

# 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  $\mu$  , 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

User name. aumi

TOA				
	Login		Help Login Page	
	User Name Password	Remember Username/Password		
		Login		

#### 2 Status

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

▼ Status	Status		Help	<u>LogOut</u>
Basic Intercom Account Network Phone Upgrade Security	Status         Model         MAC Address         Firmware Version         Hardware Version         LAN Port Type         LAN Port Type         LAN IP Address         LAN Subnet Mask         LAN DNS1         LAN DNS2         Account1         Account2	roduct Information  N-SP80VS1 C4:09:38:02:D9:CD 21.192.1.148 21.0.0.0.0.0.0  etwork Information  DHCP Auto Connected 192.168.35.26 255.255.255.255.0 192.168.35.1 8.8.8  ccount Information  S16@pbx.akuvox.com Registered None@None Disabled	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 :	

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

Intercom-Basic		Help
Account Sele	ction	Note :
Select Account No Answer Call	Auto 🗸 Disabled 🗸	Max length of characters for input box: 255: Broadsoft Phonebook server address
Push Button		127: Remote Phonebook URL AUTOP Manual Update Server
Key	Number	63: The rest of input boxes
Push Button No Answer Call	192.168.111.70	Warning :
No Answer Call	۶	Field Description :
Web Call		Submit Shortcut Submit Cancel
Web Call(Ready	) Auto 👻 Dial Out Hang Up	
Max Call Time	3	
Max Call Time	5 (2~30Minutes)	
Push To Han	g Up	
Push To Hang I	Jp Enabled 👻	
Custom Butt	on	
Apply setting to	RelayA 👻	

Sections	Description	
Account Selection	<ul> <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</li> </ul>	
	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".	
Push Button	<ul> <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>	
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

		Lo
Status	AEC Setting	Help
Interconi Basic LED Setting Relay&Input Live Stream AEC Setting ONVIF Account Network	AEC Level 700 Submit Cancel	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
Phone Upgrade		

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.

5	Relav&Input					Help
om	Relay					Note :
	Relay ID	RelayA •	- Relay	в 🗸		Max length of characters for input box:
etting ®Tanut	Relay Type	Default stat	Defa	ilt stati 👻		255: Broadsoft Phonebook server address
tream	Relay Delay(sec)	3 -	3	•		AUTOP Manual Update Server
etting	DTMF Relay Status	0 RelavA: Low	r 0 RelavB	•		63: The rest of input boxes
						Warning :
	Input					Field Description :
nt	Input ID	InputA	Inpu	itB 🔹		Submit Shortcut
rk	Input Service	Disabled •	Disa	bled .		
	Call Number					
le	Call Timer				(0~65535 Sec)	
Y I	Light Status	InputA: Normal	Inpu	B: Normal		
		Submit			Cancel	

Sections	Description
Relay	To configure some settings about unlock
	<ul> <li>Relay ID: N-SP80VS1/AS1 support 2 relays</li> </ul>
	<ul> <li>Relay Type: Different locks use different relay types.</li> </ul>
	• Delay(s): Allows door remain "open" for certain period
	The range is from 1 to 5 seconds
	<ul> <li>DTMF: Setup DTMF code for remote unlock</li> </ul>
	• 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.
	<ul> <li>Input Service: Disable by default</li> </ul>
	• 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)

		Log01
Status	Live Stream	Help
Intercom		Note :
Basic		Max length of characters for
LED Setting		255: Broadsoft Phonebook
Relay&Input		127: Remote Phonebook URL
Live Stream	1 ° ° '	& AUTOP Manual Update
E AFC Cabling	and the second sec	63: The rest of input boxes
AEC Setting	and an interest of the	Warning *
RTSP		
ONVIF		Field Description :
Multicast		Submit Shortcut
Account	K	Submit Cancel
Network		
Phone		

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

#### 3.5 AEC Setting

	Log(
Status     AEC Setting	Help
Intercom   Basic   LED Setting   Relay&Input   Live Stream   AEC Setting   RTSP   ONVIF   Account   Network   Phone   Upgrade	Note :         Max length of characters for         input box:         25: Broadsoft Phonebook         server address         127: Remote Phonebook URL &         AUTOP Manual Update Server         URL         63: The rest of input boxes         Warning :         Field Description :         Submit       Cancel

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)

B	TSP			Help
1		RTSP Bas	ic	Note :
	RTSP Server Enabled	E		Max length of characters for
ng				255: Broadsoft Phonebook
out		RTSP Stre	am	127: Remote Phonebook URL 1
n	RTSP Video Enabled			AUTOP Manual Update Server URL
g	RTSP Video Codec	H.264		63: The rest of input boxes
	()	1.264 Video Par	ameters	Warning :
	Video Resolution	VGA	•	Field Description :
	Video Framerate	30 fps		Submit Shortcut
	Video Bitrate	2048 kbps	•	
	NDFC4 Video Presentation			
	Video Paral Kara	VCA		
	Video Framerate	30 fps		
	Video Bitrate	2048 kbps		
	Μ	1JPEG Video Pa	rameters	
	Video Resolution	VGA	*	
	Video Framerate	30 fps	•	
	Video Quality	90		
	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	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)

TOA		
► Status	ONVIF	Help
▼ Intercom	Basic Setting	Note :
Basic LED Setting	Onvif Mode Discoverable 🗸	Max length of characters for input box: 255: Broadsoft Phonebook
Relay&Input Live Stream	UserName admin Password	server address 127: Remote Phonebook: URL & AUTOP Manual Update Server URL
AEC Setting RTSP	Submit Cancel	63: The rest of input boxes Warning:
ONVIF		Field Description :
Account		Submit Shortcut
Network		
▶ Phone		

Sections	Description	
Basic Setting	To setup the Onvif function parameters. It is used to connect	
	with the corresponding Onvif tool.	
	<ul> <li>Onvif Mode: Two modes - Discoverable and</li> </ul>	
	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

	Account-Basic			Help
m	SIP Account			
e l	Status	Renistered		Max length of characters for
	Account	Account 1	-	input box:
	Account Active	Enabled		server address
	Display Label	1002		127: Remote Phonebook UR
	Display Name	1002		& AUTOP Manual Update Server LIRI
	Register Name	1002		63: The rest of input boxes
	Liser Name	1002		
	Daesword			warning :
	1.237650			Field Description :
		SIP Server 1		Submit Shortcut
	Carvar ID	10 5 25 124	Doct EDED	Submit Cancel
	Basistration Dariod	1800	(30~65535c)	
	Registration Period	2000	(50-055555)	
		SIP Server 2		
	Server IP		Port 5060	
	Registration Period	1800	(30~65535s)	
	Outba	und Depro: Comore		
	Culbo	unu Proxy Server		
	Enable Outbound	Disabled	¥	
	Server IP		Port 5060	
	Backup Server IP		Port 5060	
	т	ransport Type		
	Transport Type	UDP	3.	
		NAT		
	NAT	Disabled	•	
			Port 3478	

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.



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> <li>Codec Name: The default video codec is H264.</li> </ul>	
	• 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.
	<ul> <li>Missed Call Log: To display the miss call log.</li> </ul>
	• 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.
	<ul> <li>Session Expire: Configure session expire time.</li> </ul>
	• 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.

	Network-Basic		Help
mo		LAN Port	N-4
nt erk nced	<ul> <li>DHCP</li> <li>Static IP</li> <li>IP Address</li> <li>Subnet Mask</li> <li>Default Gateway</li> <li>LAN DNS1</li> <li>LAN DNS2</li> </ul>	10.5.111.102 255.255.0.0 10.5.1.1 8.8.8.8	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 :
	Submit	Carcel	Submit Shortcut Submit Cancel

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

	ed				Help
	Loc	al RTP			Note :
	Starting RTP Port	11800		(1024~65535)	Max length of characters for input box:
	Max RTP Port	12000		(1024~65535)	255: Broadsoft Phonebook
				127: Remote Phonebook UR	
	S	SNMP			Server URL
	Active	Disabled	्र	(100.1 (CCC0.0)	63: The rest of input boxes
	Port Trusted 10			(1024~65535)	Warning :
	Thusbed IP				Field Description :
	v	LAN			Submit Shortcut
LAN Port	Active	Disabled			Submit Cancel
	VID	1	0.000	(1~4094)	
	Priority	0			
PC Port	Active	Disabled	2.5	(1- 4004)	
	Priority	0		(1~4034)	
			20		
TR069					
	Active	Disabled	•		
105	Version	1.0			
	User Name				
	Password	•••••			
Periodic Inform	Active	Disabled		1	
123332	Periodic Interval	1800		(3~24×3600s)	
CPE	URL				
	Oser Name				
	LAN Port PC Port ACS Periodic Inform CPE	Loc Starting RTP Port Max RTP Port Max RTP Port Starting RTP Port Max RTP Port Port Trusted IP VID Priority PC Port Active VID Priority PC Port Active VID Priority CPE URL User Name	Local RTP  Local RTP  Starting RTP Port  Max RTP Port  11800  Max RTP Port  12000  SNMP  Active Port Trusted IP  LAN Port Active VID 1 Priority 0  PC Port Active Disabled VID 1 Priority 0  CPE URL User Name Periodic Inform Active Disabled Periodic Inform Active User Name I800 CPE URL User Name I800	Local RTP       Starting RTP Port     11800       Max RTP Port     12000       SNMP       Active     Disabled       Port	Local RTP           Starting RTP Port         11800         (1024~65535)           Max RTP Port         12000         (1024~65535)           SNMP         Active         Disabled         •           Port         (1024~65535)         •         •           Trusted IP         (1024~65535)         •         •           LAN Port         Active         Disabled         •           VLAN         VID         1         (1~4094)           Priority         0         •         •           PC Port         Active         Disabled         •           VID         1         (1~4094)         •           Priority         0         •         •           VID         1         (1~4094)         •           Priority         0         •         •           VID         1         (1~4094)         •           Priority         0         •         •           VID         1         (1~4094)         •           VID         1         (1~4094)         •           VID         1         (1~4094)         •           VID         1         (1~4094)         •<

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.
	<ul> <li>Active: To enable or disable SNMP feature.</li> </ul>
	<ul> <li>Port: To configure SNMP server's port.</li> </ul>
	• 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.
	<ul> <li>Active: To enable or disable TR069 feature.</li> </ul>
	• 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.
	<ul> <li>URL: To configure URL address for ACS or CPE.</li> </ul>
	<ul> <li>User name: To configure username for ACS or CPE.</li> </ul>
	<ul> <li>Password: To configure Password for ACS or CPE.</li> </ul>
	<ul> <li>Periodic Inform: To enable periodically inform.</li> </ul>
	• 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

US	Time/Lang				Help
rcom		NTP		1	Note : Max length of characters for
vork	Primary Server Secondary Server	0.pool.ntp.org 1.pool.ntp.org		<ul> <li>input box:</li> <li>255: Broadsoft Pho server address</li> </ul>	
ne/Lang	Update Interval	3600	(>= 3600s)		& AUTOP Manual Update Server URL 63: The rest of input boxes
l Feature ce ticast	Sub	mit	Cancel		Warning : Field Description :
rade					Submit Shortcut Submit Cancel
inty					

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
----------------	---------------	---------

Status	Dhone-Call Feature		Liele	
• Intercom	Filone can reature	нер		
Account			Note : Max length of characters for	
	Hote	© Phone © Custom	input box:	
• Network		DND		
Phone	Account	All Account	127: Remote Phonebook URL &	
Time/Lang	DND	Disabled V	URL	
hine/Lang	Return Code When DND	486(Busy Here) 🔻	63: The rest of input boxes	
Call Feature	DND On Code		Manual	
Voice	DND Off Code		warning :	
Multicast			Field Description :	
		Submit Shortcut		
Upgrade	Active	Enabled 🔻	Submit Cancel	
Security	Intercom Mute	Disabled •		
		Others		
	Return Code When Refuse	486(Busy Here)		
	Auto Answer Delay	0 (0~5s)		
	Auto Answer Mode:	Video 🔻		
	Multicast Codec	PCMU 🔻		
	Direct IP	Enabled 🔻		
	Submit	Cancel		

Sections	Description
Mode	To enable or disable feature key sync.
	• Feature Key Sync: To enable or disable feature key sync.
	<ul> <li>Mode: Select the desired mode.</li> </ul>
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

tus			
	Voice		Help
ercom		Mic Volume	Note :
ount	Mic Volume	8 (1~15)	Max length of characters for input box:
work		Speaker Volume	255: Broadsoft Phonebook server address 127: Remote Phonebook URL &
ne/Lang	Speaker Volume	8 (1~15)	AUTOP Manual Update Server URL
l Feature		Open Door Warning	63: The rest of input boxes Warning :
ice	Open Door Warning Enabled 🗸		Field Description :
lticast Irade	Ringback Hoload		Submit Shortcut
urity	goddk opiodd		Submit Cancel
	File Format: wav, size: < 200H	KB, samplerate: 16000, Bits: 16	
		CaliTone Upload	
	参照 ファイルが選択されて	ていません。 Upload Delete	

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

		Multicast Setting			Note :
<ul> <li>Account</li> <li>Network</li> </ul>	Paging Barge				Max length of characters for input box: 255: Broadsoft Phonebook
Phone		y Active			server address 127: Remote Phonebook URL &
Time/Lang		Priority	List		AUTOP Manual Update Server URL
Call Feature	IP Address	Listening Address	Label	Priority	63: The rest of input boxes
Noico	1 IP Address	225.0.0.0:6000	ALL CALL	1	Warning :
VOICE	2 IP Address	225.0.0.0:6002	BGM1	2	Field Description :
Multicast	3 IP Address			3	Submit Chartout
Upgrade	4 IP Address			4	Submit Cancel
Security	5 IP Address			5	
	6 IP Address	1		6	
	7 IP Address	1		7	
	8 IP Address	1		8	
	9 IP Address	Q.		9	
	10 IP Address			10	

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 serve	
	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.

Upgrade-Advanced		Help	
	PNP Option	Note -	
PNP Config	Enabled 👻	Max length of characters for input box:	
5	DHCP Option	255: Broadsoft Phonebook server address	
Contra Contra	(120-204)	URL & AUTOP Manual	
(DHCP Option 66/43 is Enabled	by Default)	Update Server URL 63: The rest of input boxes	
,	Manual Autop	Warning :	
LIBI		Field Description :	
Licer Name		Submit Shortcut	
Deceword		Submit Cancel	
Common AES Key			
AES Key/MAC)			
ALS REVUERCY			
	AutoP Immediately		
A	Automatic Autop		
Mode	Power On 👻		
Schedule	Sunday 🐰		
	22 Hour(0~23)		
	0 Min(0~59)		
Clear MD5	Submit		
Export Autop Template	Export		
Submit Cancel			
	System Log		
LogLevel	3		
Export Log	Export		
	РСАР		
PCAP	Start. Stop Export		
PCAP Auto Refresh	Disabled 👻		
	Others		
Config File(.tgz/.conf/.cfg)	参照 ファイルが選択されていません。		

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.		
	<ul> <li>URL: Auto provisioning server address.</li> </ul>		
	• 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 0c11058888888.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.		
РСАР	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.		
	<ul> <li>Stop: To stop capturing packets.</li> </ul>		
	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	In display or configure others features from this page.		
	Config file: To export or import configure file for IP phone.		

# 8 Security

221727			
Status	Security-Basic		Help
Intercom	Web Pass	word Modify	Note :
Account	User Name	admin 👻	Max length of characters for input
Network	Current Password		255: Broadsoft Phonebook server
Phone	New Password		address 127: Remote Phonebook URL &
	Confirm Password		AUTOP Manual Update Server
opgrade			63: The rest of input boxes
Security	Submit	Cancel	Warning :
Basic			Field Description :
			Submit Shortcut
			Submit Cancel

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.