



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

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Production Overview

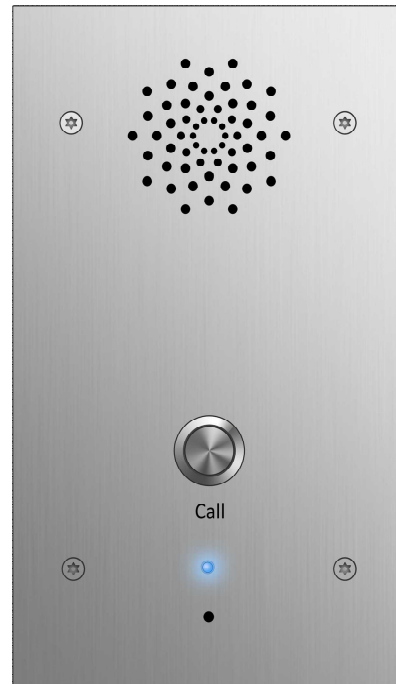
1 Production description

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

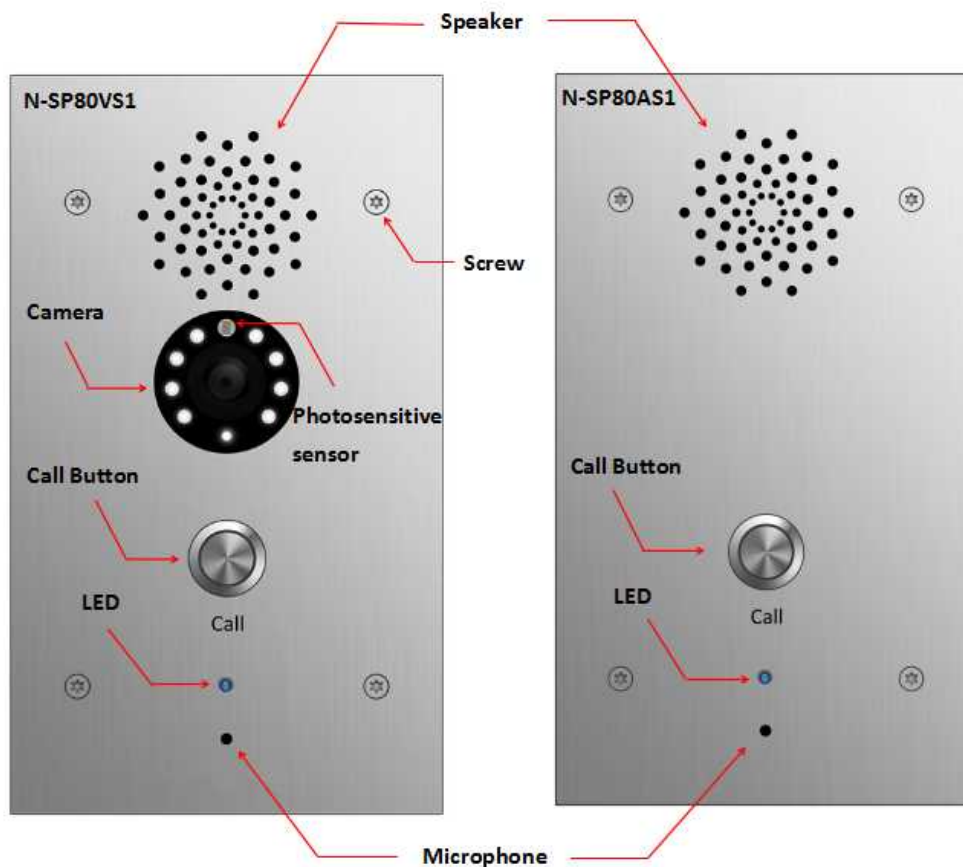
- 2x10/100Mbps Ethernet Port
- Security: Password Protection, IP address filtering, SIP over TLS, HTTPS encryption, user access log

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

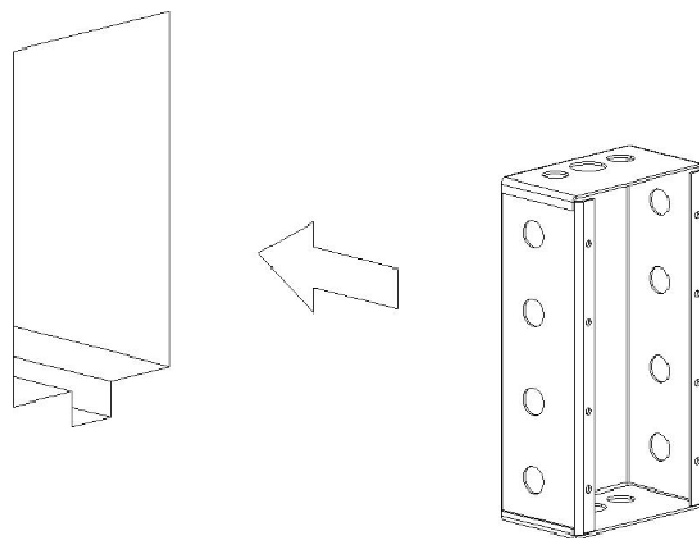
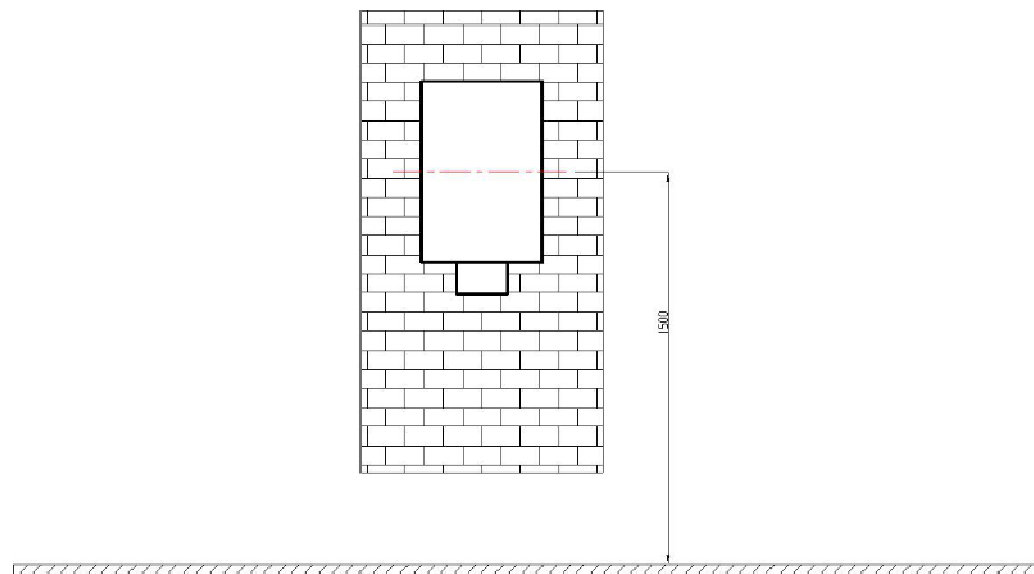
➤ SIP Features

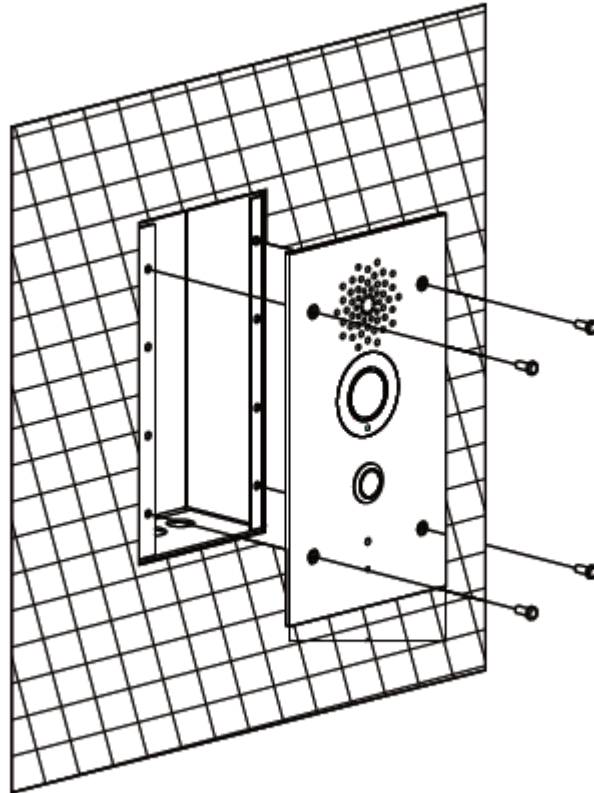
- SIP v1(RFC2543), SIP v2(RFC3261)
- Audio codecs: G.711a, G.711 μ , G.722, G.729
- Video codecs: MPEG-4/MJEG (N-SP80VS1 only)
- Speech Quality: 7kHz Audio
- Echo Cancellation
- Voice Activation Detection
- Comfort Noise Generator

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.

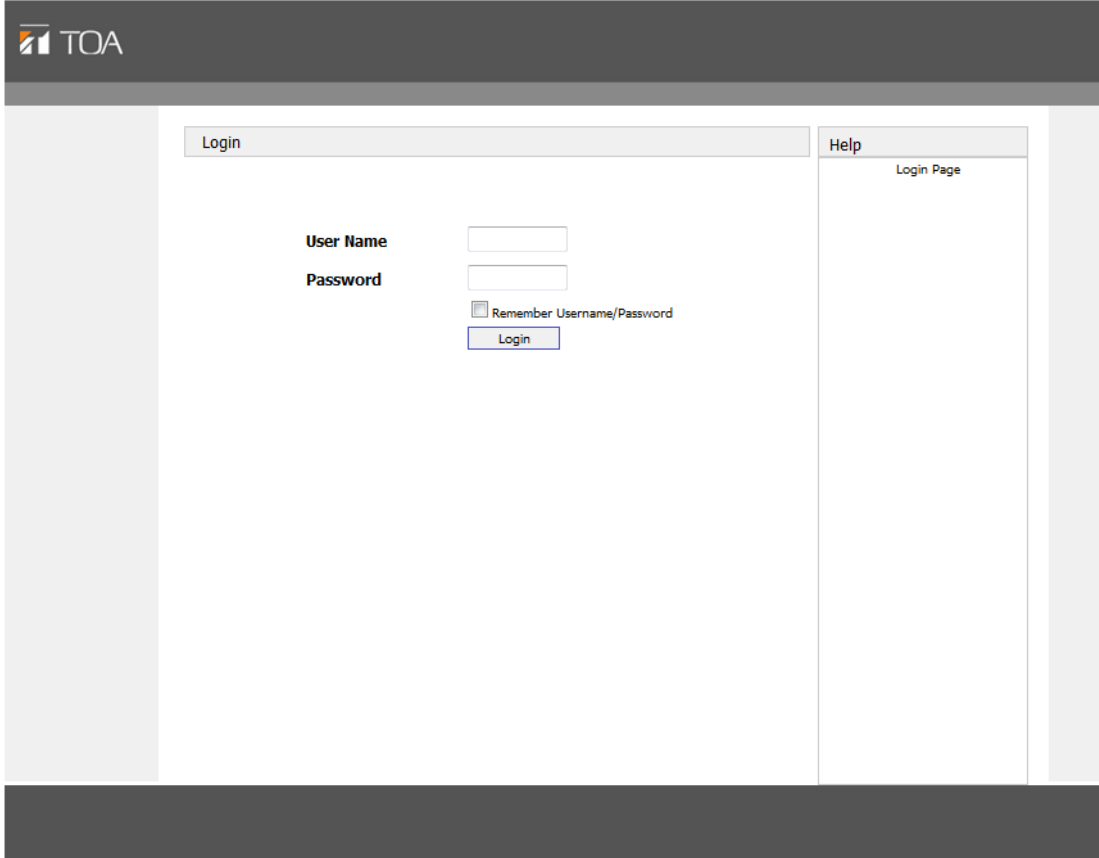
If the IP address is unknown, press the call button when LED light turns blue, after a short period of time(about 5s), the phone will announce its IP.

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



The screenshot displays the TOA web interface. At the top left is the TOA logo. Below it, there is a 'Login' tab and a 'Help' tab. The 'Login' tab is active, showing 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, showing 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 has the TOA logo and a 'LogOut' link. The left sidebar contains a navigation menu with 'Status' selected, followed by '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.akuvov.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
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

[LogOut](#)

► Status

▼ Intercom

Basic

LED Setting

Relay&Input

Live Stream

AEC Setting

RTSP

ONVIF

► Account

► Network

► Phone

► Upgrade

► Security

Intercom-Basic

Account Selection

Select Account

Auto ▼

No Answer Call

Disabled ▼

Push Button

Key	Number
Push Button	192.168.111.70
No Answer Call1	
No Answer Call2	

Web Call

Web Call(Ready)

Auto ▼

Dial Out

Hang Up

Max Call Time

Max Call Time

5

(2~30Minutes)

Push To Hang Up

Push To Hang Up

Enabled ▼

Custom Button

Apply setting to

RelayA ▼

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
Account Selection	<ul style="list-style-type: none"> 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	<ul style="list-style-type: none"> 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.

The screenshot shows the TOA web interface. On the left, a sidebar contains a tree view with the following items: Status, Intercom (expanded), Basic, LED Setting, Relay&Input, Live Stream, AEC Setting (selected), RTSP, ONVIF, Account, Network, Phone, and Upgrade. The main content area is titled 'AEC Setting' and features an 'AEC Level' input field with the value '700'. Below the input field are 'Submit' and 'Cancel' buttons. To the right of the main content area is a 'Help' panel. The 'Help' panel contains the following text: **Note :** Max length of characters for input box: 255; Broadsoft Phonebook server address 127; Remote Phonebook URL & AUTOP Manual Update Server URL 63; The rest of input boxes. **Warning :** **Field Description :** **Submit Shortcut** with 'Submit' and 'Cancel' buttons.

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.

TOA LogOut

Relay&Input

Relay

Relay ID: RelayA, RelayB
 Relay Type: Default stat
 Relay Delay(sec): 3
 DTMF: 0
 Relay Status: RelayA: Low, RelayB: Low

Input

Input ID: InputA, InputB
 Input Service: Disabled
 Call Number:
 Display Name:
 Call Timer: (0~65535 Sec)
 Light Status: InputA: Normal, InputB: Normal

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 :

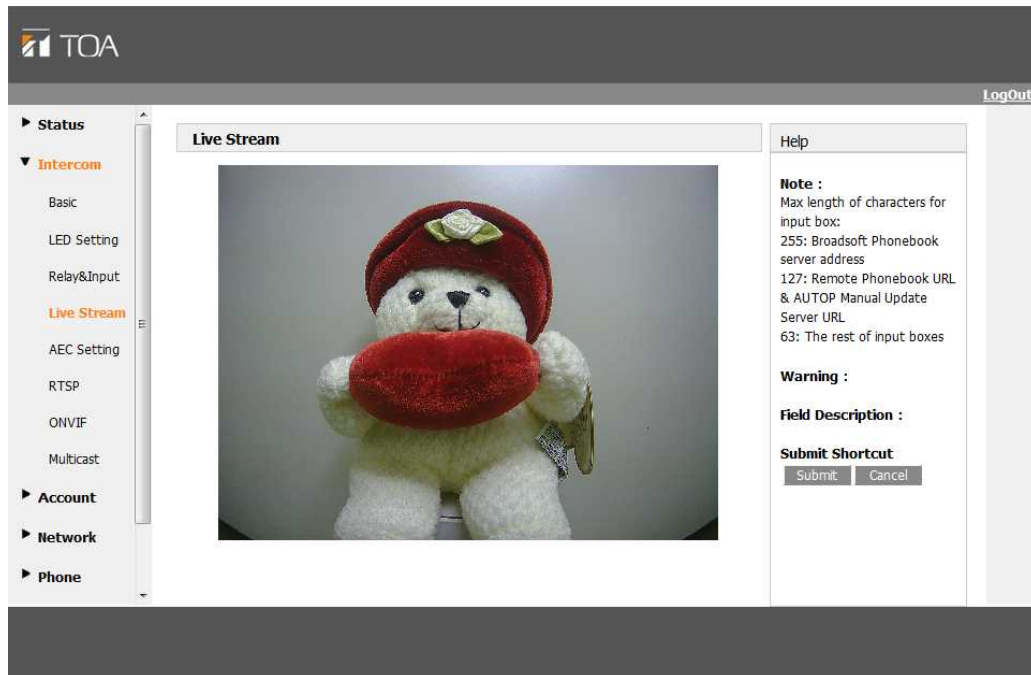
Field Description :

Submit Shortcut
 Submit Cancel

Sections	Description
Relay	<p>To configure some settings about unlock</p> <ul style="list-style-type: none"> ● 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	<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"> ● 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.

	<ul style="list-style-type: none"> ● 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.
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3.4 Live Stream(Optional)



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

3.5 AEC Setting

The screenshot displays the TOA web interface for AEC settings. On the left, a sidebar menu lists various system functions, with 'AEC Setting' currently selected. The main panel is titled 'AEC Setting' and features a single input field labeled 'AEC Level' containing the value '700'. Below this field are 'Submit' and 'Cancel' buttons. To the right of the main panel is a 'Help' section. This section includes a 'Note' explaining that the first 255 characters are for the Broadsoft Phonebook server address, the next 127 characters are for the Remote Phonebook URL & AUTOP Manual Update Server URL, and the remaining 63 characters are for other input boxes. It also contains a 'Warning' section, a 'Field Description' section, and a 'Submit Shortcut' section with its own 'Submit' and 'Cancel' buttons.

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)

TOA LogOut

► Status

▼ **Intercom**

Basic

LED Setting

Relay&Input

Live Stream

AEC Setting

RTSP

ONVIF

► Account

► Network

► Phone

► Upgrade

► Security

RTSP

RTSP Basic

RTSP Server Enabled ☐

RTSP Stream

RTSP Video Enabled ☒

RTSP Video Codec: H.264 ▼

H.264 Video Parameters

Video Resolution: VGA ▼

Video Framerate: 30 fps ▼

Video Bitrate: 2048 kbps ▼

MPEG4 Video Parameters

Video Resolution: VGA ▼

Video Framerate: 30 fps ▼

Video Bitrate: 2048 kbps ▼

MJPEG Video Parameters

Video Resolution: VGA ▼

Video Framerate: 30 fps ▼

Video Quality: 90 ▼

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
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	<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)

The screenshot shows the TOA web interface. The sidebar on the left contains the following menu items: Status, Intercom (selected), Basic, LED Setting, Relay&Input, Live Stream, AEC Setting, RTSP, ONVIF, Account, Network, and Phone. The main content area is titled 'ONVIF' and contains a 'Basic Setting' form. The form has three input fields: 'Onvif Mode' with a dropdown menu set to 'Discoverable', 'UserName' with the text 'admin', and 'Password' with masked characters '*****'. Below these fields are 'Submit' and 'Cancel' buttons. To the right of the form is a 'Help' section containing a 'Note' about input box lengths (255 for Broadsoft Phonebook server address, 127 for Remote Phonebook URL & AUTOP Manual Update Server URL, and 63 for the rest of input boxes), a 'Warning' section, and a 'Field Description' section with 'Submit Shortcut' buttons.

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

TOA LogOut

Account-Basic

SIP Account

Status: Registered
Account: Account 1
Account Active: Enabled
Display Label: 1002
Display Name: 1002
Register Name: 1002
User Name: 1002
Password:

SIP Server 1

Server IP: 10.5.35.134 Port: 5060
Registration Period: 1800 (30~65535s)

SIP Server 2

Server IP: Port: 5060
Registration Period: 1800 (30~65535s)

Outbound Proxy Server

Enable Outbound: Disabled
Server IP: Port: 5060
Backup Server IP: Port: 5060

Transport Type

Transport Type: UDP

NAT

NAT: Disabled
Stun Server Address: Port: 3478

Help

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

Warning :

Field Description :

Submit Shortcut
Submit Cancel

Submit Cancel

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

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

4.2 Account-> Advanced

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

► Status

► Intercom

▼ Account

Basic

Advanced

► Network

► Phone

► Upgrade

► Security

Account-Advanced

SIP Account

Account: Account 1

Codecs

Disabled Codecs

Enabled Codecs

G722
PCMU
PCMA
G729

>>

<<

↑

↓

Video Codec

Codec Name	<input checked="" type="checkbox"/> 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 Bargain Code	

Encryption

Voice Encryption(SRTP)	Disabled
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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 <ul style="list-style-type: none"> ● Codec Name: The default video codec is H264. ● Codec Resolution: It can support QCIF, CIF, VGA, 4CIF, 720P. ● Codec Bitrate: The lowest bitrate is 128, the highest bitrate is 2048. ● Codec payload: From 90-119.
Subscribe	To display and configure MWI, BLF, ACD subscription settings. <ul style="list-style-type: none"> ● MWI: Message Waiting Indicator which is used to indicate whether there is unread new voice message. ● BLF: BLF is short for Busy Lamp Field which is used to monitor the designated extension status. ● ACD: Automatic Call Distribution is often used in offices for customer service, such as call center. The setting here is to negotiate with the server about expire time of ACD subscription.
DTMF	To display and configure DTMF settings. <ul style="list-style-type: none"> ● Type: Support Inband, Info, RFC2833 or their combination. ● How To Notify DTMF: Only available when DTMF Type is Info. ● DTMF Payload: To configure payload type for DTMF. <p>Note: By default, DTMF type is RFC2833 which is the standard. Type Inband uses inband frequency to indicate DTMF tone which is most used to be compatible to traditional telephone server. Type Info use SIP Info message to indicate DTMF message.</p>
Call	To display and configure call-related features. <ul style="list-style-type: none"> ● Max Local SIP Port: To configure maximum local sip port for designated account. ● Min Local SIP Port: To configure minimum local sip port for designated account. ● Caller ID Header: To configure which Caller ID format to fetch for displaying on Phone UI. ● Auto Answer: If enabled, IP phone will be auto-answered when there is an incoming call for designated account.

	<ul style="list-style-type: none"> ● 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	<p>To display or configure session timer settings.</p> <ul style="list-style-type: none"> ● Active: To enable or disable this feature, If enable, the ongoing call will be disconnected automatically once the session expired unless it's been refreshed by UAC or UAS. ● Session Expire: Configure session expire time. ● Session Refresher: To configure who should be response for refreshing a session. <p>Note: UAC means User Agent Client, here stands for IP phone. UAS means User Agent Server, here stands for SIP server.</p>
BLF List	<p>To display or configure BLF List URI address.</p> <ul style="list-style-type: none"> ● BLF List URI: BLF List is short for Busy Lamp Field List. ● BLFList Pickup Code: To set the BLF pick up code. ● BLFList Bargain Code: To set the BLF barge in code.
Encryption	<p>To enable or disabled SRTP feature.</p> <ul style="list-style-type: none"> ● Voice Encryption(SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.
NAT	<p>To display NAT-related settings.</p> <ul style="list-style-type: none"> ● UDP Keep Alive message: If enabled, IP phone will send UDP keep-alive message periodically to router to keep NAT port alive. ● UDP Alive Msg Interval: Keepalive message interval. ● Rport: Remote Port, if enabled, it will add Remote Port into outgoing SIP message for designated account.
User Agent	<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 displays the TOA Network-Basic configuration interface. The 'LAN Port' section is active, showing two options: DHCP and Static IP. The Static IP option is selected, and the following fields are populated: IP Address (10.5.111.102), Subnet Mask (255.255.0.0), Default Gateway (10.5.1.1), LAN DNS1 (8.8.8.8), and LAN DNS2 (empty). The interface includes a sidebar with navigation links, a top navigation bar with a 'LogOut' link, and a right-hand help section with a note and a warning.

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

TOA LogOut

Network-Advanced

Local RTP

Starting RTP Port: 11800 (1024~65535)
Max RTP Port: 12000 (1024~65535)

SNMP

Active: Disabled
Port: (1024~65535)
Trusted IP:

VLAN

LAN Port: Active: Disabled, VID: 1 (1~4094), Priority: 0
PC Port: Active: Disabled, VID: 1 (1~4094), Priority: 0

TR069

Active: Disabled, Version: 1.0
ACS: URL, User Name, Password: *****
Periodic Inform: Active: Disabled, Periodic Interval: 1800 (3~24x3600s)
CPE: URL, User Name, Password: *****

Help

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

Warning :

Field Description :

Submit Shortcut
Submit Cancel

Submit Cancel

Sections	Description
Local RTP	<p>To display and configure Local RTP settings.</p> <ul style="list-style-type: none"> ● Max RTP Port: Determine the maximum port that RTP stream can use. ● Starting RTP Port: Determine the minimum port that RTP stream can use.
SNMP	<p>To display and configure SNMP settings.</p> <ul style="list-style-type: none"> ● 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. <p>Note: SNMP (Simple Network Management Protocols) is</p>

	Internet-standard protocol for managing devices on IP networks.
TR069	<p>To display and configure TR069 settings.</p> <ul style="list-style-type: none"> ● Active: To enable or disable TR069 feature. ● Version: To select supported TR069 version (version 1.0 or 1.1). ● ACS/CPE: ACS is short for Auto configuration servers as server side, CPE is short for Customer-premise equipment as client side devices. ● URL: To configure URL address for ACS or CPE. ● User name: To configure username for ACS or CPE. ● Password: To configure Password for ACS or CPE. ● Periodic Inform: To enable periodically inform. ● Periodic Interval: To configure interval for periodic inform. <p>Note: TR-069(Technical Report 069) is a technical specification entitled CPE WAN Management Protocol (CWMP).It defines an application layer protocol for remote management of end-user devices.</p>

6 Phone

6.1 Time/Language

Go to the path: Phone-> Time/Language

The screenshot shows the TOA web interface. On the left is a sidebar with a tree view containing: Status, Intercom, Account, Network, Phone (expanded), Time/Lang (selected), Call Feature, Voice, Multicast, Upgrade, and Security. The main content area is titled 'Time/Lang' and contains an 'NTP' configuration section. This section has four fields: 'Time Zone' (a dropdown menu showing '+9 Japan(Tokyo)'), 'Primary Server' (a text box with '0.pool.ntp.org'), 'Secondary Server' (a text box with '1.pool.ntp.org'), and 'Update Interval' (a text box with '3600' and a note '(>= 3600s)'). Below these fields are 'Submit' and 'Cancel' buttons. To the right of the NTP section is a 'Help' section. It contains a 'Note' about input box lengths, a 'Warning' section, and a 'Field Description' section. At the bottom of the Help section is a 'Submit Shortcut' with 'Submit' and 'Cancel' buttons.

Sections	Description
NTP	<p>To configure NTP server related settings.</p> <ul style="list-style-type: none"> ● Time Zone: To select local Time Zone for NTP server. ● Primary Server: To configure primary NTP server address. ● Secondary Server: To configure secondary NTP server address, it takes effect if primary NTP server is unreachable. ● Update interval: To configure interval between two consecutive NTP requests. <p>Note: NTP, Network Time Protocol is used to automatically synchronized local time with INTERNET time, since NTP server only response GMT time, so that you need to specify the Time Zone for IP phone to decide the local time.</p>

6.2 Call Feature

Go to the path: Phone->Call Feature

TOA LogOut

Phone-Call Feature

Mode **Phone**

Mode ☒ Phone ☐ Custom

DND

Account

DND

Return Code When DND

DND On Code

DND Off Code

Intercom

Active

Intercom Mute

Others

Return Code When Refuse

Auto Answer Delay (0~5s)

Auto Answer Mode:

Multicast Codec

Direct IP

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
Mode	<p>To enable or disable feature key sync.</p> <ul style="list-style-type: none"> ● Feature Key Sync: To enable or disable feature key sync. ● Mode: Select the desired mode.
DND	<p>DND (Do Not Disturb) allows IP phones to ignore any incoming calls.</p> <ul style="list-style-type: none"> ● Return Code when DND: Determine what response code should be sent back to server when there is an incoming call if DND on. ● DND On Code: The Code used to turn on DND on server's side, if configured, IP phone will send a SIP message to server to turn on DND on server side if you press DND when DND is off. ● DND Off Code: The Code used to turn off DND on server's side, if configured, IP phone will send a SIP message to server to turn off DND on server side if you press DND when DND is on.
Intercom	Intercom allows user to establish a call directly with the

	callee. <ul style="list-style-type: none"> ● Active: To enable or disable Intercom feature. ● Intercom Mute: If enabled, once the call established, the callee will be muted.
Others	<ul style="list-style-type: none"> ● Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected. ● Auto Answer Delay: To configure delay time before an incoming call is automatically answered. ● Auto Answer Mode: To set video or audio mode for auto answer by default. ● 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

The screenshot shows the TOA web interface for configuring voice settings. The main content area is titled 'Voice' and contains several sections:

- Mic Volume:** A slider control for 'Mic Volume' with a value of 8 and a range of (1~15).
- Speaker Volume:** A slider control for 'Speaker Volume' with a value of 8 and a range of (1~15).
- Open Door Warning:** A dropdown menu for 'Open Door Warning' set to 'Enabled'.
- Ringback Upload:** A section for uploading a ringback tone. It includes a file selection button, a message 'ファイルが選択されていません。' (No file selected), and buttons for 'Upload' and 'Delete'. Below the buttons, it specifies 'File Format: wav, size: < 200KB, samplerate: 16000, Bits: 16'.
- CallTone Upload:** A section for uploading a call tone. It includes a file selection button, a message 'ファイルが選択されていません。' (No file selected), and buttons for 'Upload' and 'Delete'. Below the buttons, it specifies 'File Format: wav, size: < 200KB, samplerate: 16000, Bits: 16'.

The right sidebar contains a 'Help' section with 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
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

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

7 Upgrade

7.1 Basic Upgrade

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

The screenshot displays the TOA Web UI interface for the 'Upgrade-Basic' section. On the left, a navigation menu lists various system settings, with 'Upgrade' expanded to show 'Basic' and 'Advanced' options. The main content area is titled 'Upgrade-Basic' and contains the following information:

- Firmware Version:** 21.192.1.148
- Hardware Version:** 21.0.0.0.0.0.0.0
- Upgrade:** A button labeled '选择文件' (Select File) with the status '未选择任何文件' (No file selected), followed by 'Submit' and 'Cancel' buttons.
- Reset To Factory Setting:** A 'Submit' button.
- Reboot:** A 'Submit' button.

On the right side, a 'Help' section provides additional 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 :**

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

7.2 Advanced upgrade

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

Sections	Description
PNP Option	<p>To display and configure PNP setting for Auto Provisioning.</p> <ul style="list-style-type: none"> ● PNP: Plug and Play, once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get Auto Provisioning server's address. <p>By default, this SIP message is sent to multicast address</p>

	224.0.1.75(PNP server address by standard).
DHCP Option	<p>To display and configure custom DHCP option.</p> <ul style="list-style-type: none"> ● DHCP option: If configured, IP Phone will use designated DHCP option to get Auto Provisioning server's address via DHCP. <p>This setting require DHCP server to support corresponding option.</p>
Manual Autop	<p>To display and configure manual update server's settings.</p> <ul style="list-style-type: none"> ● URL: Auto provisioning server address. ● User name: Configure if server needs an username to access, otherwise left blank. ● Password: Configure if server needs a password to access, otherwise left blank. ● Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file. ● AES Key (MAC): Used for IP phone to decipher MAC-oriented auto provisioning configuration file (for example, file name could be 0c1105888888.cfg if IP phone's MAC address is 0c1105888888). <p>Note: AES is one of many encryption, it should be configure only configure filed is ciphered with AES, otherwise left blank.</p>
Automatic AutoP	<p>To display and configure Auto Provisioning mode settings.</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>
System Log	<p>To display system log level and export system log file.</p> <ul style="list-style-type: none"> ● System log level: From level 0~7.The higher level means the more specific system log is saved to a temporary file. By default, it's level 3. ● Export Log: Click to export temporary system log file to local PC.
PCAP	<p>To start,stop packets capturing or to export captured Packet file.</p> <ul style="list-style-type: none"> ● Start:To start capturing all the packets file sent or received from IP phone. ● Stop:To stop capturing packets. <p>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.</p>
Others	<p>To display or configure others features from this page.</p> <ul style="list-style-type: none"> ● Config file: To export or import configure file for IP phone.

8 Security

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

The screenshot displays the TOA web management interface. On the left is a sidebar menu with options: Status, Intercom, Account, Network, Phone, Upgrade, and Security (which is expanded to show 'Basic'). The main content area is titled 'Security-Basic' and contains a 'Web Password Modify' form. The form includes a 'User Name' dropdown menu set to 'admin', and three input fields for 'Current Password', 'New Password', and 'Confirm Password'. Below these fields are 'Submit' and 'Cancel' buttons. To the right of the form is a 'Help' section containing a 'Note' about input box lengths (255 for server address, 127 for URL, 63 for the rest), a 'Warning' section, and a 'Field Description' section with a 'Submit Shortcut' containing 'Submit' and 'Cancel' buttons. A 'LogOut' link is visible in the top right corner of the interface.

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