Akuvox SDP-R25 User Manual

12/05/2014

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Production Overview

1. Production Description

The Akuvox SPD-R25 is the video door phone, that you can connect with your Akuvox IP Phones for remote unlock control and monitoring. You can operate the indoor handset to communicate with visitors via voice and video, and unlock the door if you wish. It's applicable in apartment, villas, Office, building and so on.

2. Features

- Key Features
  - HD Voice
  - Compatible with Asterisk and Broadsoft platforms

- Physical Features
  - Camera: 2.0 Mega Pixels
  - White balance:Auto

- Lens: 4.0mm/F2.8
- Viewing Angle (Diagonal): 50
- Minimum Illumination: 1 LUX (without LED Illumination)
- LED: 6 LEDs
- Power Requirement: DC12V
- Operating Temperature; -30C ~ 40C
- Weight: 180g
- Size (WXHXD)185 x 68 x 50 mm

➢ Phone Features

- Video resolution: 320 x 240, with 20pics per second
- Wide angle lens and IR LEDs for night vision Crystal sound quality
- Remote door opening
- Integrated microphone and speaker
- Water-proof outdoor unit: IP55
- Support all the VoIP Phones

➢ IP-PBX Features

- Video Codec: H.264
- Audio Codec: PCMU
- VAD, CNG , Echo Canceller

➢ Network Features

- SIP v1(RFC2543), V2(RFC3261)
- Static IP/DHCP for IP conguration
- 3 DTMF modes: In-Band, RFC2833, SIP INFO
- HTTP/HTTPS Web Server for Management
- NTP for Auto Time Setting
- TFTP/FTP/HTTP/HTTPS client API

➢ Administration Features

- Auto provisioning using FTP/TFTP/HTTP/HTTPS/PnP
- Dial through IP PBX Using Phone Number
- Dial through IP PBX Using URL Address
- Conguration Managements with Web, keypad on the phone, and Auto Provisioning

➢ Security Features

- Support HTTPS (SSL)
- Support SRTP for Voice Data Encryption
- Support Login for Administration
- Sip Over TLS

Configuration

1. Web Login

1.1. Obtaining the IP address

The Akuvox R25 uses Static IP by default, and the default IP address is 192.168.1.100.

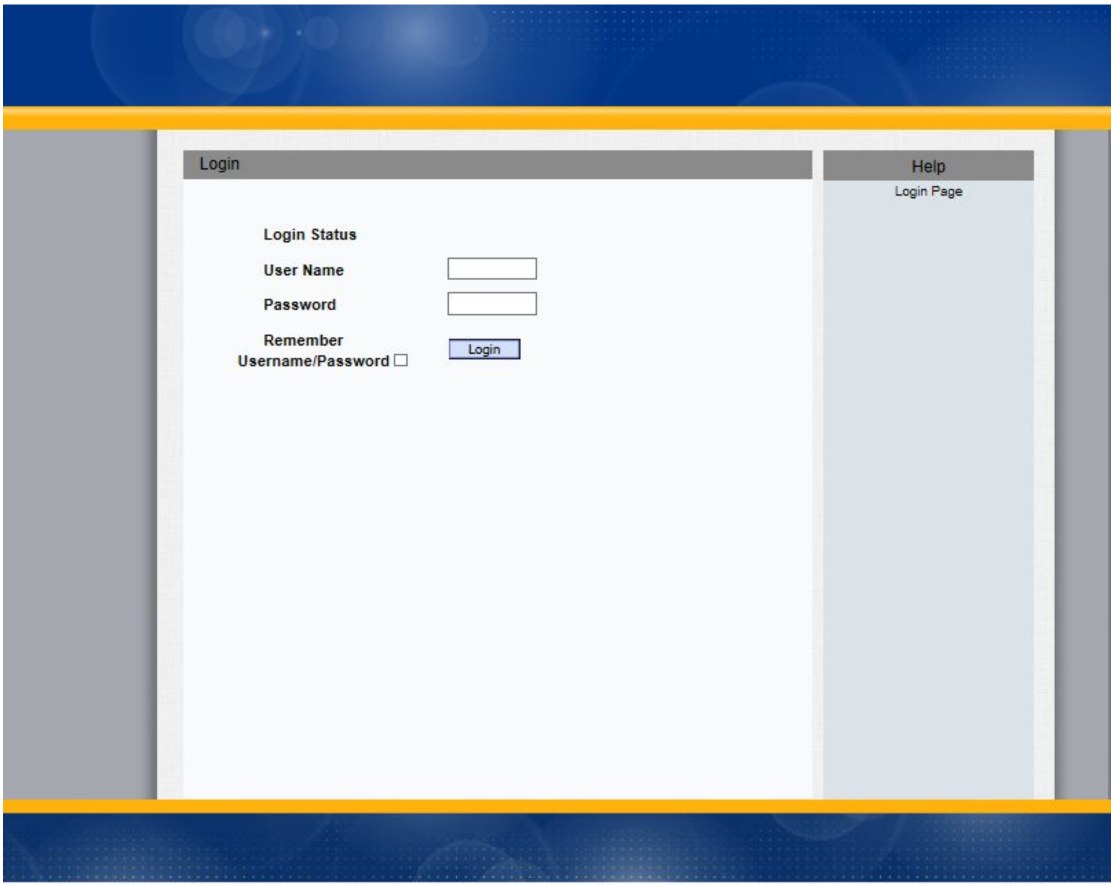
If the IP address is unknown, press the call button when the door phone is initialing, after a short period of time, the phone will announce its IP.

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,

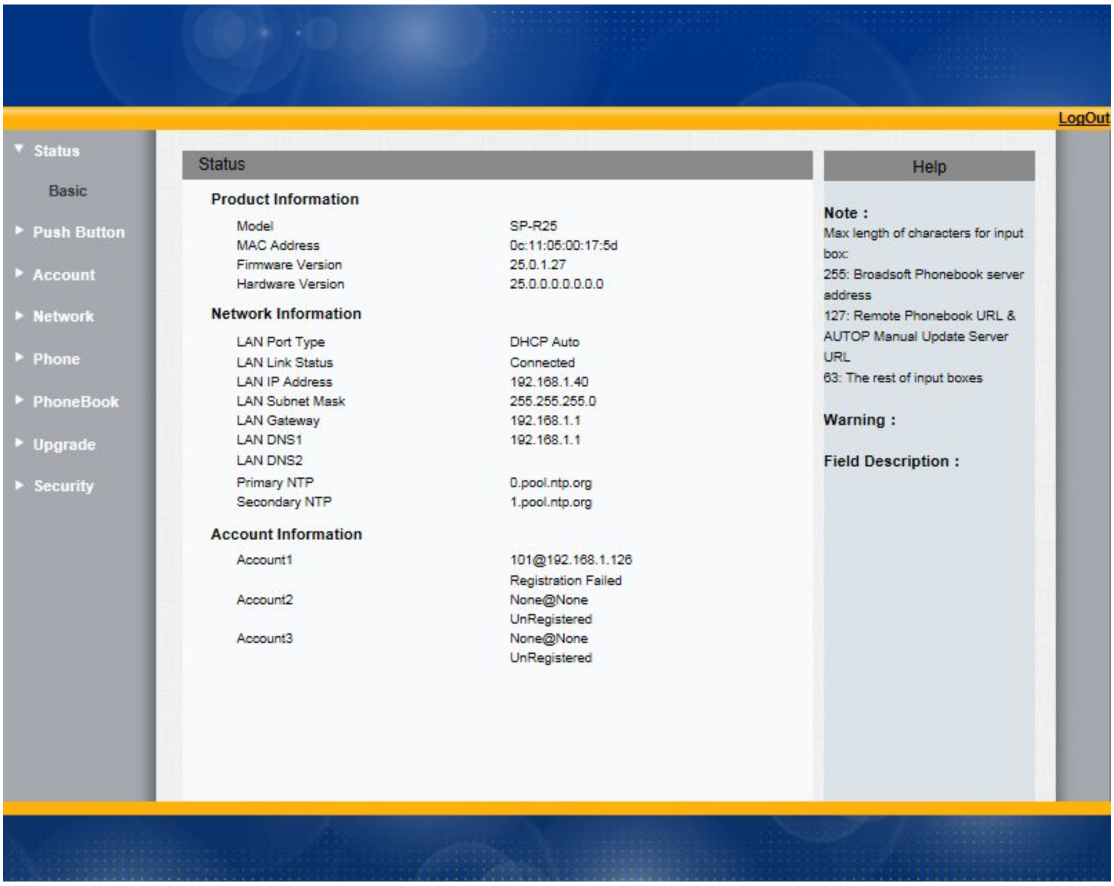
User name: admin

Password: admin



2. Status

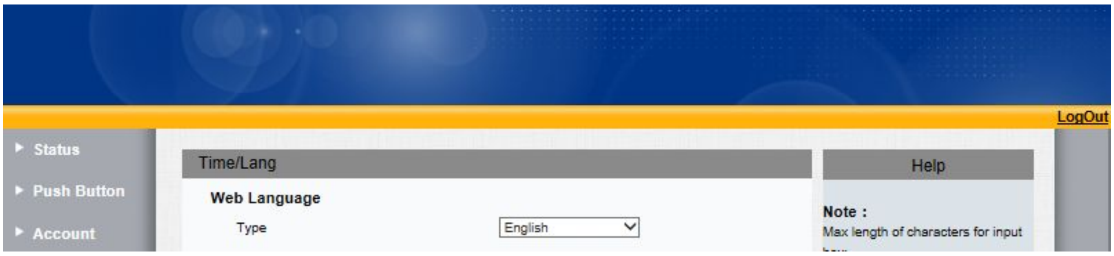
Status, including product information, network information and Account information, can be viewed from, Status > Basic.



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. Language

Web Language can be configured from, Phone > Time/Lang.

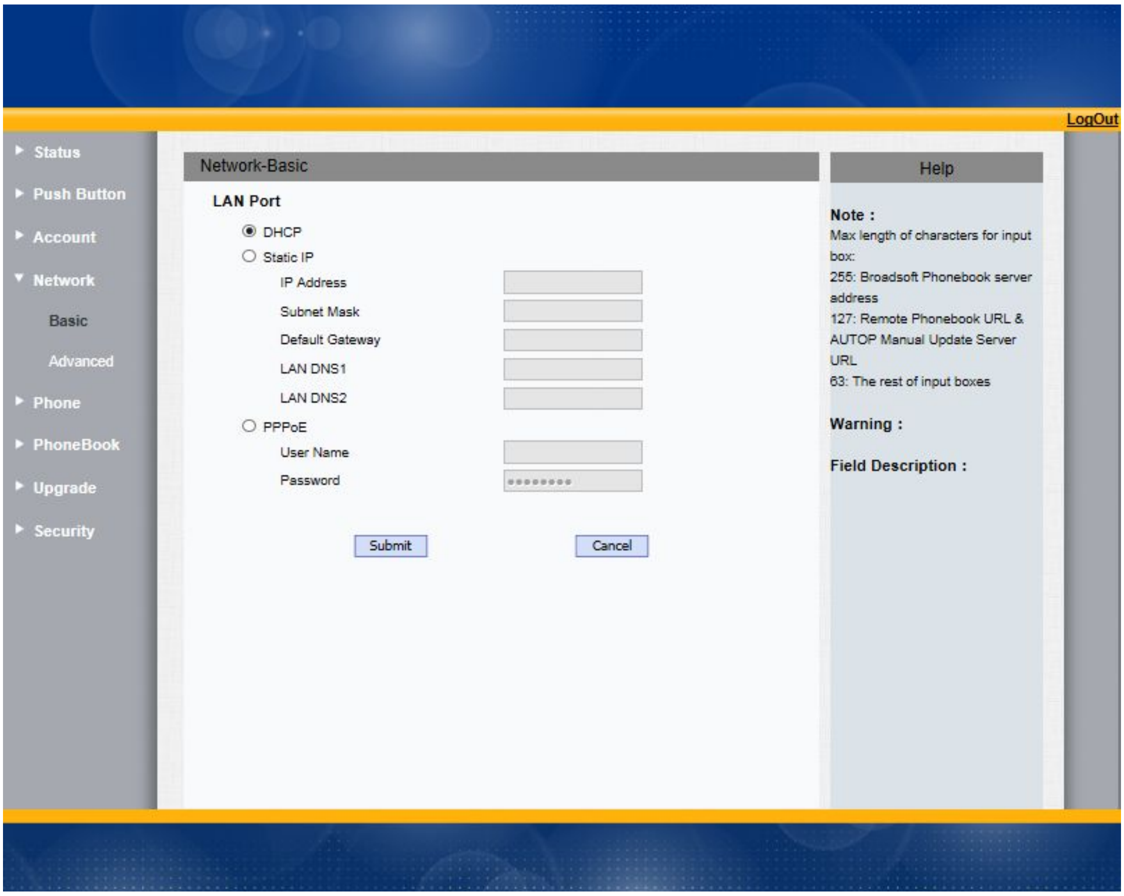


Select the desire language from the pull-down list of Type. The default language is English.

4. Network configuration

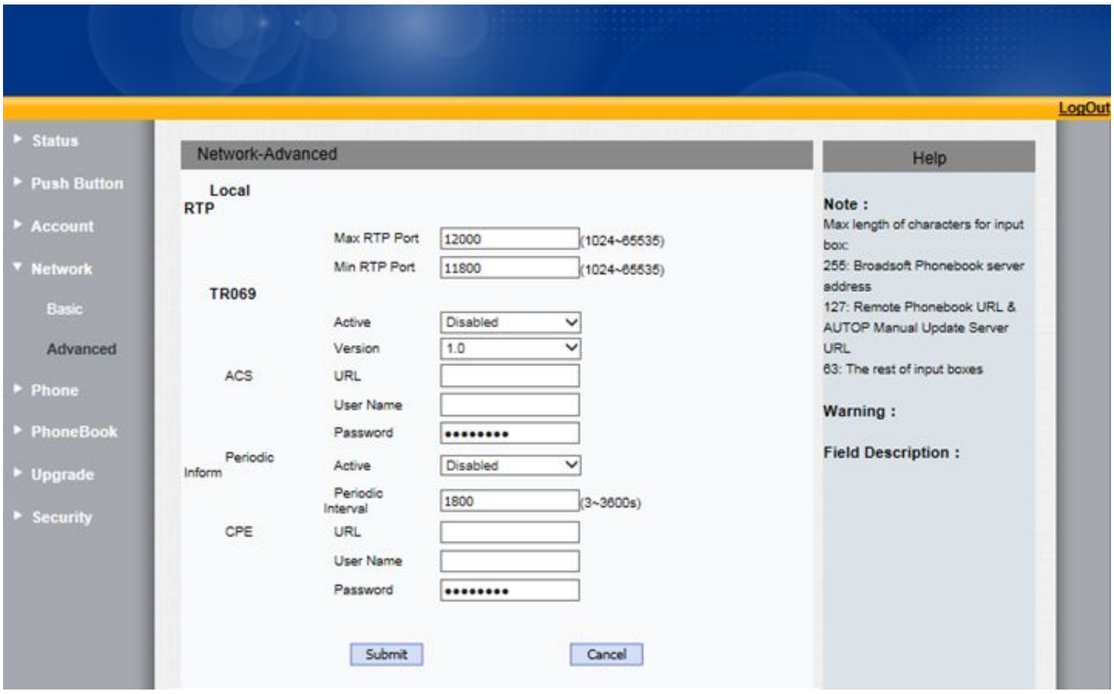
To configure the basic network settings, Go to Network > Basic.

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



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. ● PPPoE: Use PPPoE username/password to connect to PPPoE server.

For advanced settings, go to Network > Advanced

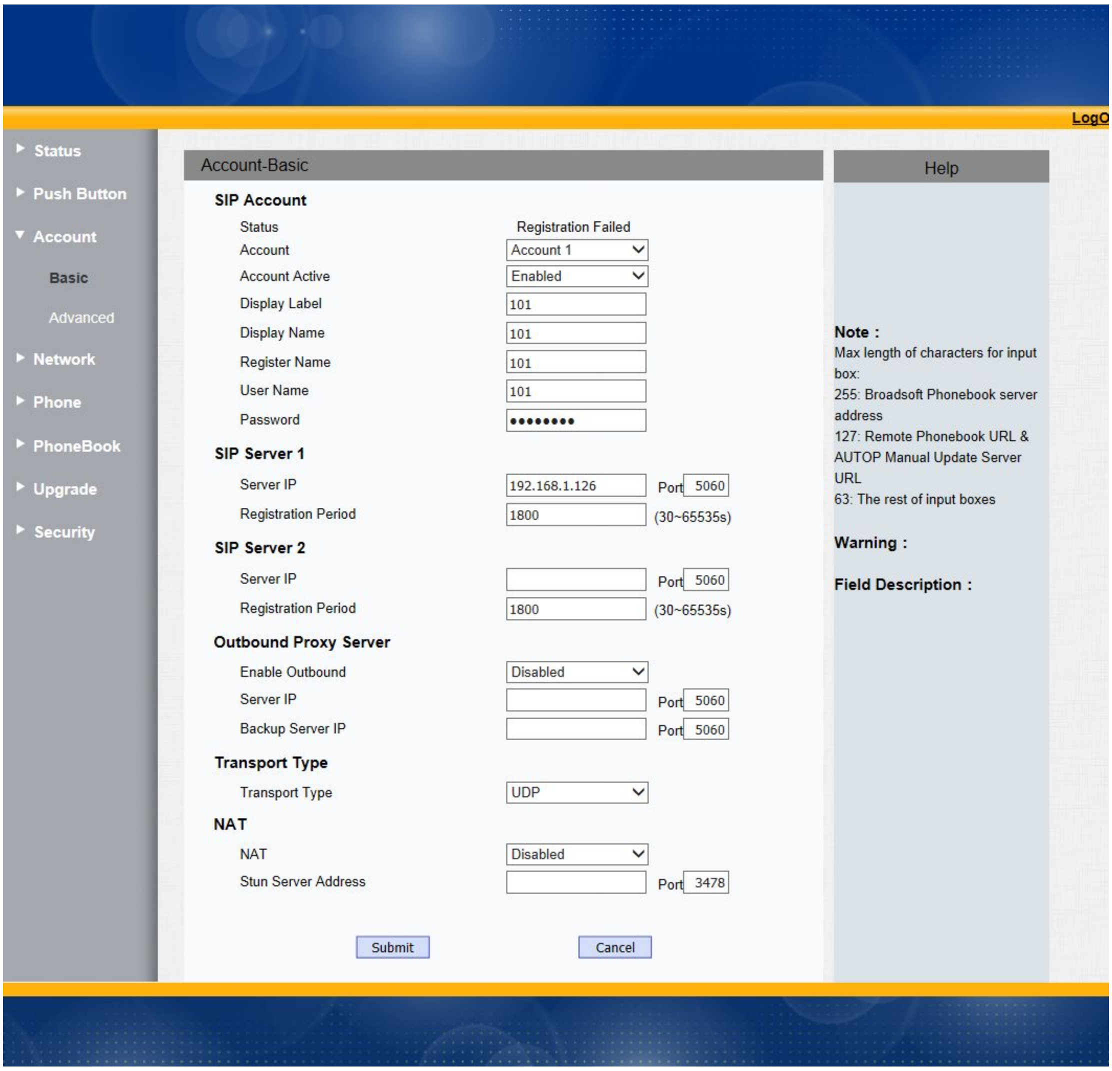


Sections	Description
Local RTP	To display and configure Local RTP settings. ● Max RTP Port: Determine the maximum port that RTP stream can use. ● Min RTP Port: Determine the minimum port that RTP stream can use.
TR069	To display and configure TR069 settings. ● Active: To enable or disable TR069 feature.

	• 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.

5. Account

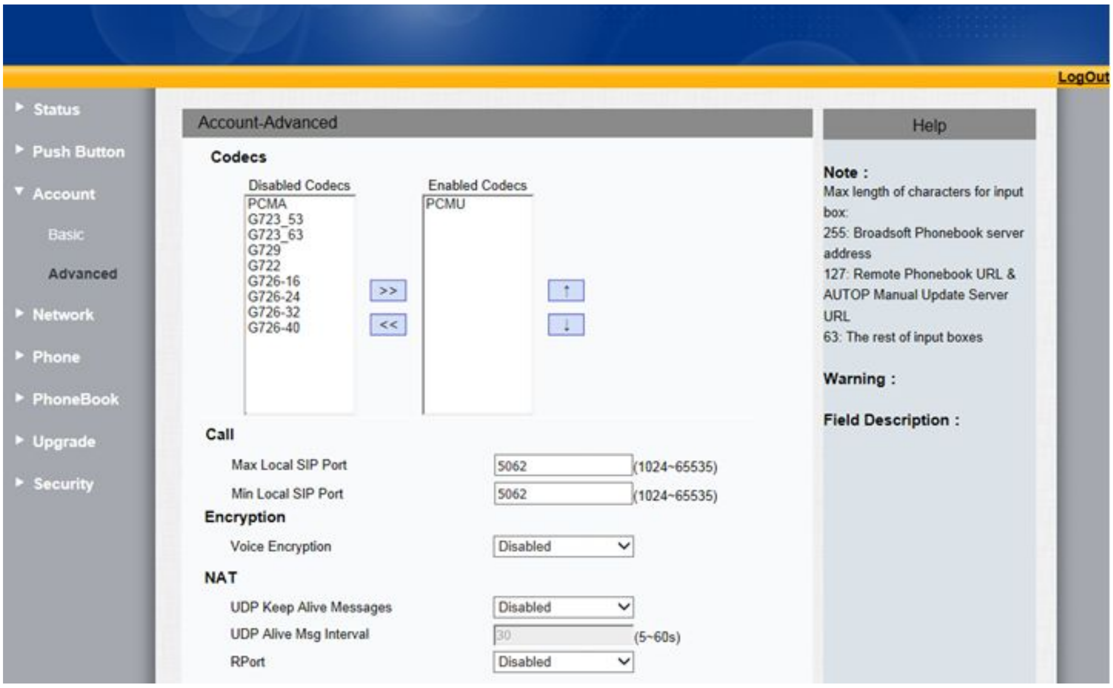
To configure your SIP account, go to Account > Basic.



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 displaying. ● 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.

For advance account settings, go to Account > Advanced.

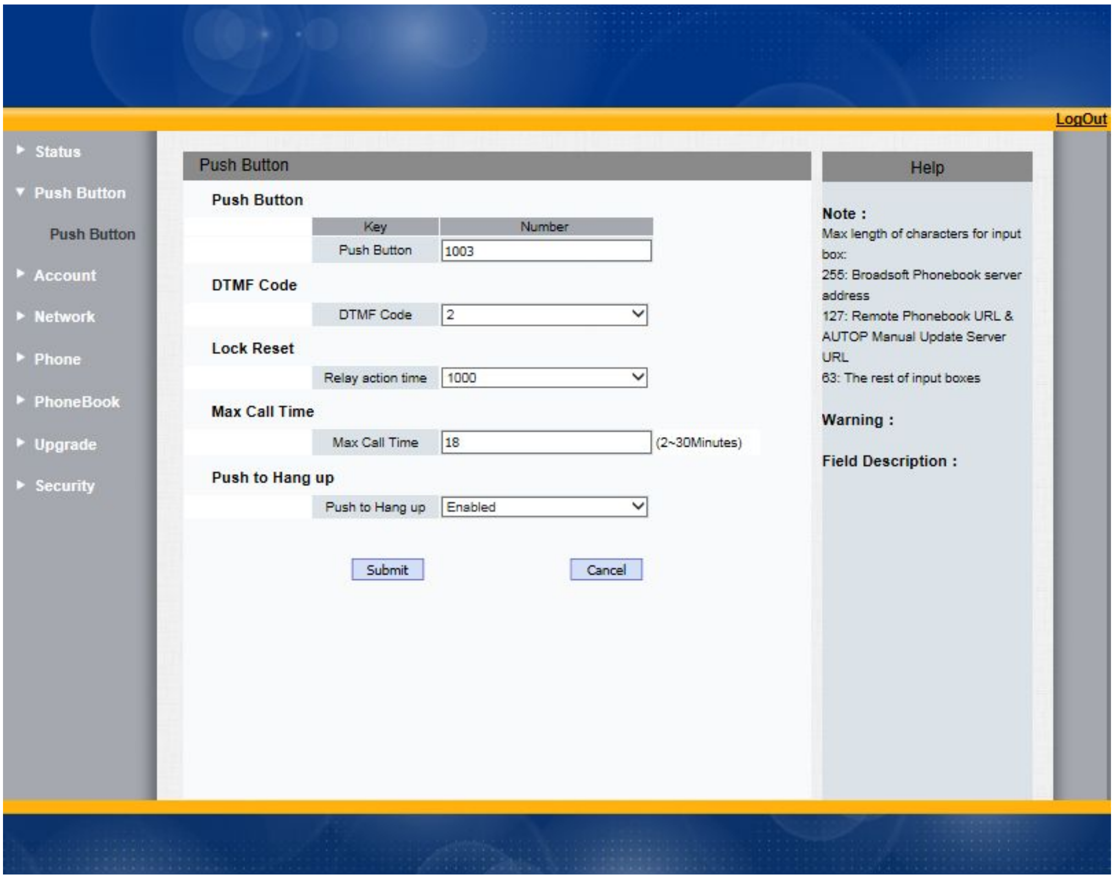


Sections	Description
Codecs	To display and configure available/unavailable codecs list. Codec means coder-decoder which is used to transfer analog signal to digital signal or vice versa. Familiar codecs are PCMU(G711U), PCMA(G711A), G722 (wid-bandth codecs), G723,G726,G729 and so on.
Call	To display 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.
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.

6. Push Button

To configure Push Button, go to Push Button.



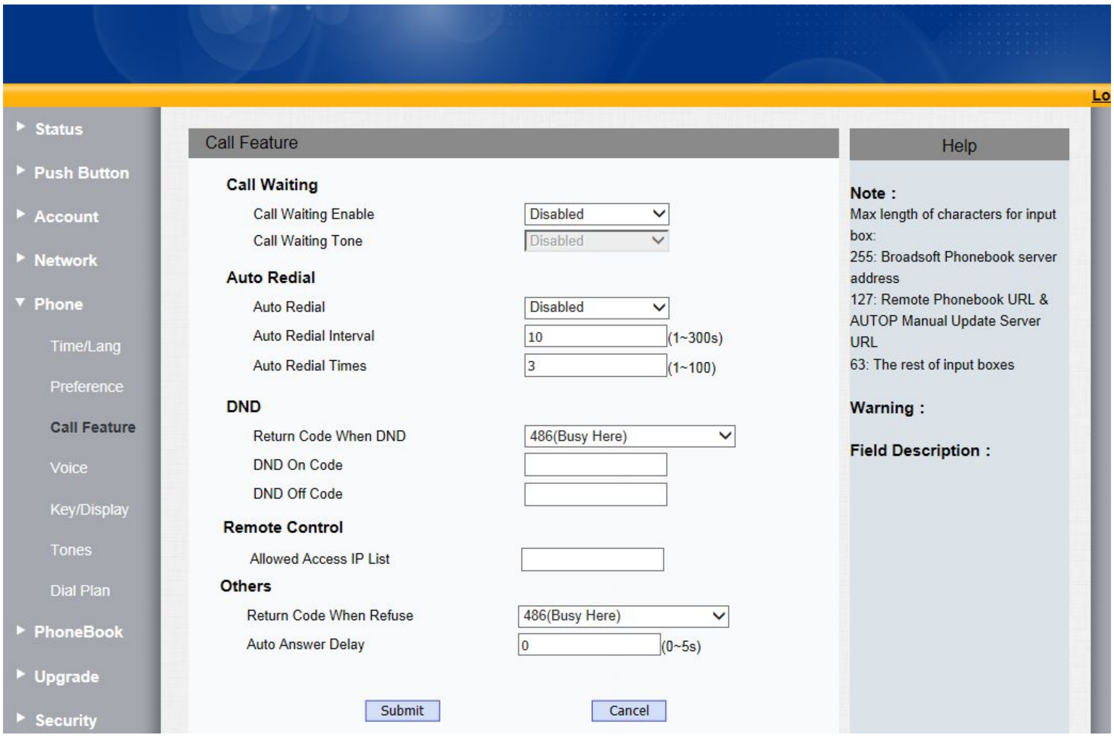
Sections	Description
Push Botton	To configure the destination number you want to contact with.
DTMF Code	To select the desired DTMF Code
Lock Reset	To set the lock reset time

Max Call Time	To configure the max call time
Push to Hang up	To enable or disable the Push to Hang up function

7. Phone

7.1. Call Feature

Call feature can be configured from, Phone > Call Feature.

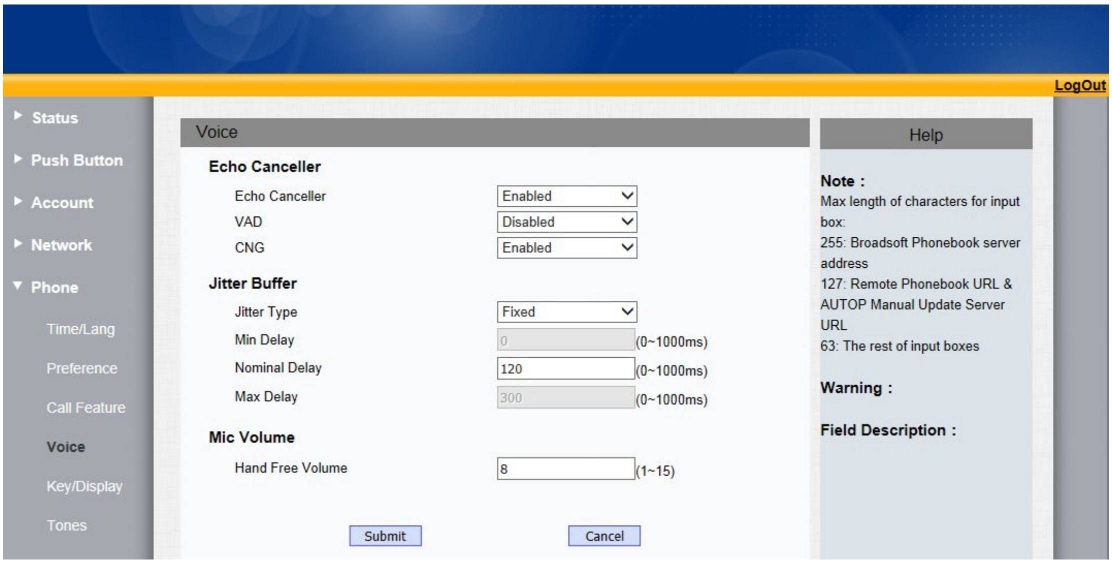


Sections	Description
Call Waiting	To enable or disable Call Waiting. ● Call Waiting Enable: If enabled, it allows IP phones to receive a new incoming call when there is already an active call. ● Call Waiting Tone: If enabled, it allows IP phones to play the call waiting tone to the waiting callee.

Auto Redial	Auto redial allows IP phones to redial an unsuccessful call for designated times within designated interval. ● Auto Redial: To enable or disable auto redial feature. ● Auto Redial Interval: Determine the interval between two consecutive attempts. ● Auto Redial Times: Determine how many times to redial.
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.
Remote Control	Remote Control allows specific host to interact with IP phone by sending HTTP or HTTPS requests. The specific action could be answering an incoming call, hangup an ongoing call and so on. ● Allowed Access IP List: To configure the allowed host address. **Note**: For now, IP phone can only support IP address, IP address list and IP address pattern as allowed hosts
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.

7.2. Voice

Voice can be configured from, Phone > Voice



Sections	Description
Echo Canceller	Echo Canceller: To remove acoustic echo from a voice communication in order to improve the voice quality . ● VAD (Voice Activity Detection): Allow IP phone to detect the presence or absence of human speech during a call. When detecting period of "silence", VAD replaces that silence efficiently with special packets that indicate silence is occurring. It can facilitate speech processing, and deactivate some processes during non-speech section of an audio session. It can avoid unnecessary coding or transmission of silence packets in VoIP applications, saving on computation and network bandwidth. ● CNG (Comfort Noise Generation): Allow IP phone to generate comfortable background noise for voice communications during periods of silence in a conversation. It is a part of the silence suppression or VAD handling for VoIP technology. CNG, in conjunction with VAD algorithms, quickly responds when periods of silence occur and inserts artificial noise until voice activity resumes. The insertion of artificial noise gives the illusion of a constant transmission stream, so that background sound is consistent throughout the call and the listener does not think the line has released.

Jitter Buffer	Jitter buffer is a shared data area where voice packets can be collected, stored, and sent to the voice processor in even intervals. Jitter is a term indicating variations in packet arrival time, which can occur because of network congestion, timing drift or route changes. The jitter buffer, located at the receiving end of the voice connection, intentionally delays the arriving packets so that the end user experiences a clear connection with very little sound distortion. IP phones support two types of jitter buffers: fixed and adaptive. Fixed: Add the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones. Adaptive: Capable of adapting the changes in the network's delay. The range of the delay time for the dynamic jitter buffer added to packets can be also configured on IP phones.
Mic Volume	To configure Microphone volume

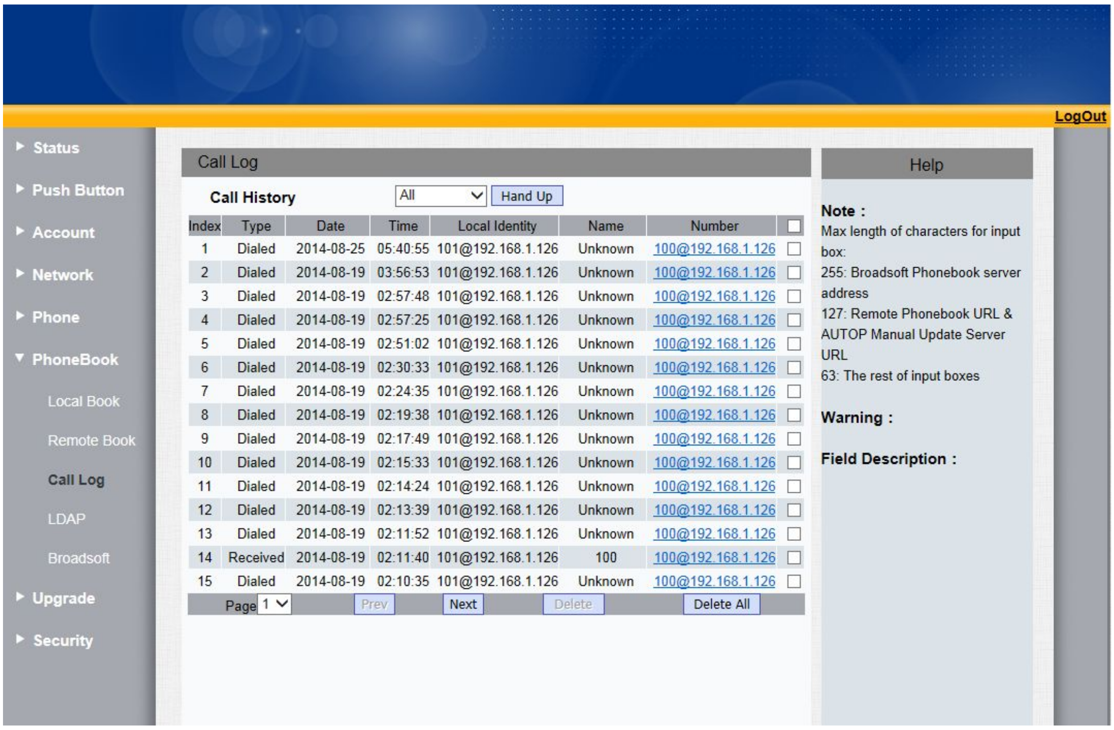
7.3. Country Ringtone

Country Ringtone can be configured from, Phone > Tone.

Select the desired country ringtone from the pull-down list of Select Country.

8. PhoneBook

8.1. Call Log

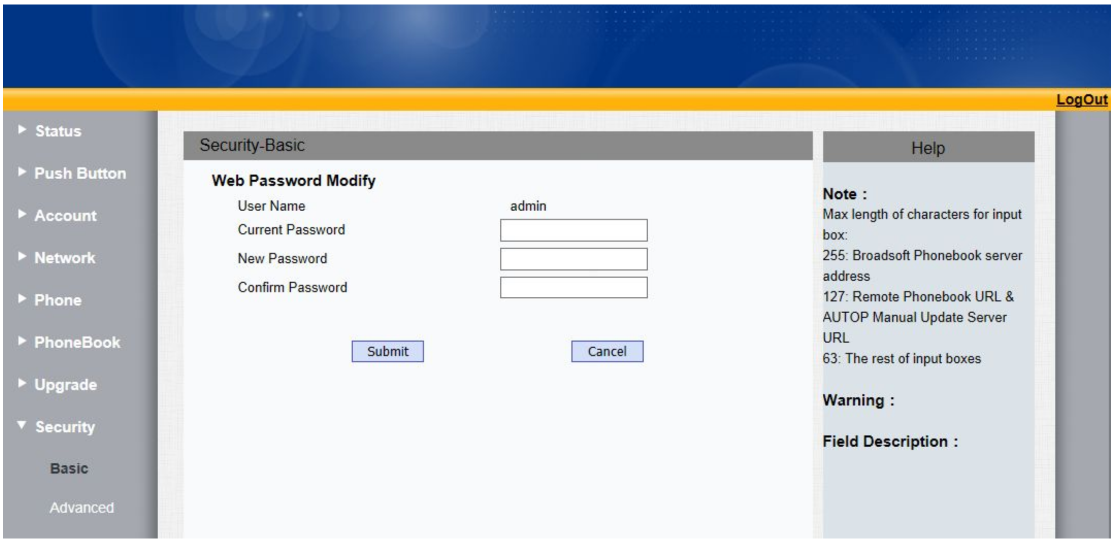


Sections	Description
Call History	To display call history records. Available call history type are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls. HangUp: To click to hangup ongoing call on the IP phone. **Note**: For "HangUp" feature, you need to have the remote control privilege to control IP phone via Web UI. Please refer to section "Remote Control" in the Web UI->Phone->Call Feature page.

9. Security

9.1. Web Password Modify

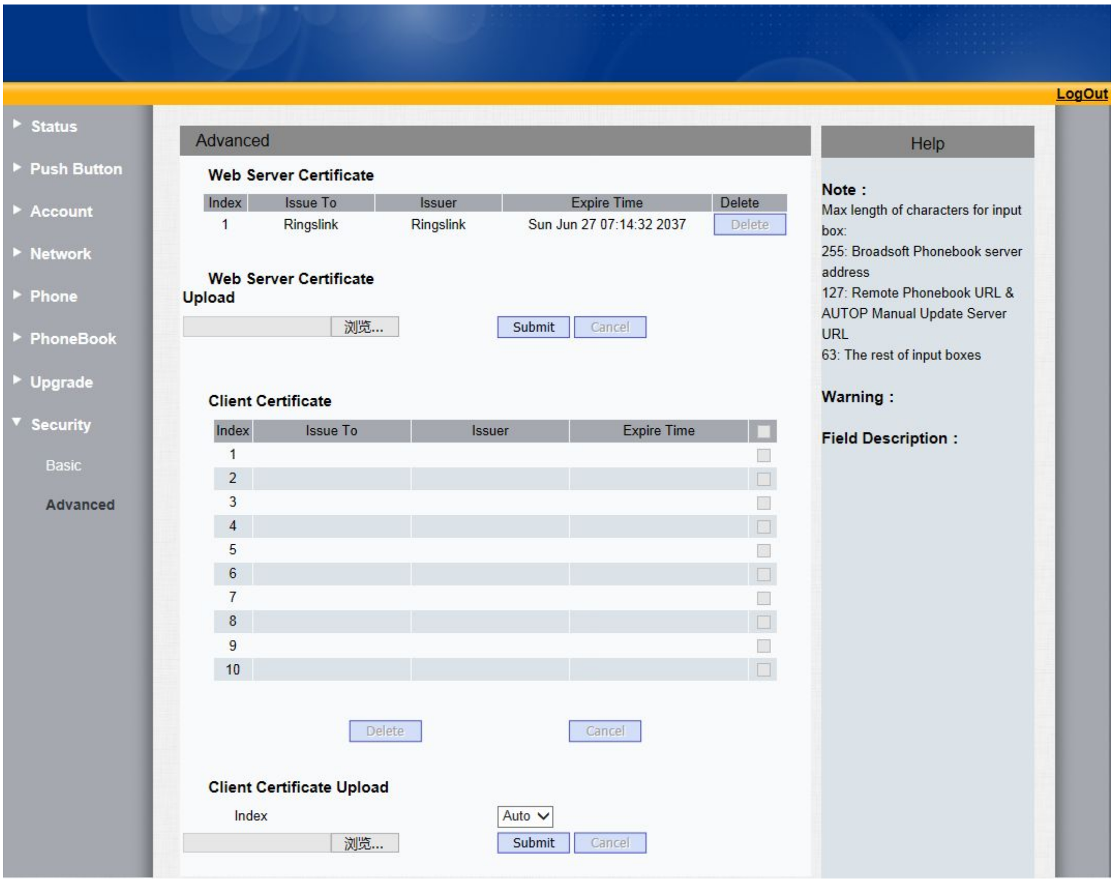
To modify web passoword, go to Security > Basic



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

9.2. Web Server Certificate

To check or upload your web server certificate, go to Security > Advanced

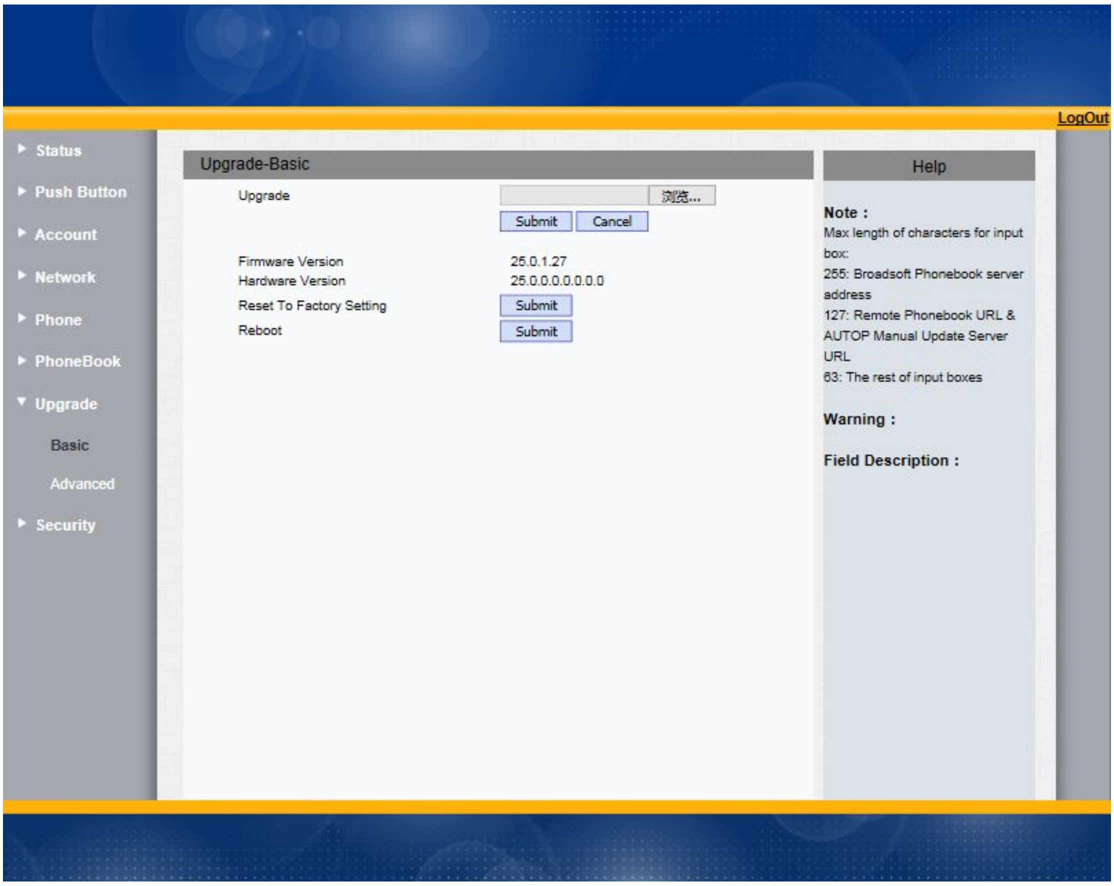


Sections	Description
Web Server Certificate	To display or delete Certificate which is used when IP phone is connected from any incoming HTTPs request. **Note**: The default certificate could not be deleted.
Web Server Certificate Upload	To upload a certificate file which will be used as server certificate.
Client Certificate	To display or delete Certificates which is used when IP phone is connecting to any HTTPs server.
Client Certificate Upload	To upload certificate files which is used as client certificate.

10. Upgrade

10.1. Basic upgrade

To upgrade your device, go to Upgrade > Basic.

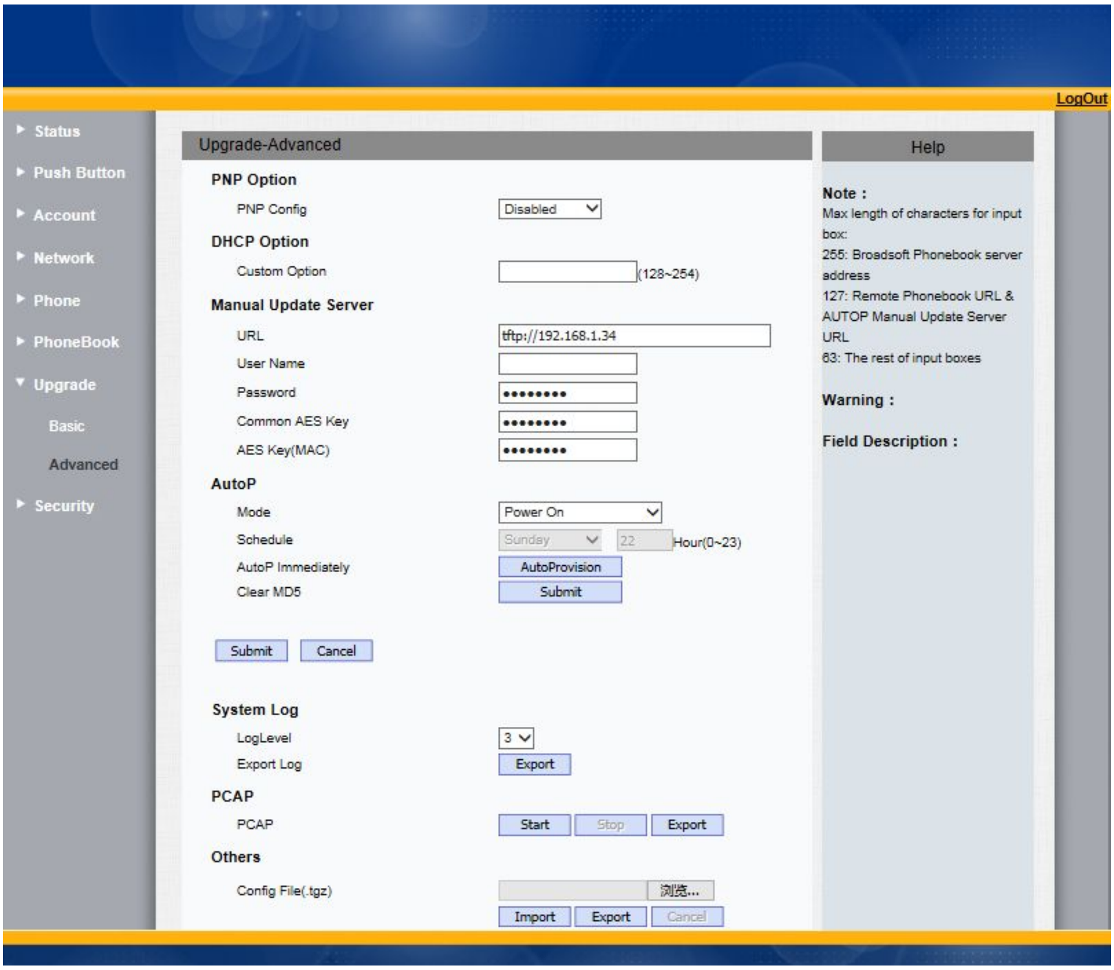


Sections	Description
Upgrade	To select upgrading rom file from local or a remote server automatically. Note: Please make sure it's right file format for right model.
Firmware version	To display firmware version, firmware version starts with MODEL name.
Hardware Version	To display Hardware version.

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.

10.2. Advanced Upgrade

To do the advanced upgrade for your device, go to 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.conf if IP phone's MAC address is 0c1105888888). Note: AES is one of many encryption, it should be configure only configure filed 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 syslog level and export syslog file. Syslog level: From level 0~7.The higher level means the more specific syslog is saved to a temporary file. By default, it's level 3.

	Export Log: Click to export temporary syslog 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.
Others	To display or configure others features from this page. Config file: To export or import configure file for IP phone.