



**N-SP80AS1 SIP Video Door Station,  
N-SP80VS1 SIP Audio Door Station  
User Manual**

# Content

<b>Production Overview .....</b>	<b>3</b>
1 Production description.....	3
2 Features.....	4
3 Panel Description .....	5
4 Installation.....	6
<b>Configuration .....</b>	<b>8</b>
1 Web Login .....	8
1.1 Obaining the IP address .....	8
1.2 Login the web.....	8
2 Status.....	9
3 Intrecom.....	10
3.1 Basic .....	10
3.2 LED Settings.....	11
3.3 Relay&Input.....	12
3.4 Live Stream(Optional) .....	13
3.5 AEC Setting .....	14
3.6 RTSP(optional).....	15
3.7 Onvif(optional) .....	16
3.7 Multicast .....	16
4 Account .....	17
4.1 Account->Basic .....	17
4.2 Account-> Advanced .....	18
5 Network.....	22
5.1 Network-> Basic .....	22
5.2 Network-> Advanced.....	22
6 Phone .....	25
6.1 Time/Language .....	25
6.2 Call Feature .....	26
6.3 Voice.....	27
6.4 Multicast .....	28
7 Upgrade .....	29
7.1 Basic Upgrade.....	29
7.2 Advanced upgrade .....	30
8 Security.....	32

# Production Overview

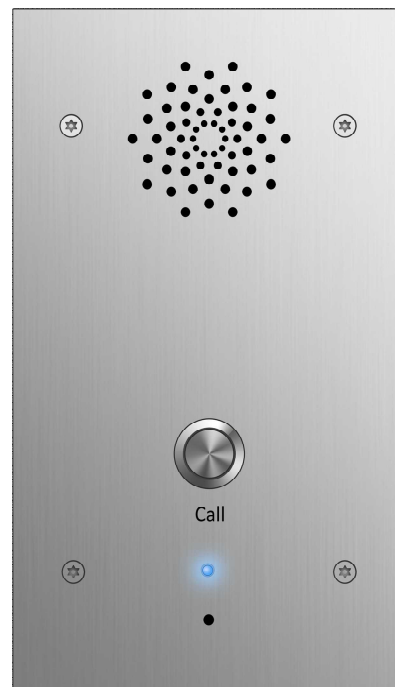
## 1 Production description

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

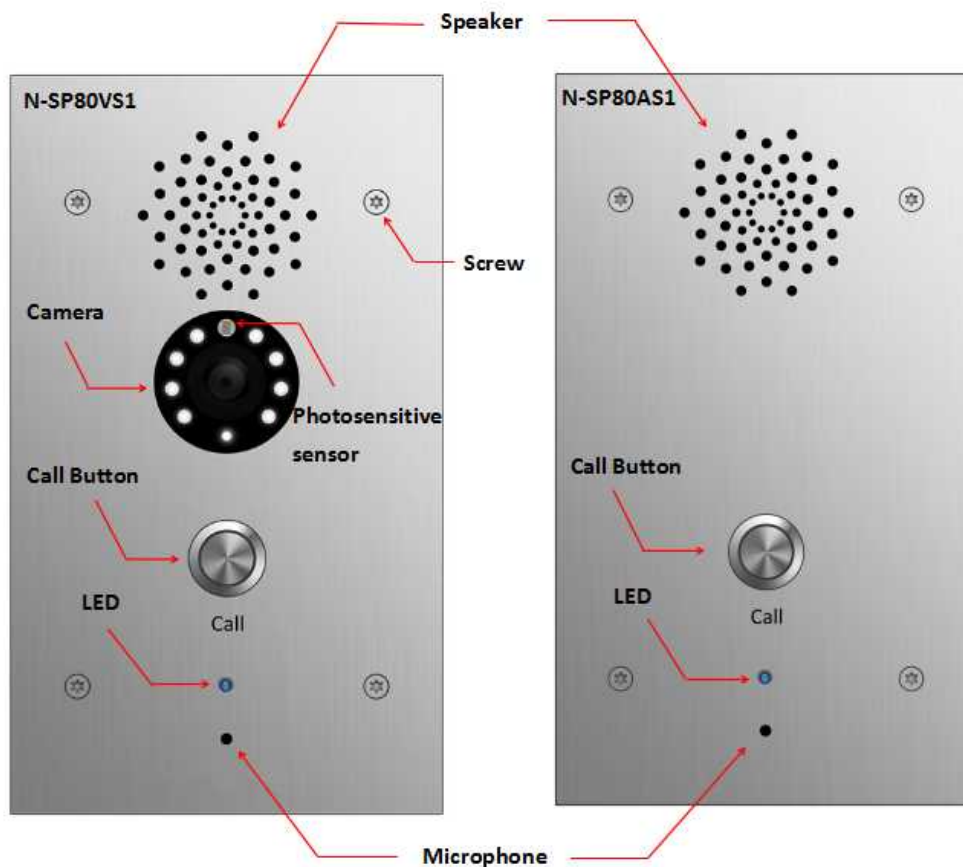
- 2x10/100Mbps Ethernet Port
- Security: Password Protection, IP address filtering, SIP over TLS, HTTPS encryption, user access log

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

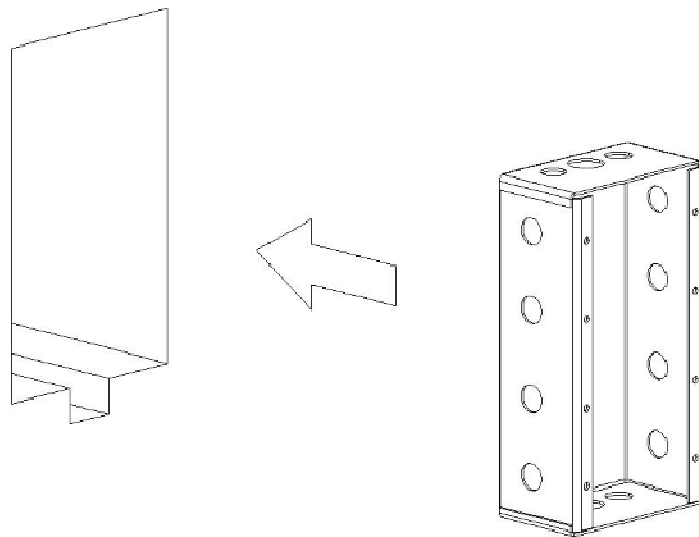
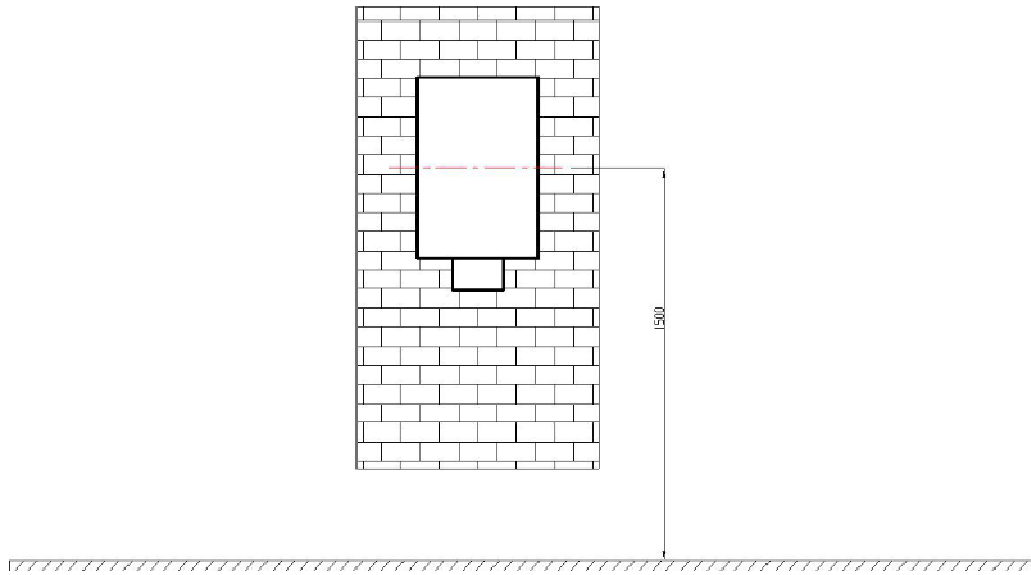
### ➤ SIP Features

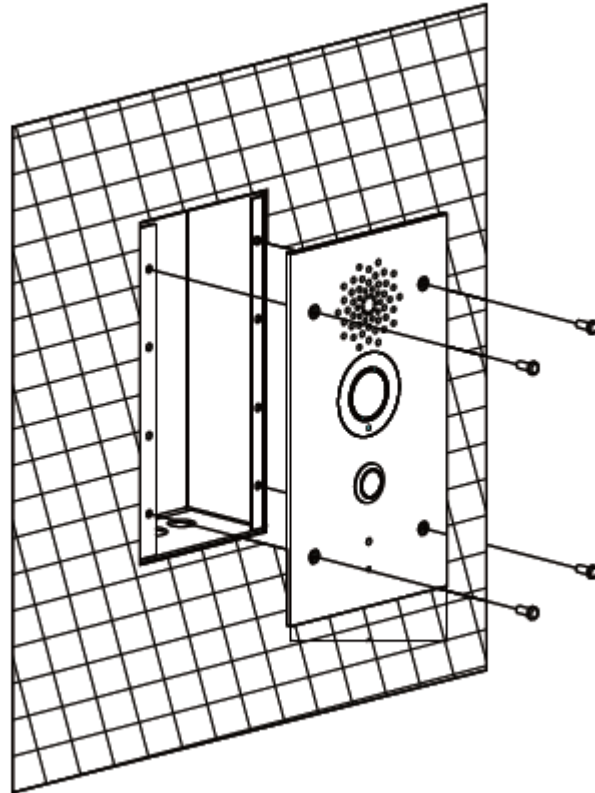
- SIP v1(RFC2543), SIP v2(RFC3261)
- Audio codecs: G.711a, G.711  $\mu$  , G.722, G.729
- Video codecs: MPEG-4/MJEG (N-SP80VS1 only)
- Speech Quality: 7kHz Audio
- Echo Cancellation
- Voice Activation Detection
- Comfort Noise Generator

## 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.

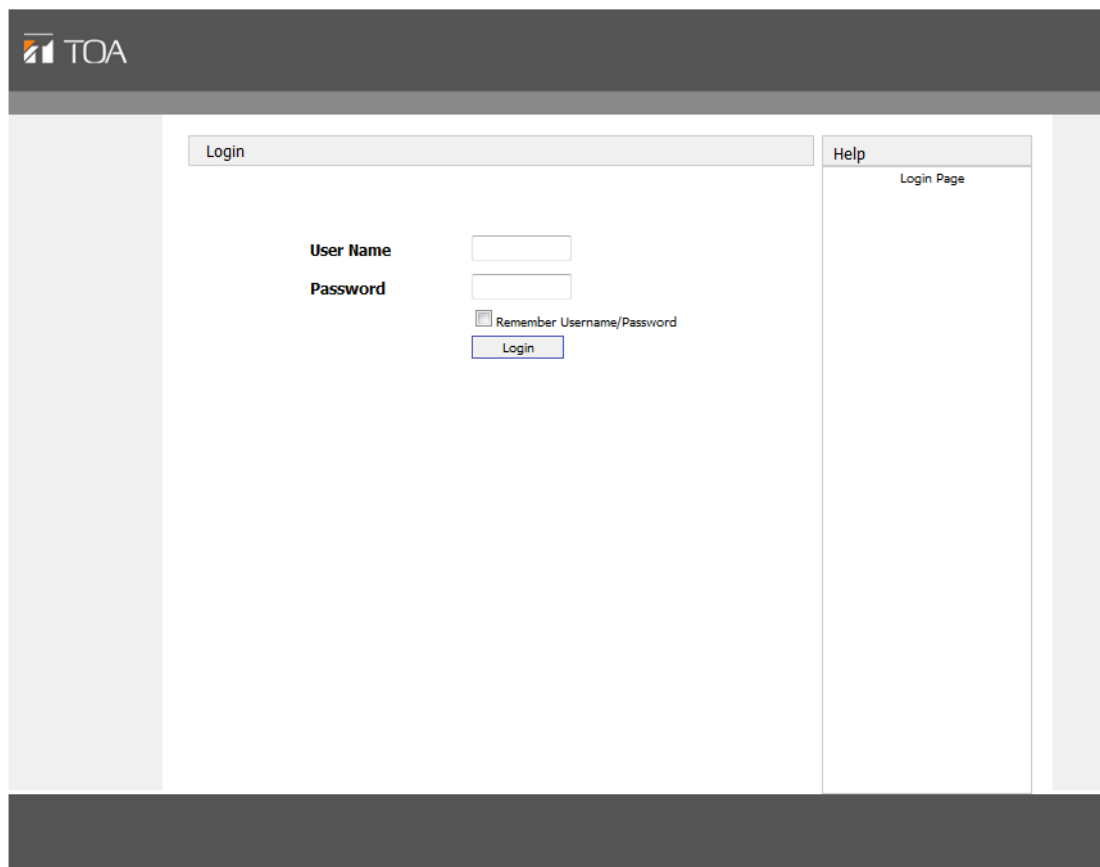
If the IP address is unknown, press the call button when LED light turns blue, after a short period of time(about 5s), the phone will announce its IP.

### 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



The screenshot shows the TOA web interface. At the top left is the TOA logo. Below it, there are two tabs: 'Login' and 'Help'. The 'Login' tab is active and contains a form with the following elements:

- User Name**: A text input field.
- Password**: A text input field.
- Remember Username/Password
- Login**: A button.

The 'Help' tab is also visible and contains the text 'Login Page'.



## 2 Status

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

The screenshot shows the TOA web interface. The top header includes the TOA logo and a 'LogOut' link. A sidebar on the left lists navigation options: Status (selected), Basic, Intercom, Account, Network, Phone, Upgrade, and Security. The main content area is titled 'Status' and contains three sections:

- Product Information:**

Model	N-SP80VS1
MAC Address	C4:09:38:D2:D9:CD
Firmware Version	21.192.1.148
Hardware Version	21.0.0.0.0.0.0
- Network Information:**

LAN Port Type	DHCP Auto
LAN Link Status	Connected
LAN IP Address	192.168.35.26
LAN Subnet Mask	255.255.255.0
LAN Gateway	192.168.35.1
LAN DNS1	8.8.8.8
LAN DNS2	
- Account Information:**

Account1	316@pbx.akuvox.com
	Registered
Account2	None@None
	Disabled

On the right side, there 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 :**

Sections	Description
<b>Product Information</b>	To display the device's information such as Model name, MAC address (IP device's physical address), Firmware version and Hardware firmware.
<b>Network Information</b>	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).
<b>Account Information</b>	To display device's Account information and Registration status (account username, registered server's address, Register result).

## 3 Intercom

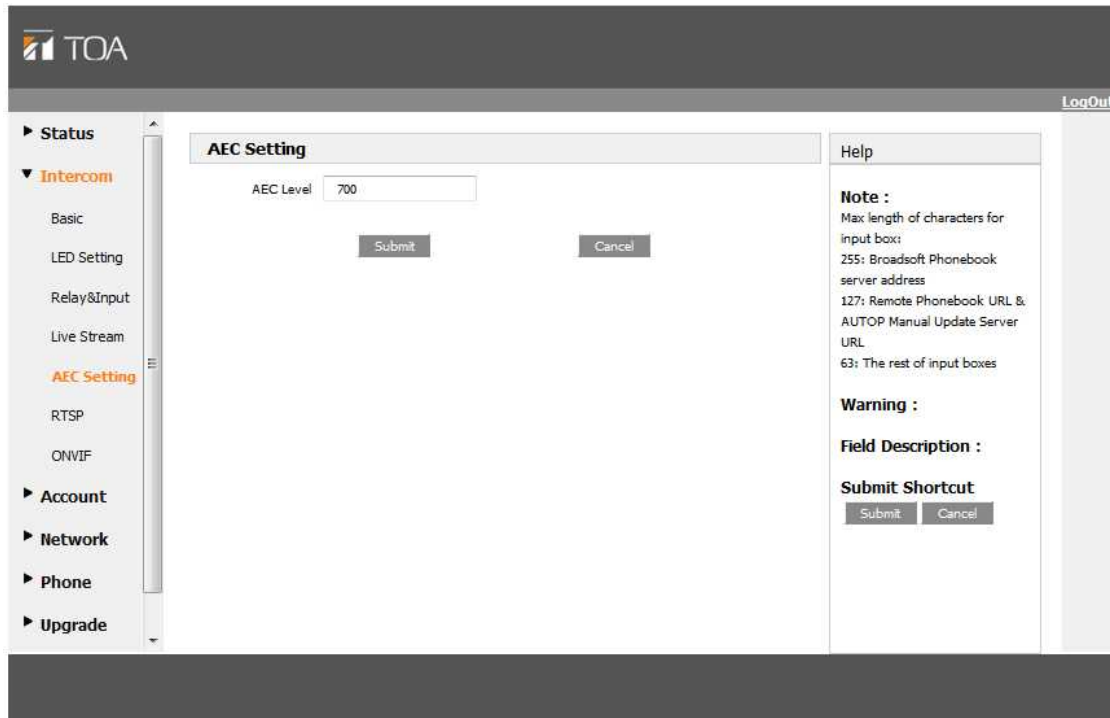
### 3.1 Basic

The screenshot displays the TOA Intercom Basic settings interface. On the left is a navigation menu with categories like Status, Intercom, Account, Network, Phone, Upgrade, and Security. The main content area is titled 'Intercom-Basic' and contains several sections: 'Account Selection' with dropdowns for 'Select Account' (set to 'Auto') and 'No Answer Call' (set to 'Disabled'); 'Push Button' with a table for 'Key' and 'Number' (Push Button: 192.168.111.70); 'Web Call' with a 'Web Call(Ready)' dropdown (set to 'Auto') and 'Dial Out'/'Hang Up' buttons; 'Max Call Time' with a text input (5) and '(2~30Minutes)'; 'Push To Hang Up' with a dropdown (set to 'Enabled'); and 'Custom Button' with 'Apply setting to' (set to 'RelayA'). A 'Help' sidebar on the right provides a 'Note' about character lengths, a 'Warning' about URLs, and a 'Field Description' for 'Submit Shortcut' with 'Submit' and 'Cancel' buttons. 'Submit' and 'Cancel' buttons are also at the bottom of the main form.

Sections	Description
<b>Account Selection</b>	<ul style="list-style-type: none"> <li>Select Account: N-SP80VS1/AS1 supports 2 accounts. You can choose one account or Auto mode for the following Intercom basic settings.</li> <li>No Answer Call: Choosing Enabled is for 3 No Answer Calls. 1<sup>st</sup> call is to the cell of "Push Button", 2<sup>nd</sup> is "No Answer Call1", 3<sup>rd</sup> is "No Answer Call2".</li> </ul>
<b>Push Button</b>	<ul style="list-style-type: none"> <li>Push Button: To configure the destination number or IP you want to contact with. No Answer Call1 is for the 2<sup>nd</sup> call when the destination number doesn't answer its call. No Answer Call2 is the next of Call1.</li> </ul>
<b>Web Call</b>	To dial out or answer the phone from website.
<b>Max Call Time</b>	To configure the max call time
<b>Push to Hang up</b>	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
<b>States</b>	There is five states: Normal, Offline, Calling, Talking and Receiving.
<b>Color Off</b>	The default status is OFF
<b>Color On</b>	It can support three color: Red, Green, Blue
<b>Blink Mode</b>	To setup the different blink frequency.

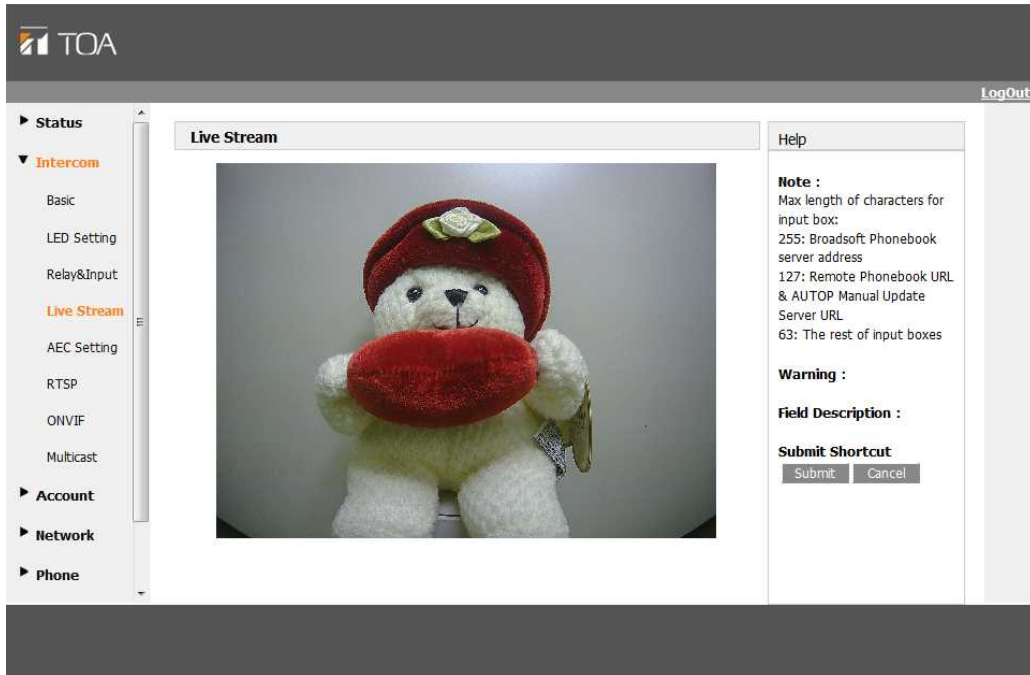
### 3.3 Relay&Input

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

Sections	Description
<b>Relay</b>	<p>To configure some settings about unlock</p> <ul style="list-style-type: none"> <li>● Relay ID: N-SP80VS1/AS1 support 2 relays</li> <li>● Relay Type: Different locks use different relay types.</li> <li>● Delay(s): Allows door remain “open” for certain period The range is from 1 to 5 seconds</li> <li>● DTMF: Setup DTMF code for remote unlock</li> <li>● Status: Different relay type will show different status.</li> </ul>
<b>Input</b>	<p>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.</p> <ul style="list-style-type: none"> <li>● Input ID: N-SP80VS1/AS1 supports 2 optical-couplers. Once the optical-coupler is triggered, it will alarm when this function is enabled.</li> <li>● Input Service: Disable by default</li> <li>● Call Number: To setup management center number for alarm.</li> </ul>

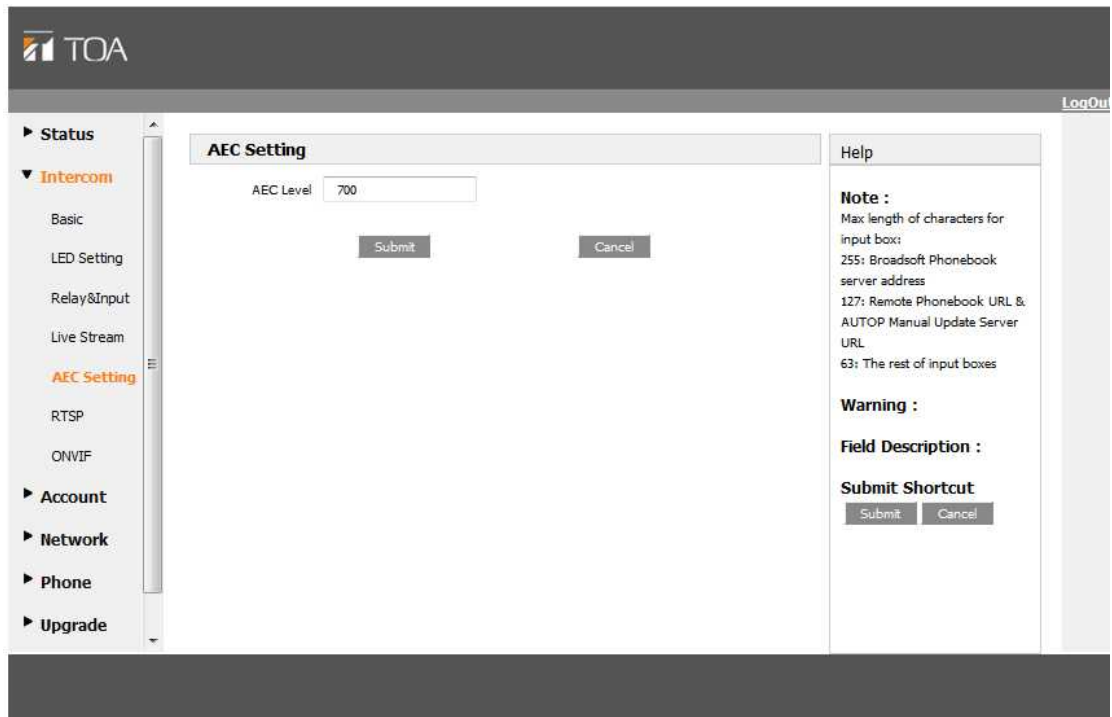
	<ul style="list-style-type: none"> <li>● Display Name: Which is sent to the other call party for displaying</li> <li>● Call Timer: Every its seconds makes call during the input is activated.</li> <li>● Light Status: Here is an indication of a status of input.</li> </ul>
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### 3.4 Live Stream(Optional)



Sections	Description
<b>Live Stream</b>	To check the real-time video from N-SP80VS1.

### 3.5 AEC Setting



Sections	Description
<b>AEC Level</b>	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)

The screenshot shows the TOA web interface for RTSP configuration. The sidebar on the left contains the following menu items: Status, Intercom (Basic, LED Setting, Relay&Input, Live Stream, AEC Setting), RTSP (ONVIF), Account, Network, Phone, Upgrade, and Security. The main content area is titled 'RTSP' and contains several sections:

- RTSP Basic:** RTSP Server Enabled (checkbox, currently unchecked).
- RTSP Stream:** RTSP Video Enabled (checkbox, checked), RTSP Video Codec (dropdown menu, H.264 selected).
- H.264 Video Parameters:** Video Resolution (dropdown menu, VGA selected), Video Framerate (dropdown menu, 30 fps selected), Video Bitrate (dropdown menu, 2048 kbps selected).
- MPEG4 Video Parameters:** Video Resolution (dropdown menu, VGA selected), Video Framerate (dropdown menu, 30 fps selected), Video Bitrate (dropdown menu, 2048 kbps selected).
- MJPEG Video Parameters:** Video Resolution (dropdown menu, VGA selected), Video Framerate (dropdown menu, 30 fps selected), Video Quality (dropdown menu, 90 selected).

At the bottom of the main content area are 'Submit' and 'Cancel' buttons. On the right side, there 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 :**

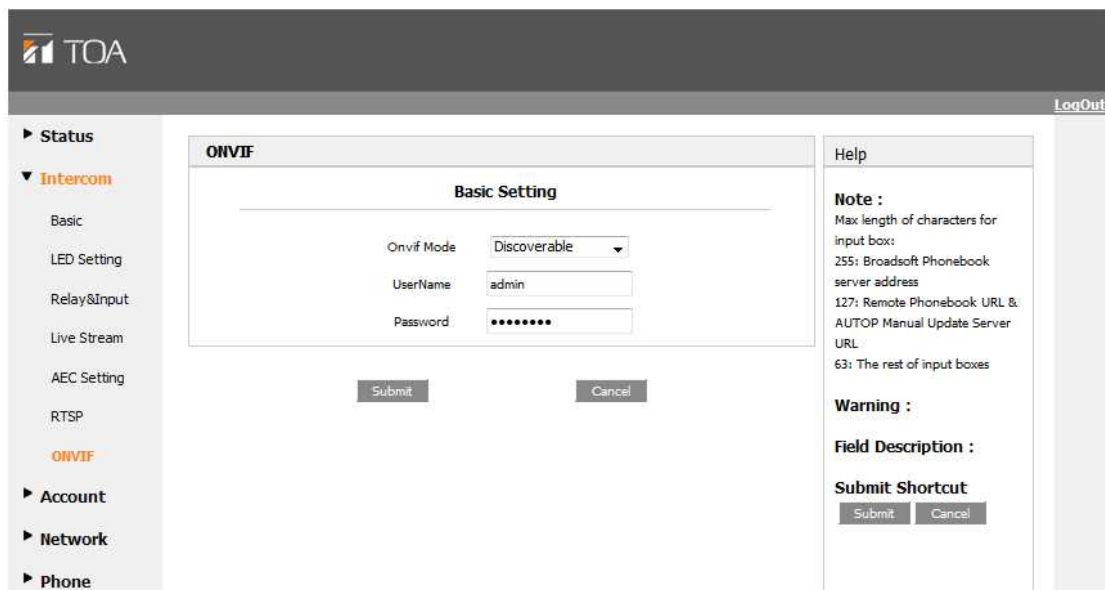
**Field Description :**

**Submit Shortcut**  
Submit Cancel

Sections	Description
<b>RTSP Basic</b>	To active the RTSP function, then N-SP80VS1 can be monitored.
<b>RTSP Stream</b>	To enabled RTSP video and select the video codec. N-SP80VS1 supports H264,H263 video codec. H264 by default.
<b>H.264 Video Parameters</b>	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
<b>MPEG4 Video Parameters</b>	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
<b>MJPEG Video Parameters</b>	<p>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</p> <p>To modify the resolution, framerate and bitrate of MJPEG</p>

### 3.7 Onvif(optional)



Sections	Description
<b>Basic Setting</b>	<p>To setup the Onvif function parameters. It is used to connect with the corresponding Onvif tool.</p> <ul style="list-style-type: none"> <li>● Onvif Mode: Two modes - Discoverable and Non-discoverable. Discoverable by default. Only Discoverable mode, then Onvif software can search N-SP80VS1.</li> <li>● User Name: To modify the user name you need. Admin by default.</li> <li>● Password: To modify the password you want. Admin by default.</li> </ul>



## 4 Account

### 4.1 Account->Basic

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

The screenshot displays the TOA web interface for configuring a SIP Account. The main content area is titled 'Account-Basic' and contains several sections:

- SIP Account:** Fields for Status (Registered), Account (Account 1), Account Active (Enabled), Display Label (1002), Display Name (1002), Register Name (1002), User Name (1002), and Password (masked).
- SIP Server 1:** Fields for Server IP (10.5.35.134), Port (5060), and Registration Period (1800).
- SIP Server 2:** Fields for Server IP (empty), Port (5060), and Registration Period (1800).
- Outbound Proxy Server:** Fields for Enable Outbound (Disabled), Server IP (empty), Port (5060), Backup Server IP (empty), and Port (5060).
- Transport Type:** Field for Transport Type (UDP).
- NAT:** Fields for NAT (Disabled) and Stun Server Address (empty), Port (3478).

At the bottom of the main form are 'Submit' and 'Cancel' buttons. The right sidebar contains a 'Help' section with a 'Note' and a 'Warning', and a 'Submit Shortcut' section with 'Submit' and 'Cancel' buttons.

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

	<p>for authentication.</p> <ul style="list-style-type: none"> <li>● Password: Used for authorization.</li> </ul>
<b>SIP Server 1</b>	<p>To display and configure Primary SIP server settings.</p> <ul style="list-style-type: none"> <li>● Server IP: SIP server address, it could be an URL or IP address.</li> <li>● Registration Period: The registration will expire after Registration period, the IP phone will re-register automatically within registration period.</li> </ul>
<b>SIP Server 2</b>	<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><b>Note:</b> Secondary SIP server is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.</p>
<b>Outbound Proxy Server</b>	<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><b>Note:</b> If configured, all SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.</p>
<b>Transport Type</b>	<p>To display and configure Transport type for SIP message</p> <ul style="list-style-type: none"> <li>● UDP: UDP is an unreliable but very efficient transport layer protocol.</li> <li>● TCP: Reliable but less-efficient transport layer protocol.</li> <li>● TLS: Secured and Reliable transport layer protocol.</li> <li>● DNS-SRV: A DNS RR for specifying the location of services.</li> </ul>
<b>NAT</b>	<p>To display and configure NAT(Net Address Translator) settings.</p> <ul style="list-style-type: none"> <li>● STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues.</li> </ul> <p><b>Note:</b> 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
	<input type="button" value="&gt;&gt;"/> <input type="button" value="&lt;&lt;"/>	G722 PCMU PCMA G729

**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 Bargain 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
<b>SIP Account</b>	To display current Account settings or to select which account to display.
<b>Codecs</b>	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.
<b>Video Codec(optional)</b>	To configure the video quality <ul style="list-style-type: none"> <li>● Codec Name: The default video codec is H264.</li> <li>● Codec Resolution: It can support QCIF, CIF, VGA, 4CIF, 720P.</li> <li>● Codec Bitrate: The lowest bitrate is 128, the highest bitrate is 2048.</li> <li>● Codec payload: From 90-119.</li> </ul>
<b>Subscribe</b>	To display and configure MWI, BLF, ACD subscription settings. <ul style="list-style-type: none"> <li>● MWI: Message Waiting Indicator which is used to indicate whether there is unread new voice message.</li> <li>● BLF: BLF is short for Busy Lamp Field which is used to monitor the designated extension status.</li> <li>● 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.</li> </ul>
<b>DTMF</b>	To display and configure DTMF settings. <ul style="list-style-type: none"> <li>● Type: Support Inband, Info, RFC2833 or their combination.</li> <li>● How To Notify DTMF: Only available when DTMF Type is Info.</li> <li>● DTMF Payload: To configure payload type for DTMF.</li> </ul> <p><b>Note:</b> 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>
<b>Call</b>	To display and configure call-related features. <ul style="list-style-type: none"> <li>● Max Local SIP Port: To configure maximum local sip port for designated account.</li> <li>● Min Local SIP Port: To configure minimum local sip port for designated account.</li> <li>● Caller ID Header: To configure which Caller ID format to fetch for displaying on Phone UI.</li> <li>● Auto Answer: If enabled, IP phone will be auto-answered when there is an incoming call for designated account.</li> </ul>

	<ul style="list-style-type: none"> <li>● 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.</li> <li>● User=phone: If enabled, IP phone will send user=phone within SIP message.</li> <li>● PTime: Interval time between two consecutive RTP packets.</li> <li>● Anonymous Call: If enabled, all outgoing call for the designated account will be anonymous number.</li> <li>● Anonymous Call Rejection: If enabled, all incoming anonymous-out call for the designated account will be rejected.</li> <li>● Is escape non Ascii character: To transfer the symbol to Ascii character.</li> <li>● Missed Call Log: To display the miss call log.</li> <li>● Prevent SIP Hacking: Enable to prevent SIP from hacking.</li> </ul>
<b>Session Timer</b>	<p>To display or configure session timer settings.</p> <ul style="list-style-type: none"> <li>● 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.</li> <li>● Session Expire: Configure session expire time.</li> <li>● Session Refresher: To configure who should be response for refreshing a session.</li> </ul> <p><b>Note:</b> UAC means User Agent Client, here stands for IP phone. UAS means User Agent Server, here stands for SIP server.</p>
<b>BLF List</b>	<p>To display or configure BLF List URI address.</p> <ul style="list-style-type: none"> <li>● BLF List URI: BLF List is short for Busy Lamp Field List.</li> <li>● BLFList Pickup Code: To set the BLF pick up code.</li> <li>● BLFList BargelIn Code: To set the BLF barge in code.</li> </ul>
<b>Encryption</b>	<p>To enable or disabled SRTP feature.</p> <ul style="list-style-type: none"> <li>● Voice Encryption(SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.</li> </ul>
<b>NAT</b>	<p>To display NAT-related settings.</p> <ul style="list-style-type: none"> <li>● UDP Keep Alive message: If enabled, IP phone will send UDP keep-alive message periodically to router to keep NAT port alive.</li> <li>● UDP Alive Msg Interval: Keepalive message interval.</li> <li>● Rport: Remote Port, if enabled, it will add Remote Port into outgoing SIP message for designated account.</li> </ul>
<b>User Agent</b>	<p>One can customize User Agent field in the SIP message; If</p>

	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.

The screenshot shows the TOA Network-Basic configuration interface. The 'LAN Port' section is active, with 'Static IP' selected. The IP address is 10.5.111.102, Subnet Mask is 255.255.0.0, Default Gateway is 10.5.1.1, LAN DNS1 is 8.8.8.8, and LAN DNS2 is empty. The interface includes 'Submit' and 'Cancel' buttons. A 'Help' section on the right contains a 'Note' about character lengths and a 'Warning' section.

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

### 5.2 Network-> Advanced

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

The screenshot shows the TOA Network-Advanced configuration interface. It includes a sidebar with navigation options like Status, Intercom, Account, Network (Basic and Advanced), Phone, Upgrade, and Security. The main content area is divided into sections: Local RTP, SNMP, VLAN, and TR069. Each section contains various configuration fields such as ports, active status, and credentials. A help section on the right provides additional information and warnings.

Sections	Description
<b>Local RTP</b>	<p>To display and configure Local RTP settings.</p> <ul style="list-style-type: none"> <li>● Max RTP Port: Determine the maximum port that RTP stream can use.</li> <li>● Starting RTP Port: Determine the minimum port that RTP stream can use.</li> </ul>
<b>SNMP</b>	<p>To display and configure SNMP settings.</p> <ul style="list-style-type: none"> <li>● Active: To enable or disable SNMP feature.</li> <li>● Port: To configure SNMP server's port.</li> <li>● Trusted IP: To configure allowed SNMP server address, it could be an IP address or any valid URL domain name.</li> </ul> <p><b>Note:</b> SNMP (Simple Network Management Protocols) is</p>

	Internet-standard protocol for managing devices on IP networks.
<b>TR069</b>	<p>To display and configure TR069 settings.</p> <ul style="list-style-type: none"> <li>● Active: To enable or disable TR069 feature.</li> <li>● Version: To select supported TR069 version (version 1.0 or 1.1).</li> <li>● ACS/CPE: ACS is short for Auto configuration servers as server side, CPE is short for Customer-premise equipment as client side devices.</li> <li>● URL: To configure URL address for ACS or CPE.</li> <li>● User name: To configure username for ACS or CPE.</li> <li>● Password: To configure Password for ACS or CPE.</li> <li>● Periodic Inform: To enable periodically inform.</li> <li>● Periodic Interval: To configure interval for periodic inform.</li> </ul> <p><b>Note:</b> 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 the TOA web interface for configuring NTP settings. The main configuration area is titled "Time/Lang" and contains a sub-section for "NTP". The fields are as follows:

- Time Zone: +9 Japan(Tokyo)
- Primary Server: 0.pool.ntp.org
- Secondary Server: 1.pool.ntp.org
- Update Interval: 3600 (>= 3600s)

Buttons for "Submit" and "Cancel" are located below the form. To the right, a "Help" sidebar contains the following information:

- 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"> <li>● Time Zone: To select local Time Zone for NTP server.</li> <li>● Primary Server: To configure primary NTP server address.</li> <li>● Secondary Server: To configure secondary NTP server address, it takes effect if primary NTP server is unreachable.</li> <li>● Update interval: To configure interval between two consecutive NTP requests.</li> </ul> <p><b>Note:</b> 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
<b>Mode</b>	<p>To enable or disable feature key sync.</p> <ul style="list-style-type: none"> <li>● Feature Key Sync: To enable or disable feature key sync.</li> <li>● Mode: Select the desired mode.</li> </ul>
<b>DND</b>	<p>DND (Do Not Disturb) allows IP phones to ignore any incoming calls.</p> <ul style="list-style-type: none"> <li>● Return Code when DND: Determine what response code should be sent back to server when there is an incoming call if DND on.</li> <li>● 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.</li> <li>● 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.</li> </ul>
<b>Intercom</b>	<p>Intercom allows user to establish a call directly with the</p>

	<p>callee.</p> <ul style="list-style-type: none"> <li>● Active: To enable or disable Intercom feature.</li> <li>● Intercom Mute: If enabled, once the call established, the callee will be muted.</li> </ul>
<b>Others</b>	<ul style="list-style-type: none"> <li>● Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected.</li> <li>● Auto Answer Delay: To configure delay time before an incoming call is automatically answered.</li> <li>● Auto Answer Mode: To set video or audio mode for auto answer by default.</li> <li>● Multicast Codec: Choose the suitable audio codec for multicast function. PCMU by default.</li> <li>● Direct IP: Direct IP call without SIP proxy.</li> </ul>

### 6.3 Voice

Go to the path: Phone->Voice

The screenshot shows the TOA web interface for the 'Voice' configuration. The left sidebar contains navigation options: Status, Intercom, Account, Network, Phone (selected), Time/Lang, Call Feature, Voice (selected), Multicast, Upgrade, and Security. The main content area is titled 'Voice' and contains several sections:

- Mic Volume:** A slider set to 8, with a range of 1 to 15.
- Speaker Volume:** A slider set to 8, with a range of 1 to 15.
- Open Door Warning:** A dropdown menu set to 'Enabled'.
- Ringback Upload:** A file upload section with a '参照...' button, a message 'ファイルが選択されていません。', and 'Upload' and 'Delete' buttons. Below it, it specifies 'File Format: wav, size: < 200KB, samplerate: 16000, Bits: 16'.
- CallTone Upload:** A file upload section with a '参照...' button, a message 'ファイルが選択されていません。', and 'Upload' and 'Delete' buttons. Below it, it specifies 'File Format: wav, size: < 200KB, samplerate: 16000, Bits: 16'.

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' section, and a 'Submit Shortcut' section with 'Submit' and 'Cancel' buttons. At the bottom of the main content area, there are 'Submit' and 'Cancel' buttons.

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

## 6.4 Multicast

- ▶ Status
- ▶ Intercom
- ▶ Account
- ▶ Network
- ▼ **Phone**
  - Time/Lang
  - Call Feature
  - Voice
  - Multicast**
  - ▶ Upgrade
  - ▶ Security

**Multicast**

**Multicast Setting**

Paging Barge:

Paging Priority Active:

**Priority List**

IP Address	Listening Address	Label	Priority
1 IP Address	<input type="text" value="225.0.0.0:6000"/>	<input type="text" value="ALL CALL"/>	1
2 IP Address	<input type="text" value="225.0.0.0:6002"/>	<input type="text" value="BGM1"/>	2
3 IP Address	<input type="text"/>	<input type="text"/>	3
4 IP Address	<input type="text"/>	<input type="text"/>	4
5 IP Address	<input type="text"/>	<input type="text"/>	5
6 IP Address	<input type="text"/>	<input type="text"/>	6
7 IP Address	<input type="text"/>	<input type="text"/>	7
8 IP Address	<input type="text"/>	<input type="text"/>	8
9 IP Address	<input type="text"/>	<input type="text"/>	9
10 IP Address	<input type="text"/>	<input type="text"/>	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 :**

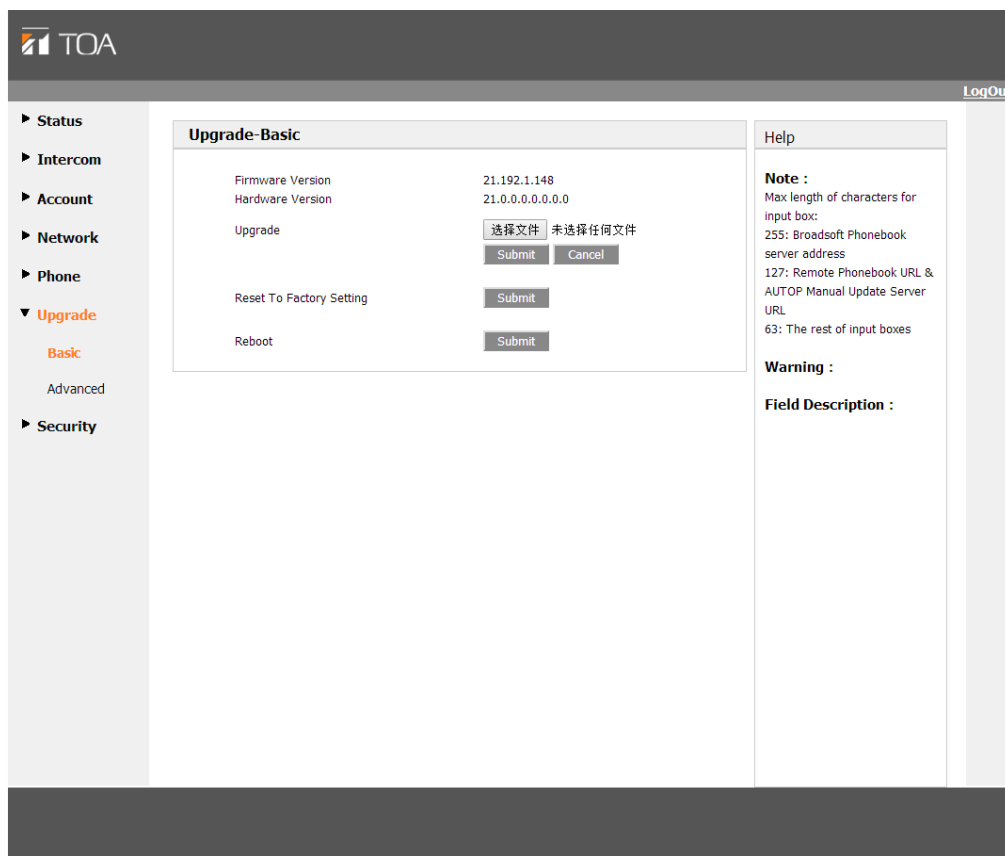
**Submit Shortcut**

Sections	Description
<b>Multicast Setting</b>	To display and configure the Multicast setting. <ul style="list-style-type: none"> <li>● Paging Barge: Setup the priority level. A call is in a higher priority than a paging below this level.</li> <li>● Paging priority Active: Enable or disable a priority between a call and a paging</li> </ul>
<b>Priority List</b>	To setup the multicast parameters. <ul style="list-style-type: none"> <li>● Listening Address: Enter the IP address you need to listen</li> <li>● Label: Input the label for each listening address</li> </ul>

# 7 Upgrade

## 7.1 Basic Upgrade

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



Sections	Description
<b>Upgrade</b>	To select upgrading rom file from local or a remote server automatically. <b>Note:</b> Please make sure it's right file format for right model.
<b>Firmware version</b>	To display firmware version, firmware version starts with MODEL name.
<b>Hardware Version</b>	To display Hardware version.
<b>Reset to Factory Setting</b>	To enable you to reset IP phone's setting to factory settings.
<b>Reboot</b>	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.

The screenshot shows the 'Upgrade-Advanced' configuration page. On the left is a navigation menu with 'Upgrade' selected. The main content area is titled 'Upgrade-Advanced' and contains several sections:

- PNP Option:** PNP Config is set to 'Enabled'.
- DHCP Option:** Custom Option is an empty text box with '(128~254)' next to it. A note below says '(DHCP Option 66/43 is Enabled by Default)'. There are 'Submit' and 'Cancel' buttons.
- Manual Autop:** Fields for URL, User Name, Password, Common AES Key, and AES Key(MAC). An 'AutoP Immediately' button is at the bottom.
- Automatic Autop:** Mode is 'Power On', Schedule is 'Sunday'. Hour is '22' (0~23) and Min is '0' (0~59). There are 'Clear MDS', 'Submit', 'Export Autop Template', and 'Export' buttons.
- System Log:** LogLevel is '3'. There is an 'Export Log' button.
- PCAP:** PCAP and PCAP Auto Refresh buttons. PCAP Auto Refresh is 'Disabled'. There are 'Start', 'Stop', and 'Export' buttons.
- Others:** Config File(.tgz/.conf/.cfg) field. Buttons: 参照..., Export (Encrypted), Import, Cancel.

On the right side, there is a 'Help' section with a 'Note' and a 'Warning' section. The 'Note' says: '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'. The 'Warning' section is empty. Below it is a 'Field Description' section and a 'Submit Shortcut' section with 'Submit' and 'Cancel' buttons.

Sections	Description
<b>PNP Option</b>	<p>To display and configure PNP setting for Auto Provisioning.</p> <ul style="list-style-type: none"> <li>● 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.</li> </ul> <p>By default, this SIP message is sent to multicast address</p>

	224.0.1.75(PNP server address by standard).
<b>DHCP Option</b>	<p>To display and configure custom DHCP option.</p> <ul style="list-style-type: none"> <li>● DHCP option: If configured, IP Phone will use designated DHCP option to get Auto Provisioning server's address via DHCP.</li> </ul> <p>This setting require DHCP server to support corresponding option.</p>
<b>Manual Autop</b>	<p>To display and configure manual update server's settings.</p> <ul style="list-style-type: none"> <li>● URL: Auto provisioning server address.</li> <li>● User name: Configure if server needs an username to access, otherwise left blank.</li> <li>● Password: Configure if server needs a password to access, otherwise left blank.</li> <li>● Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file.</li> <li>● 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).</li> </ul> <p><b>Note:</b> AES is one of many encryption, it should be configure only configure filed is ciphered with AES, otherwise left blank.</p>
<b>Automatic AutoP</b>	<p>To display and configure Auto Provisioning mode settings.</p> <p>This Auto Provisioning mode is actually self-explanatory.</p> <p>For example, mode "Power on" means IP phone will go to do Provisioning every time it powers on.</p>
<b>System Log</b>	<p>To display system log level and export system log file.</p> <ul style="list-style-type: none"> <li>● 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.</li> <li>● Export Log: Click to export temporary system log file to local PC.</li> </ul>
<b>PCAP</b>	<p>To start,stop packets capturing or to export captured Packet file.</p> <ul style="list-style-type: none"> <li>● Start:To start capturing all the packets file sent or received from IP phone.</li> <li>● Stop:To stop capturing packets.</li> </ul> <p><b>Note:</b>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.</p>
<b>Others</b>	<p>To display or configure others features from this page.</p> <ul style="list-style-type: none"> <li>● Config file: To export or import configure file for IP phone.</li> </ul>

## 8 Security

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

The screenshot shows the TOA web interface. On the left is a navigation menu with categories: Status, Intercom, Account, Network, Phone, Upgrade, and Security. Under Security, 'Basic' is selected. The main content area is titled 'Security-Basic' and contains a 'Web Password Modify' form. The form has the following fields: 'User Name' (a dropdown menu showing 'admin'), 'Current Password' (a text input field), 'New Password' (a text input field), and 'Confirm Password' (a text input field). Below the form are 'Submit' and 'Cancel' buttons. To the right of the form is a 'Help' sidebar containing a 'Note' about input box lengths, a 'Warning' section, a 'Field Description' section, and a 'Submit Shortcut' section with 'Submit' and 'Cancel' buttons.

Sections	Description
<b>Web Password Modify</b>	<p>To modify user's password.</p> <ul style="list-style-type: none"> <li>● Current Password: The current password you used.</li> <li>● New Password: Input new password you intend to use.</li> <li>● Confirm Password: Repeat the new password.</li> </ul> <p><b>Note:</b> For now, IP phone can only support user admin.</p>