

OPERATING INSTRUCTIONS

Rev. 1.00

N-8000 SIP GATEWAY

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



3. BASIC FUNCTIONS

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

Function of N-8000 SIP Gateway	Description		
Interconnection between SIP and N-8000 functions: ⓐ, ⓑ, ⓒ, ⓓ, and ⓔ	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.		
Paging from SIP to N-8000. functions: $①$	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.		
Contact control from SIP telephone functions: (9)	Dialing by a SIP telephone can control contact output of N-8640/50DS.		

Note

(a)-(g) 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

 $\ensuremath{\mathbb{A}}\xspace$ -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

- ⓐ There is a function of converting from SIP protocol to N-8000 and vice versa.
- (b) There are 5 speech paths between SIP telephone system and N-8000.
- \odot SIP Gateway converts from SIP audio to N-8000 audio, and vice versa.
- (d) N-8000 station is connected to SIP telephone via N-8000 virtual station in a SIP Gateway.
- (e) SIP telephone is connected to N-8000 station via virtual SIP Client station in a SIP Gateway.
- (f) SIP telephone makes a station paging to N-8000 system with additional dials.
- (B) SIP telephone makes contact output control of N-8640/50DS as known "Door Remote Control" function.



3.1. Details of functions

Functions of (a) and (b) are skipped.

3.1.1. \bigcirc 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. \bigcirc 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.



Tip

This "additional dial" function is only for N-8600MS with a special firmware. (The firmware is in a CD.)

3.1.3. (e) Call from SIP to N-8000

By an example, its function is explained below.

[System example]



[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]

		Selectable from 0 to 5 seconds
	5 seconds	\rightarrow
<dial from="" n-8000"103"="" sip"2004"="" to=""></dial>	3 9 9 9 Within 5 seconds	1 0 3
<dial from="" n-8000"2"="" sip"2004"="" to=""></dial>	3999	Automatic transfer to "202"/N-8000

3.1.4. (f) Paging function

Paging from SIP telephone to N-8000 system.



3.1.5. (g) 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. (A) 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.

M-8000 Software	A		1 100 mar					
<u>File</u> <u>C</u> onfiguration <u>H</u> elp								
General Exchange Multi Interface Sub-stati	ion Interface IP Stat	tion Station C	C/O Interface T	elephone Ir	nterface Auc	dio Interface	Direct Selec	t Gateway Paging Group
Equipment Content Registration					Total n	umber of ea	uipments con	nected to network : 3/192
Station Table Exchange :								
Network N-8000EX 1	N-8010EX 0							
Registration IP Station :								
Multicast Registration N-8500MS 0	N-8510/5MS 0	N-8600MS	2 N-861	.6MS	0 N-8610F	RM 0	N-8540DS	0 N-8640/50DS 0
System Settings Interface :								
Gateway Settings N-8000RS 0	N-8010RS 0	N-8400RS	0					
N-8000CO 0	N-8000AL 0	N-8000AF	0 N-800	IMI <mark>This</mark>	is for	SIP Gat	teway.	IP address is set for an IP
							Star	ting Equipment Number 4
Connected to : Enter th	e set value to the err	uinment Value s	et to the Tree	add	ress of	PC wh	<u>ich is ir</u>	nstalled N-8000 SIP Gateway.
equipmo	ent cannot be change	d here.	scan	result				
Equipment No. E	quipment name Mod		ies Web Port	+ WAN TO 4	Address WAN	Web Port		
	-8600MS	500MC 102 168	1 100 80		nucles that	(neb Port		
	-8000EX N-80	000EX 192.168.	1.11 80					
3	-8000MS	000MB 192.108.	1.101 80					
P. N-1	8000 Software							
Eile	Ele Configuration Help							
Gene	eral Exchange Multi	Interface Sub-st	tation Interface	IP Station	Station C/O	Interface Te	lephone Inter	ace Audio Interface Direct Select Gateway Paging Group
Equ	ipment Registration St	tation No. digits :	Collectiv	ve Setup				These are also for SIP Gateway.
Delete Checked	Station Table	Equipment No.	Equipment name	Line No. T	ype	Station No.	Station name	
	Communication	1	N-8600MS		I-8600MS -	102	N-8600MS	Station numbers for them are
M	Registration	2	N-8000EX	1 N	-8000MS -	201		
Mu	System Settings			2 N	-8000MS -	202	K	kinds of "access code".
	Gateway Settings			3 N	-8000MS -	203		inter of access code i
				4 N	-8000MS +	204		When N-8000 station wants to
				5	8000MS •	205		when he bood station wants to
				6	•			connect to SIP system, these
				8				
				9				station numbers can be dialed.
				10	•			
				10	•			
				10 11 12	•			
				10 11 12 13	• • •			
				10 11 12 13 14	• • •			
				10 11 12 13 14 15	• • • •			
				10 11 12 13 14 15 16	- - - - - -			
		3	N-8600MS	10 11 12 13 14 15 16 16 1 1 N	• • • • • • • • • • • • • •	101	N-8600MS	

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.

🔒 SIPGatev	way				
File Help SipGat	tewaySetting	B		SIP Gateway	
SipClie	entSetting	C DOO Term No.	SIP Client No.	N8000 Term S	tatus Term Status
1	Idle	poorteniirite	p	1	Įdle
2	Idle	D	D	2	Idle
3	Idle	ρ	p	3	Idle
4	Idle	o	p	4	Idle
5	Idle	0	0	5	Idle
6	Idle	0	o	SIP Client Stat	us
7	Idle	0	o	Client No	Client Status
8	Idle	0	0	1	Not Connect
9	Idle	ρ	p	2	Not Connect
10	Idle	p	p	3	Not Connect
				4	Not Connect
				5	Not Connect
					Exit

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.

No.	Item	Comment		
1	IP address of SIP Server	-		
2	SIP Port of SIP Server	Default is "5060"		
3	Extension number for SIP Clients	SIP Gateway can have 5 SIP Clients		
4	SIP device profile	Password requirement, and a method of Authentication ca		
		be set.		

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



1.5. ^B 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.

SIPClient Setting	Contraction of the second seco	×
SIP Server Setting	(1)	
IP Address	<u>192</u> . 168 . 11 . 200	
Port No	5060	
SIP Client Setting		
No Phone warber	Password 3 SIP Client IP Address RTP Rear Port No	
1 201	••••• 192 . 168 . 11 . 102 4001	
2 202	•••••• 192 . 168 . 11 . 102 4002	
3 203	•••••• 192 . 168 . 11 . 102 4003	
4 204	192 168 11 102 4004	
203	•••••• 192 168 . 11 . 102 4095	
	Save Cancel	

1.6. ^(C) 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.

SIPGateway Setting	X
N-2000 Setting File Name C:¥ProgramData¥TOA¥N-8 Brow N-8000 Term Setting	s 5 SIP Client Setting
No Equipment No Line No Transfer Number 1 3 7 1 101 8 2 3 2 102 3 3 3 103 4 3 4 201 5 3 5 202	No Transfer Number 1 2001 2 2002 3 2003 2 2004 5 2005
	Save Cancel

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]



[Setting 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.

N-8000 station	N-8000 setting content	Setting of N-8000 SIP Gateway
801	802 is for "Group-hunting"	virtual SIP client 1 has 2001 in SipGatewaySetting
802	803 is for "Group-hunting"	virtual SIP client 2 has 2001 in SipGatewaySetting
803		virtual SIP client 3 has 2001 in SipGatewaySetting
804	805 is for "Group-hunting"	virtual SIP client 4 has 2005 in SipGatewaySetting
805		virtual SIP client 5 has 2005 in SipGatewaySetting

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.

SIP extension	SIP client setting content	Setting of N-8000 SIP Gateway
3995	3996 is for "Busy transfer"	virtual N-8000MS 1 has 101 in SipGatewaySetting
3996		virtual N-8000MS 2 has 101 in SipGatewaySetting
3997	3998 is for "Busy transfer"	virtual N-8000MS 3 has 102 in SipGatewaySetting
3998		virtual N-8000MS 4 has 102 in SipGatewaySetting
3999		virtual N-8000MS 5 has 103 in SipGatewaySetting

[Example of N-8000 Setting software, N-8000 SIP Gateway software]

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.

N-8000 Software							-	-	-	- • ×
<u>File</u> <u>Configuration</u> <u>He</u>	elp									
C/O Interface	Telephone I	Interface	Audio	o Interfa	ce C	irect Selec	t (Gateway	Paging	Group
General	Exchange		Multi Interf	ace	Sub	-station In	terface	IP S	tation	Station
Equipment Registration	Content					Total n	umber of equ	ipments conn	ected to netw	ork : 6 / 192 -
Station Table	Exchange :									
Network	N-8000EX	1 N-80	10EX	0						
Registration	IP Station :									
Multicast Registration	N-8500MS	0 N-85	10/5MS	0 N-	8600MS	3 N-861	омз о	N-8616MS	0 N-	8610RM 0 ≡
System Settings	N-8540DS	0 N-86	40/50DS	2						
Gateway Settings	Interface :									
	N-8000RS	0 N-80	10RS	0 N-8	400RS)				
	N-8000CO	0 N-80	00AL	0 N-8	000AF	N-8000	0 IMO	N-8000DI	0 SX-	200IP 0
	Connected to : En	nter the set v quipment can	alue to the e not be chang	quipmer ged here	nt. Value set to	the Impo	ort from result			
	Equipment I	No. Equipme	nt name Mo	odel	IP Address	Web Port	WAN IP Add	ress WAN We	eb Port	
	1	N-8600M	S N-	8600MS	10.5.111.102	80				
	2	N-8640D	S N-	8640DS	10.5.111.103	80				
	3	N-8000E	X N-	8000EX	10.5.111.100	80				
	4	N-86400	S N-	8640DS	192.168.1.1	80				
	6	N-8600M	IS N-I	8600MS	192.168.1.1	80				
	Delete Checked B	Equipments								
	<u></u>									

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.

N-8000 Software					-	-	-		- • ×
<u>File</u> Configuration <u>H</u> e	lp								
C/O Interface	Telephone Int	erface	Audio Interfa	ce	Direct Select	Ga	teway	Paging	Group
General	Exchange	Mult	i Interface		Sub-station Int	erface	IP St	tation	Station
Equipment Registration	Station No. digits	: 3 🔹 Collec	tive Setup						
Station Table					C 1-11-11-11-	C 1			
Network	Equipment No.	Equipment nam	e Line No. Ty	ype	Station No.	Station name			
Communication	1	N-8600M	S I N	-8600MS	101	N-8600MS			
Multicast Registration	2	N-8640D	S 1 N	-864005	201	N-8640DS			
System Settings	- 3	N-8000E	X 1 N	-8000MS	801	SIPGW1			
Gateway Settings			2 N	-8000MS	802	STPGW2	<u> </u>		
			3 N	-8000MS	803	SI GW3			
			4 N	-8000MS	804	STPGW4			
				-8000MS	805	SIPGW5			
	-		6						
	-		7				+		E
			-		•		<u> </u>		
			•		•		L		
			9		•				
			10		•				
			11		•				
			12		•				
			13		•				
			14		•		1		
			15		•				
			16		-				
									·

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."

configuration	Heib						
C/O Interface	Telephone Interfac	ce Audio	Interface	Direct Select	Gateway	Paging	Group
General	Exchange	Multi Interf	ace	Sub-station Interface	IP S	tation	Station
General tation Selection change No. : 3 ation No. : 80 unction Settings Speed Dialing Scan Monitor	Exchange Exchange name Exchange name Exchange name Incoming call mode Automatic connection Continuous call Level Microphone sensitivity : Speaker output : Call volume : Group call (max. 15) Station No. :	Multi Interf	Priority call Priority call Priority Lev BGM Receives Door remot MI/DI/AF e Contact oul Calling stati Control out MI/DI/A Contact Called stati	Sub-station Interface Line No. : 1 el : 1 BGM BGM input : e quipment No. : F equipment No. : F equipment No. : on's No. : versation	Audio Trigg Access t Access t Call ma	er to audio trigger to audio trigger tester () Monitor e contact No Contact N () Audio evel) : 5 + : 5 + udio) : 5 + : 5 + ntrol input on mode	Station
	Calls transferred to		Recording	AF No. :		on mode	
	Call forwarding to	:	Speaker sel	ection	Call Activa	ation - push 3 tin	nes
	Group hunting t	ing to :			Access to	priority call oper	ation
	Absence transfer to	. 802	User Access	Code		emergency call	operation
	Absence didfisier to		- A attache			chick geney can t	speration

After all setting, a setting file shall be exported. As shown below.

N-8000 Software			- C.				-				
e Configuration Help											
Upload		face	Audio	Interface	Direct Select	Gate	eway	Paging		Group	
Download			ulti Interfa	ce	Sub-station Interface		IP St	ation	Sta	tion	
Sta <u>E</u> xport tat <u>Export</u>	n Check	me : 1-8 : SIPC	000EX GW1	• L	ine No. : 1 🔹						
Function Settings Speed Dialing Scan Monitor Incoming call mode Automatic connection Continuous call With call to Level Microphone sensitivity : 2 • Speaker output : 3 • Call volume : 3 • Group call (max. 15) Station No. :		call tone call tone	Priority call Priority Level : 1 BGM Receives BGM BGM input : Door remote MI/DI/AF equipment No. : Calling station indication / CCTV control Control output : MI/DI/AF equipment No. :			Audio Trigger Access to audio trigger Mode Call master Monitor Remote contact Equipment No Contact No. : Trigger Export 					
				Contact Called static	output No. :			IP Addre 10.5.111 10.5.111	ess No 102 1 103 2 100 3	Name N-8600MS N-8640DS N-8000EX	Type N-8600MS N-8640DS N-8000EX
Call	s transferred to			Recording	AF No. :			192.168.	1.12 4	N-8640DS	N-8640DS
Tim Gro Abs	e-based call forwa up hunting to sence transfer to	ording to : :	802	Speaker sele	Code			192.168.	1.13 5 1.14 6	N-8600MS	N-8600MS
				Activate	ada .					UN	CANCE

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. 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.
- $\cdot\,$ 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;

- $\cdot\,$ Extension number (telephone number) for SIP clients.
- \cdot 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) [general] context=default port=5060 bindaddr=0.0.0.0 language=ja musiconhold=default disallow=all allow=ulaw * G.711/u-law setting must be set. ;allow=alaw ;allow=gsm ;allow=ilbc dtmfmode=rfc2833 * RFC2833 must be set here for DTMF tone. * This is a setting for "3995" to be added. [3995] type=friend defaultuser=3995 secret=pass * Here is for a password. canreinvite=no host=dynamic dtmfmode=rfc2833 *RFC2833 * This is a setting for "3996" to be added. [3996] type=friend defaultuser=3996 secret=pass * Here is for a password. canreinvite=no host=dynamic dtmfmode=rfc2833 *RFC2833 Configuration idea with extensions.conf (for a setting of process of SIP Server) [default] section as below will be modified 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 2 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>
```

</include>

</user>

Item	Content
user_id	Extension number
vm-password	Voicemail password
accountcode	Authenticated extension number
effective_caller_id_name	User name notified to the corresponding client
effective_caller_id_number	Number notified to the corresponding client

Configuration idea for dialplan/default.xml (5 clients for N-8000 SIP Gateway is added, and busy transferring function is added.)

e.x.)

<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 <action application="sleep" data="1000"/> <action application="bridge" data="loopback/app=voicemail:default \${domain_name} \${dialed_extension}"/> </condition> </extension> * For User2 (3996) <extension name="N8000SIP2 Extension"> <condition field="destination_number" expression="^(3996)\$">

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Action	Data	Content
	extension_name	Unique name for a dial plan.
	destination_number	User ID for a setting target. Multi-user ID can be set.
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.

System - Call Routing - Media F	Resources 👻 Advanced Features 👻 Device 👻 Application 👻 User Management 👻 Bulk Administration 👻 Help 👻
Server	
Cisco Unified CM	
Cisco Unified CM Group	
Phone NTP Reference	
Date/Time Group	Administration
Presence Group	
Region	
Device Pool	
Device Mobility	•
DHCP	•
LDAP	▶ 1:17 AM
Location	
Physical Location	11C.
SRST	res and is subject to United States and local country laws governing import, export, transfer and use. Delivery
MLPP	 rters, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By usin away return this product immediately.
Enterprise Parameters	
Enterprise Phone Configuration	cryptographic products may be found at our <u>Export Compliance Product Report</u> web site.
Service Parameters	unications Manager please visit our <u>Unified Communications System Documentation</u> web site.
Security	Certificate
Application Server	Phone Security Profile
Licensing	SIP Trunk Security Profile
Geolocation Configuration	CUMA Server Security Profile Select a setting of "System/Security/Phone
Geolocation Filter	Security Profile"

esources 👻 Advar	nced Features 👻	Device 👻 App	olication 👻 User Mana	agement 👻 Bulk Administr	ation 👻 Help 👻
rofiles					
Name 🔻 b	egins with 🔻		Find	Clear Filter 🔂 🛥]
		No ac	tive query. Please enter	your search criteria using th	ne options above.
Add a new pr	ofile				
	Advar rofiles	Advanced Features rofiles Name begins with	esources Advanced Features Device App rofiles Name begins with No ac Add a new profile	esources Advanced Features Device Application User Mana rofiles Name begins with Find No active query. Please enter Add a new profile	esources Advanced Features Device Application User Management Bulk Administr rofiles Name begins with No active query. Please enter your search criteria using th Add a new profile

Phone Security Profile Configuration	
Next	
Status Status: Ready	1-3
Select the type of device profile you would like to create Phone Security Profile Type ¹ Third-party SIP Device (Basic)	▼
- Next	Select as shown in left, and go to the next

Phone Security Prof	ile Configuration						
Save							
-Status							
i Status: Ready							
-Phone Security Pro	file Information ————						
Product Type: Device Protocol:	Third-party SIP Device (Basic) 1-5						
Name*	Third-party SIP Device Basic - Digest Required						
Description	Third-party SIP Device (Basic)	Third-party SIP Device (Basic) - Digest Required					
Nonce Validity Time*	600						
Transport Type*	TCP+UDP	T					
🖉 Enable Digest Aut	hentication	□ 1-6					
-Parameters used in	Phone						
SIP Phone Port* 506	0						
Save 1-7		Full fill boxes as shown in left, s Digest Authentication" shall be save the setting	pecially "Enabl checked, then				

System 👻	Call Routing 👻	Media Resources 👻	Ad	vanced Features 👻	Device - Applic
Find and	AAR Group				
	Dial Rules				
	Route Filter	3		Delete Selected	
	Route/Hunt				
Status -	SIP Route F	Pattern			
(i) 35 re	Intercom				
	Class of Co	atrol			
Director	Client Matte	ur Codes			
Find Direct	Earned Aut	horization Codes		It is a singe with	*
Find Direc	Forced Aut	Dettere		 Degins with 	•
	Translation	Pattern		tern/Directory Nu	mber [*]
	Call Park				
	Directed Ca	all Park		0.4	
	Call Pickup	Group		2-1	
	Directory N	lumber			
	Meet-Me N	umber/Pattern		Selec	ct a setting of "Call Routing/
0	Dial Plan In	staller		Direc	
	Route Plan	Report			
	Transforma	ition	•		
	Mobility		×		
	Logic al Par	tition Policy Configuration	1		
	Call Control	Discovery	۲		
	External Ca	Il Control Profile			
System 👻 Cal	Routing 👻 Media	Resources 👻 Advanced F	eatur	es 🔻 Device 👻 App	lication
Find and List	Directory Numb	ers			
Add New					
Directory N	umber				
Find Directory	Number where D	irectory Number 🔻 beg	ins w	vith 🔻	Find Clear Filter 🖶 👄
				No act	ive query. Please enter your search criteria using the options above.
Add New	2-2	Add a new Dir	ecto	orv number	
			_		
Directory Nun	nber Configurati	on			
Save	2-4				
	27				
Status					
(i) Status: Re	eady				
Directory Nur	nber Informatio	n	_		
Directory Num	ber* 3999				to
Route Partition	PTN_Taka	_EXT		•	
Description	3999				2-3
Alerting Name	3999				
ASCII Alerting	Name 3999				_
🗹 Active					
Directory Nur	nber Settings —				
Voice Mail Prof	ïle	< None >			▼ (Choose <none> to use system default)</none>
Calling Search	Space	< None >			¥
Presence Grou	p* IAudio Course	Standard Presence gro	цр		Full fill boxes as shown in left, Directory number
Network Hold	Audio Source	< None >			can be a range of SIP Client numbers. N-8000 SIP
	.e.r Addio Source	< None >			"3995" to "3999."

Step 2. Adding extension number

Step 3. Adding SIP Client

System 👻 Call Routing 👻 Media Resources 👻 Advanced Features 👻	Device 🔻	Application 👻 User Manager	nent 👻 Bulk Administration 👻 Help 👻		
Find and List Phones	CTIR	oute Point			
Add New		veper vay	3-1		
Phone	Phone	Ð		_	
Find Phone where Device Name		te Destination	Select "Device/Phone"		

System - Call Routing - Media Resources - Advance	ed Features 👻 D	evice 👻 Application 👻	User Management 👻 Bulk Administration 👻 Help 👻
Find and List Phones			
Add New			
Phone			
Find Phone where Device Name	begins with 🔻		Find Clear Filter 🕂 📼
3-2		Select item or enter se	arch text 🔻
Add a new phon		No active query, Pl	ease enter your search criteria using the options above.
Add New			

Add a New Phone		
Next		
Status Status: Ready	3-3	
Create a phone using the phone type or a phone template Phone Type* Third-party SIP Device (Basic)	•	
BAT Phone Template* Not Selected	▼	
- Next	Select the phone type that is created in a creating step, then go to the next	a profile
3-4		

Phone Configuration		
Save		
- Status		
(1) Status: Ready		
Diana Tana		
Phone Type	(p i-)	
Device Protocol: SIP	e (Basic)	
Device Information		
Device is not trusted		
MAC Address*	000C29CB913B	
Description	SEP000C29CB913B	
Device Pool*	Default	View Details 2_5
Common Device Configuration	< None >	View Details
Phone Button Template*	Third-party SIP Device (Basic)	*
Common Phone Profile*	Standard Common Phone Profile	T
Calling Search Space	CSS_Taka_EXT	v
AAR Calling Search Space	< None >	T
Media Resource Group List	< None >	T
Location*	Hub_None	¥
AAR Group	< None >	T
Device Mobility Mode*	Default	View Current Device Mobility Settings
Owner User ID	< None >	T
Use Trusted Relay Point*	Default	¥
Always Use Prime Line*	Default	¥
Always Use Prime Line for Voice Message*	Default	T
Calling Party Transformation CSS	< None >	T
Geolocation	< None >	T
Suse Device Pool Calling Party Transform	nation CSS	Full fill boxes as shown in left
□ Ignore Presentation Indicators (internal		
✓ Logged Into Hunt Group		
Remote Device		

-Protocol Specific Information-		
Presence Group*	Standard Presence group	T
MTP Preferred Originating Codec*	711ulaw	T
Device Security Profile*	Third-party SIP Device Basic - Digest Required	•
Rerouting Calling Search Space	< None >	
SUBSCRIBE Calling Search Space	< None >	- 3-6
SIP Profile*	Standard SIP Profile	•
Digest User	3999	•
Media Termination Point Requir	ed	
Unattended Port	3-7	
Require DTMF Reception		
MLPP Information		
MLPP Domain < None >	T	
Save 3-8	Full fill boxes as User" can be a cl 3995, 3996, or 39	shown in left and save it. "Digest ient that is set by this step, like 99, e.g

Find ar	nd List	Phones						
	ld New	Select All Clear All	🗙 Delete Selected	set Selected 🖉 Apply Co	nfig to Selected			
Status								
(i) ²	9 recor	ds found						
Phon	e (1	- 29 of 29)						
Find Ph	one wh	nere Device Name	▼ begins with ▼		Find Clear Filter	4 -		
			S	elect item or enter search	n text 🔻			
		Device Name(Line) [▲]	Description	Device Pool	Device Protocol	S		
	8	SEP0008E101F47F	SEP0008E101F47F	Default	SIP	Unknown		
	P	SEP0008E1028CCF	SEP0008E1028CCF	Default	SIP	Unknown		
	1	SEP000C29CB913B	SEP000C29CB913B	Default	SIP 3_9	Unknown		
	6911	SEP04DAD2BFA6FD	SEP04DAD2BFA6FD	DP Taka G729	SIP	Registered with 192		
	m	SEP100C29CB913B	SEP100C29CB913B	Like shown in lef	Like shown in left, an added device is on the list			
	6	SEP110C29CB913B	SEP110C29CB913B	Click the added device				

Phone Configuration		
Save 🗙 Delete 🗋 Copy 🧣	🗍 Reset 🥒 Apply Config	Add New
Status Status: Ready		
Association Information Modify Button Items	Phone Type Product Type: Thi Device Protocol: SIP	rd-party SIP Device (Basic)
3-10	Device Information – Registration IP Address	The left information can be seen. Click the red area

Directory Number Co	onfigurati	on			
Save 3-1	3				
Status					
i Status: Ready					
Directory Number In	formatio	n			
Directory Number*	3999				
Route Partition	PTN Taka EXT				
Description	3999				0.44
Alerting Name	3999				3-11
ASCII Alerting Name	3999				
✓ Active				J	
Directory Number Se	ettings —				
Voice Mail Profile		< None >	•	(Choo	se <none> to use system default)</none>
Calling Search Space	ling Search Space CSS_Taka_EXT		•	J 3	-12
Presence Group*		Standard Presence group	•]	
User Hold MOH Audio S	Source	< None >	•]	
Network Hold MOH Aud	dio Source	< None >	*)	Full fill boxes as shown in left

1

	Voice Mail	Destination		Calling Search Space
Calling Search Space Activation Policy			Use S	ystem Default
Forward All	or 🔲		< No.	e >
Secondary Calling Search Space for Fo	rward All		< No.	ie >
Forward Busy Internal	or 🔲	3996	CSS_	Taka_EXT
Forward Busy External	🔲 or	3996	CSS_	Taka_EXT
Forward No Answer Internal	🔲 or	3996	CSS	Taka_EXT
Forward No Answer External	or	3996	CSS_	Taka_EXT
Forward No Coverage Internal	🔲 or		< No.	e >
Forward No Coverage External	🔲 or		< No	ie >
Forward on CTI Failure	🔲 or		< No	ie >
Forward Unregistered Internal	or	3996	CSS_	Taka_EXT
Forward Unregistered External	🔲 or	3996	CSS_	Taka_EXT
Io Answer Ring Duration (seconds)				
Call Pickup Group < N	one >	¥		

*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.

of others can cover a call instead of a busy station.

Multiple Call/Call Waiting Settings on Device SEP	2000C29CB913B	
Note:The range to select the Max Number of calls is: 1	1-2	
	2	
Busy Trigger*	1	(Less than or equal to Max. Calls)
	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

System 👻 Call Routing 👻 Media	Resources 👻 Advanced Features 👻 Device 👻	Application 👻	User Management 👻	Bulk Administration	Help
Find and List Users			Credential Policy D)efault	
			Credential Policy		1 1
Add New			Application User		4-1
User			End User		
Find User where First name	Leasing with	Find Clear	Role		
This oser where Trischame		Tind Clear	User Group		
		No active query.	User/Phone Add	ADE Drafile	ons above.
Add New	Select "User Managem	ont"/	End Liser CAPE D	APP PIONE	
	"End user"		SIP Realm		
					1
System - Call Routing - Med	dia Resources - Advanced Features - Devic	e - Application	 User Management 	t 👻 Bulk Administra	ation - Help -
Find and List Users					
Add New					
User				_	
Find User where First name	▼ begins with ▼	Find Cle	ar Filter 🔂 😑		
		No active quer	y. Please enter your se	arch criteria using th	e options above.
Add New 4-2	Add a new user				
System - Call Routing - Me	dia Resources Advanced Features Dev	vice - Applicatio	n 👻 User Managen	nent 👻 Bulk Admir	istration - Help -
End User Configuration		_			
Save 4_4		Ma	ke a setting of	a new user, thi	s example is for
		a S Th	SIP Client with a e digest creden	number '3999 tials shall be s	.' et in this setting
Status		wir	ndow.		et in this setting
i Status: Ready					
		Ple	ease save it unti	I these setting	S.
User Information			_		
User ID**	3999				
Password					
Confirm Password					
PIN					
Confirm PIN					
Last name "U	Jser				
Middle name			_		
First name	999			•	
Mail ID	1999		4-3	5	
Manager User ID					
Department					
User Locale	notice units a new second				
Associated PC	English, United States				
Digest Credentials			Dasswor	rd "n@ssw0r	d"
Confirm Digest Credentials					d"
•			passwor	u pesswur	u

*Until this setting, pressing save is better to continue.

- Device Information	
Controlled Devices	SED000C20CB012B
controlled Devices	
	Device Association
	▼
Available Profiles	EM_2000 4-5
	EM_2001
	•
CTI Controlled Device Profiles	
	★
	· · · · · · · · · · · · · · · · · · ·
Extension Mobility	
Available Profiles	EM_2000
	EM_2001
	Select as shown in left.
	and make a device
Controlled Profiles	assotiation
Conditioned Promes	
	i i i i i i i i i i i i i i i i i i i
	· · · · · · · · · · · · · · · · · · ·
Default Profile	Not Selected 🔻
Presence Group*	Standard Presence group
SUBSCRIBE Calling Search Sp	ace < None >
Allow Control of Device from	m CTI
Enable Extension Mobility C	ross Cluster
Directory Number Associati	0.05
Primary Extension < None >	
Mobility Information	
Enable Mobility	
Primary User Device	< None > T
Enable Mobile Voice Access	
Maximum Wait Time for Desk	Pickup [*] 10000
Remote Destination Limit*	4
Remote Destination Profiles	*
	View Details
CAPF Information	
Associated CAPF Profiles	
	View Details
Permissions Information —	
Groups Standard CCM End Use	ers 4-6
	Add to User Group
	Remove from User Group
Poles Standard COM Fail II	<u>View Details</u>
Standard CCM End Use Standard CCMUSER Ad	dministration Add to the group shown in left and
	save it.
4-/	View Details
- Save Delete Add New	-