N-SP80AS1 SIP Video Door Station,

N-SP80VS1 SIP Audio Door Station

User Manual

Content

Production Overview

1 Production description

TOA N-SP80 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. N-SP80 supports two types: N-SP80AS1(Audio) and N-SP80VS1(Video).

They are often used in locations such as: parking facilities, college campuses, medical centers, and industrial parks.

N-SP80VS1
SIP Video Door Station

N-SP80AS1
SIP Audio Door Station

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;(N-SP80VS1 only);
- MPEG-4/MJPEG compression; (N-SP80VS1 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 (N-SP80VS1 only)
- Resolution: up to 1080P (N-SP80VS1 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

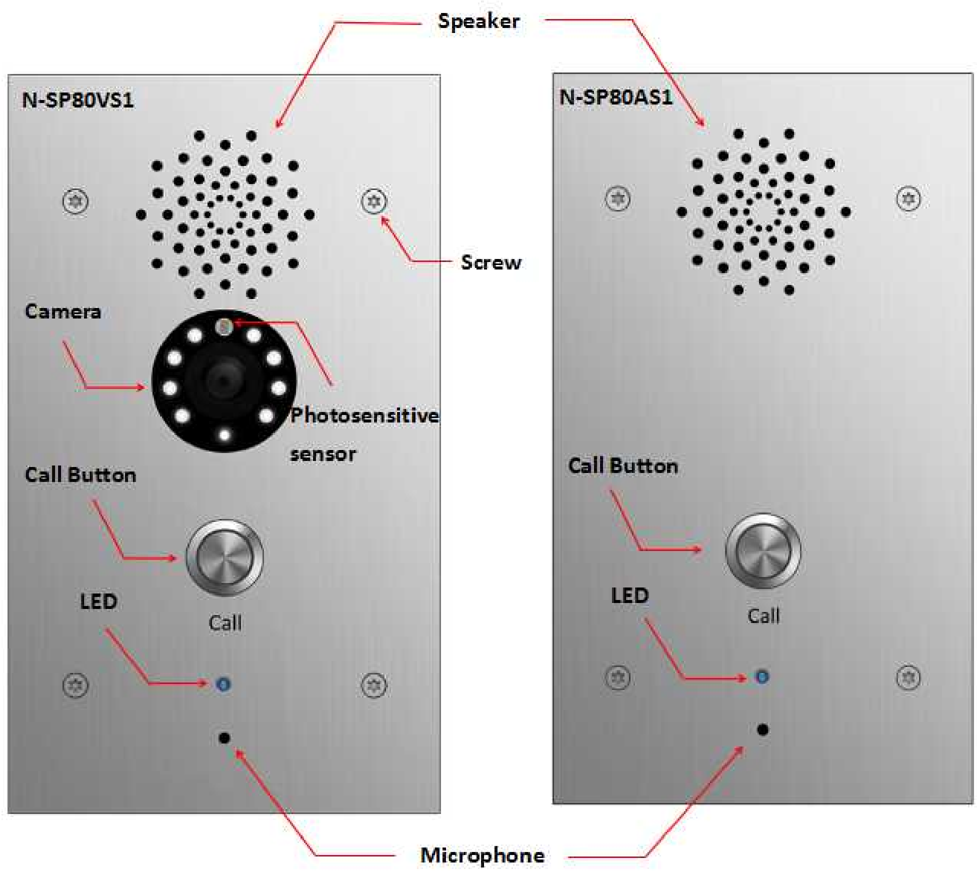
- 2x10/100Mbps Ethernet Port
- Security: Password Protection, IP address filtering, SIP over TLS, HTTPS encryption, user access log

- Protocols support: IPv4, HTTP, HTTPS, FTP, SNMP, DNS, NTP, RTSP, RTP, TCP, UDP, ICMP, DHCP, ARP

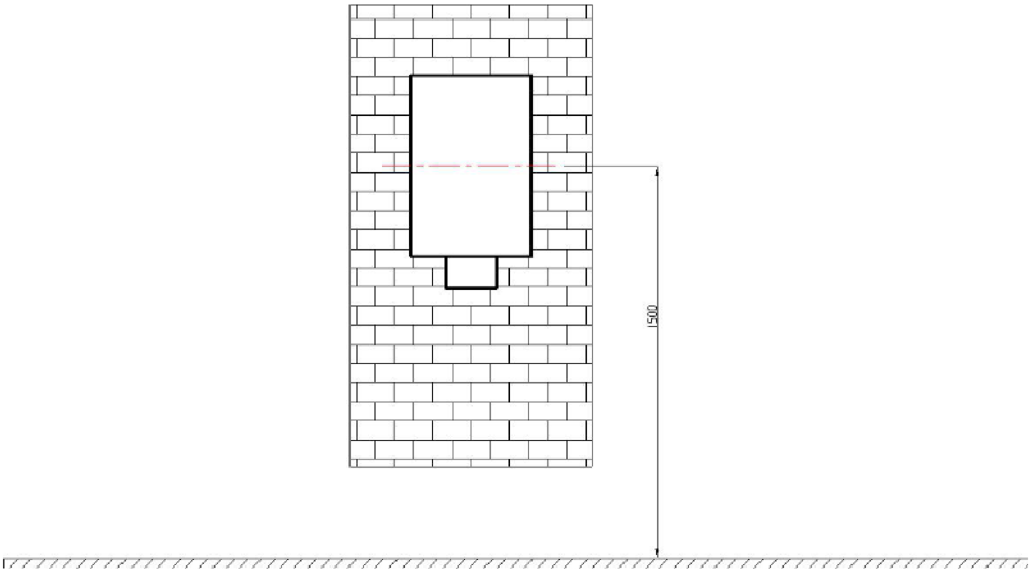
➢ SIP Features

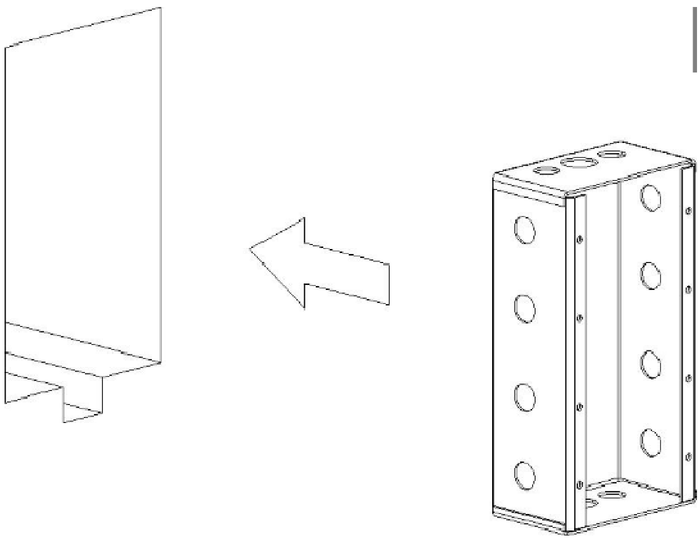
- SIP v1(RFC2543), SIP v2(RFC3261)
- Audio codecs: G.711a, G.711 μ , G.722, G.729
- Video codecs: MPEG-4/MJEG (N-SP80VS1 only)
- Speech Quality: 7kHz Audio
- Echo Cancellation
- Voice Activation Detection
- Comfort Noise Generator

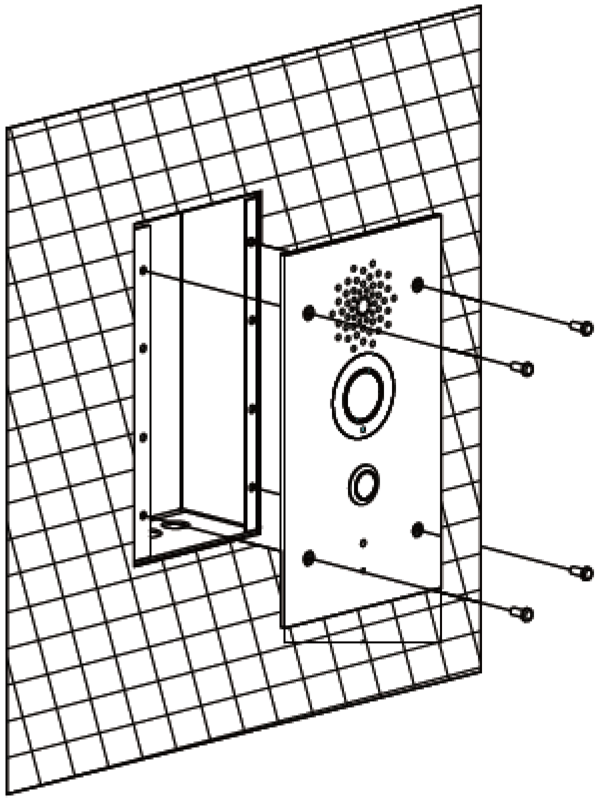
3 Panel Description



4 Installation







Installation step:

1. Use cement to fix the back cover in the wall(installation height about 1500mm)
2. Place N-SP80VS1/AS1 panel into the back cover.
3. Use screws to fix the panel.

Configuration

1 Web Login

1.1 Obtaining the IP address

The TOA N-SP80VS1/AS1 uses Static IP by default, and the default IP address is 192.168.1.102.

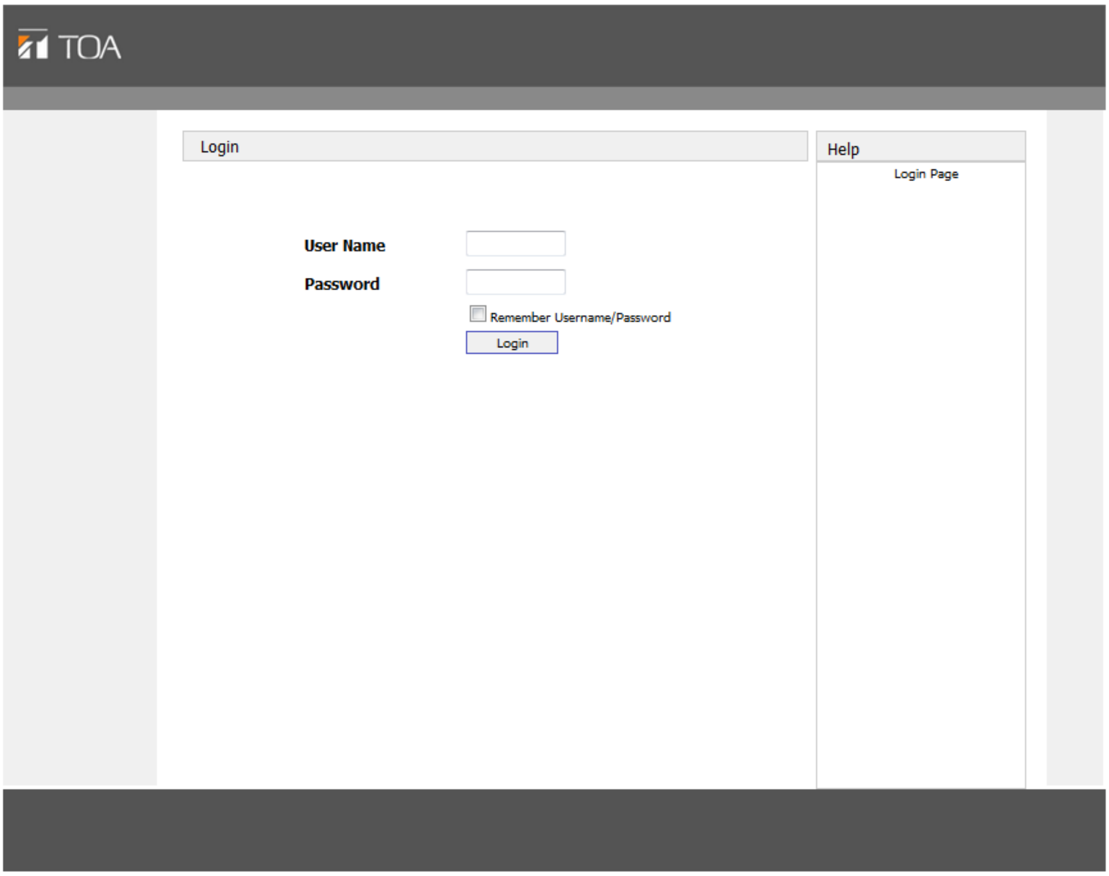
If the IP address is unknown, press the call button when LED light turns blue, after a short period of time(about 5s), the phone will announce its IP.

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:

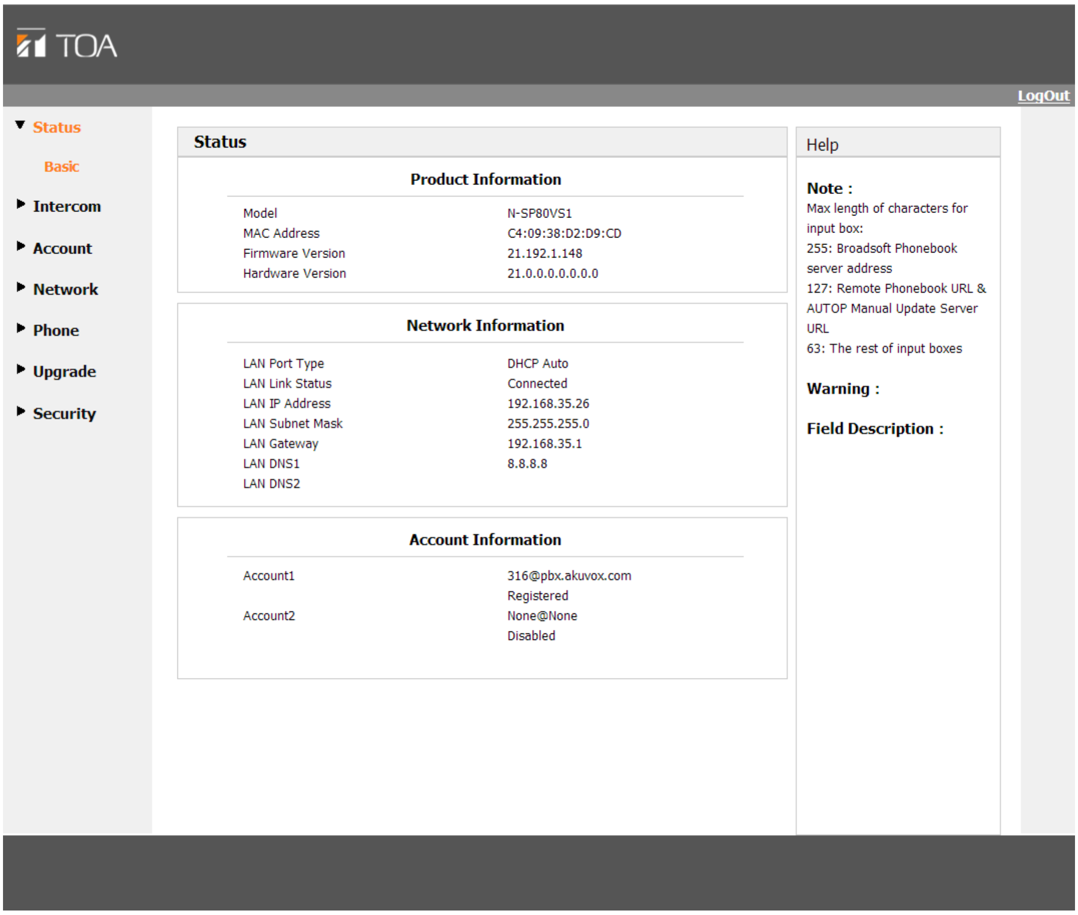
User name: admin

Password: admin



2 Status

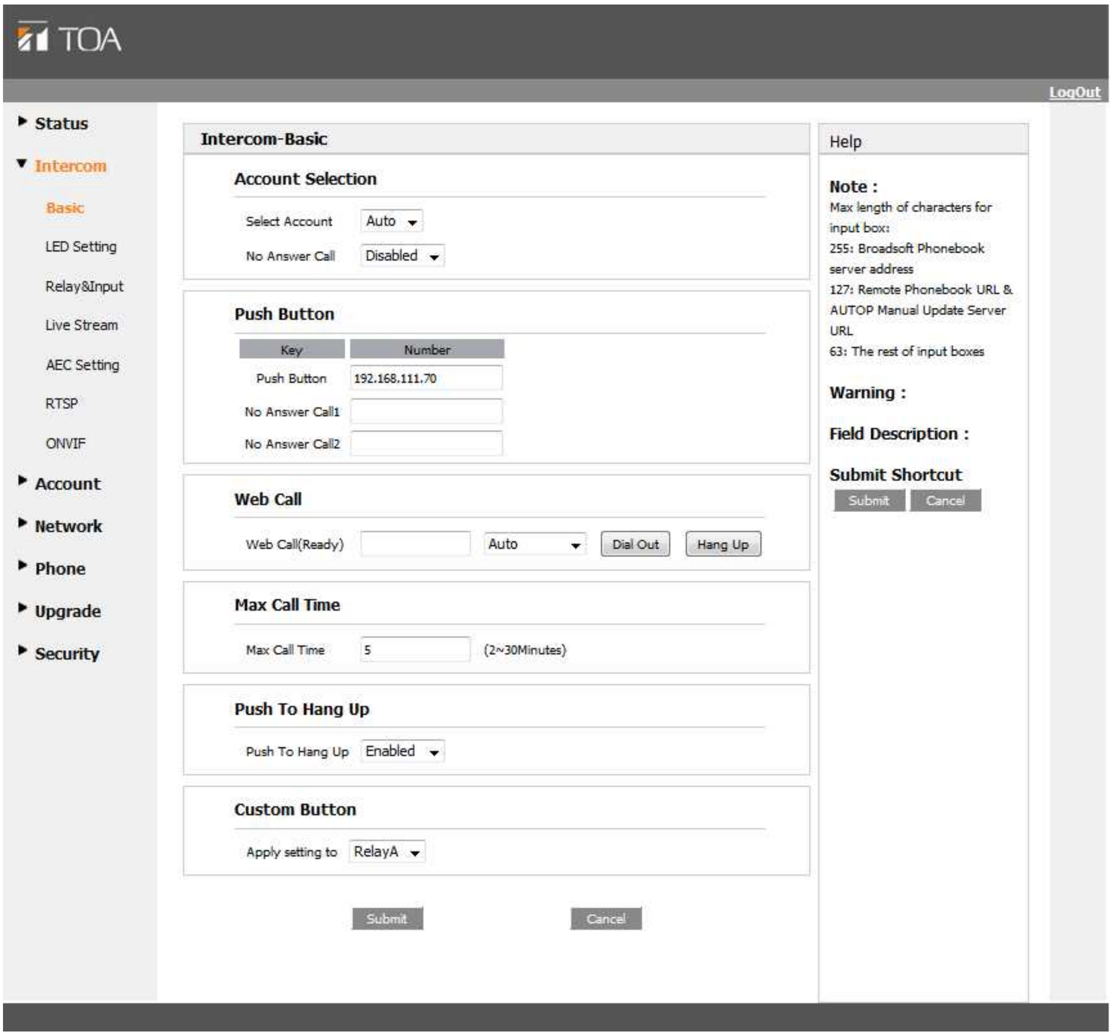
Status, including product information, network information and Account information, can be viewed from, Status -> Basic.



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

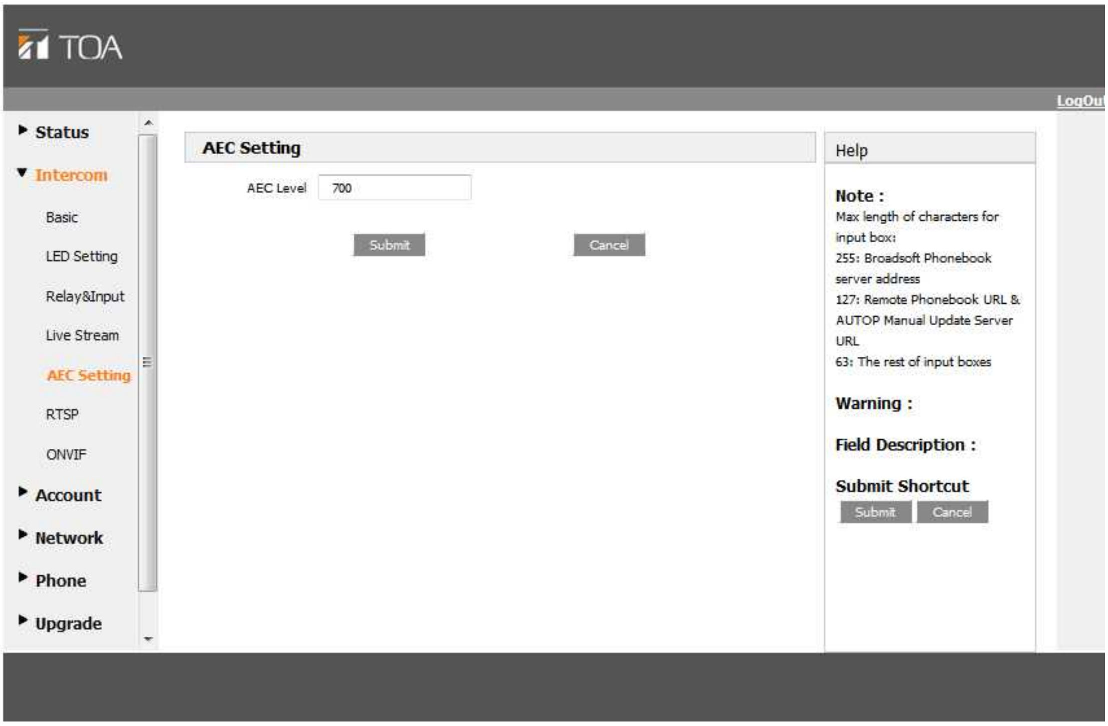
3.1 Basic



Sections	Description
Account Selection	● Select Account: N-SP80VS1/AS1 supports 2 accounts. You can choose one account or Auto mode for the following Intercom basic settings. ● No Answer Call: Choosing Enabled is for 3 No Answer Calls. 1st call is to the cell of "Push Button", 2nd is "No Answer Call1", 3rd is "No Answer Call2".
Push Button	● Push Button: To configure the destination number or IP you want to contact with. No Answer Call1 is for the 2nd call when the destination number doesn't answer its call. No Answer Call2 is the next of Call1.
Web Call	To dial out or answer the phone from website.
Max Call Time	To configure the max call time
Push to Hang up	To enable or disable the Push to Hang up function

3.2 LED Settings

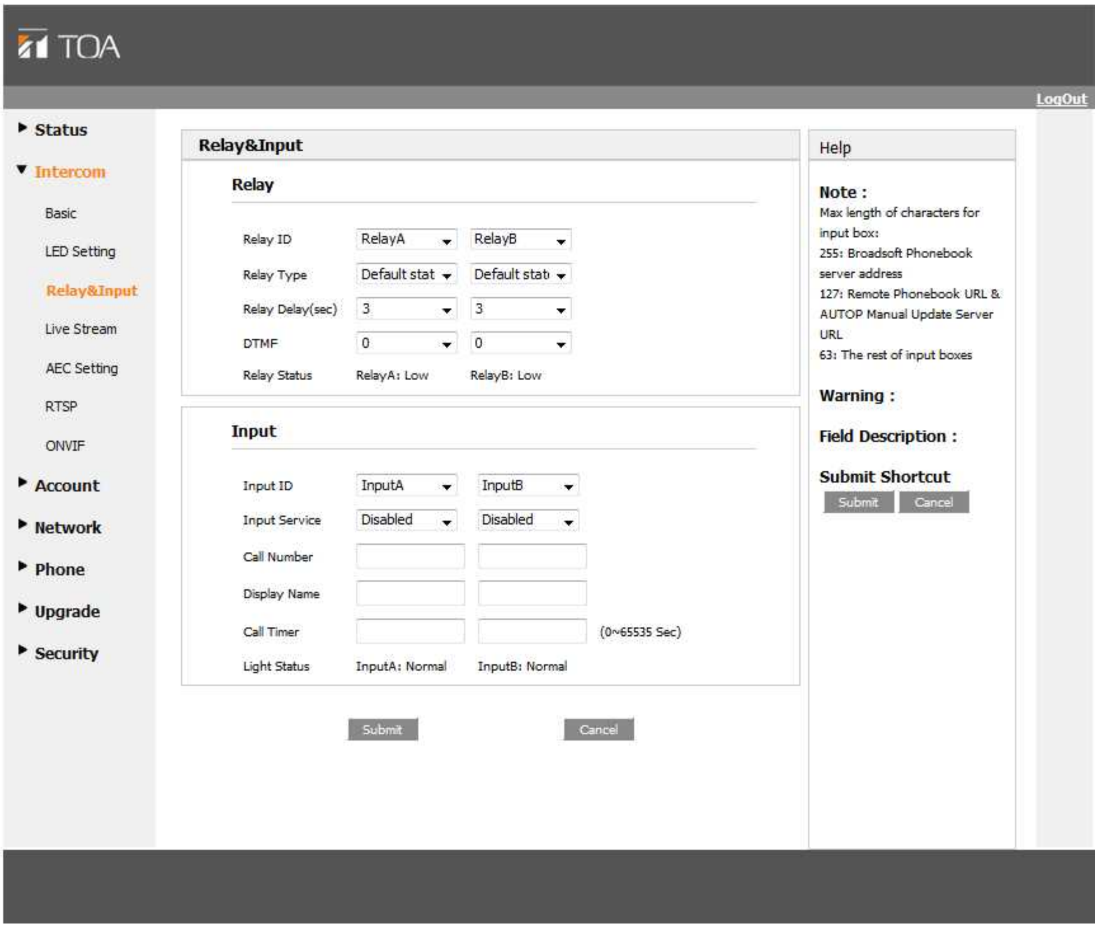
To configure the different LED blink mode of different states.



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

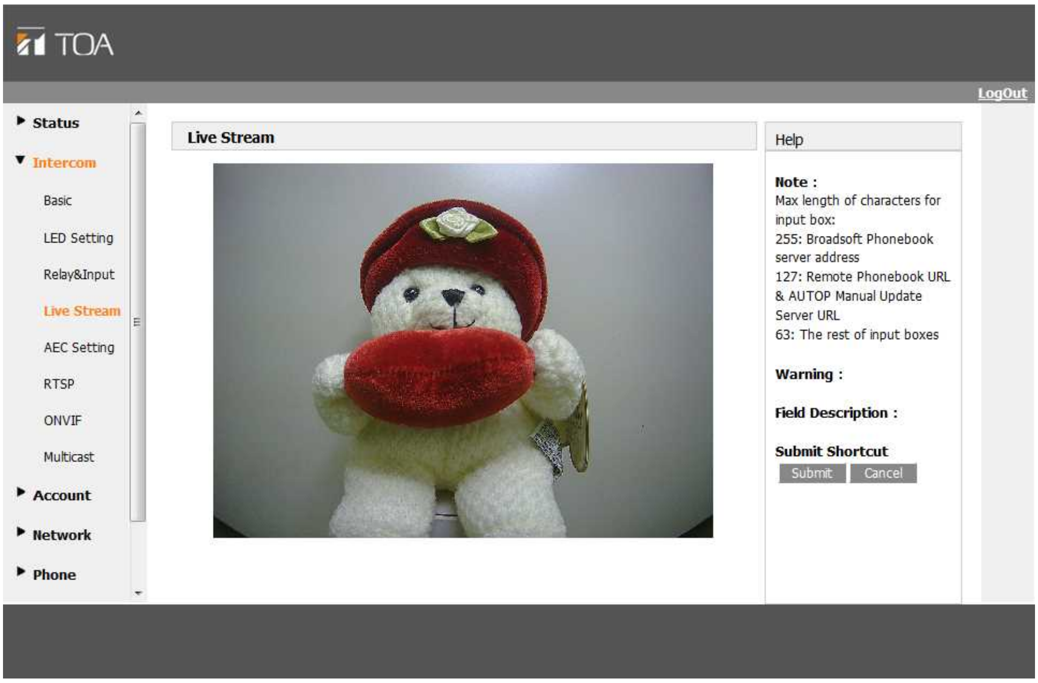
To configure unlock and alarm setting. Go to the path: Push Button-> Relay&Input.



Sections	Description
Relay	To configure some settings about unlock ● Relay ID: N-SP80VS1/AS1 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	There is a sensor that is used to anti vandal in N-SP80VS1/AS1. When N-SP80VS1/AS1 is broken by violent means. The sensor will be triggered, then management center will receive the alarm. ● Input ID: N-SP80VS1/AS1 supports 2 optical-couplers. Once the optical-coupler is triggered, it will alarm when this function is enabled. ● Input Service: Disable by default ● Call Number: To setup management center number for alarm.

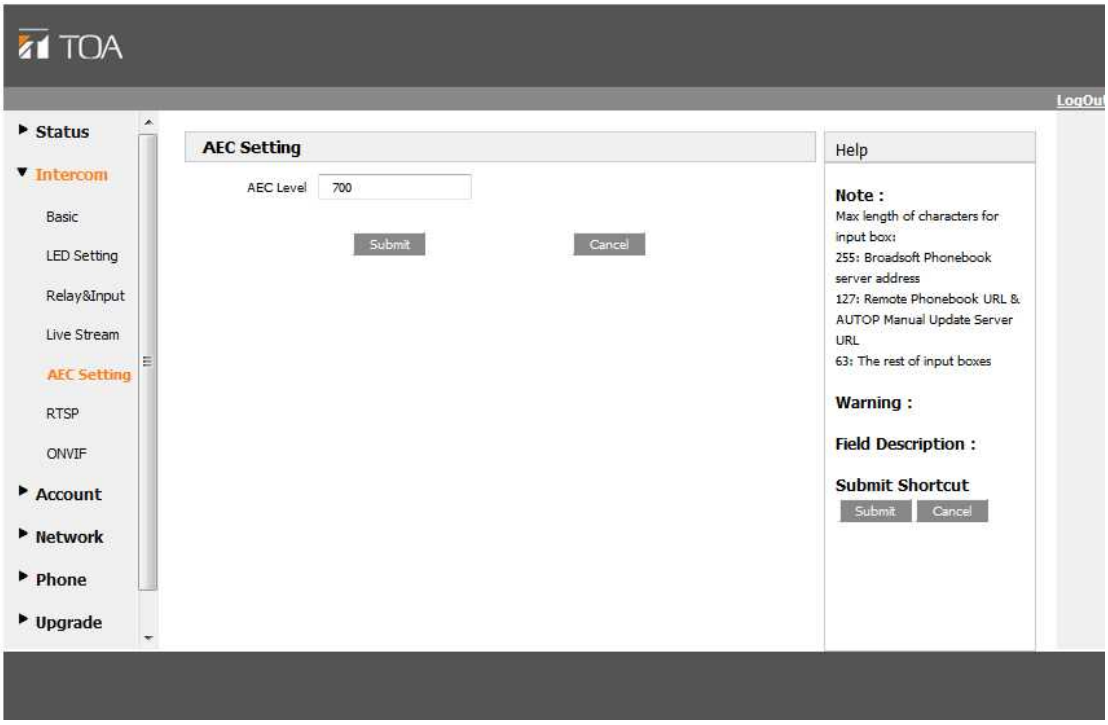
	● Display Name: Which is sent to the other call party for displaying ● Call Timer: Every its seconds makes call during the input is activated. ● Light Status: Here is an indication of a status of input.

3.4 Live Stream(Optional)



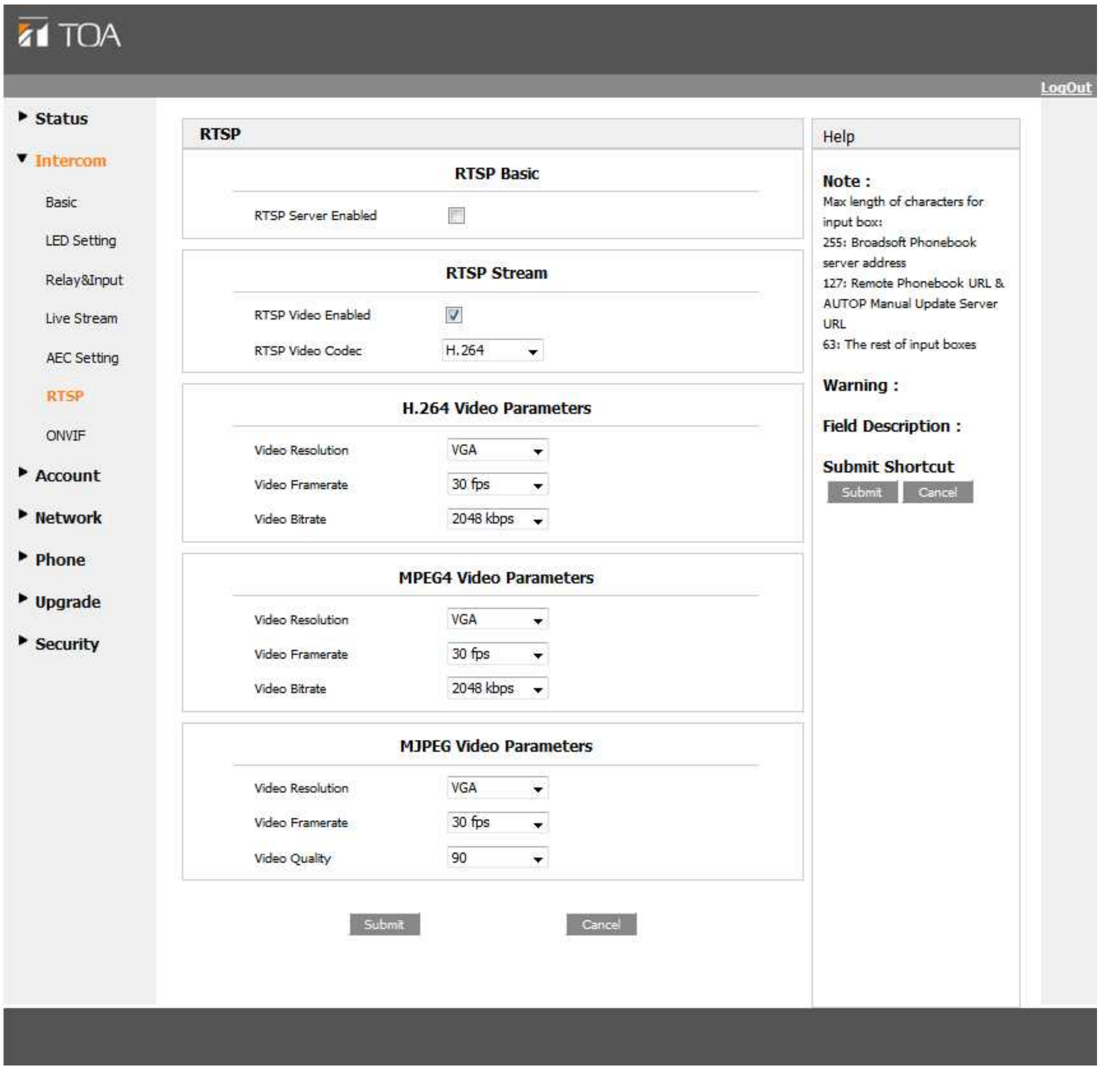
Sections	Description
Live Stream	To check the real-time video from N-SP80VS1.

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.

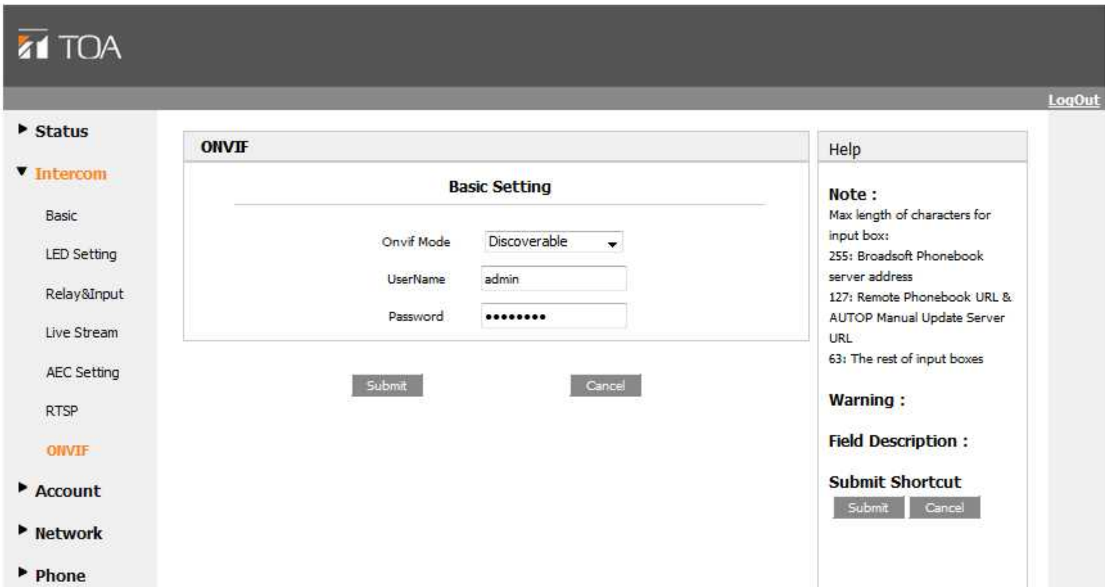
3.6 RTSP(optional)



Sections	Description
RTSP Basic	To active the RTSP function, then N-SP80VS1 can be monitored.
RTSP Stream	To enabled RTSP video and select the video codec. N-SP80VS1 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 common video function with great communication application integration ability and less core program space.

	To modify the resolution, framerate and bitrate of MPEG4
MJPEG Video Parameters	MJPEG: called Motion Joint Photographic Experts Group. It is a video encoding format.in which each image is compressed separately by JPEG.MJPEG compression can produce high quality video image and has a flexible configuration in video definition and Compressed frames To modify the resolution, framerate and bitrate of MJPEG

3.7 Onvif(optional)

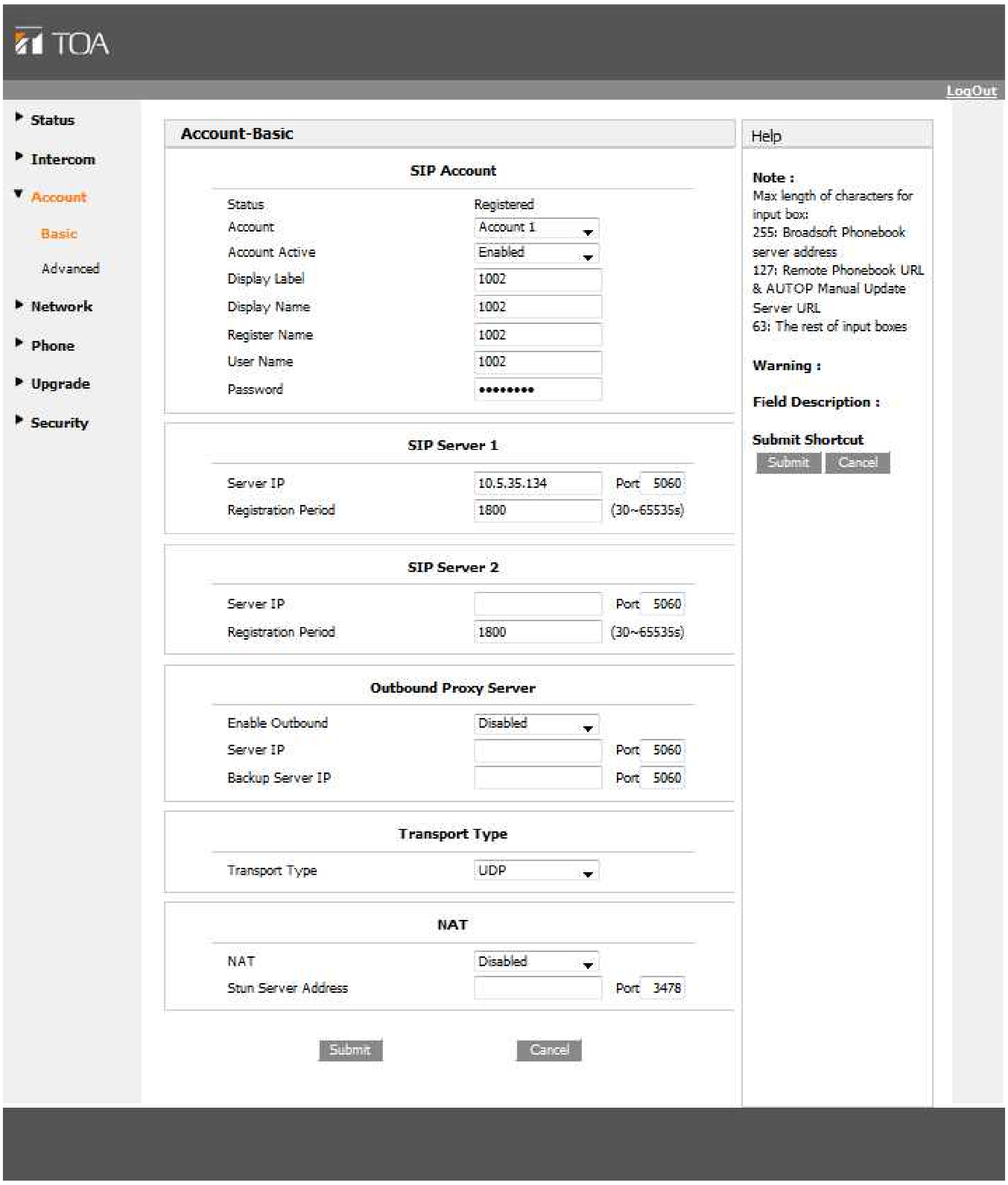


Sections	Description
Basic Setting	To setup the Onvif function parameters. It is used to connect with the corresponding Onvif tool. ● Onvif Mode: Two modes - Discoverable and Non-discoverable. Discoverable by default. Only Discoverable mode, then Onvif software can search N-SP80VS1. ● User Name: To modify the user name you need. Admin by default. ● Password: To modify the password you want. Admin by default.

4 Account

4.1 Account->Basic

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

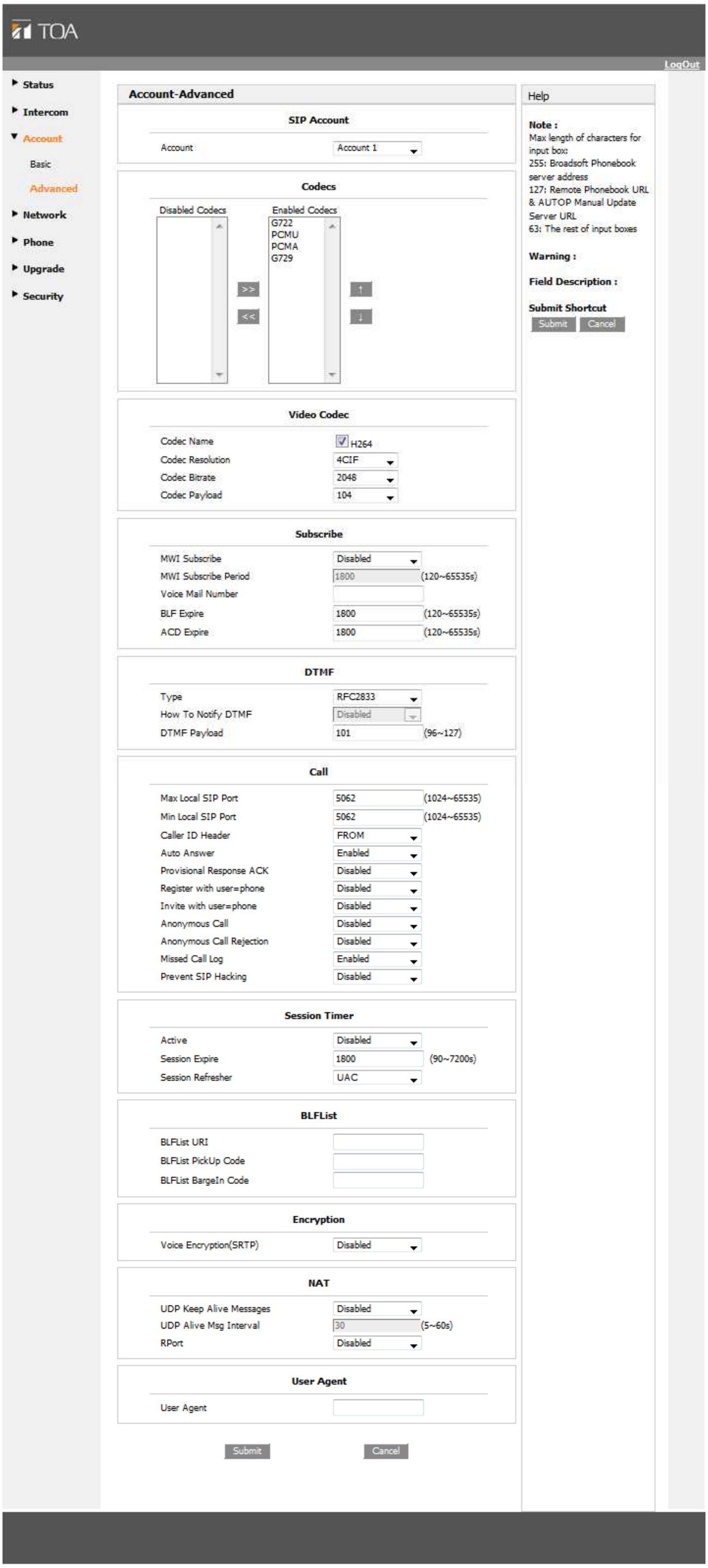


Sections	Description
SIP Account	To display and configure the specific Account settings. ● Status: To display register result. ● Display Name: Which is sent to the other call party for displaying. ● Register Name: Allocated by SIP server provider, used for authentication. ● User Name: Allocated by your 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.

4.2 Account-> Advanced

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


TOA
LogOut
Status
Intercom
Account
Basic
Advanced
Network
Phone
Upgrade
Security
Account-Advanced
SIP Account
Account
Account 1
Codecs
Disabled Codecs
Enabled Codecs
G722
PCMU
PCMA
G729
>>
<<
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 (wid-bandth codecs), G729 and so on.
Video Codec(optional)	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.

	● 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 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. **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

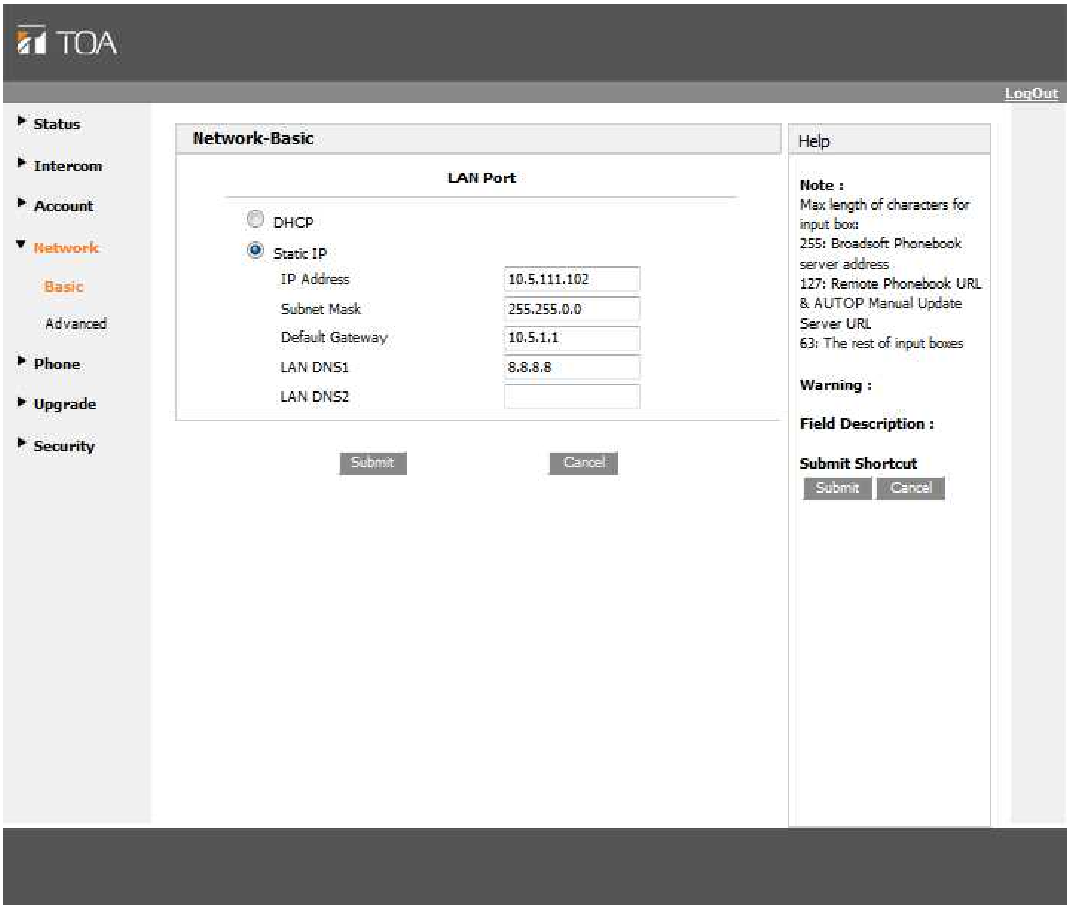
	user agent is set to specific value, user could see the information from PCAP. If user agent is not set by default, user could see the company name, model number and firmware version from PCAP

5 Network

5.1 Network-> Basic

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

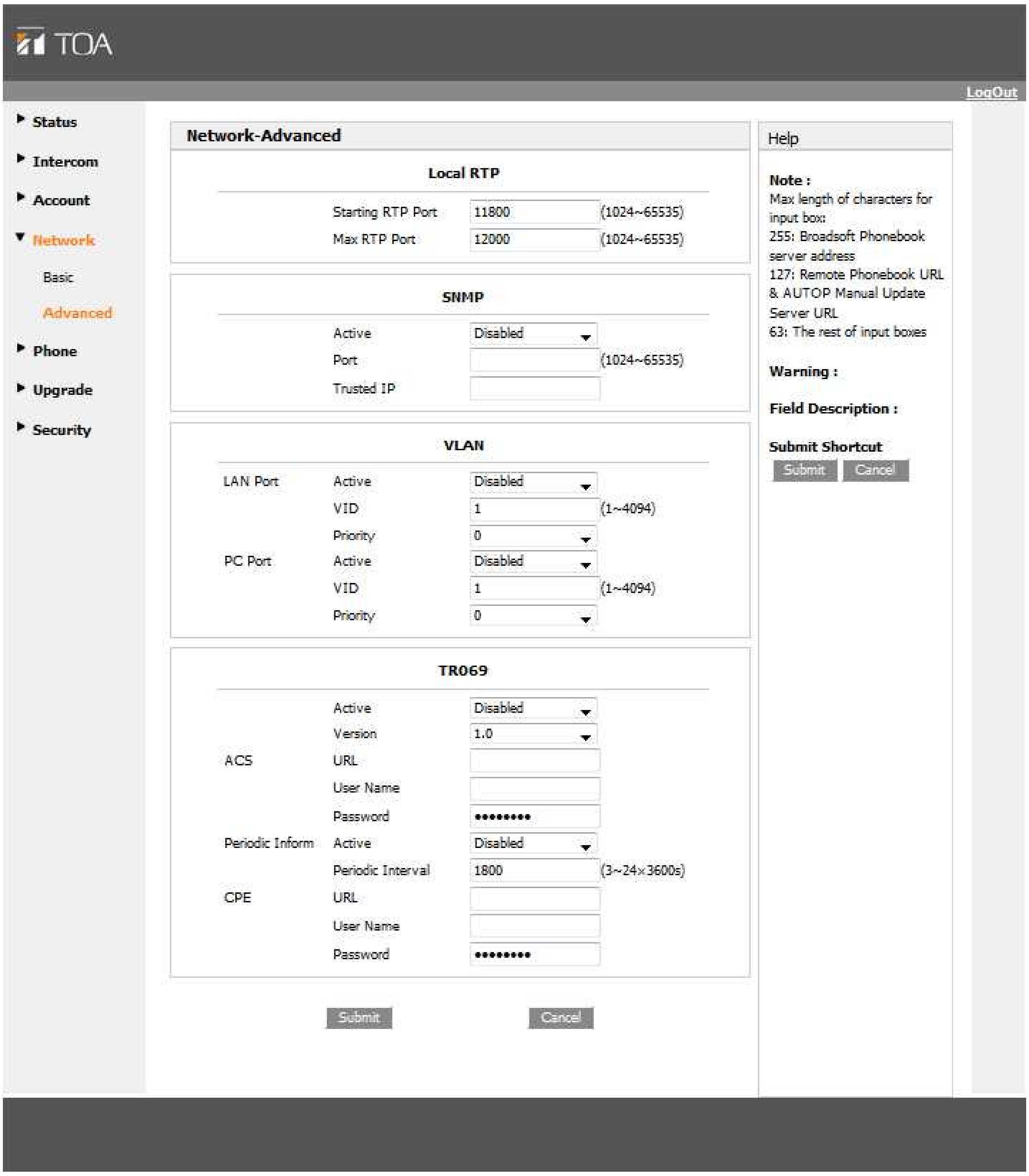
The static IP is set as default, and its IP address is 192.168.1.102.



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.

5.2 Network-> Advanced

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



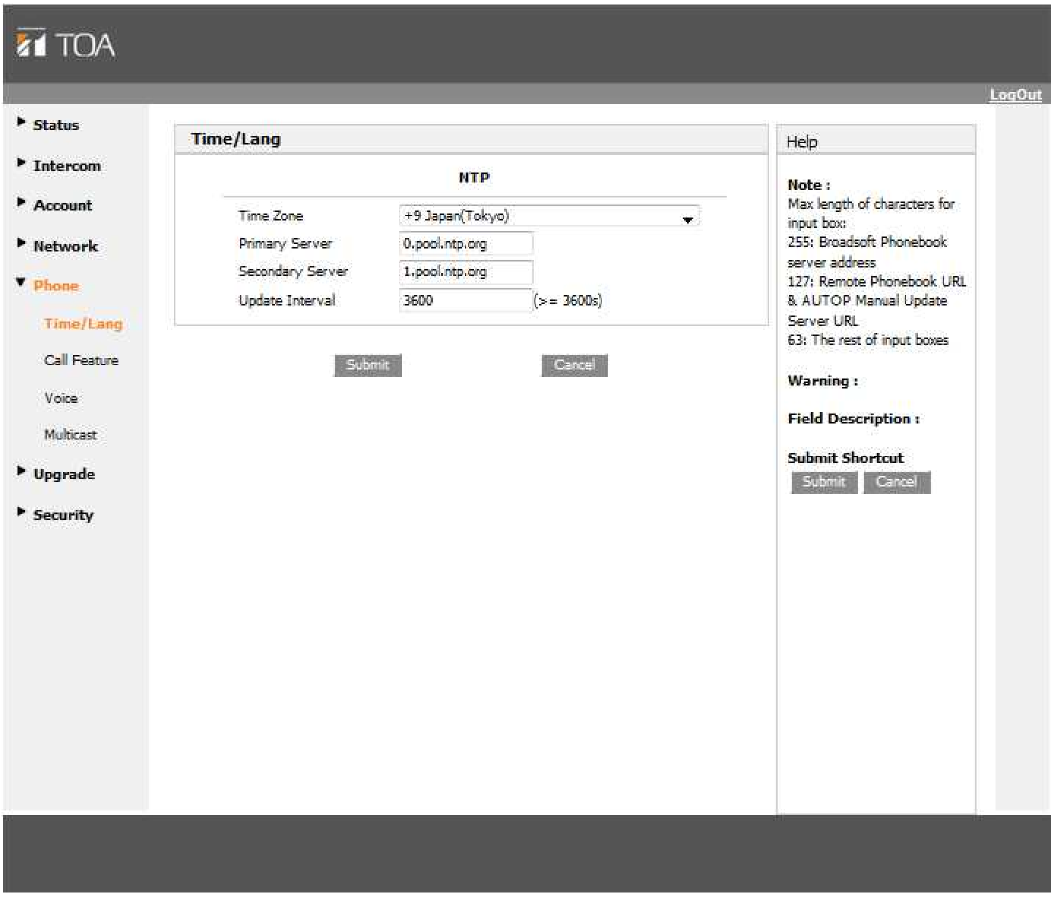
Sections	Description
Local RTP	To display and configure Local RTP settings. ● Max RTP Port: Determine the maximum port that RTP stream can use. ● Starting RTP Port: Determine the minimum port that RTP stream can use.
SNMP	To display and configure SNMP settings. ● Active: To enable or disable SNMP feature.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

6 Phone

6.1 Time/Language

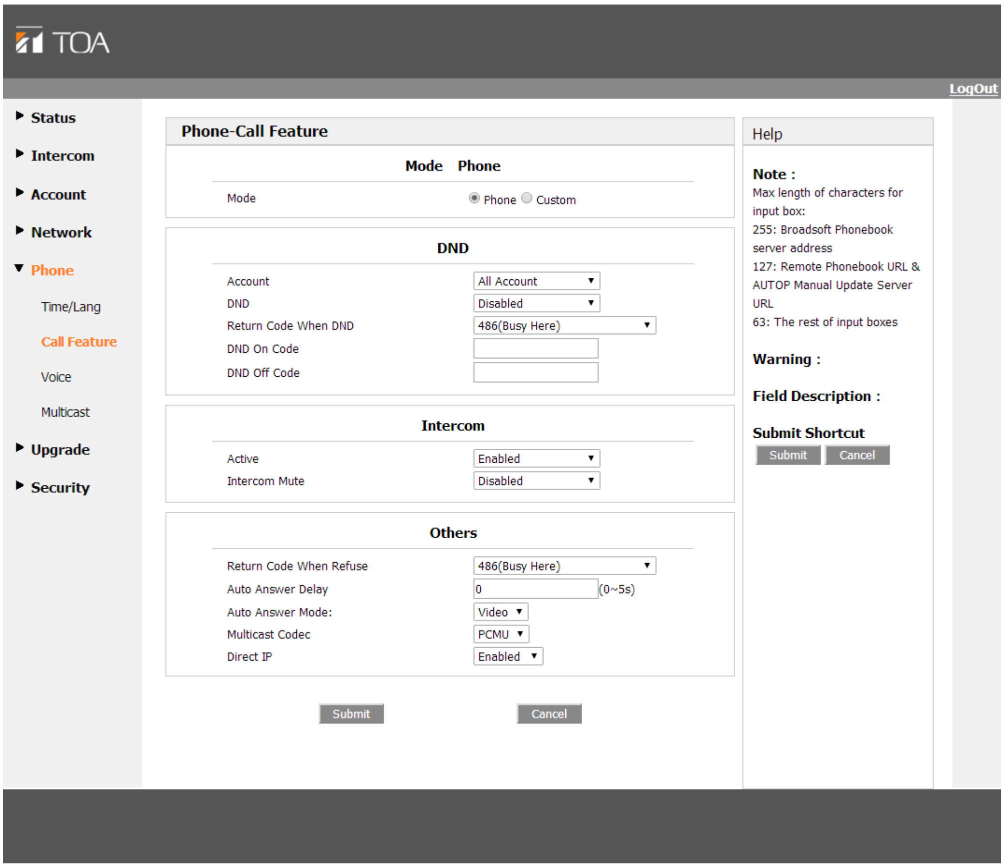
Go to the path: Phone-> Time/Language



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.

6.2 Call Feature

Go to the path: Phone->Call Feature

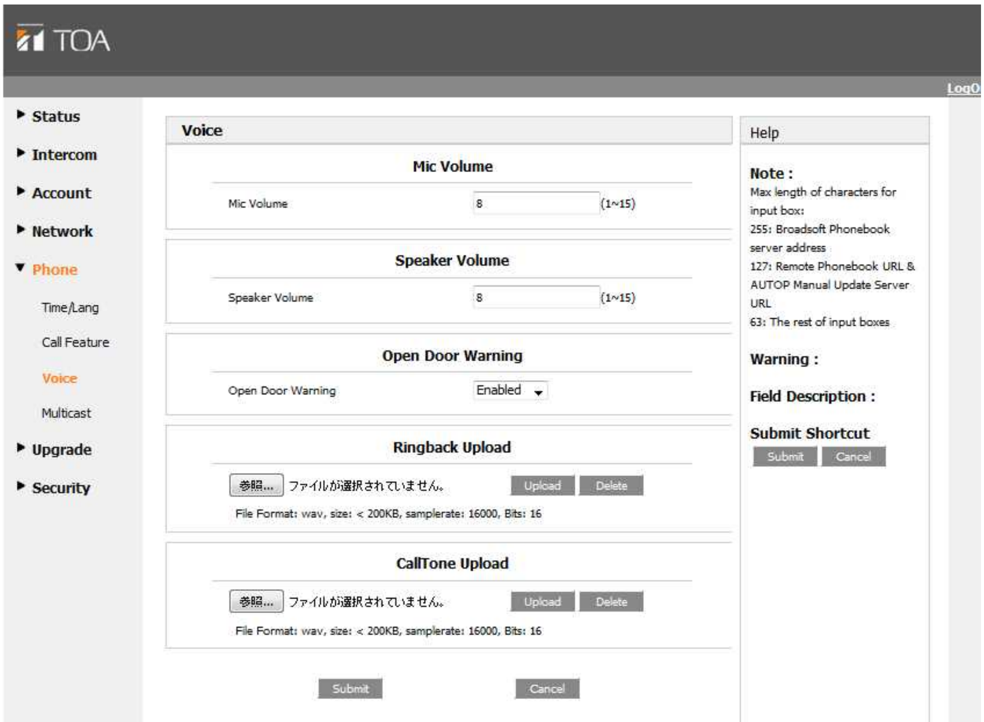


Sections	Description
Mode	To enable or disable feature key sync. ● Feature Key Sync: To enable or disable feature key sync. ● Mode: Select the desired mode.
DND	DND (Do Not Disturb) allows IP phones to ignore any incoming calls. ● Return Code when DND: Determine what response code should be sent back 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. ● Multicast Codec：Choose the suitable audio codec for multicast function. PCMU by default. ● Direct IP: Direct IP call without SIP proxy.

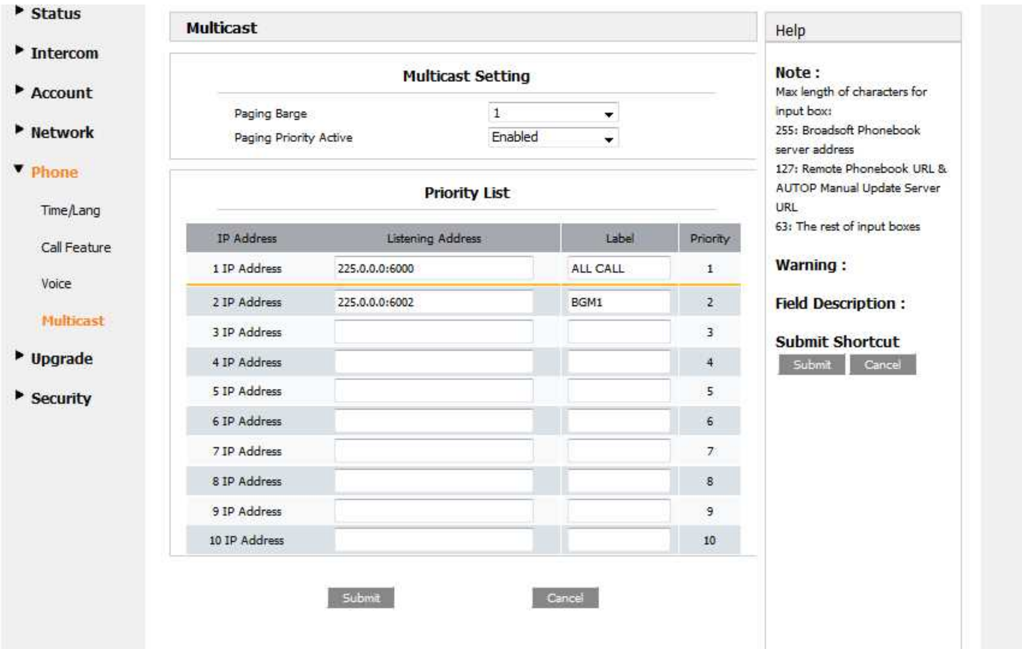
6.3 Voice

Go to the path: Phone->Voice



Sections	Description
Mic Volume	To configure Microphone volume
Speaker Volume	To configure Speaker Volume
Open Door Warning	To configure door opening voice. Disable it, you won't hear the prompt voice when the door is opened.
Ringback Upload	For a tone when N-SP80VS1/AS1 is pressed a call button.
CallTone Upload	For a tone when N-SP80VS1/AS1 is called from other device.

6.4 Multicast

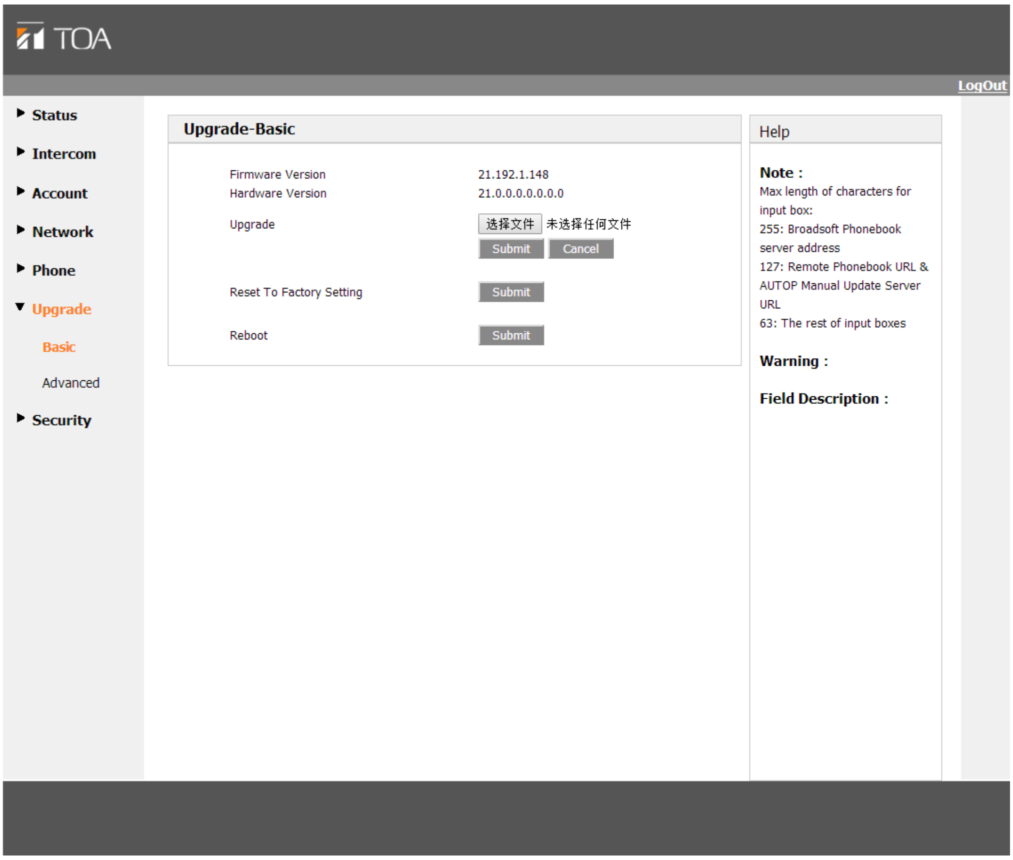


Sections	Description
Multicast Setting	To display and configure the Multicast setting. ● Paging Barge: Setup the priority level. A call is in a higher priority than a paging below this level. ● Paging priority Active: Enable or disable a priority between a call and a paging
Priority List	To setup the multicast parameters. ● Listening Address: Enter the IP address you need to listen ● Label： Input the label for each listening address

7 Upgrade

7.1 Basic Upgrade

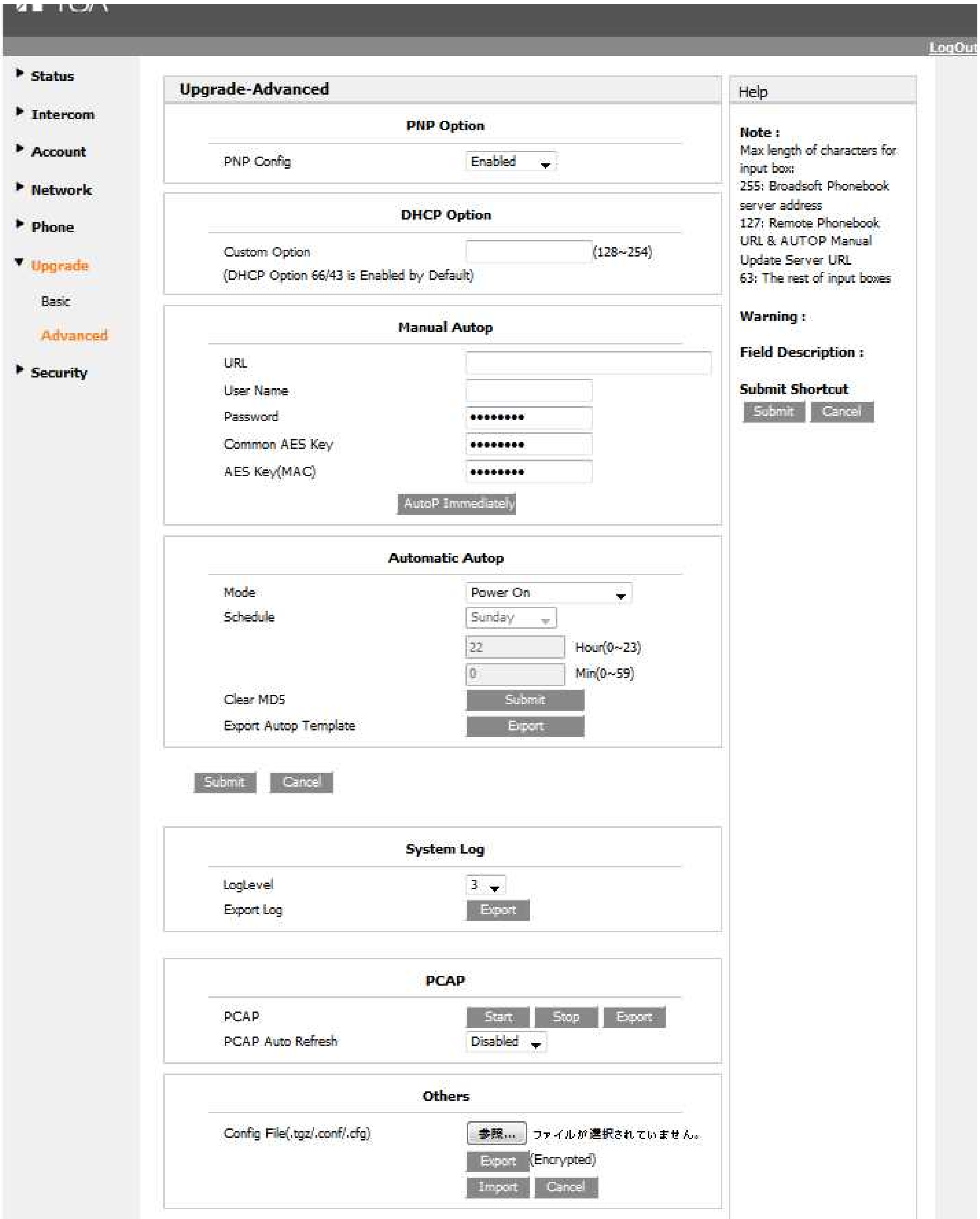
To upgrade your device, go to the path: Upgrade > Basic.



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.

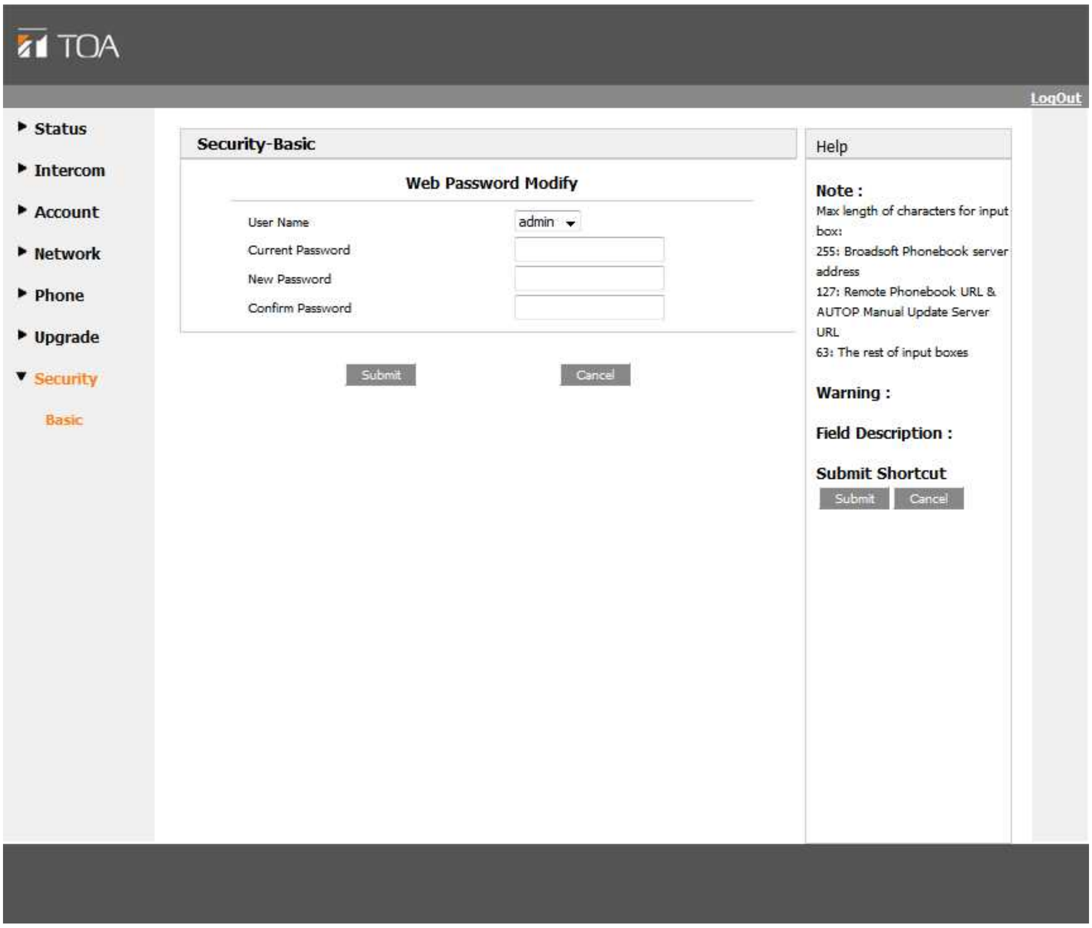


Sections	Description
PNP Option	To display and configure PNP setting for Auto Provisioning. ● PNP: Plug and Play, once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get Auto Provisioning server's address. By default, this SIP message is sent to multicast address

	224.0.1.75(PNP server address by standard).
DHCP Option	To display and configure custom DHCP option. ● DHCP option: If configured, IP Phone will use designated DHCP option to get Auto Provisioning server's address via DHCP. This setting require DHCP server to support corresponding option.
Manual Autop	To display and configure manual update server's settings. ● URL: Auto provisioning server address. ● User name: Configure if server needs an username to access, otherwise left blank. ● Password: Configure if server needs a password to access, otherwise left blank. ● Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file. ● AES Key (MAC): Used for IP phone to decipher MAC-oriented auto provisioning configuration file (for example, file name could be 0c1105888888.cfg if IP phone's MAC address is 0c1105888888). **Note:** AES is one of many encryption, it should be configure only configure filed is ciphered with AES, otherwise left blank.
Automatic AutoP	To display and configure Auto Provisioning mode settings. This Auto Provisioning mode is actually self-explanatory. For example, mode "Power on" means IP phone will go to do Provisioning every time it powers on.
System Log	To display system log level and export system log file. ● System log level: From level 0~7.The higher level means the more specific system log is saved to a temporary file. By default, it's level 3. ● Export Log: Click to export temporary system log file to local PC.
PCAP	To start,stop packets capturing or to export captured Packet file. ● Start:To start capturing all the packets file sent or received from IP phone. ● Stop:To stop capturing packets. **Note**:IP phone will save captured packets file to a temporary file,this file maximum size is 1M(mega bytes), and will top capturing once reaching this maximum size.
Others	To display or configure others features from this page. ● Config file: To export or import configure file for IP phone.

8 Security

To modify web password, go to the path: 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.