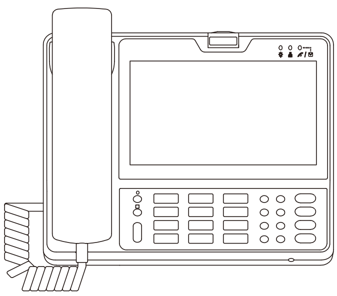
INSTRUCTION MANUAL

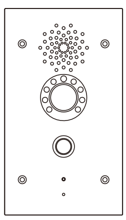
SIP VIDEO INTERCOM

N-SP80 Series

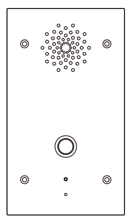
SIP MULTIMEDIA STATION	**N-SP80MS1**
SIP VIDEO DOOR STATION	**N-SP80VS1**
SIP AUDIO DOOR STATION	**N-SP80AS1**
4 SIZE BACK BOX	**YC-400**



N-SP80MS1



N-SP80VS1



N-SP80AS1

Thank you for purchasing TOA's SIP Video Intercom.
Please carefully follow the instructions in this manual to ensure long, trouble-free use of your equipment.

TABLE OF CONTENTS

1. SAFETY PRECAUTIONS

- Before installation or use, be sure to carefully read all the instructions in this section for correct and safe operation.
- Be sure to follow all the precautionary instructions in this section, which contain important warnings and/or cautions regarding safety.
- After reading, keep this manual handy for future reference.

Safety Symbol and Message Conventions

Safety symbols and messages described below are used in this manual to prevent bodily injury and property damage which could result from mishandling. Before operating your product, read this manual first and understand the safety symbols and messages so you are thoroughly aware of the potential safety hazards.

⚠ **CAUTION**	Indicates a potentially hazardous situation which, if mishandled, could result in moderate or minor personal injury, and/or property damage.

- Use the specified AC adapter and PoE switching hub for the unit. Note that the use of another adapter and PoE switching hub may cause a fire. (N-SP80MS1 only)

- Avoid touching the unit's sharp metal edge to prevent injury. (N-SP80VS1, N-SP80AS1, and YC-400)

- Use the 12 V DC power supply and PoE switching hub for the unit. Note that the use of another adapter and PoE switching hub may cause a fire. (N-SP80VS1 and N-SP80AS1 only)

2. GENERAL DESCRIPTION

The N-SP80 series intercom system is designed for use in combination with the SIP (Session Initiation Protocol) Intercom system.
This system consists of the Android based touch panel multimedia stations and the door stations. The door station is available in 2 models: the one with camera and the one without camera.
The system can be used not only in SIP server mode but also in peer-to-peer mode: the former enables the system to work by connecting to the SIP server and the latter enables it to work without using the SIP server.
In this manual, the N-SP80MS1 is described as the multimedia station, and the N-SP80VS1 and the N-SP80AS1 are collectively described as the door station.

Note: Android is a trademark of Google LLC.

3. FEATURES

- Fully compliant with SIP.
- Connected to the network via Ethernet.
- The door station with camera is compatible with ONVIF.
- Supports 2 ways of connection: Connection to the SIP server using SIP and peer-to-peer connection.
- Supports the following audio codecs: G.711, G.722, and G.729.
  (Audio codec is fixed to G.722 when in peer-to-peer connection.)
- Can be powered by means of PoE or from the DC power supply unit.
- The camera incorporated in the door station with camera has 3 mega pixels, featuring the built-in infrared light that allows images to be taken at night.
- The built-in acoustic echo canceller ensures full duplex conversation for the door stations.
- The door stations are equipped with a relay output function. Relay control can be performed from a multimedia station or other SIP telephones.
- The door stations are equipped with an external control input function. Different types of calling can be made when an external control switch is connected.
- The multimedia station is easy to operate on the screen thanks to a touch panel and GUI (Graphical User Interface) design, and in addition, usable as a telephone by intuitive dial operation with a ten-key pad.
- The multimedia station is selectable one of 3 conversation methods: handset conversation, hands-free conversation, and headset conversation.
- Compatibility with CUCM obtained by Cisco systems.
- Easy conversations between stations even under high noise environment.

Note: Cisco Systems is a registered trademark of Cisco Systems, Inc. in the United States and certain other countries.

4. HANDLING PRECAUTIONS

The transmission quality of the internet is not always guaranteed.
Therefore, when this system is connected via the internet, the following symptoms may happen when the network is congested.

- Packet loss
- Interruption of speech voice
- Generation of noise

5. USAGE MODE

5.1. SIP Server Mode

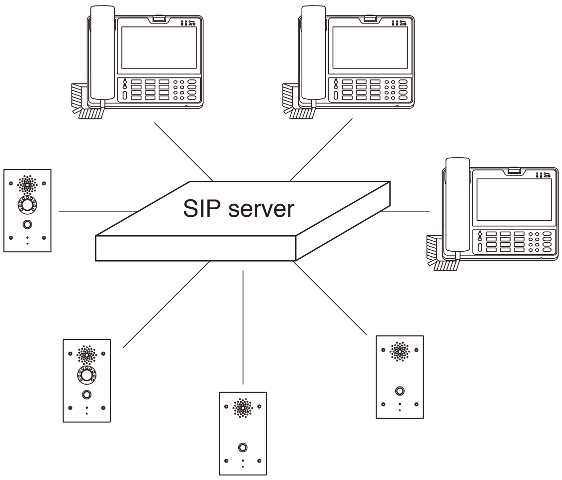
Each equipment of this system is in full conformity with SIP and can be used in connection to the SIP server (Telephone system). The SIP server mode refers to the system when connecting to the SIP server. The connection to CUCM by Cisco Systems and Asterisk by Digium is confirmed for the N-SP80 series. Since the supporting version and the SIP server need to be updated, check for the latest information on the TOA product data download site (https://www.toa-products.com/international/).

Note: Asterisk is a trademark of Digium, Inc.

[Features of SIP server mode communication]

- Audio signals are directly communicated between stations. (They may be communicated via the server.)
- The maximum number of the stations to be connected depends on the SIP server's specification.
- Communications are centrally controlled by the server.
- Transfer function can be used depending on the SIP server's specification.
- Connection to the outside line is possible depending on the SIP server's specification.

[SIP server mode]



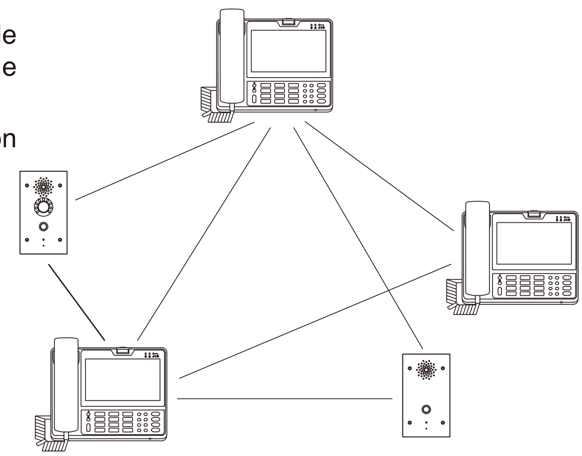
5.2. Peer-to-Peer Mode

The devices of this system can be directly connected with each other without using the SIP server. Peer-to-Peer mode refers to the system or the state where the stations are directly connected with each other without using a SIP server.

[Features of Peer-to-peer mode communication]

- Multiple one-to-one communications can be made simultaneously, enabling the number of the connectable stations within the system to be unlimited.
- SIP server is not used, allowing suppression of introduction costs.
- A 3 party conversation can be made.
- Audio codec is fixed to G.722.

[Peer-to-peer mode]

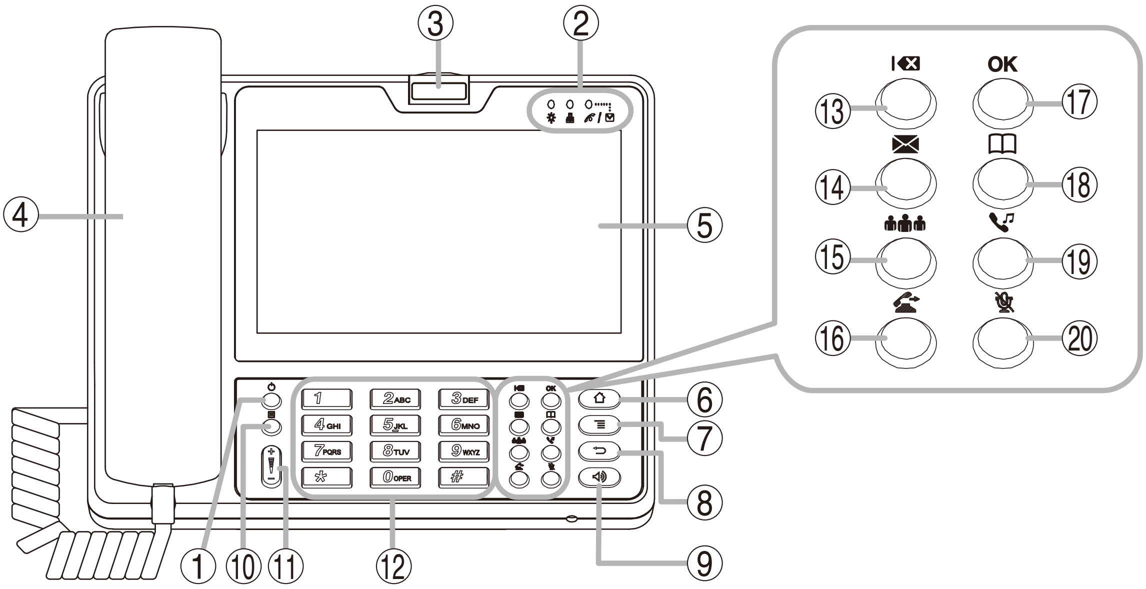


6. NOMENCLATURE AND FUNCTIONS

6.1. N-SP80MS1

- Desktop design
- Handset, Hands-free (with AEC function), Headset connectable
- Built-in touch panel
- PoE-compatible (IEEE802.3af compliant)
- Built-in camera

[Front]



1. Power ON/OFF button
Press this button to turn on the power and pressing it again will place the unit in sleep state.
Holding it down for 1 second or more allows the selection of either restarting the unit or setting the manner mode.

2. Indicators
Indicate the following operation states.

: Indicates the unit's power ON or OFF state. The unit is operating when the indicator is lit, and not operating when it is unlit.

: Indicates the network connection state. Connection to a LAN is established when the indicator is lit, and not established when it is unlit.

/ : Indicates the absence incoming call/ unread message state. The absence incoming call or unread messages are present when the indicator is flashing, and not present when it is unlit.

3. Camera
A built-in camera with 2 mega pixels.
Used when making conversation between multimedia stations.

4. Handset
Used for handset conversation.

5. Touch panel
A 7" touch panel screen.

6. Home button
Returns the display to the home screen.

7. Menu button
Indicates the setting items on the screen.

8. Back button
Returns the display to the last screen you viewed, or the home screen.

9. Speaker button
Press this button to start hands-free conversation.

10. Sleep button
Press this button to place the unit in sleep state.
Holding it down for 1 second or more allows the selection of either restarting the unit or setting the manner mode.

11. Volume control button
Adjusts the volume level of the built-in speaker.

12. Numerical keypad
Used to enter the numbers.

13. Delete button
Deletes one by one the dial numbers entered when making a call or the characters entered when registering.

14. Message button
Press this button to read or write the short message. (Only when in the SIP server mode)

15. Conference button
Starts the three-party conversation function.

16. Transfer button
Starts the transfer function during conversation. (Only when in the SIP server mode)

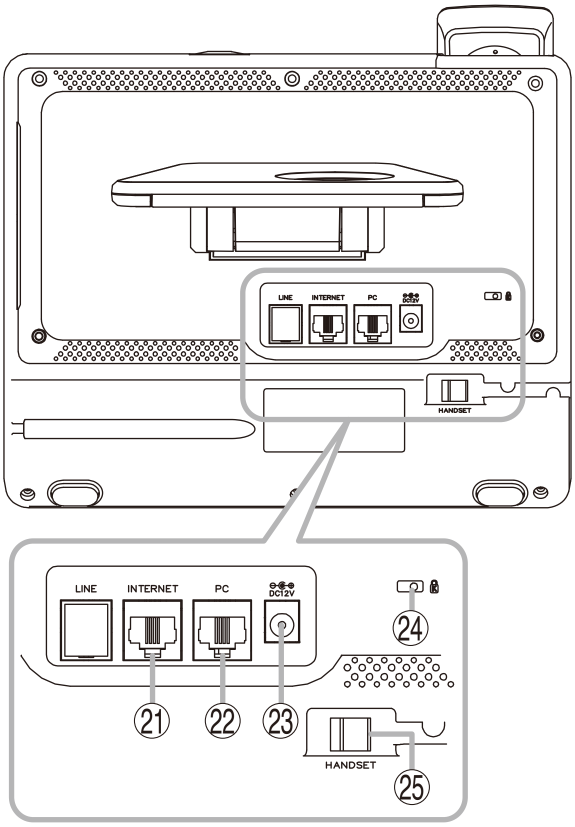
17. OK button
Confirms the selection on the screen while in the touch panel operation.

18. Contact button
Press this button when registering or selecting the telephone book.

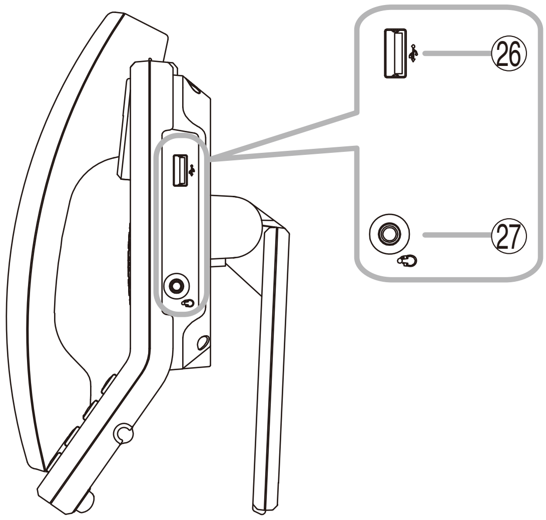
19. Hold button
Starts the hold function during conversation.

20. Mute button
Mutes the hands-free microphone and the handset microphone.

[Rear]



[Right side]



21. LAN connection terminal
Connect the LAN cable to this terminal.
This terminal can be connected to PoE switching hub.

22. PC connection terminal
Connect a PC to this terminal.

23. DC input terminal
Connect the AC adapter* to this terminal.

* Use the optional AD-1215P or its equivalent.

24. Security slot
Used to connect the security wire such as a theft preventing wire.
This is a Kensington lock slot.

25. Handset connection terminal
Connect the handset to this terminal.

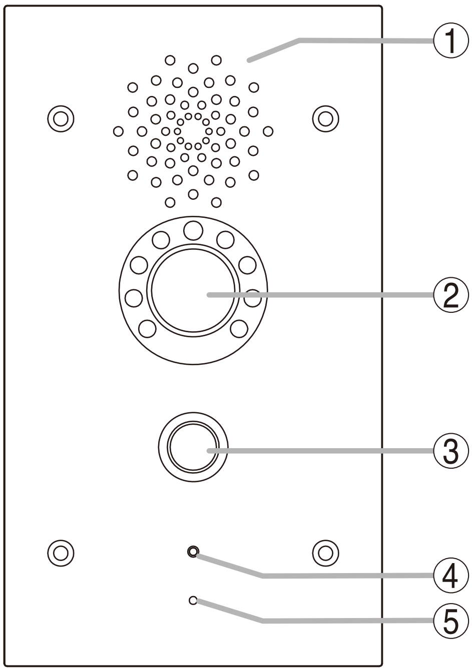
26. USB connection terminal
Connect a USB device to this terminal.

27. Headset connection terminal
Connect a headset to this terminal.
20 mW, 16 Ω/32 Ω, ø3.5 mm mini jack (3P)

6.2. N-SP80VS1, N-SP80AS1

- Designed for wall-recessed installation (YC-400 is required.)
- Hands-free (with AEC function) type
- Call button x 1, Relay output x 2, External control input x 2
- PoE compatible (IEEE802.3af compliant)
- Equipped with a camera, luminance sensor, and infrared light (N-SP80VS1 only)

[Front]



The figure shows the N-SP80VS1.

1. Speaker
Outputs speech voice from the partner station during conversation.

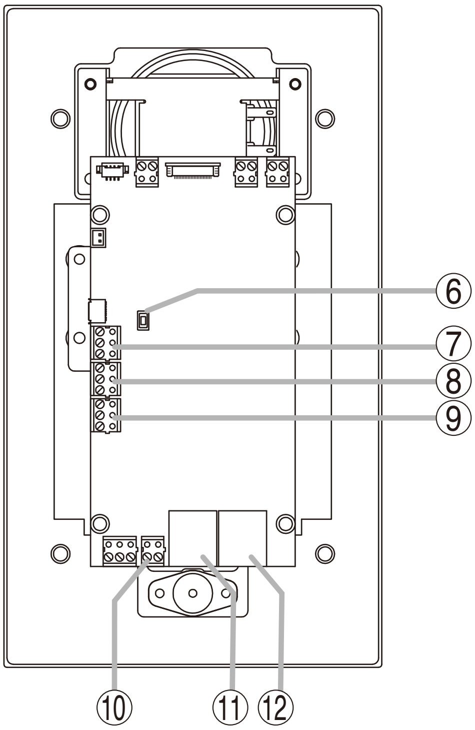
2. Camera (N-SP80VS1 only)
A built-in camera with 3 mega pixels.
Used when making conversation with a multimedia station.

3. Call button
Press this button to start conversation.
Pressing this button will call the preset partner station.

4. Operation indicator
Lights or flashes during conversation or during a call from this door station.
(The indication color and when the indicator lights or flashes can be changed by the setting.)

5. Microphone
Picks up the speaker's voice during conversation, which is then sent to the partner station.

[Rear]



The figure shows the N-SP80VS1.

6. Reset button
Press this button to restart the station.
Holding down this button for 5 seconds or more restarts the door station in the default state. (The settings data will be initialized.)

7. External control inputs 1 and 2
A special calling can be performed by connecting an external control switch or sensor output to this terminal.

8. Relay 1 connection terminal
The relay output can be controlled by the specific dial code from the partner station during conversation. Used to unlock the nearby door. (An appropriate dial code can be set.)

9. Relay 2 connection terminal
Has the same function as the Relay 1 connection terminal (8).

10. DC power input terminal
The door station can be operated by inputting 12 V DC to this terminal.

11. Ethernet connection terminal
Used when connecting to the network.

12. Ethernet connection terminal (PoE compatible)
Used when connecting to the network.
Power can be supplied when this terminal is connected to the PoE switching hub.

7. LIST OF SYSTEM FUNCTIONS

7.1. Basic Functions

7.1.1. List of the N-SP80MS1's functions

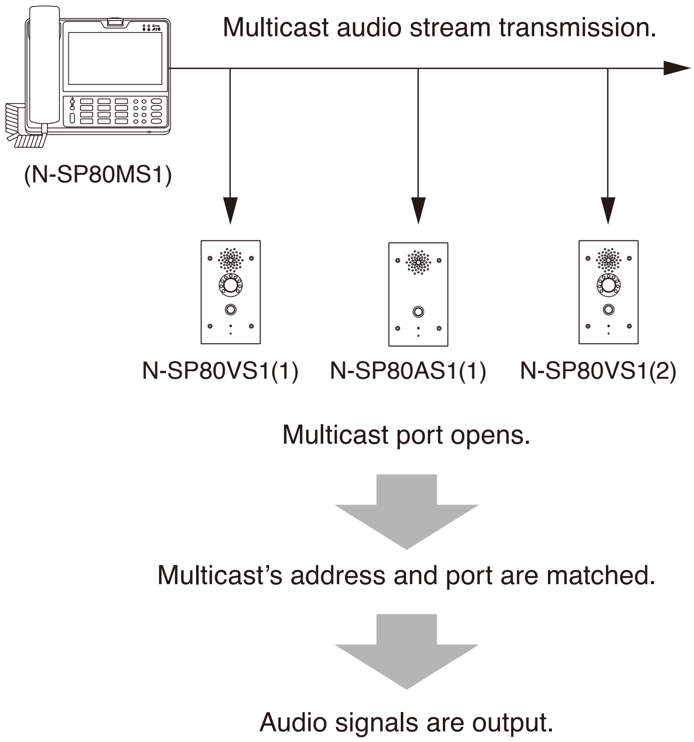
Function		General description	Reference page	
			Operation	Setting
Call	Making a call (Call by direct dialing)	You can make a call by directly pressing the other party's phone number (or IP address) using the numerical keypad (or touching the ten keys on the touch panel).	p. 15	—
	Making a call (Call from the phone book)	Call can be made from the phone book registered in advance.	p. 16	p. 49
	Receiving a call	You can select either "Response only with audio " or "Response using video" for a call reception.	p. 17	—
	Rejecting an incoming call	You can reject an incoming call.	p. 18	—
	Call option	Operation of such as holding, microphone muting, and video switching can be made during conversation.	p. 18	—
	Video option	Operation of such as screen resizing, self-view display, and other party view display can be made during video conversation.	p. 18	p. 44
	Call log	You can check call logs and make a call from the call history, too.	p. 19	p. 51
	Transfer (Blind transfer)	A method to transfer a call without confirming with the transfer destination party. Can be used when in SIP server mode.	p. 21	—
	Transfer (Attended transfer)	A method to transfer a call after confirming with the transfer destination party in advance. Can be used when in SIP server mode.	p. 22	—
Paging	Paging Call	You can make a paging call. Paging is performed using Multicast audio from a multimedia station.	p. 23	—
Others	3 party conference	Conversation among 3 parties can be made. Can be used when in SIP server mode.	p. 24	—
	Connecting to a third-party VoIP	Can be connected to a third-party VoIP.	p. 26	—
	Connection with outside line	Connection with outside line can be made. Can be used when in SIP server mode.	p. 26	—
	Door remote control	The relay output can be controlled by entering the preset dial code during conversation with the door station. Can be used to unlock the nearby door.	p. 26	—

7.1.2. List of the N-SP80VS1's and N-SP80AS1's functions

Function		General description	Reference page	
			Operation	Setting
Call	Making a call	You can make a call by pressing the Call button.	p. 27	p. 57
	Receiving a call	Call reception is automatically responded.	p. 27	—
	Cancelling a call and conversations	Pressing the Call button during a call or conversation cancels the current operation.	p. 27	p. 57
Paging	Paging function	Paging call is received automatically. Audio signals are output when the preset multicast address receives the audio stream.	p. 27	p. 68
Others	No answer forward function	Up to 3 call destinations can be assigned to the Call button by setting the order. If the first called station is absent, then the 2nd one will be called. When the 2nd one is also absent, then the 3rd one will be called.	p. 28	p. 57
	Time limit	Maximum duration of calling time and conversation time can be set.	p. 28	p. 57
	Door remote control	The relay output built in the door station can be controlled from the multimedia station. Can be used to unlock the nearby door.	p. 28	p. 59
	Call activation from an external device	A different call can be made when inputting signals from the external device to the door station's external control input.	p. 29	p. 59
	Change in various function sounds	Call transmission sound, call reception sound, and door remote control sound can be changed. Can be changed by updating the sound file.	p. 29	p. 67

8. SUMMARY OF THE PAGING FUNCTION

8.1. Paging Configuration



(Example)

N-SP80MS1 Transmission address/Port
239.0.0.0:5004

N-SP80VS1 (1) Reception address/Port
239.0.0.0:5010

N-SP80AS1 (1) Reception address/Port
239.0.0.0:5004

N-SP80VS1 (2) Reception address/Port
239.0.0.0:5004

The N-SP80MS1's audio signals are output from the N-SP80AS1 (1) and N-SP80VS1 (2).

8.2. Priority Setting of Paging and Conversation

Paging call made from the multimedia station can be received through each station's speaker.
The priority level of paging call and conversation can be set for each station. (See p. 68.)

9. MULTIMEDIA STATION'S FUNCTIONS AND OPERATIONS

9.1. Basic Usage

Tip

This device runs on Android OS. Perform settings such as clock by clicking the Setting icon.

9.1.1. Main screen

1. Status bar

Located at the upper most of the screen and displays the system status information.

2. Accounts

The multimedia station supports 6 accounts. The Account list displays the most recent account statuses.

3. Expanded screen prompt

Displays the location of the current screen page and how many pages the screens are expanded.

4. Shortcuts

Up to 4 shortcuts can be arranged in the dock area. Any number of shortcuts can be arranged in the main area.

You can replace the arranged shortcut by dragging a new one to any region as shown above.

9.1.2. Status and notification display

1. Status area

Icon	Description
	Connection to network has been established.
	Connection to network is not established.
	Mute mode
	Alarm clock setting has been completed.
	Connection by PPPoE has been established.
	PPPoE connection has failed.

2. Notification area

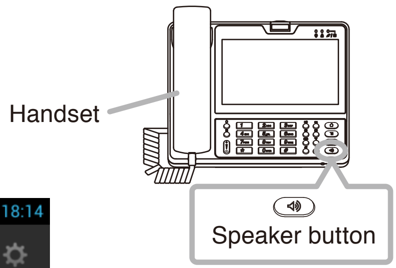
Icon	Description
	SIP account has been registered.
	Unconfirmed incoming calls exist. **Note:** You can check the number of unconfirmed incoming calls by swiping down on the notification area. (See p. 19.)
	Downloading
	Uploading
	A new mail has been received.
	An incoming event are being received.

9.2. Conversation's Functions and Operations

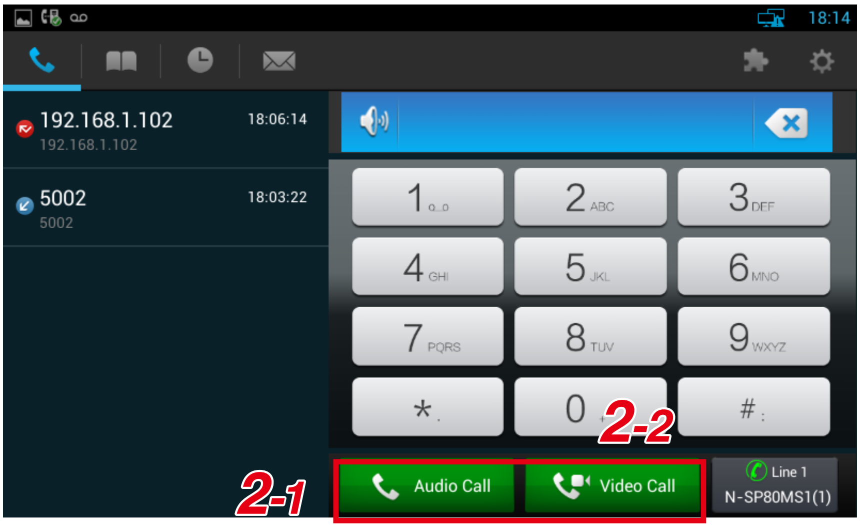
9.2.1. Making a call (Call by direct dialing)

You can make a call directly using the station's ten keys or using the ten keys, contact list, or call history on the dial screen.
The dial screen can be displayed by one of the following operations.

- Lift the handset.
- Press the station's Speaker button.
- Touch on the main screen.



(Dial screen)



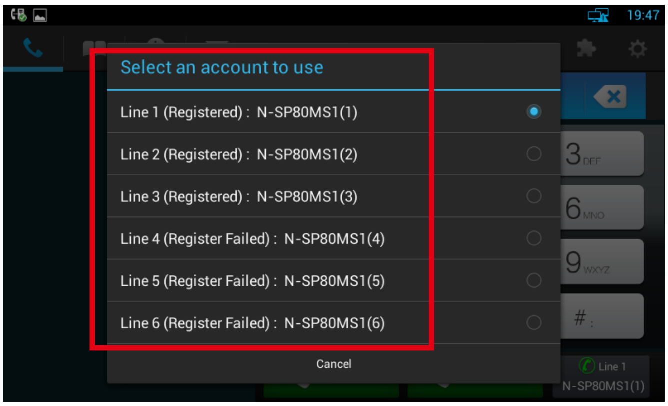
Step 1. Enter the call destination number using the ten keys on the dial screen or the station's ten keys.

Tips

- To delete the number, press on the dial screen or the station's Delete button directly. The number is deleted one by one.
- Period can be entered by holding down for one second or more.

[When multiple SIP accounts have been registered in SIP server mode]

The accounts can be switched and used.
Touch on the dial screen, then select the account number to be switched to on the displayed screen.



Step 2. Make a call.

2-1. When making an audio call, touch Audio Call on the dial screen.

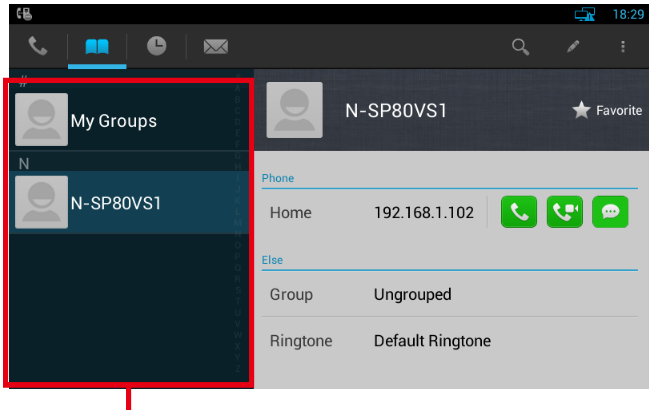
2-2. When making a video call, touch Video Call on the dial screen.

9.2.2. Making a call (Call from the phone book)

You can make a call to the contact on the dial screen.

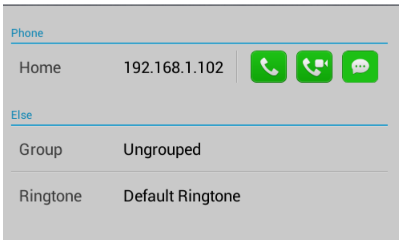
Step 1. Touch [ => Local Phone Book => All Contacts…] on the dial screen.
The contact list appears on the left side of the dial screen.

Tip
The contact can be searched promptly with the first character you enter.



Contact list

Step 2. Touch the desired contact to talk in the Contact list.
Details can be viewed in the window on the right side of the screen.



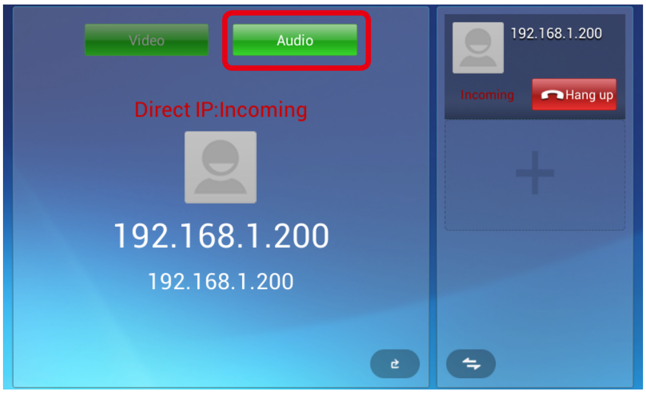
Step 3. Call the contact you touched.
3-1. To make an audio call, touch .
3-2. To make a video call, touch .

Tip
 is not used.

9.2.3. Receiving a call

[When receiving an audio call]

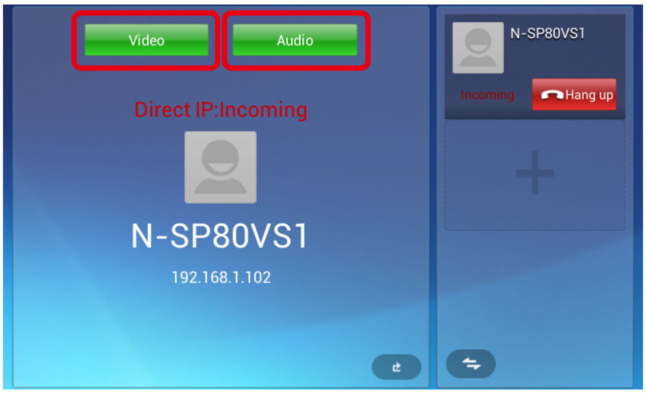
When an audio is received, Audio becomes active.



Step: Touch Audio or lift the handset to answer the call.

[When receiving a video call]

When a video call is received, Video and Audio become active.



Step 1. To make an audio response, touch Audio.

Note: Images cannot be viewed.

Step 2. To make a video response, touch Video.
Images can be viewed.
Then, you can start handsfree conversation. You can also make conversation with the caller by using the headset or lifting the handset.

9.2.4. Rejecting an incoming call

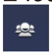
To reject an incoming call, touch 

on the call screen.

9.2.5. Call option

Call options are displayed by the icons as shown below.

The table below shows the descriptions of the call options.

Icon	Description
	Switches the current call between Hold and Restart each time the icon is touched and the display of the icon changes as well.
	Activates the transfer function.
	Mutes the microphone.
	Switches between Video call and Audio call each time the icon is touched and the display of the icon changes as well.
	Switches between speaker (handsfree) and handset conversations each time the icon is touched and the display of the icon changes as well.
	Extended function. 2 icons below appear when this icon is touched.  : Activates the 3 party conference function.  : Starts recording of conversations.

9.2.6. Video option

Images can be switched by the icon operation on the video screen during video conversation.

Icon	Description
	Switches to the full screen.
	Displays the local video.
	Displays the image of the conversation partner.

9.2.7. Call log

You can make the following 4 actions using the call logs.

- Checking the unconfirmed incoming call information notification displayed on the status bar
- Checking the call logs from the call history
- Registering a new contact from the call history
- Adding the phone number to the existing Contact list

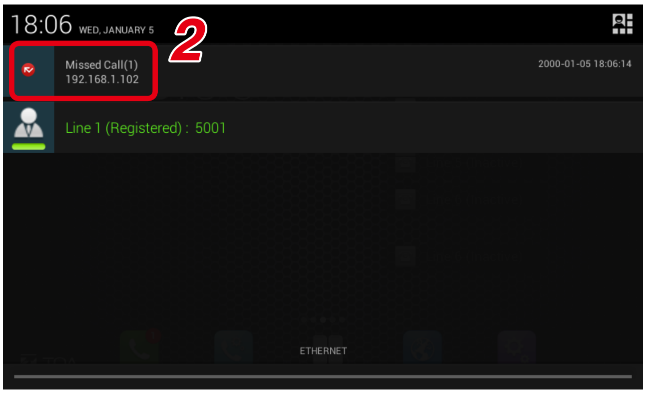
Shown below is operation.

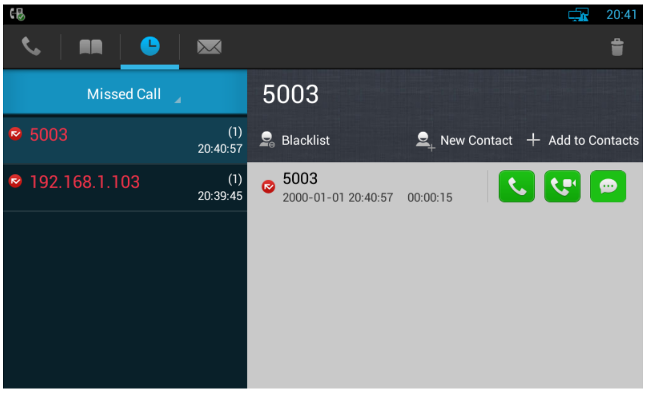
[Checking the unconfirmed incoming call information notification displayed on the status bar]

Unconfirmed incoming call information notification

Step 1. Swipe down on the status bar.
The Notification screen opens.

Step 2. Touch Missed Call(1) 192.168.1.102.
You can view the call history.



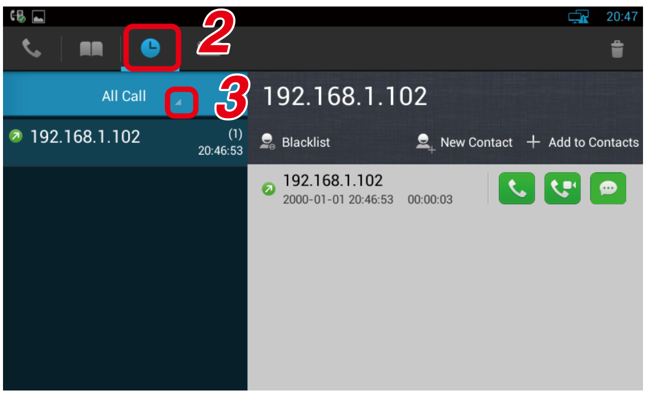


[Checking the call logs from the call history]

Step 1. Touch the phone icon on the main screen, or touch Apps on the main screen and select the phone icon.
A Dial screen opens.

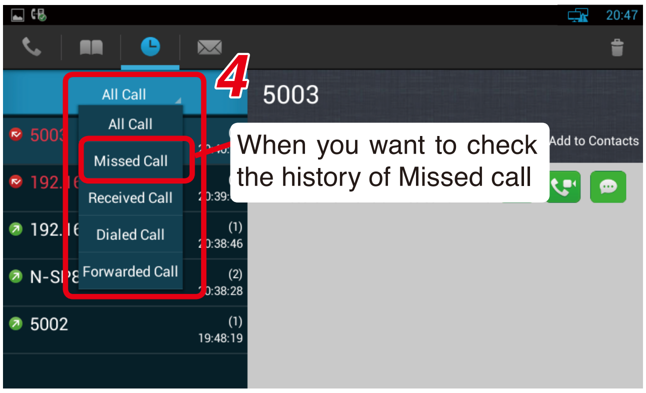
Step 2. Touch the call log icon.
All Call and the call history appear.

Step 3. Touch triangle mark of All Call.
Types of call logs appear in a pull-down menu.
You can select the type of call logs to be confirmed from All Call, Missed Call, Received Call, Dialed Call, and Forwarded Call.



Step 4. Touch the type of call logs to be confirmed from the pull-down menu.
Touching it causes the call history to appear on the left side of the screen, displaying the total number of logs in parentheses.

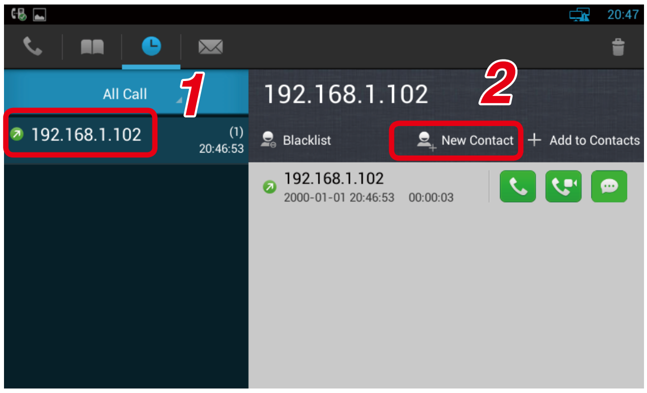
Step 5. Touch the phone number or contact to be confirmed.
The touched phone number or details of all call logs remained in the contact history can be viewed in the window on the right side of the screen.



[Registering a new contact from the call history]

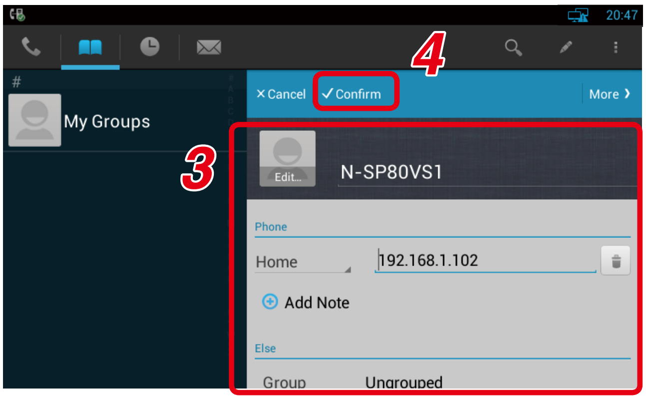
Step 1. Touch the number to be registered from the call history.

Step 2. Touch New Contact.
A registration page for a new contact appears.

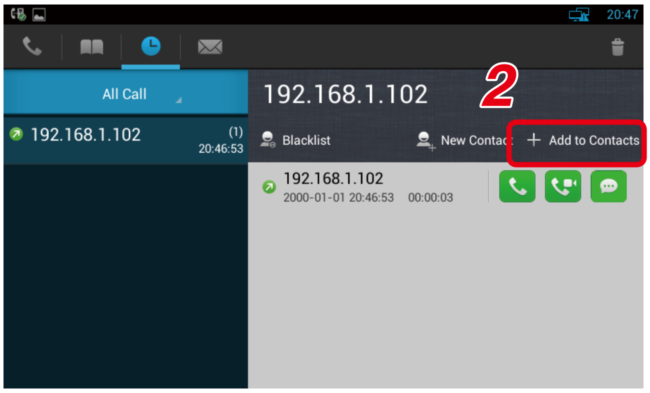


Step 3. Input the necessary items through a touch panel.

Step 4. Touch ✓Confirm.
A new contact list is created, saving the contact.



[Adding the phone number to the existing Contact list]



Step 1. Touch the phone number to add to the existing Contact list from the call history.

Step 2. Touch + Add to Contacts.
A Contact list screen appears.

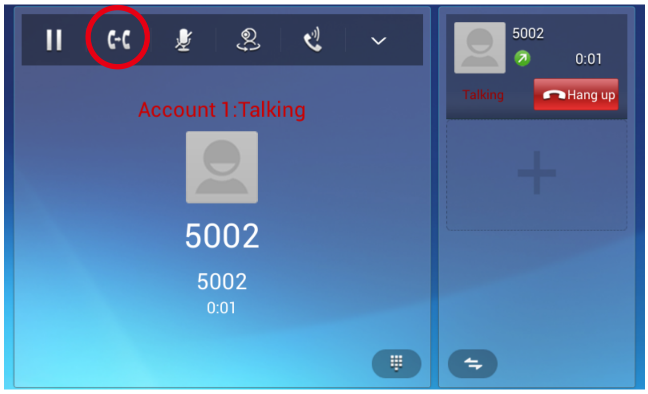
Step 3. Select the contact of which phone number you want to add.
The detailed screen of the selected contact appears.

Step 4. Enter the phone number to add in the displayed screen.

Step 5. Touch ✓Confirm.
The phone number is added to the Contact list.

9.2.8. Transfer (Blind transfer): Unconfirmed transfer (When in SIP server mode)

Shown below is the procedure to transfer a call without communicating with the transfer destination party.



Step 1. Press the station's Transfer button or touch [transfer icon] on the screen during conversation.
A dial screen for entering the phone number of the transfer destination party appears.

Conversations between the own station and the other party to be transferred are placed on hold.

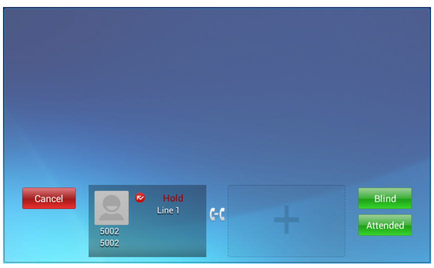
Step 2. Enter the phone number of the transfer destination party, then touch Blind.
The transfer destination party is called, then both transferred and transfer destination stations are engaged in direct communications with each other, terminating the own station call.

[Returning to the original conversation without transfer]

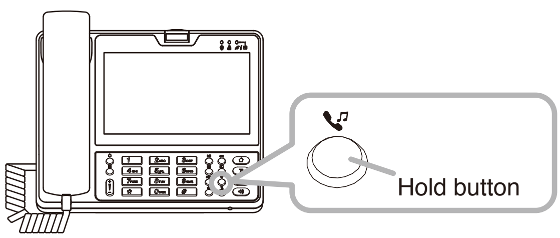
The original conversation is restored by following the procedures (1) through (3) below without touching Blind.

(1) Touch Cancel on the screen in **Step 2.**

(2) Touch Cancel on the displayed screen below.



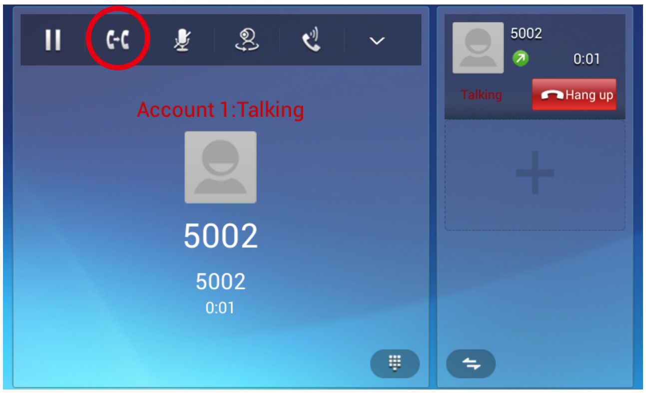
(3) Press the unit's Hold button (figure at right).



9.2.9. Transfer (Attended transfer): Confirmed transfer (When in SIP server mode)

Shown below is the procedure to transfer a call after making conversations with the transfer destination party.

Step 1. Press the station's Transfer button or touch [icon] on the screen during conversation.
A dial screen to call the transfer destination station is displayed and the conversations between the own station and the other party to be transferred are placed on hold.

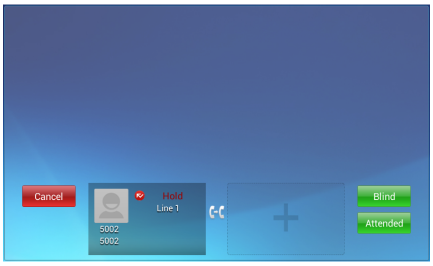


Step 2. Enter the phone number of the transfer destination party, then touch Attended.
A conversation screen with the transfer destination station appears.

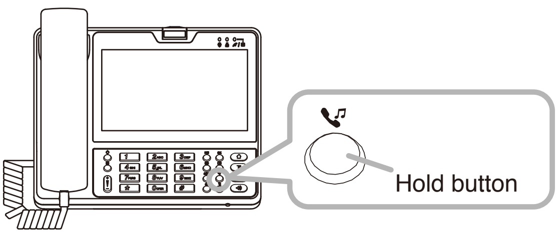
[Returning to restore to the original conversation without transfer]

The original conversation is restored by following the procedures (1) through (3) below without touching Attended.

(1) Touch Cancel on the screen in **Step 2**.
(2) Touch Cancel on the displayed screen below.

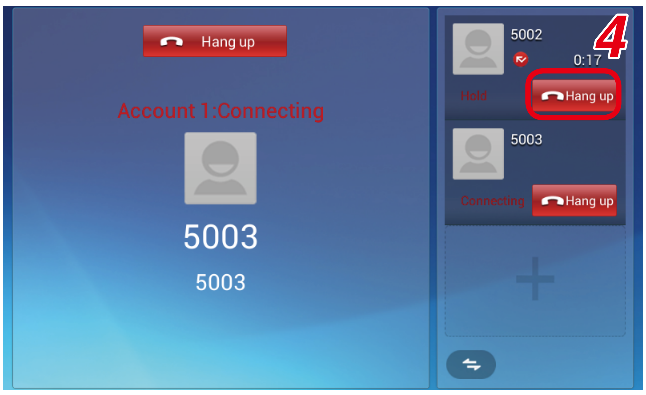


(3) Press the unit's Hold button (figure at right).



Step 3. Make conversations with the transfer destination partner.

Step 4. Hang up the handset or touch Hang up to end the conversation.
The station that was placed on hold (Station No. 5002 in **Step 1** above) and the transfer destination station (Station No. 5003 in **Step 2** above) are connected, enabling to start conversations.



9.3. Paging Call Operation

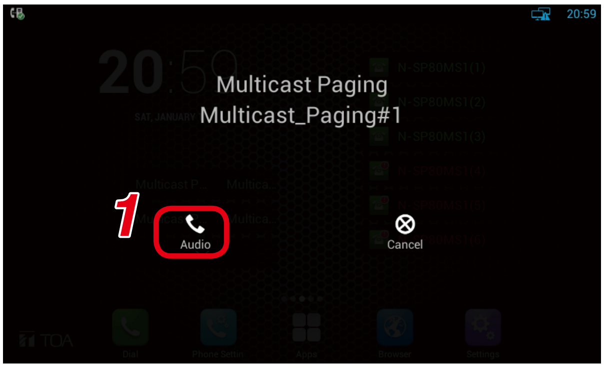
Paging calls can be made using the Multicast paging function.
Perform paging call operation by clicking the shortcut icon of the Multicast paging.

Shortcut icons of the Multicast paging

Tip
If the shortcut icon of the Multicast paging cannot be viewed on the screen, see p. 46 "Creating the shortcut of the EXT key."

Step 1. Click the shortcut icon of the Multicast paging.
A paging call screen appears.

(Paging call screen)

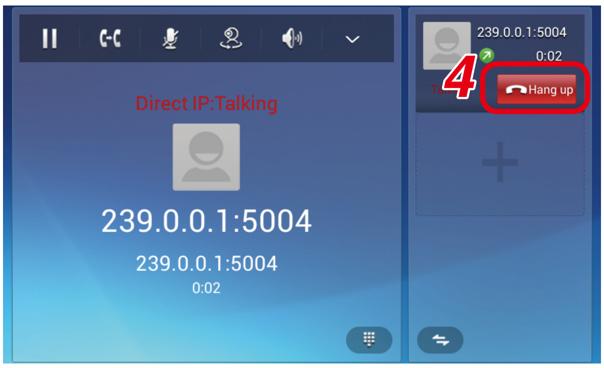


Step 2. Click Audio.
A dial screen appears.

Step 3. Lift the handset or press the Speaker button to start paging call.

Step 4. Hang up the handset or touch Hang up on the Dial screen to terminate paging call.

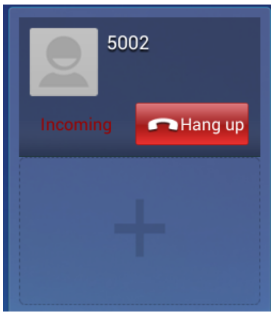
(Dial screen)



9.4. Other Functions and Operations

9.4.1. 3 party conference

- The multimedia station supports 3 lines of conversations.
- The line information during conversation appears in the windows on the right side of the conversation screen.



[3 party conference operation]

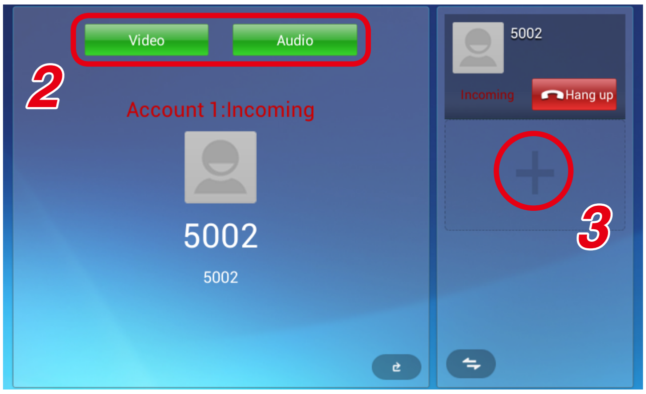
Shown below is an operation example on the screen when the multimedia station No. 5001 calls the multimedia station No. 5002 and the video door station No. 5003 to make 3 party video conference among them.

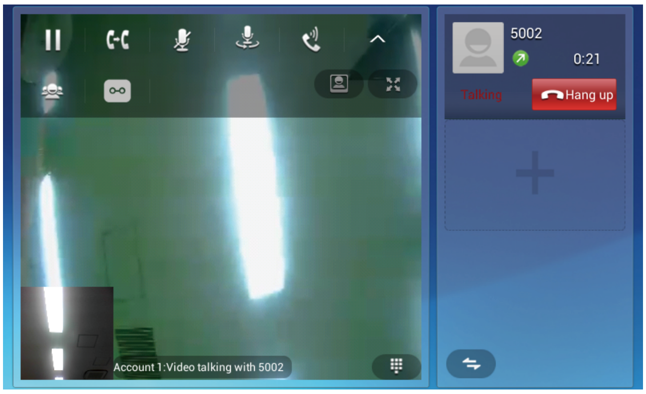
Step 1. Make conversations between 2 parties.
Operation of the conversation between 2 parties is the same as that of the normal call.

Tip
It does not matter whether the conversation is initiated by being called from a door station or another multimedia station.

Step 2. Touch Video or Audio.
A conversation screen appears.

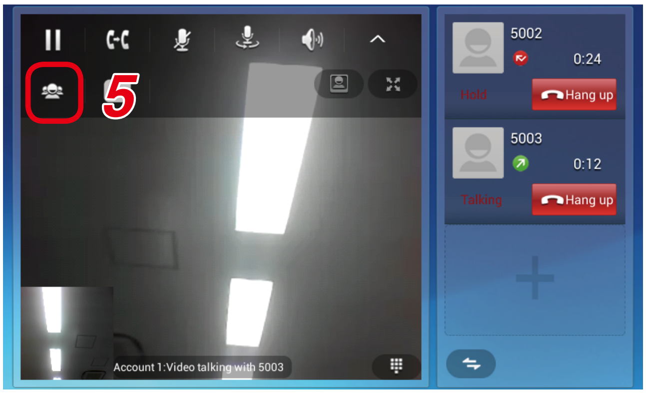
Step 3. Touch + button displayed in the window on the right side of the conversation screen.
The current conversation is placed on hold and the entry screen for calling a new station appears.





Step 4. Enter the station number to join the 3 party conference on the entry screen, then touch Video Call or Audio Call.
The conversation screen on which the added station number is displayed.

Step 5. Touch the 3 party conference icon displayed on the bar at the top of the conversation screen.



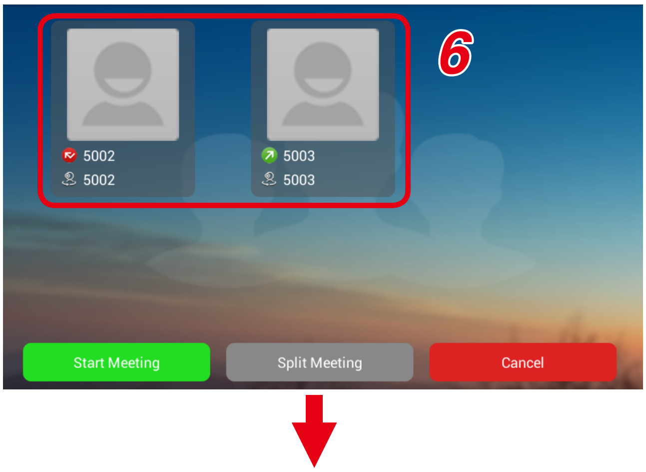
Tip

If icon is not displayed, touch icon displayed on the bar at the top.

The screen is switched and the icons of the station numbers that join the 3 party conference appears.

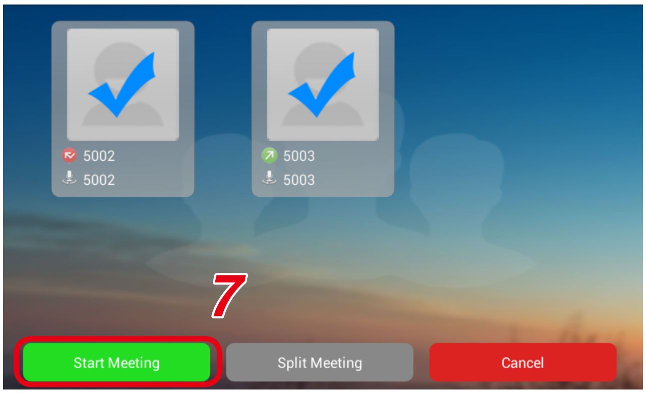
Step 6. Touch the icons of the 2 station numbers to join 3 party conference.

Check marks are put on the clicked icons.



Step 7. Touch Start Meeting.

3 party conference starts and the screen changes to the conversation screen.

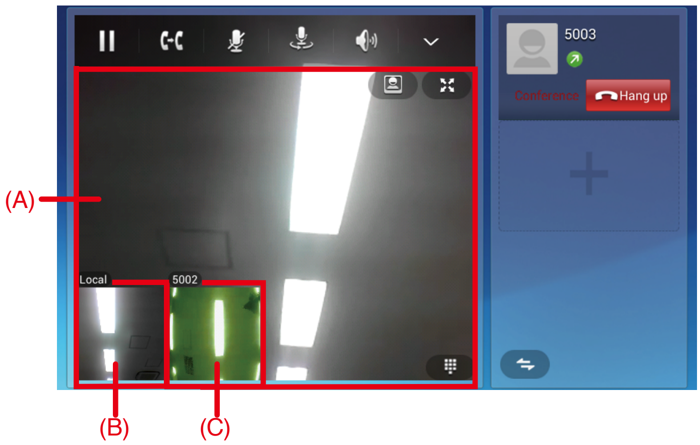


The conversation screen of the multimedia station during 3 party video conference shows the screen of each station in a divided display as shown at right.

(A): Camera image of the station added to the call destination station in **Step 4**.

(B): Camera image of the own station. "Local" indication appears at the upper left corner of the split screen.

(C): Camera image of the partner station which was engaged in 2 party conversation in **Step 1**. The station number appears at the upper left corner of the screen.



Tips

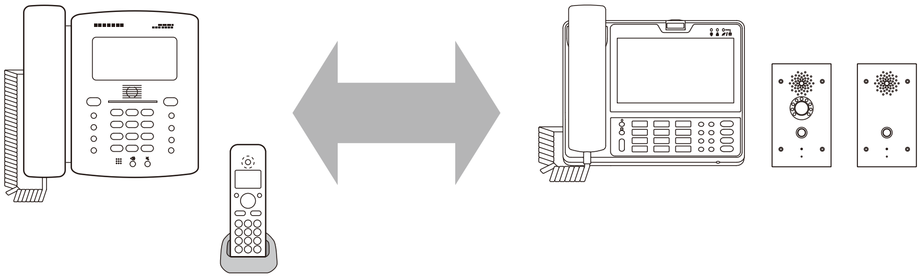
- Touch the screen (C) to switch the image displayed in the screen (C) to that displayed in the screen (A).
- The screens (A) and (C) are not displayed when the partner station is a door station.

9.4.2. Connecting to a third-party VoIP

The SIP compliant VoIP station can be generally connected to the N-SP80 Series station.
Basic operations (Call, call reception, hold, and transfer, etc) are the same as those of the N-SP80MS1.

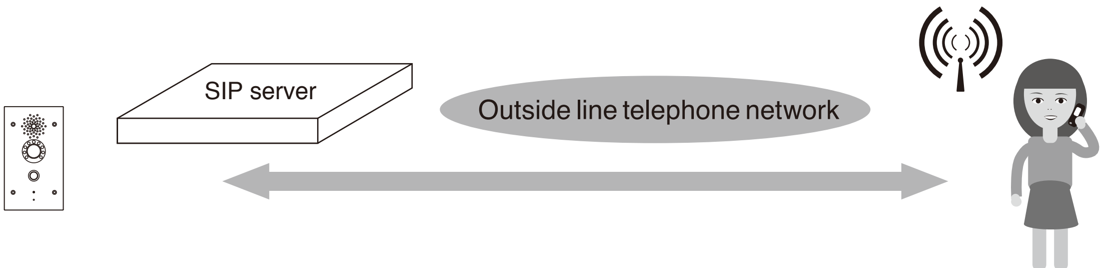
Notes

- As connection compatibility must be ensured, perform connection confirmation in advance.
- Confirm that the audio codec supports the VoIP devices to use.
  The audio codec of the N-SP80 series is fixed to G.722 when using the station in Peer-to-peer mode.



9.4.3. Connection with outside line (in SIP server mode)

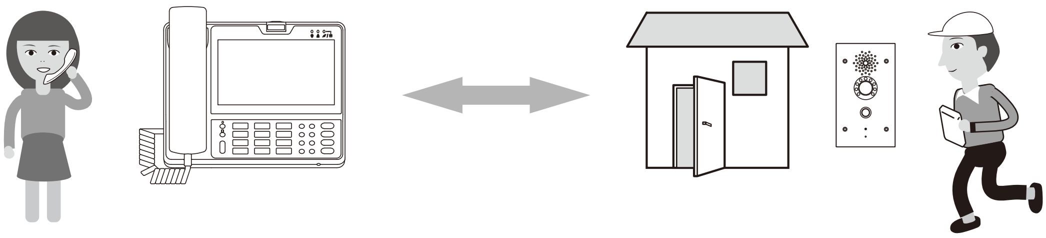
The SIP server needs to have configurations of outside line (Telephone line). When settings and operations are performed according to the assignment of the special number to the outside line, the station can call the outside line telephone.
For example, when the special number assigned to the outside line is "0," the outside line telephone can be called directly provided that the call button on the door station is set to "0-080-****-####."



9.4.4. Door remote control

When entering the preset number* assigned to the door station during conversation with it, you can remotely lock or unlock the door near the station engaged in conversation.
The sound or message which informs that the lock is opened or closed can also be output from the door station.

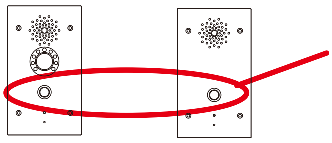
* The number set in the DTMF of the p. 59 "Intercom - Relay&Input"



10. DOOR STATION'S FUNCTIONS AND OPERATIONS

10.1 Functions for Conversations and Operations

10.1.1. Making a call



Press the Call button to call up the preset station.

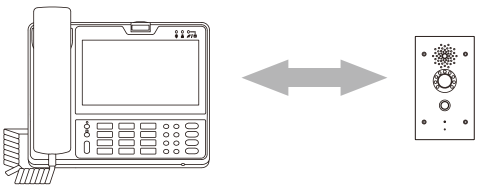
(N-SP80VS1) (N-SP80AS1)

10.1.2. Receiving a call

An incoming call can be received by the automatic response function.
As the call is automatically received, you do not need any operation.

10.1.3. Cancelling a call and conversations

Pressing the Call button during a call or conversations cancels the operation in progress.



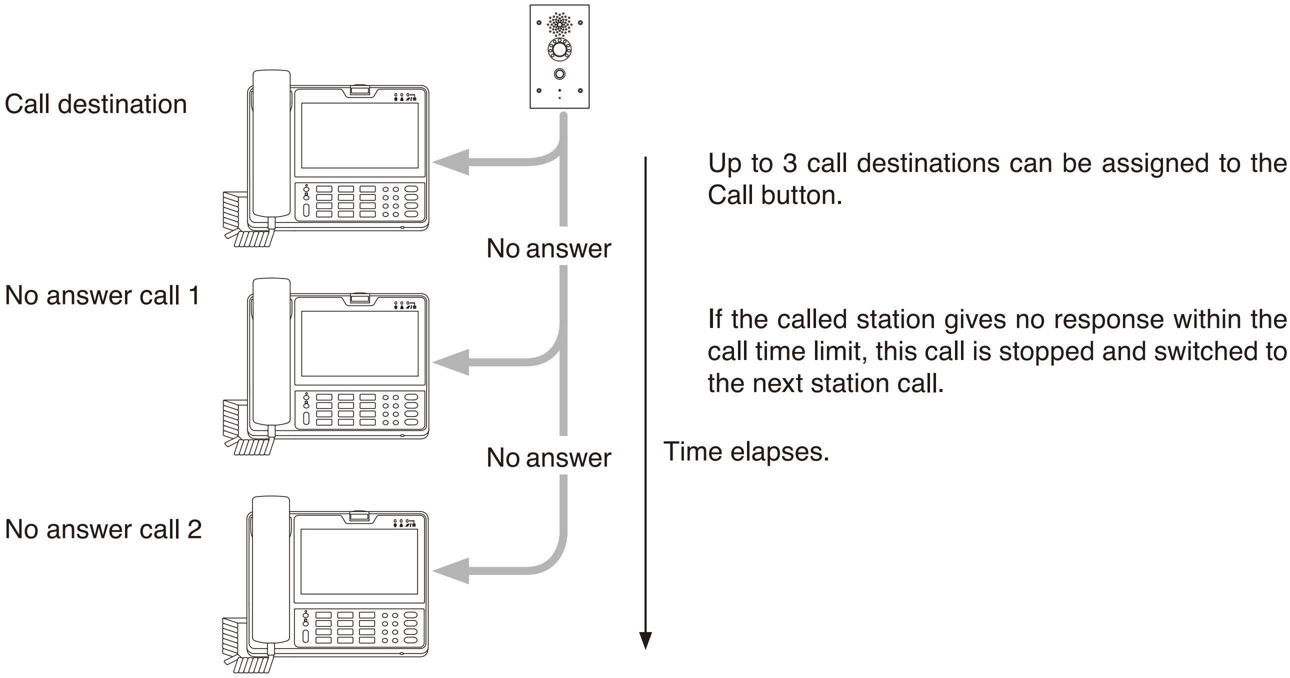
10.2. Paging Function

Paging call can be automatically received according to the preset priority level. Any operation is not required for this function.

10.3. Other Functions and Operations

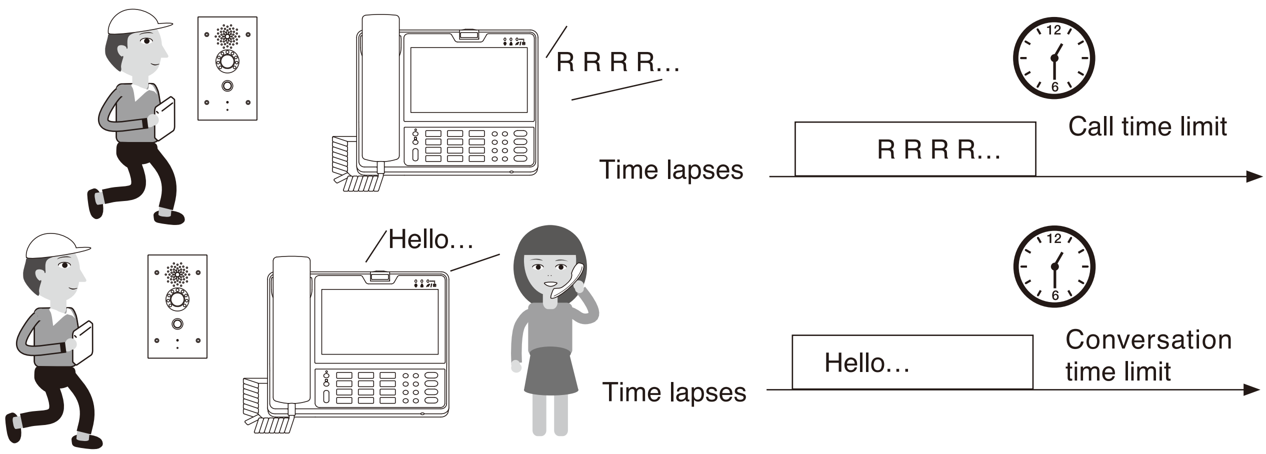
10.3.1. No answer forward function

Up to 3 call destinations can be assigned to the Call button. If the first call destination does not respond, it is switched to the second call destination. If the second call destination does not respond, it will be switched to the third call destination.
Set the transfer destination using [Account Selection: No answer call 1 and 2] of "Intercom - Basic" on p. 57.



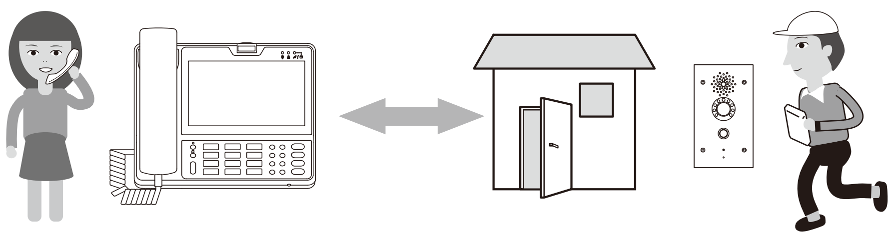
10.3.2. Time limit

Time limit can be set for the call and conversation times.
Set the time limit using [Time Limit: Call time-out and Conversation time-out] of "Intercom - Basic" on p. 57.



10.3.3. Door remote control

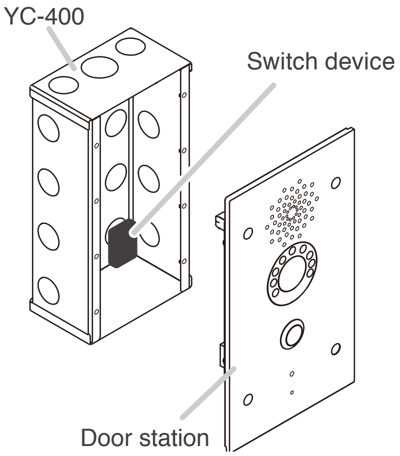
The door remote control such as opening and closing of the nearby door can be received from the multimedia station during conversation.
Set the relay output of the door remote control using [Relay] of "Intercom - Relay&Input" on p. 59.



10.3.4. Call activation from an external device

The door station can make a call to the station other than those assigned to the Call button when connecting to an external device or switch.

Note
To perform this call activation, prepare the optional YC-400 4 size back box.
The door station should be installed into the wall using the YC-400. (See p. 30 "Installation of Door station".)



(Example)
Install a switch device between the door station and the back box in advance.
Connect the switch device to the external control input so that call can be executed when the station is removed by a mischief.

10.3.5. Change in various function sounds

You can change or stop the sounds shown below. The sound can be changed by uploading the audio file from the Web setting screen. (See p. 67.)

- Incoming call sound of the door station
- Call transmission sound
- Door remote control sound

11. INSTALLATION

11.1. Safety Precautions for The Multimedia Station

- Do not install the unit near the heat generating equipment.
- Do not install the unit at the locations which could stand in someone's way or strike someone.
- Never use the unit near the locations where water is handled such as bath room, rest room, and kitchen.

11.2 Installation of Door Station

The door station is designed to be mounted into the wall in conjunction with the YC-400 wall recessed 4 size back box. For mounting, follow the procedure below.

Step 1. Remove a knockout hole in the YC-400.
Before mounting the YC-400 into the wall, punch out the knockout hole with a screwdriver or other tool to make a cable entry hole.

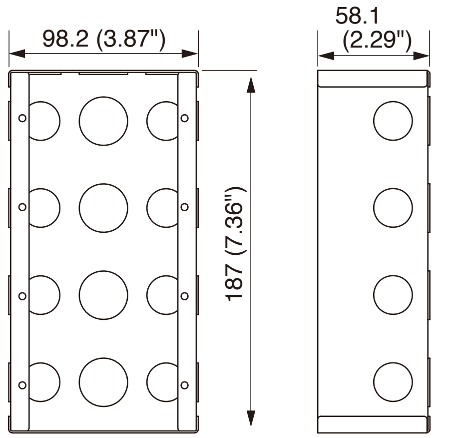
Step 2. Make a mounting hole in the wall such as a gypsum board, then mount the YC-400 into the wall.

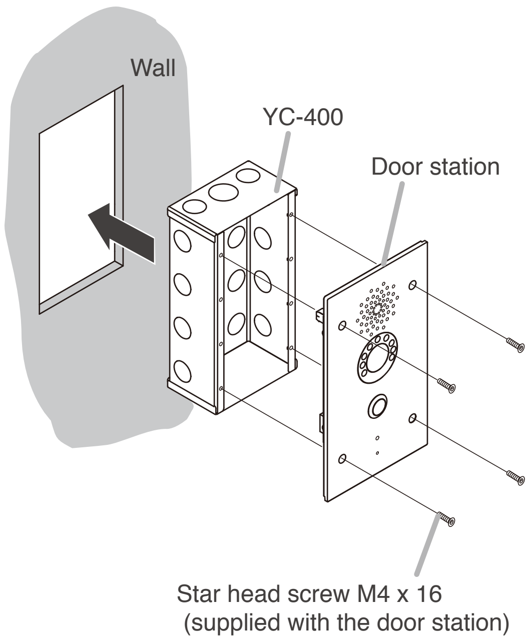
Tip
Install the YC-400 at the appropriate height from the floor (approx. 1.5 m or 5 ft).

[YC-400 dimensional drawing]

[Front] [Side]

Unit: mm (in)

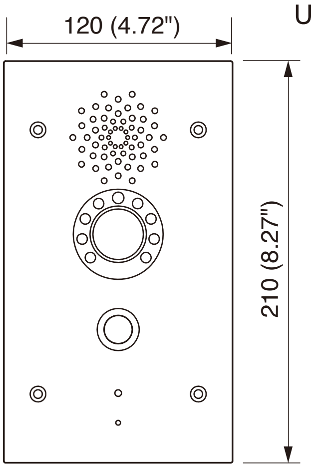




[Door station dimensional drawing]

Note
The N-SP80VS1 and the N-SP80AS1 have the same dimensions.

Unit: mm (in)



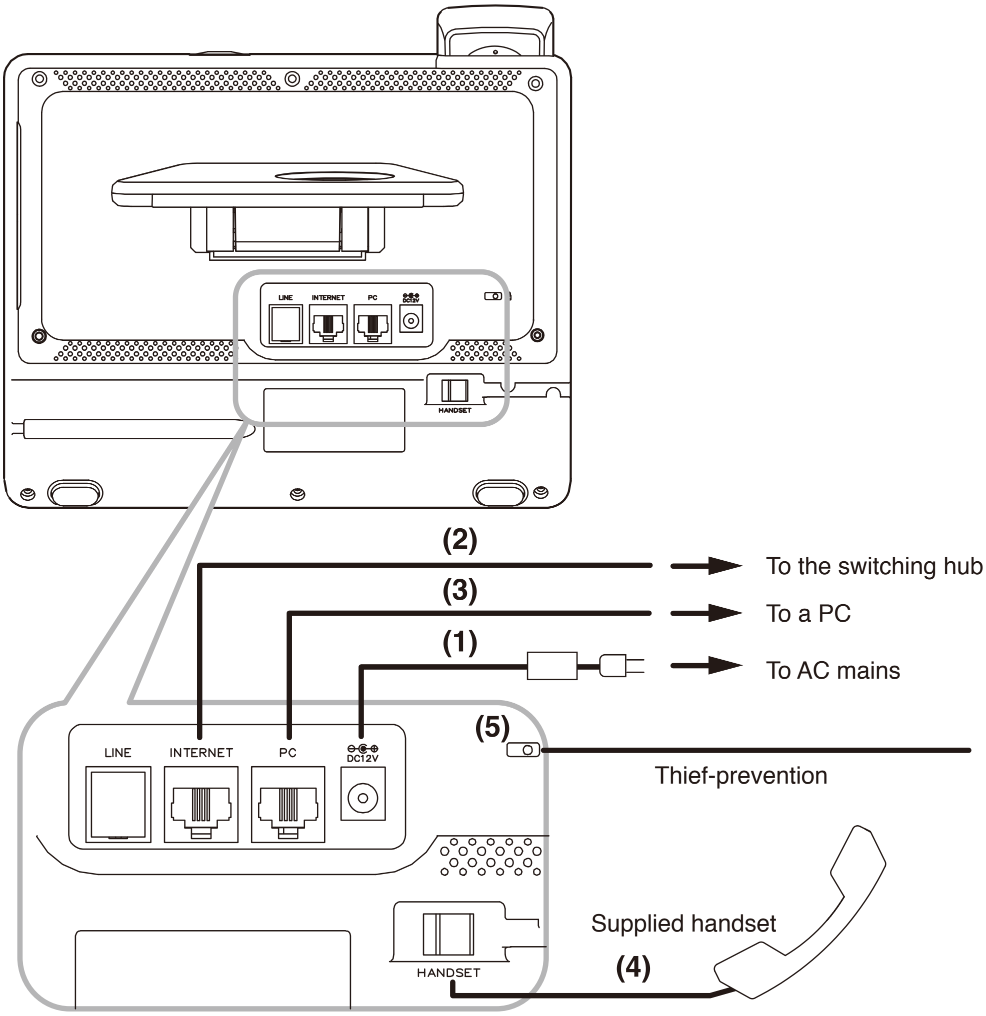
Step 3. Run the connection cable through the cable entry hole in the YC-400, then connect it to the door station.

Step 4. Secure the door station to the YC-400.
Use the screws supplied with the door station and the star head screwdriver.

12. CONNECTION

12.1. N-SP80MS1

[Rear]



12.1.1. Power supply connection

Power is supplied from the AC adaptor or the PoE-compatible switching hub.
For the power supply from the switching hub, see p. 32 "(2) LAN connection terminal."

(1) DC input terminal

Connect the AC adapter.*
* Use the optional AC adapter AD-1215P or AD-5000-2 (or its equivalent). As for the usable adapter, consult your TOA dealer.

⚠ **CAUTION**	The use of the AC adapter other than the specified one may cause a fire.

Tip
If both the AC adapter and the PoE switching hub are connected, the power will be supplied from the one that has started feeding first.

12.1.2. Switching hub and PC connections

(2) LAN connection terminal
Connect this terminal to the 100BASE-TX-compatible network.
Use the Ethernet RJ-45 connector for connection.

As this terminal can be connected to PoE switching hub, power can be supplied from the PoE switching hub when this terminal is connected to it.
In this case, use the switching hub meeting the following specification.
Specification of the usable PoE switching hub: IEEE802.3af compliant

⚠ CAUTION	When power is supplied from the PoE switching hub, be sure to use the one meeting the specified specification. The use of the switching hub other than the specified one may cause a fire.

Tip
If both the AC adapter and the PoE switching hub are connected, the power will be supplied from the one that has started feeding first.

(3) PC connection terminal
Connect a PC to this terminal.
Use the Ethernet RJ-45 connector for connection.
This terminal can also be connected to the 100BASE-TX-compatible network.

Note
This terminal is not PoE-compatible.

12.1.3. Other connections

(4) Handset connection terminal
Connect the supplied handset to this terminal.

(5) Security slot
Connect a commercially available theft preventing wire to this slot as needed.
This is a Kensington lock slot.

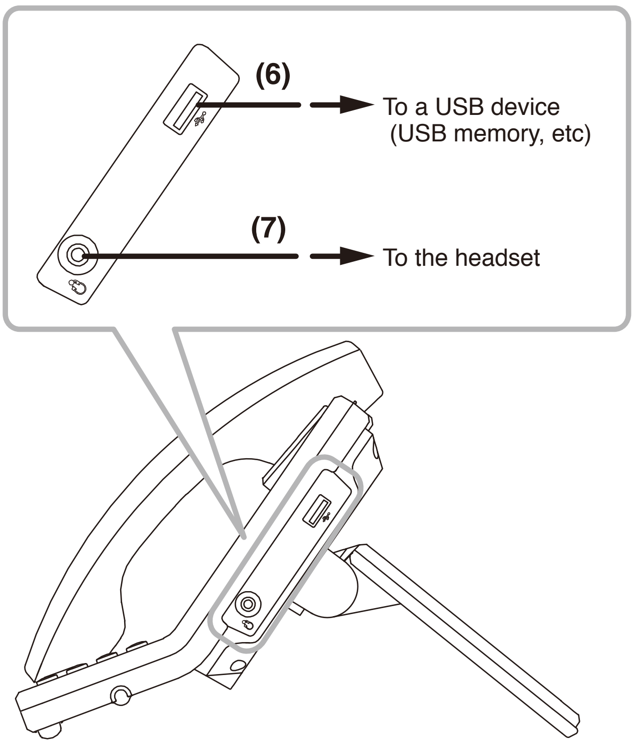
12.1.4. USB device connection

(6) USB connection terminal
Connect a USB device such as a USB memory to this terminal.
This terminal is USB 2.0 compatible.

12.1.5. Headset connection

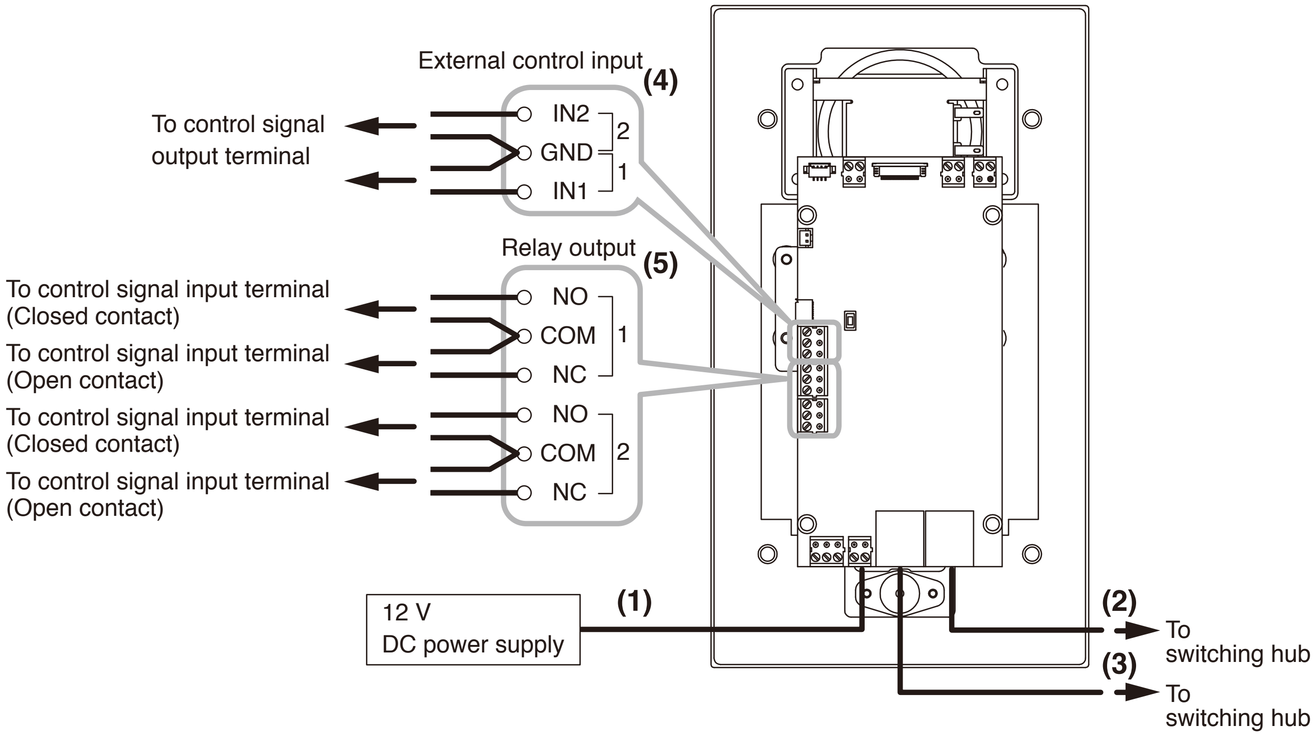
(7) Connect a headset to this terminal.
Usable headset: 16/32 Ω,
3.5 mm (0.14") dia. mini plug (3 pins)

N-SP80MS1 unit (Right side)



12.2. N-SP80VS1, N-SP80AS1

[Rear]



12.2.1. Power supply connection

Power is supplied from the 12 V DC power supply or the PoE-compatible switching hub.
For the power supply from the switching hub, see below "(2) Ethernet connection terminal."

(1) DC input terminal

Connect the 12 V DC power supply to this terminal. This terminal has no polarity. Connect the "+" and "–" cables to each terminal.
Prepare the 12 V DC power supply separately.

Tip

If both the 12 V DC power supply and the PoE switching hub are connected, the power will be supplied from the one that has started feeding first.

12.2.2. Switching hub connection

(2) Ethernet connection terminal (PoE compatible)

Connect this terminal to the 100BASE-TX-compatible network.
Use the Ethernet RJ-45 connector for connection.

As this terminal can be connected to PoE switching hub, power can be supplied from the PoE switching hub when this terminal is connected to it.
In this case, use the switching hub meeting the following specification.

Specification of the usable PoE switching hub: IEEE802.3af compliant

⚠ CAUTION	When power is supplied from the PoE switching hub, be sure to use the one meeting the specified specification. The use of the switching hub other than the specified one may cause a fire.

Tip

If both the 12 V DC power supply and the PoE switching hub are connected, the power will be supplied from the one that has started feeding first.

(3) Ethernet connection terminal

Connect this terminal to the 100BASE-TX-compatible network.
A PC can also be connected to this terminal when performing various settings.
Use the Ethernet RJ-45 connector for connection.

Note

This terminal is not PoE-compatible.

12.2.3. Other connections

(4) External control input terminal

No-voltage make contact, Open voltage: 30 V DC, Short-circuit current: 10 mA, Short-circuit duration: 200 ms or more
Connect such a device that outputs the control signals as an external control switch or sensor to this terminal.
Two channels of the control signals can be applied. The table below shows the combination of the terminals.

	Combination of the terminals
Control input 1	IN1 and GND
Control input 2	IN2 and GND

To use these terminals, you need to make the system setting on the browser. For the details, see p. 59.

(5) Relay connection terminals 1 and 2

Contact type: Relay contact output, Contact capacity: 30 V DC, 0.5 A
Connect the device that is controlled by the relay output such as an electric lock to this terminal.
Two channels of the control signals can be output. Also, a closed contact or open contact can be selected for each signal depending on the terminal to be connected.
The table below shows the combination of the terminals.

	Combination of the terminals	
	When the closed contact is selected	When the open contact is selected
Relay output 1	Relay output 1's NO and COM	Relay output 1's NC and COM
Relay output 2	Relay output 2's NO and COM	Relay output 2's NC and COM

Note

Never connect any devices to both the closed contact and open contact terminals of the same relay output at the same time, as this could result in the unit failure.

To use these terminals, you need to make the system setting on the browser. For the details, see p. 59.

13. SYSTEM SETTING USING A WEB BROWSER

13.1. Before Performing System Setting

Access the web servers of all devices using the web browser, then perform settings for each device. Settings cannot be performed while offline.
Preparations shown below are required before starting settings.

- MAC address and IP address assignment plan for the devices to use (Phone number assignment plan is also needed when in SIP server mode.)
- SIP server setting information (when in SIP server mode)
- IP address setting of a PC used in the system setting (Set the IP address so as to belong to the same system network.)

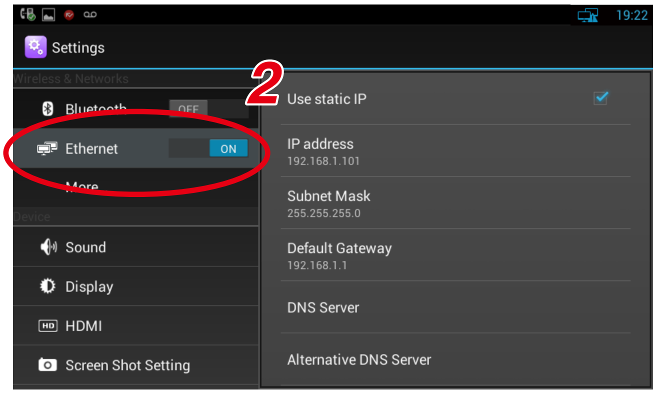
  [Verified browsers (Version)]
  · Microsoft Edge (38.14393.1066.0)
  · Google Chrome (Version 63.0.3239.132)
  · Fire fox (Version 58.0.2)

13.2. Confirming the IP Address of Each Device

13.2.1. N-SP80MS1

Step 1. Touch the Settings icon on the main screen.
The setting screen opens.

Step 2. Touch the Ethernet item.
The current IP address appears.

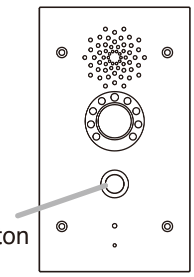


13.2.2. N-SP80VS1, N-SP80AS1

After turning on the power, perform the operation shown below within the valid time during which the IP address can be confirmed.
You can set the valid time on the station's web browser setting screen. (See p. 67 "IP Announcement.")

Step: Hold down the Call button for 3 seconds or more.
The IP address of this door station is announced.
It will be announced like "IP one nine two dot, …" in English.

Note
After tuning on the power, if you want to execute this function after the valid time expires, turn off and on the power again or press the Reset button on the unit’s rear panel.


Call button

13.3. N-SP80MS1 Setting

13.3.1. Logging in

Connect to the unit’s Web server by using the IP address.
When the IP address is 192.168.1.101, enter "http://192.168.1.101" to make connection.
For the method to confirm the IP address, see p. 35 "Confirming the IP address of Each Device."

User name and password settings are as follows.

User name: N-SP80
The user name is fixed. It cannot be changed.
Password: guest (default setting)
The password can be changed.
Enter it with up to 63 characters.
Notes
• Unusable characters : &, %, ' , =
• Password is case-sensitive.

Step 1. Start the PC's browser.

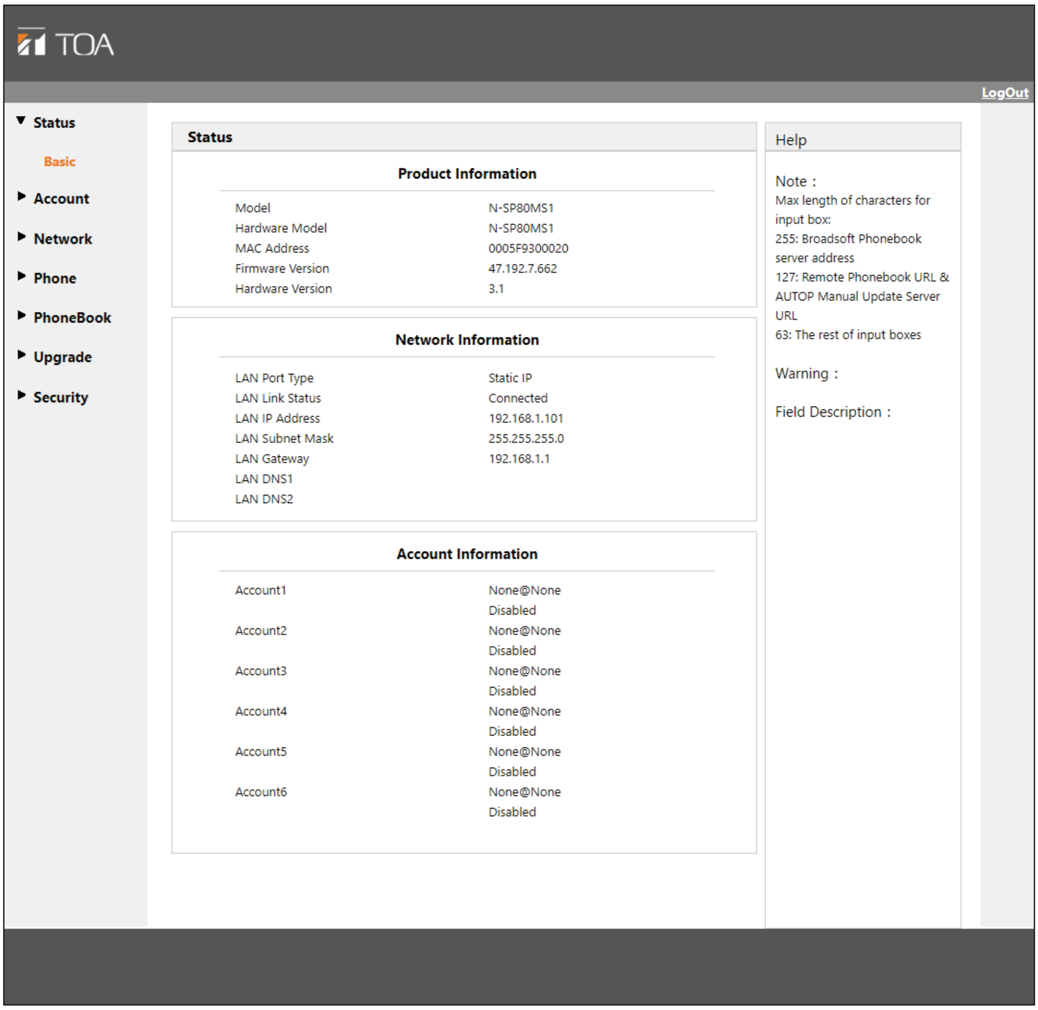
Step 2. Enter the IP address in the address bar.

Tip
The default IP address is 192.168.1.101. (Subnet mask is 255.255.255.0.)

A login screen appears.

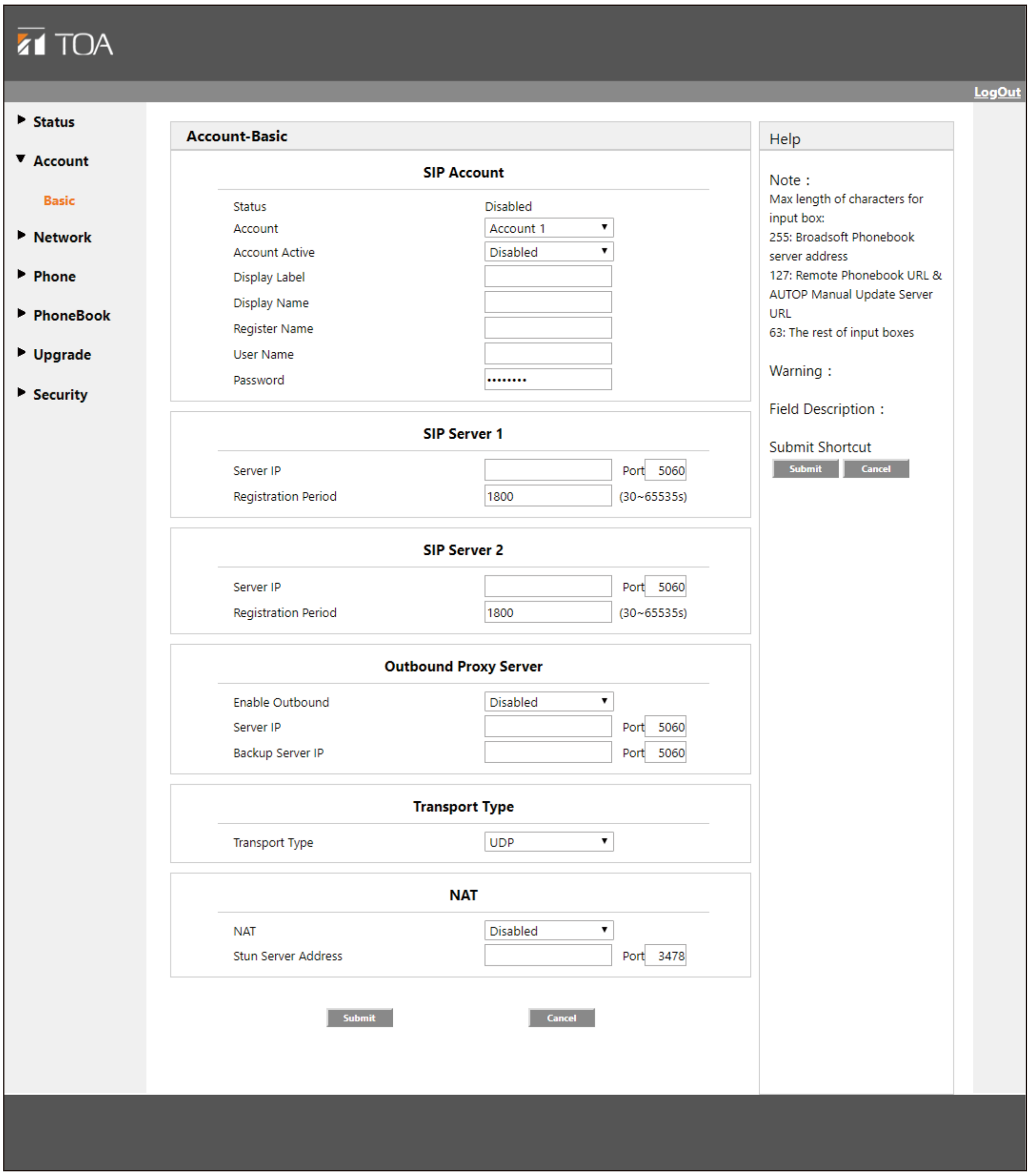
Step 3. Enter the user name and password, then click Login.

13.3.2. Status - Basic



Item	Description
Product Information	Displays the following product information: • Model • Hardware model • MAC address (Physical address of the IP device) • Firmware version and Hardware version.
Network Information	Displays the following network status (LAN port) information of the device: • LAN port type (one of DHCP static, and PPPoE) • LAN link status • LAN IP address • LAN Subnet Mask • LAN Gateway • LAN DNS1 • LAN DNS2
Account Information	Displays the account information and registration status (account user name, registered server address, and registration result) of the device.

13.3.3. Account - Basic

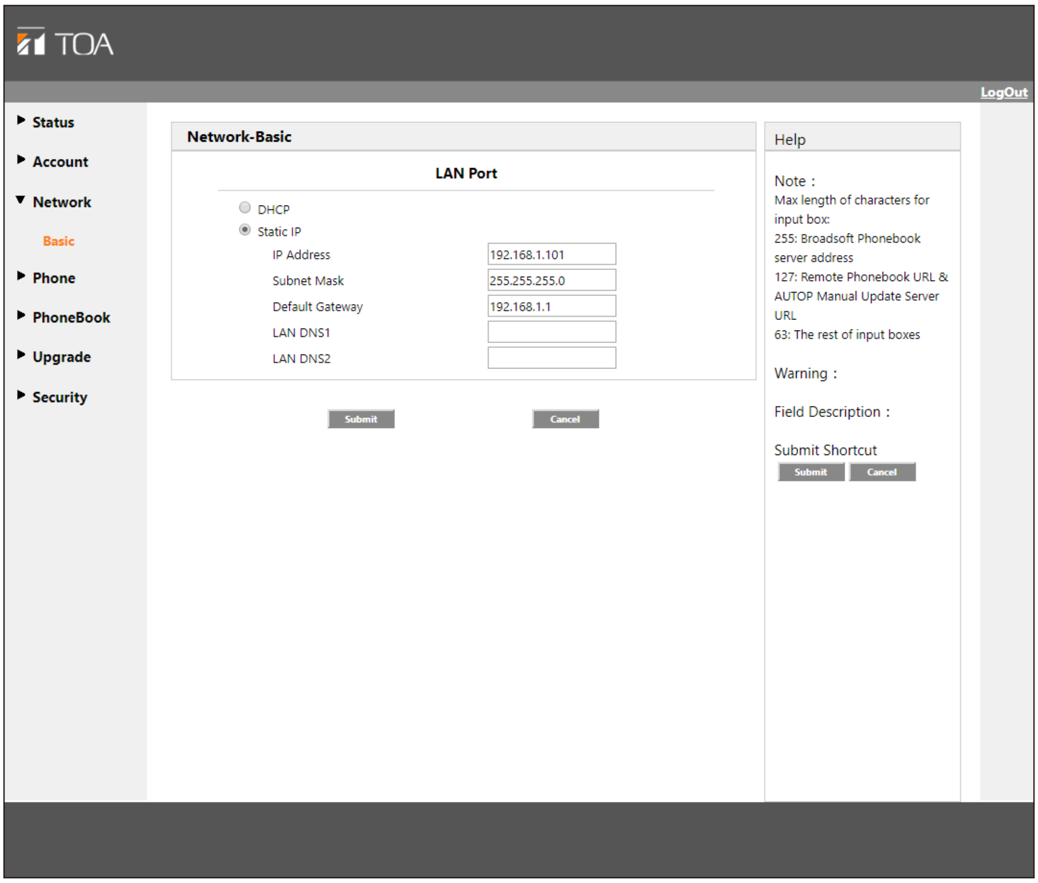


Item	Description
SIP Account	Displays or sets the specific account setting. Status: Displays the registration result. Account: Select the SIP account to set. (Select one from Account 1 through 6.) Account Active: Select Enabled/Disabled of each account. Display Label: Displayed on the station's LCD screen. Display Name: Sent to the called party and displayed. Register Name: Use the name to be registered (set) in the SIP server for authentication. User Name: Use the name to be registered (set) in the SIP server for authentication. Password: Use the password to be registered (set) in the SIP server.

Item	Description
SIP Server 1	Displays or sets the Primary SIP server setting. Server IP: SIP server address that is URL or IP address Registration Period: An interval to periodically send the registration (REGISTER) to the SIP server. Continues to retain the registration in the SIP server when the station sends registration (REGISTER) again within the registration (REGISTER) maintaining period on the SIP server.
SIP Server 2	Displays or sets the Secondary SIP server setting. This is a backup server so that the information communication station can be registered at the Secondary SIP server even if the registration to the Primary SIP server fails. **Note** The Secondary SIP server is used as a backup server. If there is no SIP server for backup in a user environment, these corresponding fields are left blank. Displays or sets the Primary SIP server setting. Server IP: SIP server address that is URL or IP address Registration Period: An interval to periodically send the registration (REGISTER) to the SIP server. Continues to retain the registration in the SIP server when the station sends registration (REGISTER) again within the registration (REGISTER) maintaining period on the SIP server.
Outbound Proxy Server*[1]	Displays or sets the outbound proxy server. **Note** • Once set, all SIP request messages are forcibly sent to the outbound proxy server from the multimedia station. • When the external SIP server (such as one for an outside line) is not used, leave the corresponding fields blank. Enable Outbound: Sets Enabled/Disabled for the connection to the outbound server. Server IP: Sets the IP address to the outbound server to connect. Buckup Server IP: Sets the backup server's IP address if there is a backup server for the outbound server.
Transport Type	Displays or sets the transfer type of the SIP message. The type is factory-preset to "UDP." UDP: An unreliable but highly effective transfer layer protocol. TCP: A reliable but less effective transfer layer protocol.
NAT	NAT (Network Address Translation): Displays or makes settings. Stun Server Address*[2] : One of the solutions to solve NAT problem. **Note** The default NAT is set to Disabled.

*[1] Used to receive all Start request messages and transfer them to the designated SIP server.
*[2] Stun stands for Simple Traversal of UDP over NAT.

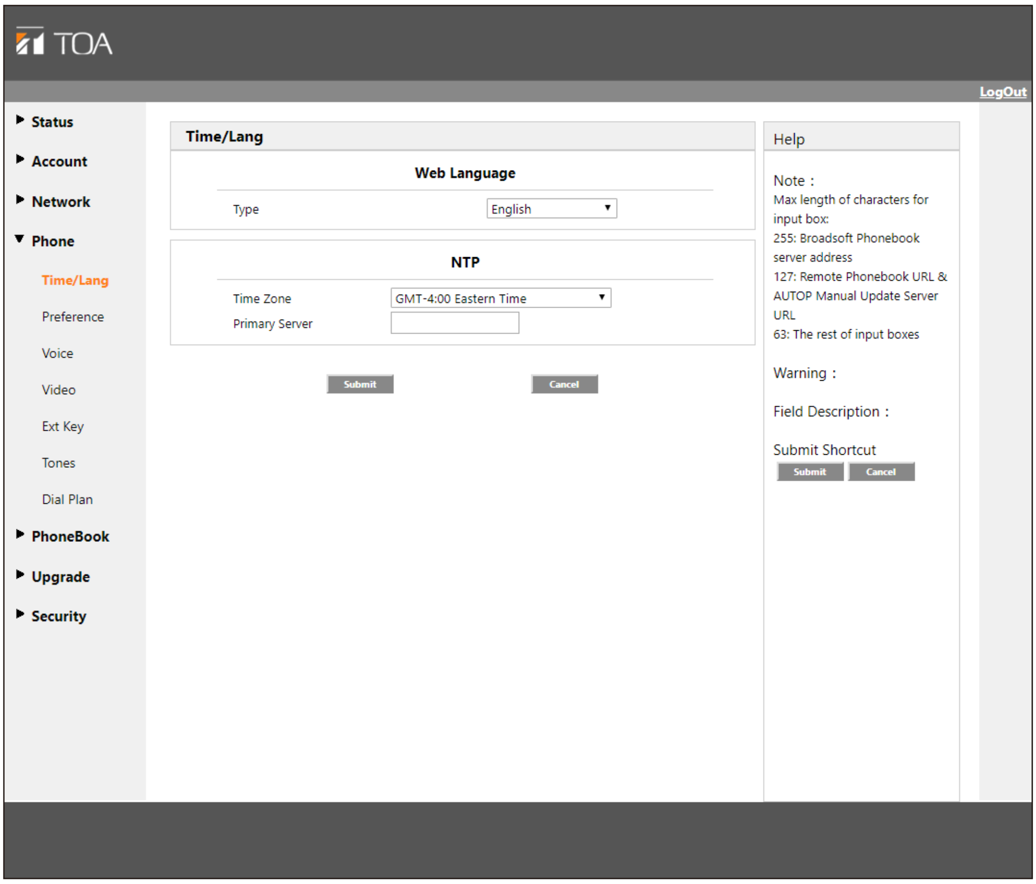
13.3.4. Network - Basic



Item	Description
LAN Port	Displays or sets the LAN port setting. DHCP: Automatically acquires the IP Address, Subnet Mask, Default Gateway, LAN DNS1*, and LAN DNS2* from the DHCP server. Static IP: It is necessary to manually set the IP Address, Subnet Mask, Default Gateway, LAN DNS1* and LAN DNS2*.

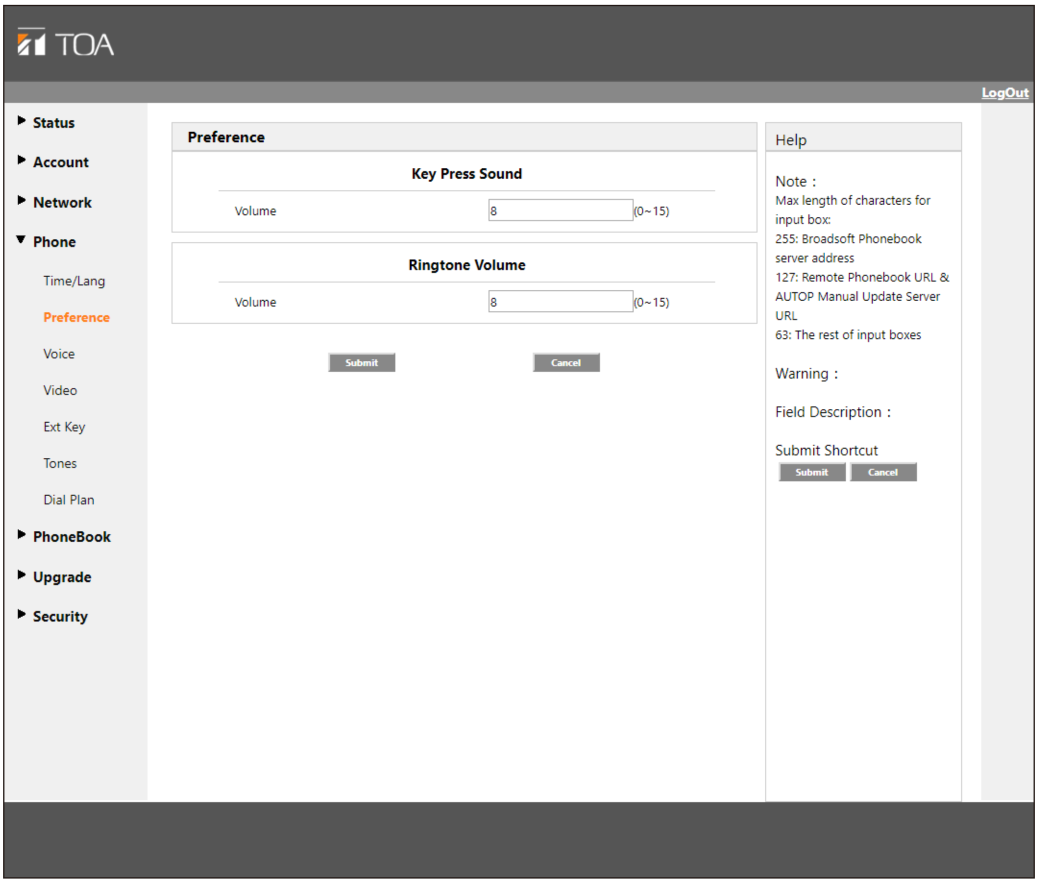
* DNS server address

13.3.5. Phone - Time/Lang



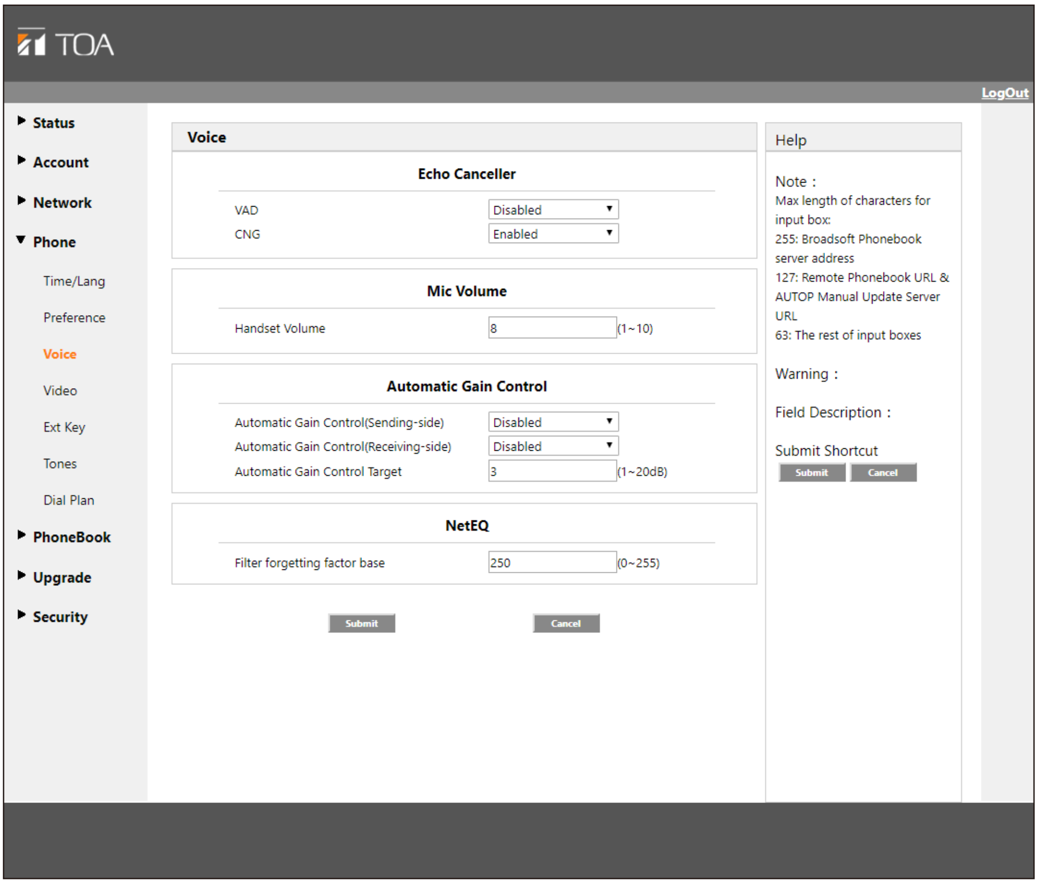
Item	Description
Web Language	Select the language used on the Web setting screen.
NTP	Performs the NTP setting for maintaining station's clock information. Time Zone: Select a reference time (time zone) to be set to the device. Primary Server: Designate the NTP server address to connect to.

13.3.6. Phone - Preference



Item	Description
Key Press Sound	Sets the key operation sound volume. Volume: Effective volume range is 0 – 15 and the default volume level is "8."
Ringtone Volume	Sets the ringing tone (call sound) volume. Volume: Effective volume range is 0 – 15 and the default volume level is "8."

13.3.7. Phone - Voice

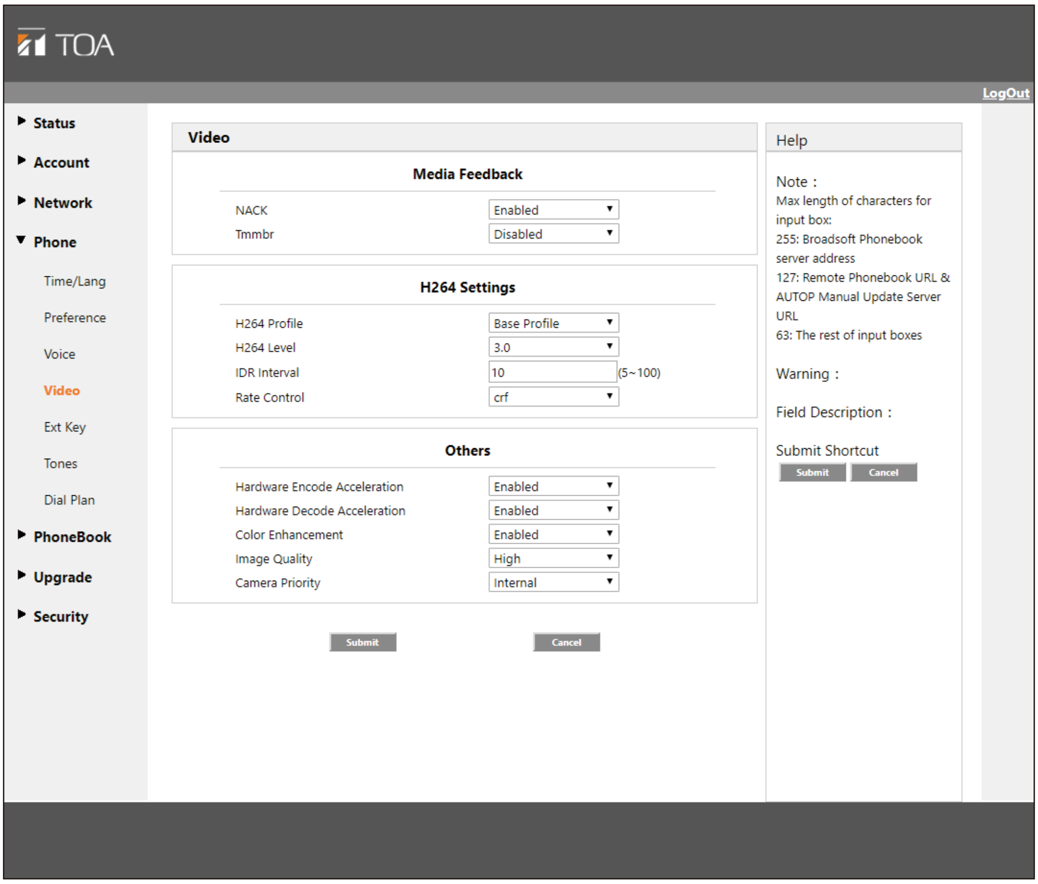


Item	Description
Echo Canceller	Eliminates an acoustic echo from the voice communications to improve the speech quality. VAD*1 (Voice activity detection): Detects presence or absence of the human voice during conversations using the multimedia station. "Disabled" is selected by default. CNG*2 (Comfort noise generation): Generates a comfortable background noise for the voice communication during the silent period in the conversation using the multimedia station. This is a part of the silence suppression or VAD handling of the VoIP technology. "Enabled" is selected by default.
Mic Volume	Sets the microphone volume in handset mode. Handset Volume: Effective volume range is 1 - 10 and the default volume level is "8."
Automatic Gain Control	The multimedia station automatically adjusts the gain of the amplifier circuit via signals. Automatic Gain Control (Sending-side): "Disabled" is selected by default. Automatic Gain Control (Receiving-side): "Disabled" is selected by default. Automatic Gain Control Target: 1 dB – 20 dB, "3 dB" is selected by default.
Net EQ	Filter forgetting factor base: 0 – 255, "250" is selected by default.

*1 When "Silence" period is detected, the VAD efficiently replaces it with the special packet showing that silence is present. This facilitates the audio processing, disabling some processes to be performed in a non-audio section during conversation. It is possible to avoid unnecessary coding or transmission of a silent packet, saving the arithmetic processing and network bands.

*2 CNG reacts in conjunction with the VAD algorithm immediately when a silent period occurs, inserting an artificial noise in this period until the audio activity resumes.
Inserting an artificial noise gives the listeners the illusion that constant transmission stream is present. So, they do not notice that the line has been opened as the ambient sound continues to exist during conversations.

13.3.8. Phone - Video



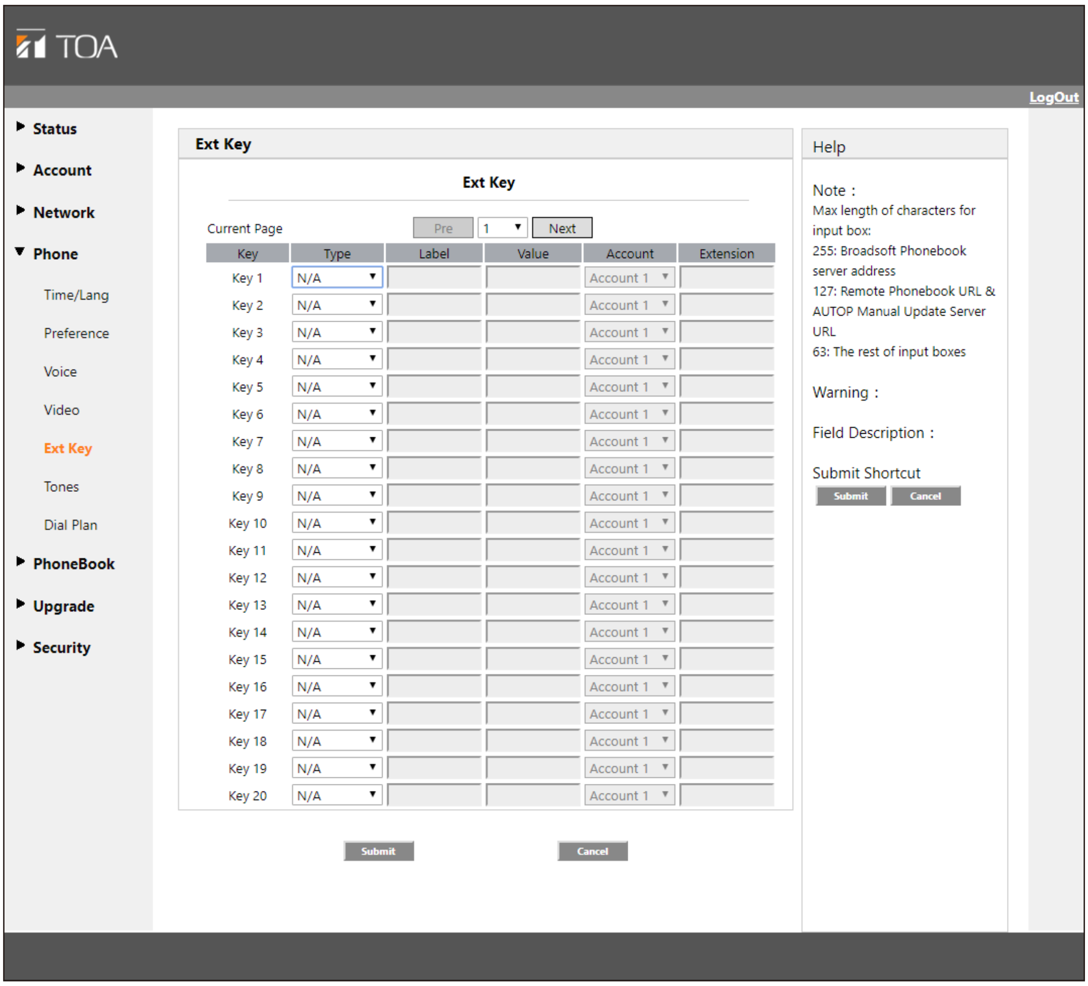
Item	Description
Media Feedback	NACK: "Enabled" is selected by default. Tmmbr: Sends the temporarily maximum media bit rate request. "Disabled" is selected by default.
H.264[*1] Settings	Sets the video parameters related to H.264. H264 Profile: 4 modes are available: Base Profile, Main Profile, High Profile, and Extend Profile modes. Each different profile makes up a different coding function and video quality. H264 Level: A different profile has the corresponding level value. IDR[*2] Interval: Used to control both coding and decoding processes. Rate Control: Select the H.264 video bit rate.
Others	Hardware Encode Acceleration: Enables the hardware encoder enhancement when needed. "Enabled" is selected by default. Hardware Decode Acceleration: "Enabled" is selected by default. Color Enhancement: Improves the multimedia station's display color. "Enabled" is selected by default. Image Quality: Select "High," "Middle," or "Low." "High" is selected by default. Camera Priority: "Internal" is selected by default.

[*1] A video stream compression standard. The video stream quality is nearly the same as that of H.263, but the bit rate of H.264 is half that of H.263. This type of compression is sometimes referred to as MPEG-4 part 10.

[*2] Stands for Instantaneous Decoding Refresh.

13.3.9. Phone - Ext Key

To use the function assigned to the EXT key, it is necessary to create the shortcut of the function-assigned EXT key you want to use on the multimedia station's main screen.
For the method to create the shortcut of the EXT key, see p. 46.

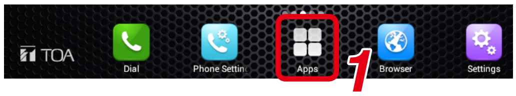


Item	Description
Current Page	4 pages of extension numbers are provided for the multimedia station. 20 extension number keys are provided on each page.
Key	A specific function can be assigned to each key. List of the functions assignable to each key is as follows; • Pickup • Group Pickup • Intercom • History • Redial • ACD • BLF • BLF List • Call Return • Hot desking • Record • DTMF • Multicast Paging

[Creating the shortcut of the EXT key]

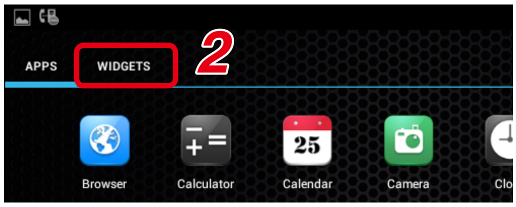
Step 1. Touch  in the lower part of the main screen.
An application list screen appears.

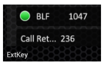
(Lower part of the main screen)



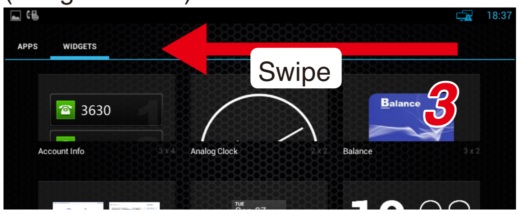
Step 2. Touch WIDGETS in the upper part of the application list screen.
A widget screen appears.

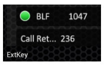
(Upper part of the application list screen)

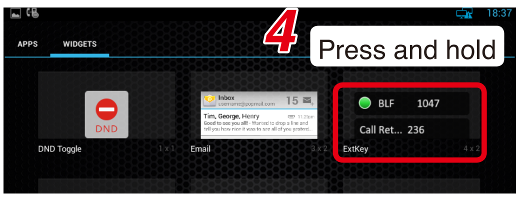


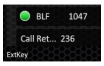
Step 3. Swipe the displayed widget screen left until  icon of the EXT key appears.

(Widget screen)



Step 4. If  appears, hold it down for 1 second or more.
The icon becomes slightly larger, and you can move it.

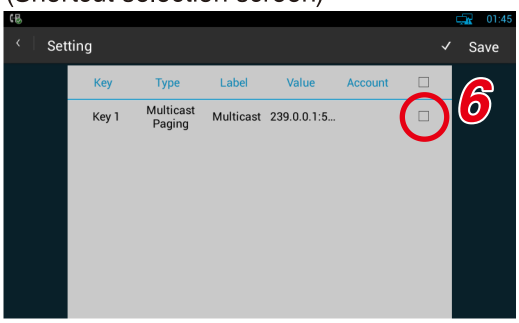


Step 5. Drag  to the shortcut area of the main screen, then drop it at the desired location.
A selection screen for the EXT key's shortcut you want to display appears.

(Main screen).

Step 6. Check the checkbox for the EXT key of which function you want to display on the main screen.
The shortcut icon is created and arranged on the main screen.

(Shortcut selection screen)



13.3.10. Phone - Dial Plan

Settings and edits of "Replace Rule" and "Dial Now" can be performed by switching the screen.

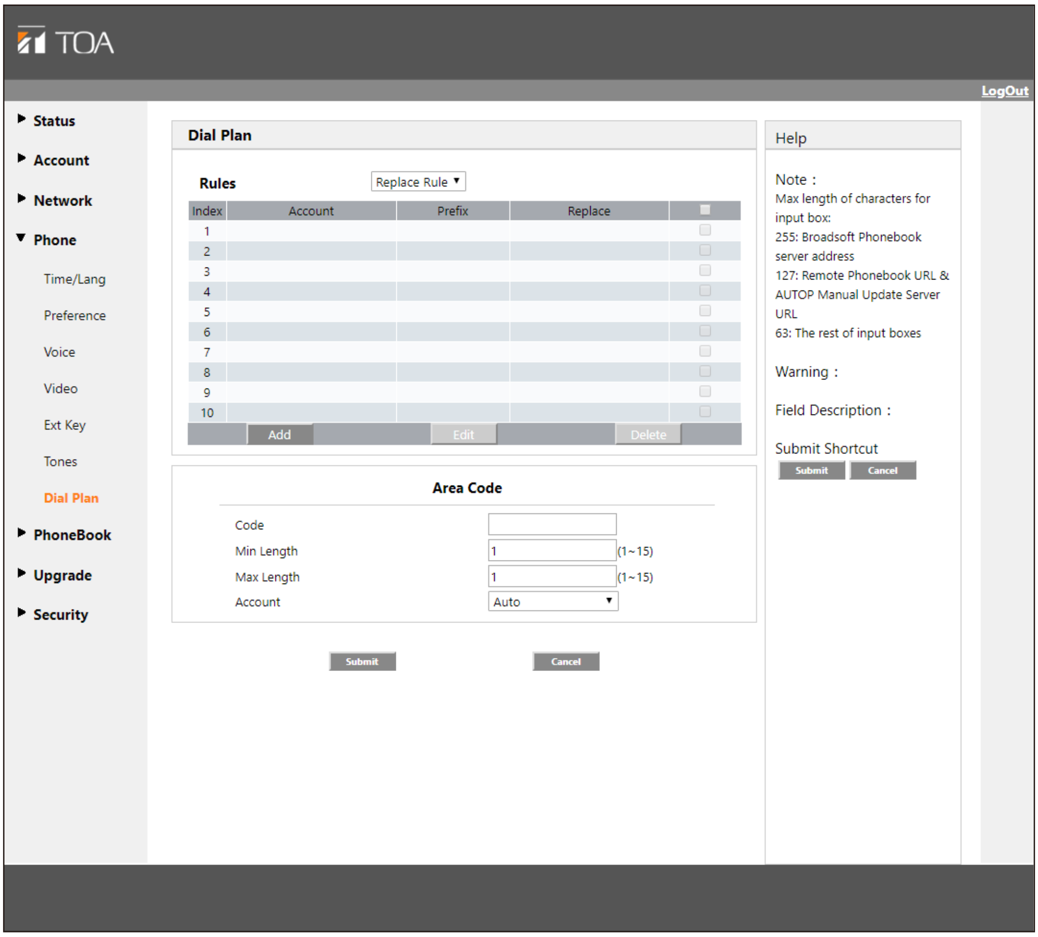
- Replace Rule: An abbreviated number can be set for the phone number or IP address. (Shown below.)
- Dial Now: Makes the settings of the phone number that can directly call the other party only by dialing without performing conversation start operation*1. (See. p. 48.)

*1 An operation to start conversations by pressing the station's Speaker button (p. 7) or touching 

or 

on the screen.

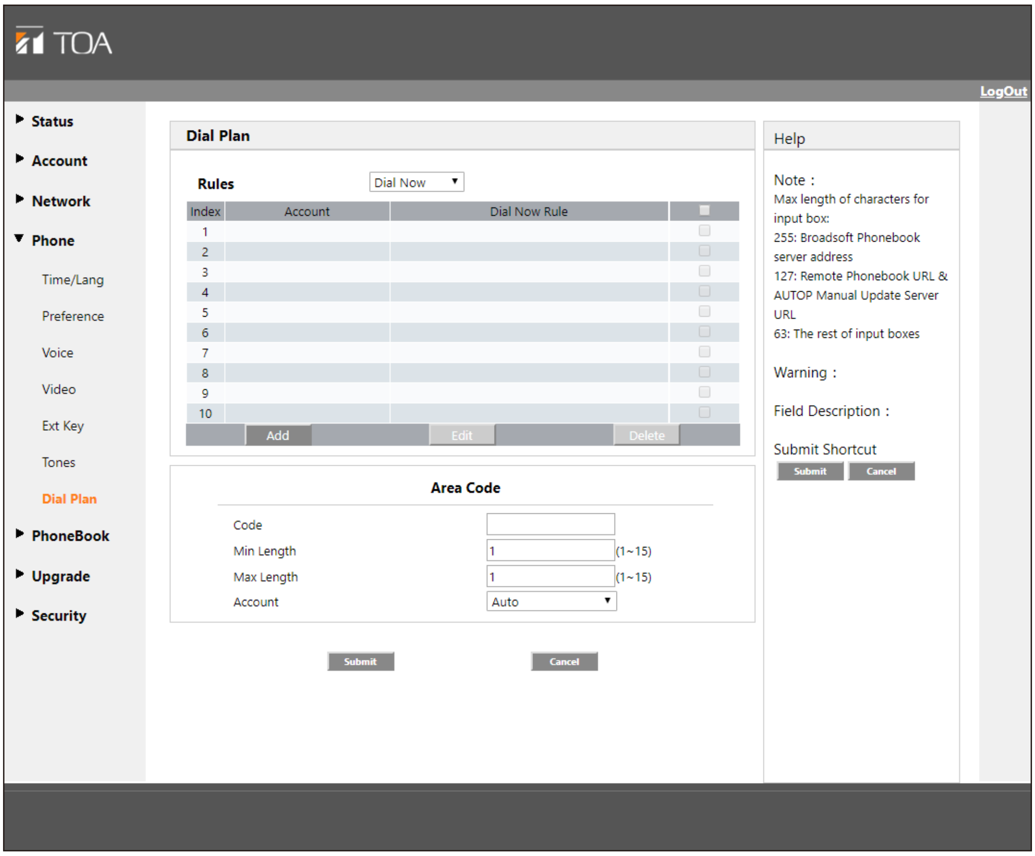
[Replace Rule]



Item	Description
Rule	Displays or edits "Replace Rule" or "Dial Now" by selecting either one. Click "Add" to register the rule or "Edit" to edit the registered rule. • Account: Sets the account to use. "Auto" is selected by default. • Prefix: Sets the abbreviated number to be assigned to the phone number or IP address. • Replace: Sets the phone number or IP address to which abbreviated number is assigned.
Area Code*2	Shows the geographical areas in the country. The multimedia station automatically gives the area code before the call number when the entered number matches the predefined area code rule. **Note** Only one area code is supported.

*2 NPAs (stands for Numbering Plan Areas).

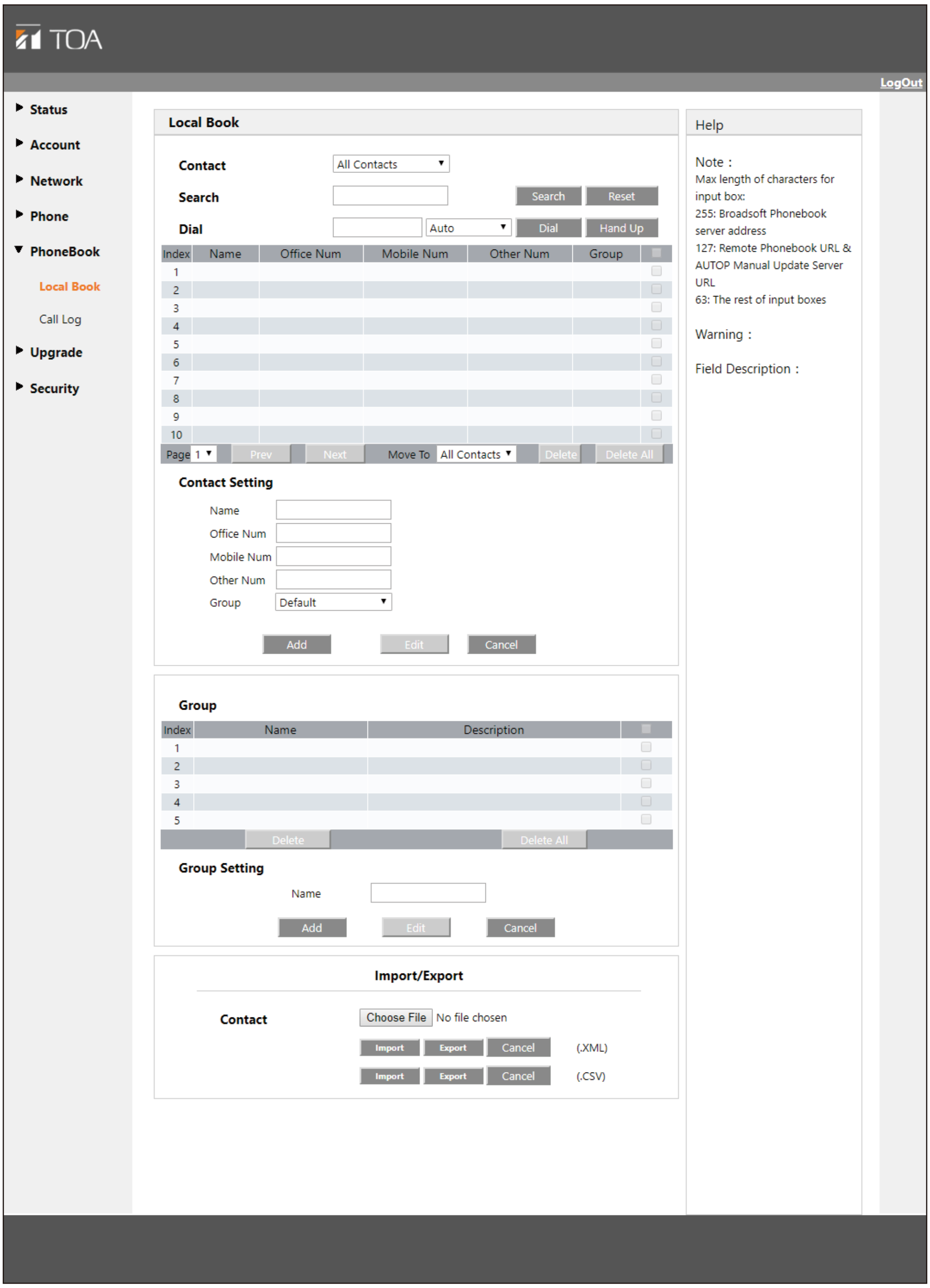
[Dial Now]



Item	Description
Rule	Displays or edits "Replace Rule" or "Dial Now" by selecting either one. Click "Add" to register the rule or "Edit" to edit the registered rule. • Account: Sets the account to use. "Auto" is selected by default. • Dial Now Rule: Sets the existing phone number that you can directly call.
Area Code*	Shows the geographical areas in the country. The multimedia station automatically gives the area code before the call number when the entered number matches the predefined area code rule. **Note** Only one area code is supported.

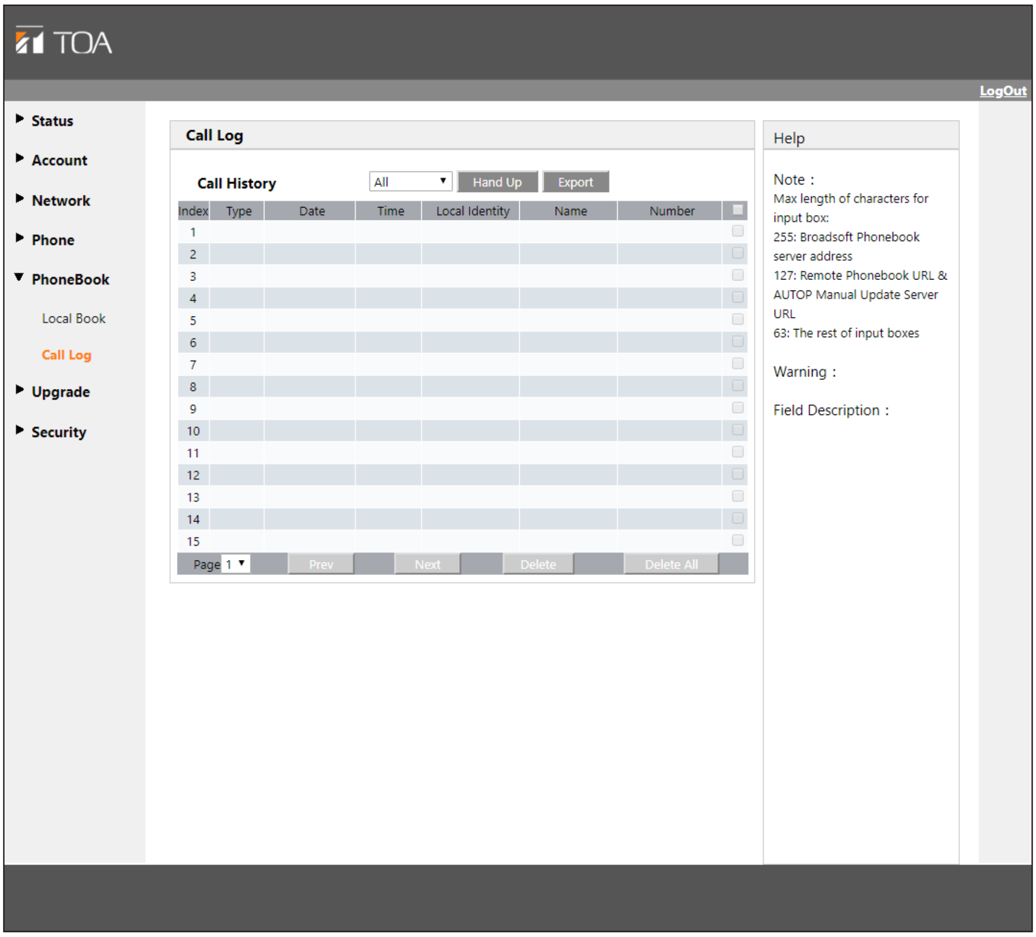
* NPAs (stands for Numbering Plan Areas).

13.3.11. Phone Book - Local Book



Item	Description
Contact	All Contacts: Displays or edits all local contacts. Favorites: Displays or edits the frequently used contact. Black List: Not used.
Search	Searches the designated contact from the local phone book.
Dial	Not used.
Contact Setting	Click "Add" when registering the contact setting such as Name, Office Num, Mobile Num, Other Num, and Group, and click "Edit" when editing the registered contact setting.
Group	Displays or deletes the group.
Group Setting	Displays or registers the group name and description. Click "Add" when registering the group and click "Edit" when editing the registered rule.
Import/Export	Imports or exports the Contact ("All Contacts" and "Favorites") in the form of an XML file or a CSV file.

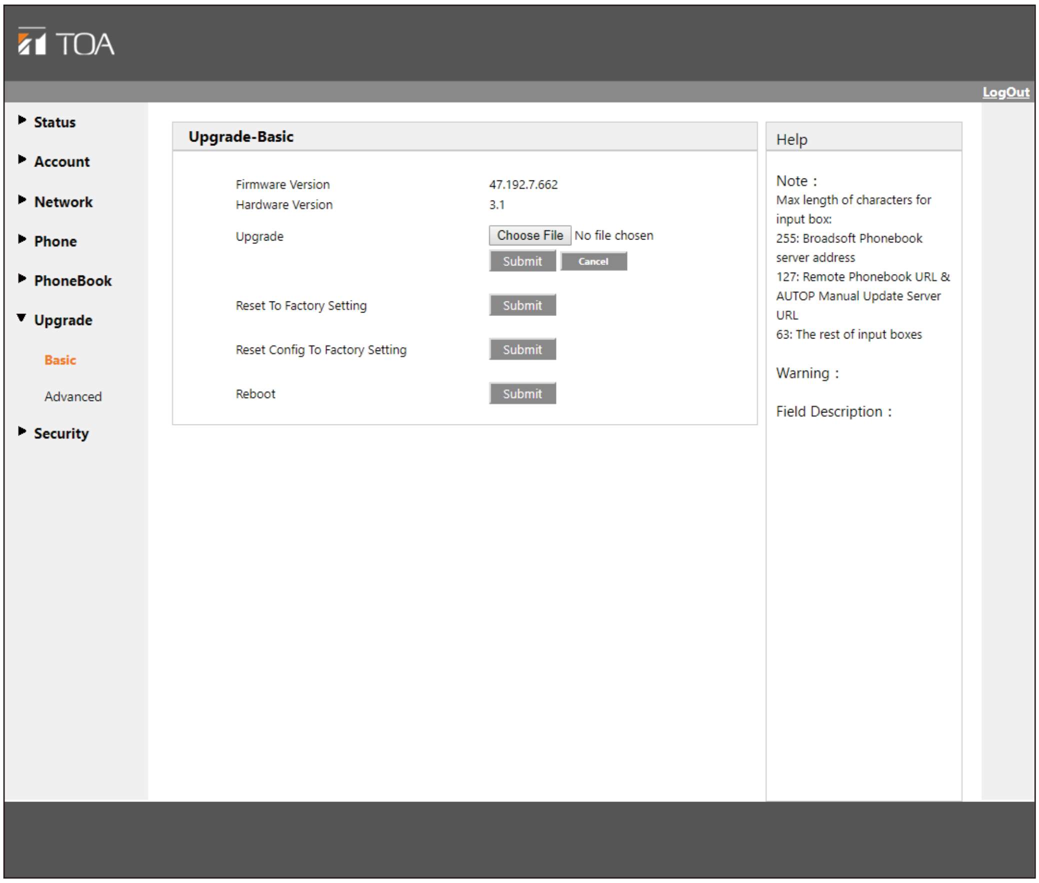
13.3.12. PhoneBook - Call Log



Item	Description
Call History	Displays the call history. Type of display can be selected from those listed below. • All* • Dialed • Received • Missed • Forwarded

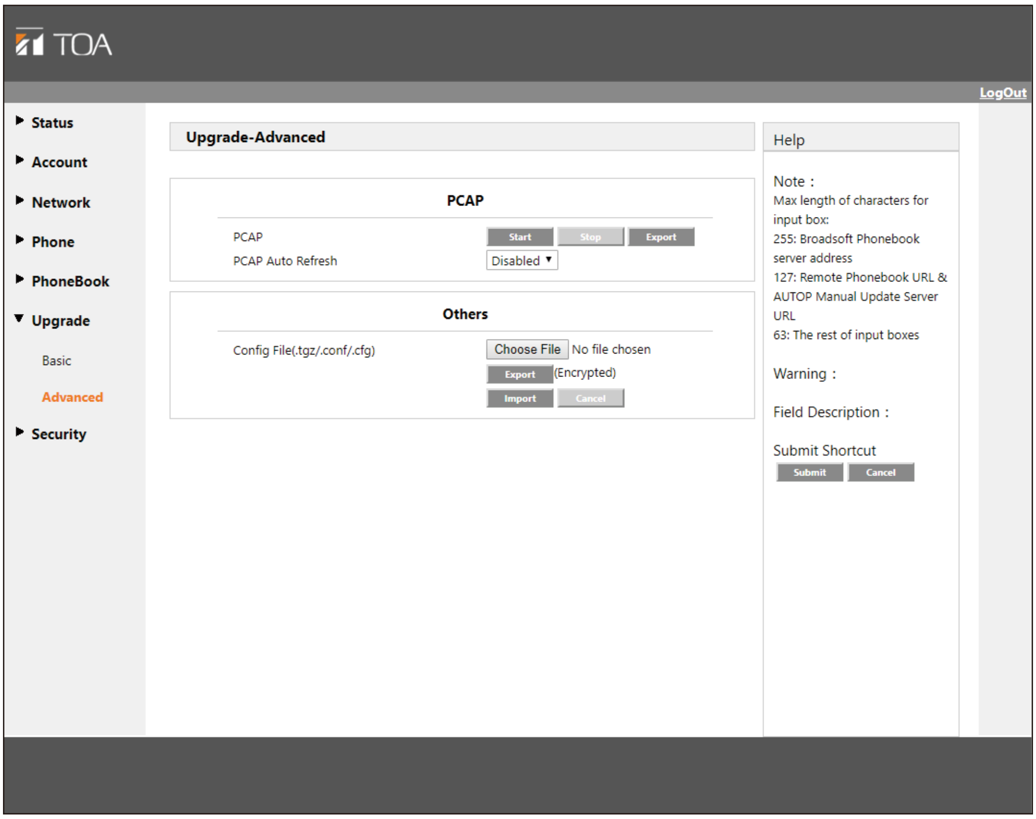
* All outgoing and incoming calls

13.3.13. Upgrade - Basic



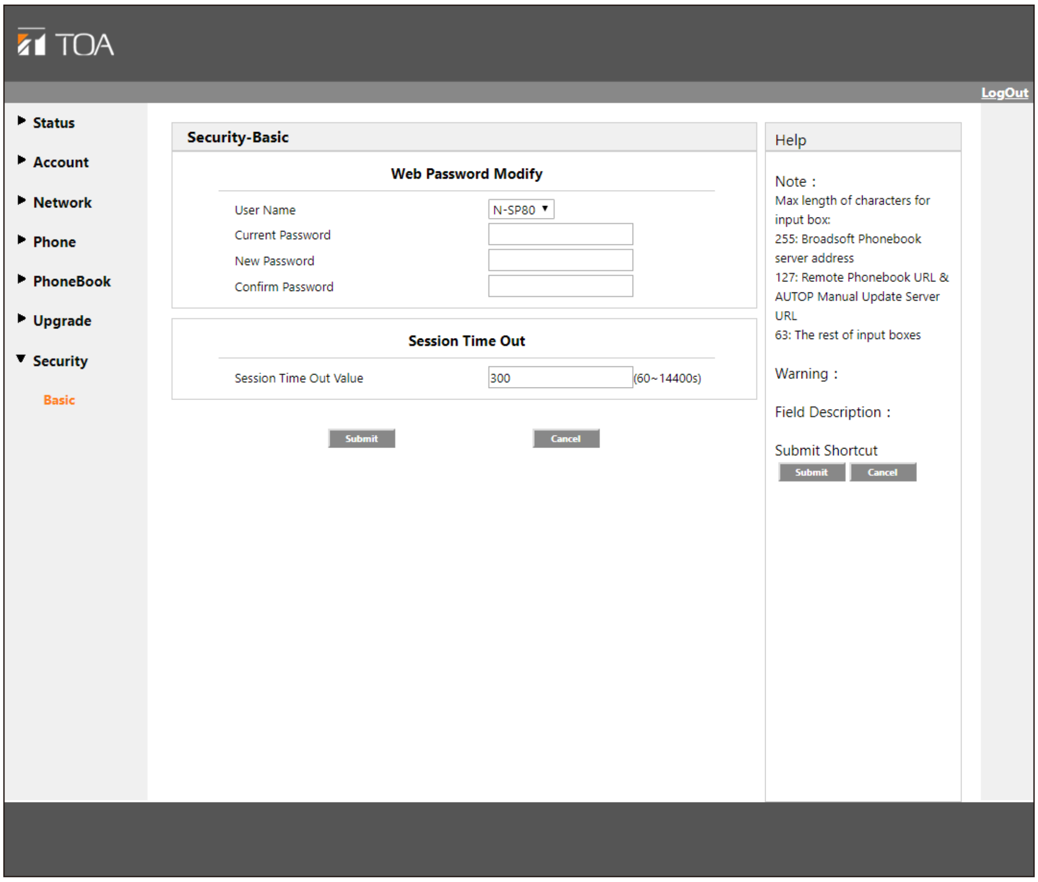
Item	Description
Firmware Version	Displays the firmware version.
Hardware Version	Displays the hardware version.
Upgrade	Select the file to be automatically upgraded from the local or remote server. **Note** Confirm that the file format is appropriate for that device. Example of a file: r47p-47.192.7.662-toa.zip
Reset To Factory Setting	Returns the IP address and account information to the initial value. • IP address: 192.168.1.101 • Account information: Blank
Reset Config to Factory Setting	Resets the Config file (setting file) to factory default. The IP address and account information remain unchanged.
Reboot	Click "Submit" to reboot the device.

13.3.14. Upgrade - Advanced



Item	Description
PCAP	Starts or stops the packet capturing. Exports the captured packets file as well. Start: Starts capturing all the packet files received by or sent from the multimedia station. Stop: Stops the packet capturing. Export: Captures the saved packet file. PCAP Auto Refresh: When "Enabled" is selected for automatic update, the packet data is constantly overwritten. **Note** The multimedia station saves the captured packets files into the temporary file. The maximum size of this file is 1 MB. When the saved data reaches this limit, the station stops capturing data.
Others	Exports or imports the setting file of the multimedia station. Setting files in .tgz/.conf/.cfg format can be handled. Export: Encrypts and downloads the setting file. This file is downloaded in .tgz format. Import: Uploads the file set in the Config file to the device, updating the device settings.

13.3.15. Security - Basic



Item	Description
Web Password Modify	Changes the user password. User Name: N-SP80 The user name is fixed. It cannot be changed. Current Password: guest (default setting) The password can be changed. New Password: Enter it with up to 63 characters. **Note** • Unusable characters : &, %, ' , = • Password is case-sensitive. Confirm Password: Enter the new password again. **Note** Set the security through the Web only.
Session Time Out	Session Time Out Value: Limits the log-in time. When the log-in time exceeds the set time limit, you need to re-login to the Web site. "300" is selected by default.

13.4. N-SP80VS1 and N-SP80AS1 Settings

13.4.1. Logging in

Connect to the unit's Web server by using the IP address.
When the IP address is 192.168.1.102, enter "http://192.168.1.102" to make connection.
For the method to confirm the IP address, see p. 35 "Confirming the IP address of Each Device."

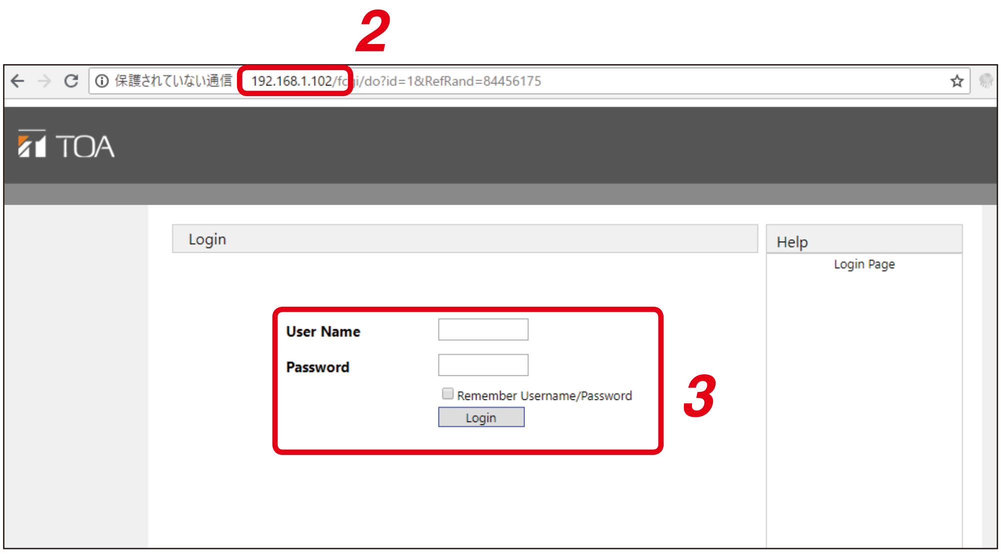
User name and password settings are as follows.

User name: N-SP80
The user name is fixed. It cannot be changed.

Password: guest (default setting)
The password can be changed.
Enter it with up to 63 characters.

Notes

• Unusable characters : &, %, ' , =
• Password is case-sensitive.



Step 1. Start the PC's browser.

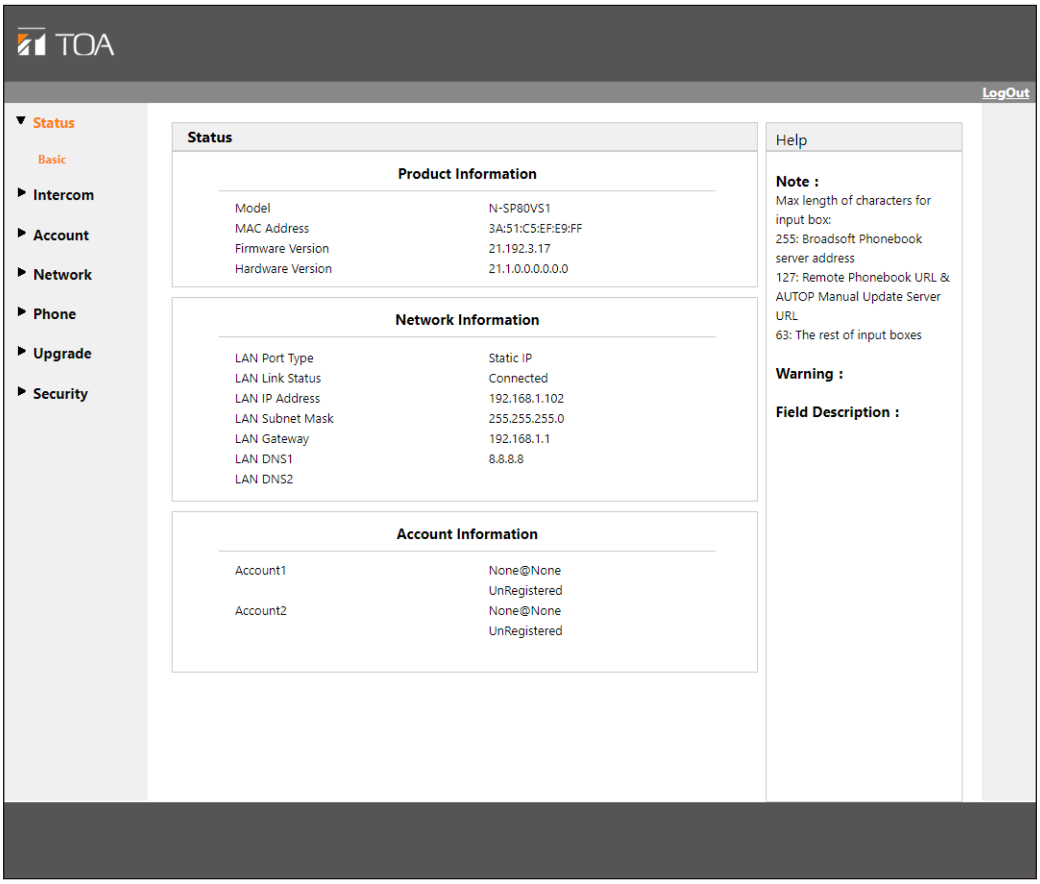
Step 2. Enter the IP address in the address bar.

Tip
The default IP address is 192.168.1.102. (Subnet mask is 255.255.255.0.)

A login screen appears.

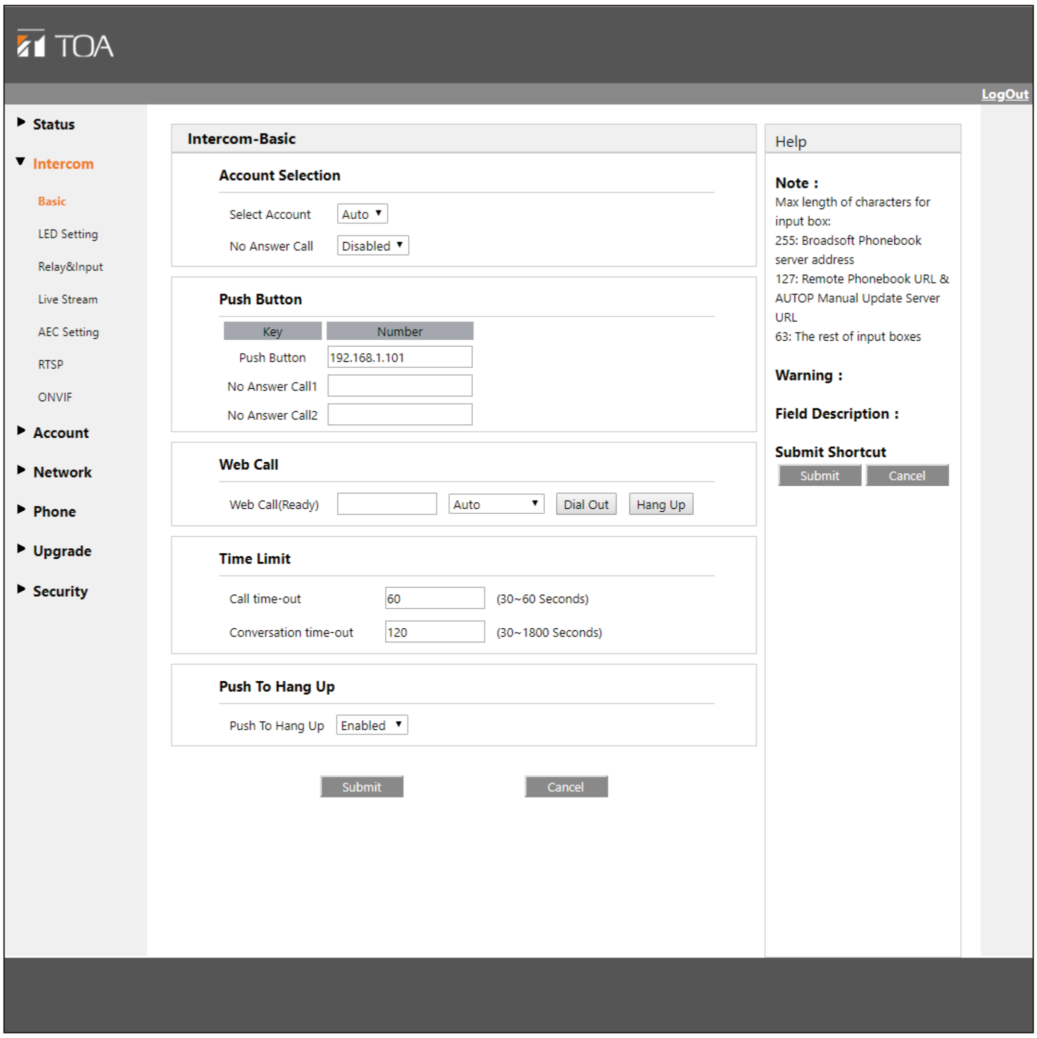
Step 3. Enter the user name and password, then click Login.

13.4.2. Status - Basic



Item	Description
Product Information	Displays the following product information: • Model • MAC address (Physical address of the IP device) • Firmware version and Hardware version.
Network Information	Displays the following network status (LAN port) information of the device: • LAN port type • LAN link status • LAN IP address • LAN Subnet Mask • LAN Gateway • LAN DNS1 • LAN DNS2
Account Information	Displays the account information and registration status (account user name, registered server address, and registration result) of the device.

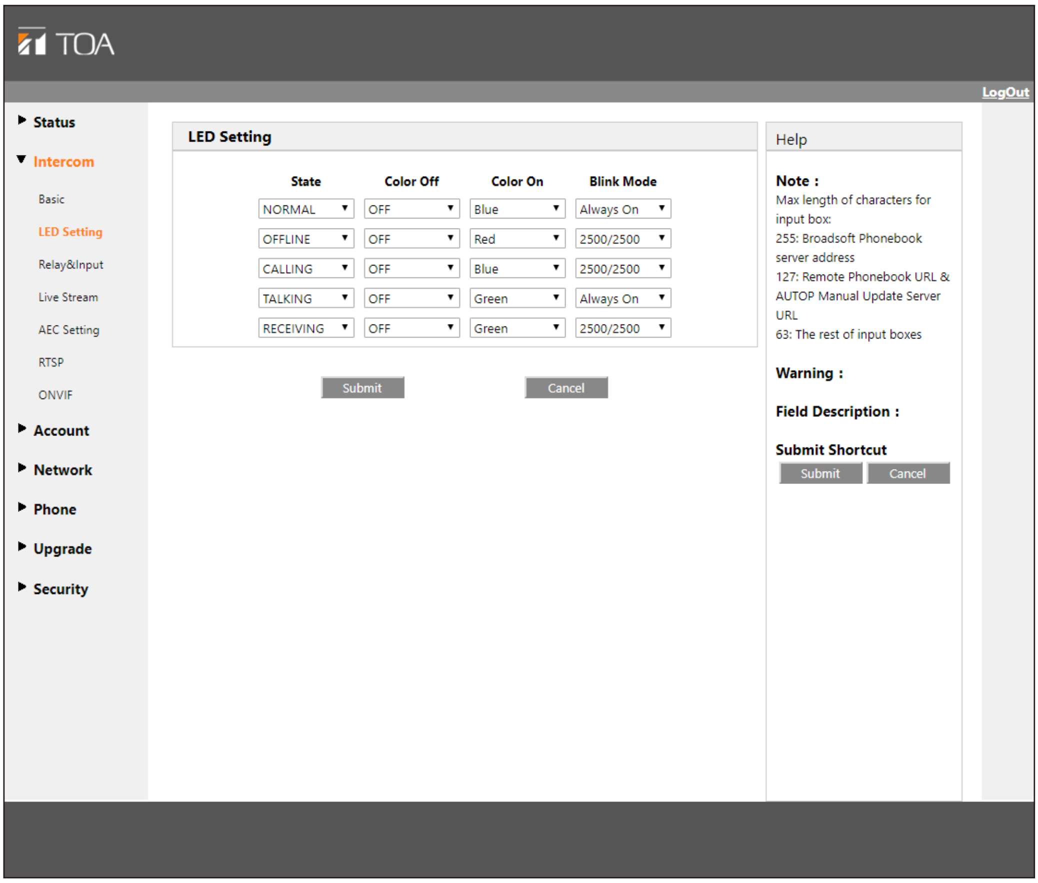
13.4.3. Intercom - Basic



Item	Description
Account Selection	Select Account: The door station supports 2 accounts. You can select a single account or Automatic mode with the Intercom basic settings below. "Auto" is selected by default. No Answer Call: Transfers the call to another station when the called party does not answer. "Disabled" is selected by default.
Push Button	Push Button: Up to 3 call destinations can be set. If the main call destination station does not answer for the given period, the call to that station stops and moves to the next station assigned to No answer call destination 1. Up to 2 No answer destinations can be set. "Disabled" is selected by default.
Web Call	Remotely calls the station number entered in this field from the multimedia station connected to this screen.
Time Limit	Sets the maximum call time-out and conversation time-out durations. Call time-out: "60" seconds is set by default. Conversation time-out: "120" seconds is set by default.
Push To Hang Up	Sets "Push to Hang Up" function* to "Enabled" or "Disabled." "Enabled" is selected by default.

* A function that can "Hang up" the call by pressing the Call button again while the door station is making a call or engaged in conversation.

13.4.4. Intercom - LED Setting



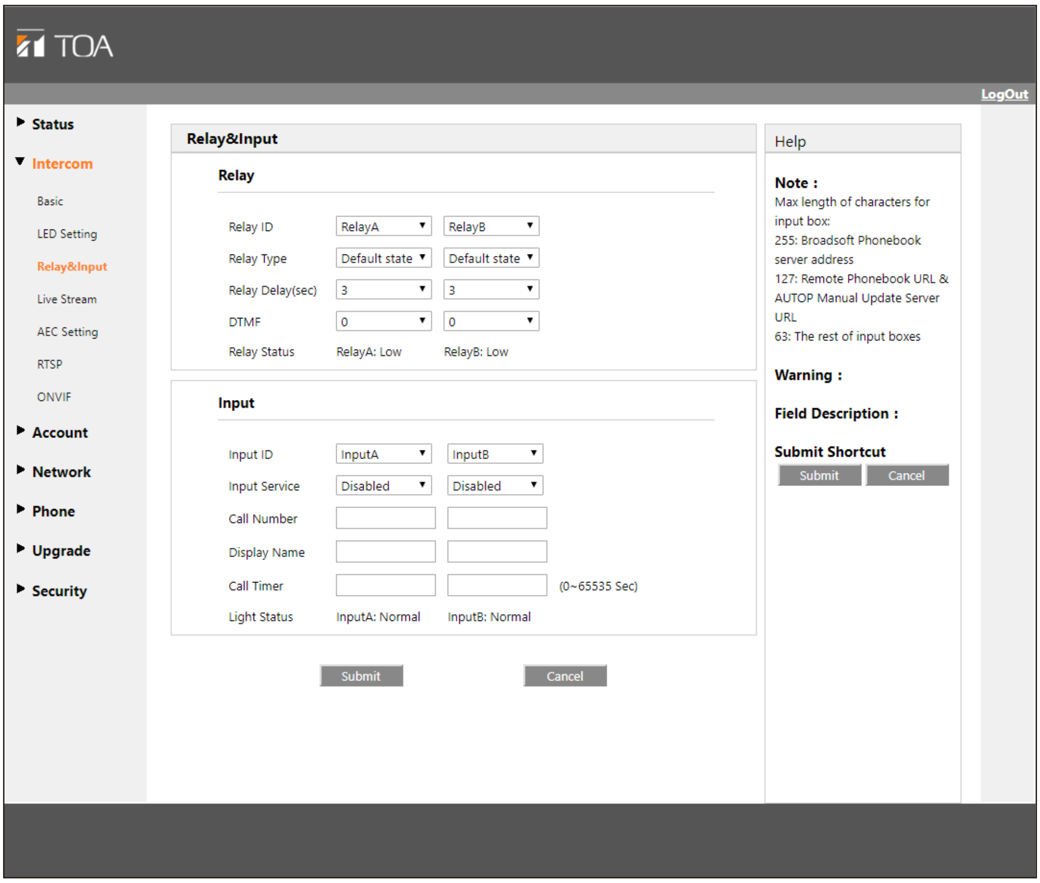
<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>LED Setting</td><td>State: Designates the door station's state to decide the LED’s lighting pattern.
<table>
<tr><th>State</th><th>Description of state</th></tr>
<tr><td>NORMAL</td><td>Standby</td></tr>
<tr><td>OFFLINE</td><td>Connection to Network and SIP not established</td></tr>
<tr><td>CALLING</td><td>Calling in progress</td></tr>
<tr><td>TALKING</td><td>Conversation in progress</td></tr>
<tr><td>RECEIVING</td><td>Receiving an incoming call</td></tr>
</table>
Color Off: (Not used)
Color On: Sets color while flashing or lit.
Select red, blue, green, or Off.
Blink Mode: Sets Lit, Unlit, and Flashing patterns
(msec) Lit: "Always On"
Unlit: "Always Off"
Flashing* : "500/500", "1000/1000", "2500/2500", "3000/3000"</td></tr>
</table>

* Each setting value shows the flashing interval as Lit duration/Unlit duration (msec).

The table below shows the LED’s initial settings.

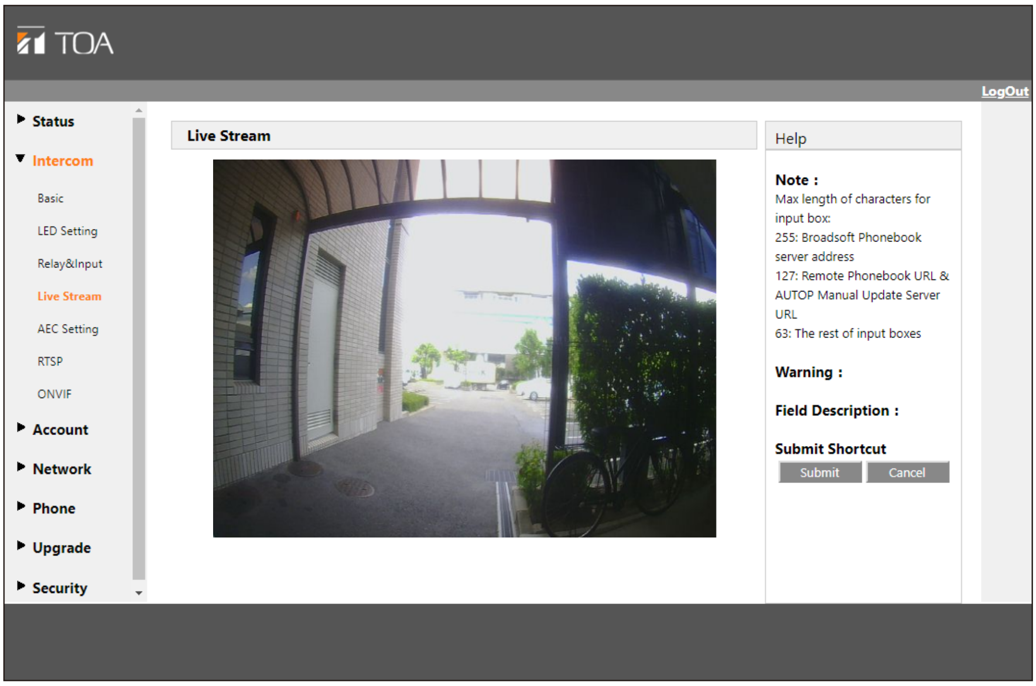
State	Color Off	Color On	Blink Mode
NORMAL	OFF	Blue	Always On
OFFLINE	OFF	Red	2500/2500
CALLING	OFF	Blue	2500/2500
TALKING	OFF	Green	Always On
RECEIVING	OFF	Green	2500/2500

13.4.5. Intercom - Relay&Input



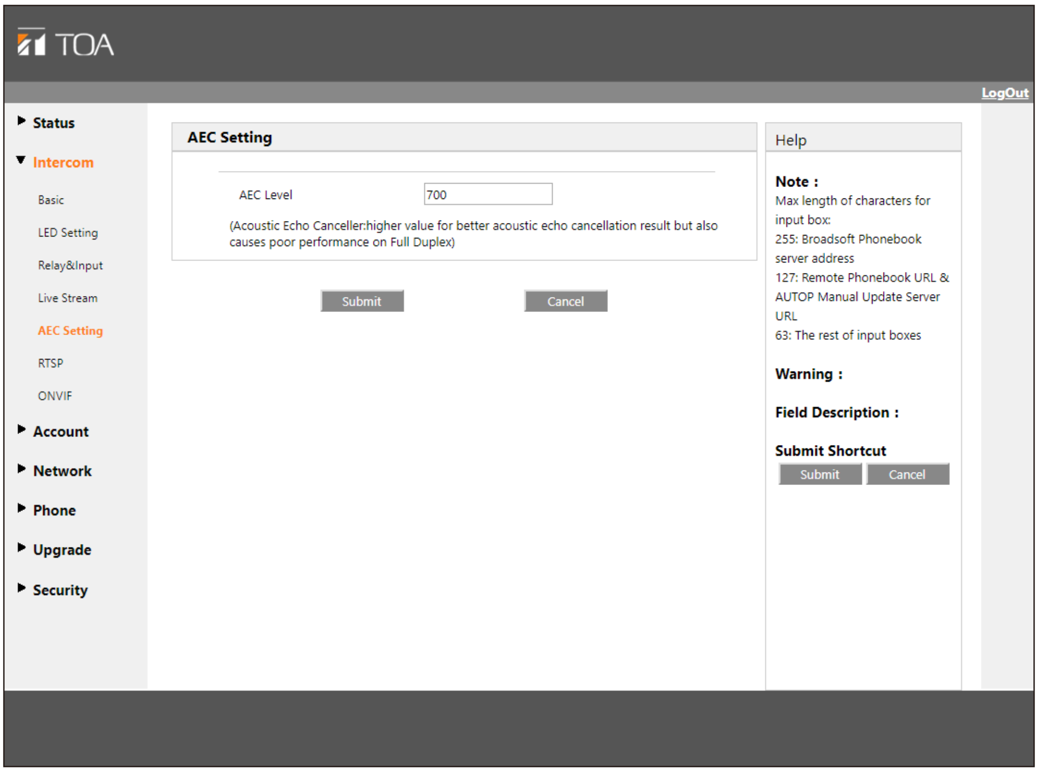
Item	Description
Relay	Performs the settings related to the door lock release when the door remote control function is used. Relay ID: The door station has 2 relay control outputs. Relay Type: Each lock is controlled using a different relay type. Relay Delay(sec): Leaves the door open for the given period of time. The time range is 1 to 5 seconds. DTMF: Sets the DTMF code with which the lock release is remotely controlled. Relay Status: Displays each relay type status.
Input	Connects to the external control switch. Example of use: Install a sensor switch functioned as a vandal prevention device between the door station and the back box. Connect the sensor's output to the door station's input. If the door station is destroyed violently, the sensor is activated and outputs an alarm signal. **Note** The door station has no sensor function. Input ID: The door station has 2 built-in photo couplers on the inputs. If the photo coupler operates, an alarm signal is output when this function is set to "Enabled." Input Service: "Disabled" is selected by default. Call Number: Sets the call number of the alarm control center. Display Name: The calling party's name is sent to and displayed on the called station. Call Timer: Continues a call for the set time while the input is being activated. Light Status: Displays the input status.

13.4.6. Intercom - Live Stream



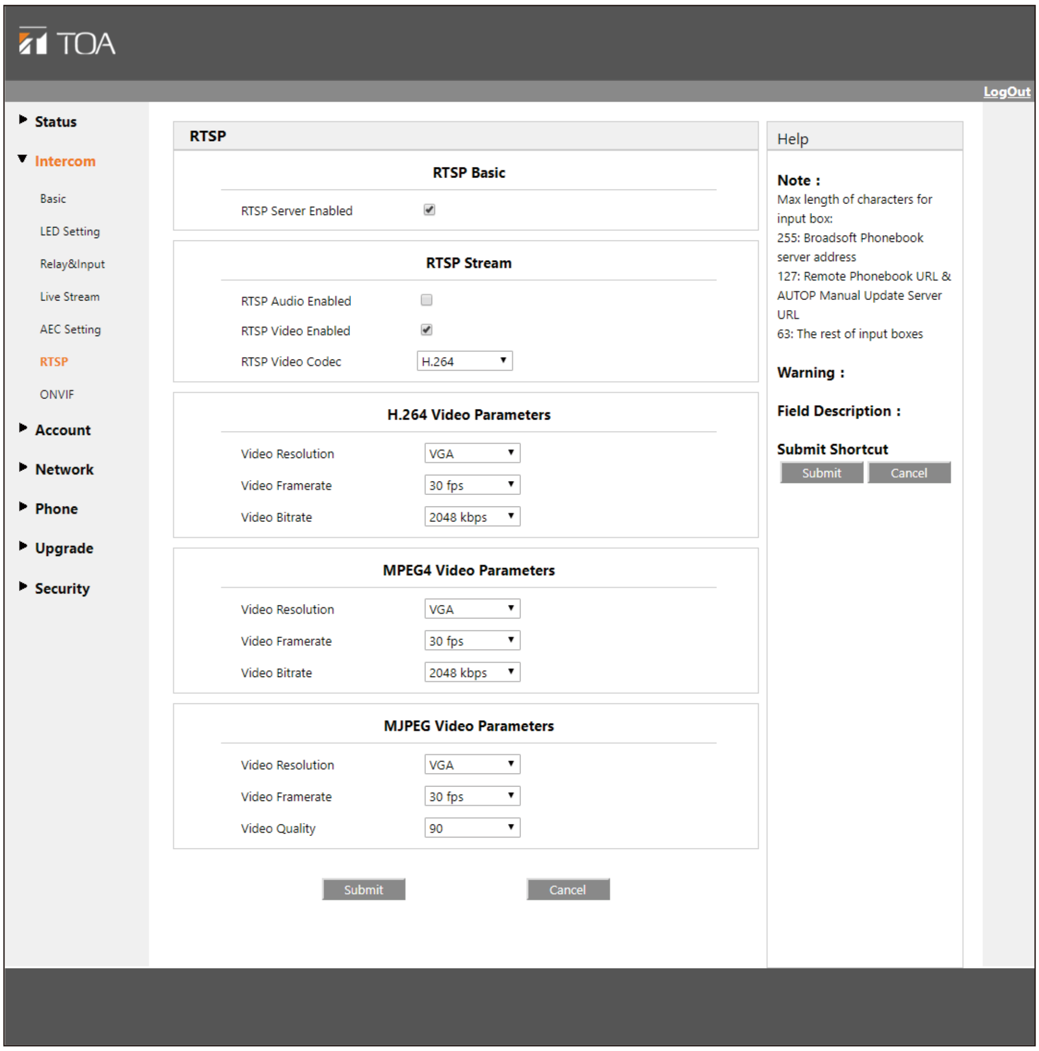
Item	Description
Live Stream	Allows to monitor the real-time images from the N-SP80VS1.

13.4.7. Intercom - AEC Setting



Item	Description
AEC Setting	AEC (Acoustic Echo Canceller) is used to adjust an echo effect during conversation. "700" is set by default. The higher the level is, the more the echo is suppressed. (The station is placed in half-duplex conversation state.)

13.4.8. Intercom - RTSP



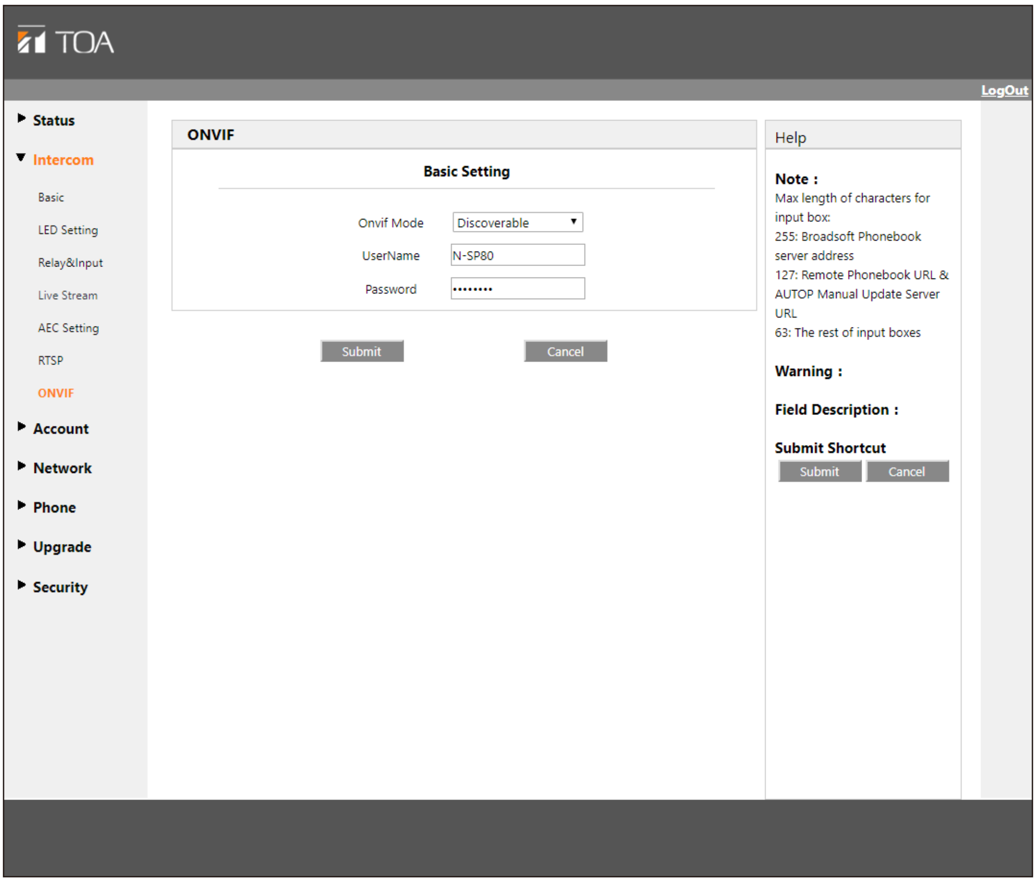
Item	Description
RTSP Basic	Activating the RTSP function allows the N-SP80VS1 to monitor.
RTSP Stream	To enable the RTSP video, select the video codec. The N-SP80VS1 supports H.264 and H.263 video codec. "H.264" is selected by default.
H.264*1 Video Parameters	Changes the resolution, frame rate, and bit rate of H.264 parameter.
MPEG4*2 Video Parameters	Changes the resolution, frame rate, and bit rate of MPEG4.
MJPEG*3 Video Parameters	Changes the resolution, frame rate, and bit rate of MJPEG.

*1 A video stream compression standard. The video stream quality is nearly the same as that of H.263, but the bit rate of H.264 is half that of H.263. This type of compression is sometimes referred to as MPEG-4 part 10.

*2 One of the network video image compression standards

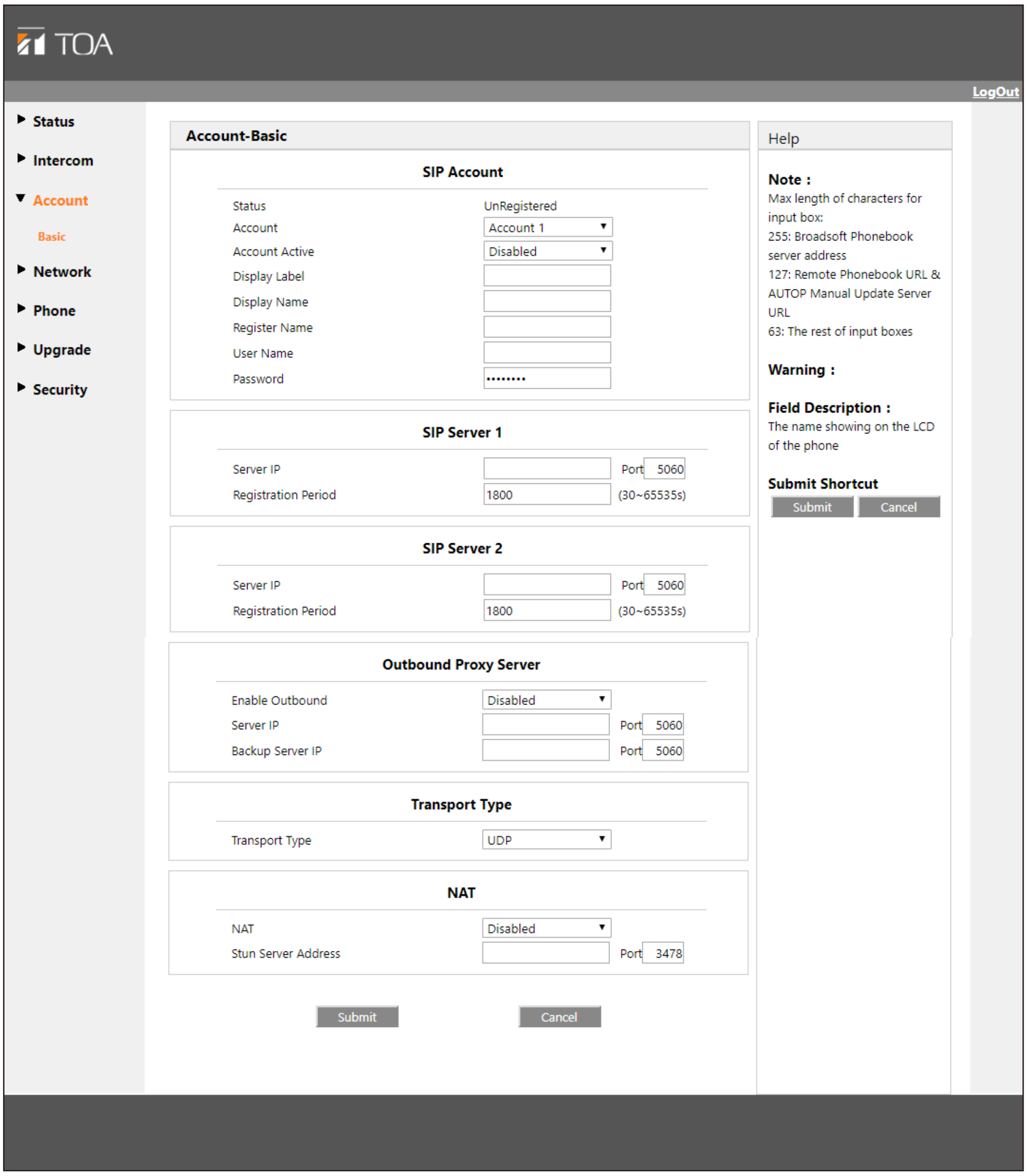
*3 MJPEG stands for Motion Joint Photographic Experts Group. A video encoding format where each frame is individually compressed by JPEG.
High quality video images are generated with the MJPEG compression, enabling the video resolution and compression frame to be set flexibly.

13.4.9. Intercom - ONVIF



Item	Description
Basic Setting	Sets the ONVIF function parameters. Use this function to connect the station to the corresponding ONVIF tool. ONVIF Mode: Sets Discoverable or Non-discoverable function mode. "Discoverable" is selected by default. The N-SP80VS1 can be detected only in Discoverable mode using the ONVIF software. User Name: Change the user name as needed. "N-SP80" is set by default. Password: Change to the desired password. "guest" is set by default. Enter with up to 63 characters. **Note** Symbols below and 2-byte characters cannot be used. & % ' =

13.4.10. Account - Basic

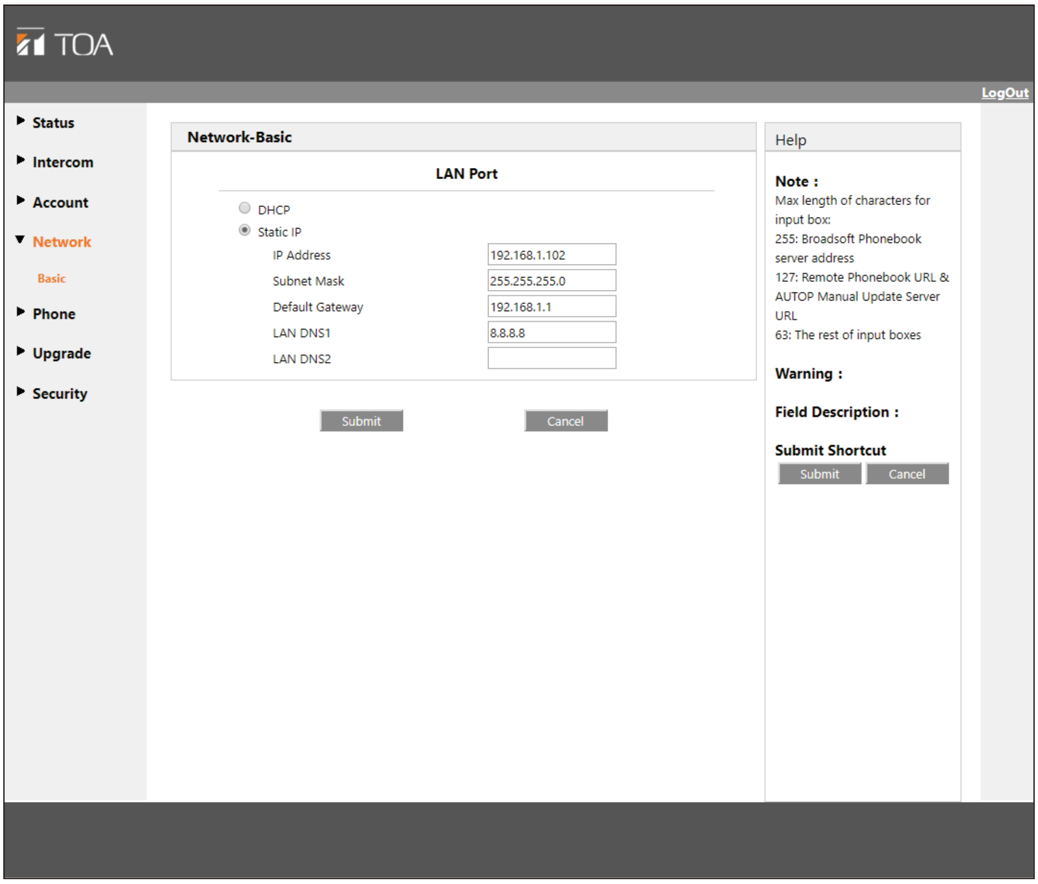


Item	Description
SIP Account	Displays or sets the specific account. Status: Displays the registration result. Account: Selects the account to set. (Account 1 or Account 2) Account Active: Validates the selected account. Display Label: A station name that is recorded in log. Display Name: Sent to the called party and displayed. Register Name: Use the name to be registered (set) in the SIP server for authentication. User Name: Use the name to be registered (set) in the SIP server for authentication. Password: Use the password to be registered (set) in the SIP server.
SIP Server 1	Displays or sets the Primary SIP server settings. Server IP: SIP server address that is URL or IP address Registration Period: An interval to periodically send the registration (REGISTER) to the SIP server. Continues to retain the registration in the SIP server when the station sends registration (REGISTER) again within the registration (REGISTER) maintaining period on the SIP server.

Item	Description
SIP Server 2	Displays or sets the Secondary SIP server setting. This is a backup server so that the IP station can be registered at the Secondary SIP server even if the registration to the Primary SIP server fails. **Note** The Secondary SIP server is used as a backup server. If there is no SIP server for backup in a user environment, these corresponding fields are left blank.
Outbound Proxy Server*[1]	Enable Outbound: Sets Enabled/Disabled for the connection to the outbound server. Server IP: Sets the IP address of the outbound server to connect. Backup Server IP: Sets the backup server's IP address if there is a backup server for the outbound server.
Transport Type	Selects the SIP message's transfer type from a pull-down menu. UDP: An unreliable but highly effective transfer layer protocol. TCP: A reliable but less effective transfer layer protocol. TLS: A safe and reliable transfer layer protocol. DNS-SRV: DNS RR to designate the server location.
NAT	Displays or sets the NAT (Network Address Translator) settings. Stun Server Address*[2]: One of the solutions to solve NAT problem. **Note** "Disabled" is selected for NAT by default.

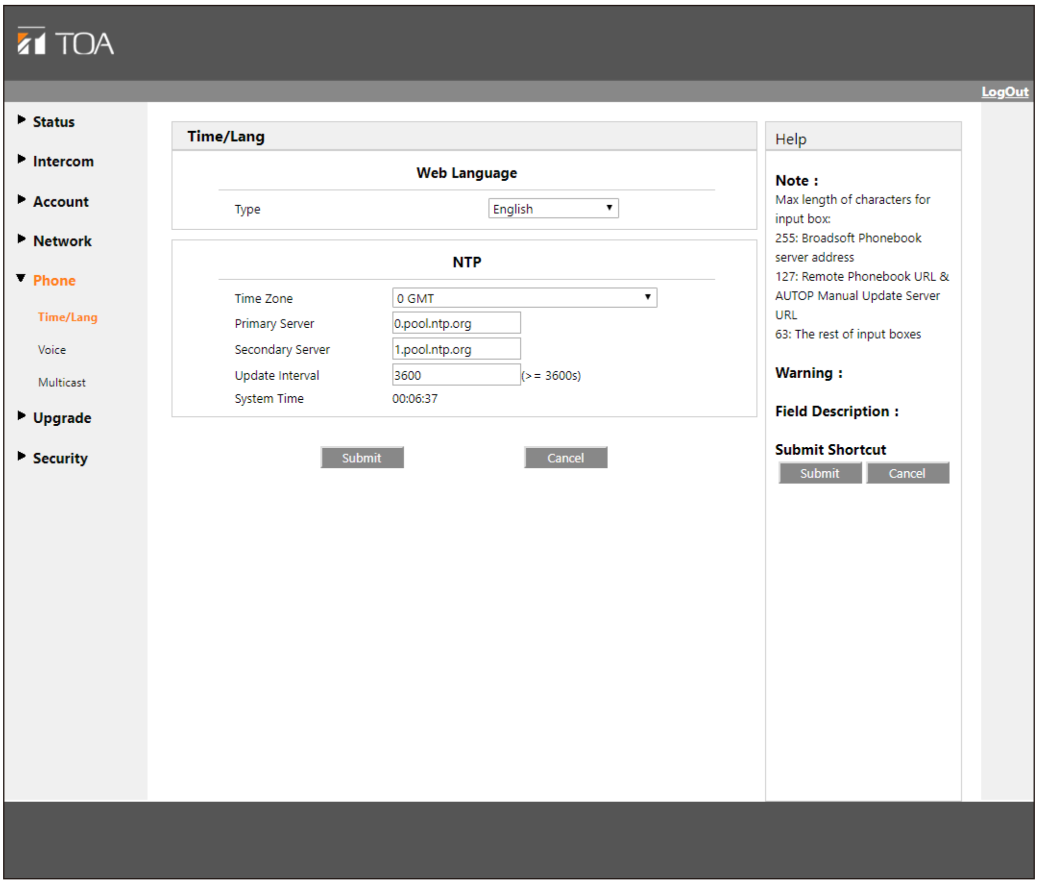
*[1] Used to receive all Start request messages and transfer them to the designated SIP server.
*[2] Stun stands for Simple Traversal of UDP over NAT.

13.4.11. Network - Basic



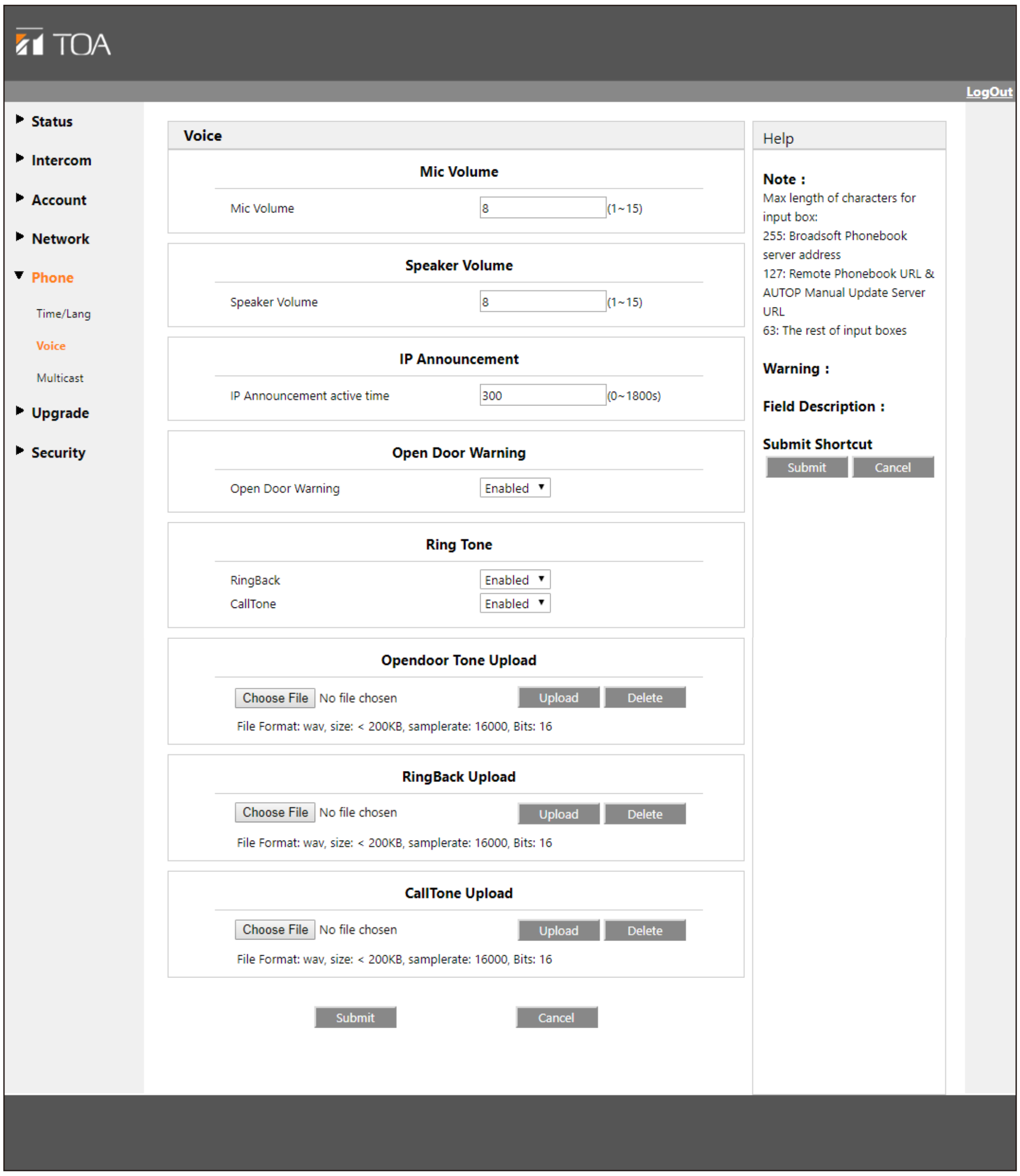
Item	Description
LAN Port	Displays or sets LAN port setting. DHCP: Automatically acquires the IP address, Subnet Mask, Default gateway, and DNS server address from the DHCP server. Static IP: It is necessary to manually set the IP address, Subnet Mask, Default gateway, and DNS server address.

13.4.12. Phone - Time/Lang



Item	Description
Web Language	Selects language used on the Web setting screen.
NTP	Performs the NTP (Network Time Protocol) server related settings. Time Zone: Selects the local time zone for the NTP server. Primary Server: Sets the Primary NTP server address. Secondary Server: Sets the Secondary NTP server address. Becomes accessible when the Primary NTP server cannot be accessed. Update interval: Sets the interval between 2 consecutive NTP requests. System Time: Clock of the currently connected device. **Note** NTP is used to automatically synchronize the local time with the internet time. As the NTP server supports only the GMT time, it is necessary to specify the time zone in order to set station's local time.

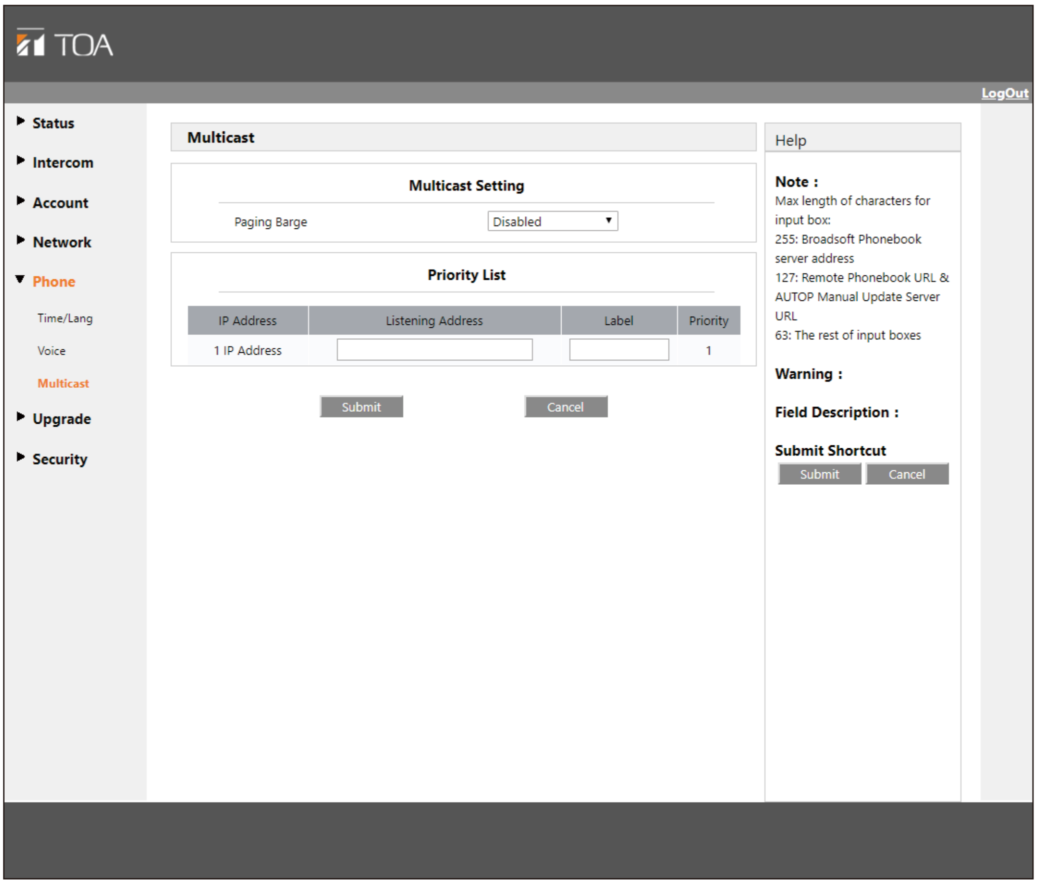
13.4.13. Phone - Voice



Item	Description
Mic Volume	Sets the microphone's volume level.
Speaker Volume	Sets the speaker's volume level.
IP Announcement	Performs the settings about the voice announcement* of IP address. Becomes active within the preset period (second) from power-on.
Open Door Warning	Sets the door remote's control sound. Select "Disabled" if you do not want to listen to the voice alarm message when the door lock is opened.
Ring Tone	RingBack: "Enabled" is selected by default. CallTone: "Enabled" is selected by default.
Opendoor Tone Upload	Uploads the sound when the door remote control function operates.
RingBack Upload	Uploads the ring tone that sounds when the door station's Call button is pressed.
CallTone Upload	Uploads the call tone that sounds when the door station receives an incoming call from the other device.

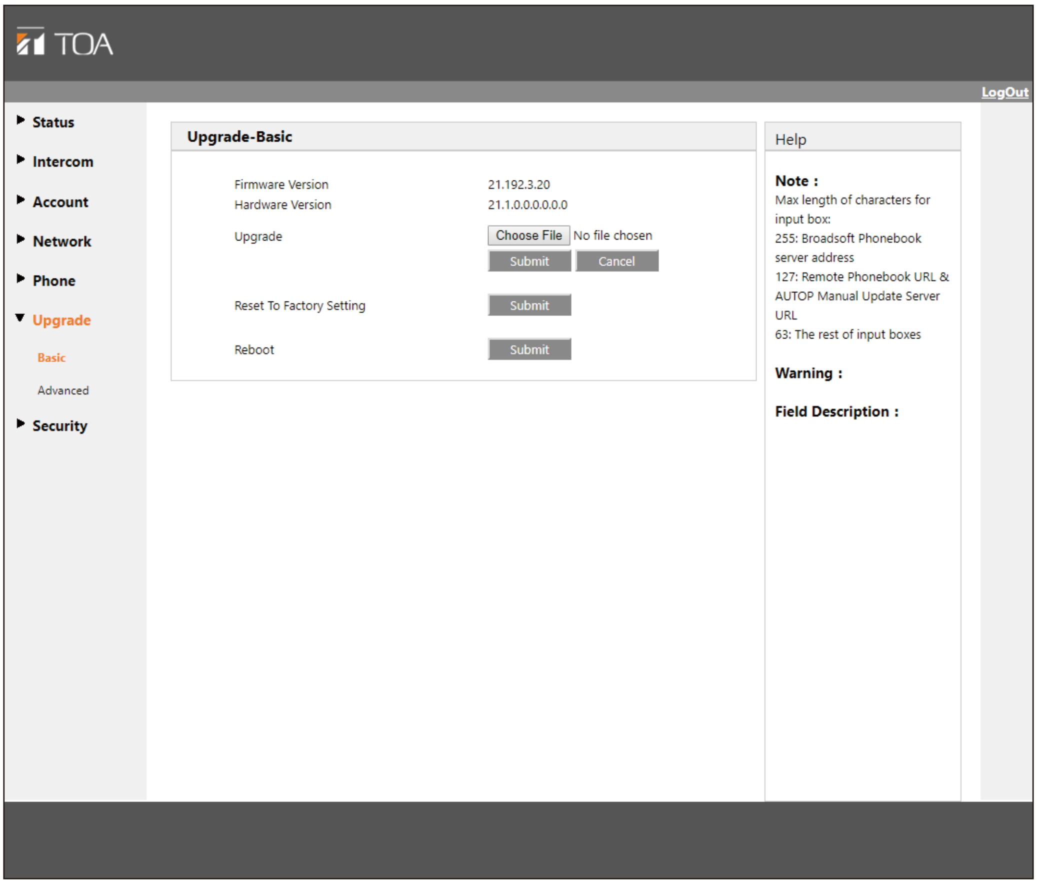
* The IP address will be announced from the station's speaker if you hold down the door station's Call button for 5 seconds or more.

13.4.14. Phone - Multicast



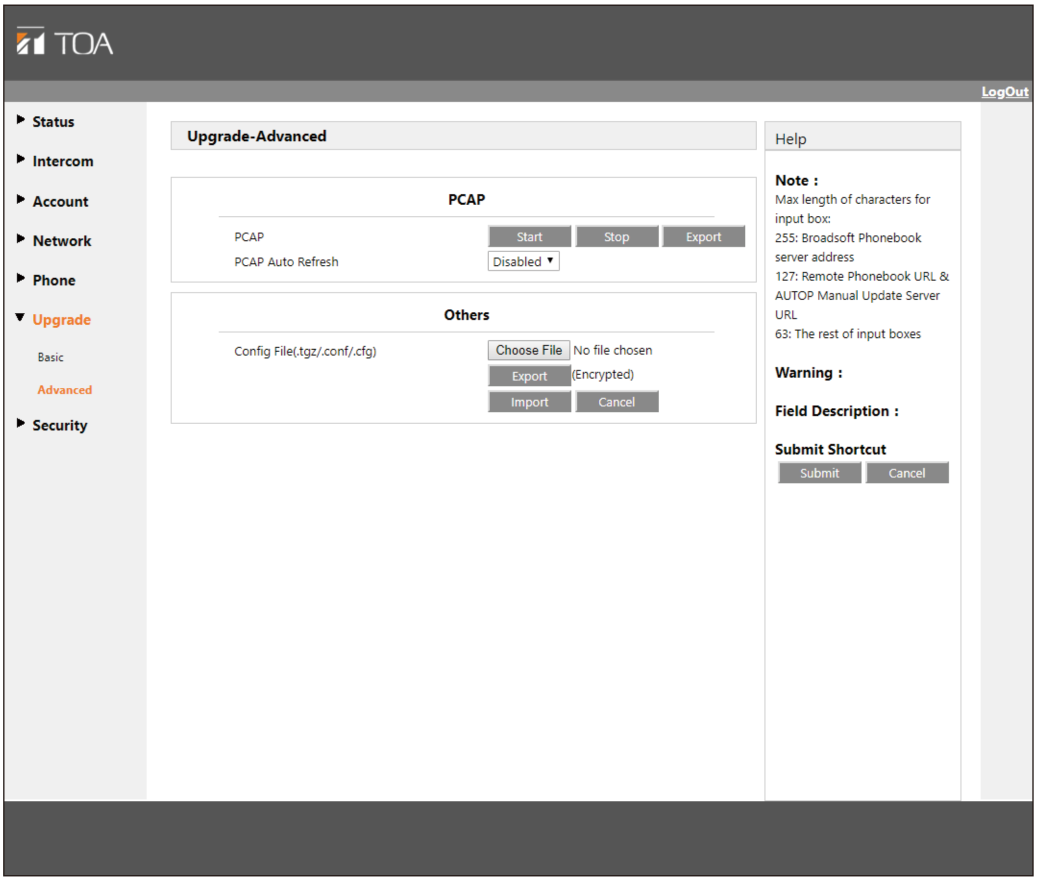
Item	Description
Multicast Setting	Displays or sets the Multicast setting. Paging Barge: Enables or disables the priority level between a call and a paging call. 1 (Enabled): The priority level of the paging call is higher than that of the call. Disabled: The priority level of the paging call is lower than that of the call.
Priority List	Sets the Multicast parameters. Listening Address: Enter IP address from which you want to listen to the paging. Label: Enter the label of each listening address.

13.4.15. Upgrade - Basic



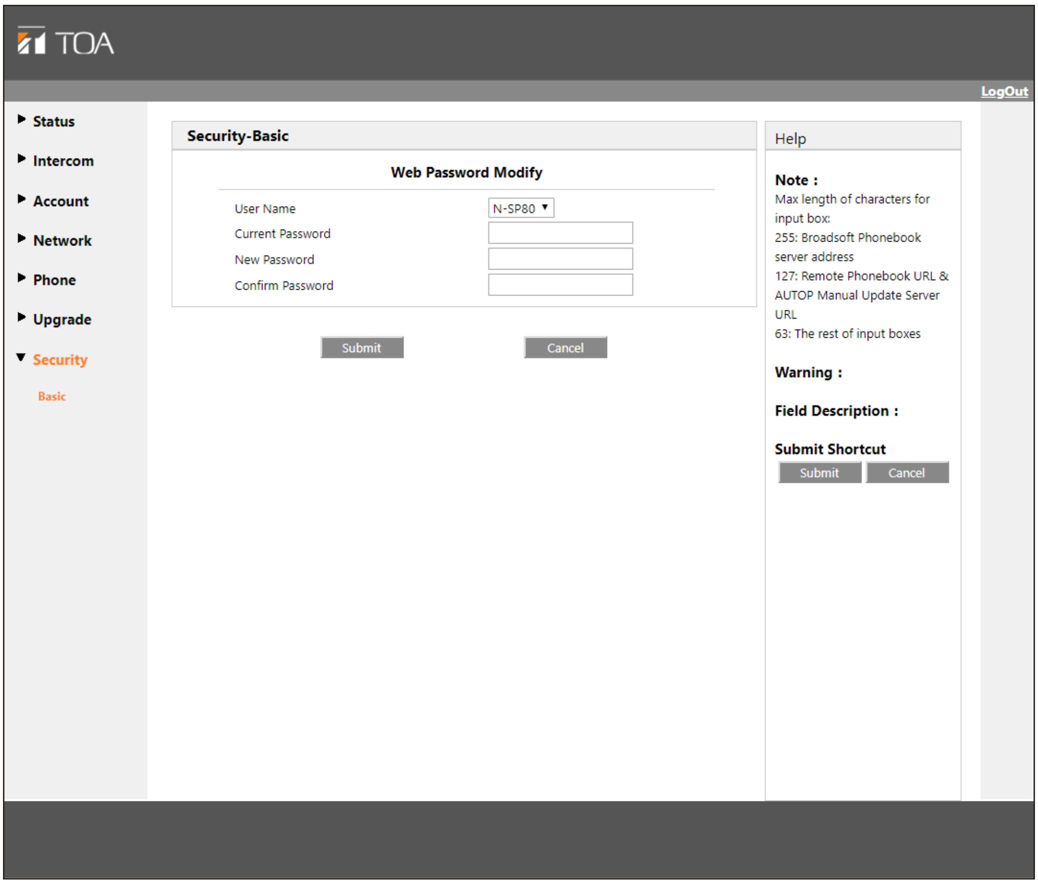
<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>Firmware Version</td><td>Displays the firmware version.</td></tr>
<tr><td>Hardware Version</td><td>Displays the hardware version.</td></tr>
<tr><td>Upgrade</td><td>Automatically selects the ROM file for upgrade from the local or remote server.
Note
Check if the file format is correct for that model.
Example of the file name: 21.192.2.33.rom</td></tr>
<tr><td>Reset to Factory Setting</td><td>Returns the IP address and account information to the initial value.
• IP address: 192.168.1.102
• Account information: Blank</td></tr>
<tr><td>Reboot</td><td>Remotely restarts the currently connected station.</td></tr>
</table>

13.4.15. Upgrade - Advanced



Item	Description
PCAP	Starts or stops packet capturing. Exports the captured packets file as well. Start: Starts capturing all the packet files sent to and received from the IP telephone. Stop: Stops the packet capturing. Export: Exports the capturing packets files. PCAP Auto Refresh: When "Enabled" is selected for automatic update, the packet data is constantly overwritten. "Disabled" is selected by default. **Note** The IP telephone saves the captured packets file into the temporary file. The maximum size of this file is 1 MB. When the saved data reaches this limit, the station stops capturing packets file.
Others	Config file: Exports or imports the IP telephone's setting files.

13.4.16. Security - Basic



Item	Description
Web Password Modify	Changes the user password. Current password: Enter the currently used password. New password: Enter the new password you want to use. Enter with up to 63 characters. **Note** Symbols below and 2-byte characters cannot be used. Confirm password: Enter the new password again. **Note** Currently, the IP telephone supports only user "N-SP80."

14. TROUBLE SHOOTING

Symptom	Remedy	Reference page	
		Operation	Setting
The station can neither make a call nor receive an incoming call.	Reboot the station or turn off and on the power. • In the case of the multimedia station, reboot it. • In the case of the door station, turn off and on the power.	 p. 7 p. 9	 — —
	Confirm the settings shown below according to the set mode. • When in Peer-to-peer mode Confirm the IP address setting. If it has returned to the initial value, reset it. [IP address's initial value] · Multimedia station: 192.168.1.101 · Door station: 192.168.1.102 • When in SIP server mode Confirm the SIP account's basic settings. If the displayed name and registered name have returned to the initial settings, reset them. [Displayed name's initial setting] · Multimedia station: No name assigned (Blank) · Door station: No name assigned (Blank) [Registered name's initial setting] · Multimedia station: No name assigned (Blank) · Door station: No name assigned (Blank)	 p. 35 p. 35 — — — —	 p. 40 p. 65 p. 38 p. 63 p. 38 p. 63

15. SPECIFICATION

15.1. N-SP80MS1 SIP Multimedia Station

Power Supply	PoE (IEEE802.3af) or 12 V DC (use of the optional AC adapter)
Power Consumption	12 W or less
Speech Method	Hands-free or Handset or Headset conversation
Audio Bandwidth	G.722 codec: up to 7 kHz
Hands-free	Speaker: 5 cm (1.97") Cone-type, Maximum output 2 W, 8 Ω Microphone: Omni-directional electret condenser microphone
Handset	Receive path: 3.6 cm (1.42") Cone-type, Maximum output 30 mW, 32 Ω Send path: Electret condenser microphone
Headset	Speaker: 20 mW, 16 Ω/32 Ω, ø3.5 mini jack (3P)
Display	7 inch capacitive touch screen TFT LCD, 800 x 480 pixels, 16:9 wide screen
Camera	Image device: 1/5" CMOS Number of effective pixels: 2 M pixels Maximum resolution: 1080 p Other features: Free-Rotation
Video	Image size: QCIF, QVGA, CIF, 4CIF, VGA Bit rate: 64 kbps – 2 Mbps
Network	Network I/F: 10BASE-T/100BASE-TX (Auto-Negotiation) Network protocol: UDP/TCP/IP, RTP/RTCP, ARP/RARP, NAT, NTP, IGMP, SIP, etc. Packet transmission system: Unicast, Multicast Paging: Multicast transmission x 1 Connector: RJ45, 2 ports (one supports PoE (IEEE802.3af)) Quantifying bit number: Maximum 16 bits Voice encoding method: G.711 μ-law/A-law, G.722, G.729 Video compression method: H.263, H.264
External Interface	USB 2.0
Installation Method	Desktop
Operating Temperature	0 °C to 40 °C (32 °F to 104 °F)
Operating Humidity	10 % to 90 %RH (no condensation)
Finish	Body, Handset: ABS resin, black
Dimensions	240 (w) x 214 (h) x 117 (d) mm or 9.45" x 8.43" x 4.61" (excluding a curl cord section)*
Weight	1.1 kg (2.43 lb)

* Numerical values are for reference only.

Note: The design and specifications are subject to change without notice for improvement.

• **Accessories**

Handset 1
Carl cord 1
LAN cable 1

• **Optional products**

AC adaptor: AD-1215P, AD-5000-2

15.2. N-SP80VS1 SIP Video Door Station

Power Supply	PoE (IEEE802.3af) or 12 V DC
Power Consumption	12 W or less
Speech Method	Hands-free conversation
Audio Bandwidth	G.722 codec: up to 7 kHz
Hands-Free	Speaker: 5 cm (1.97") Cone-type, Maximum output 1 W, 8 Ω Microphone: Omni-directional electret condenser microphone
Control Input	2 channels, no-voltage make contact input, open circuit voltage: 5 V DC, short-circuit current: 10 mA or less
Control Output	2 channels, relay output, normal open/normal close output, withstand voltage: 30 V DC, control current: 1 A
DC Power Input	12 V DC
Operation Button	Call button x 1, Reset button x 1
Indication	Status LED
Camera	Image device: 1/3" CMOS Number of effective pixels: 3 M pixels Maximum resolution: 1080 p Angle of View: (Horizontal) 120°, (Vertical) 64° IR light: IR LED Day & night mode: Color camera (day-mode)/black & white camera (night-mode), Automatic switching
Network	Network I/F: 10BASE-T/100BASE-TX (Auto-Negotiation) Network Protocol: IPv4, HTTP, HTTPS, FTP, SNMP, DNS, NTP, RTSP, RTP, TCP, UDP, ICMP, DHCP, ARP, SIP Packet transmission system: Unicast, Multicast Paging: Multicast receive x 1 Connector: RJ45, 2 ports (one supports PoE (IEEE802.3af)) Quantifying bit number: Maximum 16 bits Voice encoding method: G.711 μ-law/A-law, G.722, G.729 Video compression method: H.264, MPEG-4, MJPEG Security: Password protection, IP address filtering, SIP over TLS, HTTPS
Speech Features	SIP: SIPv1 (RFC2543), SIPv2 (RFC3261) Audio features: Acoustic echo canceller, VAD (Voice Activity Detection), Comfort noise generator Others: Auto answer, Volume control, Peer-to-peer connection (Direct IP connection without SIP server)
Installation Method	Flush-mount
Operating Temperature	–20 °C to +55 °C (–4 °F to 131 °F)
Operating Humidity	10 to 90 %RH (no condensation)
Dust/Water Protection	IP65 (Panel)
Finish	Panel: Stainless steel
Dimensions	120 (w) x 210 (h) x 49.5 (d) mm (4.72" x 8.27" x 1.95")
Weight	910 g (2.01 lb)
Applicable Box	4 size back box: YC-400

Note: The design and specifications are subject to change without notice for improvement.

• **Accessories**

Star head screw M4 x 16 for box mounting 4
Star head screwdriver 1

15.3. N-SP80AS1 SIP Audio Door Station

Power Supply	PoE (IEEE802.3af) or 12 V DC
Power Consumption	12 W or less
Speech Method	Hands-free conversation
Audio Bandwidth	G.722 codec: up to 7 kHz
Hands-Free	Speaker: 5 cm (1.97") Cone-type, Maximum output 1 W, 8 Ω Microphone: Omni-directional electret condenser microphone
Control Input	2 channels, no-voltage make contact input, open circuit voltage: 5 V DC, short-circuit current: 10 mA or less
Control Output	2 channels, relay output, normal open/normal close output, withstand voltage: 30 V DC, control current: 1 A
DC Power Input	12 V DC
Operation Button	Call button x 1, Reset button x 1
Indication	Status LED
Network	Network I/F: 10BASE-T/100BASE-TX (Auto-Negotiation) Network Protocol: IPv4, HTTP, HTTPS, FTP, SNMP, DNS, NTP, RTSP, RTP, TCP, UDP, ICMP, DHCP, ARP, SIP Packet transmission system: Unicast, Multicast Paging: Multicast receive x 1 Connector: RJ45, 2 ports (one supports PoE (IEEE802.3af)) Quantifying bit number: Maximum 16 bits Voice encoding method: G.711 μ-law/A-law, G.722, G.729 Security: Password protection, IP address filtering, SIP over TLS, HTTPS
Speech Features	SIP: SIPv1 (RFC2543), SIPv2 (RFC3261) Audio features: Acoustic echo canceller, VAD (Voice Activity Detection), Comfort noise generator Others: Auto answer, Volume control, Peer-to-peer connection (Direct IP connection without SIP server)
Installation Method	Flush-mount
Operating Temperature	–20 °C to +55 °C (–4 °F to 131 °F)
Operating Humidity	10 % to 90 %RH (no condensation)
Dust/Water Protection	IP65 (Panel)
Finish	Panel: Stainless steel
Dimensions	120 (w) x 210 (h) x 49.5 (d) mm (4.72" x 8.27" x 1.95")
Weight	870 g (1.92 lb)
Applicable Box	4 size back box: YC-400

Note: The design and specifications are subject to change without notice for improvement.

• **Accessories**

Star head screw M4 x 16 for box mounting 4
Star head screwdriver 1

15.4. YC-400   4 Size Back Box

Finish	Surface-treated steel plate, silver
Dimensions	98.2 (w) x 187 (h) x 58.1 (d) mm (3.87" x 7.36" x 2.29")
Weight	450 g (0.99 lb)