Thank you for purchasing TOA's N-8000 SIP Gateway.
Please carefully follow the instructions in this manual to ensure long, trouble-free use of your equipment.

TOA Corporation
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Chapter 1

General Description
1. GENERAL DESCRIPTION

This manual is described for N-8000 SIP Gateway that has interconnecting functions between N-8000 and SIP system.

2. SYSTEM IMAGE

![Diagram of N-8000 system and SIP system](image)

3. BASIC FUNCTIONS

N-8000 SIP Gateway has new functions by software working on Windows7 Professional 32/64bits, as below.

<table>
<thead>
<tr>
<th>Function of N-8000 SIP Gateway</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Interconnection between SIP and N-8000 functions: ②, ④, ⑥, ⑧, and ⑩</td>
<td>This software converts a call from N-8000 to SIP system and vice versa. One software has 5 speech paths at the same time. Transferring functions are also available from N-8000 to SIP system.</td>
</tr>
<tr>
<td>Paging from SIP to N-8000. functions: ⑦</td>
<td>This software brings a “zone paging” and an “All-call paging” to the system. SIP telephone can make any paging. One server has 5 paging channels at the same time.</td>
</tr>
<tr>
<td>Contact control from SIP telephone functions: ⑪</td>
<td>Dialing by a SIP telephone can control contact output of N-8640/50DS.</td>
</tr>
</tbody>
</table>

Note

②-⑩ indicates functions’ name which are explained in Section2.
4. SETTING STRUCTURE

There are 2 software settings to create a system, and another setting is needed for the other system which is SIP system.

**Note**

A~D indicates settings which are explained in Section3 and 4.
Chapter 2

Functions
1. CONDITIONS OF SIP GATEWAY

N-8000 SIP Gateway is tested with a SIP server of Cisco Systems, Asterisk, FreeSwitch, and Sipelia of Genetec Security Center.

2. SIP GATEWAY STRUCTURE

A SIP Gateway consists of 5 virtual master stations with one virtual N-8000EX, 5 virtual SIP clients, and audio processing functions. There are 5 channels of speech paths between SIP telephone and N-8000. SIP Gateway has an audio processing function.
3. SIP GATEWAY FUNCTIONS

3. SIP GATEWAY FUNCTIONS

There is a function of converting from SIP protocol to N-8000 and vice versa.

There are 5 speech paths between SIP telephone system and N-8000.

SIP Gateway converts from SIP audio to N-8000 audio, and vice versa.

N-8000 station is connected to SIP telephone via N-8000 virtual station in a SIP Gateway.

SIP telephone is connected to N-8000 station via virtual SIP Client station in a SIP Gateway.

SIP telephone makes a station paging to N-8000 system with additional dials.

SIP telephone makes contact output control of N-8640/50DS as known “Door Remote Control” function.

3.1. Details of functions

Functions of ③ and ④ are skipped.

3.1.1. Audio conversion

A SIP Gateway converts audio from N-8000 to SIP telephone and vice versa. Both sides have a different sampling frequency and audio codec.

- N-8000  16 kHz, 8 kHz/ G.722
- SIP system  8 kHz/ G.711 u-law
3.1.2. Call from N-8000 to SIP
By an example, its function is explained below.

[System example]

[Setting concept]
When N-8000 calls 801, virtual SIP client 1 has a setting to call to 2001.

Automatically transfer to 2001.

When N-8000MS(102) calls 805, virtual SIP client 5 has a setting to call to 2005,
within a selected time (0 to 5 seconds), N-8600MS dials 2004, then talk.

[Operation]

Tip
This "additional dial" function is only for N-8600MS with a special firmware. (The firmware is in a CD.)
3.1.3. Call from SIP to N-8000
By an example, its function is explained below.

[Setting concept]
When SIP telephone calls 3995, virtual N-8000MS 1 has a setting of calling 101.
→ SIP telephone and N-8000 101 starts to talk.

When SIP telephone calls 3999, virtual N-8000MS 5 has a setting of calling 202,
Within a selected time (0 to 5 seconds), SIP telephone dials 103, then talk.

[Operation]
3.1.4. Paging function
Paging from SIP telephone to N-8000 system.

3.1.5. Control functions
Door remote control from SIP telephone for Door stations.
Chapter 3

Settings
1. SETTINGS

1.1. Setting image for N-8000 side and SIP telephone side, and a main of SIP Gateway

There are 3 steps of setting which are setting of N-8000 system, setting of SIP system, and setting of SIP gateway.

1.2. Setting of N-8000 system

This N-8000 SIP Gateway is to be set as a N-8000EX and 5 N-8000MS, virtually. The following is an example of a setting for SIP Gateway.

**Note:** This Equipment No. of virtual N-8000EX is used for the setting of SIP Gateway.
1.3. Preparation for B and C

The following is a window of SIP Gateway.

Settings are by a clicking File, there are menu of B: “SipClientSetting” and C: “SipGatewaySetting.”

1.4. Preparation for SIP Client setting

Connecting to SIP system, the following info. is needed.

<table>
<thead>
<tr>
<th>No.</th>
<th>Item</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>IP address of SIP Server</td>
<td>–</td>
</tr>
<tr>
<td>2</td>
<td>SIP Port of SIP Server</td>
<td>Default is “5060”</td>
</tr>
<tr>
<td>3</td>
<td>Extension number for SIP Clients</td>
<td>SIP Gateway can have 5 SIP Clients</td>
</tr>
<tr>
<td>4</td>
<td>SIP device profile</td>
<td>Password requirement, and a method of Authentication can be set.</td>
</tr>
</tbody>
</table>

Registration command to a SIP server is executed right after starting this software up, and by every 140 seconds.

SIP Gateway Setting

<SIP Server>
IP address = 192.168.1.100 (e.g.)
Port for SIP = 5060 (e.g.)

Every 140 sec.

SIP Server
IP address = 192.168.1.100 (e.g.)
Port for SIP = 5060 (e.g.)
1.5. ⑵ SIP Client Setting

Here is for all about SIP system.

**Step 1.** Setting for SIP Server. The port number with 5060 is a default of SIP protocol.

**Step 2.** SIP virtual Client telephone number which is registered to SIP server.

**Step 3.** A password for a SIP virtual Client for an authentication.

**Step 4.** Settings of IP address and RTP port for each virtual SIP client.
1.6. SIP Gateway Setting

Step 5. A location of N-8000 setting file which is <system name>.pi8 file.

Step 6. Waiting time for specific dialing from N-8000 or SIP telephone. After this time, SIP Gateway connects automatically to a certain station which is set in 4 for calling from N-8000, or in 5 for calling from SIP telephone.

Step 7. N-8000 equipment number, these should be matched to a setting of N-8000Next setting file selected in 1.

Step 8. Connecting N-8000 number which is called from SIP telephone side. Each virtual SIP Client has an individual target of N-8000 station.

Step 9. Connecting SIP telephone number which is called from N-8000 side. Each virtual N-8000 station has an individual target of SIP telephone.
1.7. Setting example of N-8000 SIP Gateway

The following is an example of a setting that N-8000’s call to SIP.

[System example]

When N-8000 side calls 801, N-8000 SIP Gateway connects to 2001 and 3 speech paths are prepared for this function, and when N-8000 side calls 804, N-8000 SIP Gateway connects to 2005 and 2 speech paths are prepared for this function.

<table>
<thead>
<tr>
<th>N-8000 station</th>
<th>N-8000 setting content</th>
<th>Setting of N-8000 SIP Gateway</th>
</tr>
</thead>
<tbody>
<tr>
<td>801</td>
<td>802 is for “Group-hunting”</td>
<td>virtual SIP client 1 has 2001 in SipGatewaySetting</td>
</tr>
<tr>
<td>802</td>
<td>803 is for “Group-hunting”</td>
<td>virtual SIP client 2 has 2001 in SipGatewaySetting</td>
</tr>
<tr>
<td>803</td>
<td></td>
<td>virtual SIP client 3 has 2001 in SipGatewaySetting</td>
</tr>
<tr>
<td>804</td>
<td>805 is for “Group-hunting”</td>
<td>virtual SIP client 4 has 2005 in SipGatewaySetting</td>
</tr>
<tr>
<td>805</td>
<td></td>
<td>virtual SIP client 5 has 2005 in SipGatewaySetting</td>
</tr>
</tbody>
</table>

When SIP telephone calls 3995, N-8000 SIP Gateway connects to 101 and 2 speech paths are prepared for this function, and when SIP telephone calls 3997, N-8000 SIP Gateway connects to 102 and 2 speech paths are prepared for this function, and when SIP telephone calls 3999, N-8000 SIP Gateway connects to 103.

<table>
<thead>
<tr>
<th>SIP extension</th>
<th>SIP client setting content</th>
<th>Setting of N-8000 SIP Gateway</th>
</tr>
</thead>
<tbody>
<tr>
<td>3995</td>
<td>3996 is for “Busy transfer”</td>
<td>virtual N-8000MS 1 has 101 in SipGatewaySetting</td>
</tr>
<tr>
<td>3996</td>
<td></td>
<td>virtual N-8000MS 2 has 101 in SipGatewaySetting</td>
</tr>
<tr>
<td>3997</td>
<td>3998 is for “Busy transfer”</td>
<td>virtual N-8000MS 3 has 102 in SipGatewaySetting</td>
</tr>
<tr>
<td>3998</td>
<td></td>
<td>virtual N-8000MS 4 has 102 in SipGatewaySetting</td>
</tr>
<tr>
<td>3999</td>
<td></td>
<td>virtual N-8000MS 5 has 103 in SipGatewaySetting</td>
</tr>
</tbody>
</table>
In the N-8000 Setting software, N-8000 SIP Gateway is set as N-8000EX. IP address and port is for a PC that N-8000 SIP gateway software is installed.

**Note:** In the later step, equipment No. will be used.

The next is setting for virtual N-8000MS. Line 1 to 5 are used for N-8000 SIP Gateway, 801 to 805 are set according to this example.
This is for the function that calling to 801 in the N-8000 system has 3 speech paths that goes to 2001 in SIP system.
The following example is a setting for 801 to add 802 in the box of “Group hunting to.” Additionally a setting for 802 is needed to transfer to 803. While 804 in the N-8000 system connecting to 2005 in SIP system needs 2 speech paths, then a setting for 804 needs to have 805 in the box of “Group hunting to.”

After all setting, a setting file shall be exported. As shown below.
The next step is a setting of N-8000 SIP Gateway. N-8000 SIP Gateway uses a N-8000 Setting file that is `<system_name>.pi8`. Then the equipment No. that is programmed in the N-8000 Setting software (in this example, it’s 3). In the left with “Transfer Number” requires numbers that N-8000 SIP Gateway connects to N-8000 system side by a request from SIP system. The right box for “Transfer Number” is for SIP telephone number that N-8000 SIP Gateway connects by a request from N-8000 system.

A setting of SIP server is also needed. All SIP server has a different way for the setting, therefore the following is just basic setting contents.

- 5 SIP clients for N-8000 SIP Gateway shall be prepared. Extension numbers shall be 3995 to 3999 for this example.
- Passwords for each extension numbers are set.

Finally SIP server setting is needed for N-8000 SIP Gateway setting. That can be done by selecting from a menu of “File/SipClientSetting.”

That’s all.
Chapter 4

Setting of SIP server
### 1. SETTING OF SIP SERVER

What N-8000 SIP Gateway requires are as below;
- Extension number (telephone number) for SIP clients.
- Corresponding password for each extension number.

Additional setting like transferring function can be set by other settings.

#### 1.1. Setting for Asterisk

Files for a basic setting are with sip.conf and extensions.conf.

Configuration idea with sip.conf (for Extension number and password)

```conf
[general]
context=default
port=5060
bindaddr=0.0.0.0
language=ja
musiconhold=default
disallow=all
allow=ulaw
;allow=alaw
;allow=gsm
;allow=ilbc
dtmfmode=rfc2833
[3995]  * RFC2833 must be set here for DTMF tone.
type=friend
defaultuser=3995
secret=pass
canreinvite=no
host=dynamic
dtmfmode=rfc2833
[3996]  * This is a setting for “3996” to be added.
type=friend
defaultuser=3996
secret=pass
canreinvite=no
host=dynamic
dtmfmode=rfc2833

Configuration idea with extensions.conf (for a setting of process of SIP Server)

[default] section as below will be modified

```conf
exten => 3995,1,Dial(SIP/3995,12)  * 3995 means a station number that is set in sip.conf file.
exten => 3995,2,Congestion
exten => 3995,102,Busy
exten => 3996,1,Dial(SIP/3996,12)  * 3996 means a station number that is set in sip.conf file.
exten => 3996,2,Congestion
exten => 3996,102,Busy
exten => 3997,1,Dial(SIP/3997,12)  * 3997 means a station number that is set in sip.conf file.
exten => 3997,2,Congestion
exten => 3997,102,Busy

```conf
exten => _.,1,Answer()
exten => _.,2,Wait(2)
exten => _.,3,Playback(pbx-invalid)
exten => _.,3,Congestion
1.2. Setting for FreeSwitch

Files for a basic setting are with directory/default/****(Extension number).xml and dialplan/default.xml. Configuration idea with (Extension number).xml (Extension number and password can be set in this file.)

**Step 1.** New user setting file is added.
That file like 3995.xml can be created by copying 1000.xml. The file is for one user.

**Step 2.** The new setting file is modified from a file made in the step 1.
   e.x.) 1000.xml is copied and created 3995.xml to 3999.xml.

Modification idea for 3995.xml
<include>
   <user id="3995">
      <params>
         <param name="password" value="${default_password}"/>
         <param name="vm-password" value="3995"/>
      </params>
      <variables>
         <variable name="toll_allow" value="domestic,international,local"/>
         <variable name="accountcode" value="3995"/>
         <variable name="user_context" value="default"/>
         <variable name="effective_caller_id_name" value="Extension 3995"/>
         <variable name="effective_caller_id_number" value="3995"/>
         <variable name="outbound_caller_id_name" value="${outbound_caller_name}"/>
         <variable name="outbound_caller_id_number" value="${outbound_caller_id}"/>
         <variable name="callgroup" value="techsupport"/>
      </variables>
   </user>
</include>

<table>
<thead>
<tr>
<th>Item</th>
<th>Content</th>
</tr>
</thead>
<tbody>
<tr>
<td>user_id</td>
<td>Extension number</td>
</tr>
<tr>
<td>vm-password</td>
<td>Voicemail password</td>
</tr>
<tr>
<td>accountcode</td>
<td>Authenticated extension number</td>
</tr>
<tr>
<td>effective_caller_id_name</td>
<td>User name notified to the corresponding client</td>
</tr>
<tr>
<td>effective_caller_id_number</td>
<td>Number notified to the corresponding client</td>
</tr>
</tbody>
</table>
Configuration idea for dialplan/default.xml (5 clients for N-8000 SIP Gateway is added, and busy transferring function is added.)

**Example:**

```xml
<extension name="N8000SIP1_Extension">  
  *  For User1 (3995)
  <condition field="destination_number" expression="^3995$">  
    <action application="export" data="dialed_extension=$1"/>
    <action application="set" data="hangup_after_bridge=true"/>
    <action application="set" data="continue_on_fail=true"/>
    <action application="bridge" data="user/$(dialed_extension)@$domain_name"/>
    <action application="answer"/>
    <action application="transfer" data="3996"/>  
    * 3996 is a station number transferred from 3995
  </condition>
</extension>

<extension name="N8000SIP2_Extension">  
  *  For User2 (3996)
  <condition field="destination_number" expression="^3996$">  
    :  
    :  
    :  
  </condition>
</extension>
```

<table>
<thead>
<tr>
<th>Action</th>
<th>Data</th>
<th>Content</th>
</tr>
</thead>
<tbody>
<tr>
<td>extension_name</td>
<td>Unique name for a dial plan.</td>
<td></td>
</tr>
<tr>
<td>destination_number</td>
<td>User ID for a setting target. Multi-user ID can be set.</td>
<td></td>
</tr>
</tbody>
</table>
| set            | hangup_after_bridge                 | Whether the next dial plan is processed or not in case of “hung-up bridge status.”
|                |                                     | true: Stop the current process.                                       |
|                |                                     | false or no definition: Continue the current process.                 |
| set            | continue_on_fail                    | Whether the next dial plan is processed or not in case of busy or off-line status on the receiving station.
|                |                                     | true or error-code: Continue the current process.                     |
|                |                                     | false: Stop the current process.                                      |
| bridge         | user/$(dialed_extension)@$domain_name | Receiving process.                                                    |
| answer         |                                     | Answer to a call from others or session from others.                  |
| transfer       | From 3996 to 3999                   | In case that this client/user is occupied by any reasons, a client/user in this section is transferred from this client/user. |
| sleep          | 1000                                | This is a time of lasting dial plan's process.                        |
| bridge         | loopback/app=voicemail:default      | In case of no call, the caller who made a call to this client/user will receive a voicemail. |
1.3. Setting of Cisco Call Manager

1.3.1. Registration procedure of SIP Client into Cisco SIP Server

Step 1. Create a new profile with Digest Authentication for Telephone Security *only the beginning.

1-1

1-2

1-3

1-4
### Chapter 4  Setting of SIP server

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
</table>

#### Phone Security Profile Configuration

<table>
<thead>
<tr>
<th>Product Type</th>
<th>Third-party SIP Device (Basic)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Name</td>
<td>Third-party SIP Device Basic - Digest Required</td>
</tr>
<tr>
<td>Description</td>
<td>Third-party SIP Device (Basic) - Digest Required</td>
</tr>
<tr>
<td>Nonce Validity Time</td>
<td>500</td>
</tr>
<tr>
<td>Transport Type</td>
<td>TCP+UDP</td>
</tr>
</tbody>
</table>

**Enable Digest Authentication**  
Full fill boxes as shown in left, specially "Enable Digest Authentication" shall be checked, then save the setting

---

**Parameters used in Phone**

| SIP Phone Port | 5060 |

---

Operating Instructions  4-6
Chapter 4   Setting of SIP server

Step 2. Adding extension number

2-1. Select a setting of "Call Routing/Directory Number"

Add a new Directory number

Directory Number Configuration

2-3. Fill in the boxes as shown in the left.
Directory number can be a range of SIP Client numbers. N-8000 SIP Gateway has 5 SIP Clients, then a range is like "3995" to "3999."
Chapter 4   Setting of SIP server

Step 3. Adding SIP Client

3-1
Select "Device/Phone"

3-2
Add a new phone

3-3
Select the phone type that is created in a profile creating step, then go to the next

3-4

### Chapter 4  Setting of SIP server

#### Phone Configuration

**Status**
- Status: Ready

**Phone Type**
- Product Type: Third-party SIP Device (Basic)
- Device Protocol: SIP

#### Device Information

<table>
<thead>
<tr>
<th>Device is not trusted</th>
<th>MAC Address*</th>
<th>000C29CB913B</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Device Pool*</td>
<td>SIP0200C29CB913B</td>
<td></td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Phone Button Template*</td>
<td>Third-party SIP Device (Basic)</td>
<td></td>
</tr>
<tr>
<td>Common Phone Profile*</td>
<td>Standard Common Phone Profile</td>
<td></td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>CSS_Take_EXT</td>
<td></td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_None</td>
<td></td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Device Mobility Mode*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Owner User ID</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Use Trusted Relay Point*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Always Use Prime Line*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Message*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

- **Use Device Pool Calling Party Transformation CSS**
- **Ignore Presentation Indicators (internal calls only)**
- **Logged Into Hunt Group**
- **Remote Device**

---

#### Protocol Specific Information

<table>
<thead>
<tr>
<th>Presence Group</th>
<th>Standard Presence group</th>
</tr>
</thead>
<tbody>
<tr>
<td>MTP Preferred Originating Codec*</td>
<td>711law</td>
</tr>
<tr>
<td>Device Security Profile*</td>
<td>Third-party SIP Device Basic - Digest Required</td>
</tr>
<tr>
<td>Rerouting Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP Profile*</td>
<td>Standard SIP Profile</td>
</tr>
<tr>
<td>Digest User</td>
<td>3999</td>
</tr>
</tbody>
</table>

- **Media Termination Point Required**
- **Unattended Port**
- **MLPP Information**
  - MLPP Domain: < None >

---

**Save**

---

*Full fill boxes as shown in left*

*3-5*

*3-6*

*3-7*

*3-8*

---

Operating Instructions  4-9
Chapter 4 Setting of SIP server

Find and List Phones

Find Phone where Device Name begins with

Phone Configuration

Association Information

Phone Type

Directory Number Configuration

Directory Number Information

Directory Number Settings

Operating Instructions 4-10
3-14

"e.g., "ring transferring setting" like 3995 transferred to 3996, 3996 transferred to 3997, ......, is one of the idea to use easily a function of SIP Gateway.

3-15

This setting is for a condition that how many calls can be received by a SIP client. The number is 1 or 2.
Step 4. Adding users

Make setting of a new user, this example is for a SIP Client with a number '3999.' The digest credentials shall be set in this setting window.

Please save it until these settings.

*Until this setting, pressing save is better to continue.
Select as shown in left, and make a device association.

Add to the group shown in left, and save it.