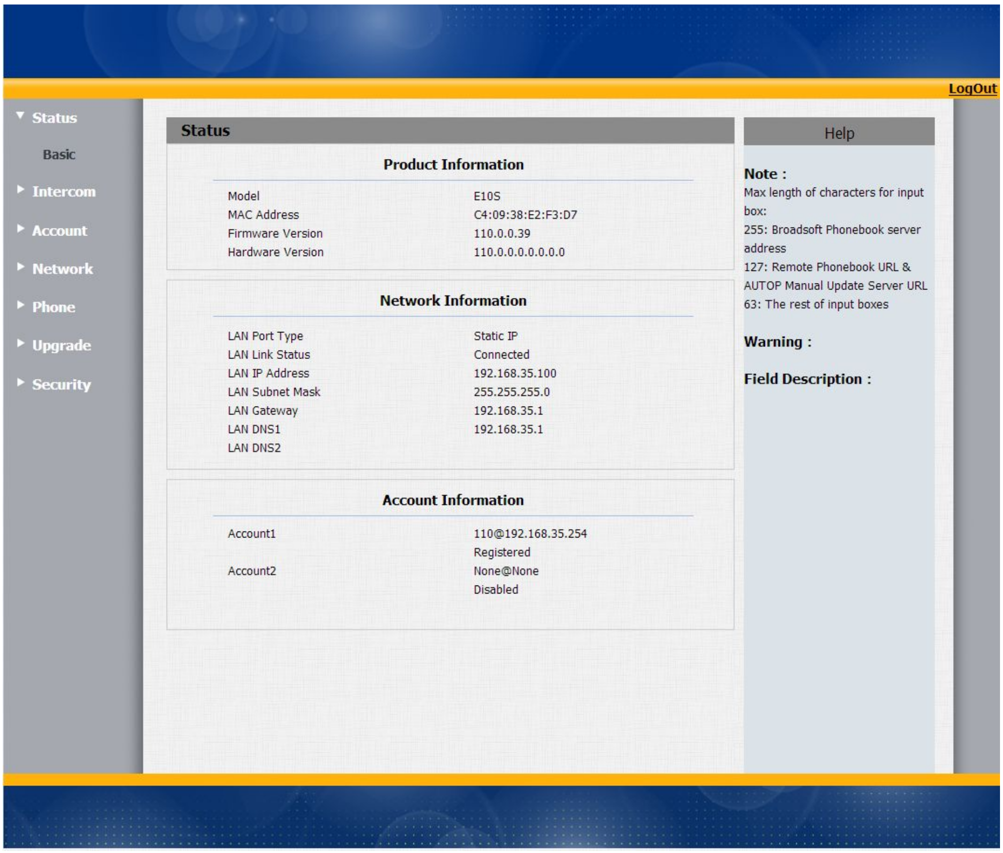
User name: admin

Password: admin



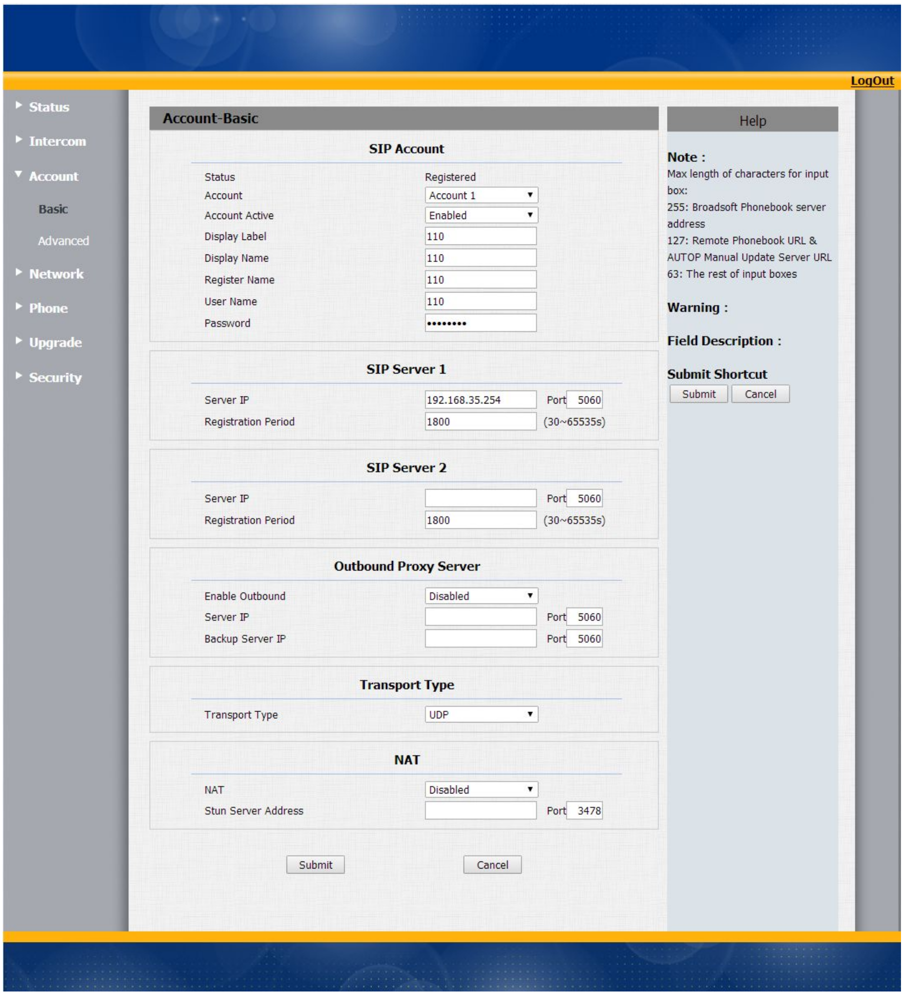
3.2 Status-Basic

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).

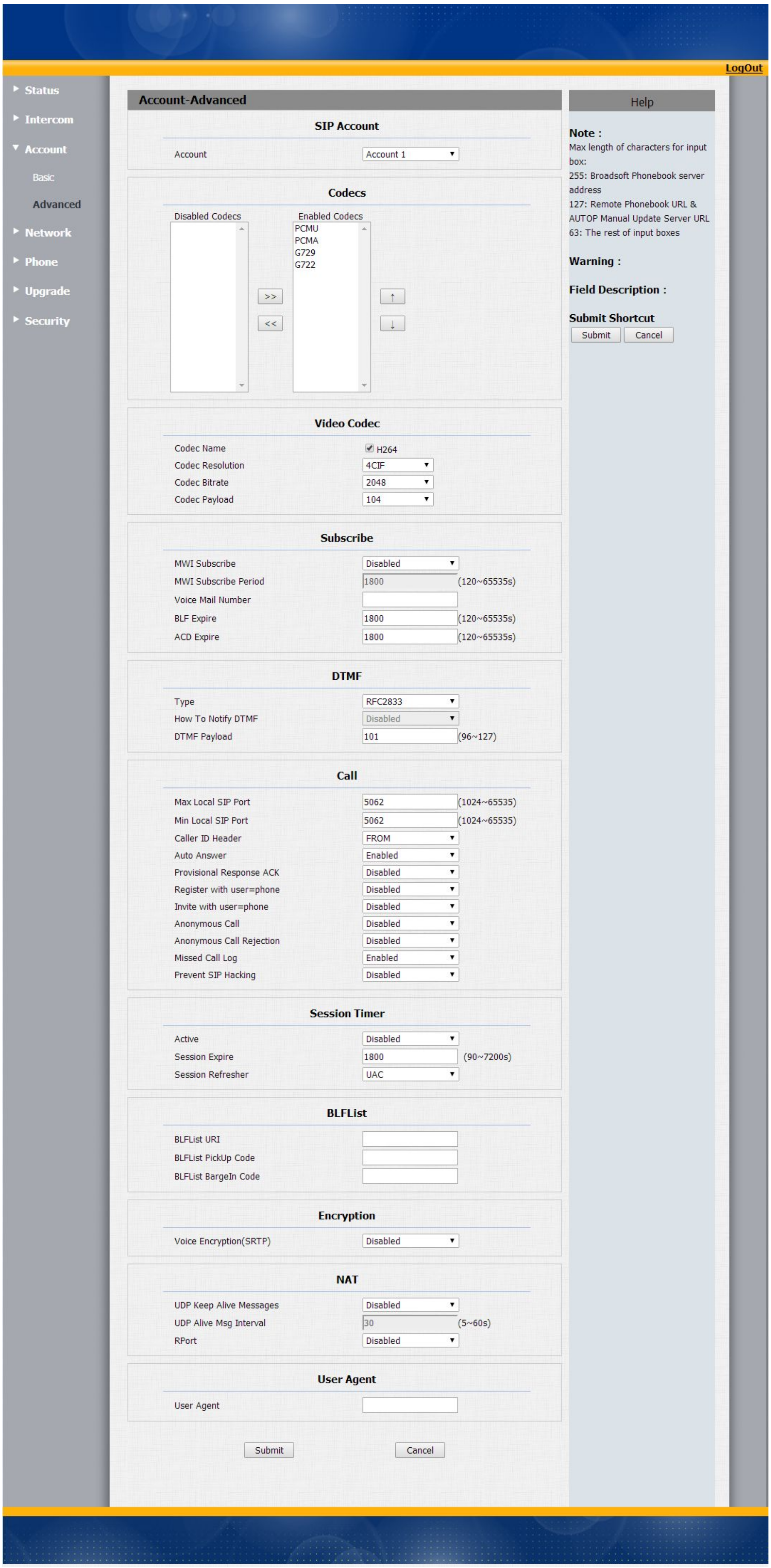
3.3 Intercom- Basic



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.

3.4 Intercom-Advanced

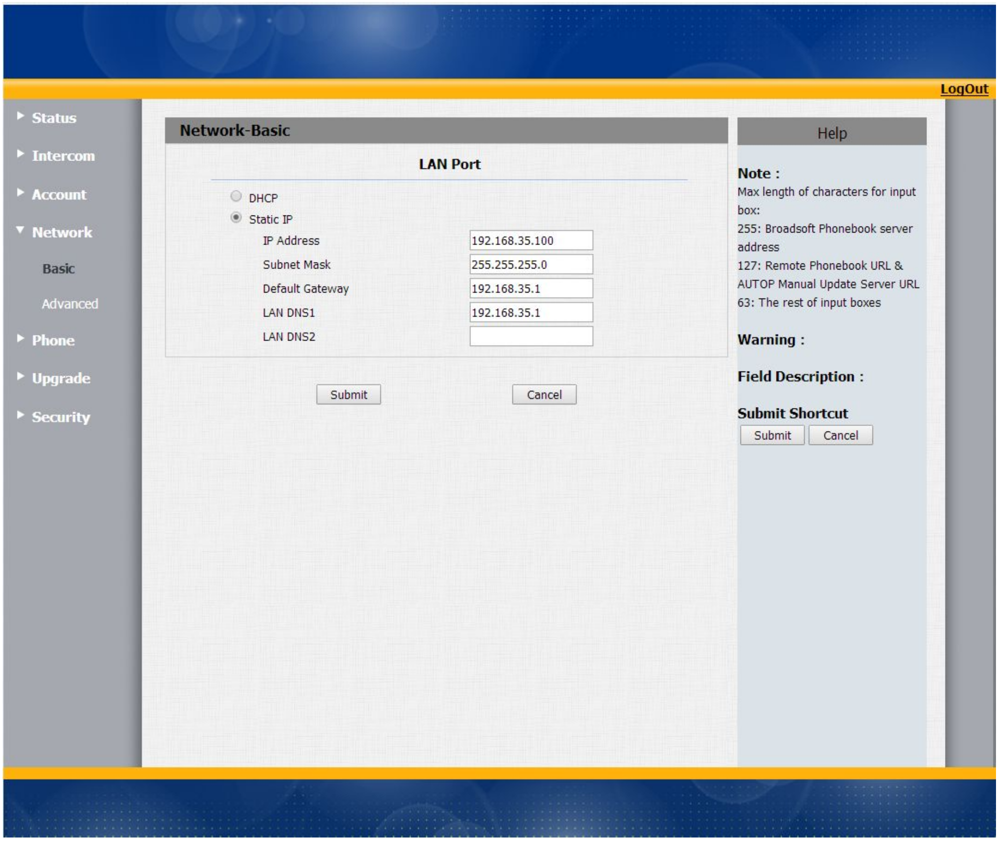

LogOut
Status
Intercom
Account
Basic
Advanced
Network
Phone
Upgrade
Security
Account-Advanced
SIP Account
Account
Account 1
Codecs
Disabled Codecs
Enabled Codecs
PCMU
PCMA
G729
G722
>>
<<
Video Codec
Codec Name
H264
Codec Resolution
4CIF
Codec Bitrate
2048
Codec Payload
104
Subscribe
MWI Subscribe
Disabled
MWI Subscribe Period
1800
(120~65535s)
Voice Mail Number
BLF Expire
1800
(120~65535s)
ACD Expire
1800
(120~65535s)
DTMF
Type
RFC2833
How To Notify DTMF
Disabled
DTMF Payload
101
(96~127)
Call
Max Local SIP Port
5062
(1024~65535)
Min Local SIP Port
5062
(1024~65535)
Caller ID Header
FROM
Auto Answer
Enabled
Provisional Response ACK
Disabled
Register with user=phone
Disabled
Invite with user=phone
Disabled
Anonymous Call
Disabled
Anonymous Call Rejection
Disabled
Missed Call Log
Enabled
Prevent SIP Hacking
Disabled
Session Timer
Active
Disabled
Session Expire
1800
(90~7200s)
Session Refresher
UAC
BLFList
BLFList URI
BLFList PickUp Code
BLFList BargeIn Code
Encryption
Voice Encryption(SRTP)
Disabled
NAT
UDP Keep Alive Messages
Disabled
UDP Alive Msg Interval
30
(5~60s)
RPort
Disabled
User Agent
User Agent
Submit
Cancel
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

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

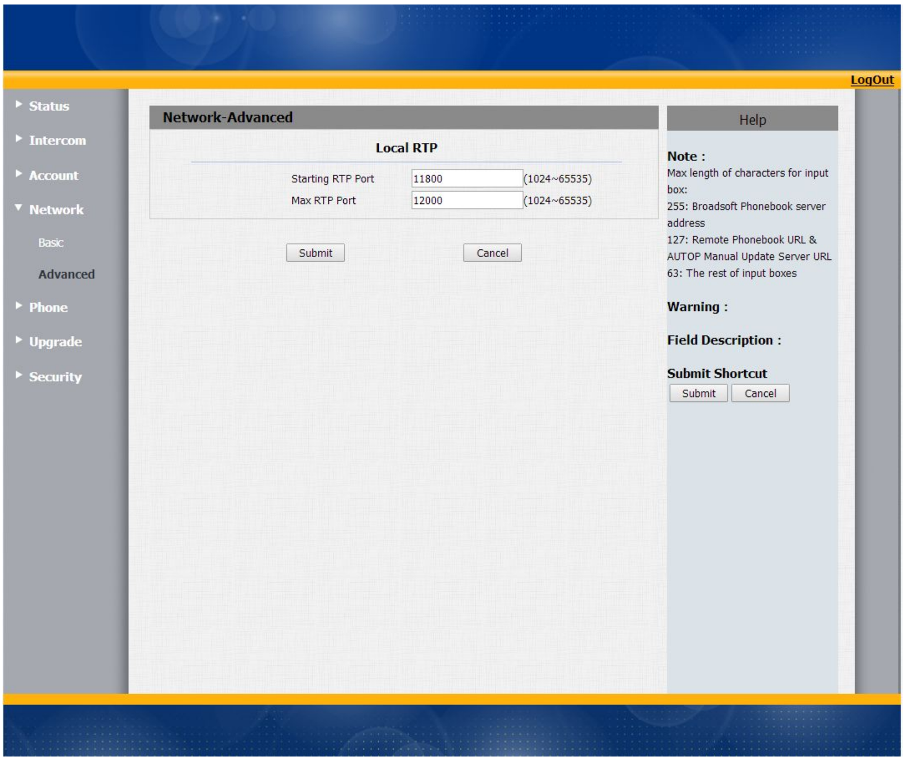
	into outgoing SIP message for designated account.
User Agent	One can customize User Agent field in the SIP message; If user agent is set to specific value, user could see the information from PCAP. If user agent is not set by default, user could see the company name, model number and firmware version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

3.5 Network-Basic



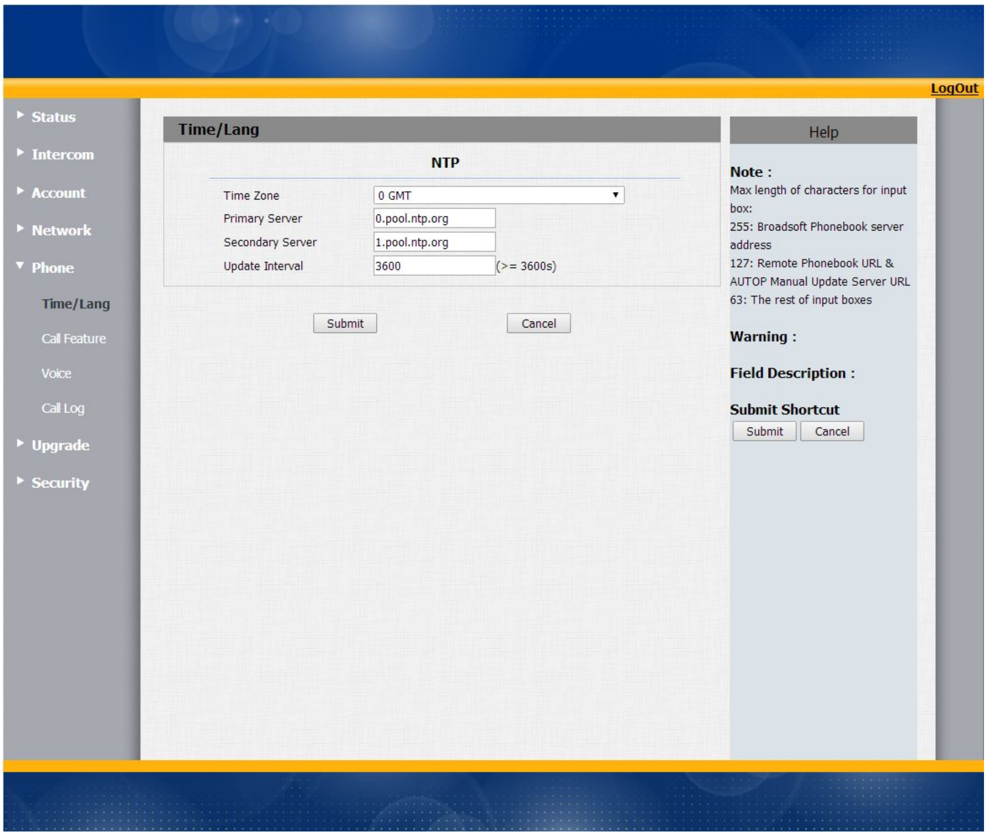
Sections	Description
LAN Port	To display and configure LAN Port settings. ● DHCP: If selected, IP phone will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically. ● Static IP: If selected, you have to set IP address, Subnet Mask, Default Gateway and DNS server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

3.6 Network-Advanced



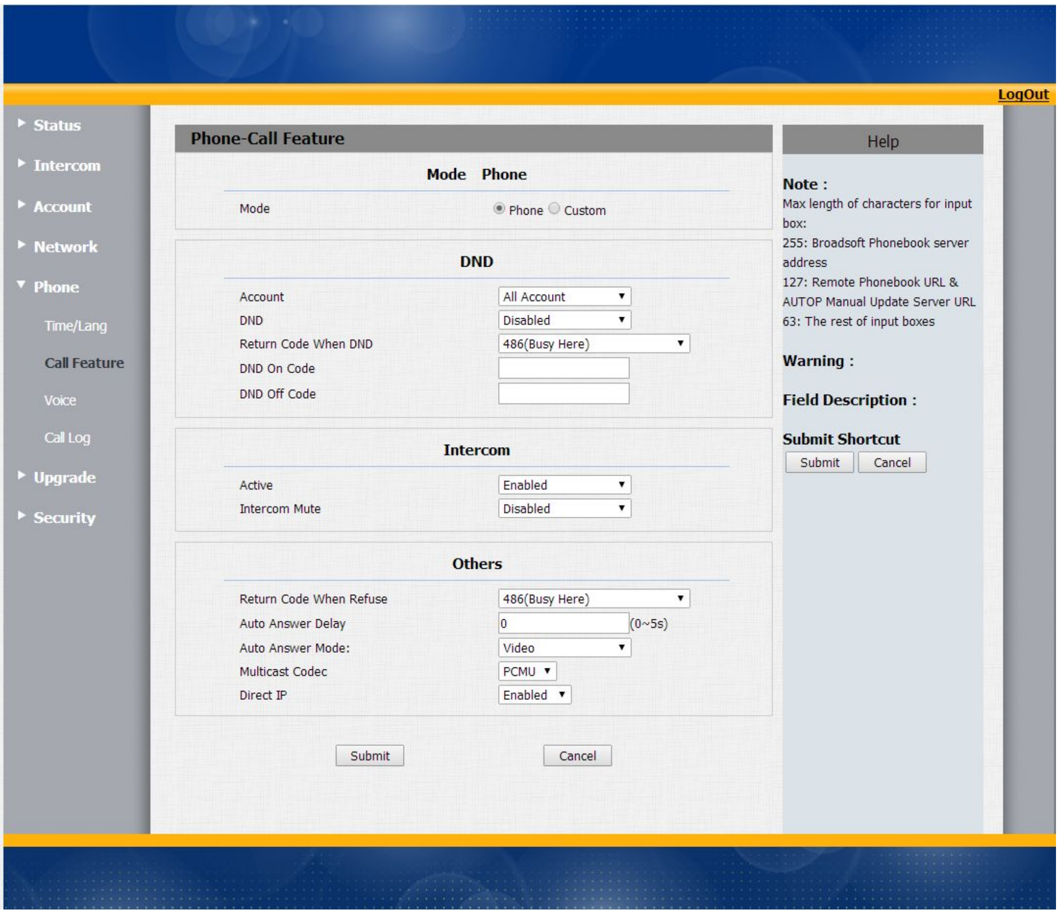
Sections	Description
Local RTP	To display and configure Local RTP settings. ● Max RTP Port: Determine the maximum port that RTP stream can use. ● Starting RTP Port: Determine the minimum port that RTP stream can use.

3.7 Phone-Time/Lang



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.

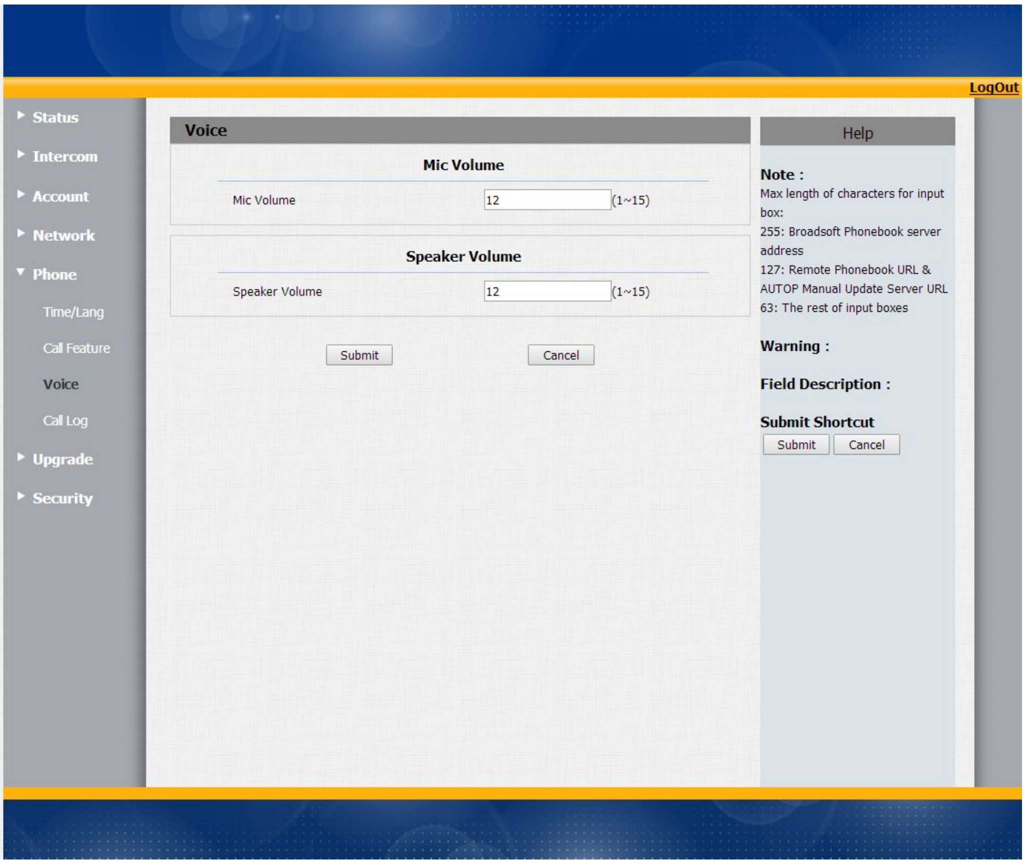
3.8 Phone-Call Feature



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

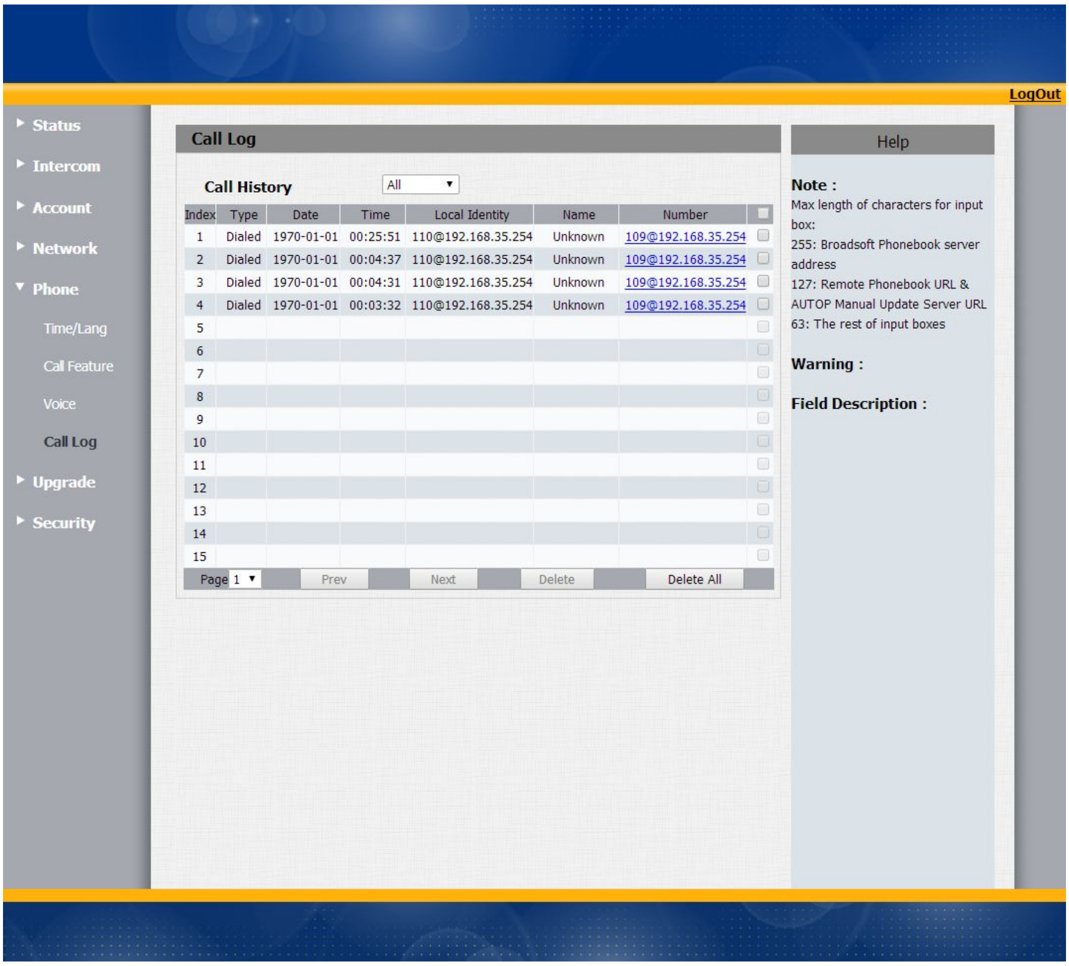
	callee will be muted.
Others	● Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected. ● Auto Answer Delay: To configure delay time before an incoming call is automatically answered. ● Auto Answer Mode: To set video or audio mode for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

3.9 Phone-Voice



Sections	Description
Mic Volume	To configure Microphone volume , from 1-15. 12 by default.
Speaker Volume	To configure Speaker Volume,from 1-15,12 by default.

3.10 Phone-Call Log

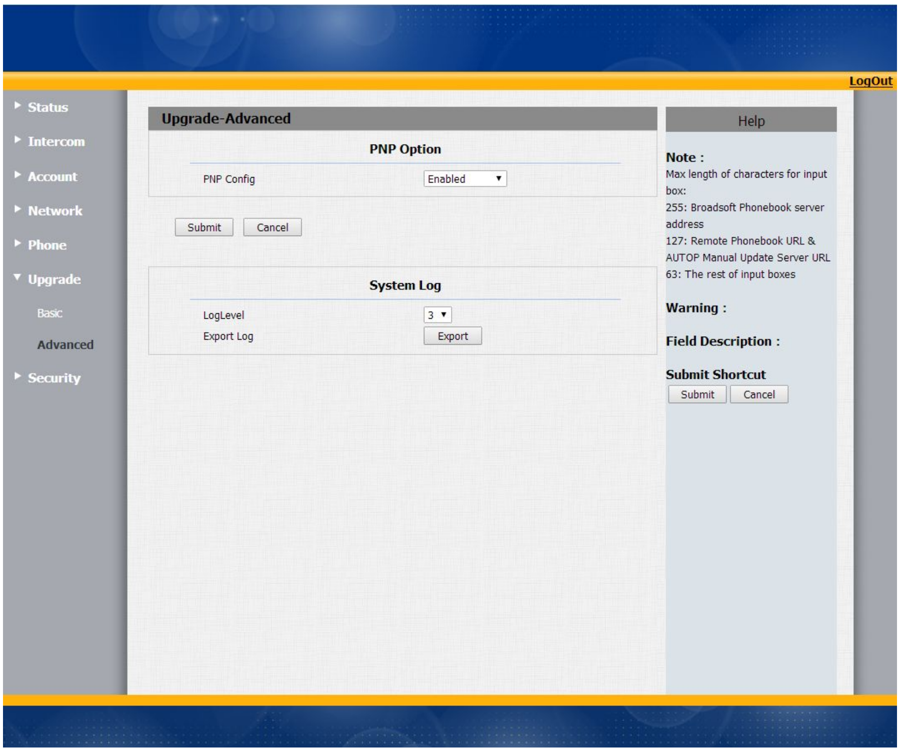


Sections	Description
Call History	To display call history records. Available call history types are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls. Users can check the call history in detail. Tick the number to delete or delete all logs. E10S supports 100 call logs.

3.11 Upgrade-Basic

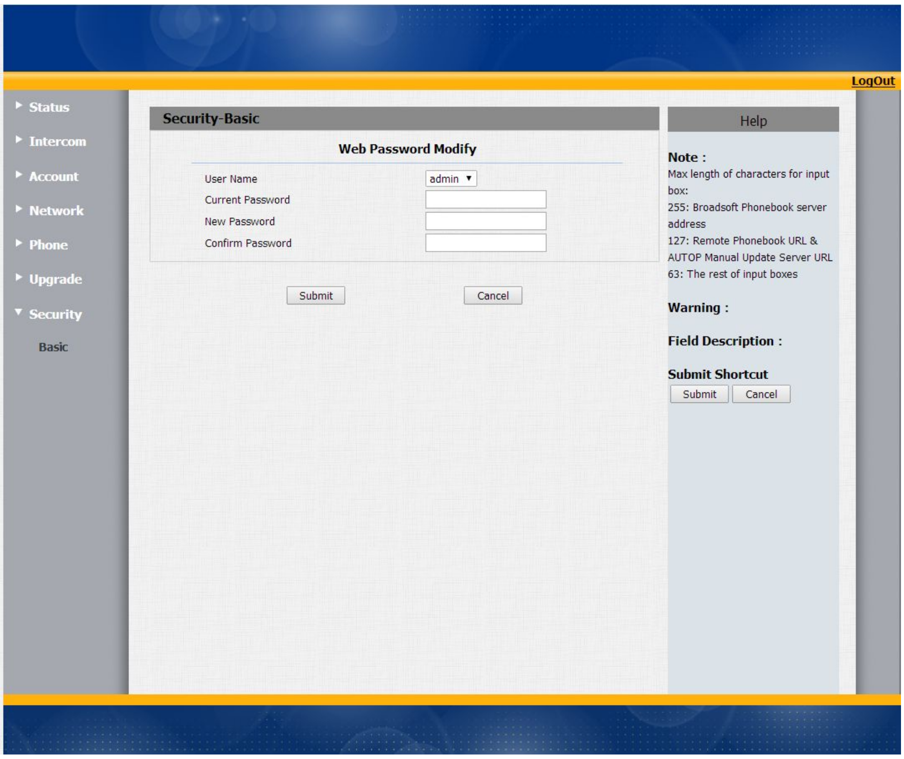
Sections	Description
Upgrade	To select upgrading rom file from local or a remote server automatically. **Note:** Please make sure it's right file format for right model.
Firmware version	To display firmware version, firmware version starts with MODEL name.
Hardware Version	To display Hardware version.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

3.12 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).
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.

3.13 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.