

UNIVERGE SV9300

System Manual

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INTRODUCTION

PURPOSE

This manual explains the system description and the programming procedure for the UNIVERGE SV9300.

NOTE: *As for the parts described as [9300V3 STEP2 software required] in this manual support 9300V3 STEP2 (SC-4351 LYRA BSC PROG-V3.2.0) software or later.*

OUTLINE OF THIS MANUAL

This manual consists of two chapters and three appendixes. The following paragraphs summarize Chapters 1 through 2 and Appendixes A, B, C and D.

CHAPTER 1 GENERAL INFORMATION

This chapter explains the system outline, system conditions and services for the system.

NOTE: *As for the equipment name and function of chassis and circuit blades and system capacity, refer to the System Hardware Manual.*

CHAPTER 2 SYSTEM DATA PROGRAMMING

This chapter explains the programming procedure of the system data, IP Station setup procedure and IP Station login/logout procedure.

APPENDIX A SYSTEM SPECIFICATIONS

This appendix explains the specifications of the system.

APPENDIX B VOICE QUALITY MEASURES IN VoIPDB

This appendix describes how to reduce echoes in SV9300 and the setup method.

APPENDIX C TCP/UDP/RTP PORT NUMBER CONDITIONS

This appendix explains the default TCP/UDP/RTP port number and range of the available TCP/UDP/RTP port numbers used in SV9300.

APPENDIX D DT700/DT800 SERIES CONFIGURATION MENU LIST

This appendix explains the configuration menu used for DT700/DT800 Series configuration settings.

NOTE: *As for the configuration menu used for DT900 Series configuration settings, refer to the DT900 Setup Guide.*

TERMS IN THIS MANUAL

PBX System Designation

PBX system is designated as “PBX” or “system” usually.

When we must draw a clear line between the PBX systems, they are designated as follows.

SV9300 : UNIVERGE SV9300

SV8300 : UNIVERGE SV8300

SV9500 : UNIVERGE SV9500

SV8500 : UNIVERGE SV8500

SV8100 : UNIVERGE SV8100

SV7000 : UNIVERGE SV7000

2400 IPX: NEAX 2400 IPX Internet Protocol eXchange

Attendant Console Name

Attendant Console is usually designated as “Attendant Console”.

When the console type is limited by a service, it is designated as follows:

DESKCON: Desk Console (SN716 DESKCON)

Paging Adapter Name

With the release of the new Paging Adapter model, the current model “PGD(2)-U10 ADP” is integrated to the new model “IP8WW-2PGDAD-A”. Therefore, as for the term of “PGD(2)-U10 ADP” or “PGD(2)-U10” described in this manual, please read them as “IP8WW-2PGDAD-A” or “Paging Adapter”.

Terminal Name

The term of “DESI-less terminal” described in this manual is the same meaning as the “Self-Labeling terminal”.As for the following terminal names, please read “DESI-less” as “Self-labeling”.

DT700/DT800/DT900 Series DESI-less: DT700/DT800/DT900 Series Self-Labeling

DT820 DESI-less: DT820 Self-Labeling

DT710 DESI-less: DT710 Self-Labeling

In this manual, the following terminals are usually designated as each common term unless other type of terminal is specified.

COMMON TERMS		TERMINAL NAMES				
Multiline Terminal	Digital Multiline Terminal	D ^{term}		D ^{term} 85 (Series i)		
		DT500 Series		DT510 DT530		
		DT400 Series		DT410 DT430		
		DT300 Series		DT310 DT330		
		IP Station	IP Multiline Terminal	IP Enabled Digital Multiline Terminal		D ^{term} 85 (Series i) (IP Adapter Type)
				D ^{term} IP INASET		
				D ^{term} IP		D ^{term} 85 (Series i) (IP Bundled Type)
	DT900 Series			DT920 Series	DT920 DT920 Self-Labeling	
				DT930 Series	DT930 DT930 Touch Panel	
	DT800 Series			DT820/820C DT830/DT830CG/DT830DG		
	DT700 Series			DT710 DT730/DT730CG/DT730DG DT750 DT770G		
	Soft Phone		D ^{term} SP30 SP350			
	SIP Wireless Terminal		MH240			
	IP Single Line Telephone (SIP)	Standard SIP Terminal	GT890 (ST500 for GT890 is installed.)			
ST500 (for iOS or for Android)						
Third-party SIP Terminal						

NOTE: *DT770G (Cradle Phone) is normally used together with a smart device. The operation and the condition depend on the specifications of application on smart device.*

REFERENCE MANUAL

During installation, also refer to the manuals below:

System Hardware Manual:

Contains the installation procedure for the PBX system and the hardware installation procedure for the SV9300.

Command Manual:

Contains the Customer Administration Terminal (CAT) operation, command functions and data required for programming the system and the Resident System Program.

System Data Programming Manual:

Contains the Customer Specifications Sheets and the System Data Programming Sheets.

Programming Manual:

Contains procedure for programming each business, hotel, ISDN and OAI feature.

System Maintenance Manual:

Contains the system maintenance services and the recommended troubleshooting procedure.

Networking Manual:

Contains the system description and the programming procedure for the CCIS, Q-SIG and Remote Unit System.

SMDR/MCI/PMS Interface Specifications:

Contains the interface specifications for Station Message Detail Recording (SMDR), Message Center Interface (MCI), Property Management System (PMS) and the PMS operation.

PC Programming Manual:

Contains the functional description and the installation procedure for the PCPro.

DT900 Setup Guide:

Contains the setup procedure for the DT900 Series using terminal configuration menu list such as network settings, PC port settings etc.

DT500/DT900 Series Multiline Terminal Installation Manual

Contains the hardware installation procedure for the DT500/DT900 Series including cable connections, optional adapter and wall mount etc.

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GENERAL INFORMATION

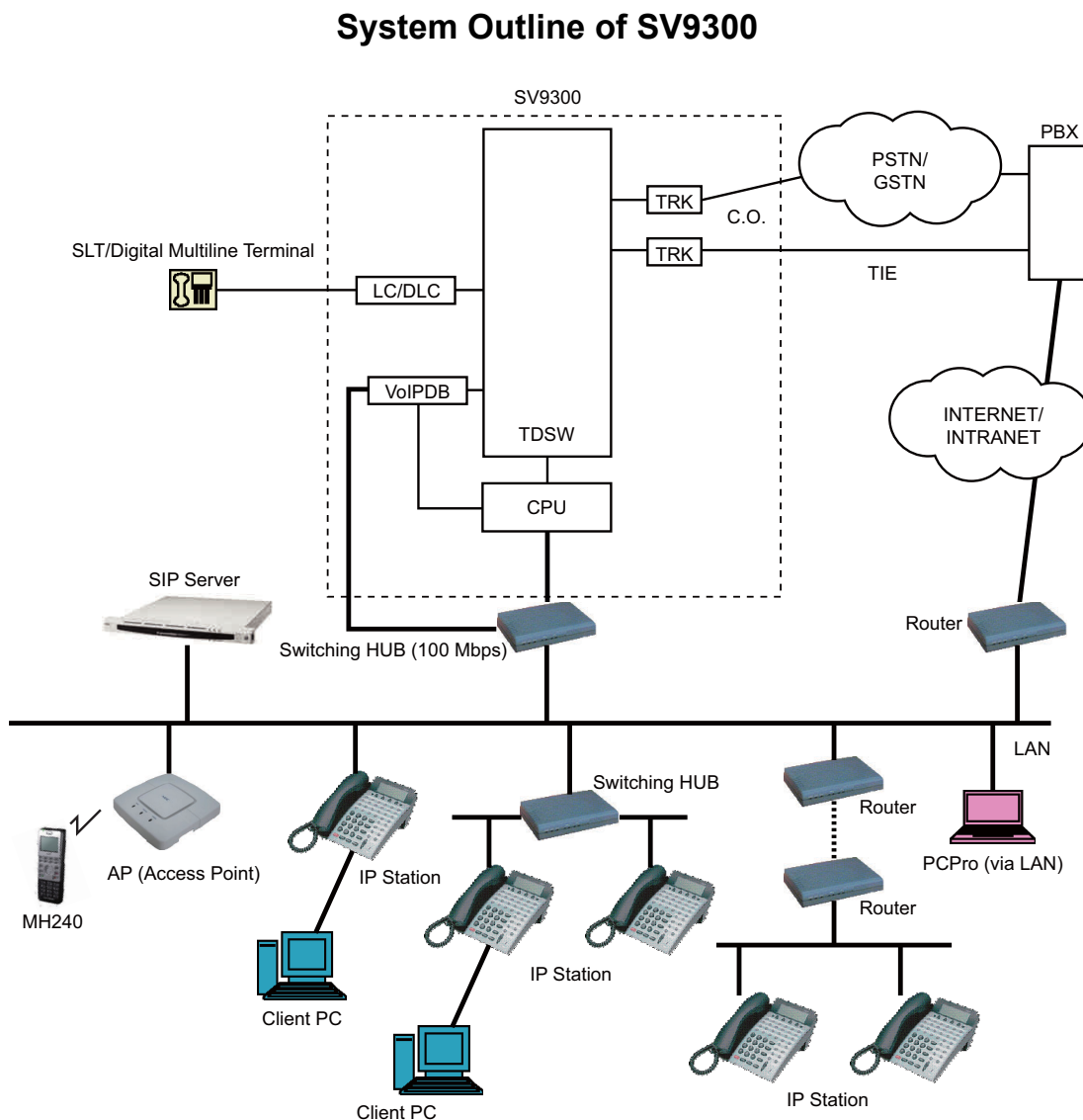
This chapter explains the system outline, system conditions and services for the system.

SYSTEM OUTLINE

SV9300 is an IP communication system that integrates voice terminals through Peer-to-Peer connection to the IP network. The system is a hybrid system to accommodate both IP Stations and the Legacy PBX's terminals (Legacy terminal). Line/Trunk blades and Application blades can be mounted in the system to provide the Legacy PBX features that use the Time Division Switch (TDSW).

NOTE: *As for the Dual CPU system, refer to the "Chapter 12 Installing the Dual CPU System" in the System Hardware Manual. As for the Failover system, refer to the "Chapter 13 Installing the Failover System" in the System Hardware Manual.*

The illustration below shows the typical system outline.



STATION-TO-STATION CONNECTION

Station-to-Station connection is available on the LAN.

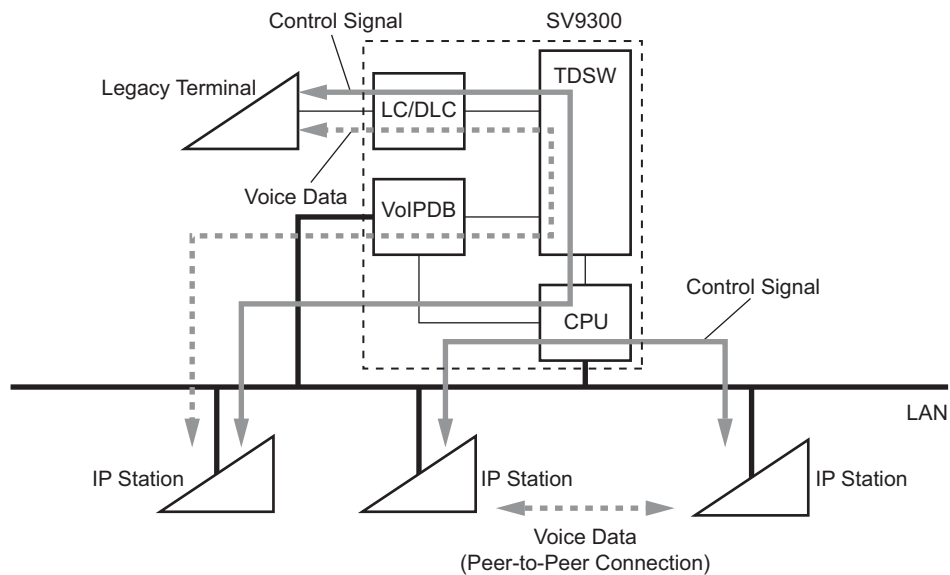
For IP Station-to-IP Station connection (Peer-to-Peer connection), the voice data is transmitted and received directly between IP Stations on the LAN.

For IP Station-to-Legacy terminal connection, the VoIPDB is required to transmit and receive the voice data. This daughter board is used to control and convert the voice data.

The CPU blade manages control signals in both types of connections.

The illustration below shows a system outline of Station-to-Station Connection.

System Outline of Station-to-Station Connection



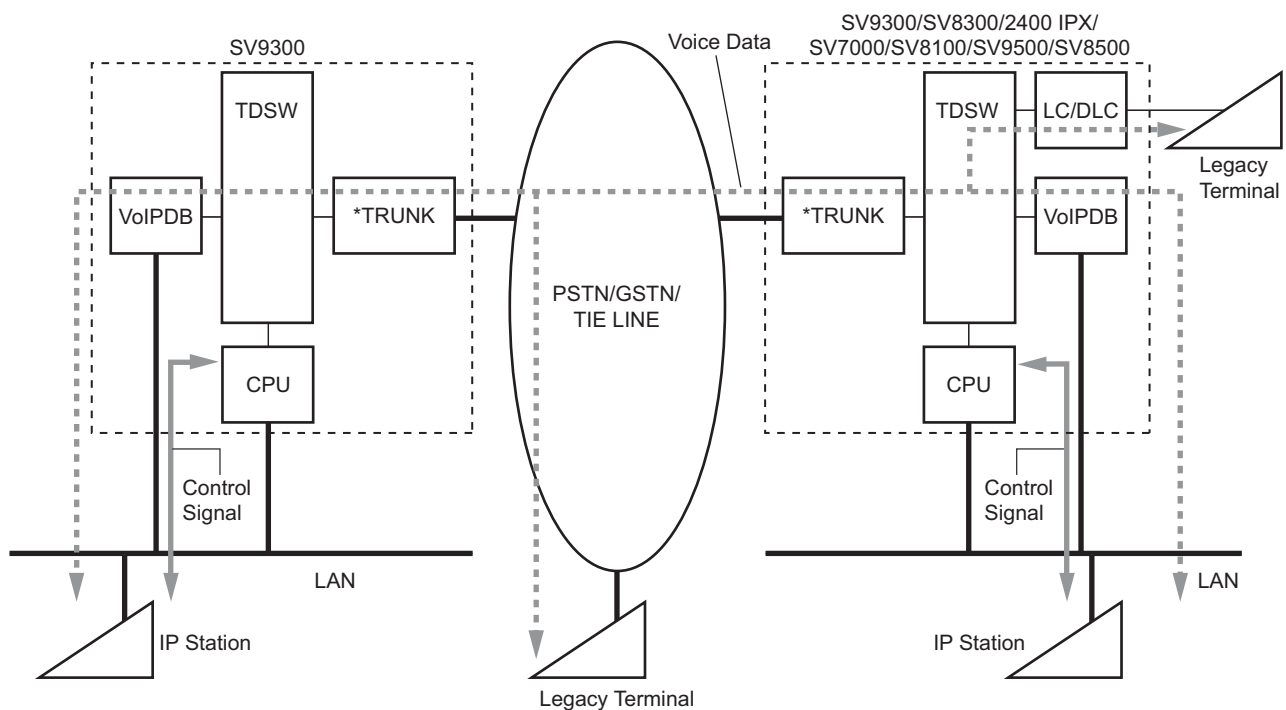
PUBLIC NETWORK/TIE LINE NETWORK CONNECTION

The system can be connected with a Public Network or Tie Line Network.

When the IP Station communicates with the IP Station/Legacy terminal in the destination office via Public Network or Tie Line Network, the VoIPDB and the trunk blade are required to transmit and receive the voice data.

The illustration below shows a system outline of Public Network/Tie Line Network Connection.

System Outline of Public Network/Tie Line Network Connection



INTER-OFFICE CONNECTION

By connecting to IP network with using IP trunk (P2P CCIS using VoIPDB) or SIP trunk, the office can be connected to another office. The communication protocol between offices is No. 7 CCIS or Session Initiating Protocol (SIP). With IP network by No. 7 CCIS, service functions between offices can be used. With IP network by SIP, terminals and devices conformed to SIP can be connected to the network.

Connections between offices via IP network are as following:

- CCIS connection (via IP trunk [P2P CCIS]) → IP stations can be peer-to-peer connected.
- SIP connection → IP stations cannot be peer-to-peer connected.

CCIS CONNECTION

The system can be connected with the IP network by No. 7 Common Channel Inter-office Signaling (CCIS) via the IPT (P2P CCIS), when the destination office is SV9300/SV8300/2400 IPX/SV7000/SV8100/SV9500/SV8500.

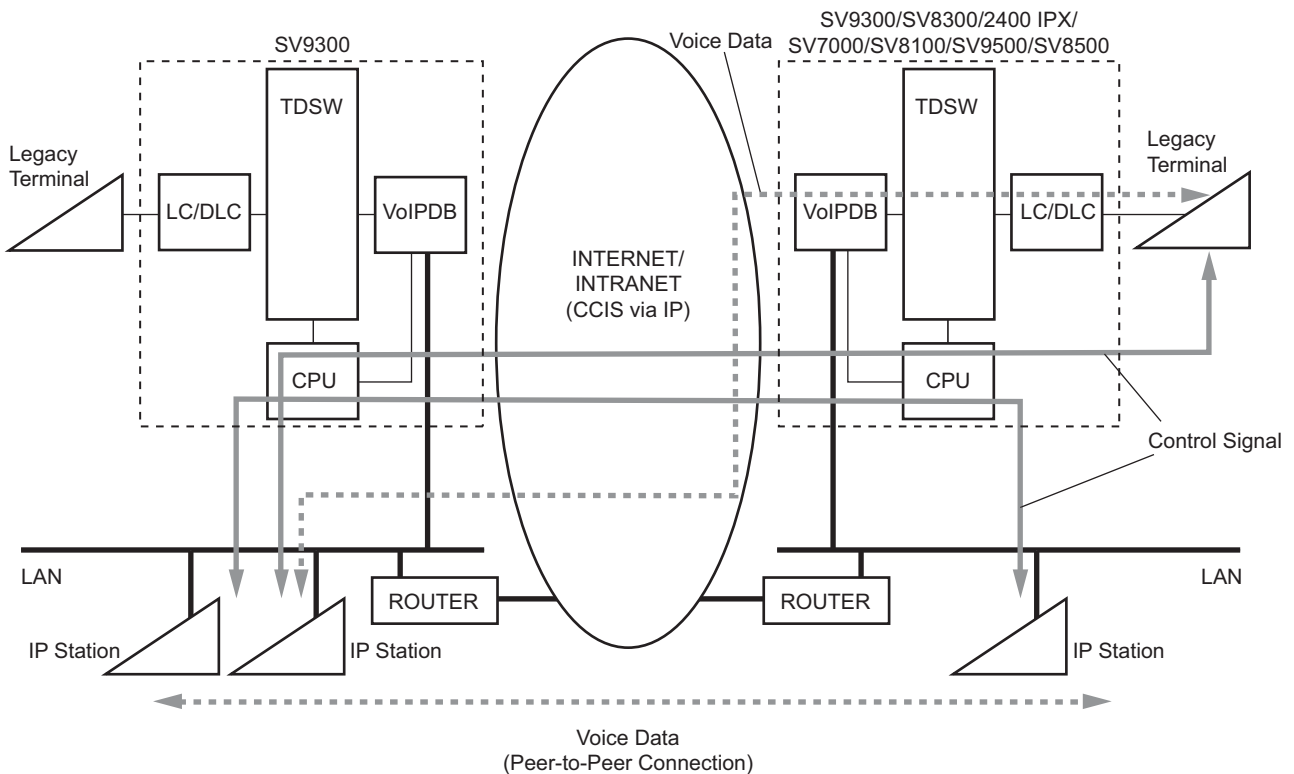
For IP Station-to-IP Station connection via CCIS (Peer-to-Peer connection), the voice data is transmitted and received directly between IP Stations via the IP network (CCIS via IP).

For IP Station-to-Legacy terminal connection via CCIS, VoIPDB is required to transmit and receive the voice data. This daughter board is used to control and convert the voice data.

The CPU blade has a built-in IPT (P2P CCIS) and the IPT (P2P CCIS) manages control signals in both types of connections.

The illustration below shows a system outline of CCIS Connection.

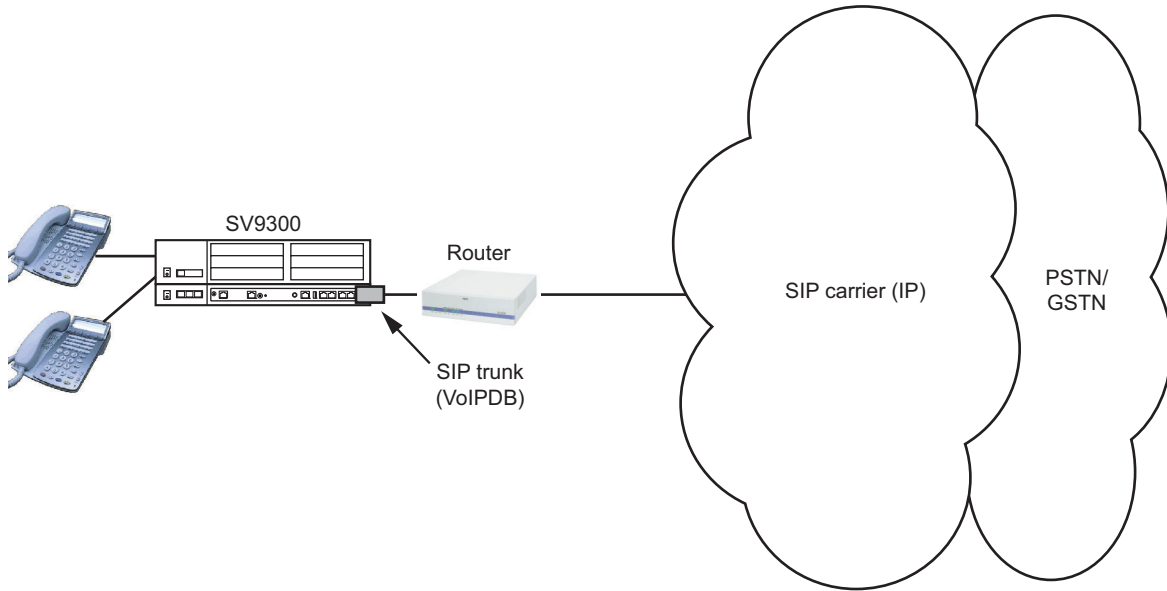
System Outline of CCIS Connection



SIP CONNECTION

The system provides the IP phone service by SIP carrier via SIP trunk on SV9300.

System Outline of SIP Connection

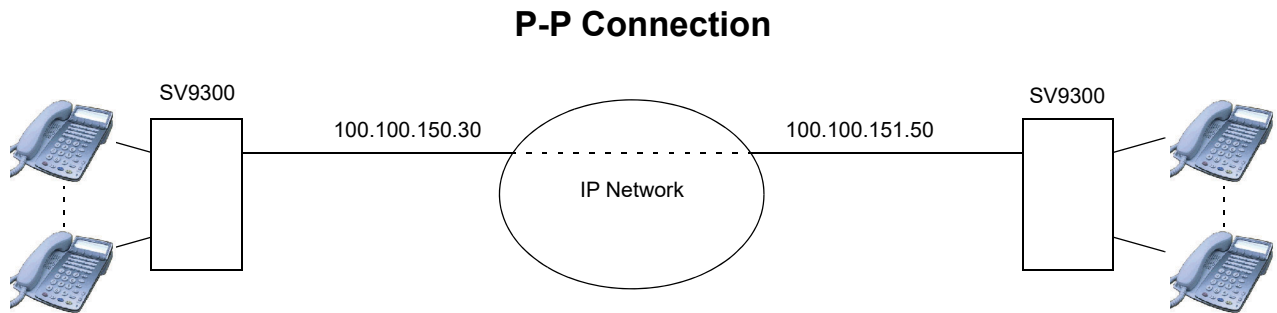


P-P/P-MP CONNECTION

In the IP network by No. 7 CCIS or SIP, there are 2 ways of connection topologies between offices using IP trunk or SIP trunk: P-P (Point-to-Point) connection and P-MP (Point-to-Multipoint) connection.

P-P Connection

This connection topology specifies one destination (IP address) per SIP trunk and connects to one office.

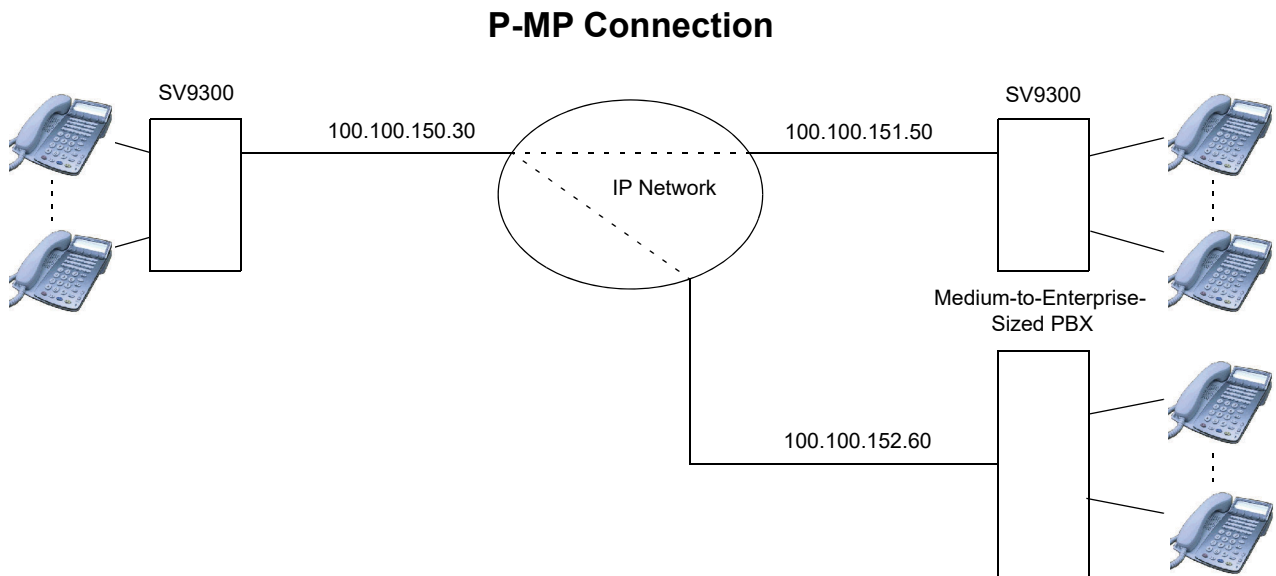


P-MP Connection

This connection topology specifies multiple destinations (IP addresses) per IP trunk or SIP trunk and connects to multiple offices.

In the case of IP trunk, up to 255 destinations (IP addresses) can be specified.

In the case of SIP trunk, up to 63 destinations (IP addresses: include own address) can be specified.



SYSTEM CONDITIONS

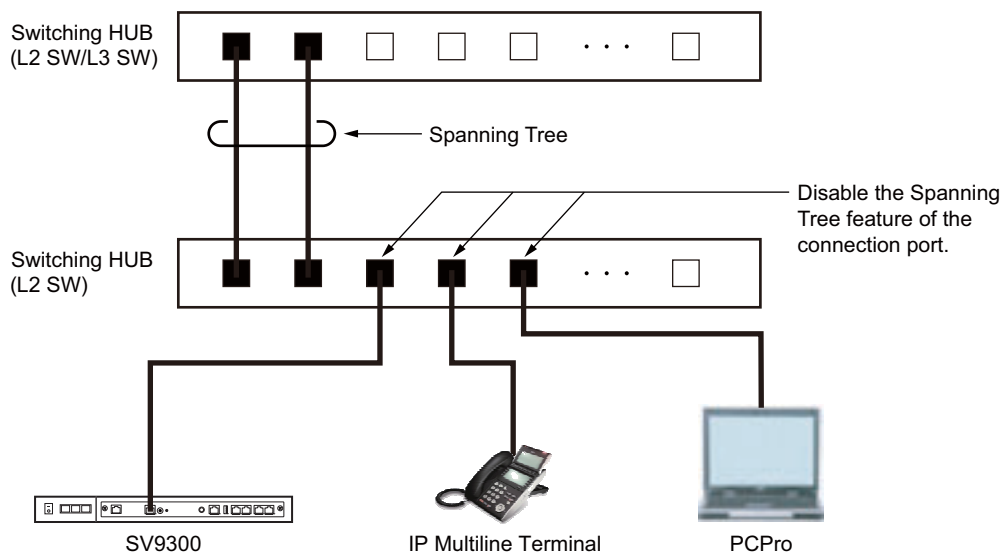
Conditions for Overall System

- The DTMF sender signal width (Inband) of Multiline Terminal is 112-128 ms.
- When the CPU blade/VoIPDB/IP Station is connected to the port of switching HUB that Spanning Tree (IEEE 802.1d) is enabled, communication failures shown below may occur. Disable the Spanning Tree feature to the connection port on the switching HUB.
 - IP Station fails to connect to SV9300
 - IP Station cannot communicate with the IP Station.
 - IP Station cannot communicate with the SLT/Digital Multiline Terminal.
 - Remote Unit cannot change over to the normal mode in the Remote UNIT over IP system.

Also, disable the Spanning Tree feature as for the following connections.

- Connections between Units
- Remote Unit/Dual CPU system/Failover
- IP trunk/SIP trunk
- Application connections such as PCPro/SMDR

Connection Image



- When connecting the CPU blade/VoIPDB/IP Station and the port of switching HUB which LACP (Link Aggregation Control Protocol: IEEE 802.3ad) function is available, communication failures shown below may occur. Set up the LACP function invalid by the switching HUB.
 - Remote Unit cannot change over to the normal mode in the Remote UNIT over IP system.

- When ETHERNET port, VOIP port of CPU blade and IP terminals are connected to a switching HUB, match the communication speed of these ports and terminals to that of connection port of the switching HUB by the following settings. For the setting of communication speed, it is recommended to use Auto Negotiation or 100Mbps (Full-Duplex).

- ETHERNET port side : Set by CM0B Y=0XX>09 (default: Auto Negotiation)
- VOIP port side : Set by CM0B Y=1XX>09 (default: Auto Negotiation)
- IP terminal side : Set from the configuration menu of each terminal

In addition, confirm that the connection is linked up by the set communication speed by such as LED for each connection port of CPU blade and a switching HUB. If any of the following cases applies, a voice interruption and a reboot of the IP terminal may occur by collisions.

- When linked up by 100 Mbps (Half-Duplex) or 10 Mbps (Half-Duplex)
- When linked up by the communication speed differed from CPU blade and a switching HUB
- For IP-PAD/IPT (P2P CCIS)/SIP trunk, the VoIPDB is required.
- There are the following conditions when setting the data by CM0B.
 - Set the system data by CM0B Y=1XX/2XX when mounting the VoIPDB in SV9300 and using VOIP port. Y=1XX is associated with setting the system data for Control Signals, and Y=2XX is associated with setting the system data for Voice Packets (RTP).
 - Set the system data by CM0B Y=0XX when not mounting the VoIPDB in SV9300 and using an Ethernet port.
 - When setting the IP address for Maintenance Port (Y=0XX) and VOIP Port (Y=1XX), set the different segment for each using port respectively.
 - Only one Default Gateway Address can be set for each unit.
 - Set the Default Gateway Address to the Maintenance port (Y=0XX) when not using VoIPDB.
 - Set the Default Gateway Address to the VOIP port (Y=1XX) when using VoIPDB.
- Jitter buffer of GPZ-64IPLD/GPZ-128IPLD (VoIPDB) is controlled as 200 ms. on SV9300, even if the minimum value of jitter buffer is set to more than 210 ms. (set by CM67 Y=04/CM67 Y=21/22/23/24>08).
- For the Dual CPU system, ACT CPU uses various kinds of IP addresses (STBY CPU uses them when switching from ACT CPU to STBY CPU).
- When resetting VoIPDB, at the same time of the reset operation, “FK=125: VoIPDB notification” is registered as fault according to the system data (CMEA Y=2>125).

- The number of VoIPDB channels used by the connection configurations is as follows.
 - To connect within a same Unit:

		Unit A (A=01-50)					
		TDM Terminal	D ^{term} IP/ D ^{term} SP30	DT700/ DT800/ DT900 Series/ SP350	Standard SIP Terminal NOTE 4	SIP Trunk	IPT (P2P CCIS) NOTE 1
Unit A (A=01-50)	TDM Terminal	None	1 channel	1 channel	1 channel	1 channel	1 channel
	D ^{term} IP/ D ^{term} SP30	1 channel	None	None NOTE 2	2 channels	2 channels	None
	DT700/ DT800/ DT900 Series/ SP350	1 channel	2 channels NOTE 2	None	2 channels	2 channels	None NOTE 2
	Standard SIP Terminal NOTE 4	1 channel	2 channels	2 channels	2 channels NOTE 3	2 channels	2 channels
	SIP Trunk	1 channel	2 channels	2 channels	2 channels	2 channels	2 channels
	IPT (P2P CCIS) NOTE 1	1 channel	None	None NOTE 2	2 channels	2 channels	None

(b) To connect between an own Unit and other Unit:

UA: Unit A UB: Unit B

		Unit A (A=01-50)					
		TDM Terminal	D ^{term} IP/ D ^{term} SP30	DT700/ DT800/ DT900 Series/ SP350	Standard SIP Terminal NOTE 4	SIP Trunk	IPT (P2P CCIS) NOTE 1
Unit B (B=02-50)	TDM Terminal	UA: 1 channel UB: 1 channel	UA: None UB: 1 channel	UA: None UB: 1 channel	UA: 2 channels UB: 1 channel	UA: 2 channels UB: 1 channel	UA: None UB: 1 channel
	D ^{term} IP/ D ^{term} SP30	UA: 1 channel UB: None	UA: None UB: None	UA: None UB: None NOTE 2	UA: 2 channels UB: None	UA: 2 channels UB: None	UA: None UB: None
	DT700/ DT800/ DT900 Series/ SP350	UA: 1 channel UB: None	UA: None UB: None NOTE 2	UA: None UB: None	UA: 2 channels UB: None	UA: 2 channels UB: None	UA: None UB: None NOTE 2
	Standard SIP Terminal NOTE 4	UA: 2 channels UB: 1 channel	UA: 2 channels UB: None	UA: 2 channels UB: None	UA: 2 channels UB: 2 channels NOTE 3	UA: 2 channels UB: 2 channels	UA: None UB: 2 channels
	SIP Trunk	UA: 1 channel UB: 2 channels	UA: None UB: 2 channels	UA: None UB: 2 channels	UA: 2 channels UB: 2 channels	UA: 2 channels UB: 2 channels	UA: None UB: 2 channels

NOTE 1: IPT (P2P CCIS) can be accommodated in the Unit01 only.

NOTE 2: When sending a DTMF signal, one VoIPDB channel is consumed per Unit.

NOTE 3: For a Peer-to-Peer connection, a VoIPDB channel is not consumed (0 channel is applied).

NOTE 4: For details of resource channels of Standard SIP Terminal, refer to the Conditions for Resource Channels When Using Standard SIP Terminal. [📄 Page 1-65](#)

Conditions for Port Number Check (for RTP Packets)

- Whether to check a port number can be specified by the following system data. The settings are reflected by the next call after changing the system data.
 - Port number check for RTP packets: CM0B Y=2XX>137

Conditions for Overall IP Station

- For the IP Station (except Soft Phone), an AC-DC adapter or inline power patch panel is required.
- The hold tone for IP Station is only “Minuet”. The hold tone set by CM48 Y=3 is not effective for IP Station.

Conditions for Overall Soft Phone

- Soft keys are not available.
- Be sure to set B mode (CM12 Y=24) for the kind of Multiline Terminal.
- Do not assign two or more Speaker keys for Line/Trunk Feature keys and Fixed keys.

Conditions for Live Recording Service

- Live Recording interacted with UM8000 is the same operation as other analog telephones. However, while the D^{term}SP30/SP350 interacted with the terminal, the operation according to the guidance is only available. The LCD indication to the D^{term}SP30/SP350 is not provided.
- For D^{term}SP30/SP350, you cannot answer during monitoring the Live Recording by UM8000 interaction.
- Because the D^{term}SP30/SP350 has no soft key, you cannot use the features such as Interruption/Restart, Re-record, Deletion during UM8000 interaction (recording is available by the feature key).

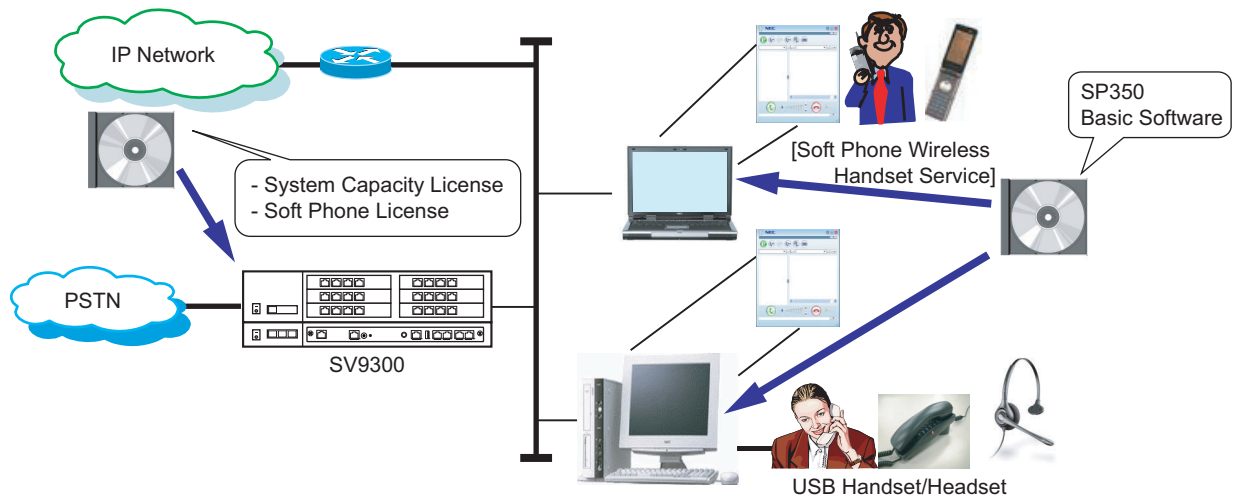
Conditions for D^{term}SP30

- When using a D^{term}SP30 with SV9300, do not perform the setting of high-tone quality CODEC from the D^{term}SP30 (the high-tone quality CODEC is set to effective in default setting).
- D^{term}SP30 displays the button data assigned by CM90 Y=00 onto the function panel as label. The displayed content can be arranged on D^{term}SP30.
- For D^{term}SP30, short tones to be heard when using the features such as Executive Right of Way (Executive Override) or Call Waiting differ from Multiline Terminal/analog stations.
 - Multiline Terminal/analog stations: 3 burst tones
 - D^{term}SP30: 1 burst tone
- When no answer is returned for the recall of Outgoing Trunk Queuing, only the time history is listed in the call history of D^{term}SP30 (records of calling numbers are not listed).

Conditions for Soft Client SP350

- By accommodating Soft Client SP350 (hereinafter “SP350”) to SV9300, the various services such as Voice Calls, Instant Message and Data Conference become possible. The illustration below shows the example of SP350 use in a system.

Example of SP350 Use in System



General Conditions

- SP350 can be accommodated to Unit01-50 as with other IP Stations.
- To use SP350, the following licenses are required:
 - System Version License
 - System Capacity License
 - Client License for Soft Phone ($D^{\text{term}}\text{SP30/SP350}$)*
 - Client License for Soft Phone ACD ($D^{\text{term}}\text{SP30/SP350}$)*
- * As for the client licenses for Soft Phone/Soft Phone ACD, the licenses are required for the number of terminals to be used.
- SP350 displays the button data assigned by CM90 Y=00 (F0XXX, F1XXX, F50XX) onto the function panel as label. The displayed content can be arranged on $D^{\text{term}}\text{SP350}$.
- SP350 can be accommodated with $D^{\text{term}}\text{SP30}$ at the same time. However, Data Conference and Instant Message services cannot be used between SP350 and $D^{\text{term}}\text{SP30}$.
- The following features are not available for SP350.
 - Reverse Contrast
 - Double-sized characters indication
- For SP350, you cannot allocate the trunk to the Line/Trunk Feature key.

- When providing Answering Sub Line call with handset, do the following setting from SP350 in addition to the system data programming (CM15 Y=082, 281).
 - Make the Answer preset ineffective by the user setting of SP350.
 - Set the Answer key to the appropriate function key (F1) - (F4) by the handset setting of SP350.
- When providing Ringing Line Pickup, do the following setting from SP350 in addition to the system data programming (CM15 Y=082, 281).
 - Make the Answer preset effective by the user setting of SP350.
- The table below shows the list of feature availability.

×: Available –: Not available

FEATURES	AVAILABILITY of SP350	REMARKS
Presence	×	For details of the availability of Presence Service use, see “Conditions for Presence Service”. Page 1-17
Data Conference (for 2-Party)	×	A connection between offices is not available with SV9500/SV8500/SV7000. For details of the availability of Data Conference use, see “Conditions for Data Conference”. Page 1-20
Data Conference (for 8-Party maximum)	×	
Desk Top sharing	–	
White Board sharing	×	A connection between offices is not available with SV9500/SV8500/SV7000.
Chat	×	
File Transfer	×	
Instant Message	×	For details of the availability of Instant Message use, see “Conditions for Instant Message”. Page 1-23

- The available CODEC types/Payload sizes for SP350 are as follows.

CODEC TYPE	PAYLOAD SIZE	REMARKS
G.711	20 ms. 40 ms.	Bandwidth: 64 Kbps
G.722	20 ms. 40 ms.	Bandwidth: 64 Kbps
G.729a	20 ms. 40 ms.	Bandwidth: 8 Kbps

- SV9300 uses the following UDP port numbers as a default when using SP350. As for the port numbers to be used for SP350 and OW5000, refer to the manual of each product.

FEATURES TO BE USED FOR SP350	UDP PORT FOR SV9300		UDP PORT FOR DESTINATION
	SOURCE	LISTEN	DESTINATION
Start up/basic call control	5080		5080 (SP350)
Presence service	5082	NOTE	111 (OW5000) NOTE
Data Conference	3456 5080		3456 (SV9300) 5080 (SP350)
Instant Message	3456 5080		3456 (SV9300) 5080 (SP350)

NOTE: *The Main Unit of SV9300 communicates with the Presence Server via UDP as a client of RPC (Remote Procedure Call). The number of UDP ports to be used by SV9300 is one port and port number 5082 is used by the default setting (the port number can be changed by CM0B Y=101>43). Besides, UDP port for the destination device also communicates with the port number specified by OW5000, in addition to the fixed UDP port (111).*

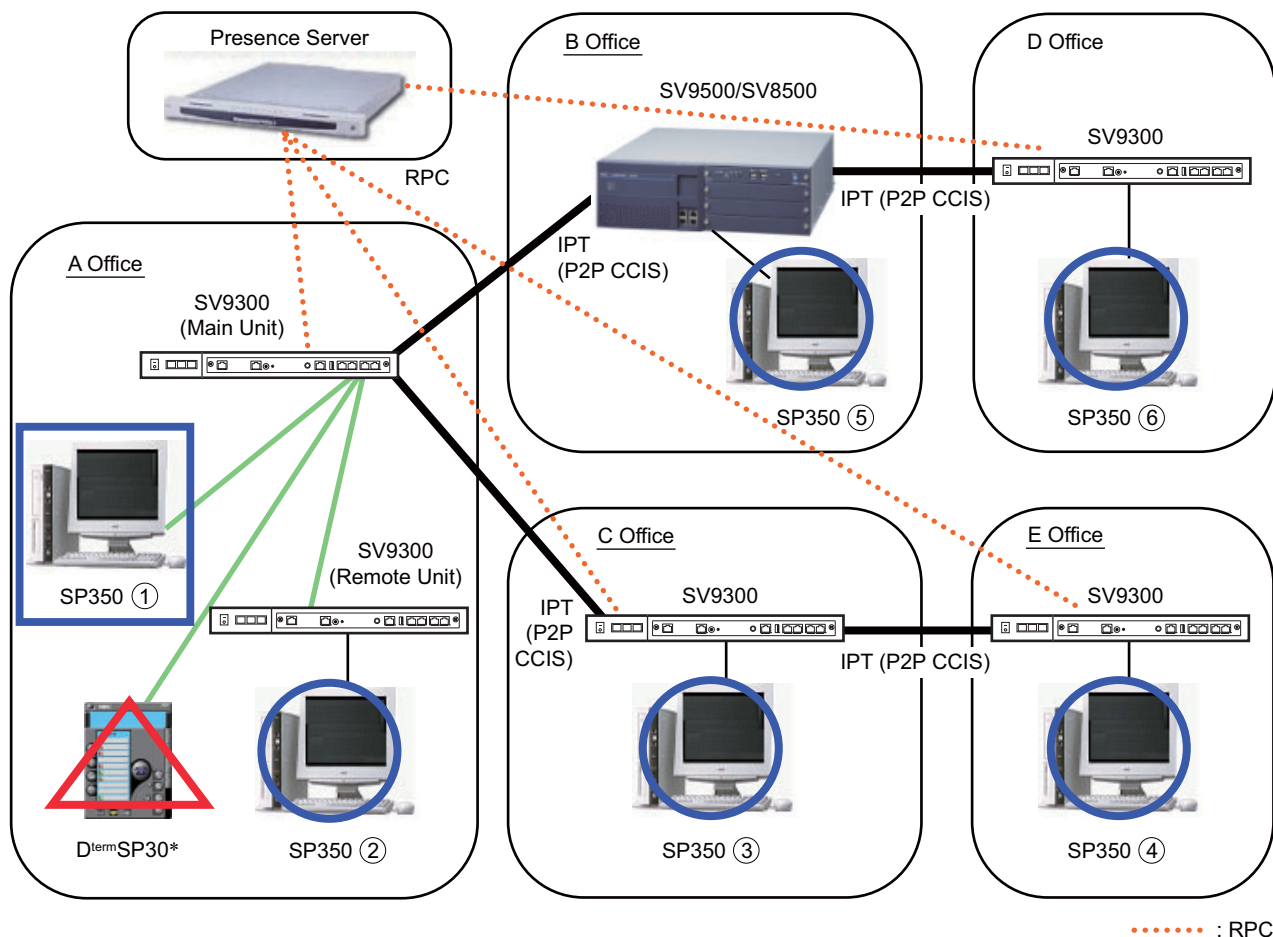
- Set the Protected Login Mode for the ID registration method for SP350. The Automatic Login Mode (Authentication by MAC address) and the Fixed Connection Mode are not available.
- By the fault information display (Fault Kind: 036 [Detect terminal connection]), you can confirm that the SP350 is started (the fault information table must be set by CMEA Y=2>036).
- When the office data (CM90 Y=0) is changed, restart SP350.
- To configure Forwarding settings in the Forwarding settings panel, use Call Forwarding - Busy Line Screen. (Call Forwarding - No Answer Screen does not support this settings.) In this case, Call Forwarding - Busy Line and Call Forwarding - No Answer are commonly set. For individually setting, set with access code.
- In the Forwarding Settings Panel of SP350, Logout Forwarding is not supported. Set with access code.

- When CAS hooking is assigned in the Fixed functions of the Telephone Function Menu, in the Action Screen (if displayed), Hooking appears in the Forwarding icon's right-clicking menu. Select Hooking to use CAS hooking. Otherwise, right-click menu does not appear.
- When no answer is made to the recall from the Outgoing Trunk Queuing (Trunk camp on), the history of SP350 has only the time of the call, not the dial No. of the caller.

Conditions for Presence Service

- This service allows more flexible communication with the co-workers according to the status of the called party whom the user is trying to reach. For example, when the called party is on the desk, the user makes a call, when the called party is busy or away from the desk, the user sends an Instant Message, or when the called party is away from the office, the user sends an E-mail. The illustration below shows the example of the Presence service use.

Example of Presence Service Use (Example of Presence Service Use for SP350 ① in A Office)



..... : RPC

* You cannot refer the two-way presence between SP350 and D^{term}SP30 (the presence of D^{term}SP30 can be referred only from SP350 side).

- To use the Presence service, a Presence Server (OW5000) is required.
- Only one Presence Server is available.
- For SP350 under Remote Unit on survival mode (because of such as a network failure), the Presence service is not available.
- When the station data (CM10 Y=01) is changed or deleted, the target terminal must be logged out in advance (the Presence status may not be updated up to 30 minutes when the station data (CM10 Y=01) is changed or deleted without log out).
- When the network failure occurs between the SV9300 and the Presence server, the Presence status may not be updated up to 30 minutes after the restoration.
- Because the Presence service is realized by the communication between each office and the Presence Server via RPC (Remote Procedure Call), any interface type (such as IPT [P2P CCIS]/ISDN) is acceptable between offices.
- For OW5000, you cannot refer the presence from the terminal other than SP350.
- The table on next page shows the availability of the Presence service between the Presence Server and each terminal.

Presence Service between Presence Server (OW5000) and Each Terminal

- ×: Reference for two-way presence is available
- : Reference for two-way presence is not available
- Δ: Terminal A can refer the presence of Terminal B **NOTE**
- ▲: Terminal B can refer the presence of Terminal A **NOTE**

NOTE: Available only when the call status of the legacy stations is obtained between the office accommodated the station that the presence is referred and OW5000.

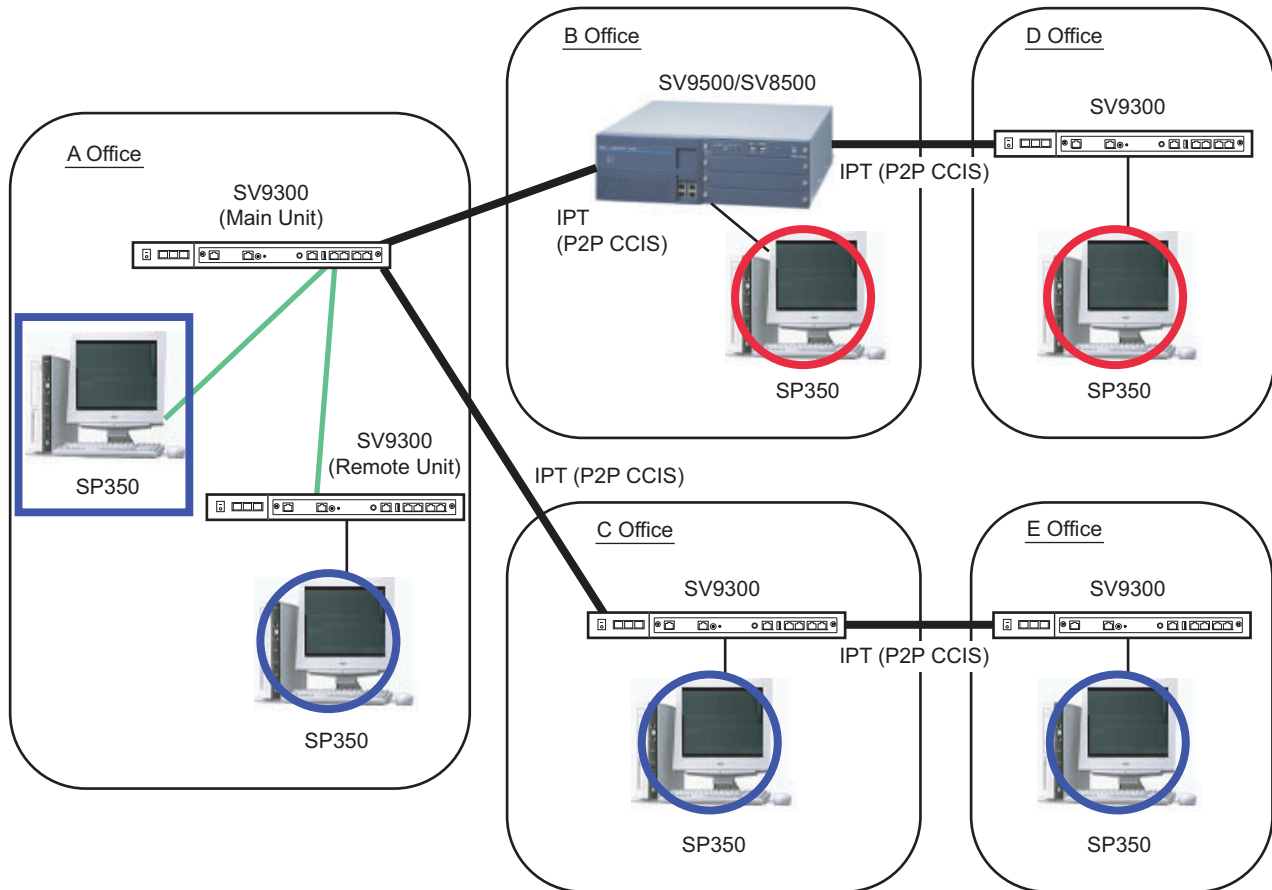
TERMINAL B \ TERMINAL A (Stations accommodated to SV9300)		SP350			D ^{term} SP30 (Only for Enhanced Protocol Mode)/ OTHER STATION		
		SP350	Δ	×	SP350	Δ	×
Own Office	SV9300 accommodation	SP350		×		▲	
		D ^{term} SP30 (only for enhanced protocol mode)/other station		Δ		–	
Other Office	SV9300/SV8300 accommodation	SP350		×		▲	
		D ^{term} SP30 (only for enhanced protocol mode)/other station		Δ		–	
	SV9500/SV8500 accommodation	SP350		×		▲	
		D ^{term} SP30 (only for enhanced protocol mode)/other station		Δ		–	
		D ^{term} SP30 (SIP mode)		×		▲	

* For OW5000, you cannot refer the presence from the terminal other than SP350.

Conditions for Data Conference

- Up to 8-party can be participated to Data Conference with SP350.
Data Conference provides the services such as Video Conference/Communication Board (Data sharing/White Board)/File Transfer/Chat without preparing any private server.
The illustration below shows the example of the Data Conference use.

Example of Data Conference Use



- Data Conference is available between Main Unit and SP350 accommodated to the Remote Unit, and between SP350s accommodated to SV9300 which is connected to IPT (P2P CCIS).
- Data Conference is not available via C.O.line (including a SIP trunk) and between SV9300 and SV9500/SV8500/SV7000 which is connected by IPT (P2P CCIS).
- For IP Bandwidth service, you cannot manage the bandwidth for data field such as Video Conference.
- Before starting Data Conference, Voice Conference (including 2-party call) must be started in advance.

- Up to 8-party can be participated to Data Conference per conference call. When the Data Conference is started with three or more parties, CFT (Conference Trunk) is used. When the Data Conference is started with four or more parties, the setting for conference is required.
- When the Data Conference is started in a condition that nine or more parties is participated to one conference call, the participants cannot be specified or selected other than the host of the conference.
- The conditions when utilizing Data Conference during a Three-Party Conference are as follows.
 - When the station that is started the Three-Party Conference and the station to be started the Data Conference are the same, each of the Three-Party can be participated to the Data Conference.
 - When the station that is started the Three-Party Conference differs from the station to be started the Data Conference, and both stations are in a same office, each of the Three-Party can be participated to the Data Conference.
 - When the station that is started the Three-Party Conference differs from the station to be started the Data Conference, and both stations are not in a same office, only the station that is started the Three-Party Conference and the station to be started the Data Conference can be participated the Data Conference. **NOTE**

NOTE: *After a Three-Party Conference is started, when the station that is started the Three-Party Conference made a Call Forwarding, the conditions for Data Conference may be changed depending on the management state of conference participants.*

- The conditions when utilizing Data Conference during a conference (Call-up type [One by One Call]) are as follows.
 - When the Voice Conference caller and the Data Conference caller are the same, all of participants can be participated to the Data Conference.
 - When the Voice Conference caller differs from the Data Conference caller, and both callers are in a same office, all of participants can be participated to the Data Conference.
 - When the Voice Conference caller differs from the Data Conference caller, and both callers are not in a same office, only the Voice Conference caller and the Data Conference caller can be participated to the Data Conference. **NOTE**

NOTE: *After Voice Conference is started, when the station that is convened the Voice Conference is released, the conditions for Data Conference may be changed depending on the management state of conference participants.*

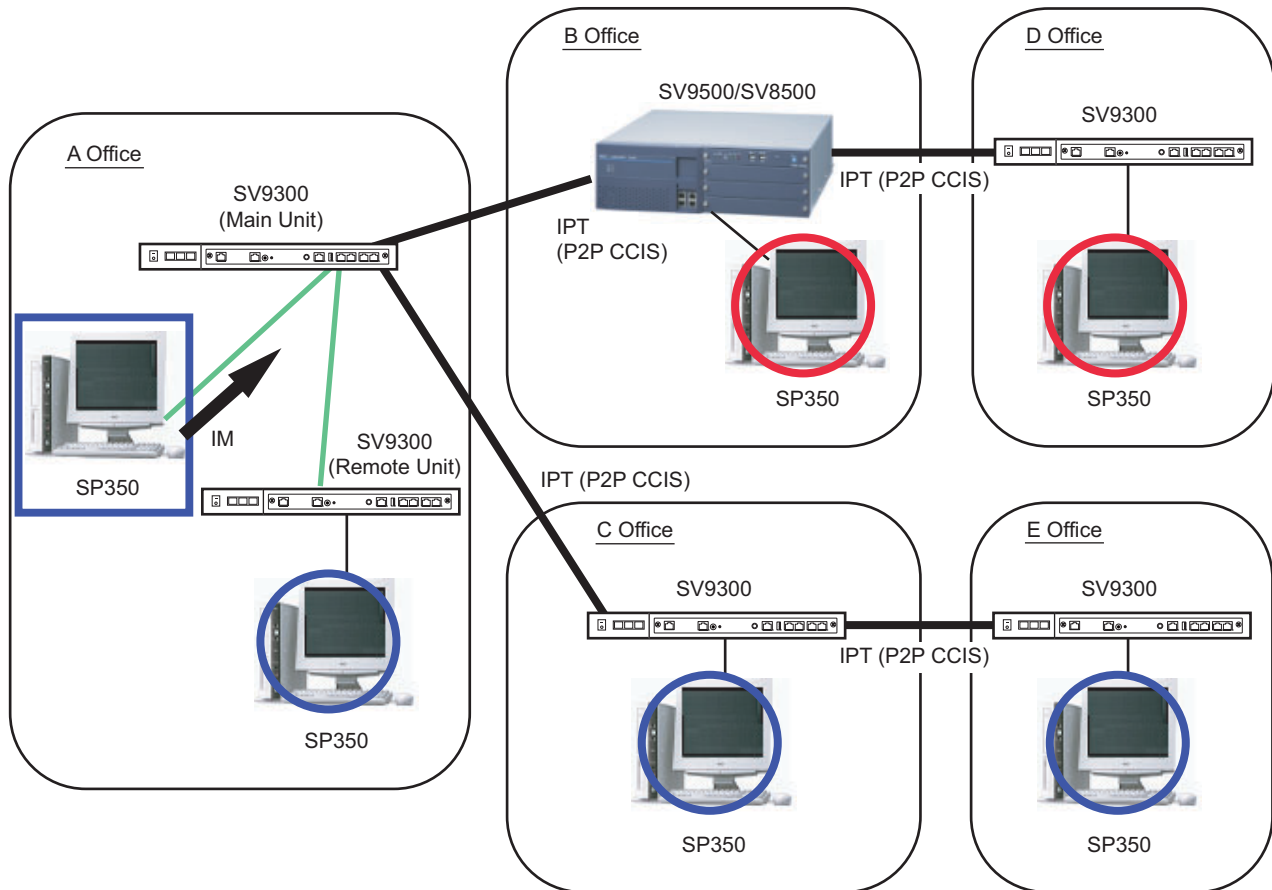
- The conditions when starting the Data Conference during the conference (Call-up type [Group Call]/ Participation type [Optional Call]) are as follows.
 - When the Data Conference caller and CFT (Conference Trunk) are in a same office, all of participants can be participated to the Data Conference.
 - When the Data Conference caller and CFT (Conference Trunk) are not in a same office, the Data Conference is not available.

- When the Data Conference is started between the offices, set the same DRS port number for each office (set by CM0B Y=1XX>31).
- When Data Conference is performed in a network with the open numbering system, the CCIS route to the opposite office must be specified the trunk route of the first choice. In addition, the calling number must be assigned the number including access code + office number + station number within 8 digits to the opposite office.
- The conditions when utilizing Data Conference between SP350 accommodated to SV9300 and SP350 accommodated to SV9500/SV8500/SV7000 are as follows.
 - To utilize Data Conference with SP350 accommodated to SV9500/SV8500/SV7000, the information on SV9300 must be registered by CMBC Y=16, 17 on the list of IP addresses of SV9500/SV8500/SV7000's MGC* that can be communicated.
 - *MGC: Media Gateway Controller
 - The list of IP addresses of MGC that can be communicated is created by the greeting process* between SV9300 and SV9500/SV8500/SV7000. The list created by this process becomes the greeting list.
 - *Greeting process: the process to repeat Read requests and Responses each other between SV9300 and SV9500/SV8500/SV7000.
 - To restrict Data Conference with SP350 accommodated to SV9500/SV8500/SV7000, the inquiries for the opposite office must be restricted to set the following system data.
 - CMBC Y=16: NONE (No data)*
 - *: The setting of CMBC Y=17 is also required when the setting of CMBC Y=16 is changed.
 - CM8A Y=5000-5255>181: 0 (Restrict)
 - To utilize Data Conference with SP350 accommodated to SV9500/SV8500/SV7000, the maximum of 50 SV9300s can be accommodated in a same network.
 - The UDP port number to be used by Data Conference with SP350 accommodated to SV9500/SV8500/SV7000 is the same as with SP350 between SV9300s.
 - For Data Conference with SP350 accommodated to SV9500/SV8500/SV7000, the greeting process is initially performed to the IP address of the destination office set by system data. In addition, by performing the greeting process to the office newly registered to the greeting list, the greeting processes are eventually performed between all offices in a same network. Therefore, it is necessary to make the communication for greeting processes possible between all offices.
 - Before Data Conference with SP350 accommodated to SV9500/SV8500/SV7000 can be available, a certain amount of time is required after the network connection is established between offices (up to time of the number of nodes × 1 minute is required).

Conditions for Instant Message

- Instant Messages (IM) can be sent to up to 8-party simultaneously between SP350s. The party received the Instant Message can be called to the sender after confirming the message, or sent the message to the sender. The illustration below shows the example of the Instant Message use.

Example of Instant Message Use



- Instant Message is available between Main Unit and SP350 accommodated to the Remote Unit, and between SP350s accommodated to SV9300 which is connected to IPT (P2P CCIS).
- The sending/receiving of Instant Message is not available via C.O.line (including a SIP trunk) and between SV9300 and SV9500/SV8500/SV7000 which is connected by IPT (P2P CCIS).
- When Instant Message is sent/received between the offices, set the same DRS port number for each office (set by CM0B Y=1XX>31).
- Instant Message can be sent/received regardless of the state (such as busy) of the terminal (available if the terminal is an online state).
- The sending/receiving of Instant Message in an own office cannot be restricted by system data. The sending/receiving of Instant Message between the offices can be restricted by system data.

- Instant Message cannot be sent to the SP350 which is not registered.
- Call-Forwarding and Station Hunting are not available for Instant Message. Instant Message is sent to only the station assigned as the destination.
- When Instant Message is sent, even when the setting for adding/deleting the reception number on called office is effective, the setting is not applied. In addition, the abbreviated dialing cannot be used as the destination number.
- When Instant Message is performed in a network with the open numbering system, the CCIS route to the opposite office must be specified the trunk route of the first choice. In addition, the destination number of Instant Message must be assigned the number (calling number) including access code + office number + station number to the opposite office.
- The conditions when utilizing Instant Message between SP350 accommodated to SV9300 and SP350 accommodated to SV9500/SV8500/SV7000 are as follows.
 - To utilize Instant Message with SP350 accommodated to SV9500/SV8500/SV7000, the information on SV9300 must be registered by CMBC Y=16, 17 on the list of IP addresses of SV9500/SV8500/SV7000's MGC* that can be communicated.
 - *MGC: Media Gateway Controller
 - The list of IP addresses of MGC that can be communicated is created by the greeting process* between SV9300 and SV9500/SV8500/SV7000. The list created by this process becomes the greeting list.
 - *Greeting process: the process to repeat Read requests and Responses each other between SV9300 and SV9500/SV8500/SV7000.
 - To restrict Instant Message with SP350 accommodated to SV9500/SV8500/SV7000, the inquiries for the opposite office must be restricted to set the following system data.
 - CMBC Y=16: NONE (No data)*
 - *: The setting of CMBC Y=17 is also required when the setting of CMBC Y=16 is changed.
 - CM8A Y=5000-5255>181: 0 (Restrict)*
 - *: When no data is set, the Instant Message transmission from SV9300 to SV9500/SV8500/SV7000 is not restricted.
 - To utilize Instant Message with SP350 accommodated to SV9500/SV8500/SV7000, the maximum of 50 SV9300s can be accommodated in a same network.
 - The UDP port number to be used by Instant Message with SP350 accommodated to SV9500/SV8500/SV7000 is the same as with SP350 between SV9300s.
 - For Instant Message with SP350 accommodated to SV9500/SV8500/SV7000, the greeting process is initially performed to the IP address of the destination office set by system data. In addition, by performing the greeting process to the office newly registered to the greeting list, the greeting processes are eventually performed between all offices in a same network. Therefore, it is necessary to make the communication for greeting processes possible between all offices.
 - Before Instant Message with SP350 accommodated to SV9500/SV8500/SV7000 can be available, a certain amount of time is required after the network connection is established between offices (up to time of the number of nodes × 1 minute is required).

Conditions for IP Multiline Terminal (DT700/DT800/DT900 Series)

- When Call Forwarding-All Calls is set to DT700/DT800/DT900 Series station, the station receives RTP instead of the announcement after the station goes off-hook operation.
- For originating/terminating calls between DT700/DT800/DT900 Series and DT700/DT800/DT900 Series/IP^{term}85, idle channels in VoIPDB is required. If no idle channel exists in VoIPDB, a call to the terminal is not performed.
- An External Hold Tone Source through Pin JACK on the CPU blade or PGD(2)-U10 ADP can be sent to DT700/DT800/DT900 Series via VoIPDB. For details, refer to the “Programming Manual”.
- The conditions when using XML applications with XML-based terminals are shown below.

① Service Conditions

- XML applications provide various services by working with XML-based terminals and an Application Server. By downloading XML-based applications from an Application Server and operating from an XML Browser, displaying various kinds of information and making a phone call become available.
- XML applications are available with terminals equipping XML Browser.
- XML applications correspond to the Application Server which provides application contents based on NEC XML Interface.
- The XML Browser can be used for the following modes. For details of the XML Browser setting, See “Config Setting for XML Application”. [📄 Page 2-113](#)

(a) Fixed Mode (Fixed Mode: Enable)

This is the mode that always displays an XML Browser. A screen for a phone call cannot be displayed during the Fixed Mode. The Fixed Mode is available by config setting of the terminal or application contents.

NOTE: *For the operation method of the Fixed Mode, refer to the manual of each application.*

(b) Normal Mode (Fixed Mode: Disable)

This is the mode that can be used the terminal not only for an XML Browser but also for a telephone. A telephone conversation is available when the call is terminated to the terminal while an XML Browser is running. However, the called party information is not displayed on the LCD of the terminal. To display the called party information, the XML application must be quit by Exit key.

NOTE: *If you do the operations mentioned above while an XML Browser is running, the office data must be assigned to the key of the terminal for key allocations in advance.*

- When using DT700/DT800/DT900 Series terminal on Fixed Mode, the Security key becomes ineffective. In addition, do not register anything to the Hold key.
- The conditions when making a call from an XML Browser are shown below.
 - (a) A Prime Line is used when making a call. When the Prime Line is busy, a call from an XML Browser is not available. The operation of the application at the time differs depending on the applications. For the operation of the application, refer to the manual of each application.
 - (b) The operations of SV9300 when making a call from an XML Browser are shown below.
 - When destination station is busy..... a caller hears BT
 - When destination station is an unassigned number a caller hears BT or announcement
 - When terminal of destination station is not login a caller hears BT
 - When Call-Forwarding is set to the destination station ... a call terminates to the transferred destination
 - When originating station is busy or hears RBT a call is not sent
 - (c) There are the following restrictions when making a call from an XML Browser.
 - Maximum dial number digits: 24 digits (access codes are included)
 - Available destination to call: Station, Trunk, Tie line
 - Handling of access code: operated as access code features
- When making a call from an XML Browser, the registration of numbering plan for dial number received from an XML Browser to SV9300 is required.

NOTE: *Register the dial number to the Application Server when making a call from an XML Browser. Register the dial number with access code, if required. For details of the registration method, refer to the manual of each application.*

② Operating Procedure for XML Application

■ Starting XML Browser

The method of starting an XML Browser differs depending on the Fixed Mode setting (Enable/Disable).

- For Fixed Mode (Fixed Mode: Enable)
 1. Login to the SV9300.
 2. An XML Browser is automatically started (an application content registered to the Home URL is displayed).
- For Normal Mode (Fixed Mode: Disable)

Manual start from the Menu or automatic start by config setting is available.

 1. Press **Menu** key.
 2. Select **Service** with cursor key and push the center of the cursor key.

■ **Exiting XML Browser**

1. Press and hold Exit key (Soft key).
The XML Browser is exited.

NOTE 1: *For a screen display after exiting the XML Browser differs depending on the state of the terminal upon termination.*

NOTE 2: *This operation is required only when the Fixed Mode is set to “Disable”. Start an XML Browser from the Menu screen of the terminal when restarting the XML Browser.*

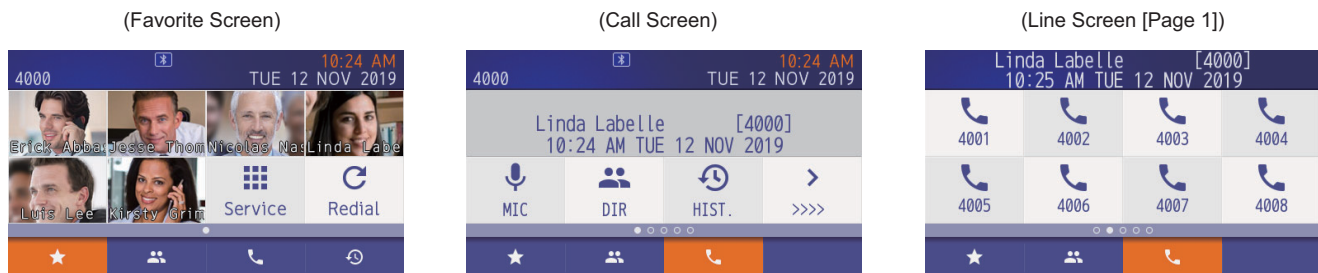
Conditions for DT900 Series

[9300V7 software required]

The DT900 provides Portal Mode display on LCD in addition to the traditional LCD display format (hereafter described as Classic Mode Display). The following figures show the example of screen format for DT900 firmware V2.2.0.0 or later.

NOTE: *The following conditions are supported for 9300V7 (V7.2.0) software or later.*

[Portal Mode Display] **NOTE 1**



[Classic Mode Display] **NOTE 2**

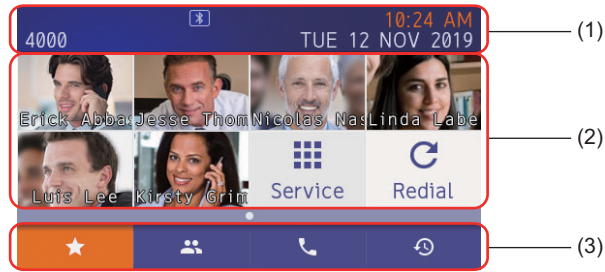


NOTE 1: *For DT900 firmware V2.2.0.0 or later, the Home Screen for the Portal Mode Display can be selected among the above screens. For details, refer to the “<Conditions on Portal Mode Display>”. [Page 1-32](#)*

NOTE 2: *In a Classic Mode, the LCD indication is the same as DT800 Series.*

NOTE 3: *The screen names described in the UNIVERGE SV9300 System Manual Issue 5.1 has been changed. The Favorite Screen (the former Shortcut Screen) and the Call Screen (the former System Screen) are used as the new screen names.*

- Favorite Screen

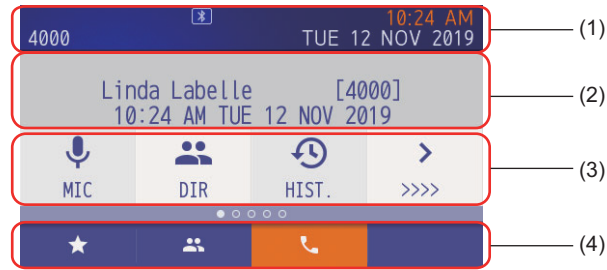


Item No.	Display Area	Description (Favorite Screen)
(1)	Information Area	<p>The following information sent from SV9300 is displayed.</p> <ul style="list-style-type: none"> - My Line information (Station Number/Name) - Calendar and clock - Status Icon <p style="text-align: right;">NOTE 1</p>
(2)	Shortcut Display Area	<p>The following items can be registered (Max.8)</p> <ul style="list-style-type: none"> - Telephone number which is frequently dialed. - The contact person which is registered in Terminal Phonebook. - The service menu button provide by XML application. - Redial button.
(3)	Local Soft Key Display Area	<p>The following Soft Key controlled by terminal is displayed.</p> <ul style="list-style-type: none"> - Favorite Screen The Favorite Screen is displayed. - Directory The Terminal Phonebook is displayed. If the Terminal Phonebook is not used, the system directory is displayed. - Call Screen The Call Screen is displayed which is controlled by SV9300. - Call History The incoming and outgoing Call Histories recorded in SV9300 are displayed. <p style="text-align: right;">NOTE 2</p>

NOTE 1: The calendar format is specified by terminal menu setting. The system data programming (CM12 Y=58: Type of Clock/Calendar Display on Multiline Terminal) is ineffective.

NOTE 2: The Local Soft key of the currently displayed screen is visualized in orange color.
Example: Favorite Screen

- Call Screen




Item No.	Display Area	Description (Call Screen)
(1)	Information Area	The same information as Information Area of Favorite Screen is displayed. NOTE 1
(2)	System Information Display Area	The information sent form SV9300 is displayed on upper and lower line. NOTE 2, NOTE 3
(3)	System Soft Key Display Area	The Soft Keys controlled by SV9300 are displayed. NOTE 4, NOTE 5, NOTE 6
(4)	Local Soft Key Display Area NOTE 8	The following Soft Key controlled by terminal is displayed. <ul style="list-style-type: none"> - ★ Favorite Screen The Favorite Screen is displayed. - 👤 Directory The Terminal Phonebook is displayed. NOTE 7 - 📞 Call Screen The Call Screen is displayed which is controlled by SV9300.

NOTE 1: When the date and time setting is changed while the system is operating, to reflect the calendar and system clock information displayed on the Information Area, it may take up to an hour (the setting is updated from SV9300 every hour at two minutes after the hour).

NOTE 2: The calendar and clock is displayed both Information Area and System Information Display Area. It can be deleted on System Information Display Area by system data programming (CM12 Y=58: 29).


NOTE 3: When the date and time setting is changed while the system is operating, the SV9300 reflects the setting within a minute.


NOTE 4: The same characters of Soft Key as Classic Mode Display are appeared which are set by system data programming (CM9A).

NOTE 5: *The icon is displayed corresponded to the character information and it cannot be changed. However, as for the following services, the  icon is all displayed.*

- *During Preset Dialing*
- *While the System Directory is displayed*
- *While the Incoming/Outgoing call histories are displayed*
- *During CAT mode*

NOTE 6: *As for the character information on the System Soft Key Display Area, the Double-sized characters indication is not available.*

NOTE 7: *The  Directory is not displayed when the Terminal Phonebook feature is not enabled by the system data programming (CM13 Y=100: 1 (Default)).*

NOTE 8: *The Local Soft key of the currently displayed screen is visualized in orange color.
Example:  Call Screen*

<Conditions on Portal Mode Display>

- This service is available from 9300V7 software or later. If the registered system version license is 9300V6 or former, the Classic Mode Display is applied.
- The display mode is selectable by system data programming with system basis or each station basis. (Default: Portal Mode). If the display mode is changed, terminal reset is required.
- The display mode can be changed by terminal menu operation (Default: Auto). If the display mode is changed, terminal reset is required. Following shows the display mode patterns by terminal settings.

Terminal Setting	Display Mode
Auto	Display mode is applied according to the system data programming.
Classic	Classic Mode Display is applied regardless of system data programming.

- Following shows the service conditions for each terminal model.

Terminal Type	Conditions
DT930	Portal Mode or Classic Mode Display can be selected. If the registered system version license is invalid, the Classic Mode Display is applied.
DT930 Touch Panel Model NOTE	Only the Portal Mode Display is available. If the registered system version license is invalid, this terminal cannot be used.
DT920 Self-Labeling Model NOTE	Portal Mode or Classic Mode Display can be selected. If the registered system version license is invalid, this terminal cannot be used.
DT920	Classic Mode Display is applied regardless of system data programming, since this model does not support Portal Mode Display.

NOTE: As for the line keys (Line/Trunk Feature keys) of DT930 Touch Panel Model and DT920 Self-Labeling Model, the same character information and icons as the DT800 Series DESI-less are displayed.

(a)The character information of line keys: assigns using CM9C (same as DT800 Series)

(b)The icons of line keys: displays the fixed icons depending on the state of the terminal




*1 The icon to be displayed cannot be changed.

*2 When the line key is changed during an operation, a reset of the terminal is required to reflect the change of icon display.

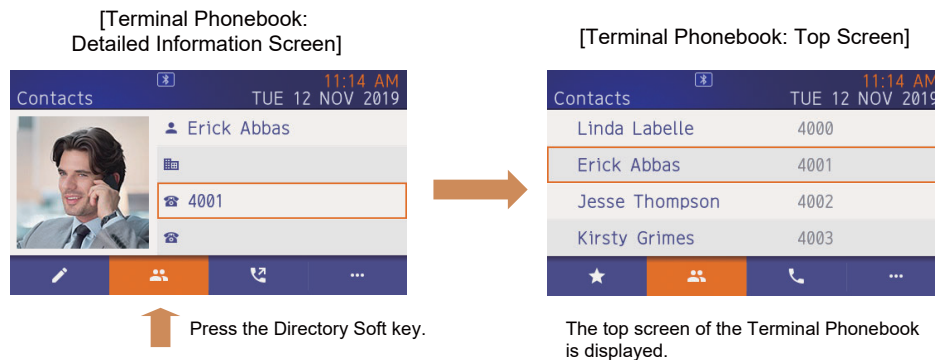
*3 The status icon to be displayed is similar to that of DT800 Series DESI-less.

*4 For All One Touch keys, the icons which invoke an image of one touch operation are displayed.

(c) When using a CAT mode, The CAT mode menu of line keys can also be used. The character information of line keys display the CAT mode menu. However, as for the icon to be displayed, the status icon for each line key is displayed. The (⋮) icon does not appear.

- The Line Screen is available for DT930 Touch Panel Model and DT920 Self-Labeling Model.
- Displaying of soft key functional description by pressing the help key is not supported.
- When pressing the Soft key on the currently displayed screen, each screen switches to the top of the screen.
 - When pressing the  Favorite while the Favorite Screen is displayed:
Switches the screen to the top of the Favorite Screen.
 - When pressing the  Directory while the Terminal Phonebook is displayed:
Switches the screen to the top of the Terminal Phonebook.
 - When pressing the  Soft Key of the telephone features while the Call Screen, or the Line Screen is displayed: Switches the Call Screen.

Example: For a screen transition of the Terminal Phonebook



- When pressing the Back Key in an idle state while the menu for the telephone services is displayed on the Call Screen or the Line Screen, the screen switches to the Call Screen. Also, the service will end for the following cases.
 - While the System Directory is displayed
 - While the Incoming/Outgoing call histories are displayed
 - During CAT mode

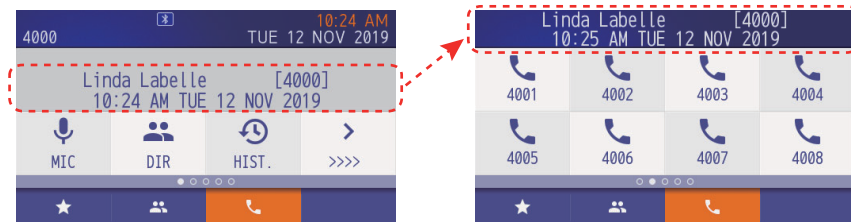
NOTE: As for the Favorite Screen and the operation when pressing the Back Key while the Terminal Phonebook is displayed, refer to the manual of the terminal.

- When starting up the DT900 Series, or pressing the Home Key of the DT900 Series, the screen displayed in the Portal Mode (Home Screen) can be selected by the system data setting (CM08>2003 [Default: Favorite Screen]/CM12 Y=103 [Default: As per CM08>2003]). Also, you can select the screen from the User menu of the terminal.

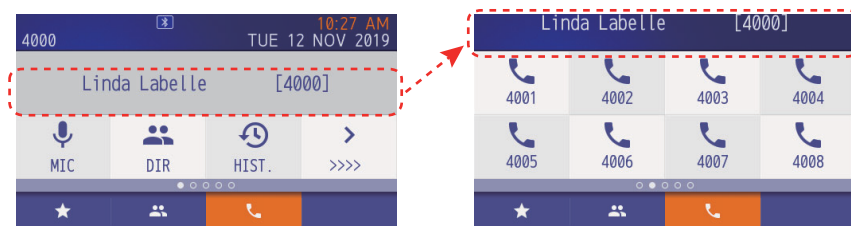
NOTE: When “Automatic” (default) is set by the User menu of the terminal, the system data setting has a priority. When other than “Automatic” (default) is set, the setting of the terminal has a priority. As for the details of the User menu, refer to the manual of the terminal.

- Conditions on Line Screen
 - The called party in a call, call duration, information during setting the service and others are displayed in the System Information Area (2nd and 3rd rows) on the Call Screen. The information is also displayed to the upper part on the Line Screen. When the station line is in an idle state, in the information area on the Line Screen, the same calendar, station number and name information as the System Information Area (the 2nd row and the 3rd row) on the Call Screen are displayed.

a) In the case with calendar indication



b) In the case without calendar indication



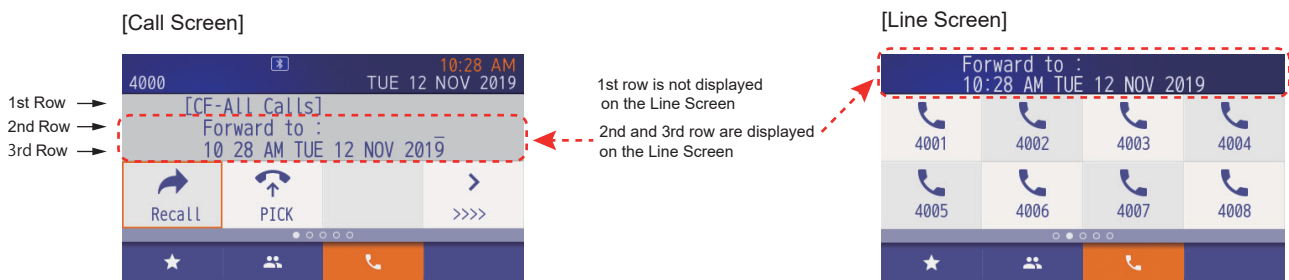
NOTE 1: The display format and language of the calendar and clock follow the system data setting (CM12 Y=63).

NOTE 2: When the second data of CM12 Y=59 is set to “29”, the calendar and clock are not displayed on the Line Screen.

NOTE 3: When the second data of CM15 Y=210 is set to “0”, the station number and the station name are not displayed on the Line Screen.

- For the following services, the service name is displayed in the System Information Display Area (the first row) on the Call Screen, but not on the Line Screen. To confirm the service name, manually switch to the Call Screen.
 - Call Forwarding services (Call Forwarding-All Calls/-Busy Line/-No Answer/-I'm here (Destination), IP Station Logout, Mobility Access)
 - Dual Ringing
 - Ecology
 - Malicious Call

Example: a display image for Call Forwarding-All Calls



- 1st row: Displays the service name
- 2nd row: Service Information Display (Forwarding destination)
- 3rd row: Service Information Display (Calendar and Clock)

- Even if the Double-sized characters indication is provided by the system data setting (CM12 Y=72), the normal indication is applied on the Line Screen (the Double-sized characters indication is not applied).
- When an incoming call is received to the DT920 Self-Labeling Telephone or a recall is executed, you can switch to the setting screen of the applicable Line Screen without displaying the Call Screen (CM12 Y=104).

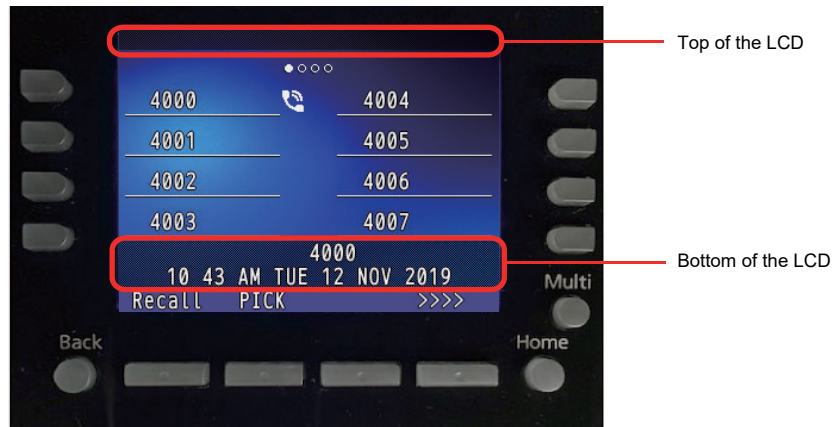
- The following table shows the displayed screens when an incoming call is received to the terminal and answered, or the call is abandoned by the calling party.

Conditions	Displayed Screen
<p>When an incoming call is received and answered, or the call is abandoned by the calling party while the following each screen is displayed.</p> <ul style="list-style-type: none"> - Favorite Screen - Call Screen - The screen on Terminal Phonebook 	<p>Displays the original screen.</p>
<p>When the current screen is switched to the Line Screen, then an incoming call is answered or the call is abandoned by the calling party.</p>	<p>Displays the Line Screen.</p>

- The automatic switching to the Line Screen is intended for My Line and Sub Line. For an incoming call to the Sub Line, the current screen is automatically switched to the Line Screen with CLI on the terminal LCD. During a call in progress, when an incoming call is terminated to the Sub Line, the CLI is displayed for about six seconds but the Line Screen is not automatically switched. Also, in the case that a call in progress is finished, the screen is not switched to the corresponding Line Screen. If there is no LCD indication to the Sub Line, the current screen is not automatically switched to the Line Screen.
- When pressing a Fixed Feature Key (Speaker, Transfer key etc.), a Line/Trunk Features key or a Dial key while the Line Screen is displayed, the current Line Screen is continuously displayed. To use the System Soft Key, manually switch to the Call Screen. However, the screen is switched to the Call Screen for the following cases.
 - While the System Directory is displayed
 - While the Incoming/Outgoing call histories are displayed
 - During CAT mode

<Classic Mode Display on the DT920 Self-Labeling Telephone>

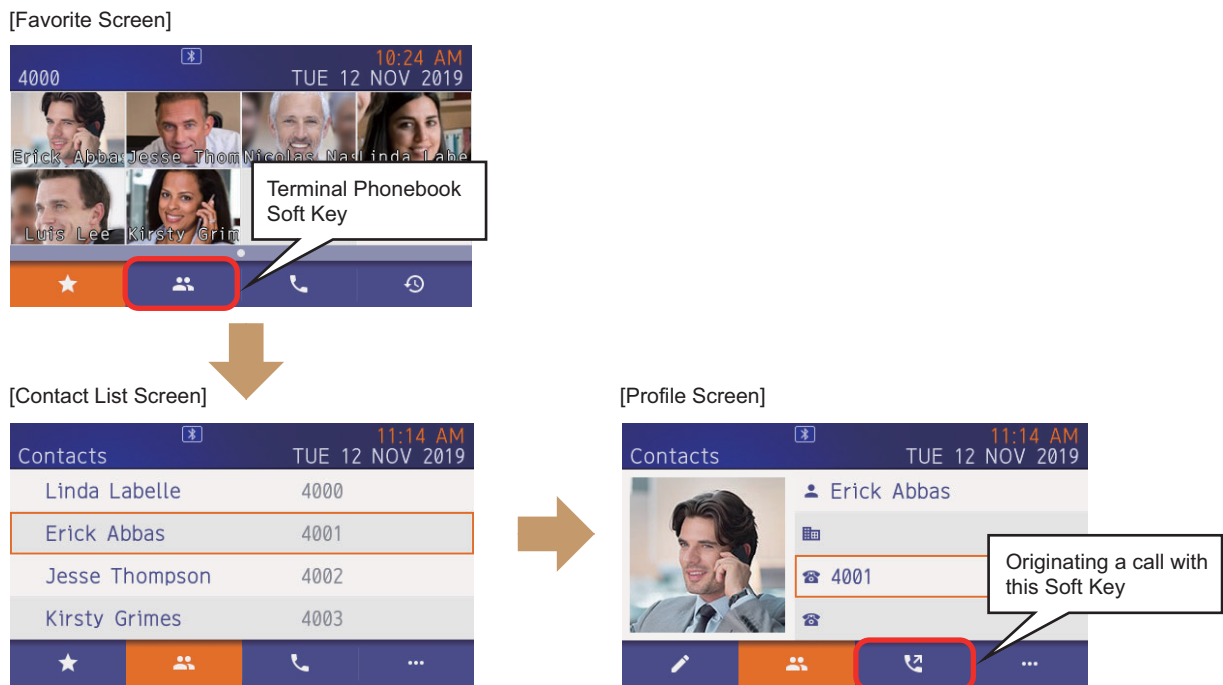
- Classic Mode Display on the DT920 Self-Labeling is available with 9300V7(V7.2.0) and DT900 firmware V2.1.0.0 or later.
- In the Classic Mode on the DT920 Self-Labeling Telephone, the LCD information is displayed at the top and bottom of the LCD.




- At the top of the LCD: displays the service name when each service is assigned.
The service names to be displayed are as follows.
 - Call Forwarding services (Call Forwarding-All Calls/-Busy Line/-No Answer/-I'm here (Destination), IP Station Logout, Mobility Access)
 - Dual Ringing
 - Ecology
 - Malicious Call
- At the bottom of the LCD:
 - For an idle state: displays the calendar, my line number and name
 - For a call in progress: displays information such as the called party number and name
 - For an incoming call: displays information such as the calling party number and name

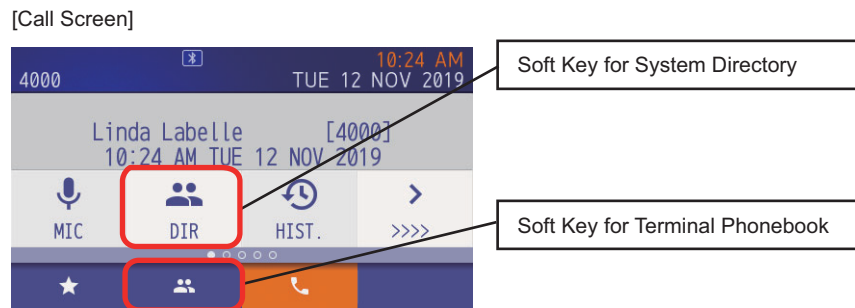
<Conditions on Terminal Phonebook>

- The DT900 provides Terminal Phonebook in addition to the personal directory (Station Speed Dialing) and system directory (System Speed Dialing). The Terminal Phonebook directory data is stored in terminal itself. The Terminal Phonebook supports photo data corresponded to the name and telephone number and it is displayed on the LCD. The following figures show the example of screen for DT900 firmware V2.2.0.0 or later.



- This service is available for DT920 Self-Labeling Telephone and DT930.
- This service can be enabled by station basis with system data programming. If the system data is changed, terminal reset is required.
- The directory data of Terminal Phonebook is stored in terminal itself and it is not deleted if the terminal is reset or power is disconnected. The directory data is not deleted after the Set Relocation (Station Data Swap) feature is performed.
- The Terminal Phonebook can be displayed during a call in progress. However, the call origination from Terminal Phonebook is available when the terminal is idle state.
- The Terminal Phonebook data is created by PCPro for each station with ZIP file format and downloaded to the terminal (refer to the PC Programming Manual for detail).
- The directory data can edit or delete by terminal operations. However, the updated phonebook data cannot be uploaded to the PCPro. If the terminal downloads the phonebook data from PCPro, the phonebook data is over written and updated data by terminal operation is erased.

- Even if the Terminal Phonebook is enabled by the system data programming (CM13 Y=100: 0), the incoming and outgoing Call Histories on the system side are displayed when pressing the Soft Key .
- The language displayed with Terminal Phonebook follows the downloaded directory data regardless of language selection of LCD display set by system data programming.
- This feature and system directory can be used with the same terminal under the following conditions (System Directory: Station Speed Dialing and System Speed Dialing).
 - The Soft Key is different as shown in the following figure.



- When the Terminal Phonebook is disabled by system data programming (CM13 Y=100: 1), the System Directory is displayed by pressing the Soft Key for Terminal Phonebook.
- The following feature cannot display the name information stored in Terminal Phonebook.

Service Feature	Remarks
<ul style="list-style-type: none"> - Caller ID Display - Call Redirect - Incoming Call History (CID Call Back) - CPN To Network-Present - Call ID Class 	The calling party's name information stored in system directory is displayed.
<ul style="list-style-type: none"> - Outgoing Call History (Stack Dial) 	The called party's name information stored in system directory is displayed.
<ul style="list-style-type: none"> - Calling Name and Number Display - Calling Name Display-CCIS 	The name information set by system data programming (CM77) is displayed.

Conditions for DT820 Terminal

[9300V3 STEP2 software required]

- The DT820 provides the following additional functions by registering the terminal license.
 - Gigabit Ethernet LAN port (Max.1000 Mbps) for external device connection
(Fast Ethernet LAN port [Max.1000 Mbps] is provided without terminal license.)
 - Up to 16/32 programmable keys can be used for 8LDX terminal
(8 keys are provided without terminal license.)

As for the registering the terminal license, refer to “Procedure for Registering Terminal License of DT820”. [☞ Page 2-117](#)

- There are two types of terminal license model for each market.
 - Floating License Model **[For North America]**
This model is able to add the Gigabit Ethernet function and the Line Key expansion function to the DT820 by registering the terminal license after factory shipment.

The lineup of this model is as follows.

Terminal Name	H/W Model Name	Description
DT820 (6D)	ITY-6D-1/1A (BK) TEL	6 Line Key with Fast Ethernet
DT820 (8LDX)	ITY-8LDX-1 (BK) TEL	DESI-less 8 Line Key with Fast Ethernet

- Node-lock License Model **[For Australia/EMEA/Asia]**
This model is already added the Gigabit Ethernet function and the Line Key expansion function to the DT820 by registering the terminal license before shipment.

The lineup of this model is as follows.

Terminal Name	H/W Model Name	Description
DT820 (6D)	ITY-6D-1P (BK) TEL	6 Line Key with Fast Ethernet
	ITY-6DG-1A/1P (BK) TEL	6 Line Key with Gigabit Ethernet
DT820 (8LDX)	ITY-8LDX-1P (BK) TEL	DESI-less 8 Line Key with Fast Ethernet
	ITY-32LDG-1P (BK) TEL	DESI-less 32 Line Key with Gigabit Ethernet

- DT820 consumes the following licenses.
 - One IP Port License/per terminal
 - One System Port License/per terminal
- The following number of DT820 that can be accommodated including other IP Multiline Terminals.
 - Up to 1024 terminals/per Unit
 - Up to 1536 terminals/per system

- The Power Factor of DT820 with POE from GCD-ETIA is 4.
- Color-coding of a Call Indicator Lamp for DT820 can only be used with three colors (red/green/yellow).
- Three-color rotation can be used only for DT820 (8LDX).
- DT820 does not support the Shortcut menu.
- For Chinese Language Display, both Simplified Chinese and Traditional Chinese can be displayed.
- The way of firmware update is the same as other DT800 Series. To update the IP Station firmware by specifying the terminal type, assign the following system data.
 - For DT820 (6D) : CM0C Y=00-07: 23 (DT820)
 - For DT820 (8LDX): CM0C Y=00-07: 24 (DT820 [DESI-less])
- For DT820 firmware upgrade, the default directory, default file name and time for upgrading per terminal is as shown below. Leave the default directory, default file name and time for upgrading per terminal as it is.

Type of IP Station	TFTP Server (Default File Name)	FTP Server (Default Directory + Default File Name)	Time for Upgrading Per Terminal
DT820 (6D)	ityisipe.tgz	/DT820E/ityisipe.tgz	Approx. 3 minutes
DT820 (8LDX)	ityisipex.tgz	/DT820D/ ityisipex.tgz	Approx. 3 minutes

- By assigning CM90 Y=00: F1090 (Headset) to the Line/Trunk feature key of DT820, you can use an EHS headset.

NOTE: Use an ADP-80 exclusive cable for DT820 when using an EHS headset. As for the operation of an EHS headset, refer to the manual of the terminal.

- The number of DT820's line keys can be read by CM90 Y=90. The number of line keys is read as follows depending on the terminal.
 - For DT820 (6D) : 12 buttons
 - For DT820 (8LDX): 32 buttons
- When the DT820 is connected to SV9300 using the 9300V3 software or before, the DT820 operates as DT710.
- The terminal type of DT820 can be read by CMFA Y=01. The terminal type is read as follows depending on the software version.
 - When using 9300V3 STEP2 software or later:
 - For DT820 (6D) : 23 (DT820)
 - For DT820 (8LDX): 24 (DT820 [DESI-less])
 - When using 9300V3 software or before:
 - For DT820 (6D) : 12 (DT710)
 - For DT820 (8LDX): 13 (DT710 [DESI-less])

- Terminal reset (CM12 Y=89) from PCPro is available.
- The following services and optional equipment are not available for DT820.
 - Live recording by external equipment
 - Smart Device Linkage
 - Line key extended console (8LK-L/16LK-L)
 - DSS Console **NOTE**

NOTE: *DSS Console cannot be used as the side option of DT820. However, DT820 can be used in cooperation with DSS Console accommodated to DLC blade.*

Conditions for SIP Wireless Terminal (MH240)

- The Wireless LAN access point which supports IEEE802.11b/g is required when using MH240 as an IP Station.
- A maximum of 1024 MH240 can be accommodated in a system (one MH240 is counted as an IP Station).
- To use MH240, the NEC IP Phone License is required for a client license.
- There is no call history for an incoming call with no CLI.
- The incoming operation for a Tie Line Call is the same for C.O. line call.
- Do not assign the key functions beyond 8 function keys because MH240 has only 8 function keys.
- If a MH240 is out of cell (Zone), the calling station keeps hearing RBT when calling to the MH240 which is out of cell (Zone). If Call Forwarding Logout is set to a MH240, the terminated call which is out of cell (Zone) can be transferred to the MH240.
- The function related to the Voice Call termination (such as Internal Zone Group Paging and Hands-Free Answerback-CCIS) is not available.
- The Mute Key setting (CM90 Y=00: F5013) of the Multiline Terminal is not available. To mute the terminal, use the Mute function by a built-in function of the terminal (Press OK key during a telephone conversation).
- Only Automatic Login Mode (Authentication by MAC Address) is available for the Login method to SV9300. In addition, an override is not available.
- The following features are not supported at the moment.
 - The features related to Multicast
 - SIP encryption, SRTP (However, the encryption for wireless zones is supported)
 - NAT Traversal
- JitterBufferSize, E/C are set to the terminal local.
- Codec/PayloadSize/JitterBufferSize to be supported are as follows.
 - Codec: G.711 μ -law, G.711 A-law, G.729a
 - PayloadSize: 20 ms.-40 ms.

NOTE 1: *The Codec/PayloadSize setting of the equipment to be connected with MH240 must be corresponded to the value as shown above (set the appropriate value definitely including “G.711/20 ms” in a Codec list).*

NOTE 2: *The connection for a codec by a transmission/reception direction is not supported.*

NOTE 3: *For a PayloadSize by a transmission/reception direction, the RTP beep tones might ring because they are calculated by the transmission PayloadSize of the MH240 side.*

- Unlike the existing DT Series, the following features function as local features of terminal-side.
 - The switching feature of contrast level on LCD
 - Directory feature, outgoing/incoming call history (the Center Directory is not supported)
 - Ringing Tone selection, a LED control feature when a call is terminated (Ringing Tone or LED pattern can be changed for each kind of station number, trunk number and registered directory number in Config menu)
- If a calling party number is the number which is registered in the directory of the MH240, the incoming operation of the registered directory number has a high priority (the function of the sorting Ringing Tone by station/trunk line does not work).
- In following cases, their conditions are displayed on the screen of the MH240. The screen display is unconditionally replaced for the appropriate lines until each state is canceled
 - During keylock
 - When a mute feature is effective during a telephone conversation
 - During a handsfree state
- If call is terminated during display of the screen related to the network setting of Config menu, the call termination display screen will not be changed.
- If MH240 has left in state that cannot be login to a SV9300 over long time, the another wireless connection has been required to connect to the SV9300 again by pressing the **OK** key continuously.
- The available range for a PAD control is from +6 dB (Gain) to -3 dB (Loss). When the PAD data of SV9300 is beyond the range as noted above, the PAD data is assigned by an upper/lower limit value.
- The area for saving tones inside of a MH240 are only for three countries. Therefore, when making the MH240 work with country type other than a downloaded tone, the tone file must be updated with the exclusive tool before use. For details, refer to the Manual of the MH240.
- The wireless antenna symbol does not appear unless the login is done to the SV9300. When confirming IP packets communication between the wireless zones before the login, select “Tool”, “Site Survey”, “PING” in that order on the Menu of the MH240, then execute the PING command.
- The lights of the function keys go off approximately 5 seconds after their keys is lit (due to the power saving, the lights of the function keys also go off in conjunction with the back light of the LCD. When charging the MH240 with a cradle connection, the lights of the function keys stays on).

NOTE 1: *The timer value to go off can be changed by the Config setting of the MH240.*

NOTE 2: *The state of the terminal can be confirmed even after the function keys lights off because the icon to indicate for which line keys is lit is displayed on the LCD.*

NOTE 3: *Press the any key including **OK** key to light the function keys which went off again. The line keys is lit again with LCD lighting up.*

- The FW download is not available from SV9300. Download with the exclusive tool (IP Phone/Easy Startup). This tool is available for not only FW download but also for batch download such as Config information of the MH240 and Tone data.
- The MH240-capable DHCP options are as follows.
 - Subnet Mask: 1
 - Router (Default Gateway): 3
 - DNS server: 6
 - Vendor unique information: 43
 - Vendor class identifier: 60
 - SIP server: 120
 - AutoConfig FTP server: 141
- Provide network design not to send multicast packets for the wireless zone connected to the MH240. If multicast packets is kept running in a wireless zones, the wireless device of the MH240 in itself always receives packets even if the packets is not for the MH240. Therefore, the waiting time is extremely shorten even if there is no operation.
- The ToS setting is not available for the MH240.

Conditions for Call Back from Call History of I755 IP DECT Terminal

[9300V3 software required]

- When the call is terminated to the I755 IP DECT Terminal, by notifying calling number with a trunk access code, you can call back to the called party from the call history.
- The following table shows the availability of kinds of trunks that can be added a trunk access code to a calling number.

×: Available Δ: Partly available -: Not available

Kind of Trunk	Availability	Remarks
COT	×	Number Display
ODT	-	
LDT	-	
DID	-	
ISDN (BRI)	×	
ISDN (PRI)	×	
Q-SIG	×	
T1 1.5Mbps	-	
E1 2Mbps	-	
MFC	×	
CCIS (CCT)	×	
IPT (P2P CCIS)	-	
SIP Trunk	Δ	Not available for Point-to-Multipoint connection

- When adding a trunk access code, the following system data setting is required.
 - CM13 Y=87: 0 (Adding trunk access code to calling number)
 - CM35 Y=189: X-XX (Trunk access code to be added)
- A maximum of two digits can be added for a trunk access code.
- A trunk access code is not added when the call is originated with no CLI from the trunk of calling party or by Point-to-Multipoint connection (CMA7 Y=46: 0) of SIP trunk.
- A calling number that can be notified is up to 24 digits including a trunk access code. If the calling number exceeds 24 digits, a trunk access code is not added.
- For an incoming call to a Standard SIP Terminal, a trunk access code is not included for the calling number notified by SMDR or OAI. For a call back, a trunk access code is included for the called number notified by SMDR or OAI.

Conditions for IP Single Line Telephone (SIP)

General Conditions

- Standard SIP Terminal can be connected to the system when satisfying the following conditions.
 - Standard SIP Terminal which is compliant with IETF RFC3261 and RFC5359 (NEC products/third party products are included).
 - The terminal that passed the operation test in advance.
- To use Standard SIP Terminal, the following licenses are required.
 - System Version License
 - System Capacity License (Each Standard SIP Terminal requires this license)
 - Standard SIP Terminal License
- One System Capacity License and one Standard SIP Terminal License are consumed per Standard SIP Terminal at registration.
- No license is consumed for SIP Converter registration.
- To accommodate Standard SIP Terminal, VoIPDB is required (voice data (RTP) is sent/received by VoIPDB).
- Standard SIP Terminal can be accommodated as follows.

NUMBER OF STANDARD SIP TERMINALS	ACCOMMODATED UNITS
Maximum of 512 terminals per Unit Maximum of 1024 terminals per system	Unit01-50

However, to accommodate the number of Standard SIP Terminals in the above, there are following conditions.

- The number of Standard SIP Terminals shall be up to 1536 terminals in a system including IP Multiline Terminals and other TDM Terminals.
- SIP Converter can be registered as follows.

NUMBER OF SIP CONVERTERS	REGISTRATED UNITS
Maximum of 32 converters per Unit Maximum of 128 converters per system	Unit01-50

- The number of simultaneous calls on Standard SIP Terminal depends on the following idle channels or the number of using channels.
 - The number of idle SIP Converter channels (up to 3 channels per SIP Converter)
 - The number of idle VoIPDB channels (up to 128 channels per VoIPDB)
 - The number of used VoIPDB channels for each connection configuration (see “Conditions of VoIPDB Channels” [Page 1-70](#))

Conditions for Standard SIP Terminal Accommodation

- Numbers of up to 8 digits (0-9, A (*), B (#)) can be assigned as station numbers for Standard SIP Terminal.
- For the following cases, Standard SIP Terminal receives terminal's local tone (BT/ROT) or becomes idle state.
 - When the called party disconnects the call during the call
 - When the called party is busy
 - When the called number is restricted (including access codes)
- DTMF signals are sent/received between VoIPDB and Standard SIP Terminal.
- When a connected party of Standard SIP Terminal made a Call Forwarding, the Standard SIP Terminal remains receiving hold tone until the call forwarded destination answered. The hold tone is not changed to RBT (only when the second data of CM08>1007 is set to 0).
- When a calling status continues for a certain period, you can set the operation by system data (CM41 Y=0>155) whether the call is considered as invalid and to disconnect the call (default: 30 minutes). Calls to be disconnected are only for the calls which are receiving RBT. Calls which are in progress are not disconnected.
- When a connected party of Standard SIP Terminal is replaced by Call Forwarding, SV9300 does not notify the station number of the newly connected party to the Standard SIP Terminal. Therefore, there is a case that the station number displayed on the Standard SIP Terminal is different from that of the actual connected party.
- When Standard SIP Terminal is in conditions which are unable to terminate a call such as an off hook state, the power is off, or the cable is pulled out, services such as Station Hunting/Announcement/Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out and Announcement-Standard SIP station No Answer will be functioned according to the system data settings.

 [Page 2-49, Page 2-55](#)

- The setting value of CM41 Y=0>85 (the message reply timer for Standard SIP station Off Hook/Power Off/Cable Pulled Out) must be set within “01-07” (4-28 seconds). This feature will not be effective when the setting value “08” (32 seconds) or higher is set.
- When you call up Standard SIP Terminal for long periods of time, depends on the type of Standard SIP Terminal, the terminal may have the feature to automatically disconnect the call by tens of seconds to tens of minutes. In that case, disable the feature for automatic disconnection by the setting of the terminal due to occur the following restrictions. When you cannot disable the setting, set the enough time for automatic disconnection, or set the feature such as Call Forwarding-No Answer not to occur problems. When the call is automatically disconnected, Standard SIP Terminal operates as follows.
 - The terminal on calling side continues to receive RBT even after the terminal on called side is automatically disconnected.
 - The Standard SIP station on called side cannot be originated or terminated until the terminal on calling side disconnects the call.

- When the system is reset, a call origination from Standard SIP Terminal or a call termination to Standard SIP Terminal is not available while the periodic registration timer settings (CM41 Y=0>151/152/153 [default: 1 hour]). In this case, the call origination or call termination is available by the reset of Standard SIP Terminal (Power ON/OFF of the terminal).
- The following table shows the summary of Standard SIP Terminal.

ITEM	DESCRIPTION	REMARKS
CODEC	G.711 (64 Kbps) G.722 (64 Kbps) G.729a (8 Kbps)	A CODEC type and a payload size can be specified by CM67 Y=00.
Payload size	20 ms./30 ms./40 ms.	
DTMF relay	Out-band (RFC2833) In-band (Voice Pass Through)	
Jitter Buffer	10-300 ms.	
QoS (Quality of Service)	IP Precedence/DiffServ	
PAD	[-16 dB] - [+ 16 dB] (default: 0 dB)	Set to VoIPDB
Echo canceller	G.168 (up to 64 ms.), NLP (Non Linear Processor)	Set to VoIPDB
Silence Suppression	Not supported	

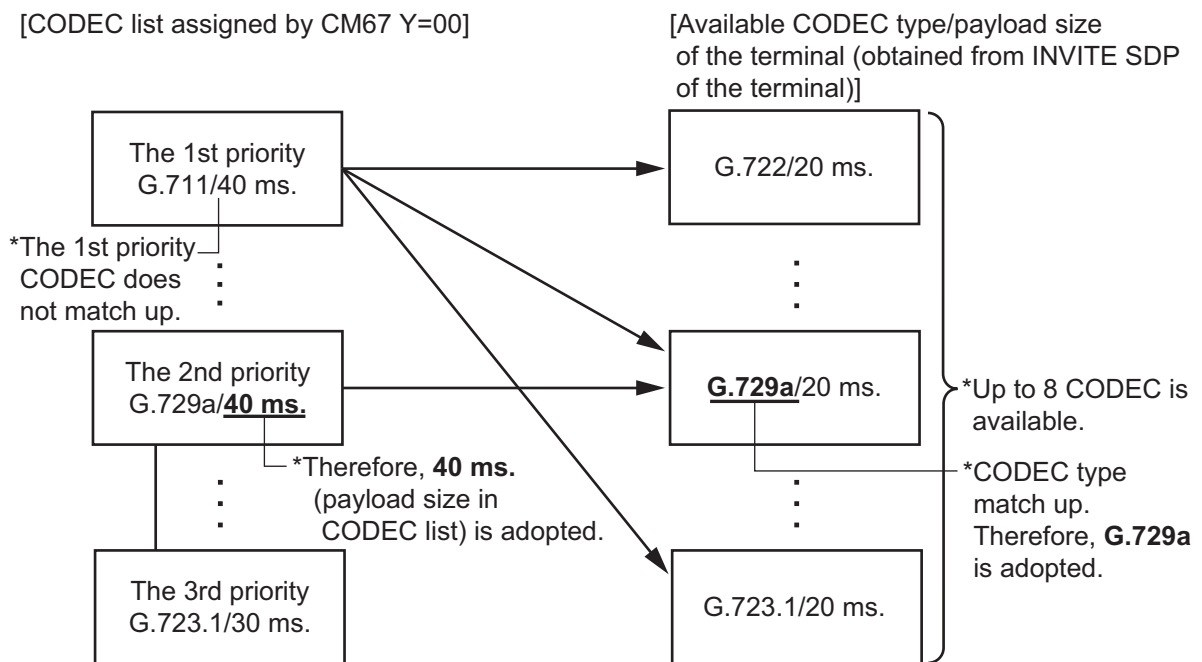
NOTE: G.722 is ineffective for GPZ-64IPLD/GPZ-128IPLD (VoIPDB). When setting only G.722 on CODEC list, the CODEC is converted into G.711.

- A CODEC type and a payload size for Standard SIP Terminal is determined by the following procedures.

(a) When receiving INVITE from Standard SIP Terminal:

Checks whether to match the CODEC type obtained from INVITE SDP from the 1st priority of the CODEC list (set by CM67 Y=00) in order (see the Example A on next page), then uses the matched CODEC type. For payload size, the one in the priority of the chosen CODEC is used. In this example, G.729a/40ms. is adopted as the CODEC type and the payload size of Standard SIP Terminal.

Example A:

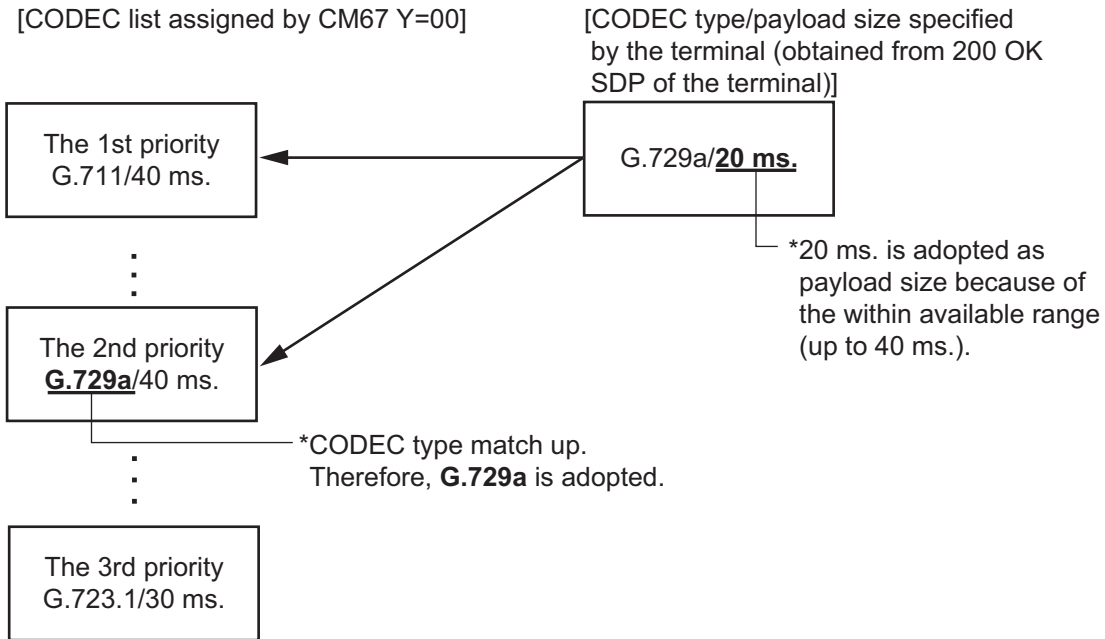


(b) When receiving 200 OK from Standard SIP Terminal:

Checks whether the CODEC in 200 OK SDP exists in the CODEC list (set by CM67 Y=00) or not (see the Example B on next page). If the CODEC exists, the CODEC is used. If not, the call is disconnected.

For payload size, if a payload size (200 OK SDP) is within an available range (up to 40ms.), the payload size is used. Otherwise, the one in the priority of the CODEC chosen from the CODEC list is used. In this example, G.729a/20ms. is adopted as the CODEC type and the payload size of Standard SIP Terminal.

Example B:



- If any CODEC in the list does not agree with available CODEC for Standard SIP Terminal, it results in media negotiation NG, and “488 Not Acceptable” is returned.
- Services on Standard SIP Terminal such as Video/Conference are not available.
- Keep Alive (session timer) of call session is not provided.

NOTE: *If a failure occurred during a calling, resources of SV9300 are released by the following elements.*

- *Disconnection of the call by the called party*
- *The expiration of term of Standard SIP Terminal REGISTER validity*
- *A new REGISTER from Standard SIP Terminal*

Conditions for Standard SIP Terminal Start up

- Authentication for Standard SIP Terminal functions as follows depending on the combinations of SIP Converter settings (CMBC Y=05) and settings for each terminal (CM1D Y=32).

◀: Default

Terminal Data SETTING (CM1D Y=32)	SIP Converter SETTING (CMBC Y=05)	AUTHENTICATION at REGISTRATION	AUTHENTICATION at CALLING
02/15◀: Allow the Authentication	1◀: Allowed	Allowed	Allowed
	0 : Restricted	Allowed	Restricted
03 : Restrict the Authentication	1◀: Allowed	Restricted	Restricted
	0 : Restricted		

- You can set by the system data (CM1D Y=32) whether to provide Authentication to Standard SIP Terminal.
- A user name for Authentication to be set to the terminal is fixed to the station number (CM10 Y=04) of the Standard SIP Terminal.
- The password for Authentication to be set to the terminal when executing Authentication, set the same password as one assigned by the system data (CM2B Y=00). Assign the password “0000” when no data is assigned.
- When the password is changed by the system data (CM2B Y=00), also the password assigned to the terminal must be changed.
- When starting up the Standard SIP Terminal that does not send REGISTER (the terminal without REGISTER), the registration of the terminal without REGISTER (CMBC Y=15*) is required after the setting of the IP address (CM12 Y=97) and the port number (CM12 Y=98) in addition to the IP Single Line Telephone (SIP) Data Assignment.
*: A system reset is required after the data setting.
- During a calling between Standard SIP Terminals, there is a case that the call between them remains without end for some reason. The session timer is set for releasing the call. The period of the session timer follows on the system data settings as shown the table below. In addition, if either of one is unable to continue the call, the call can be disconnected by ending the call on the other party.

CONDITIONS	SYSTEM DATA SETTINGS
Standard SIP Terminal does not support the “UPDATE” message.	Periodic registration timer settings: CM41 Y=0>151/152/153 (default: 1 hour)
Standard SIP Terminal supports the “UPDATE” message.	Session timer settings: CM15 Y=489: 0 (allow) CM41 Y=0>166 (default: 3 minutes)

Conditions of Call Hold

- For Standard SIP Terminal, you can select the following Hold Tone sources by the system data settings.

◀: Default

HOLD TONE SOURCE SELECTION METHODS	SYSTEM DATA	STANDARD SIP TERMINAL HOLD TONE SOURCE SELECTION METHODS (CM08>1007)	STANDARD SIP TERMINAL HOLD TONE SOURCE SELECTION METHODS for EACH STATION (CM13 Y=74)	SENDING PATTERN of HOLD TONE (CM48 Y=0)	HOLD TONE SOURCE KINDS for EACH TENANT (CM64 Y=1)	UNIT No. of EXTERNAL HOLD TONE SOURCE (CM64 Y=17)
When hearing Hold Tone Source on CPU blade of SV9300 (for system)	1 ▶: Hold Tone Source on CPU blade (selected by CM48 Y=0)	-	-	0000: No Tone 0500: Hold Message 1400: Hold Tone Source on CPU blade 1500: Internal Tone Generator NONE ▶: Hold Tone Source on CPU blade	-	-
When hearing Hold Tone Source on CPU blade of SV9300 (for each station)	0: Hold Tone Source on Standard SIP station	0: Hold Tone Source on CPU blade (selected by CM48 Y=0)	-	0000: No Tone 0500: Hold Message 1400: Hold Tone Source on CPU blade 1500: Internal Tone Generator NONE ▶: Hold Tone Source on CPU blade	-	-
When hearing External Hold Tone Source of SV9300 (for system)	-	-	-	1300: External Hold Tone Source on CPU blade	-	-

Continued on next page

◀: Default

HOLD TONE SOURCE SELECTION METHODS	SYSTEM DATA	STANDARD SIP TERMINAL HOLD TONE SOURCE SELECTION METHODS (CM08>1007)	STANDARD SIP TERMINAL HOLD TONE SOURCE SELECTION METHODS for EACH STATION (CM13 Y=74)	SENDING PATTERN of HOLD TONE (CM48 Y=0)	HOLD TONE SOURCE KINDS for EACH TENANT (CM64 Y=1)	UNIT No. of EXTERNAL HOLD TONE SOURCE (CM64 Y=17)
When hearing External Hold Tone Source of SV9300 (for each tenant)	-	-	-	1300: External Hold Tone Source on CPU blade	00-09: External Hold Tone Source No. by PGD(2)-U10 ADP	01-50: Unit No. of External Hold Tone Source
	-	-	-	1300: External Hold Tone Source on CPU blade	11: External Hold Tone Source through Pin JACK on the CPU blade	01-50: Unit No. of External Hold Tone Source
When hearing Hold Tone Source of Standard SIP Terminal	0: Hold Tone Source on Standard SIP Terminal	0: Hold Tone Source on Standard SIP Terminal	1◀: As per CM08>1007	-	-	-

- When a call is placed on hold and an External Tone Source is received during a call with Standard SIP Terminal, the LCD indication which means during a call is continued but a held party is not displayed on the LCD.

Conditions of Call Transfer

- If no dialing operation is provided within 15 seconds after pressing the Transfer key on Call Transfer, the call is considered as an error and the call origination from the terminal is rejected (from SV9300, “488 Not Acceptable Here” is sent to the INVITE of 2nd Call).
- The call transferred party remains the state of hearing Hold Tone even after the call transferring party is released (the tone is not changed from Hold Tone to RBT).

Conditions of Call Forwarding-Not Available

- For Call Forwarding when the called Standard SIP station is in states such as off hook, power off and cable pulled out, there is no limitation for the number of registrations when a Call Forwarding destination is a station. You can verify the number of registrations by CMEC Y=2 when a Call Forwarding destination is an outside number. For details, refer to the Call Forwarding-All Calls feature in the “Programming Manual”.

Conditions of Consultation Hold

- Pressing Hold key again does not make the Standard SIP Terminal return to the original call.
- When the 3rd party disconnects the call, the Standard SIP Terminal does not return to the original call but continues to place the 2nd party on hold.

Conditions of Message Waiting

- Assign IP address of the SV9300 for SUBSCRIBE sending destination on the Standard SIP Terminal.
- Set the second data of CM13 Y=03 to “0” when using the Message Waiting service.
- When a message is sent to the Message Waiting service, the SIP message (NOTIFY) defined by RFC3842 is sent to the Standard SIP Terminal. The display method after receiving the SIP message, such as lamp and icon depends on the terminal specifications.
- If a message is sent to the Message Waiting service while the Standard SIP Terminal is not operating, the SIP message (NOTIFY) defined by RFC3842 is sent to it after the terminal’s start up.
- When a message is set to the Message Waiting Service while the Standard SIP Terminal is originating/terminating a call/being connected, the SIP message (NOTIFY) defined by RFC3842 is sent to it after the call is disconnected.
- The Standard SIP Terminal cannot originate/terminate a call while the SIP message (NOTIFY) defined by RFC3842 is sent to the terminal.
- The number of received SIP messages is not displayed.

Conditions for Call Origination and Service Settings

- Only “Preset Dialing” (dialing a Access Code + additional number at once) is available as the dialing method for Standard SIP Terminal.

- When using an Access Code with additional number (such as Call Forwarding-All Calls), you can use “Preset Dialing” (dialing a Access Code + additional number at once), or “Overlap Sending” (dialing an additional number after dialing an Access Code) **[9300V3 software required]**.

The following table shows the setting methods for each call origination method.

◀: Default

System Data Call Origination Method	CM08>1035	CM08>1046 [9300V3 software required]
Preset Dialing NOTE 1	0 : Hearing RBT 1◀: Service Set Tone (SST)	Data assignment is not required.
Overlap Sending [9300V3 software required] NOTE 2, NOTE 3, NOTE 4	1◀: Service Set Tone (SST)	1◀: Available

NOTE 1: *By using the service setting announcement feature when setting a service with an Access Code, you can use the announcement from VRS instead of Service Set Tone (SST) (however, it is not available when using Call Park for Single Line station and Call Park-System).*

NOTE 2: *When setting the services by Overlap Sending, it is recommended to set the DTMF sending method to RFC2833.*

NOTE 3: *When a call is originated with an Access Code which is not able to use the Overlap Sending, the call is disconnected after a lapse of a few seconds.*

NOTE 4: *When a call is originated with an Access Code, the call becomes a state which is in use because SV9300 detects a dial number (DTMF signal) from a Standard SIP Terminal.*

- The following table shows the availability of services using Access Codes for Preset Dialing and Overlap Sending.

×: Available –: Not available

Service Name	Access Code	Preset Dialing	Overlap Sending [9300V3 software required]
Call Park-System Set	A008	×	-
Call Park-System Retrieve	A009	×	-
Call Forwarding-All Calls Set	A010	×	×
Call Forwarding-All Calls Cancel	A011	×	-
Call Forwarding-No Answer/Busy Line Set	A012	×	×
Call Forwarding-No Answer/Busy Line Cancel	A013	×	-
Call Forwarding-Busy Line Set	A014	×	×
Call Forwarding-Busy Line Cancel	A015	×	-
Call Forwarding-No Answer Set	A016	×	×
Call Forwarding-No Answer Cancel	A017	×	-
Call Forwarding-I'm here (Destination) Set	A018	×	×
Call Forwarding-I'm here (Destination) Cancel	A019	×	×
Call Pickup-Group	A020	×	-
Call Pickup-Direct	A021	×	×
Do Not Disturb Set	A022	×	-
Do Not Disturb/Return Message Schedule Cancel	A023	×	-
Wake Up Call/Timed Reminder Set	A024	×	×
Wake Up Call/Timed Reminder Cancel	A025	×	-
Wake Up Call/Timed Reminder Check	A026	×*	-
Wake Up Call Set from Predetermined Station (Single Wake Up time operation)	A027	×	-

* Timed Reminder Check is available only when using Speech Synthesis.

Continued on next page

×: Available –: Not available

Service Name	Access Code	Preset Dialing	Overlap Sending [9300V3 software required]
Wake Up Call Set from Predetermined Station (Multiple Wake Up time operation)	A028	×	-
Maid Status	A029	×	×
Monitor	A033	×	×
Call Pickup-Designated Group	A037	×	×
MW Lamp Control Set	A040	×	-
MW Lamp Control Reset	A041	×	-
Day Night Mode Change by Station Dialing	A043	×	×
ACD/UCD Station Busy Out Set	A044	×	-
ACD/UCD Station Busy Out Reset	A045	×	-
Call Hold	A046	×	-
TAS Answer A	A047	×	-
TAS Answer B	A048	×	-
TAS Answer C	A049	×	-
TAS Answer D	A050	×	-
TAS Answer E	A051	×	-
Trunk Hold	A058	×	-
Trunk Answer	A059	×	-
Call Park-Tenant Set/Retrieve	A062	×	-
System Speed Dialing Origination	A067	×	-
Last Number Call (Last Number Redial)	A069	×	-
Paging Answer Zone 0	A070	×	-
Paging Answer Zone 1	A071	×	-
Paging Answer Zone 2	A072	×	-

Continued on next page

×: Available –: Not available

Service Name	Access Code	Preset Dialing	Overlap Sending [9300V3 software required]
Paging Answer Zone 3	A073	×	-
Paging Answer Zone 4	A074	×	-
Paging Answer Zone 5	A075	×	-
Paging Answer Zone 6	A076	×	-
Paging Answer Zone 7	A077	×	-
Paging Answer Zone 8	A078	×	-
Paging Answer Zone 9	A079	×	-
Speaker/Radio Paging Cancel (Delay Operation)	A080	×	-
Account Code	A085	×	-
Authorization Code	A086	×	-
Forced Account Code	A087	×	-
Direct Data Entry	A097	×	×
Voice Response System Access Record	A100	×	-
Voice Response System Access Replay	A101	×	-
Voice Response System Access Delete	A102	×	-
Announcement Service Record	A103	×	×
Announcement Service Group 0 Replay	A104	×	-
Announcement Service Group 1 Replay	A105	×	-
Announcement Service Group 2 Replay	A106	×	-
Announcement Service Group 3 Replay	A107	×	-
Announcement Service Group 4 Replay	A108	×	-
Announcement Service Delete	A109	×	×
Voice Message Waiting Service-Individual (Setting of station numbers to be sent)	A114	×	×

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×: Available –: Not available

Service Name	Access Code	Preset Dialing	Overlap Sending [9300V3 software required]
Voice Message Waiting Service-System/Individual (Reset of station numbers to be sent)	A119	×	×
Voice Message Waiting Service-System/Individual Retrieve	A120	×	-
Sending of Hooking Signal to C.O. line/Centrex from PB telephone	A158	×	-
Voice Message Waiting Service-Individual All Clear when the called station does not answer	A165	×	-
Split Call Forwarding-All Calls Set	A180	×	×
Split Call Forwarding-All Calls Cancel	A181	×	-
Split Call Forwarding-Busy Line/-No Answer Set	A182	×	×
Split Call Forwarding-Busy Line/-No Answer Cancel	A183	×	-
Whisper Page	A188	×	-
Call Forwarding-Not Available Set	A189	×	-
Call Forwarding-Not Available Cancel	A190	×	-
Call Forwarding-Not Available Replay	A191	×	-
System Clock Setup by Station Dialing	A197	×	×
Call Park-System Set which retrieved by dialing station number	A198	×	-
Call Park-System Retrieve by dialing station number	A199	×	-
Station Class change with Station Authorization Code	A230	×	-
Station Authorization Code/Password Change	A231	×	-
Pad Lock Set by Station Authorization Code	A232	×	-

Continued on next page

×: Available –: Not available

Service Name	Access Code	Preset Dialing	Overlap Sending [9300V3 software required]
Pad Lock Reset by Station Authorization Code	A233	×	-
Call Pickup-Group (Pilot)	A234	×	-
Call Forwarding-Logout/Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out Set	A241	×	×
Call Forwarding-Logout/Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out Cancel	A242	×	-
System Speed Dialing origination (1-8 digits abbreviated Code: depends on CM42>77)	A243	×	-
Mobility Access Mode Set (Trunk Access Code 1)	A256	×	×
Mobility Access Mode Cancel	A257	×	-
Do Not Disturb-Override/Call Forwarding-All Calls Override	A260	×	×
Dual Ringing Set	A261	×	-
Dual Ringing Cancel	A262	×	-
Peer-to-Peer Connection for Station-to-Station Call with Standard SIP station Set	A263	×	-
Peer-to-Peer Connection for Station-to-Station Call with Standard SIP station Cancel	A264	×	-
Mobility Access Mode Set (Trunk Access Code 2)	A267	×	×
Mobility Access Mode Set (Trunk Access Code 3)	A268	×	×
Mobility Access Mode Set (Trunk Access Code 4)	A269	×	×
Wake Up Call Set with Snooze [9300V3 software required]	A274	×	×

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×: Available –: Not available

Service Name	Access Code	Preset Dialing	Overlap Sending [9300V3 software required]
Malicious Call Set by Malicious Call number specification [9300V3 software required]	A275	×	×
Malicious Call Delete by Malicious Call number specification [9300V3 software required]	A276	×	×

Conditions for OAI Support on Standard SIP Terminal

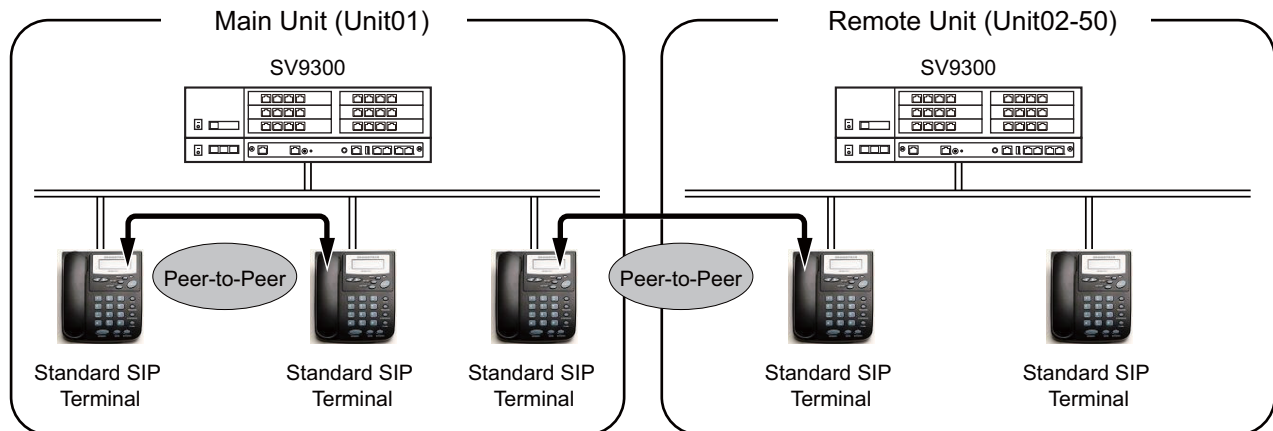
- OAI is supported on the Standard SIP Terminal.
- This feature is available regardless of country code.
- OAI and the license for Standard SIP Terminal are required to operate this feature.
- The terminal type of Standard SIP Terminal to be notified by SMFN is determined by CM08>804: 0.
- For the Call Transfer by SCF FN=07 (TransferCall), the Standard SIP Terminal does not receive RBT during calling to the transfer destination. Without receiving RBT, the terminal goes to talking status by answering on the transfer destination.
- Even if the connected party is changed by SCF FN=04 (DequeueCall), SCF FN=07 (TransferCall), or SCF FN=09 (Join2Parties), the connected party information displayed on the Standard SIP Terminal is not changed. Therefore, there is a case that the displayed number is different from the one of actual connected party.
- With this feature, Peer-to-Peer connection using Standard SIP Terminal is available.

There are following constrains when using Peer-to-Peer connection between Standard SIP Terminals.

- For SCF FN=03 (RingCall), SCF FN=04 (DequeueCall) and SCF FN=09 (Join2Parties), the calls are connected via VoIPDB (Peer-to-Peer connection is not available).
- The following SCF are not available for Peer-to-Peer connection between Standard SIP Terminals. SCF FN=06 (ListenToCall), SCF FN=07 (TransferCall), SCF FN=08 (ConferenceCall), SCF FN=10 (StopTransfer), SCF FN=19 (TransferCampOn), SCF FN=20 (SendDTMFConn)
- SMFN (ANSWER_NORMAL) and SMFN (ANSWER_HOLD) are not available for Peer-to-Peer connection between Standard SIP Terminals.

Conditions for Peer-to-Peer Connection with Standard SIP Terminal

- For Standard SIP Terminals within Main Unit, or Standard SIP Terminals between Main Unit and Remote Units, this feature allow you to send the voice data (RTP) without via VoIPDB and call by Peer-to-Peer connection.



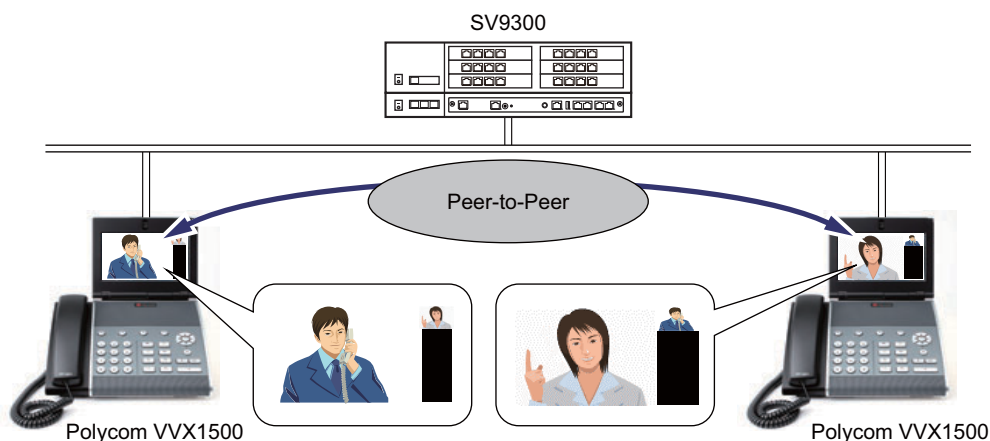
- Peer-to-Peer connection is available only when you call between Standard SIP Terminals.
- Peer-to-Peer connection between Standard SIP Terminals via IPT(P2P CCIS)/SIP trunk is not supported.
- During Peer-to-Peer connection, the available features are only basic call origination/termination, call hold (other features such as Call Transfer are not available).
- Whisper Page/Exclusive Override are not available against the Standard SIP Terminal connected via Peer-to-Peer. If the terminal using these features are operated, “restricted” is displayed and ROT is heard on the operating terminal.
- Because call monitoring and recording using Voice Mail feature are not available for the Standard SIP Terminal connected via Peer-to-Peer.
- There are the following operation methods for Peer-to-Peer connection.
 - The operation method using an access code (CM20: A263)
By dialing Access Code for Peer-to-Peer connection + station number when originating a call, you can change over the call via VoIPDB to call via Peer-to-Peer.
 - The operation method using RTP route settings (CM15 Y=229: 0)
By doing the Peer-to-Peer settings by system data in advance, all the calls between Standard SIP Terminals can be connected via Peer-to-Peer.
- * By making the Peer-to-Peer setting effective (CM15 Y=229: 0) as a RTP route setting, and dialing Access Code for Peer-to-Peer connection cancel (CM20: A264) + station number when originating a call, you can also change over the call via Peer-to-Peer to call via VoIPDB.

NOTE: As for the system data programming, refer to the “(16) Peer-to-Peer/VoIPDB connection settings for Standard SIP station”. [📄 Page 2-61](#)

- The conditions when using access codes (CM20: A263/A264) are as follows.
 - Access Code is given priority over RTP route settings (CM15 Y=229). When without Access Code, Standard SIP Terminal functions as per RTP route settings (CM15 Y=229).
 - When using access codes, use the preset dialing with access code + station number. When a call is originated by dividing an access code and a station number, the call origination is restricted.
 - When an access code are used via terminals other than Standard SIP Terminals or trunks, the caller using the access code cannot originate a call and ROT is heard on the operating terminal.
 - When a call is originated with an access code to the terminals other than Standard SIP Terminals or trunks, the access code is deleted from the called party number before the call termination on the called party.
 - During a call, you cannot change over the call from the VoIPDB connection to the Peer-to-Peer connection.
- When using RTP route settings (CM15 Y=229), make the Peer-to-Peer connection effective for both the Standard SIP Terminals on the calling side and the called side. When only for either of the two sides is effective, the call is connected via VoIPDB (not via Peer-to-Peer).
- SIP messages are controlled via VoIPDB and only voice data (RTP) is connected via Peer-to-Peer.

Conditions for Video Phone with Standard SIP Terminal

- Video Phone is available between Standard SIP Terminals within a same system.



NOTE: Only ST500 and Polycom VVX1500 are supported for Video Phone. The Video Phone connection is not available between ST500 and Polycom VVX1500.

- Video Phone is available only where the Peer-to-Peer connection is established.
- The operation related to Video Phone depends on the specifications of ST500/Polycom VVX1500.
- Whisper Page/Exclusive Override are not available for Video Phone. If the terminal using these features are operated, “restricted” is displayed and ROT is heard on the operating terminal.

- When no license is registered, Video Phone via Peer-to-Peer connection is not available but only Voice Call via VoIPDB can be used.
- Video Phone between Standard SIP Terminal via IPT(P2P CCIS)/SIP trunk is not available.
- SIP messages are controlled via VoIPDB.
- Voice and Video data is transmitted and received over Peer-to-Peer connection between Standard SIP Terminal.
- The conditions for available operation methods and access codes for Video Phone feature are the same in Peer-to-Peer connection between Standard SIP Terminals.
- As for the system data programming of Video Phone feature, refer to the “(16) Peer-to-Peer/VoIPDB connection settings for Standard SIP station”. [📄 Page 2-61](#)

Conditions for Standard SIP Terminal Accommodation in Remote Unit

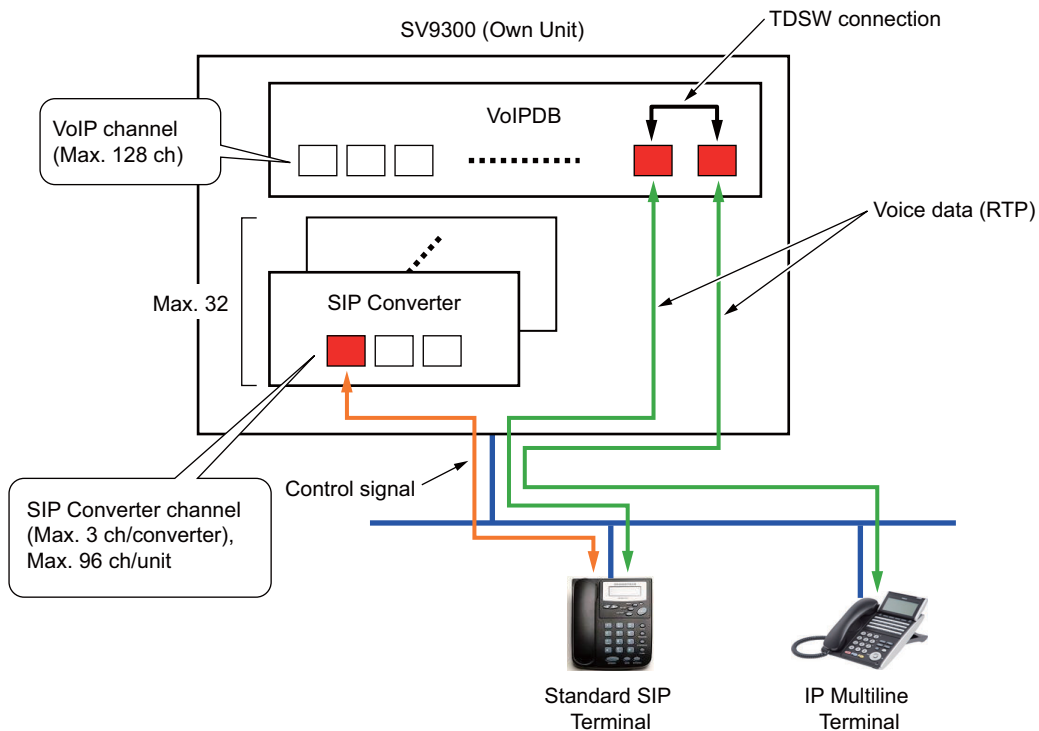
- When accommodating Standard SIP Terminal to Remote Unit, up to 1024 stations can totally be accommodated in both Main Unit and Remote Units.
- When accommodating Standard SIP Terminal to Remote Unit, the registration to SIP Converter is required to the Remote Unit which accommodates the Standard SIP Terminal.
- At the Standard SIP Terminal settings, as for the destination of the registration, specify the IP address of the VOIP port on the Unit that you want to login.
- The Peer-to-Peer connection between Standard SIP Terminals is available within Main Unit or between Main Unit and Remote Unit.
- Standard SIP Terminal without REGISTER can be accommodated to only Main Unit (Unit01).
- Conditions when Remote Unit is switched from Normal Mode to Survival Mode are as follows.
 - Standard SIP Terminal registered to Remote Unit cannot be originated or terminated until re-registration is completed.
 - When switching from Normal Mode to Survival Mode, the periodic REGISTER from the Standard SIP Terminal which is registered during Normal Mode is assumed as a new registration and treated (same for the periodic REGISTER during Survival Mode when switching from Normal Mode to Survival Mode).
- For DTMF methods, DTMF signal sending or receiving by In-band DTMF and Out-band DTMF (RFC2833) is available.
- For the connection between Standard SIP Terminals or Standard SIP Terminal and SIP trunk, DTMF setting between Units when the In-band DTMF signal is received can be selected from In-band DTMF or Out-band DTMF by using system data (CM67 Y=26)
- While Remote Unit is in Survival Mode, the number of Standard SIP Terminals at the expiration of the term of temporary license validity becomes equal to the number of them connected to the Unit in Normal Mode.

Conditions for Resource Channels When Using Standard SIP Terminal

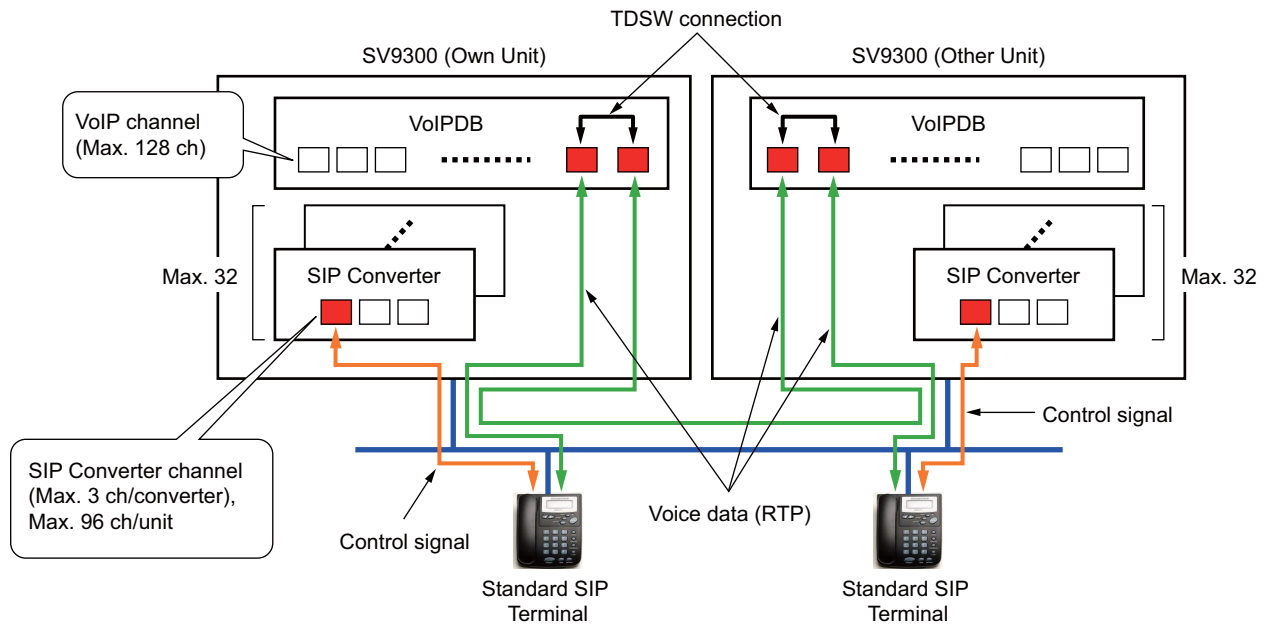
- Standard SIP Terminal uses the following resource channels.
 - SIP Converter channel (for SIP message control)
 - VoIPDB channel (for voice data (RTP) sending/receiving)
- When using Standard SIP Terminal accommodated to a Remote Unit, SV9300 uses the SIP Converter channel and the VoIPDB channel of the Remote Unit that the terminal is accommodated.
- As for the call patterns for Standard SIP Terminal, there are connection via VoIPDB and Peer-to-Peer connection. The use of channel resource differs depending on the each call pattern. The following illustrations show the connection examples via VoIPDB and Peer-to-Peer.

Connection Examples

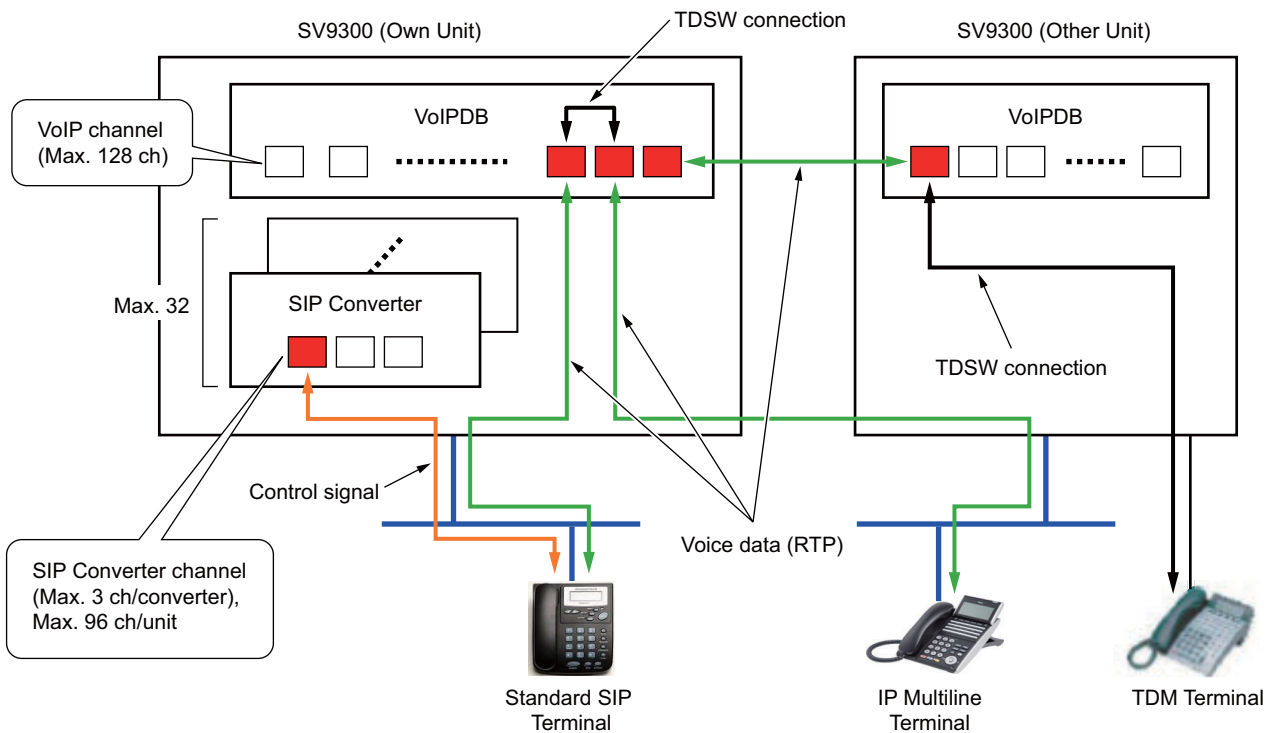
(a) For a connection via VoIPDB (for a call within an own Unit [Standard SIP Terminal-D^{term}IP])



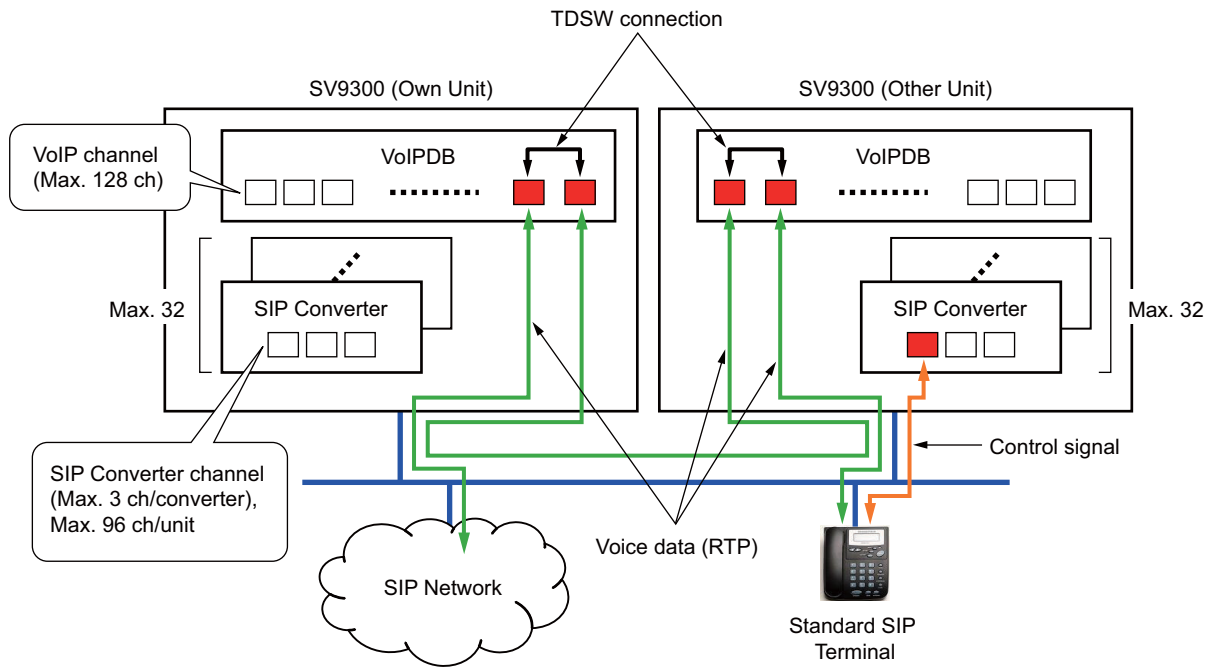
- (b) For a connection via VoIPDB (for a call between an own Unit [Standard SIP Terminal] and other Unit [Standard SIP Terminal])



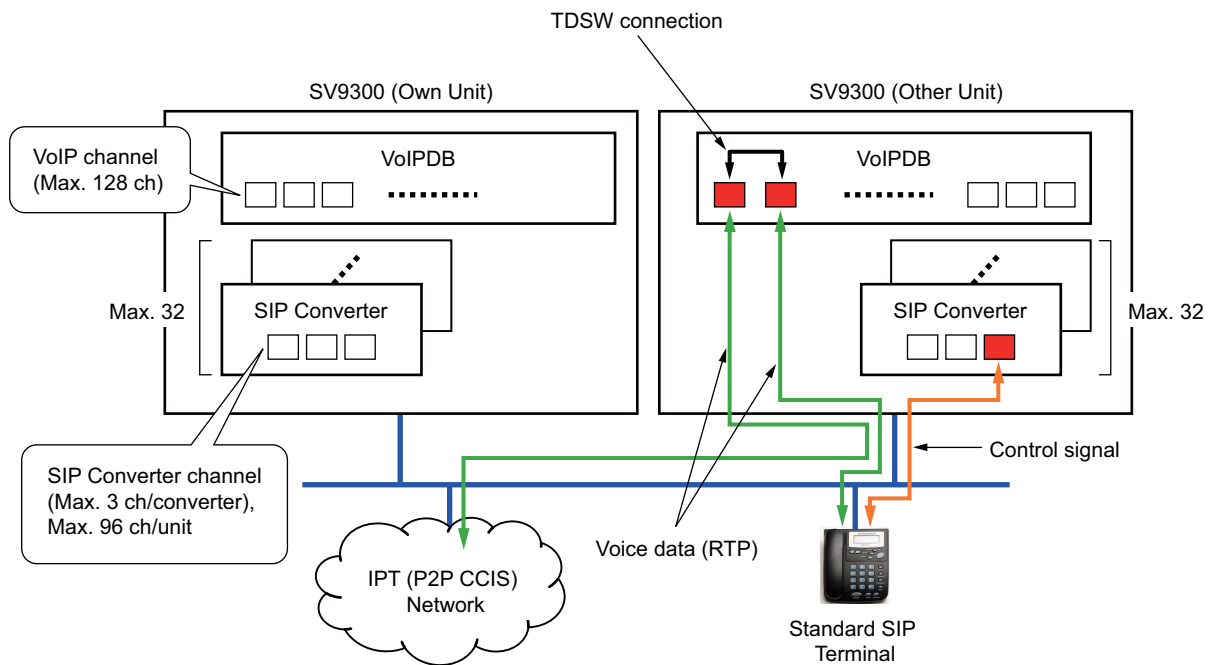
- (c) For a connection via VoIPDB (for a call between an own Unit [Standard SIP Terminal] and other Unit [D^{term}IP/TDM Terminal])



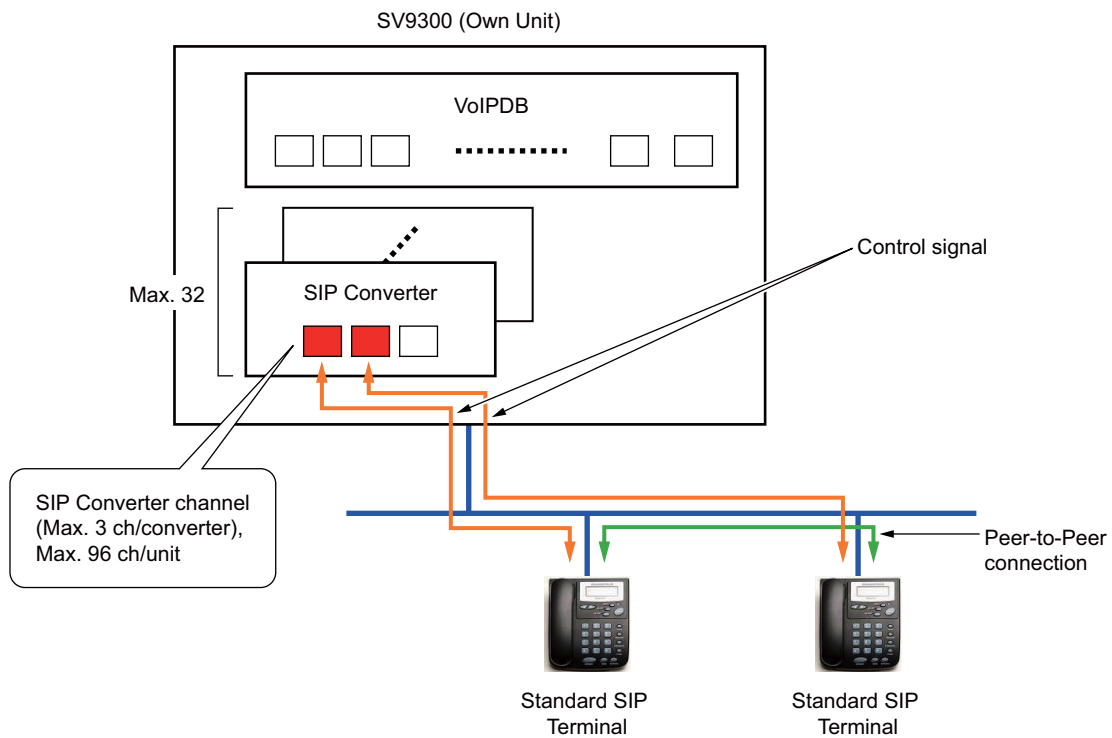
- (d) For a connection via VoIPDB (for a call between an own Unit [IP trunk] and other Unit [Standard SIP Terminal])



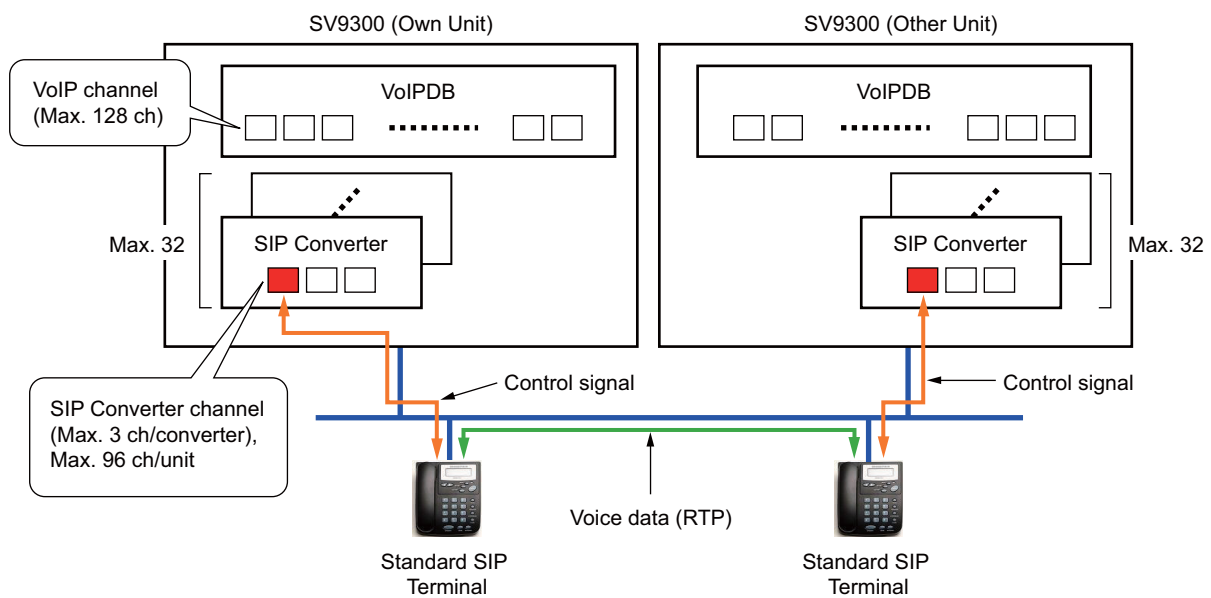
- (e) For a connection via VoIPDB (for a call between an own Unit [IPT (P2P CCIS)] and other Unit [Standard SIP Terminal])



- (f) For a Peer-to-Peer connection (for a call within an own Unit [Standard SIP Terminal-Standard SIP Terminal])



- (g) For a Peer-to-Peer connection (for a call between an own Unit [Standard SIP Terminal] and other Unit [Standard SIP Terminal])



Conditions of SIP Converter Channels

- One SIP Converter channel is used while the Standard SIP Terminal is operated as follows.
 - From start up of the Standard SIP Terminal to the completion of start up (on SIP message, until completion of the initial registration).
 - From call origination/termination to the completion of call (on SIP message, from Initial INVITE receiving/sending to 200 OK receiving/sending for BYE).
- The number of SIP converter channels used by the connection configurations of Standard SIP Terminal is as follows.

(a) To connect within a same Unit:

		Unit A (A=01-50)					
		TDM Terminal	D ^{term} IP/ D ^{term} SP30	DT700/ DT800/ DT900 Series/ SP350	Standard SIP Terminal	SIP Trunk	IPT (P2P CCIS) NOTE
Unit A (A=01-50)	Standard SIP Terminal	1 channel	1 channel	1 channel	2 channels	1 channel	1 channel

NOTE: IPT (P2P CCIS) can be accommodated in the Unit01 only.

(b) To connect between an own Unit and other Unit (when Standard SIP Terminal is accommodated in an own Unit):

UA: Unit A UB: Unit B

		Unit B (B=02-50)				
		TDM Terminal	D ^{term} IP/ D ^{term} SP30	DT700/DT800/ DT900 Series/ SP350	Standard SIP Terminal	SIP Trunk
Unit A (A=01-50)	Standard SIP Terminal	UA: 1 channel UB: None	UA: 1 channel UB: None	UA: 1 channel UB: None	UA: 1 channel UB: 1 channel	UA: 1 channel UB: 1 channel

- (c) To connect between an own Unit and other Unit (when Standard SIP Terminal is not accommodated in an own Unit):

UA: Unit A UB: Unit B

		Unit A (A=01-50)				
		TDM Terminal	D ^{term} IP/ D ^{term} SP30	DT700/DT800/ DT900 Series/ SP350	SIP Trunk	IPT (P2P CCIS) NOTE
Unit B (B=02-50)	Standard SIP Terminal	UA: None UB: 1 channel	UA: None UB: 1 channel	UA: None UB: 1 channel	UA: None UB: 1 channel	UA: None UB: 1 channel

NOTE: IPT (P2P CCIS) can be accommodated in the Unit01 only.

- If there is no idle SIP Converter channel, Standard SIP Terminal may not be started up and a call by Standard SIP Terminal may not be available.

Conditions of VoIPDB Channels

- Since Standard SIP Terminal is always connected via VoIPDB of the accommodated Unit regardless of the connected party (except Peer-to-Peer connection), SV9300 always uses an one VoIPDB channel for Standard SIP Terminal and an one VoIPDB channel for a connection party. However, when a connection party is a TDM terminal, VoIPDB channel is not used for the connection party.
- When you call to a TDM terminal accommodated in an other Unit (Unit02-50), SV9300 uses an one VoIPDB channel for each Unit for the call path connection between an Own Unit (Unit01) and the other Unit (Unit02-50). In addition, when the terminal on the other Unit (Unit02-50) is an IP terminal, the VoIPDB channel on the other Unit (Unit02-50) is not used (the VoIPDB channel on the Own Unit is used).
- For Standard SIP Terminal-to-Standard SIP Terminal call between Units, four VoIPDB channels are seized because the terminal other than Standard SIP Terminal may response before the call starts.
- When SV9300 recognized that a call is between Standard SIP Terminals after the call starts, the call is connected by Peer-to-Peer. Seized VoIPDB channels are released and no VoIPDB channel is consumed for the call.

- The number of VoIPDB channels used by the connection configurations of Standard SIP Terminal is as follows.

(a) To connect within a same Unit:

		Unit A (A=01-50)					
		TDM Terminal	D ^{term} IP/ D ^{term} SP30	DT700/ DT800/ DT900 Series/ SP350	Standard SIP Terminal	SIP Trunk	IPT (P2P CCIS) NOTE
Unit A (A=01-50)	Standard SIP Terminal	2 channels	2 channels	2 channels	2 channels	2 channels	2 channels

NOTE: IPT (P2P CCIS) can be accommodated in the Unit01 only.

(b) To connect between an own Unit and other Unit (when Standard SIP Terminal is accommodated in an own Unit):

UA: Unit A UB: Unit B

		Unit B (B=02-50)				
		TDM Terminal	D ^{term} IP/ D ^{term} SP30	DT700/DT800/ DT900 Series/ SP350	Standard SIP Terminal	SIP Trunk
Unit A (A=01-50)	Standard SIP Terminal	UA: 2 channels UB: 1 channel	UA: 2 channels UB: None	UA: 2 channels UB: None	UA: 2 channels UB: 2 channels NOTE	UA: 2 channels UB: 2 channels

NOTE: IPT (P2P CCIS) can be accommodated in the Unit01 only. For a Peer-to-Peer connection, a VoIPDB channel is not consumed (0 channel is applied).

- (c) To connect between an own Unit and other Unit (when Standard SIP Terminal is not accommodated in an own Unit):

UA: Unit A UB: Unit B

		Unit A (A=01-50)				
		TDM Terminal	D ^{term} IP/ D ^{term} SP30	DT700/DT800 / DT900 Series/ SP350	SIP Trunk	IPT (P2P CCIS) NOTE
Unit B (B=02-50)	Standard SIP Terminal	UA: 1 channel UB: 2 channels	UA: None UB: 2 channels	UA: None UB: 2 channels	UA: 2 channels UB: 2 channels	UA: None UB: 2 channels

NOTE: IPT (P2P CCIS) can be accommodated in the Unit01 only.

- If there is no idle VoIPDB channel, Standard SIP Terminal operates as follows.
 - For call origination from Standard SIP Terminal: Error response 503 is sent
 - For call termination to Standard SIP Terminal : Calls are not terminated

Conditions for IM (Instant Message) between Standard SIP Stations

IM (Instant Message) between Standard SIP Stations is available.

- This function is available with only Standard SIP stations equipped with IM function.
- IM to/from the station other than Standard SIP station, such as multi-line terminal, is not supported.
- To use this function, any license is not required, and any office data setting is not required.
- Characters (characters, symbols, etc.) and maximum number of characters to be used depends on the specification of the Standard SIP station.
- Regardless of the status or settings of the destination Standard SIP station, IM is sent.
- IM transmitting to Standard SIP station without IM function fails.
- IM transmitting to Standard SIP station that is not logged in fails.
- IM transmitting to stations other than Standard SIP station or to trunk fails.

Conditions for Call Back from Call History of Standard SIP Terminal

[9300V3 software required]

- When the call is terminated to a Standard SIP Terminal, by notifying calling number with a trunk access code, you can call back to the called party from the call history.
- The following table shows the availability of kinds of trunks that can be added a trunk access code to a calling number.

×: Available Δ: Partly available -: Not available

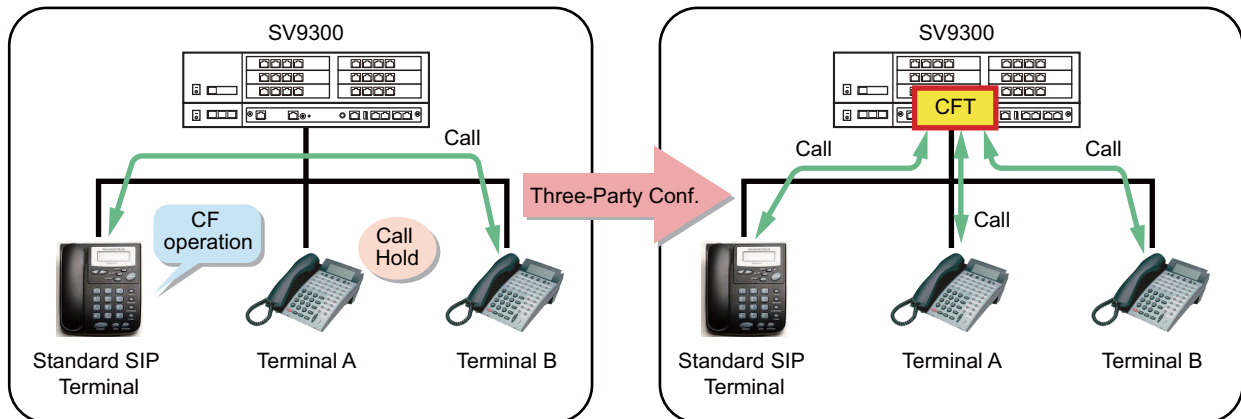
Kind of Trunk	Availability	Remarks
COT	×	Number Display
ODT	-	
LDT	-	
DID	-	
ISDN (BRI)	×	
ISDN (PRI)	×	
Q-SIG	×	
T1 1.5Mbps	-	
E1 2Mbps	-	
MFC	×	
CCIS (CCT)	×	
IPT (P2P CCIS)	-	
SIP Trunk	Δ	Not available for Point-to-Multipoint connection

- When adding a trunk access code, the following system data setting is required.
 - CM13 Y=87: 0 (Adding trunk access code to calling number)
 - CM35 Y=189: X-XX (Trunk access code to be added)
- A maximum of two digits can be added for a trunk access code.
- A trunk access code is not added when the call is originated with no CLI from the trunk of calling party or by Point-to-Multipoint connection (CMA7 Y=46: 0) of SIP trunk.
- A calling number that can be notified is up to 24 digits including a trunk access code. If the calling number exceeds 24 digits, a trunk access code is not added.
- For an incoming call to a Standard SIP Terminal, a trunk access code is not included for the calling number notified by SMDR or OAI. For a call back, a trunk access code is included for the called number notified by SMDR or OAI.

Conditions of Three-Party Conference

[9300V3 software required]

- Three-Party Conference is available using Conference Trunk (CFT) in SV9300 when you perform a conference operation from a Standard SIP Terminal during consultation hold.
- * A conference operation differs by each terminal.



- Call Hold and Call Transfer features cannot be used from Standard SIP Terminal during Three-Party Conference.
- CODEC of the Standard SIP Terminal during Three-Party Conference is applied the one used just prior to the Three-Party Conference.
For example, CODEC of the Standard SIP Terminal for the following case is G.729.
 - CODEC for a held call of Standard SIP Terminal: G.711 μ -low
 - CODEC for a call with Standard SIP Terminal is in progress: G.729
- If the operator using Standard SIP Terminal is the host of the conference, a Four-Party Conference cannot be used.
- To restrict the Three-Party Conference by the operation of Standard SIP Terminal, set the second data of CM13 Y=88 to 0.
- If the terminal put on hold is released during consultation hold, the conference does not switch to a Three-Party Conference even if conference operations from the Standard SIP Terminal are performed.
- The LCD indication of Standard SIP Terminal during a Three-Party Conference complies with the specification of the Standard SIP Terminal.
- Three-Party Conference is not available when a Standard SIP Terminal is used by a Peer-to-Peer connection.

- The following table shows the type of terminals that can be used the Three-Party Conference by the operation from a Standard SIP Terminal.

×: Available -: Not available

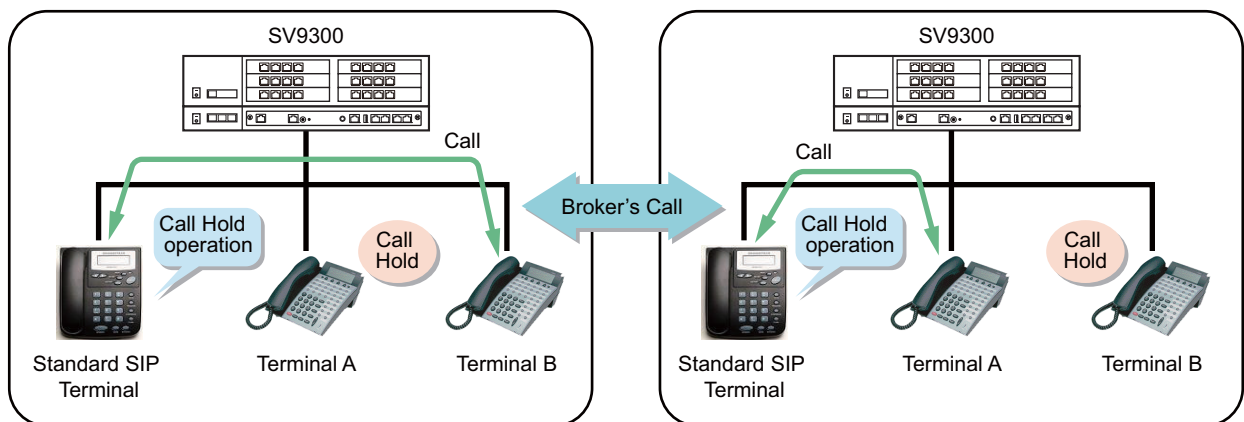
Type of Terminals	Availability
Single Line Telephone	×
DT300/DT400/DT500 Series	×
D ^{term}	×
DT700/DT800/DT900 Series	×
D ^{term} IP	×
Soft Phone	×
Standard SIP Terminal	×
SIP Wireless Terminal (MH240)	×
ISDN Terminal	-
Virtual Station	×
Attendant Console	-
Mobile Phone (Mobility Access)	×

Conditions of Broker's Call

[9300V3 software required]

- You can switch the Call Hold side and the Call side freely when you perform a Broker's Call operation from a Standard SIP Terminal during consultation hold.

* The way of operation differs by each terminal.



* You can connect to the line before the switch when you perform a Call Hold operation again.

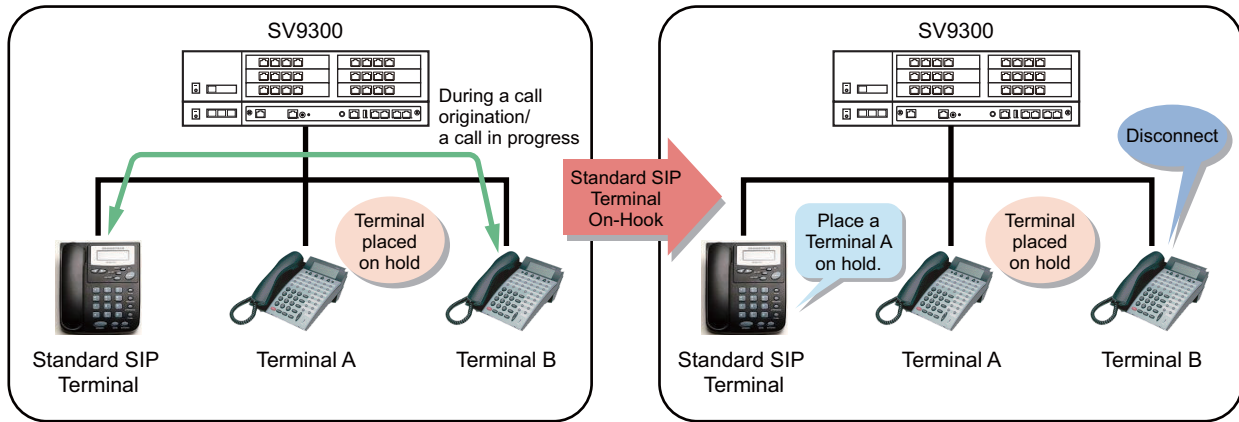
- To restrict the Broker's Call by the operation of a Standard SIP Terminal, set the second data of CM13 Y=89 to 0.
- Broker's Call is not available when a Standard SIP Terminal is used by a Peer-to-Peer connection.
- The following table shows the type of terminals that can be used the Broker's Call by the operation from a Standard SIP Terminal.

×: Available -: Not available

Type of Terminals	Availability
Single Line Telephone	×
DT300/DT400/DT500 Series	×
D ^{term}	×
DT700/DT800/DT900 Series	×
D ^{term} IP	×
Soft Phone	×
Standard SIP Terminal	×
SIP Wireless Terminal (MH240)	×
ISDN Terminal	-
Virtual Station	×
Attendant Console	-
Mobile Phone (Mobility Access)	×

Conditions for Operation When Standard SIP Station with a Held Call Goes On-Hook
[9300V3 software required]

- When a Standard SIP station is in a status with a held call and goes on-hook while an outgoing call is originated or a call is in progress, you can select the following operations at that time: Hold Transfer, or disconnection only for the outgoing call or the call in progress.
- * The way of operation differs by each terminal.



- When a Standard SIP station with a held call goes on-hook while an outgoing call is originated or a call is in progress, the operations are as follows depending on the system data settings.

System Data Settings		Operations
CM08>1047: 0	CM13 Y=93: 1	An outgoing call is abandoned or a call in progress is disconnected.
CM08>1047: 1	-	Hold Transfer
CM08>1047: 0	CM13 Y=93: 0	

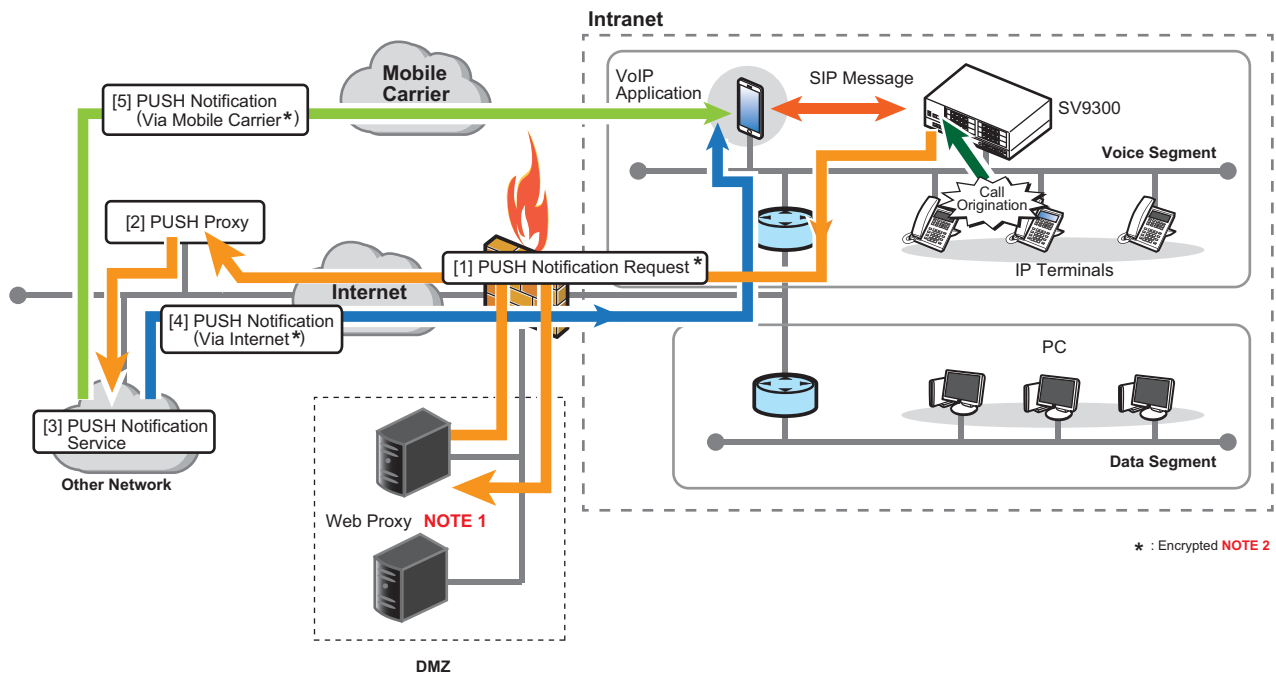
Conditions for PUSH Notification

[9300V6 software required]

- This service is a function to request PUSH Notification service to send the PUSH Notification to ST500/ST450 (hereafter “VoIP Application”) from the SV9300.
- The PUSH Notification Request is made via the PUSH Proxy to the PUSH Notification Service provided by the provider. The PUSH Notification Service delivers the information with PUSH Notification to VoIP application, which is running in the background, on the mobile device such as a smartphone.
- The VoIP Application running in the background is placed in the sleep state. Therefore, it cannot receive incoming events by SIP message. However, when the PUSH Notification is delivered by the PUSH Notification Service, it is possible to temporarily process the SIP message. The SV9300 requests the PUSH Notification Service to send PUSH Notification when sending the incoming event by SIP message to the VoIP Application.
- Operation for terminating an incoming call to the VoIP Application
 - (1) When the SV9300 receives a call request to a VoIP Application, it sends PUSH Notification Request (HTTPS) to the PUSH Proxy before calling the VoIP Application.
 - (2) PUSH Proxy sends PUSH Notification Request to a specified PUSH Notification Service when the PUSH Proxy receives the PUSH Notification Request from the SV9300. And then, PUSH Notification Service sends the PUSH Notification to the VoIP Application.
 - (3) When the VoIP Application receives the PUSH Notification, it updates its registration to the SV9300. **NOTE**
 - (4) At the same time when the SV9300 receives the registration update, it starts calling the VoIP Application to terminate the incoming call. **NOTE**

NOTE: *The actions of VoIP Application which receives the PUSH Notification differ depending on the version of the application or the device OS. For the details, refer to the manuals of each terminal.*

- With this service, the SV9300 sends a PUSH Notification Request for delivering an incoming call or sending a MWI (Message Waiting Indication: RFC3842) notification to the VoIP Application.



- NOTE 1:** Whether to use a Web Proxy (Web Proxy server) can be set by CM0B Y=1XX>165. When a Web Proxy is not used (CM0B Y=1XX>165: 3), the PUSH Notification Request is directly sent from SV9300 to a PUSH Proxy.
- NOTE 2:** When communicating SIP messages and communication packets (RTP) via Wi-Fi, the PUSH Notification must be used the encrypted communication via Mobile Carrier network specified by NEC.

ITEM	DESCRIPTION	REMARKS
[1] PUSH Notification Request	<ul style="list-style-type: none"> • A message that is sent from the SV9300 to the PUSH Notification Service via the PUSH Proxy, requesting for sending a PUSH Notification to the application on mobile devices. 	
[2] PUSH Proxy	<ul style="list-style-type: none"> • A server that relays the PUSH Notification Request to the specified PUSH Notification Service upon receiving the request from the SV9300. 	<ul style="list-style-type: none"> • An Internet-based server

ITEM	DESCRIPTION	REMARKS
[3] PUSH Notification Service	<ul style="list-style-type: none"> • A service that sends the PUSH Notifications to an application on mobile devices. the PUSH Notification Service to be used depends on the device OS. 	<ul style="list-style-type: none"> • An Internet-based server is used to send a PUSH Notification to the application. • For Android FCM (Firebase Cloud Messaging) • For iOS APNs (Apple Push Notification service)
[4] PUSH Notification (Via Internet)	<ul style="list-style-type: none"> • A message that is sent to the application on the mobile devices from the PUSH Notification Service. • The application can receive the notification even if it is not active on the device. 	
[5] PUSH Notification (Via Mobile Carrier)	<ul style="list-style-type: none"> • If a SIM card is installed in the terminal and it is connected to the mobile phone carrier network, a message that is sent to the application on the mobile devices from the PUSH Notification Service. • The PUSH notification via the carrier network takes priority over the Internet. • If the terminal and the carrier network are not connected, PUSH notification is sent via the Internet. 	

System Requirements

- The table below shows the requirements for using this service.

ITEM		REQUIREMENTS
VoIP Application	Supported OS	<ul style="list-style-type: none"> iOS: V11 or later NOTE 1 Android: as per the ST500 (Android) version
	Supported application version	<ul style="list-style-type: none"> For PUSH Notification <ul style="list-style-type: none"> ST500 (Android): V7.0 or later [9300V8 software required] ST500 (iOS): V2.0 or later ST450: V8.0.0 or later For Representative Call Notification (PUSH Notification) <ul style="list-style-type: none"> ST500 (Android/iOS): V7.0 or later [9300V8 software required]
SV9300 Software Version		9300V6 (software rev.1) or later NOTE 2
PUSH Proxy		<ul style="list-style-type: none"> usa01.nec-pushproxy.com (for North America) nld01.nec-pushproxy.com (for EMEA) aus01.nec-pushproxy.com (for Australia) other01.nec-pushproxy.com (for other markets except China)
PUSH Notification Service		<ul style="list-style-type: none"> For Android Firebase Cloud Messaging (FCM) For iOS Apple Push Notification Service (APNs)

NOTE 1: *When the terminal's OS is iOS11 or later and the PUSH Notification service is not used, the operation of an incoming call notification from the SV9300 to the VoIP application that is running in the background is not guaranteed.*

NOTE 2: *The system version license of 9300V6 or later is also required.*

Service Conditions

<Important Notice>

- When the SV9300 is in the following states, the SV9300 requests a re-registration to VoIP applications by sending the PUSH Notification. **[9300V9 software required]**
 - When a system reset is performed and a system starts operations
 - When a changeover to survival mode occurs in the Remote Unit and a system starts operations
 - When a system changeover occurs in dual CPU system and a system starts operations
 - When a mode changeover occurs in failover mode and a system starts operations **NOTE 1**

NOTE 1: *To perform a re-registration to VoIP applications in a failover system, set the destination of the PUSH Notification sent to the Primary and Secondary Unit to the same settings.*

NOTE 2: *When using 9300V8 software or before, and the VoIP application with PUSH Notification enabled is in logout state due to the above conditions, it cannot receive incoming calls. To receive incoming calls, you have to set the VoIP application to foreground once or restart the VoIP application to register it to the SV9300.*

- Re-registration to SV9300 is performed when user confirms the indication of PUSH Notification and starts up the VoIP application. **[9300V9 software required]**
- To request the re-registration to the VoIP application with the PUSH Notification, it is necessary to register the VoIP application once to SV9300. **[9300V9 software required]**
- It is recommended to set the data for the services such as Station Hunting/Announcement/Call Forwarding- Not Available when the Standard SIP terminal is unable to receive a call.

<Basic Conditions>

- The terminals where the VoIP Application runs will receive the PUSH Notifications from the PUSH Notification Service. Therefore, each terminal must contain a SIM card or be able to access the Internet via Wi-Fi. It is recommended to use the terminal with SIM card even when using Wi-Fi. (It may take time until become available when the terminal moves to wireless radio range from outside. Also, depending on the quality of the network, incoming calls may not be received, therefore it is required to be careful for studying the system and network configuration.)
If a SIM card is installed in the terminal and it is connected to the mobile phone carrier network, the PUSH notification via the carrier network takes priority over the Internet. If the terminal and the carrier network are not connected, PUSH notification is sent via the Internet.
- The SV9300 requires Internet connection because the PUSH Proxy runs on an Internet-based server. For security reasons, it is strongly recommended that a firewall be set up between the SV9300 and the PUSH Proxy. Refer to the “For IP Station” of the TCP/UDP/RTP PORT NUMBER CONDITIONS. [📄 Page C-3](#)
- To access the PUSH Proxy, FQDN for the PUSH Proxy must be configured on the SV9300 and setting the address for the DNS server (primary/secondary) is required for IP address resolution.

- This service feature is only available for situations in which the PUSH Notification can be sent to the VoIP Application via the PUSH Proxy connected to the SV9300.
- The VoIP Application must complete the communication with the PUSH Notification Service and register itself to the SV9300 in advance of receiving a PUSH Notification.

NOTE: *The registration information contains the application ID and the device token which are required for the PUSH Notification.*

- Application ID: An ID that uniquely identifies the application to send the PUSH Notification

- Device token: An identifier to send the PUSH Notification (for Android, this indicates a registration token)

- In this service, the SV9300 uses the PUSH Notification Service such as FCM/APNs via the PUSH Proxy. It is the service that depends on external network service and cannot be involved in troubles with the following factors:
 - Network path failure
 - PUSH Proxy failure
 - PUSH Notification Service failure
 - Communication failure between PUSH Notification Service and the mobile deviceFor this reason, out of the obstacles found in the message on the SV9300 side, it is necessary to deal with the network path/external network service side rather than the SV9300.
- Unlike the other SIP terminals, the VoIP application with PUSH Notification enabled does not periodically send REGISTER update to the SV9300. But the SV9300 maintains the login state of the VoIP application even if it does not receive REGISTER update. As a result, even if the VoIP application is out of service area or unusable state, it will not be logged out and the license will remain consumed.
- This feature is not available for the following Standard SIP stations.
 - Standard SIP stations that do not support this feature
 - VoIP Application version that does not support this feature
 - VoIP Application on which the PUSH Notification Service is not set to enable

<Conditions for Distinctive Ringing on ST500>

[9300V9 software required]

- When an internal/external incoming call and a Nurse call (routine/emergency call) arrives to a VoIP application, you can assign to each station whether to provide Caller ID, Name and Ringer Tone Pattern for PUSH Notification (assigned by CM13 Y=102).
- To use this service, the System Version License for 9300V9 is required.
- This service is only supported for the VoIP application (ST500 Ver.7.0 or later) for iOS/Android.

- The ringing tone added to a PUSH Notification is used to distinguish routine/emergency calls for nurse calls. The ringing tone added to an internal or an external call depends on the setting of phone-book registered to a smartphone.

<Conditions for Terminal Accommodation>

- The VoIP Applications using the PUSH Notification Service and the ones not using the service/Standard SIP terminals can be accommodated together under the SIP converter in the same Unit.
- Use “Standard SIP Station Connection Status (CM12 Y=96)” to identify if a station is a VoIP application using the PUSH Notification Service or the one not using the service/Standard SIP terminal.
- While a VoIP Application with PUSH Notification service enabled is logged in to the system, if another VoIP Application without the service enabled/unsupported by this service logs in with the same station number, an override automatically occurs. The overridden VoIP Application, the one with the service enabled, will be logged out and not be able to use this service feature.
- A VoIP Application can use only one station number when it logs in. It cannot use multiple profiles for multiple logins at the same time. Only the profile used for the latest login is effective.
- For the VoIP Application using the PUSH Notification Service, periodic registration monitoring is not performed. The data specified in “Periodic Registration Timer for Standard SIP station (CM41 Y=0>151/152/153 (Default: 1 hour))” is not reflected, and the VoIP Application is handled as perpetual login. To make it log out, clear the data (enter CCC) in “Standard SIP station connection status CM12 Y=96” on the PCPro.
- During the period after a call is originated to a VoIP Application using the PUSH Notification Service before the VoIP Application receives the PUSH Notification and transitions to the foreground, Ring Back Tone can be sent to the calling party. Set this in CM08>503 or CM35 Y=368.

<Conditions for External Network Connection>

- Configure the settings for external network connection only when the VoIP Applications that require the PUSH Notification Service are accommodated in the system.
- For external network connection, DNS resolution using the FQDN (Fully Qualified Domain Name) for the PUSH Proxy is performed.
- Only DNS resolution, translating a domain name to an IP address is supported. If the DNS server returns more than one IP address for one domain name, only the first IP address is used.
- A DNS resolution is performed at the following timings:
 - When system reset is performed
 - When Time-to-Live (TTL) in the DNS expires
 - When DNS cache table is cleared by the system data (CM0B Y=1XX>164)
- The DNS resolution data is always cached in the system. The expiration for the cache depends on the Time-to-Live (TTL) sent from the DNS server on DNS resolution. If the DNS resolution cannot be updated for reasons such as DNS server failure, the existing cache data is used for continuous operation.

- The PUSH Notification Requests are sent from the Unit where the VoIP Applications log in. Therefore, configure the DNS server settings for all the Units where the VoIP Applications will log in. You can set primary DNS and secondary DNS server address for each Unit.
- The maximum number of sessions for connection between SV9300 and PUSH proxy is 128 per unit.

<Conditions of External Network Connection Error>

- The SV9300 does not monitor the connection status with the PUSH Proxy. Instead, when a connection error occurs on sending a PUSH Notification Request, it stores the fault information and outputs the external alarm (NJ/MN) in accordance with the data set in the system data (CMEA Y=2>132). Thereafter, the SV9300 stores fault information (IP network error restoration) and outputs the external alarm (NJ/MN) according to the setting of system data (CMEA Y = 2>133) when a normal response is returned from the PUSH proxy in response to the PUSH Notification Request from the SV9300.

Example of Fault Information Display

Index	Date Time	FK	Alarm	UNIT-SLOT-CKT	Detail
1	31/05/18 14:51:08	13 2	MJ	01	32 01 00 00 18 05 04 13 - 23 38 00 00 00 00 00 00 01 FF FF 00 02 02 01 FF - FF 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 - 00 00 00 00 00 00 00 00
2	31/05/18 15:23:22	13 3	MJ	01	33 01 00 00 18 05 04 13 - 23 38 00 00 00 00 00 00 01 FF FF 00 02 02 01 FF - FF 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 - 00 00 00 00 00 00 00 00

NOTE: Binary data only is displayed in “Detail” column, which is used for analyzing the fault.

- As a result of DNS inquiry, if an error response is returned from the DNS server, the following processing is executed.
 - If an error is received by DNS inquiry, retry will be performed twice.
 - If error responses except for NAME ERROR is received after the above retrying, DNS inquiry will be executed again after 60 minutes.

NOTE: If the NAME ERROR “the domain name referenced in the query does not exist” is returned in response to a DNS query, DNS queries will not be sent after that. To perform a DNS query again, clear the DNS cache table (CM0B Y=1XX>164) or reset the system.

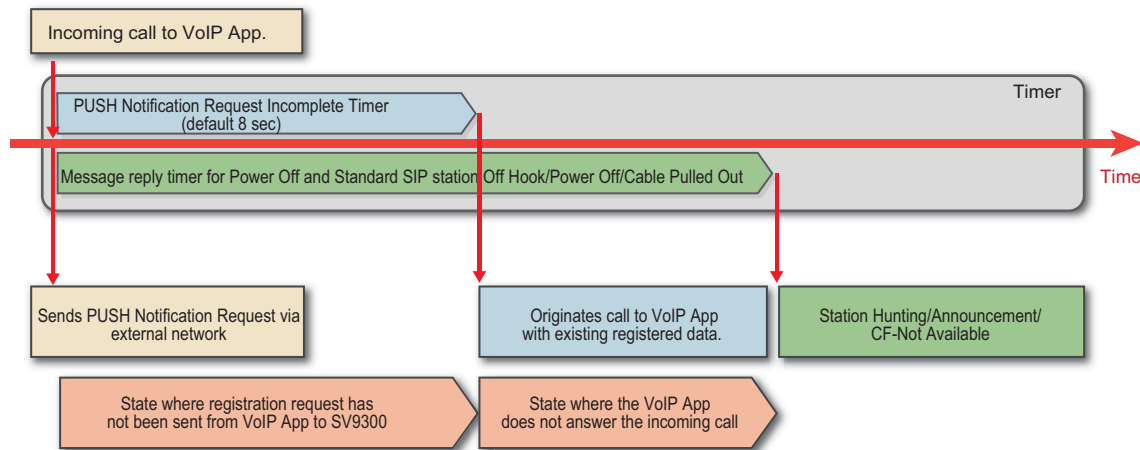
- In the following cases, the SV9300 sends the incoming call message (INVITE) to the VoIP Application using the registration data of the VoIP Application in the system.
 - When the IP address of PUSH Proxy cannot be received due to DNS unresolved.
 - When an error response is received from PUSH Proxy for the PUSH Notification Request.

– When the SV9300 does not receive registration request from the VoIP application within “PUSH Notification Request Incomplete Timer (CM41 Y=0>223 (8 sec. by default)),” after sending the PUSH Notification Request to the PUSH Proxy.

- When the Standard SIP terminal is unable to receive a call due to the power being off or being out of service area, services such as Station Hunting/Announcement/Call Forwarding-Not Available will function according to the system data settings. If these services are used, the timer setting of "CM41 Y=0>85" should be set longer than that of "CM41 Y=0>223 (default=8sec.)".

e.g.) CM41 Y=0>223 : PUSH Notification Request Incomplete Timer (default=8 sec.)

CM41 Y=0>=85 : Message reply timer for Power Off and Standard SIP station Off Hook/Power Off/Cable Pulled Out: 12-28 sec. (must be longer than 8 sec.)



<Relationship with Other Services>

- The SV9300 can send the PUSH Notification Requests even under the following conditions: the Remote Unit is in the survival mode, CPU1 is active in dual mode system, and the system is in failover mode.
- When accommodating the VoIP Applications in the Primary Unit in a failover system, configure the PUSH Notification Request destination settings for the Secondary Unit, too.

Conditions for Representative Call Pick-Up on ST500

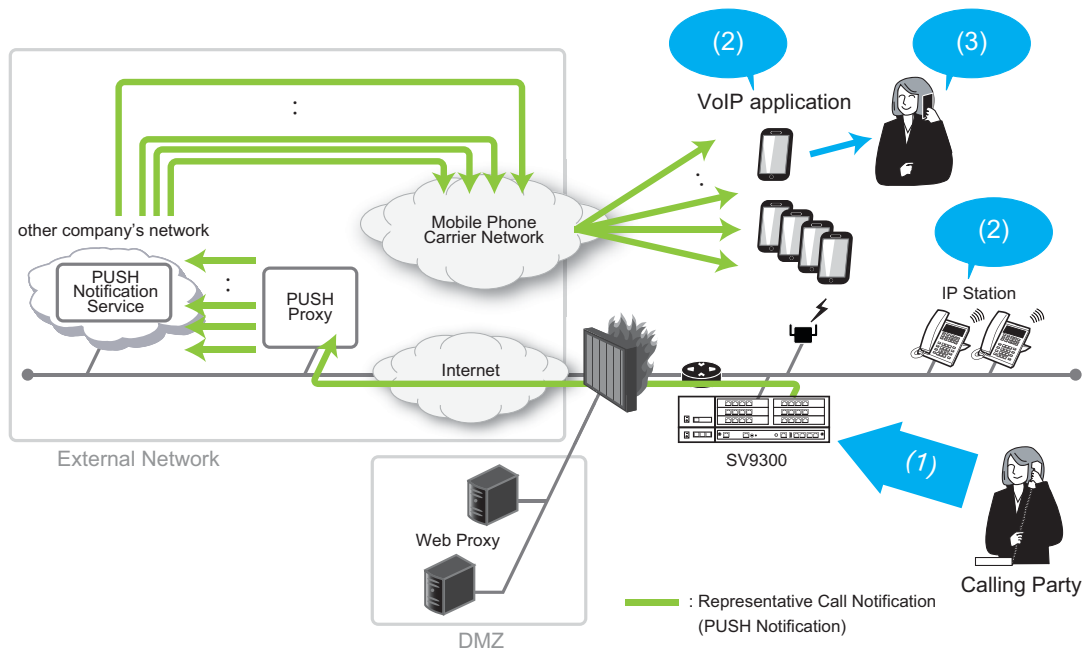
[9300V9 software required]

- This feature is the function to send a Representative Call PUSH Notification (hereafter “Representative Call Notification”) to VoIP applications (ST500) assigned by the system data in advance for the representative call group of the group representative station, if an incoming call arrives to the destination (group representative station) of a representative number from a station or a trunk.

The VoIP application received the Representative Call Notification displays an incoming call notification on the smartphone and answers the incoming call of the group representative station when a user do the answer operation.

NOTE: This feature uses the PUSH Notification feature. For the conditions of the PUSH Notification feature, refer to the “Conditions for PUSH Notification”. [☞ Page 1-78](#)

- The flow of the Representative Call Pick-up on ST500 is shown as follows.



(1) An incoming call arrives to the representative number

When an incoming call arrives to the group representative station, a Representative Call Notification is sent to the VoIP application assigned to the group representative station in advance.

- (2) For IP Stations, the line of the representative number flashes. For a VoIP application, the notification that the incoming call has been received is displayed on the VoIP application or in the notification area on the device OS.

(3) To answer the call to the group representative station, tap the notification.

After the call answer, the system deletes the notification from the VoIP applications other than answered line (when a Representative Call Notification on the VoIP application is displayed in fix, the notification displayed on the VoIP application is grayed out).

- The following terms are used in this feature.

TERMS	DESCRIPTION
Group Representative Station	Virtual station that assigns as the destination for a representative number. When an incoming call arrives to the group representative station, sends the Representative Call Notification to multiple VoIP applications assigned in advance.
VoIP Application	Application software that provides the extension features for smartphones and can be used as a station accommodated in the SV9300. In this feature, it means ST500.
Representative Call Notification	PUSH notification that notifies the Representative Call Notification to the VoIP applications.

Basic Conditions

- To use this feature, System Version License for 9300V9 is required.
- This feature supports only for the VoIP application (ST500 Ver7.0 or later) for iOS and Android. Also, it is necessary to make the PUSH notification setting valid on the screen of the VoIP application.

NOTE: *As for the operation of the VoIP application, refer to the operation manual of the application.*

- If there is a significant difference between the set time and the current time of a VoIP application, the Representative Call Notification may not be sent to the VoIP application. Set the time of the VoIP application accurately.
- This feature can also be used in the Wi-Fi network.
- For port numbers to be used by the Representative Call Notification to the PUSH Proxy, you can use 128 ports from the base port assigned by CM0B Y=1XX>182 (default: 55500-55627).
- For the following cases, a setting of the DNS feature is required for a DNS resolution of PUSH proxy/Web proxy FQDN.
 - When not using a Web Proxy (assigned by CM0B Y=1XX>165: 3)
 - When using a Web Proxy (Domain name) (assigned by CM0B Y=1XX>165: 1)

Setting Conditions

- Assign the virtual station (CM11) for the destination (group representative station) of the group representative number.
- The following table shows the number of ST500 registrations and group registrations.

ITEMS	NUMBER OF REGISTRATION	REMARKS
The number of ST500 registrations	Max. 64 ST500 terminals/per group NOTE 1, NOTE 2	<ul style="list-style-type: none"> • Assigned by CM57 Y=39 • Assign the Unit number and the group number that send the Representative Call Notification to the group representative station (virtual station).
The number of group registrations	Max. 128 groups	<ul style="list-style-type: none"> • Assigned by CM12 Y=107 • An ST500 terminal can be assigned to two or more representative call groups (Max.128 groups) <p style="text-align: right;">NOTE 3</p>

NOTE 1: *The stations that can be sent the Representative Call Notification are only for ST500.*

NOTE 2: *As for ST500 terminals to be registered in a representative call group, accommodate all of them to the same Unit.*

NOTE 3: *For a high traffic that exceeds 16 consecutive incoming call notification to a same ST500 terminal, registration to a same ST500 station is up to 16 groups because a representative call cannot be notified to the destination normally, or you cannot answered to a representative call notification.*

- For a Representative Call Notification, you can set the group name (max. 16 characters) of each representative call group that displays on the ST500 (assigned by CM77 Y=14).

NOTE: *This feature does not support Chinese and Russian.*

Conditions for Representative Call Notification

- When an incoming call arrives to a group representative station, a Representative Call Notification is sent to a ST500 registered to the representative call group assigned for the group representative station.
- A Representative Call Notification is sent to ST500 at the timing when an incoming call arrives to a group representative station from a station or a trunk.

NOTE 1: *When a group representative station is assigned to a Multiline terminal as the sub line, if an incoming call arrives to the group representative station, the Representative Call Notification to ST500 is later than calling to the Multiline terminal.*

NOTE 2: *Depending on the network environment or the load conditions of the PUSH Proxy (AWS), Apple and Google, the Representative Call Notification may be delayed or not be informed.*

- If one minute has elapsed during calling to a group representative station, the calling to the group representative station continues but the Representative Call Notification to ST500 is automatically deleted.

NOTE: *To receive the notification continuously, prepare another representative call group same as the current group representative station and set the timer for Call Forwarding-No Answer to less than one minute. The notification is transferred from the current group representative station to another group representative station, and vice versa.*

<Conditions for Answering the Representative Call Notification>

- ST500 user can answer the representative call by tapping “Answer” of the incoming call notification.
- To answer to a Representative Call Notification, the Call Pickup-Direct feature to group representative stations is automatically used. Therefore, an access code setting for Call Pickup-Direct is required in advance.
 - CM20 Y=0-3: A021 (Call Pickup-Direct)
 - CM64 Y=28 (Access code for answering Representative Call Notification)

NOTE: *For details, refer to the “CALL PICKUP-DIRECT” of Chapter 1 BUSINESS FEATURE in the Programming Manual.*

- Even if the Call Pickup-Direct feature is manually used via a station other than representative call group during sending a Representative Call Notification, you can answer the incoming call to the group representative station.
- Behaviors and operations after answering a Representative Call Notification are same as the PUSH notification. Call histories remain in a VoIP application.
- When an incoming call arrives to a group representative station during an incoming call to a VoIP application or calling by a VoIP application, the Representative Call Notification is sent to the VoIP application but you cannot answer. Answer the Representative Call Notification after the incoming call to the VoIP application is ended or the call is released.
- If two or more user operates to answer the Representative Call Notification at the same time, user who could not answer the notification hear a reorder tone (ROT). Call histories also remain in the VoIP application even if not answered.

<Conditions for Disabling the Representative Call Notification>

- The PUSH notification for disabling the Representative Call Notification is sent for the following cases.
 - When answered by the station in the group for the incoming call to the group representative station
 - When the incoming call is abandoned by a calling party and the group representative station becomes idle state
- When the PUSH notification for disabling answer to the Representative Call Notification cannot reach ST500 due to the load conditions of the SV9300 or network congestion, the Representative Call Notification remains on the device operating ST500. To avoid this case, the remained Representative Call Notification is automatically deleted after one minute of receiving the Representative Call Notification.
- When the PUSH notification for disabling answer is received without answering the Representative Call Notification, call histories do not remain in the VoIP application.

Relationship with Other Services

- This feature can also be used for the following cases.
 - Changeover to survival mode occurs in the Remote Unit
 - System changeover occurs in the Dual CPU system
 - Mode changeover occurs in the failover system
- In the call transfer features, the Representative Call Notification is performed for the following cases.
 - When an incoming call arrives to the group representative station by Call Transfer (with attendant assistance) or Call Transfer-All Calls (without attendant assistance) from a station/trunk. **NOTE**
 - When an incoming call arrives to the group representative station by Call Forwarding-All Calls/Call Forwarding-Busy Line/Call Forwarding-No Answer from a station/trunk.

NOTE: *When an incoming call arrives to the group representative station by Call Transfer (with attendant assistance) or Call Transfer-All Calls (without attendant assistance), an intervening station number is used for the calling party number displayed for the Representative Call Notification. The intervening station number is displayed after the intervening station is released, too.*

- The conditions for the incoming call features are as follows.

INCOMING CALL FEATURES	CONDITIONS
Station Hunting	<ul style="list-style-type: none"> The Station Hunting feature can be used with this feature. When using two or more representative numbers or amount of call traffic is high, and you want to response with the same group, assign the multiple group representative stations. you can reduce the rate of busy call by grouping the Station Hunting with them.
UCD	<ul style="list-style-type: none"> The UCD feature can be used with this feature. When using two or more representative numbers or amount of call traffic is high, and you want to response with the same group, assign the multiple group representative stations. you can reduce the rate of busy call by grouping the UCD with them. <p>Furthermore, while group representative stations are all busy, the system waits until one of the group representative stations becomes idle. As soon as a group representative station becomes idle, the representative call arrives to the group representative station and the Representative Call Notification is sent to ST500 at the same time.</p>
Do Not Disturb	<ul style="list-style-type: none"> When the Do Not Disturb feature is assigned to the group representative station, a Representative Call Notification is not sent.
Group Call by Pilot Number Dialing	<ul style="list-style-type: none"> A pilot station (virtual station) for Group Call by Pilot Number Dialing cannot be used with a group representative station (virtual station).

- You cannot answer with a Representative Call Notification by using OAI (SCF). However, you can monitor incoming calls to group representative stations and answer or end calls on ST500 by SMF.
- When you answer the Representative Call Notification with ST500, the a call record can be output to the SMDR as the billing for incoming calls of ST500.
- If not answered the Representative Call Notification, the call record of abandoned incoming call to ST500 cannot be output to the SMDR.

NOTE: *When an incoming call is abandoned by a calling party, the call record of abandoned incoming call to a group representative station can be output to the SMDR.*

Maintenance Conditions

- As with a PUSH notification, the following failure information is also output for a Representative Call Notification.
 - Fault Kind Number 132: IP Network failure
 - Fault Kind Number 133: IP Network failure returned to normal condition

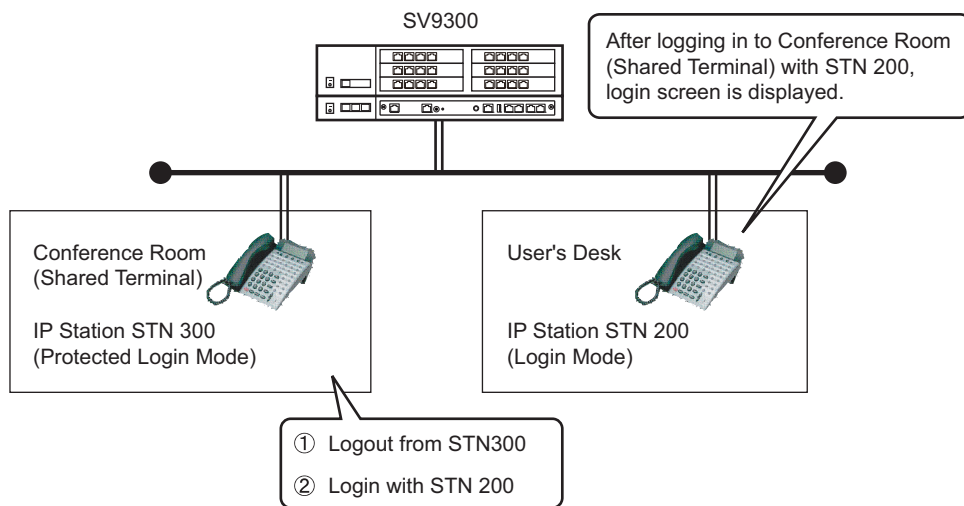
Conditions for IP Station Login/Logout

- For the IP Station login, there are three types of method as shown below.

LOGIN METHOD for IP STATION	DESCRIPTION
Automatic Login Mode (Authentication by MAC Address)	<ul style="list-style-type: none"> • IP Station is registered by MAC Address Authentication. • Used when using other terminals to login is not necessary.
Protected Login Mode	<ul style="list-style-type: none"> • IP Station is registered by Password Authentication. • Used when using other terminals to login is necessary.
Fixed Connection Mode	<ul style="list-style-type: none"> • IP Station is registered by both MAC Address and Password Authentication. • Used when using such as shared terminals (e.g. conference room phone) to login is necessary. • Automatic Login Mode is used to register IP Station usually, but with this mode, Login Mode can be used to register IP Station if needed.

NOTE: *IP station login operation is not supported for Standard SIP Terminal.*

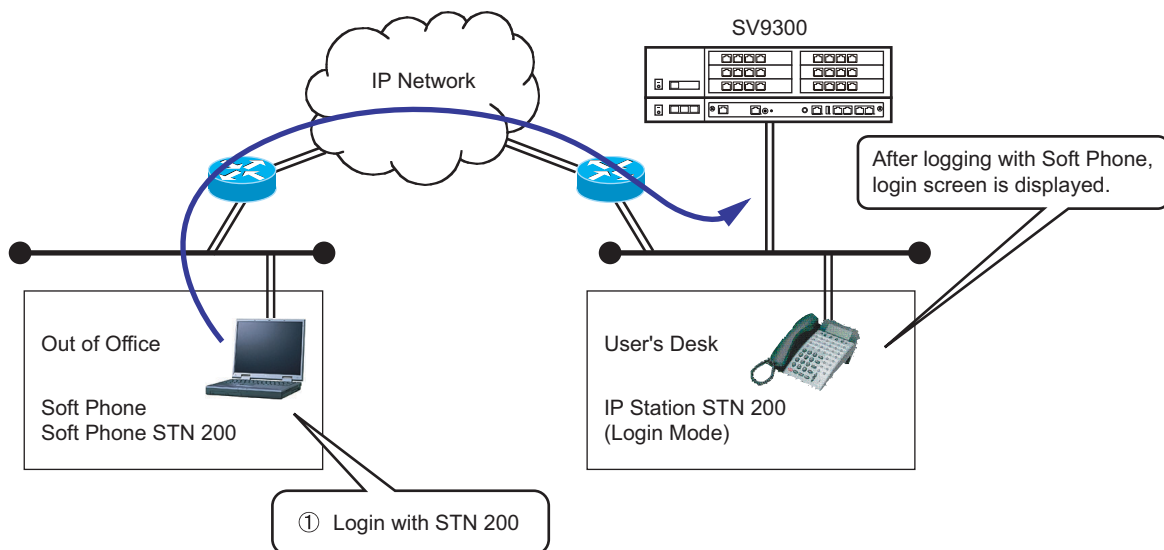
- Examples of using fixed connection mode
 - When a shared IP Station as station number 300 is used in Fixed Connection Mode in a conference room, and also the shared IP Station is used as station number 200 on the user's desk in Login Mode.
 1. Logout from the shared IP Station as station number 300.
 2. Login to the shared IP Station with station number 200 as the user's desk's station number.



NOTE: Logout from the shared IP Station to return to the former status. The shared IP Station is reconnected as station number 300 automatically. The IP Station on the user's desk is needed to login manually.

- When IP Station on the user's desk is used as station number 200 in Fixed Connection Mode, and also Soft Phone out of the office is used as station number 200.

Connect SV9300 from Soft Phone, and login to Soft Phone with station number 200.



NOTE: Logout from the Soft Phone or power off the PC to return to the former status. The IP Station on the user's desk is reconnected as station number 200 automatically.

- For the common conditions of the three types of IP Station login method are shown below.
 - Registering Protected Login Mode for Soft Phone by CM15 Y=480 is recommended.
 - When these login methods are used between the units of Remote UNIT over IP, use User Mobility simultaneously. Protected Login Mode and Fixed Connection Mode can be used on User Mobility. The terminal which has registered MAC Address for Fixed Connection Mode can not be registered to other unit.
 - To change login method, do the following operation.
 - Change from Automatic Login Mode (Authentication by MAC Address) to Protected Login Mode
 1. Change login method by CM15 Y=480.
 2. Logout or clear the MAC Address by CM12 Y=90.
 3. Login again.
 - Change from Fixed Connection Mode to Automatic Login Mode (Authentication by MAC Address)/Protected Login Mode
 1. Clear the MAC Address of Fixed Connection Mode by CM12 Y=92.
 2. Change login method by CM15 Y=480.
 3. Logout or clear the MAC Address by CM12 Y=90.
 4. Login again.

Change from Automatic Login Mode (Authentication by MAC Address)/Protected Login Mode to Fixed Connection Mode

1. Change login method by CM15 Y=480.
2. Clear the MAC Address of Fixed Connection Mode by CM12 Y=92.
3. Restart the terminal.

After changing login method, System Data Backup by CMEC Y=6>0: 0 is needed. Also, when login method of the terminal in Remote Unit is changed, System Data Copy by CMEC Y=8 is needed.

- When the terminal station number registered as Automatic Login Mode (Authentication by MAC Address)/Fixed Connection Mode is cleared, the registered MAC Address assigned by CM12 Y=90, 92 is cleared. After that, System Data Backup by CMEC Y=6>0: 0 is needed. Also, if the terminal is in Remote Unit, System Data Copy by CMEC Y=8 is needed.
- The conditions of Fixed Connection Mode are shown below.
 - IP Station and Soft Phone can be used for Fixed Connection Mode.
 - Fixed Connection Mode is not available on the conditions as follows.
 - D^{term}85 (Series i) with IP adapter
 - Terminal whose number of Line/Trunk/Feature Keys (12/16/18/24) is set different from actual one.
 - There are two ways to register MAC Address as follows.
 - Automatic MAC Address registration by CM12 Y=90: 0
 - Manual MAC Address registration by CM12 Y=92
 - The password to login in Fixed Connection Mode is set by CM2B Y=00 (same programming data as Protected Login Mode). Password setting is needed.
 - The maximum number of terminals used in Fixed Connection Mode is 256.
 - When logging out from the terminal registered to Fixed Connection Mode and logging in to the terminal with other station number, register Protected Login Mode or Fixed Connection Mode. Automatic Login Mode (Authentication by MAC Address) can not be registered.
 - When the terminal logged in with Fixed Connection Mode logs out and [Cancel] key + [Exit] key (soft key) are pressed with the login screen displaying, the terminal is reconnected with the registered station number. For Soft Phone, restarting the terminal is required after pressing [Cancel] key (soft key).
 - When logging out from the terminal registered to Fixed Connection Mode and operating the following when the terminal is logged in with other station number, the terminal is reconnected with the registered station number. The login screen is not displayed.
 - System Reset
 - IP Station setup
 - Unplugging cable and unplugging terminal detection
 - Logout

- When re-registering the terminal is needed (e.g. terminal failure), do the following operation to register MAC Address.
 - Automatic MAC Address registration by CM12 Y=90
 1. Disconnect the terminal from the network
 2. Clear the MAC Address of the terminal by CM12 Y=92: CCC and then by CM12 Y=90: CCC.
 3. Connect a new terminal to the network and login.
 4. Register a MAC Address for the new terminal by CM12 Y=90: 0.
 5. Execute System Data Backup by CMEC Y=6>0: 0. When the terminal is accommodated in a Remote Unit, execute System Data Copy by CMEC Y=8 to the Remote Unit.
 - Manual MAC Address registration by CM12 Y=92
 1. Disconnect the terminal from the network
 2. Clear the MAC Address of the terminal by CM12 Y=92: CCC and then by CM12 Y=90: CCC.
 3. Register a MAC Address for a new terminal by CM12 Y=92: XXXXXXXXXXXXX.
 4. Execute System Data Backup by CMEC Y=6>0: 0. When the terminal is accommodated in a Remote Unit, execute System Data Copy by CMEC Y=8 to the Remote Unit.
 5. Connect the new terminal to the network.

Conditions for Station-to-Station Connection

- A Peer-to-Peer connection between IP Stations is available. For details, refer to the following table.

×: Available -: Not available

	DT700/DT800/ DT900 Series	D ^{term} IP	Soft Phone	Standard SIP Terminal
DT700/DT800/ DT900 Series	×	×	×	-
D ^{term} IP	×	×	×	-
Soft Phone	×	×	×	-
Standard SIP Ter- minal	-	-	-	×

NOTE: *This communication can be used by setting the second data of CM15 Y=229 to 0 for both terminals of the call and by resetting the terminals.*

- VoIPDB is required when an IP station connects to Single Line Telephone/Multiline Terminal/Standard SIP Terminal.

Conditions for Public Network/Tie Line Network Connection

- For the IP Station communication between offices, the VoIPDB is required.
- Peer-to-Peer connection is not available in this connection.

Conditions for CCIS Connection

- In the voice communication via the Internet, the connection and communication delay may occur and the voice quality may deteriorate.
- The IPT (P2P CCIS) does not support Dynamic Host Configuration Protocol (DHCP) service.
- Peer-to-Peer connection between IP Stations via CCIS is available only when the destination office is SV9300 or 2400 IPX.
- Only one IPT (P2P CCIS) can be mounted in this system. Mount IPT (P2P CCIS) in the Unit01.
- The IPT (P2P CCIS) can be connected to a maximum of 127 trunks.
- The IPT (P2P CCIS) provides only Point-to-Multipoint connection.
- When a call over Peer-to-Peer connection via CCIS is put on hold and then answered at the same station, Elapsed Time Display returns to 0: 00: 00.
- Conditions for Link Down Notice for CCIS connection are shown below.
 - Link Down Notice is available only for Digital Multiline Terminal and IP Station accommodated in the SV9300. This is not available for a single line telephone and Attendant Console.
 - For message display, Multiline Terminal with 24-digit or more LCD is recommended. 16-digit LCD may not display all messages properly.
 - Notification message can be displayed regardless of idle or busy state of Multiline Terminal, writing the message over the present display. After six seconds, the display returns to the time display automatically.
 - The system detects a Link Down on the condition that TCP connection between offices is interrupted. The Link Down is notified to the Multiline Terminal at 15-20 seconds later from the system detects the Link Down.
 - Link Down Notice is available only for the CCIS connection via IPT (P2P CCIS). CCIS connection with CCT/DTI blade or DIO/ODT blade is not available.
- For the number of VoIPDB channels used by IPT (P2P CCIS), refer to the number of VoIPDB channels used by the connection configurations in the “Conditions for Overall System”. [Page 1-11](#)
- External Hold Tone can be sent via IPT (P2P CCIS) (however, a source of External Hold Tone must be connected to Unit01. For details, refer to the “Music on Hold” in the “Features and Specifications Manual (Business/Hotel)”.

[9300V5 software required]

- When the link between offices connected by CCIS via IPT (P2P CCIS) is interrupted, the lamp of Multiline Terminal button becomes the state as shown below. Then press the button, the LCD of the Multiline Terminal displays the following.

COLOR AND STATE OF BUTTON		STATE AND OPERATION	LCD DISPLAY
Red/Flashing (Momentarily)	0.125 seconds ON- 0.125 seconds OFF	Link Down occurrence	-
Red/Flashing (Slowly)	0.5 seconds ON- 0.5 seconds OFF	Press the button after Link Down occurrence	Link Down to CCIS
OFF	-	Link restoration	-
OFF	-	Press the button after Link restoration	Normal Condition: CCIS

- When the link between offices recovers, the flashing lamp of the button goes out.

Conditions for IP-PAD

- The IP-PAD is required for the following connections.
 - IP Station-to-Legacy terminal connection
 - IP Station-to-Public Network/Tie Line Network connection
 - IP Station-to-Conference Trunk (CFT) connection
 - IP Station-to-Voice Response System (VRS) connection

Conditions for SIP Connection

- Combination of Point-to-Point Connection and Point-to-Multipoint Connection in a system is not allowed.
- The number of available SIP trunk channels are as follows.

9300 Software	Number of Available SIP Trunk Channels
9300V3 STEP2 software or later	A maximum of 127 channels per Unit A maximum of 512 channels per system
9300V3 software or before	A maximum of 100 channels per Unit A maximum of 512 channels per system

- Processing power condition for SIP trunk is 2400 BHCA/unit and 5000 BHCA/system.
- The type of voice encoding is G.711, G.729a and T.38 (UDPTL).

- The following numbers can be notified as calling numbers.
 - Station number of My Line/Sub Line (set by CM13 Y=70, 71)
 - ISDN/SIP subscriber number (set by CM12 Y=12, 13/CM12 Y=46, 47/CM12 Y=51, 52)
 - Representative number (Registration number) (set by CMBA Y=32)
- Automated Attendant (CM76 Y=01: D09) can be used with the SIP trunk connection. However, the following features are not supported.

FEATURES	SYSTEM DATA
DTMF out-band mode for SIP trunk (with RFC2833)	CMBA Y=53: 03
Voice Response System Access Delete	CM20 Y=0-3: A102 NOTE
Mobility Access	

NOTE: *Voice Response System Access Record/Replay (CM20 Y=0-3: A100/A101) is available.*

- Remote Access to System (DISA) (CM76 Y=01: D16) can be used with the SIP trunk connection. However, the following features are not supported.

FEATURES	SYSTEM DATA
DTMF out-band mode for SIP trunk (with RFC2833)	CMBA Y=53: 03
Automated Attendant Handling of all PBR busy when 2 nd announcement and DT are connected	CM30 Y=33 NOTE 1
Answering System for Day Mode	CM64 Y=0 NOTE 2
Answering System for Night Mode	CM64 Y=2 NOTE 2
Voice Response System Access Delete	CM20 Y=0-3: A102 NOTE 3
Mobility Access	

NOTE 1: *In SIP trunk connections, Answering System for Day/Night Mode surely results in release processes regardless of system data settings.*

NOTE 2: *In SIP trunk connections, Answering System for Day/Night Mode surely results in DT connections regardless of system data settings.*

NOTE 3: *Voice Response System Access Record/Replay (CM20 Y=0-3: A100/A101) is available.*

- For the number of VoIPDB channels used by SIP trunk, refer to the number of VoIPDB channels used by the connection configurations in the “Conditions for Overall System”. [📄 Page 1-11](#)

- Call Transfer via the SIP trunk is available by performing the Call Transfer operation (the order for receiving REFER message) from the SIP network side.
The conditions for Call Transfer with the SIP connection are as follows.
 - When the Call Transfer is made from the SIP trunk, SV9300 connects the call on the SIP network side with two SIP trunk channels (REFER message is not sent).
 - When SIP trunk receives the order for Call Transfer from the SIP network, and the call is originated as the transferred call, the same trunk number is used.
 - Because a transferred call after receiving REFER message is always originated to the SIP network, the transferred call cannot be connected to other trunks or stations.
 - When the second data of CMBA Y=141>00-31 is set to 1 (default), “403 Forbidden” is sent for REFER message (order for Call Transfer) from the SIP network.
 - When no license is registered, “403 Forbidden” is sent for REFER message (order for Call Transfer) from the SIP network.
 - For a transferred call after receiving REFER message, the representative number assigned by CMBA Y=32 is used as the calling number (the call is originated as the anonymous call when no data is set to CMBA Y=32).
 - Even when Calling Name Display feature is effective, representative number assigned by CMBA Y=32 is notified as the calling number.
 - The number listed in “Refer-to” on the REFER message is used as an incoming call number.
 - Adding or deleting an incoming call number to a transferred call is not available.
 - When the transferred call from the SIP trunk cannot be established due to the failure at the transferred destination, A NOTIFY message (status notification) that Call Transfer is not available is sent to the caller who made Call Transfer. Whether the call is released or disconnected depends on the specifications of the SIP network which received the NOTIFY message.
 - The alternate routing for failures does not function even when the call origination to the transferred destination failed.
- The following is the features to be provided via SIP trunk for IP phone service.
 - Caller ID Display
 - Direct Inward Dialing (DID)
 - Registration Number “+” Addition/Deletion
 - Fault Registration
 - Fragmented IP Packet Receiving
 - Multiple CODEC Selection
 - Session Timer

- The services provided by SV9300 are listed in “List of PBX Services”. [☞ Page 1-104](#)
Since some of these services are not available depending on the terminal, set the services based on the list.
For the further information, refer to the Programming Manual.
Please note that the manuals above do not cover the following.
 - Terminating System (translation method of called number) **NOTE**
 - (a) Terminated via Tie line
PBX defines the termination destination with Dial Number Development assuming that the called number was originated from Tie line.
 - Station/Attendant Console/trunk (at tandem connection) can be specified as a destination.
 - When PBX defines to send to station, the last 1 to 8 digits of the number can be handled as the station number.
 - When PBX defines to send to trunk, the number can be deleted/added with LCR number development. (Delete: Maximum 10 digits/Add: Maximum 32 digits)
 - (b) Terminated via DID
 - The last 1 to 8 digits of the called number is handled as the DID number.
 - The number can be converted into the following Terminating System with DID Number Development.
 - Station
 - TAS
 - Automated Attendant
 - DISA
 - Maximum 1000 DID numbers can be handled.
 - DID number which is not registered is treated as Tie Line termination.
 - Method of origination operation
PBX originates a call with LCR Number Development.
 - Holding/Transferring features
Holding/Transferring features are not available with SIP method.
 - Setting Ringing Tone
The ringing tone for SIP transmission can be set by CM08>649.

NOTE: *All of the types of the called numbers are handled similarly on the PBX. When you want to discriminate between the called numbers which are recognized as representative numbers on such an upper SIP server and the called numbers which are recognized as DID numbers, assign an appropriate terminating system to each called number, e.g. Fragmented IP packet receiving for representative numbers, Direct Inward Dialing (DID) for station terminating.*

List of PBX Services

×: Available –: Not available Δ: Out of Target

SERVICES	DIGITAL MULTILINE TERMINAL	SLT	IP STATION	SOFT PHONE	REMARKS
Hot Line-OUTSIDE	×	×	×	×	
Call Hold	×	×	×	×	
Station Speed Dialing	×	×	×	×	
System Speed Dialing	×	×	×	×	
Last Number Call (Last Number Redial)	×	×	×	×	
Toll Restriction	×	×	×	×	
Direct Inward Dialing (DID)	×	×	×	Δ	
DID Name Display	×	–	×	Δ	
Station Camp-On (Camp-On)	×	×	×	Δ	
Music on Hold	×	×	×	Δ	
Group Diversion	×	×	×	Δ	
DAY/NIGHT MODE Change by System Clock	×	×	×	×	
Queue Limit For TAS	×	×	×	×	
Distinctive Ringing	×	Δ	×	Δ	
Automated Attendant	×	×	×	×	
Remote Access to System (Direct Inward System Access)	×	×	×	×	
Call Forwarding Set by DISA	×	×	×	×	
Day/Night Mode Change by Station Dialing	×	×	×	×	
Night Connection-Fixed Night Connection-Flexible	×	×	×	×	

- Conditions for Link Down Notice for SIP connection are shown below.
 - Link Down Notice via SIP is available only for Point-to-Point connection.
 - Link Down Notice via SIP is available only when the SIP trunk is re-registered periodically to the SIP server (CMBA Y=70: 0).
 - Link Down Notice via SIP is provided only when the periodic SIP trunk re-registration (CMBA Y=70: 0) is failed to the SIP server.
 - When the link between offices connected by the SIP trunk is interrupted, the lamp of Multiline Terminal button becomes the state as shown below. Then press the button, the LCD of the Multiline Terminal displays the following.

COLOR AND STATE OF BUTTON		STATE AND OPERATION	LCD DISPLAY
Red/Flashing (Momentarily)	0.125 seconds ON- 0.125 seconds OFF	Link Down occurrence	-
Red/Flashing (Slowly)	0.5 seconds ON- 0.5 seconds OFF	Press the button after Link Down occurrence	Link Down to SIP-TRK
OFF	-	Link restoration	-
OFF	-	Press the button after Link restoration	Normal Condition: SIP-TRK

- When the link between offices recovers, the flashing lamp of the button goes out.

Conditions for Point-to-Multipoint connection

- Point-to-Multipoint connection is available only when the destination office is SV9300 or NEAX2000 IPS.
- One SIP trunk can be accommodated in a system (in Unit01).
- This feature can be assigned a maximum of 63 Destination IP Addresses.
- Assign the Destination IP Address Pattern number to the LCR Pattern number.
- This feature can be assigned a maximum of 8 IP address to one of destination office.
- When SIP carrier connection is used, Point-to-Multipoint connection cannot be used together.
- This feature cannot use the connection by Domain Name.
- This feature cannot specify the destination office by the Fully Qualified Domain Name (FQDN).
- This feature is not available by performing the Call Transfer operation (the order for receiving REFER message) from the SIP network side.
- This feature cannot be received to SV9300 when the calling party number did not receive from the destination office for Point-to-Multipoint connection. Therefore, set up in the destination office to be able to notify the calling party number.
- When Voice encoding is switched by LCR Pattern number, LCR Pattern number is decided based on the calling party number notified from the destination office for Voice encoding of incoming call. Therefore, set up LCR Pattern number so that DPC and Profile number (CM8A Y=5000-5255>168, 179) can develop the calling party number by Closed Numbering Plan.
- The profile for SIP trunk used for calling can be carried on individually by dialing number. In this case, specifying the same profile is recommended.

Conditions for Point-to-Point Connection

- SIP trunks can be accommodated in all units (Unit01-50).
- Call control for SIP trunk is made in Unit01. SIP messages in SIP trunk accommodated in Unit02-50 communicates with Unit01 with using communication between units. Because the communication of SIP trunk between units is operated using the listening port of the SIP trunk (default: UDP port 5060), be aware of configuration of Firewall and bandwidth reservation.
- If multiple SIP trunks are used for connecting to SIP server (carrier), be aware of the following:
 - Maximum number of SIP servers (carriers) to be connected is 8/unit and 50/system.
 - If the contract for SIP line (ID/Password) with the SIP server (carrier) covers only one account, SIP trunk line must not be accommodated in multiple units.
 - If Sip trunks are accommodated in multiple units, each unit requires an account of the contract for SIP line (ID/Password) with the SIP server (carrier).
 - If the contract for SIP line (ID/Password) with the SIP server (carrier) covers 2 or more accounts and the destination SIP servers have individual IP addresses, multiple SIP trunks can be set and operated in a unit.
- The ring tone heard in the SIP trunk accommodated in Unit02-50 is generated in the accommodated unit.
- The PAD value between Unit02-50 and the carrier can be modified by CM68 or CM67.
- The PAD value between units is fixed to 0 dB, regardless of office data setting.
- In the case SIP trunks are accommodated in Unit02-50, different ToS/Diffserv can be set to SIP control packets between Unit01 and Unit02-50 and to SIP control packets between Unit02-50 and carrier.
- For security, take enough care to set the router/Firewall connecting to the global network.
- As for the SIP trunk profile to be used when transmitting, specify an individual profile for each SIP server (carrier).
- When transmitting a single SIP server (carrier), assigning a SIP trunk profile to each dialing number cannot be done.

Conditions for DNS Server Setting

- Each unit can set the Primary DNS and the Secondary DNS. This means, in a system, up to 50 pairs of DNS servers ((Primary DNS × 1, Secondary DNS × 1) × 50) can be set.
- For the SIP line requiring IP address resolution, set up DNS server of the unit where the SIP line is accommodated. For example, required DNS resolution of the SIP line in Unit05 shall be made with using the DNS server for Unit05.
- Up to 128 characters of the domain name (Fully Qualified Domain Name - FQDN) of the SIP server can be resolved by DNS.
- One of the following executes DNS resolution:
 - System reset
 - Time-to-live (TTL) of DNS expires
 - DNS cache table is cleared by CMBA Y=99
- “DNS resolution” means resolving domain name and converting it to IP address. When 2 or more IP addresses are returned from the DNS server with a single domain name, the first IP address is adopted.
- The data of DNS resolution is preserved as a cache. The effective time of the cache follows the time-to-live (TTL) that is reported from the DNS server at the time of DNS resolution. Even when DNS server failure occurs and DNS resolution cannot update, the operation will continue using the information in cache.
- Conditions for error occurrence with DNS query are as following:
 - When an error occurs with DNS query, Fault Kind 43 (SIP trunk failure) is registered.
 - When the detail of the error with DNS query is Name error (FQDN queried does not exist in the DNS server), no more DNS queries will be made afterwards. For another DNS query, rebooting the device or clearing DNS cache table (CMBA Y=99) is required.

Conditions for FAX in SIP Trunk

- As for FAX signal in the case of cross-unit, specify the FAX protocol used between units by CM67 Y=20. Specify the FAX protocol used between SIP trunk carriers by CMBA Y=119.
- FAX communication is not supported in a state in which the tandem connection is made on SIP trunk-SIP trunk.
- When delay, packet loss or such occurs in IP segment, FAX communication may fail.

Conditions for SIP Trunk Source IP Address Check Function

- For unauthorized access prevention, this feature confirms that the source of the SIP message (source IP address of the IP header) matches the IP address of the SIP server that is registered to SV9300.
- This function is available only when Point-to-Point connection is performed (unavailable when Point-to-Multipoint connection is performed).
- This function is specified with CM0B Y=1XX>53 (default: not provide).
- By checking the source IP address, if the address is considered incorrect, no answer is replied and the packet is discarded though the SIP message is received.
- Regardless of the setting of this function, RTP sends/receives packets only with the other party that established the session.

Conditions for RTP Monitoring Function

- This function monitors RTP reception status during a call. If no RTP has been received during a fixed period (approx. 10 seconds), this function recovers the call in progress. This feature prevents to become a state in which a channel is being occupied for the disconnect message has not reach, due to network failure.
- This function is effective for whole the system. Set with CM08>672.
- This function is effective only for the call using IP trunk the (P2P CCIS and SIP trunk).
- RTP packets sent and received between units will also be monitored.
- If a call is restored by this function, fault information is registered.

Conditions for DRS

DRS=Device Registration Server

- The System-based DRS executes IP Station registration.
- The Network-based DRS is not available for the IP Station registration.

Conditions for Legacy Interface

- All Line/Trunk blades used in SV9300 are available for the system.

Conditions for Maintenance

- PCPro can be used as the maintenance program for SV9300. Direct connection (RS-232C), Modem connection and IP connection are available to connect to the PCPro.
- You can check the condition of LAN cable connection by transmitting the ping packet to the SV9300 from PC on the LAN.
- PCPro is needed to use license data.

SERVICES

Business/Hotel/Data/CCIS/ISDN

All Business/Hotel/Data/CCIS/ISDN features are available.

NOTE: *Add-On Module [For North America]/Add-On Module-Digital Multiline Terminal [Other than North America] is not available for IP Station.*

OAI

Built-in OAI is available on the CPU blade, excluding the OAI with RS-232C and FLF [For North America].

NOTE: *The port number of SV9300 for OAI is 60030 by default. The port number 1024, 1025, 1039 are available by CM0B Y=001>41.
A maximum of four applications can be connected to the system at the same time, using the port number mentioned above.*

SYSTEM DATA PROGRAMMING

This chapter explains the programming procedure of the system data, IP Station setup procedure and IP Station login/logout procedure.

HOW TO READ THIS CHAPTER

In the programming procedure, the meanings of (1), (2) and markings are as follows.

- (1) : 1st Data
(2) : 2nd Data
◀ : Default
With the system data clear command (CM00, CM01), the data with this marking is automatically assigned for each command.
- RESET** : A reset of the CPU blade is required after data setting.
Press RESET switch on the CPU blade.
- OFF LINE** : Command with this marking can be used only under Off-Line mode of the CPU blade. To set Off-Line mode,
(1) Set SENSE switch on the CPU blade to “E” or “F”.
(2) Press RESET switch on the CPU blade.
- BLADE RESET** : Commands which require a reset of the blade by CME0 Y=3 after data setting.
- IPT (P2P CCIS) RESET** : Commands which require a reset of the IPT (P2P CCIS) by CME0 Y=5>01 after data setting.

PRECAUTIONS

System Data Backup

CAUTION

- If you operate as follows without system data backup after system data setting or service memory setting (registration of the features such as “Call Forwarding” and “Speed Dialing” from a station), the data that has been set is invalid.

You must execute the system data backup before the following operations.

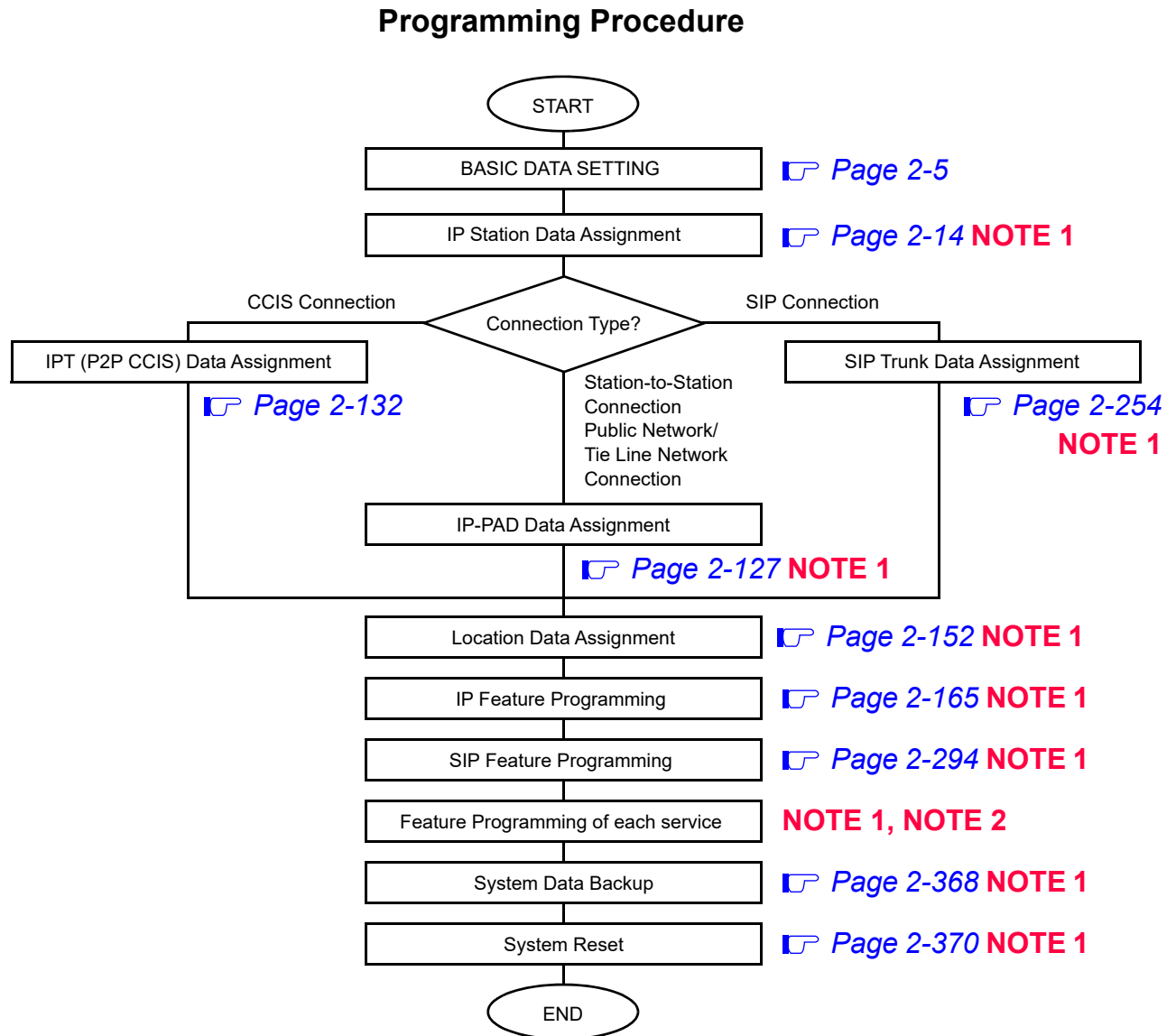
- Turning Off the system
- System Reset (reset of CPU blade)
- Changing the CPU blade to Off-Line Mode
- Changing the CPU blade to On-Line Mode after system data setting under Off-Line Mode

In addition, the VRS data backup is also required when any data is recorded or changed in the VRS.

- You can execute the system data backup by the following two ways.
 - Executing the system data backup once a day at the time set by CM43 Y=5>00
(If no data is set, the default setting is 3:00 a.m.)
As the system data backup is performed, the VRS data backup is also performed at the same time.
 - Executing the system data backup from PCPro/CAT by CMEC Y=6>0: 0
- Do not reset the CPU blade while “SYSD” LED on the CPU blade is flashing.

PROGRAMMING PROCEDURE

Program the system data according to the following procedure.



NOTE 1: *The data assignment can be executed under Off-Line mode or On-Line mode of the CPU blade. When setting the data under Off-Line mode, system reset is required after data setting.*

NOTE 2: *For feature programming of each service, refer to the Programming Manual.*

BASIC DATA SETTING

For initial system setup, you must execute the following.

- System Data Memory All Clear [☞ Page 2-6](#)
- SRAM Memory All Clear [☞ Page 2-6](#)
- License Activation Data Loading [☞ Page 2-6](#)
- Setting System Clock [☞ Page 2-7](#)
- Setting Nation Code [☞ Page 2-8](#)
- Setting A-law/ μ -law [☞ Page 2-9](#)
- Setting Kind of Tone/Music on Hold/Tone [☞ Page 2-10](#)
- VLAN Data Assignment [☞ Page 2-12](#)

System Data Memory All Clear

You can select the system data memory clear according to the system configuration.

START	DESCRIPTION	DATA
START <div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM00</div> END	Clear all system data memory for the system. <div style="text-align: center; border: 1px solid black; border-radius: 10px; padding: 2px;">OFF LINE</div>	(1) 1 (2) CCC

SRAM Memory All Clear

You can select the SRAM memory clear according to the system configuration.

START	DESCRIPTION	DATA
START <div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM00</div> END	Clear all SRAM memory for the system. <div style="text-align: center; border: 1px solid black; border-radius: 10px; padding: 2px;">OFF LINE</div>	(1) 02 (2) CCC

NOTE: *Be careful handling CM00>02. Because all stored data in a SRAM area is cleared when the SRAM area memory clear is executed by this command. Usually, this command is used only when initially starting the system, accommodating additional Units, replacing the CPU blade, or SRAM data is corrupted by such as power discontinuity of SV9300 (because the battery backup for SRAM is effective for approximately one week, the SRAM data is corrupted when SV9300 has been powered off for more than one week).*

License Activation Data Loading

The License Activation data must be loaded into the CPU blade to activate the system. Refer to the PC Programming Manual for details.

Setting System Clock

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; display: inline-block;">CM02</div>	Assign the system clock data. NOTE: <i>Date/Time for Standard SIP station cannot be assigned. Change the Date/Time on the terminal side, if necessary.</i>	(1) 0: Calendar Year (2) 2014-2099 (Ex. 2014) (1) 1: Date (2) MM DD WW MM : 01-12 (Month) DD : 01-31 (Date) WW: 00 (Sun) 01 (Mon) 02 (Tue) 03 (Wed) 04 (Thu) 05 (Fri) 06 (Sat) (1) 2: Time (2) HH MM SS HH : 00-23 (Hour) MM : 00-59 (Minute) SS : 00-59 (Second)
<u>END</u>		

Setting Nation Code

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM31</div>	Specify the nation code. <div style="text-align: center; margin-top: 10px;"> RESET </div>	<ul style="list-style-type: none"> • Y=0 (1) 0 (2) 01 : Australia 03 : North America 04 : Asia/Africa/Europe/Latin America/Middle East/Russia 05 : Austria/Belgium/Denmark/Germany/Italy/South Africa/Spain/Sweden/Switzerland/The Netherlands/UK/Brazil/China/International/Latin America/Asia [For EMEA] NOTE 2 15 : New Zealand NONE◀: As per CPU blade
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>		

NOTE 1: The default of CM31 Y=0>0 depends on each nation code of the CPU program.

For Australia/NZ: 01◀

For North America: 03◀

For Asia/Africa/Europe/Latin America/Middle East/Russia: 04◀

For Austria/Belgium/Denmark/Germany/Italy/South Africa/Spain/Sweden/Switzerland/

The Netherlands/UK/Brazil/China/International/Latin America/Asia: 05◀

NOTE 2: In the case of EMEA, the default of CM31 Y=0>0 is same as North America (nation code 03).

Therefore, you must set the nation code to 05 by this command.

Setting A-law/ μ -law [For Asia/EMEA]

START	DESCRIPTION	DATA
CM04	<p>Specify the A-law/μ-law for the Main Unit.</p> <p>NOTE: <i>Set the same value as Remote Unit (CM04 Y=14-59). You cannot set both of A-law and μ-law in the same Remote UNIT over IP system.</i></p>	<ul style="list-style-type: none">• Y=10-13 (Unit01-04) <p>(1) 00</p> <p>(2) 0 : A-law 1 : μ-law 2 : Not used 3 ◀: Depends on the CPU</p>
END		

Setting Kind of Tone/Music on Hold/Tone

(1) Kind of Tone

Kind of Tone is decided by CM04 Y=00>01, 03 data setting. To select the Kind of Tone for each country, set the data below.

START	DESCRIPTION	DATA
CM04	Set the Kind of Tone for TDM (System-basis).	<ul style="list-style-type: none"> • Y=00 (1) 03 (2) 01 : Japan 02 : North America 03 : Australia 04 : A-law countries 05 : Hong Kong 06 : Malaysia 07 : Singapore 08 : UK 09 : Mexico 10 : Taiwan 11 : New Zealand 12 : Korea 13 : China 14 : Thailand 15 : Brazil 16 : Netherlands 17 : Germany 18 : Italy 19 : Austria 20 : Belgium 21 : Spain 22 : Sweden 23 : UK 24 : Denmark 25 : Greece 26 : Switzerland 27 : South Africa 28 : Russia NONE◀: As per CM31 Y=0>0
END		

For EMEA

(2) Music on Hold

To provide Internal Hold Tone source on the CPU blade, set the data below.

NOTE: *This data setting is effective only for the legacy terminal.
For IP Station, this data setting is not effective. IP Station uses the tone source in IP Adapter (Minuet).*

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM48</div>	Set the music for Internal Hold Tone. NOTE: <i>Set CM48 Y=0>00/01/02: 1400 after this data setting.</i>	<ul style="list-style-type: none"> • Y=3 (1) 01 (2) 00 : Nocturne 01 : Minuet 02 : Fur Elise 03 : The Maiden's Prayer 04 : When the saints go marching in 06 : Spring (by four seasons) 08 : Ich bin ein Musikante (German folk song) 10 : Amaryllis (French folk song) NONE◀: Minuet
END		

(3) Periodic Time Indication Tone

To provide Periodic Time Indication Tone, set the data below.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM41</div>	Specify the timing interval for Periodic Time Indication Tone.	<ul style="list-style-type: none"> • Y=0 (1) 09 (2) 01-17: 32-548 seconds (32 seconds increments) If no data is set, the default setting is 192-196 seconds.
END		

VLAN Data Assignment

NOTE 1: The system supports only Tag VLAN.

NOTE 2: Switching HUB that supports VLAN is required to provide the VLAN function.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin-bottom: 10px;">CM0B</div> <div style="border-left: 1px solid black; border-right: 1px solid black; border-bottom: 1px solid black; height: 500px; margin-left: 10px;"></div> <div style="border: 1px solid black; padding: 2px; width: fit-content; margin-left: 10px; margin-top: 10px;">A</div>	Provide the VLAN with the system. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 10px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=0XX (Maintenance Port [0] + Unit No. [01-50]) 1XX (VOIP Port [1] + Unit No. [01-50]) (1) 50 (2) 0 : To provide 1◀: Not provided
	Specify the priority of the VLAN. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 10px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=0XX (Maintenance Port [0] + Unit No. [01-50]) 1XX (VOIP Port [1] + Unit No. [01-50]) (1) 51 (2) 0-7◀: Priority 0-7 NOTE
	Assign the VLAN ID to the system. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 10px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=0XX (Maintenance Port [0] + Unit No. [01-50]) 1XX (VOIP Port [1] + Unit No. [01-50]) (1) 52 (2) 1-4094 : VLAN ID 1-4094 NOTE 2 NONE◀: No data
	Assign the IP address for the VLAN. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 10px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=0XX (Maintenance Port [0] + Unit No. [01-50]) 1XX (VOIP Port [1] + Unit No. [01-50]) (1) 00 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address for the VLAN (Maximum 15 digits) NONE◀: 192. 168. 1. 1 (for Maintenance Port)/No data (for VOIP Port)
	NOTE: <i>The higher number has higher priority.</i>	
NOTE: <i>One VLAN ID can be set per system.</i>		
NOTE: <i>The second data must be entered including the periods (.).</i>		

A	DESCRIPTION	DATA
CM0B	Assign the Subnet Mask for the VLAN.	<ul style="list-style-type: none"> Y=0XX (Maintenance Port [0] + Unit No. [01-50]) 1XX (VOIP Port [1] + Unit No. [01-50])
	<p>NOTE: <i>The second data must be entered including the periods (.).</i></p>	<p>(1) 01</p> <p>(2) XXX.XXX.XXX.XXX: 255.0.0.0-255.255.255.252: Subnet Mask for the VLAN (Maximum 15 digits) NONE◀: 255. 255. 255. 0 (for Maintenance Port)/No data (for VOIP Port)</p>
Assign the IP address (RTP) for the VLAN of the VoIPDB.	RESET	<ul style="list-style-type: none"> Y=2XX (VOIP Port [2] + Unit No. [01-50]) <p>(1) 00: IP Address</p> <p>(2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address for the VLAN (Maximum 15 digits) NONE◀: No data</p>
<p>NOTE: <i>The second data must be entered including the periods (.).</i></p>		
END		

IP STATION DATA ASSIGNMENT

This section explains the following data assignment to accommodate IP Stations.

- IP Station Basic Data Assignment [Page 2-15](#)
- Soft Client SP350 Data Assignment [Page 2-18](#)
- IP Multiline Terminal (DT700/DT800/DT900 Series) Data Assignment [Page 2-23](#)
- SIP Wireless Terminal (MH240) Data Assignment [Page 2-31](#)
- I755 IP DECT Terminal Data Assignment [Page 2-32](#)
- IP Single Line Telephone (SIP) Data Assignment [Page 2-34](#)
- PUSH Notification Service Data Assignment [Page 2-71](#)
- Representative Call Pick-Up on ST500 Data Assignment [Page 2-80](#)
- IP Station Login/Logout Data Assignment [Page 2-83](#)

IP Station Basic Data Assignment

In addition to the system constitution data for IP Station, do the following programming.

NOTE: *IP Station Basic Data Assignment is intended for IP Multiline Terminals. As for the basic data assignment for Standard SIP Terminal, refer to the “(2) Standard SIP station data assignment”.*

 [Page 2-37](#)

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM10</div>	Assign the station number of the IP Station to the port number of the IP Station.	<ul style="list-style-type: none"> • Y=01 (1) 0000-1535: Virtual Port No. (2) FX-FXXXXXXXXX: Station No. of IP Station X: 0-9, A (*), B (#) NONE◀: No data <p>NOTE: <i>When IP Station is assigned by this command, the second data of each office data below is automatically set to “0” (To provide/To store).</i></p> <ul style="list-style-type: none"> - CM13 Y=41: 0 (To store the call record when answering a station call) - CM13 Y=49: 0 (To store the call record when handling an unanswered call) - CM13 Y=60: 0 (To store the call record when answering a trunk call) - CM13 Y=61: 0 (To store the call record when handling an unanswered trunk call)
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM12</div>	Specify the kind of IP Station.	<ul style="list-style-type: none"> • Y=24 (1) X-XXXXXXXXX: My Line No. (2) 0 : B mode 7◀: A mode <p>NOTE 1: <i>Set the second data to “0” (B mode) for Soft Phone.</i></p> <p>NOTE 2: <i>As for the key layout of Line/Trunk/Feature Keys and One Touch Keys of the Multiline Terminal, refer to the Appendix A TERMINAL KEY ASSIGNMENT in the Command Manual.</i></p>
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">A</div>		

A	DESCRIPTION	DATA
CM20	Assign the number of digits for station number.	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) 801 : 1 digit Station <li style="padding-left: 20px;">802 : 2 digits Station <li style="padding-left: 20px;">803 : 3 digits Station <li style="padding-left: 20px;">804 : 4 digits Station <li style="padding-left: 20px;">805 : 5 digits Station <li style="padding-left: 20px;">806 : 6 digits Station <li style="padding-left: 20px;">807 : 7 digits Station <li style="padding-left: 20px;">808 : 8 digits Station
CM90	Assign a My Line key on the IP Station, if required.	<ul style="list-style-type: none"> • Y=00 (1) My Line No. + <input type="text"/> + Key No. (2) X-XXXXXXXX: Station No.
CM93	Assign a Prime Line to the IP Station.	<ul style="list-style-type: none"> (1) X-XXXXXXXX: My Line No. (2) X-XXXXXXXX: Station No.
CM0B	Assign the base port number of the UDP port for IP Multiline Terminal which is NEC original UDP port (PROTIMS port).	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 30 (2) 01024-65534: UDP Base Port No. <li style="padding-left: 20px;">NONE◀ : 50000 [50000-52047 (2048 ports are used)]
	<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	
	<p>NOTE 1: Follow the typical default setting.</p> <p>NOTE 2: One UDP port is required to a IP Station.</p> <p>NOTE 3: Be sure that the use range of each port assigned by CM0B Y=2XX>40, Y=1XX>36-39 does not overlap the setting of the UDP port for IP Multiline Terminal.</p>	
B		

B	DESCRIPTION	DATA
CM0B	<p>Assign the port number of the Registration Admission Status (RAS) port which is NEC original UDP port for System-based DRS.</p> <p style="text-align: right;">(RESET)</p> <p>NOTE: Follow the typical default setting.</p> <p>Assign the port number of the UDP port for DT700/DT800/DT900 Series/SP350 voice control packet.</p> <p style="text-align: right;">(RESET)</p> <p>NOTE: Follow the typical default setting.</p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 31 (2) 01024-65534: UDP Port No. 1024-65534 00000-01023: UDP Port No. 3456 NONE◀ : UDP Port No. 3456 <ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 32 (2) 01024-65534: UDP Port No. 1024-65534 NONE◀ : UDP Port No. 5080
CM13	<p>Specify whether the DTMF signal is sent from DT700/DT800/DT900 Series to D^{term}IP.</p> <p>NOTE: This data is used when IP Multiline Terminal or BCT (IVR) is connected to the system. This data is assigned to the station number of IP Multiline Terminal/BCT (IVR).</p>	<ul style="list-style-type: none"> • Y=68 (1) X-XXXXXXXX: Station No. (2) 0 : To send 1◀: Not sent
CM08	<p>Specify whether DT700/DT800/DT900/SP350 RTP warning tone is sent.</p> <p>Specify the call is disconnected or continued when the connection failure between SV9300 and DT700/DT800/DT900 occurs.</p> <p>NOTE: DT700/DT800/DT900 has Keeping Call function and can continue the call until the terminal goes on-hook even when SV9300 is reset, as long as they are in station-to-station calling (Peer-to-Peer) within same system. For the calls connected via VoIP, however, the voice is interrupted at system reset, and this situation will continue until on-hook. When the Keeping Call function of DT700/DT800/DT900 is set to ineffective, the stations are forcibly reset after 3 minutes elapsed from SV9300 reset.</p>	<ul style="list-style-type: none"> (1) 1004 (2) 0 : Not sent 1◀: To send <ul style="list-style-type: none"> (1) 1006 (2) 0 : To disconnect 1◀: To continue
END		

Soft Client SP350 Data Assignment

In addition to the IP Station Basic Data Assignment, this section explains the following data assignment to use each feature for SP350.

- (1) Presence service data assignment : [Page 2-18](#)
- (2) Data Conference and Instant Message data assignment : [Page 2-19](#)
- (3) Answering Sub Line call/Ringing Line Pickup data assignment: [Page 2-20](#)
- (4) Fault information data assignment : [Page 2-21](#)
- (5) SP350 status confirmation : [Page 2-21](#)

- (1) Presence service data assignment
To use Presence service, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM0B</div>	<p>Assign the UDP port number for the Presence Server.</p> <p style="text-align: right;">RESET</p> <p>NOTE 1: <i>Follow the typical default setting.</i></p> <p>NOTE 2: <i>This data can be assigned only for Unit No. 01.</i></p>	<ul style="list-style-type: none"> • Y=101 (VOIP Port [1] + Unit No. [01]) (1) 43: UDP Port for Presence Server (2) 01024-65534: UDP Port No. 1024-65534 NONE◀ : UDP Port No. 5082
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBC</div>	<p>Assign the IP address for the Presence Server.</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=03 (1) 001 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data
<p><u>END</u></p>		

(2) Data Conference and Instant Message data assignment

To use Data Conference and Instant Message services, do the following programming.

NOTE: Before doing the following programming, "IPT (P2P CCIS) DATA ASSIGNMENT" must be set in advance. [Page 2-132](#)

START	DESCRIPTION	DATA
CM0B	Assign the port number of the Registration Admission Status (RAS) port which is NEC original UDP port for System-based DRS. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 10px;">RESET</div> <p>NOTE: Follow the typical default setting.</p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 31 (2) 01024-65534: UDP Port No. 1024-65534 00000-01023: UDP Port No. 3456 NONE◀ : UDP Port No. 3456
CM8A	Allow the Data Conference and Instant Message services to the LCR Pattern No.	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 181: Setting of Data Conference/Instant Message feature (2) 0 : Restricted 1◀: Allow
END		

• To use Data Conference and Instant Message services between SP350 accommodated to SV9300 and SP350 accommodated to SV9500/SV8500/SV7000, do the following programming.

START	DESCRIPTION	DATA
CMBC	Assign the IP addresses of the own office and the destination office to perform the greeting process for SP350 between offices. <p>NOTE: When this data is set or changed, be sure to initialize the greeting list by CMBC Y=17.</p> <p>Initialize the greeting list created by the greeting process.</p>	<ul style="list-style-type: none"> • Y=16 (1) 000: Greeting No. 000 001: Greeting No. 001 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Addresses of own office and destination office (Maximum 15 digits) NONE◀: No data • Y=17 (1) 000: Fixed value (2) CCC: Greeting List Initialize
END		

(3) Answering Sub Line call/Ringing Line Pickup data assignment

To answer the incoming call to Sub Line with USB handset, and to set Ringing Line Pickup, do the following programming.

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required stations.	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Allow Ringing Line Pickup in Service Restriction Class C assigned by CM12 Y=07.	<ul style="list-style-type: none"> • Y=082 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0: Allow
CM90	Assign the setting of Multiline Terminal ringing tone by Day Mode/Night Mode. NOTE: This data is effective when CM08> 577 is set to 0.	<ul style="list-style-type: none"> • Y=01 (1) My Line No. + [] + Key No. (01-24) (2) 0 : Day Mode: No ringing/Night Mode: Ringing 1 : Day Mode: Ringing/Night Mode: No ringing 2 : Day Mode: No ringing/Night Mode: No ringing 3◀: Day Mode: Ringing/Night Mode: Ringing
END		

NOTE: In addition of programming in the above, SP350 needs the following setting.

For Answering Sub Line call:

- Make the Answer preset ineffective by the user setting.
- Set the Answer key to the appropriate function key (F1)-(F4) by the handset setting.

For Ringing Line Pickup:

- Make the Answer preset effective by the user setting.

(4) Fault information data assignment

To confirm the start up of SP350 by Fault Information Display (Fault Kind: 036 [Detect terminal connection]), do the following programming.

START	DESCRIPTION	DATA
<p>START</p> <div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 10px auto;">CMEA</div> <p style="text-align: center;">END</p>	<p>Register the fault information into memory and the control of the external alarm.</p>	<ul style="list-style-type: none"> • Y=2 (1) 036: Detect terminal connection (2) 0 : Fault memory store/No output of external alarm <li style="padding-left: 20px;">1 : Fault memory store/External alarm is MN alarm <li style="padding-left: 20px;">2 : Fault memory store/External alarm is MJ alarm <li style="padding-left: 20px;">3◀: No fault memory store/No external alarm output

(5) SP350 status confirmation

To confirm the SP350 status, do the following programming.

START	DESCRIPTION	DATA
<p>START</p> <div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 10px auto;">CMFA</div> <div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 10px auto; text-align: center;">A</div>	<p>Read the SP350 status.</p>	<ul style="list-style-type: none"> • Y=01 (1) X-XXXXXXXX: Station No. (2) 02 : D^{term}IP INASET <li style="padding-left: 20px;">03 : D^{term}85 (D^{term} Series i) (IP adapter type) <li style="padding-left: 20px;">05 : D^{term}85 (D^{term} Series i) (IP Bundled type) <li style="padding-left: 20px;">10 : D^{term}SP30 <li style="padding-left: 20px;">12 : DT710 <li style="padding-left: 20px;">13 : DT730/DT710 (Self-Labeling) <li style="padding-left: 20px;">14 : DT750 <li style="padding-left: 20px;">15 : MH240 <li style="padding-left: 20px;">16 : SP350

A	DESCRIPTION	DATA
CMFA		17 : DT730DG
		18 : DT730CG
		19 : DT770G
		21 : DT830/DT830DG
		22 : DT830CG
	NOTE 1: <i>The terminal type is read as "12" (DT710) when using 9300V3 software or before.</i>	23 : DT820 NOTE 1 [9300V3 STEP2 software required]
	NOTE 2: <i>The terminal type is read as "13" (DT730/DT710 (Self-Labeling)) when using 9300V3 software or before.</i>	24 : DT820 (Self-Labeling) NOTE 2 [9300V3 STEP2 software required]
	NOTE 3: <i>The terminal type is read as "22" (DT830CG) when using 9300V6 software or before.</i>	25 : DT930CG NOTE 3 [9300V7 software required]
	NOTE 4: <i>The terminal type is read as follows.</i> - When using 9300V3 STEP2 to 9300V6 software: "23" (DT820) - When using 9300V3 software or before: "12" (DT710)	26 : DT920 NOTE 4 [9300V7 software required]
		27 : DT920 (Self-Labeling) [9300V7 software required]
		28 : DT930 (Touch Panel) [9300V7 software required]
		41 : D ^{term} 85 (D ^{term} Series i)
		42 : D ^{term} 85 (D ^{term} Series i) Russian
		43 : DT310
		44 : DT330
		45 : DT330 Chinese
	NOTE 5: <i>DT510 is available from 9300V9 software or later. The terminal type between DT510 and DT410 cannot be identified by the system data. Also, DT510 is identified as DT410 when using 9300V6 software or before (DT510 cannot be accommodated to SV9300 when using 9300V7/V8 software).</i>	46 : DT510/DT410 NOTE 5
		47 : DT430
		48 : DT430 Chinese
		49 : DT530 [9300V7 software required]
		50 : DT530 Chinese [9300V7 software required]
		70 : Paging Adapter
		80 : Standard SIP station
		FF : Other
		NONE ◀: Logout Status/Disconnected

END

IP Multiline Terminal (DT700/DT800/DT900 Series) Data Assignment

In addition to the IP Station Basic Data Assignment, this section explains the following data assignment related to DT700/DT800/DT900 Series.

- (1) DT700/DT800/DT900 Series REGISTER/SUBSCRIBE data assignment : [📄 Page 2-24](#)
- (2) DT700/DT800/DT900 Series Security (Security Lock) : [📄 Page 2-27](#)
- (3) DT700/DT800/DT900 Series Security Cancel (Security Mode Cancel) : [📄 Page 2-28](#)
- (4) DT700/DT800/DT900 Series menu time out timer data assignment : [📄 Page 2-28](#)
- (5) DT900 Series LCD Display Mode data assignment : [📄 Page 2-29](#)
- (6) DT900 Series Terminal Phonebook data assignment : [📄 Page 2-29](#)
- (7) DT900 Series Portal Mode data assignment : [📄 Page 2-30](#)

(1) DT700/DT800/DT900 Series REGISTER/SUBSCRIBE data assignment

To assign the expire value for DT700/DT800/DT900 Series REGISTER/SUBSCRIBE, do the following programming.

START	DESCRIPTION	DATA
CM41	Assign the expire value for DT700/DT800/DT900 Series REGISTER.	<ul style="list-style-type: none"> • Y=0 (1) 130: REGISTER (Day) (2) 00-99 : 0-99 days NONE◀: 7 minutes
	<p>NOTE 1: Usually, this data setting is not required.</p>	
	<p>NOTE 2: This data implies the expire value for the DT700/DT800/DT900 Series REGISTER (terminal registration) and is set by the combination of CM41 Y=0>130-132 (day/hour/minute). When CM41 Y=0>130-132 (day/hour/minute) are all set to NONE, the expire value "7 minutes" is set as default.</p>	<ul style="list-style-type: none"> (1) 131: REGISTER (Hour) (2) 00-23 : 0-23 hours NONE◀: 7 minutes
	<p>NOTE 3: The DT700/DT800/DT900 Series updates the REGISTER at half interval of the value set by this data.</p>	<ul style="list-style-type: none"> (1) 132: REGISTER (Minute) (2) 00-59 : 0-59 minutes NONE◀: 7 minutes
	<p>NOTE 4: The load to SV9300 increases if small amount of data (such as 1 minute) is set. Therefore, it is recommended to set the value more than 3 minutes.</p>	
	<p>NOTE 5: A reset of the terminal is required after this data setting.</p>	
A		

A	DESCRIPTION	DATA
CM41	Assign the expire value for DT700/DT800/DT900 Series SUBSCRIBE for Server.	<ul style="list-style-type: none"> • Y=0 (1) 133: SUBSCRIBE for Server (Day) (2) 00-99 : 0-99 days NONE◀: 1 hour
	NOTE 1: Usually, this data setting is not required.	
	<p>NOTE 2: This data implies the expire value for the DT700/DT800/DT900 Series SUBSCRIBE (terminal control signal from DT700/DT800/DT900 Series to SV9300) and is set by the combination of CM41 Y=0>133-135 (day/hour/minute). When CM41 Y=0>133-135 (day/hour/minute) are all set to NONE, the expire value "1 hour" is set as default.</p>	<ul style="list-style-type: none"> (1) 134: SUBSCRIBE for Server (Hour) (2) 00-23 : 0-23 hours NONE◀: 1 hour
	<p>NOTE 3: The DT700/DT800/DT900 Series updates the SUBSCRIBE at half interval of the value set by this data.</p>	
	<p>NOTE 4: The load to SV9300 increases if small amount of data (such as 1 minute) is set. Therefore, it is recommended to set the value more than 1 hour.</p>	
	<p>NOTE 5: A reset of the terminal is required after this data setting.</p>	
B		

B	DESCRIPTION	DATA
CM41	Assign the expire value for DT700/DT800/DT900 Series SUBSCRIBE for Client.	<ul style="list-style-type: none"> • Y=0 (1) 136: SUBSCRIBE for Client (Day) (2) 00-99 : 0-99 days NONE◀: 1 hour
	NOTE 1: <i>Usually, this data setting is not required.</i>	
	<p>NOTE 2: <i>This data implies the expire value for the DT700/DT800/DT900 Series SUBSCRIBE (terminal control signal from SV9300 to DT700/DT800/DT900 Series) and is set by the combination of CM41 Y=0>136-138 (day/hour/minute). When CM41 Y=0>136-138 (day/hour/minute) are all set to NONE, the expire value "1 hour" is set as default.</i></p>	<ul style="list-style-type: none"> (1) 137: SUBSCRIBE for Client (Hour) (2) 00-23 : 0-23 hours NONE◀: 1 hour
	<p>NOTE 3: <i>The DT700/DT800/DT900 Series updates the SUBSCRIBE at half interval of the value set by this data.</i></p>	<ul style="list-style-type: none"> (1) 138: SUBSCRIBE for Client (Minute) (2) 00-59 : 0-59 minutes NONE◀: 1 hour
	<p>NOTE 4: <i>The load to SV9300 increases if small amount of data (such as 1 minute) is set. Therefore, it is recommended to set the value more than 1 hour.</i></p>	
	<p>NOTE 5: <i>A reset of the terminal is required after this data setting.</i></p>	
END		

(2) DT700/DT800/DT900 Series Security (Security Lock)

DT700/DT800/DT900 Series Security Mode enables the DT700/DT800/DT900 Series to be locked by pressing Security key on the terminal. So the terminal operation can be restricted by this service. To register the Security Lock for DT700/DT800/DT900 Series, do the following programming.

NOTE: *DT800/DT900 Series does not equip with Security key on the terminal. To use this function, assign Security key to a Line/Trunk feature key by CM90 Y=00: F5036 in advance. For details, refer to "Multiline Terminal" of the Programming Manual.*

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required stations.	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Allow the Security Lock for DT700/DT800/DT900 Series. NOTE: <i>Reboot (re-login to SV9300) of the terminal is required when this date is changed while the system is operating.</i>	<ul style="list-style-type: none"> • Y=199 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0 : Allow 1◀: Restricted
CM42	Assign the number of password entries when the Security Lock is canceled. NOTE: <i>When entering incorrect password for the number of times set by this command, DT700/DT800/DT900 Series will be in password lock state.</i>	<ul style="list-style-type: none"> (1) 194 (2) 00 : No limit 01-99 : 1-99 times NONE◀: 3 times
	Assign the time to lock the DT700/DT800/DT900 Series when the number of password entries exceeded the limit. NOTE: <i>When the DT700/DT800/DT900 Series is in password lock state, a password entry is restricted until the time set by this command is elapsed.</i>	<ul style="list-style-type: none"> (1) 195 (2) 00 : Not unlocked 01-99 : 1-99 minutes NONE◀: 10 minutes
END		

- (3) DT700/DT800/DT900 Series Security Lock Cancel (Security Lock Cancel)
To cancel the Security Lock forcibly for DT700/DT800/DT900 Series, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM12</div>	<p>Cancel the Security Lock for DT700/DT800/DT900 Series.</p> <p>NOTE: <i>When entering “CCC”, regardless of the result, “OK” is displayed immediately.</i></p>	<ul style="list-style-type: none"> • Y=93 (1) X-XXXXXXXX: Station No. (2) CCC: To cancel (forced)
END		

- (4) DT700/DT800/DT900 Series menu time out timer data assignment
To assign DT700/DT800/DT900 Series menu time out timer, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM12</div>	<p>Assign DT700/DT800/DT900 Series menu time out timer.</p> <p>NOTE: <i>Set the second data to “00” and reset the station when using DTPlusWare.</i></p>	<ul style="list-style-type: none"> • Y=76 (1) X-XXXXXXXX: Station No. (2) 00 : No time out 01-16 : 1-16 minutes NONE◀: 4 minutes
END		

(5) DT900 Series LCD Display Mode data assignment
[9300V7 software required]

To set the LCD display mode for DT900 Series, do the following programming.

START	DESCRIPTION	DATA
CM12	Specify the LCD display mode to each station (for Portal mode support terminal). NOTE: <i>A reset of the terminal is required when this data is set or changed.</i>	<ul style="list-style-type: none"> • Y=102 (1) X-XXXXXXXX: Station No. (2) 0 : Classic mode 1 : Portal mode NONE◀: As per CM08>2002
CM08	Specify the LCD display mode for the system (for Portal mode support terminal). NOTE: <i>A reset of the terminal is required when this data is set or changed.</i>	<ul style="list-style-type: none"> (1) 2002 (2) 0 : Classic mode 1◀: Portal mode
END		

(6) DT900 Series Terminal Phonebook data assignment
[9300V7 software required]

To use the Terminal Phonebook feature for DT900 Series, do the following programming.

START	DESCRIPTION	DATA
CM13	Specify whether to provide the Terminal Phonebook feature. NOTE 1: <i>To use the Terminal Phonebook feature, download the Terminal Phonebook data after setting or changing this data. For details, refer to "Terminal Phonebook" of the PC Programming Manual.</i> NOTE 2: <i>A reset of the terminal is required when this data is set or changed.</i>	<ul style="list-style-type: none"> • Y=100 (1) X-XXXXXXXX: Station No. (2) 0 : Available 1◀: Not available
END		

(7) DT900 Series Portal Mode data assignment

[9300V7 software required]

To set the data related to the Portal Mode for DT900 Series, do the following programming in addition to the “(5) DT900 Series LCD Display Mode data assignment” [Page 2-29](#), “(6) DT900 Series Terminal Phonebook data assignment” [Page 2-29](#).

NOTE: The following system data programming is supported for 9300V7 (V7.2.0) software or later.

START	DESCRIPTION	DATA
CM12	<p>Specify the Home screen in the Portal Mode to the required stations.</p> <p>NOTE 1: When the second data 2 is assigned for DT930 other than Touch Panel model, the Favorite Screen is set as the Home Screen.</p> <p>NOTE 2: A reset of the terminal is required when this data is set or changed.</p>	<ul style="list-style-type: none"> • Y=103 (1) X-XXXXXXXX: Station No. (2) 0 : Favorite Screen 1 : Call Screen 2 : Line Screen NONE◀: As per CM08>2003
CM08	<p>Specify the Home screen in the Portal Mode for the system.</p> <p>NOTE: When the second data 02 is assigned for DT930 other than Touch Panel model, the Favorite Screen is set as the Home Screen.</p> <p>NOTE: A reset of the terminal is required when this data is set or changed.</p>	<ul style="list-style-type: none"> (1) 2003 (2) 01 : Call Screen 02 : Line Screen NONE◀: Favorite Screen
CM12	<p>Assign the automatic change time of the display to Line Screen for Multiline Terminal (for Portal Mode).</p> <p>NOTE: A reset of the terminal is required when this data is set or changed.</p>	<ul style="list-style-type: none"> • Y=104 (1) X-XXXXXXXX: Station No. (2) 0 : Not changed 1-5 : 1-5 second(s) later NONE◀: Immediately
END		

SIP Wireless Terminal (MH240) Data Assignment

In addition to the IP Station Basic Data Assignment, to accommodate MH240 as an IP Station, do the following programming.

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class for the MH240 to the required stations.	<ul style="list-style-type: none"> • Y=02 (1) X-XXXXXXXX: Station No. (2) XXZZ XX: 00-15◀: Service Restriction Class A ZZ : 00-15◀: Service Restriction Class B
CM15	Restrict the Voice Call in Service Restriction Class B assigned by CM12 Y=02. NOTE: <i>Set the second data to "0" when MH240 is used.</i> Restrict the Internal Zone Paging Access/All Zone Internal Paging in Service Restriction Class A assigned by CM12 Y=02. NOTE: <i>Set the second data to "0" when MH240 is used.</i>	<ul style="list-style-type: none"> • Y=067 (1) 00-15: Service Restriction Class B assigned by CM12 Y=02 (2) 0: Restricted <ul style="list-style-type: none"> • Y=049 (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 0: Restricted
<u>END</u>		

I755 IP DECT Terminal Data Assignment

[9300V3 software required]

To call back from the call history of I755 IP DECT Terminal by adding a trunk access code to a calling number when an external call is terminated to the I755 IP DECT Terminal, do the following programming in addition to the IP Station Basic Data Assignment.

START	DESCRIPTION	DATA
CM13	<p>Add the trunk access code to the calling number when an external call is terminated to the I755 IP DECT Terminal.</p> <p>NOTE: <i>Do not set this command when adding a trunk access code for other services or on a terminal side.</i></p>	<ul style="list-style-type: none"> • Y=87 (1) X-XXXXXXXX: IP Station No. assigned by CM10 Y=01 (2) 0: To add
CM35	<p>Assign the trunk access code to be added when storing call histories to the I755 IP DECT Terminal.</p> <p>NOTE: <i>The trunk access code assigned by this command is used for the following services.</i></p> <ul style="list-style-type: none"> • <i>Voice Mail Transfer</i> • <i>Live Recording</i> • <i>Call origination by ISDN trunk key</i> • <i>Call termination to a Caller ID station</i> • <i>Call termination to a Standard SIP Terminal</i> • <i>Call Back to Mobile Phone</i> 	<ul style="list-style-type: none"> • Y=189 (1) 00-63: Trunk Route No. (2) X-XX: Trunk Access Code to be added X : 0-9, A (*), B (#) NONE◀: No data
END		

To not indicate the Voice Mail icon on I755 IP DECT terminal when the service feature (Incoming Call History (CID Call Back), Message Reminder and so on) that turns on MW lamp except for Voice Mail is executed, do the following programming.

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required station (I755 IP DECT terminal).	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Assign the MW lamp control data to each station (I755 IP DECT terminal).	<ul style="list-style-type: none"> • Y=284 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0: Not lit • Y=285 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 1◀: To light • Y=286 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0 : Not lit 1◀: To light
<u>END</u>		

IP Single Line Telephone (SIP) Data Assignment

This section explains the following data assignment to accommodate Standard SIP station.

- (1) SIP Converter data assignment : [Page 2-35](#)
- (2) Standard SIP station data assignment : [Page 2-37](#)
- (3) Authentication data assignment for Standard SIP station : [Page 2-39](#)
- (4) Session timer data assignment for Standard SIP station : [Page 2-41](#)
- (5) Periodic registration timer assignment for Standard SIP station : [Page 2-42](#)
- (6) Release timer data assignment for Standard SIP station : [Page 2-43](#)
- (7) Reverse signal data assignment for Standard SIP station : [Page 2-43](#)
- (8) Operation settings on a calling side when the call is disconnected by Standard SIP station while incoming call is receiving : [Page 2-44](#)
- (9) Hold Tone source settings for Standard SIP station : [Page 2-44](#)
- (10) Standard SIP station connection status confirmation : [Page 2-46](#)
- (11) Data assignment for Standard SIP station forced idle return : [Page 2-48](#)
- (12) Station Hunting/Announcement/Call Forwarding settings for Standard SIP station : [Page 2-49](#)
- (13) Announcement settings when Standard SIP station is no answer : [Page 2-55](#)
- (14) OAI data assignment for Standard SIP station : [Page 2-57](#)
- (15) Data assignment for Standard SIP Terminal without REGISTER : [Page 2-58](#)
- (16) Peer-to-Peer/VoIPDB connection settings for Standard SIP station : [Page 2-61](#)
- (17) Data assignment for accommodating the Standard SIP station in Remote Units : [Page 2-63](#)
- (18) Data assignment for Call Back from call history of Standard SIP station : [Page 2-68](#)
[9300V3 software required]
- (19) Data assignment for Three-Party Conference/Broker's Call with Standard SIP Terminal : [Page 2-69](#)
[9300V3 software required]
- (20) Operation settings when Standard SIP station with a held call goes on-hook : [Page 2-70](#)
[9300V3 software required]

(1) SIP Converter data assignment

To accommodate Standard SIP station, the registration of SIP Converter is required. To register SIP Converter, do the following programming.

START	DESCRIPTION	DATA
CM10	<p>Assign the SIP Converter No. to the Virtual Port number for Standard SIP station.</p> <p style="text-align: right;">(RESET)</p> <p>NOTE 1: <i>Simultaneous call by three Standard SIP stations is available per SIP converter.</i></p> <p>NOTE 2: <i>A reset of the Standard SIP station is required to enable this command setting after the System Reset.</i></p>	<ul style="list-style-type: none"> • Y=01 (1) 0000-1535: Virtual Port No. (2) EECXXX XXX : 000-127 (SIP Converter No.) NONE◀: No data
CM0B	<p>Assign the UDP base port number for SIP Converter.</p> <p style="text-align: right;">(RESET)</p> <p>NOTE 1: <i>This port No. is applied as the port No. of the destination of message from Standard SIP station.</i></p> <p>NOTE 2: <i>Be aware that the UDP port No. with this command must not overlap the following port numbers:</i></p> <ul style="list-style-type: none"> • DT700/DT800/DT900 Series/SP350 (CM0B Y=1XX>32 [default: 5080]) • SIP trunk (CM0B Y=1XX>36 [default: 5060]) 	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 33: UDP Base Port for SIP Converter (2) 01024-65534: UDP Port No. NONE◀ : 5070
CMBC	<p>Specify whether the authentication for each call is allowed or not.</p> <p>NOTE 1: <i>For normal operation, be sure to set the second data to "1" (Allowed) to restrict the unauthorized connection.</i></p> <p>NOTE 2: <i>The authentication for each call is not provided when the authentication is restricted by CM1D Y=32.</i></p>	<ul style="list-style-type: none"> • Y=05 (1) 1XX: SIP Converter ID [1] + Unit No. [01-50]) (2) 0 : Restricted 1◀: Allowed
A		

A	DESCRIPTION	DATA
CMBC	Specify the Type of Service (TOS) field precedence of SIP Converter Control Packet.	<ul style="list-style-type: none"> • Y=06 (1) 1XX: SIP Converter ID [1] + Unit No. [01-50]) (2) 0-7 : PRECEDENCE 0-7 for control packet NONE◀: PRECEDENCE 6
	RESET	
	NOTE: <i>DiffServ Code Point (DSCP) assigned by CMBC Y=07 is ineffective when this command is set.</i>	
	Specify the DiffServ Code Point (DSCP) of SIP Converter Control Packet.	<ul style="list-style-type: none"> • Y=07 (1) 1XX: SIP Converter ID [1] + Unit No. [01-50]) (2) 00-FE : DSCP of control packet NONE◀: No data
	RESET	
	NOTE: <i>Type of Service (TOS) assigned by CMBC Y=06 is ineffective when this command is set. Clear (CCC) this data to enable TOS field assigned by CMBC Y=06.</i>	
END		

(2) Standard SIP station data assignment

To accommodate Standard SIP station, do the following programming.

START	DESCRIPTION	DATA
CM10	Assign the Standard SIP station number.	<ul style="list-style-type: none"> • Y=04 (1) 000-1023: Virtual Port No. (2) X-XXXXXXXX: Standard SIP Station No. NONE◀ : No data
CM1D	<p>Set the terminal type to “Standard SIP station” for the station number assigned by CM10 Y=04.</p> <p>Set the terminal type to the Standard SIP station the number assigned by CM10 Y=04.</p>	<ul style="list-style-type: none"> • Y=14 (1) X-XXXXXXXX: Standard SIP Station No. (2) 0: Standard SIP station <ul style="list-style-type: none"> • Y=15 (1) X-XXXXXXXX: Standard SIP Station No. (2) 05: Standard SIP Station
CM12	Assign Service Restriction Class for Standard SIP station to the required stations.	<ul style="list-style-type: none"> • Y=02 (1) X-XXXXXXXX: Standard SIP Station No. (2) XXZZ XX: 00-15◀: Service Restriction Class A ZZ : 00-15◀: Service Restriction Class B
CM15	<p>Specify whether Calling Name Display-Standard SIP station is allowed or not.</p> <p>NOTE: <i>Whether calling name is properly displayed or not depends on the specification of Standard SIP Terminal.</i></p>	<ul style="list-style-type: none"> • Y=123 (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 0 : Allowed 1◀: Restricted
CM08	<p>Specify the kind of Tone when a service is set by access code from Standard SIP station.</p> <p>Specify whether to provide the feature access code by Overlap Sending from Standard SIP station.</p> <p>[9300V3 software required]</p> <p>NOTE: <i>This command is valid only when the second data of CM08>1035 is set to 1 (Service Set Tone (SST)).</i></p>	<ul style="list-style-type: none"> (1) 1035 (2) 0 : Hearing RBT 1◀: Service Set Tone (SST) <ul style="list-style-type: none"> (1) 1046 (2) 0 : Not available 1◀: Available
A		

A	DESCRIPTION	DATA
CM13	<p>Specify whether the Message Waiting is provided to the Standard SIP station assigned by CM10 Y=04.</p> <p>NOTE: <i>This feature must be supported on the Standard SIP Terminal side when using a Message Waiting feature.</i></p>	<ul style="list-style-type: none"> • Y=03 (1) X-XXXXXXXX: Standard SIP Station No. (2) 0 : To provide 1 ◀: Not provided
	<p>Set the Request-URI field included in a SIP request message for Standard SIP station. [9300V3 software required] [North America Only]</p> <p>NOTE: <i>A reset of Standard SIP Terminal is required after this data setting.</i></p>	<ul style="list-style-type: none"> • Y=85 (1) X-XXXXXXXX: Standard SIP Station No. (2) 0 : Contact header user field of REGISTER message 1 ◀: Standard SIP station No.
	<p>Specify whether to add Server header and User-Agent header for Standard SIP station.</p>	<ul style="list-style-type: none"> • Y=94 (1) X-XXXXXXXX: Standard SIP station No. (2) 0 : Not added 1 ◀: To add
<u>END</u>		

(3) Authentication data assignment for Standard SIP station

To provide the authentication to Standard SIP station, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM1D</div>	<p>Allow the authentication to the Standard SIP station.</p> <p>NOTE 1: <i>Be sure to set the second data to "03" (Restricted) when not executing the authentication.</i></p> <p>NOTE 2: <i>Be sure to set the second data to "15" (Allowed) when using Brute-force Login Attempt Protection.</i></p> <p>NOTE 3: <i>To execute the authentication, settings of user name and password for Standard SIP station are required.</i></p> <ul style="list-style-type: none"> • <i>User name: set the Standard SIP station number assigned by CM10 Y=04.</i> • <i>Password: set the password assigned by CM2B Y=00.</i> <p>NOTE 4: <i>A reset of the Standard SIP Terminal is required (Power ON/OFF of the terminal) when this data is set/changed.</i></p>	<ul style="list-style-type: none"> • Y=32 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 03 : Restricted 15◀: Allowed
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM42</div>	<p>Specify the maximum number of digits for Standard SIP registration password.</p> <p>NOTE: <i>The number of digits assigned by this command is the same as that of assigned for Station Authorization Code and IP Multiline Terminal.</i></p>	<ul style="list-style-type: none"> (1) 73 (2) 01-08 : 1-8 digits NONE◀: 4 digits
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">A</div>		

A	DESCRIPTION	DATA
CM2B	Assign a Station Authorization Code to each Standard SIP station.	<ul style="list-style-type: none"> • Y=00 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) X-XXXXXXXX: Password X : 0-9, A (*), B (#) NONE◀: 0000
	NOTE 1: <i>The number of digits of the second data depends on the setting of CM42>73.</i>	
	NOTE 2: <i>To let the Standard SIP station function when no data is set for a authentication (such as conducting test operation by maintenance personnel), set "0000" (4 digits) as a password regardless of the setting of CM42>73.</i>	
	NOTE 3: <i>The setting of the Standard SIP station number is not available when originating/terminating a call or during a call ("WAIT BUSY NOW" is displayed).</i>	
	NOTE 4: <i>A reset of the Standard SIP Terminal is required (Power ON/OFF of the terminal) when this data is set/changed.</i>	
<u>END</u>		

(4) Session timer data assignment for Standard SIP station

To assign the session timer for the Standard SIP station, do the following programming.

NOTE 1: *This data assignment is effective when the standard SIP Terminal supports the “UPDATE” message.*

NOTE 2: *When executing the session monitoring of Standard SIP Terminal which does not support the “UPDATE” message, do the programming with “(5) Periodic registration timer assignment for Standard SIP station”.*

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required Standard SIP stations.	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Standard SIP Station No. (2) 00-15◀: Service Restriction Class C
CM15	Allow the session timer feature in Service Restriction Class C assigned by CM12 Y=07. NOTE: <i>When CM15 Y=489: 0 (allow) is set for the office which is monitoring the session by periodic registration, it is recommended to set the timer value of periodic registration (CM41 Y=0>151/152/153) to default (1 hour).</i>	<ul style="list-style-type: none"> • Y=489 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0: Allow
CM41	Assign the session timer of the Standard SIP station. NOTE 1: <i>Assign the same data as the session timer value assigned on the terminal side.</i> NOTE 2: <i>A reset of the Standard SIP Terminal is required (Power ON/OFF of the terminal) when this data is set/changed.</i>	<ul style="list-style-type: none"> • Y=0 (1) 166 (2) 00 : 1 day (1440 minutes) 01 : 90 seconds 02-99 : 2-99 minutes NONE◀: 5 minutes
<u>END</u>		

- (5) Periodic registration timer assignment for Standard SIP station
To assign the periodic registration timer for the Standard SIP station, do the following programming.

START	DESCRIPTION	DATA
CM41	<p>Assign the periodic registration timer for Standard SIP station.</p> <p style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 10px;">RESET</p> <p>NOTE 1: <i>A reset of the Standard SIP Terminal is required (Power ON/OFF of the terminal) when this data is set/changed.</i></p> <p>NOTE 2: <i>This data is set by the combination of CM41 Y=0>151-153 (day/hour/minute). When CM41 Y=0>151-153 (day/hour/minute) are all set to NONE, the expire value "1 hour" is set as default.</i></p> <p>NOTE 3: <i>The minimum value of the periodic registration timer for Standard SIP station is "3 minutes" (when less than 3 minutes is set to the second data of CM41 Y=0>151-153 respectively, "1 hour" is set as the periodic registration timer).</i></p>	<ul style="list-style-type: none"> • Y=0 (1) 151: REGISTER (Day) (2) 00-99 : 0-99 days NONE ◀: NOTE 2 (1) 152: REGISTER (Hour) (2) 00-23 : 0-23 Hours NONE ◀: NOTE 2 (1) 153: REGISTER (Minute) (2) 00-59 : 0-59 Minutes NONE ◀: NOTE 2
END		

(6) Release timer data assignment for Standard SIP station

To assign the release timer when calling to the Standard SIP station for long call duration, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM41</div>	Assign the release timer when calling to Standard SIP station for a long call duration. NOTE: <i>This data is reflected immediately after the data setting. However, the set timer before change remains as for a preset Long Call Duration Release Timer for Standard SIP station.</i>	<ul style="list-style-type: none"> • Y=0 (1) 155: Long Call Duration Release Timer for Standard SIP station (2) 00 : Long Call Duration Release Timer for Standard SIP station is ineffective 03-99 : 3-99 minutes NONE◀: 30 minutes
<u>END</u>		

(7) Reverse signal data assignment for Standard SIP station

To assign the timing until sending the reverse signal to the Standard SIP station for connecting the line, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM41</div>	Assign the timing until sending the reverse signal to the Standard SIP station for connecting the line. NOTE: <i>This timer data is used for setting the timing of pseudo-answer signal to Standard SIP station when a replying signal from the opposite system is not received after sending an outgoing call from Standard SIP station.</i>	<ul style="list-style-type: none"> • Y=0 (1) 159 (2) 00 : Not sent 01-99 : 4-396 seconds (4 seconds increments) NONE◀: Not sent
<u>END</u>		

(8) Operation settings on a calling side when the call is disconnected by Standard SIP station while incoming call is receiving.

To specify the operation on a calling side when the call is disconnected by Standard SIP station while the incoming call is receiving, do the following programming.

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required Standard SIP stations.	<ul style="list-style-type: none"> Y=07 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 00-15◀: Service Restriction Class C
CM15	In Service Restriction Class C assigned by CM12 Y=07, specify whether calling station side is disconnected or continued when the call is disconnected by Standard SIP station while incoming.	<ul style="list-style-type: none"> Y=292 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0 : Continued 1◀: Disconnected
CM35	In Service Restriction Class C assigned by CM12 Y=07, specify whether calling trunk side is disconnected or continued when the call is disconnected by Standard SIP station while incoming.	<ul style="list-style-type: none"> Y=359 (1) 00-63: Trunk Route No. (2) 0 : Disconnected (To be charged due to an instant connection) 3◀: Continued
END		

(9) Hold Tone source settings for Standard SIP station.

(a) When hearing Hold Tone source on CPU blade of SV9300 (for system).

START	DESCRIPTION	DATA
CM08	Select the Hold Tone source on CPU blade (selected by CM48) for Standard SIP station.	(1) 1007 (2) 1◀: Hold Tone Source on CPU blade (selected by CM48 Y=0)
CM48	Specify the Hold Tone Sending pattern.	<ul style="list-style-type: none"> Y=0 (1) 00: C.O. Line 01: Tie Line 02: Station (2) 0000 : No Tone 0500 : Hold Message 1400 : Hold Tone Source on CPU blade NONE◀: Hold Tone Source on CPU blade
END		

(b) When hearing Hold Tone source on CPU blade of SV9300 (for each station).

START	DESCRIPTION	DATA
CM08	Select the Hold Tone source on Standard SIP station.	(1) 1007 (2) 0: Hold Tone Source on Standard SIP station
CM13	Select the Hold Tone source on CPU blade (selected by CM48 Y=0) for Standard SIP station.	<ul style="list-style-type: none"> Y=74 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 0: Hold Tone Source on CPU blade (selected by CM48 Y=0)
CM48	Specify the Hold Tone Sending pattern.	<ul style="list-style-type: none"> Y=0 (1) 00: C.O. Line 01: Tie Line 02: Station (2) 0000 : No Tone 0500 : Hold Message 1400 : Hold Tone Source on CPU blade NONE◀: Hold Tone Source on CPU blade
END		

(c) When hearing External Hold Tone source on CPU blade of SV9300 (for system/each tenant). Set the External Hold Tone source. For details, refer to the “MUSIC ON HOLD” in the “Programming Manual”.

(d) When hearing Hold Tone source on Standard SIP station.

START	DESCRIPTION	DATA
CM08	Select the Hold Tone source on Standard SIP station.	(1) 1007 (2) 0: Hold Tone Source on Standard SIP Station
CM13	Select the Hold Tone source specified by CM08>1007.	<ul style="list-style-type: none"> Y=74 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 1◀: As per CM08>1007
END		

(10) Standard SIP station connection status confirmation

To read the terminal type of Standard SIP station and confirm the Standard SIP station connection status, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CMFA</div>	<p>Read the terminal type of the Standard SIP station.</p> <p>NOTE 1: <i>The terminal type is read as "12"(DT710) when using 9300V3 software or before.</i></p> <p>NOTE 2: <i>The terminal type is read as "13"(DT730/DT710 (Self-Labeling)) when using 9300V3 software or before.</i></p> <p>NOTE 3: <i>The terminal type is read as "22"(DT830CG) when using 9300V6 software or before.</i></p> <p>NOTE 4: <i>The terminal type is read as follows.</i> - When using 9300V3 STEP2 to 9300V6 software: "23" (DT820) - When using 9300V3 software or before: "12" (DT710)</p>	<ul style="list-style-type: none"> • Y=01 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 02 : D^{term}IP INASET 03 : D^{term}85 (D^{term} Series i) (IP adapter type) 05 : D^{term}85 (D^{term} Series i) (IP Bundled type) 10 : D^{term}SP30 12 : DT710 13 : DT730/DT710 (Self-Labeling) 14 : DT750 15 : MH240 16 : SP350 17 : DT730DG 18 : DT730CG 19 : DT770G 21 : DT830/DT830DG 22 : DT830CG 23 : DT820 NOTE 1 [9300V3 STEP2 software required] 24 : DT820 (Self-Labeling) NOTE 2 [9300V3 STEP2 software required] 25 : DT930CG NOTE 3 [9300V7 software required] 26 : DT920 NOTE 4 [9300V7 software required] 27 : DT920 (Self-Labeling) [9300V7 software required] 28 : DT930 (Touch Panel) [9300V7 software required] 41 : D^{term}85 (D^{term} Series i) 42 : D^{term}85 (D^{term} Series i) Russian 43 : DT310 44 : DT330 45 : DT330 Chinese
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">A</div>		

A	DESCRIPTION	DATA
CMFA	<p>NOTE 5: DT510 is available from 9300V9 software or later. The terminal type between DT510 and DT410 cannot be identified by the system data. Also, DT510 is identified as DT410 when using 9300V6 software or before (DT510 cannot be accommodated to SV9300 when using 9300V7/V8 software).</p> <p>Read the Standard SIP station status.</p> <p>NOTE: The following status is displayed if the Standard SIP station number is entered to the first data.</p> <ul style="list-style-type: none"> When the terminal is connected to the system: IP address and connection status (A: Busy) When the terminal is not connected to the system: "NONE" 	<p style="text-align: right;">NOTE 5</p> <p>46 : DT510/DT410</p> <p>47 : DT430</p> <p>48 : DT430 Chinese</p> <p>49 : DT530</p> <p style="text-align: right;">[9300V7 software required]</p> <p>50 : DT530 Chinese</p> <p style="text-align: right;">[9300V7 software required]</p> <p>70 : Paging Adapter</p> <p>80 : Standard SIP station</p> <p>FF : Other</p> <p>NONE : Logout Status/Disconnected</p> <p>• Y=02</p> <p>(1) X-XXXXXXXX:Standard SIP Station No. assigned by CM10 Y=04</p> <p>(2) XX...X: Z</p> <p>XX...X : IP Address of Standard SIP Terminal</p> <p>Z : Standard SIP Terminal status (A/N)</p> <p>A: Busy</p> <p>N: Idle</p> <p>FF : The terminal is not IP Station</p> <p>NONE◀: IP Station is logout status/ IP Station has never been busy</p>

END

(11) Data assignment for Standard SIP station forced idle return

To read the connection status of the Standard SIP station and execute the forced idle return, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM12</div>	<p>Read the Standard SIP station connection status (Only display) and execute the forced idle return of the station.</p> <p>NOTE 1: <i>An forced idle return is executed even if the specified Standard SIP station is during a call. Then, the call is disconnected.</i></p> <p>NOTE 2: <i>Re-registration of the Standard SIP Terminal by CMBC Y=15 is required after the forced idle return.</i></p> <p>NOTE 3: <i>“NONE” is displayed when the station assigned to the first data is other than a Standard SIP station. “DATA NOT FOUND” is displayed for unassigned numbers.</i></p> <p>NOTE 4: <i>Entering “CCC” when unused Standard SIP station number (CM12 Y=96: 0) is assigned, “DATA ERROR” is displayed (an forced idle return is not executed).</i></p>	<ul style="list-style-type: none"> • Y=96 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 0 : Not connected 1 : Connecting 2 : Connected (PUSH Notification) [from 9300V6 to 9300V8 software] 3 : Connected (PUSH Notification: terminal info not registered) [9300V9 software required] 4 : Connected (PUSH Notification: terminal info registered) [9300V9 software required] CCC : Idle return (forced) NONE◀: Other than Standard SIP station
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">END</div>		

(12) Station Hunting/Announcement/Call Forwarding settings for Standard SIP station

To provide Station Hunting/Announcement/Call Forwarding when the called Standard SIP station is in an off hook state, the power is off, or the cable is pulled out, do the following programming.

NOTE: *The service priority of the Station Hunting/Announcement/Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out and Announcement-Standard SIP station No Answer is as follows.*

*High priority: Station Hunting-Standard SIP station Off Hook/Power Off/Cable Pulled Out
Announcement-Standard SIP station Off Hook/Power Off/Cable Pulled Out
Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out*

↑
↓

Low priority : Announcement-Standard SIP station No Answer

Example:

Station Hunting functions when both Station Hunting and Call Forwarding are set.

- (a) To provide Station Hunting when the called Standard SIP station is in an off hook state, the power is off, or the cable is pulled out:

START	DESCRIPTION	DATA
CM08	Provide the Station Hunting-Standard SIP station Off Hook/Power Off/Cable Pulled Out.	(1) 510 (2) 0: Available
	Specify whether the RBT is sent when the called Standard SIP station is in an off hook state, the power is off, or the cable is pulled out.	(1) 503 (2) 0 : Not sent 1 ◀: To send
	NOTE: <i>CM08>503 is effective only for Station-to-Station call.</i>	

A

A	DESCRIPTION	DATA
CM18	<p>Assign the Standard SIP station numbers included in the station hunting group, one by one, as shown below.</p> <p><u>Station Hunting-Terminal</u> 1st Operation : (1) Station A (2) Station B 2nd Operation: (1) Station B (2) Station C</p> <p><u>Station Hunting-Circular</u> 1st Operation : (1) Station A (2) Station B 2nd Operation: (1) Station B (2) Station C 3rd Operation : (1) Station C (2) Station A</p> <p>Specify the kind of station included in the station hunting group.</p>	<ul style="list-style-type: none"> • Y=0 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 <ul style="list-style-type: none"> • Y=1 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 0◀: Member station of Station Hunting-Terminal 1 : Pilot station of Station Hunting-Terminal All stations of Station Hunting-Circular
CM41	<p>Specify the timer for detecting Standard SIP station Off Hook/Power Off/Cable Pulled Out.</p> <p>NOTE: <i>The second data must be set within "01" (0-4 seconds) to "07" (24-28 seconds) for Standard SIP station (this timer is not effective when the second data "08" or higher is set).</i></p>	<ul style="list-style-type: none"> • Y=0 (1) 85 (2) 01-07: 4-28 seconds (4 second increments) <p>If no data is set, the default setting is 8-12 seconds.</p>
END		

- (b) To provide an announcement when the called Standard SIP station is in an off hook state, the power is off, or the cable is pulled out:

START	DESCRIPTION	DATA
CM12	Assign a tenant number to a Standard SIP station number.	<ul style="list-style-type: none"> Y=04 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 00 : Tenant 00 01◀-63: Tenant 01-63
CM41	<p>Specify the message duration for Announcement-Standard SIP station Off Hook/Power Off/Cable Pulled Out.</p> <p>Specify the timer for detecting Standard SIP station Off Hook/Power Off/Cable Pulled Out.</p> <p>NOTE: <i>The second data must be set within "01" (0-4 seconds) to "07" (24-28 seconds) for Standard SIP station (this timer is not effective when the second data "08" or higher is set).</i></p>	<ul style="list-style-type: none"> Y=0 (1) 84 (2) 01-99: 4-396 seconds (4 second increments) <p>If no data is set, the default setting is 116-120 seconds.</p> <ul style="list-style-type: none"> Y=0 (1) 85 (2) 01-07: 4-28 seconds (4 second increments) <p>If no data is set, the default setting is 8-12 seconds.</p>
CM48	Provide the Announcement Service.	<ul style="list-style-type: none"> Y=5 (1) 02: Standard SIP station Off Hook/Power Off/Cable Pulled Out (2) 0500: To provide
CM49	Assign the function for each Voice Response System.	<ul style="list-style-type: none"> Y=00 (1) XXX: 000-015: VRS No. (2) 1500-1563: Message Group No. 00-63 for Announcement-Standard SIP station Off Hook/Power Off/Cable Pulled Out
	Assign the message group for Standard SIP station Off Hook/Power Off/Cable Pulled Out to each tenant.	<ul style="list-style-type: none"> Y=12 (1) 00-63: Tenant No. (2) 00-63: Message Group No.
A		

To record/replay/delete a message:

A	DESCRIPTION	DATA
CM12	Assign Service Restriction Class A to a certain station.	<ul style="list-style-type: none"> • Y=02 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) XX ZZ XX: 00-15◀: Service Restriction Class A
CM15	Allow Service Restriction Class A for Announcement Service.	<ul style="list-style-type: none"> • Y=033 Voice Response System Access: Record/Replay/Delete (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 1◀: Allow
CM29	Assign a Numbering Plan Group to each tenant.	<ul style="list-style-type: none"> (1) 00-63: Tenant No. (2) 710-713 : Numbering Plan Group 0-3 NONE◀: Numbering Plan Group 0
CM20	Assign the access codes to record/replay/delete the message.	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A100: Record A101: Replay A102: Delete
<u>END</u>		

- (c) To provide Call Forwarding when the called Standard SIP station is in an off hook state, the power is off, or the cable is pulled out:

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required Standard SIP stations.	<ul style="list-style-type: none"> Y=07 <ol style="list-style-type: none"> X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 00-15◀: Service Restriction Class C
CM15	Allow Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out in Service Restriction Class C assigned by CM12 Y=07.	<ul style="list-style-type: none"> Y=481 <ol style="list-style-type: none"> 00-15: Service Restriction Class C assigned by CM12 Y=07 03◀: Allow (Send RBT when no destination is set)
CM20	Assign the access code for Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out, which is used to set from Standard SIP station, Set and Cancel, respectively.	<ul style="list-style-type: none"> Y=0-3 Numbering Plan Group 0-3 <ol style="list-style-type: none"> X-XXXX: Access Code A241: Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out Set A242: Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out Cancel
CME6	When setting the destination of Call Forwarding-Standard SIP station Off Hook/Power Off/Cable Pulled Out from PCPro/CAT, set the destination to the station number assigned by CM10 Y=04.	<ul style="list-style-type: none"> Y=06 <ol style="list-style-type: none"> X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 Destination No.: X-XXXXXXXX: Station No. E000 : Desk Console YY...Y: Outgoing Trunk Access Code (1-4 digits) + [] + Called No. (Maximum 26 digits)
CM08	Specify whether the RBT is sent when the called Standard SIP station is in an off hook state, the power is off, or the cable is pulled out.	<ol style="list-style-type: none"> 503 0 : Not sent 1◀: To send
	NOTE: <i>CM08>503 is effective only for Station-to-Station call.</i>	
A		

A	DESCRIPTION	DATA
CM41	<p>Specify the timer for detecting Standard SIP station Off Hook/Power Off/Cable Pulled Out.</p> <p>NOTE: <i>The second data must be set within "01" (0-4 seconds) to "07" (24-28 seconds) for Standard SIP station (this timer is not effective when the second data "08" or higher is set).</i></p>	<ul style="list-style-type: none">• Y=0 <p>(1) 85 (2) 01-07: 4-28 seconds (4 second increments)</p> <p>If no data is set, the default setting is 8-12 seconds.</p>
<u>END</u>		

(13) Announcement settings when Standard SIP station is no answer

To provide an announcement when the called Standard SIP station is no answer, do the following programming

START	DESCRIPTION	DATA
CM12	Assign a tenant number to a Standard SIP station number.	<ul style="list-style-type: none"> Y=04 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 00 : Tenant 00 01◀-63: Tenant 01-63
CM08	Provide Standard SIP station No-Answer feature. Specify the type of No-Answer Timer.	<ul style="list-style-type: none"> (1) 504 (2) 0: Available (1) 085 (2) 0 : No-Answer timer (CM41 Y=0>86) 1◀: No-Answer timer (CM41 Y=0>01)
CM41	Specify the message reply timer for Announcement-Standard SIP station No Answer. Specify the duration of message for Announcement-Standard SIP station No Answer.	<ul style="list-style-type: none"> Y=0 (1) 01 (2) 01-30: 4-120 seconds (4 second increments) If no data is set, the default setting is 32-36 seconds. Y=0 (1) 75 (2) 01-99: 4-396 seconds (4 second increments) If no data is set, the default setting is 116-120 seconds.
	Specify the message reply timer for Announcement-Standard SIP station No Answer. NOTE: <i>CM41 Y=0>86 is effective only when CM08>085 is set to 0.</i>	<ul style="list-style-type: none"> Y=0 (1) 86 (2) 01-99: 4-396 seconds (4 second increments) If no data is set, the default setting is 36-40 seconds.
A		

A	DESCRIPTION	DATA
CM48	Provide the Announcement Service.	<ul style="list-style-type: none"> • Y=5 (1) 00: Standard SIP station No-Answer (2) 0500: To provide
CM49	Assign the function for each Voice Response System.	<ul style="list-style-type: none"> • Y=00 (1) XXX: 000-015: VRS No. (2) 1300-1363: Standard SIP station No-Answer Message Group No. 00-63
	Assign the message group for Standard SIP station No Answer to each tenant.	<ul style="list-style-type: none"> • Y=10 (1) 00-63: Tenant No. (2) 00-63: Message Group No.
B		

To record/replay/delete a message:

B	DESCRIPTION	DATA
CM12	Assign Service Restriction Class A to a certain station.	<ul style="list-style-type: none"> • Y=02 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) XX ZZ XX: 00-15◀: Service Restriction Class A
CM15	Allow Service Restriction Class A for Announcement Service.	<ul style="list-style-type: none"> • Y=033 Voice Response System Access: Record/Replay/Delete (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 1◀: Allow
CM29	Assign a Numbering Plan Group to each tenant.	<ul style="list-style-type: none"> (1) 00-63: Tenant No. (2) 710-713 : Numbering Plan Group 0-3 NONE◀: Numbering Plan Group 0
CM20	Assign the access codes to record/replay/delete the message.	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A100: Record A101: Replay A102: Delete
<u>END</u>		

(14) OAI data assignment for Standard SIP station

To use Standard SIP station with OAI features, do the following programming.

START	DESCRIPTION	DATA
CM08	Assign the type of Standard SIP station for OAI SMFN.	<ul style="list-style-type: none"> (1) 804 (2) 0: Single Line Telephone
<u>END</u>		

(15) Data assignment for Standard SIP Terminal without REGISTER

To assign the data for Standard SIP Terminal without REGISTER, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM12</div>	<p>Assign the IP address for Standard SIP Terminal without REGISTER.</p> <p>NOTE 1: <i>Set this data to the related station before registering Standard SIP Terminal without REGISTER by CMBC Y=15.</i></p> <p>NOTE 2: <i>When the IP address is changed or deleted by specifying the registered Standard SIP station, "WAIT, BUSY NOW" is displayed (the IP address is not changed or deleted).</i></p> <p>NOTE 3: <i>When changing the IP address assigned by this data, delete the registered terminal once by CMBC Y=15. Then register the terminal again.</i></p>	<ul style="list-style-type: none"> • Y=97 (1) X-XXXXXXXXX: Station No. of Standard SIP Terminal without REGISTER (2) XXXXXXXXXXXXX: 000000000001-25525525254: IP Address of Standard SIP Terminal without REGISTER (Maximum 12 digits) NONE◀: No data/other than Standard SIP Terminal
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">A</div>		

A	DESCRIPTION	DATA
CM12	Assign the port number for Standard SIP Terminal without REGISTER.	<ul style="list-style-type: none"> • Y=98 (1) X-XXXXXXXX: Station No. of Standard SIP Terminal without REGISTER (2) 00000-65534: Port No. of Standard SIP Terminal without REGISTER NONE◀ : 5060
B	<p>NOTE 1: <i>When using the port number other than 5060 (default), set this data to the related station before registering Standard SIP Terminal without REGISTER by CMBC Y=15.</i></p> <p>NOTE 2: <i>When the IP address is changed or deleted by specifying the registered Standard SIP station, "WAIT, BUSY NOW" is displayed (the IP address is not changed or deleted).</i></p> <p>NOTE 3: <i>When changing the port number assigned by this data, delete the registered terminal once by CMBC Y=15. Then register the terminal again.</i></p>	

B	DESCRIPTION	DATA
CMBC	Register the Standard SIP Terminal without REGISTER.	<ul style="list-style-type: none"> • Y=15 (1) 000-1023: Standard SIP Terminal ID without REGISTER (2) X-XXXXXXXX: Station No. of Standard SIP Terminal without REGISTER (1-8 digits) <p>CCC : Clear NONE◀ : No data</p>
	RESET	
	NOTE 1: <i>Whether the registered Standard SIP station number is available or not, can be confirmed by CMFA Y=02.</i>	
	NOTE 2: <i>Clear (CCC) this data once when CMFA Y=02 is in "N: Idle" state. Then set this data again.</i>	
	NOTE 3: <i>When the data for Standard SIP station or CM12 Y=97 is not assigned to the specified station in registering the terminal, "DATA ERROR" is displayed.</i>	
	NOTE 4: <i>When the terminal is deleted by specifying the Standard SIP Terminal ID while the call is in progress, "WAIT, BUSY NOW" is displayed.</i>	
	NOTE 5: <i>Standard SIP station without REGISTER can be accommodated only for the Main Unit (Unit01).</i>	
<u>END</u>		

(16) Peer-to-Peer/VoIPDB connection settings for Standard SIP station

To select the connection method in Station-to-Station connection between Standard SIP stations, do the following programming.

NOTE: *The following programming is also required for Video Phone feature for Standard SIP station.*

- (a) To select the connection method in Station-to-Station connection between Standard SIP stations with an access code:

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM20</div>	<p>Select the connection method in a Station-to-Station connection between Standard SIP stations with an access code.</p> <p>NOTE 1: <i>Dial with the access code + station number when originating a call.</i></p> <p>NOTE 2: <i>When the call is originated only by the station number (without access code), the RTP route on the calling side and the called side set by CM15 Y=229: 0/1 is applied.</i></p> <p>NOTE 3: <i>Set the second data to “A264” when using the Call Forwarding feature.</i></p> <p>NOTE 4: <i>Set the second data to “A263” (Peer-to-Peer connection) when using Video Phone feature.</i></p>	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A263: Peer-to-Peer connection A264: Peer-to-Peer disconnection
END		

(b) To select the RTP route for station-to-station connection using Standard SIP station:

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class for Standard SIP station to the required stations.	<ul style="list-style-type: none"> • Y=02 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) XXZZ XX: 00-15◀: Service Restriction Class A
CM15	Select the RTP route for station-to-station connection using Standard SIP station.	<ul style="list-style-type: none"> • Y=229 (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 0 : Peer-to-Peer connection 1◀: VoIPDB connection
	<p>NOTE 1: <i>When using Peer-to-Peer connection, set the second data to “0” for both the Standard SIP stations on the calling side and the called side.</i></p>	
	<p>NOTE 2: <i>When the call is originated with the access code (set by CM20 Y=0-3: A263/A264), the RTP route set by set by CM20 Y=0-3: A263/A264 is applied.</i></p>	
	<p>NOTE 3: <i>Set the second data to “0” (Peer-to-Peer connection) when using Video Phone feature.</i></p>	
	<p>NOTE 4: <i>A reset of the Standard SIP Terminal is required (Power ON/OFF of the terminal) when this data is set/changed.</i></p>	
END		

(17) Data assignment for accommodating the Standard SIP station in Remote Units

To accommodate the Standard SIP station in a Remote Unit, do the following programming in addition to the “IP-PAD DATA ASSIGNMENT” [Page 2-127](#), “(1) SIP Converter data assignment” [Page 2-35](#), “(2) Standard SIP station data assignment” [Page 2-37](#), “(3) Authentication data assignment for Standard SIP station” [Page 2-39](#).

START	DESCRIPTION	DATA
CM0B	Assign the UDP Port number for Standard SIP control packet between Units to default (5076). RESET	<ul style="list-style-type: none"> Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 143 (2) 01024-65534: UDP Port No. 1024-65534 NONE◀ : 5076
CMAD	Assign the Unit number of the SIP Converter for the Remote Unit. RESET	<ul style="list-style-type: none"> Y=50 (1) 000-127: SIP Converter No. (2) 02-50 : Unit No. NONE◀: Unit01
CM67	Set the DTMF method between Units to Out-band DTMF. NOTE: <i>When accommodating Standard SIP station to the Remote Unit, assign the Unit number of the Remote Unit that the terminal is accommodated.</i>	
	NOTE 1: <i>This data is effective only for Standard SIP station/SIP Trunk-Standard SIP station/SIP trunk connections between Units.</i>	<ul style="list-style-type: none"> Y=26 (1) XXZZ XX : 00-63: Location No. ZZ : 00-63: Location No. (2) 1◀: Out-band DTMF
	NOTE 2: <i>Assign the location number of the Unit by CM0B Y=1XX>10.</i>	
	NOTE 3: <i>This data settings is ineffective when standard SIP stations are interconnected through a Peer-to-Peer connection.</i>	
END		

■ Setting Example

The table below shows the setting example to accommodate Standard SIP Terminal (A), (B) to a Main Unit (Unit01), Standard SIP Terminal (C), (D) to a Remote Unit (Unit10) and to call between them.

- A setting example for Main Unit (Unit01)

CM	Y No.	1ST DATA	2ND DATA	REMARKS
Settings related to Unit01				
0B	101	00	172.16.10.1	Assign the IP address for VOIP port of Unit01 to 172.16.10.1.
0B	101	01	255.255.0.0	Assign the Subnet Mask for VOIP port of Unit01 to 255.255.0.0.
0B	201	00	172.16.10.2	Assign the IP address (RTP) for VOIP port of Unit01 to 172.16.10.2.
Settings related to SIP Converter				
0B	101	33	CCC	Clear the port for SIP Converter and set to default (5070).
10	01	0000	EEC000	Assign the SIP Converter No. 000 to Unit01.
AD	50	000	01	Assign the Unit for SIP Converter 000 to Unit01.

Continued on next page

CM	Y No.	1ST DATA	2ND DATA	REMARKS
Settings related to Standard SIP station				
10	04	000	30100	Assign the station number for Standard SIP Terminal (A) to 30100.
10	04	001	30101	Assign the station number for Standard SIP Terminal (B) to 30101.
1D	14	30100	0	For the station number 30100 assigned by CM10 Y=04, set the terminal type to Standard SIP station.
1D	14	30101	0	For the station number 30101 assigned by CM10 Y=04, set the terminal type to Standard SIP station.
1D	15	30100	05	For the station number 30100 assigned by CM10 Y=04, set the terminal type to Standard SIP station.
1D	15	30101	05	For the station number 30101 assigned by CM10 Y=04, set the terminal type to Standard SIP station.
Settings related to password				
2B	00	30100	30100	Assign the Authentication password for Standard SIP station to 30100.
2B	00	30101	30101	Assign the Authentication password for Standard SIP station to 30101.

- A setting example for Remote Unit (Unit10)

CM	Y No.	1ST DATA	2ND DATA	REMARKS
Settings related to Unit10				
0B	110	00	172.16.20.1	Assign the IP address for VOIP port of Unit10 to 172.16.20.1.
0B	110	01	255.255.0.0	Assign the Subnet Mask for VOIP port of Unit10 to 255.255.0.0.
0B	210	00	172.16.20.2	Assign the IP address (RTP) for VOIP port of Unit10 to 172.16.20.2.
Settings related to SIP Converter				
0B	110	33	CCC	Clear the port for SIP Converter and set to default (5070).
10	01	0001	EEC001	Assign the SIP Converter No. 001 to Unit10.
AD	50	001	10	Assign the Unit for SIP Converter 001 to Unit10.

Continued on next page

CM	Y No.	1ST DATA	2ND DATA	REMARKS
Settings related to Standard SIP station				
10	04	002	32000	Assign the station number for Standard SIP Terminal (C) to 32000.
10	04	003	32001	Assign the station number for Standard SIP Terminal (D) to 32001.
1D	14	32000	0	For the station number 32000 assigned by CM10 Y=04, set the terminal type to Standard SIP station.
1D	14	32001	0	For the station number 32001 assigned by CM10 Y=04, set the terminal type to Standard SIP station.
1D	15	32000	05	For the station number 32000 assigned by CM10 Y=04, set the terminal type to Standard SIP station.
1D	15	32001	05	For the station number 32001 assigned by CM10 Y=04, set the terminal type to Standard SIP station.
Settings related to password				
2B	00	32000	32000	Assign the Authentication password for Standard SIP station to 32000.
2B	00	32001	32001	Assign the Authentication password for Standard SIP station to 32001.

- Common data setting example for system

CM	Y No.	1ST DATA	2ND DATA	REMARKS
20	0	3	805	Assign the digits of Standard SIP station number to 5 digits.
42	-	73	05	Assign the digits of Authentication password for Standard SIP station to 5 digits.

(18) Data assignment for Call Back from call history of Standard SIP station

[9300V3 software required]

When an external call is terminated to Standard SIP Terminal, to call back from the call history of Standard SIP Terminal by adding a trunk access code to a calling number, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM13</div>	<p>Add the trunk access code to the calling number when an external call is terminated to Standard SIP Terminal.</p> <p>NOTE: <i>Do not set this command when adding a trunk access code for other services or on a terminal side.</i></p>	<ul style="list-style-type: none"> • Y=87 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 0: To add
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM35</div>	<p>Assign the trunk access code to be added when storing call histories to the Standard SIP Terminal.</p> <p>NOTE: <i>The trunk access code assigned by this command is used for the following services.</i></p> <ul style="list-style-type: none"> • Voice Mail Transfer • Live Recording • Call origination by ISDN trunk key • Call termination to a Caller ID station • Call termination to a Standard SIP Terminal • Call Back to Mobile Phone 	<ul style="list-style-type: none"> • Y=189 (1) 00-63: Trunk Route No. (2) X-XX: Trunk Access Code to be added X : 0-9, A (*), B (#) NONE◀: No data
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>		

(19) Data assignment for Three-Party Conference/Broker's Call with Standard SIP Terminal
[9300V3 software required]

To perform Three-Party Conference/Broker's Call from Standard SIP Terminal, do the following programming in addition to the (1) SIP Converter data assignment [☞ Page 2-35](#), (2) Standard SIP station data assignment [☞ Page 2-37](#), (3) Authentication data assignment for Standard SIP station [☞ Page 2-39](#).

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM13</div>	<p>Provide the Three-Party Conference with Standard SIP Terminal.</p> <p>NOTE: <i>Set this command to the Standard SIP Terminal which is used to operate Three-Party Conference.</i></p> <p>Provide the Broker's Call with Standard SIP Terminal.</p> <p>NOTE: <i>Set this command to the Standard SIP Terminal which is used to operate Broker's Call.</i></p>	<ul style="list-style-type: none"> • Y=88 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 1◀: To provide <ul style="list-style-type: none"> • Y=89 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 1◀: To provide
<u>END</u>		

(20) Operation settings when Standard SIP station with a held call goes on-hook

[9300V3 software required]






To specify the operation when Standard SIP station with a held call goes on-hook while an outgoing call is originated or a call is in progress, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM08</div>	Specify the operation when Standard SIP station is holding the other call goes on-hook while originating a call or in a call.	(1) 1047 (2) 0 : As per CM13 Y=93 1◀: Hold Transfer
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM13</div>	When CM08>1047: 0 is set, specify the operation when Standard SIP station is holding the other call goes on-hook while originating a call or in a call.	<ul style="list-style-type: none"> • Y=93 (1) X-XXXXXXXX: Standard SIP Station No. assigned by CM10 Y=04 (2) 0 : Hold Transfer 1◀: Disconnect
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">END</div>	<p>NOTE: <i>Assign this command to Standard SIP Terminals.</i></p>	

PUSH Notification Service Data Assignment

[9300V6 software required]

This section explains the following data assignment to be necessary for PUSH Notification Service.

- (1) External Network Connection for PUSH Notification :  Page 2-71
- (2) PUSH Notification Request Incomplete Timer :  Page 2-78
- (3) Caller ID/Name/Ringer Tone Pattern for PUSH Notification :  Page 2-78
- (4) Clearing DNS Cache Table for PUSH Notification Service :  Page 2-79
- (5) Fault Information Store and External Alarm :  Page 2-79

- (1) External Network Connection for PUSH Notification
 - (a) When not using a Web Proxy server:

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin-bottom: 5px;">CM0B</div> <div style="border-left: 1px solid black; border-right: 1px solid black; border-bottom: 1px solid black; width: 100%; height: 100%; position: relative;"> <div style="position: absolute; top: -10px; left: 50%; transform: translate(-50%, -50%);">START</div> <div style="position: absolute; bottom: -10px; left: 50%; transform: translate(-50%, -50%);">A</div> </div>	<p>Set the Web Proxy server to “not used”. [9300V8 software required]</p> <p style="text-align: right;">RESET</p> <p>NOTE: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p> <p>Assign the IP address [Primary] for the DNS server.</p> <p style="text-align: right;">RESET</p> <p>NOTE: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 165: Web Proxy server use for PUSH Notification (2) 3◀: Not used <ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 160: Primary IP Address for DNS server (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data

A	DESCRIPTION	DATA
CM0B	<p>Assign the IP address [Secondary] for the DNS server.</p> <p style="text-align: right;">RESET</p> <p>NOTE: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 161: Secondary IP Address for DNS server (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data
	<p>Assign the Domain name for the PUSH Proxy (FQDN).</p> <p style="text-align: right;">RESET</p> <p>NOTE 1: <i>Refer to “SYSTEM REQUIREMENTS” for FQDN setting.</i></p> <p>NOTE 2: <i>Set this data for all the Units where the VoIP Applications that require PUSH Notification Service will log in.</i></p> <p>NOTE 3: <i>FQDN setting is available only from the PCPro (System Data Mode). CAT mode is not available to set FQDN.</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 163: PUSH Proxy Domain Name (FQDN) (2) Domain Name (Maximum 128 characters) NONE◀: No data
	<p>Assign the Port number for PUSH Notification Request sending.</p> <p style="text-align: right;">RESET</p> <p>NOTE 1: <i>128 ports from the base port number set by this data are used.</i></p> <p>NOTE 2: <i>With the default value NONE, port number 55000-55127 are allocated.</i></p> <p>NOTE 3: <i>Change only the Unit that needs to be changed from the default.</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 170: Start port number for PUSH Notification Request sending) (2) XXXX-XXXXX (Port number 1024-65407) NONE◀: 55000
END		

(b) When using a web proxy (IP address):
[9300V8 software required]

START	DESCRIPTION	DATA
CM0B	<p>Set the Web Proxy server to “Used (IP Address)”.</p> <p style="text-align: right;">RESET</p> <p>NOTE: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 165: Web Proxy server use for PUSH Notification (2) 0: Used (IP Address)
	<p>Assign the IP address for a Web Proxy server (for PUSH Notification).</p> <p style="text-align: right;">RESET</p> <p>NOTE: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 166: IP address for a Web Proxy server (for PUSH Notification) (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data
	<p>Assign the port number for a Web Proxy server (for PUSH Notification).</p> <p style="text-align: right;">RESET</p> <p>NOTE: <i>Change the port setting only for the Unit which is necessary to change the default.</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 174: Web Proxy server Port No. (for PUSH Notification) (2) XXXX-XXXXX (Port number 1-65534) NONE◀: 8080

A

A	DESCRIPTION	DATA
CM0B	Assign the Domain name for the PUSH Proxy (FQDN).	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 163: PUSH Proxy Domain Name (FQDN) (2) Domain Name (Maximum 128 characters) NONE◀: No data
	RESET	
	NOTE 1: Refer to “SYSTEM REQUIREMENTS” for FQDN setting.	
	NOTE 2: Set this data for all the Units where the VoIP Applications that require PUSH Notification Service will log in.	
	NOTE 3: FQDN setting is available only from the PCPro (System Data Mode). CAT mode is not available to set FQDN.	
	Assign the Port number for PUSH Notification Request sending.	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 170: Start port number for PUSH Notification Request sending) (2) XXXX-XXXXX (Port number 1024-65407) NONE◀: 55000
	RESET	
	NOTE 1: 128 ports from the base port number set by this data are used.	
	NOTE 2: With the default value NONE, port number 55000-55127 are allocated.	
	NOTE 3: Change only the Unit that needs to be changed from the default.	
END		

(c) When using a web proxy (Domain name):
[9300V8 software required]

START	DESCRIPTION	DATA	
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin-bottom: 10px;">CM0B</div> <div style="border-left: 1px solid black; border-right: 1px solid black; border-bottom: 1px solid black; height: 400px; margin-left: 5px;"></div> <div style="border: 1px solid black; padding: 2px; width: fit-content; margin-top: 10px; text-align: center;">A</div>	Set the Web Proxy server to “Used (Domain name)”.	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 165: Web Proxy server use for PUSH Notification (2) 1: Used (Domain name) 	
	<div style="border: 1px solid black; border-radius: 10px; padding: 2px 10px; display: inline-block;">RESET</div>	<p>NOTE: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p>	
	Assign the IP address [Primary] for the DNS server.	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 160: Primary IP Address for DNS server (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data 	
	<div style="border: 1px solid black; border-radius: 10px; padding: 2px 10px; display: inline-block;">RESET</div>	<p>NOTE: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p>	
	Assign the IP address [Secondary] for the DNS server.	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 161: Secondary IP Address for DNS server (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data 	
	<div style="border: 1px solid black; border-radius: 10px; padding: 2px 10px; display: inline-block;">RESET</div>	<p>NOTE: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p>	

A	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM0B</div>	Assign the Fully Qualified Domain Name (FQDN) for Web Proxy server with character (for PUSH Notification). <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 167: FQDN for Web Proxy server with character (for PUSH Notification) (2) Domain Name (Maximum 128 characters) NONE◀: No data
	<p>NOTE 1: <i>Set this data for all the Units where the VoIP Applications that require the PUSH Notification Service will log in.</i></p> <p>NOTE 2: <i>FQDN setting is available only from the PCPro (System Data Mode). CAT mode is not available to set FQDN.</i></p>	
	Assign the port number for a Web Proxy server (for PUSH Notification). <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 174: Web Proxy server Port No. (for PUSH Notification) (2) XXXX-XXXXX (Port number 1-65534) NONE◀: 8080
	Assign the Domain name for the PUSH Proxy (FQDN). <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 163: PUSH Proxy Domain Name (FQDN) (2) Domain Name (Maximum 128 characters) NONE◀: No data
	<p>NOTE 1: <i>Refer to “SYSTEM REQUIREMENTS” for FQDN setting.</i></p> <p>NOTE 2: <i>Set this data for all the Units where the VoIP Applications that require PUSH Notification Service will log in.</i></p> <p>NOTE 3: <i>FQDN setting is available only from the PCPro (System Data Mode). CAT mode is not available to set FQDN.</i></p>	
B		

B	DESCRIPTION	DATA
CM0B	Assign the Port number for PUSH Notification Request sending.	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 170: Start port number for PUSH Notification Request sending) (2) XXXX-XXXXX (Port number 1024-65407) NONE◀: 55000
	<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	
	<p>NOTE 1: 128 ports from the base port number set by this data are used.</p>	
	<p>NOTE 2: With the default value NONE, port number 55000-55127 are allocated.</p>	
	<p>NOTE 3: Change only the Unit that needs to be changed from the default.</p>	
<u>END</u>		

(2) PUSH Notification Request Incomplete Timer

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM41</div>	<p>After the SV9300 sends a PUSH Notification Request, if the registration request (REGISTER) is not returned within the period of time set by this data, the SV9300 will send the call termination message (INVITE) according to the currently registered data.</p> <p>NOTE: <i>Normally the default value should not be changed.</i></p>	<ul style="list-style-type: none"> • Y=0 (1) 223: PUSH Notification Request Incomplete Timer (2) 02-32: 2-32 sec. 8 sec.◀
END		

(3) Caller ID/Name/Ringer Tone Pattern for PUSH Notification
[9300V9 software required]

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM13</div>	<p>Specify whether to provide the Caller ID/Name/Ringer Tone Pattern for PUSH Notification.</p>	<ul style="list-style-type: none"> • Y=102 (1) X-XXXXXXXX: Station No. (2) 0 : Not given 1◀: To give
END		

(4) Clearing DNS Cache Table for PUSH Notification Service

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM0B</div>	<p>It is possible to clear the DNS cache table used for PUSH Notification Service.</p> <p>NOTE: <i>When an IP address has been cached in the DNS cache table, the IP address cached is displayed in the second data. (PCPro Command Mode Screen).</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 164: DNS cache table (2) CCC: Clear
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">END</div>		

(5) Fault Information Store and External Alarm

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CMEA</div>	<p>Specify whether or not the fault information is stored into the memory and the type of external alarm to be output when a connection error to the PUSH Proxy occurs.</p> <p>Specify whether or not the fault information is stored into the memory and the type of external alarm to be output when a connection to the PUSH Proxy returns to the normal condition.</p>	<ul style="list-style-type: none"> • Y=2 Fault information and external alarm (1) 132: IP network error occurrence (2) 0 : Fault Memory store/No output of External Alarm <li style="padding-left: 20px;">1 : Fault Memory store/External Alarm is MN alarm <li style="padding-left: 20px;">2 : Fault Memory store/External Alarm is MJ alarm <li style="padding-left: 20px;">3◀: No Fault Memory store/No output of External Alarm • Y=2 Fault information and external alarm for each fault kind (1) 133: IP network error restoration (2) 0 : Fault Memory store/No output of External Alarm <li style="padding-left: 20px;">1 : Fault Memory store/External Alarm is MN alarm <li style="padding-left: 20px;">2 : Fault Memory store/External Alarm is MJ alarm <li style="padding-left: 20px;">3◀: No Fault Memory store/No output of External Alarm
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">END</div>		

Data Assignment for Representative Call Pick-Up on ST500

[9300V9 software required]

This section explains the following data assignment to be necessary for Representative Call Pick-Up on ST500.

- (1) Representative Call Notification Data Assignment : [☞ Page 2-80](#)
- (2) Group Name Assignment for Representative Call Notification : [☞ Page 2-82](#)
- (3) Data Assignment for Answering a Representative Call Notification : [☞ Page 2-82](#)

(1) Representative Call Notification Data Assignment

START	DESCRIPTION	DATA
CM11	Assign the Virtual Line station number for a destination when an incoming call arrives to a group representative station.	<ul style="list-style-type: none"> (1) 0000-0999: Virtual Port No. (2) X-XXXXXXXX: Virtual Line Station No.
CM12	To the Virtual Line station assigned by CM11, assign the Unit number and the Group number for a representative call notification. NOTE: <i>Do not assign the same group number for a representative call notification to multiple Unit numbers.</i>	<ul style="list-style-type: none"> • Y=107 (1) X-XXXXXXXX: Station No. (2) XX ZZZ XX : 01-50: Unit No. ZZZ : 000-127: Group No. NONE◀: No data
CM57	To the Group number assigned by CM12 Y=107, assign the station number of ST500 for a representative call notification. NOTE: <i>For ST500 terminals assigned within a representative call group, accommodate all of them to the same Unit.</i>	<ul style="list-style-type: none"> • Y=39 (1) XXXZZ XXX: 000-127: Group No. ZZ : 00-63: Serial No. within a group (2) X-XXXXXXXX: Station No. NONE◀: No data
A		

START	DESCRIPTION	DATA
CM0B	Assign the Port number for PUSH Notification Request sending (for a representative call notification).	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 182: Base port number for PUSH Notification Request [for a representative call notification] (2) XXXX-XXXXX (Port number 1024-65407) NONE◀: 55500
	<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	
	<p>NOTE 1: 128 ports from the base port number set by this data are used.</p>	
	<p>NOTE 2: With the default value NONE, port number 55500-55627 are allocated.</p>	
	<p>NOTE 3: Change only the Unit that needs to be changed from the default.</p>	
CM12	Read the Standard SIP station connection status (Only display).	<ul style="list-style-type: none"> • Y=96 (1) X-XXXXXXXX: ST500 Station No. assigned by CM57 Y=39 (2) 0 : Not connected 1 : Connecting 3 : Connected (PUSH Notification: terminal info not registered) 4 : Connected (PUSH Notification: terminal info registered) NONE◀: Other than Standard SIP station
<u>END</u>		

(2) Group Name Assignment for Representative Call Notification







START	DESCRIPTION	DATA
<p>CM77</p>	<p>Assign the desired group name to display on ST500 for each group when the representative call notification is sent to ST500.</p> <p>NOTE: <i>Chinese and Russian are not supported.</i></p>	<ul style="list-style-type: none"> • Y=14 Character using PCPro (1) 000-127: Group No. (2) A-Z, 0-9: Character (Maximum 16 characters) NONE◀: No data
<p>END</p>		

(3) Data Assignment for Answering a Representative Call Notification

START	DESCRIPTION	DATA
<p>CM20</p>	<p>Assign the access code for Call Pickup-Direct.</p> <p>NOTE: <i>This command assigns the access code to answer the representative call notification using the Call Pickup-Direct feature. an additional setting is not necessary, if it is already assigned.</i></p>	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A021: Call Pickup-Direct NONE◀: No data
<p>CM64</p>	<p>Assign the access code for answering a representative call notification.</p> <p>NOTE: <i>Set the access code for Call Pickup-Direct assigned to the first data of CM20 Y=0-3: A021.</i></p>	<ul style="list-style-type: none"> • Y=28 (1) 00-63: Tenant No. (2) X-XXXX: Access Code NONE◀: No data
<p>CM15</p>	<p>Specify whether to allow an answer by Call Pickup-Direct when a representative call notification is received.</p> <p>NOTE: <i>Set the second data to "1"(Allow) for the station of ST500 to notify a representative call.</i></p>	<ul style="list-style-type: none"> • Y=014 (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 0 : Restricted 1◀: Allow
<p>END</p>		

IP Station Login/Logout Data Assignment

This section explains the following data assignment to be necessary for IP Station Login/Logout.

- (1) For IP Station Login (Automatic Login Mode) :  Page 2-83
- (2) For IP Station Login (Protected Login Mode) :  Page 2-87
- (3) For IP Station Login (Fixed Connection Mode) :  Page 2-89
- (4) Confirmation of IP Station status and terminal station number :  Page 2-93
- (5) Reading the connecting Unit No. for IP Station :  Page 2-94
- (6) For IP Station Logout (Automatic Login Mode, Protected Login Mode, Fixed Connection Mode) :  Page 2-95

NOTE: IP Station login operation is not supported for Standard SIP Terminal.

- (1) For IP Station Login (Automatic Login Mode)
To register the IP Station ID for Automatic Login mode, do the following programming.

START	DESCRIPTION	DATA
CM08	Specify the ID registration method for Automatic Login Mode.	(1) 513 (2) 1◀: As per CM15 Y=480
CM12	Assign Service Restriction Class C to the required stations.	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Provide the Automatic Login Mode to Service Restriction Class C assigned by CM12 Y=07.	<ul style="list-style-type: none"> • Y=480 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 7◀: Automatic Login Mode (Authentication by MAC Address)
	<p>NOTE 1: This data is effective only when CM08>513 is set to 1.</p> <p>NOTE 2: When a change is made from Automatic Login Mode (by MAC Address) to Fixed Connection Mode, logout operation is required. Before login again, make a backup of your office data by CMEC Y=6>0: 0.</p> <p>NOTE 3: Set the second data to “7” when using the SIP Wireless Terminal (MH240) because the terminal supports only Automatic Login Mode.</p>	
A		

A	DESCRIPTION	DATA
CM2B	Assign the IP Station registration password for the Automatic Login Mode (for maintenance personnel), if required.	<ul style="list-style-type: none"> • Y=10 (1) 00 (2) Z-ZZZZZZZZ: Password for reset setup Z : 0-9, A (*), B (#) NONE◀: No data
CM08	Specify if the system encodes the station number when the IP Stations login to the network.	<ul style="list-style-type: none"> (1) 514 (2) 0 : To encode by original method 1◀: Not encoded
	Specify if the system encodes the password when the IP Stations login to the network.	<ul style="list-style-type: none"> (1) 515 (2) 0 : Not encoded 1◀: As per CM08>517
	Specify the encoding method of the password.	<ul style="list-style-type: none"> (1) 517 (2) 0 : MD5 1◀: Original method
	NOTE 1: Do not set the 2nd data 0, when the destination IP Station does not support the MD5 encoding.	
	NOTE 2: This data is effective when CM08>515 is set to 1.	
<u>END</u>		

NOTE: When changing login mode, logout operation is required after office data setting.


- When replacing the IP Station which has been set to Automatic Login mode with new one individually, do the following programming before registering the ID of new IP Station.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM12</div>	<p>Clear the authentication by MAC Address of the IP Station.</p> <p>NOTE 1: <i>Execute the system data backup by CMEC Y=6>0: 0 after this data setting.</i></p> <p>NOTE 2: <i>If you clear the registered IP Station MAC Address while the system is operating, IP Station is reset even though it is connected.</i></p>	<ul style="list-style-type: none"> • Y=90 (1) X-XXXXXXXX: Station No. (ID) (2) CCC
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>		

- When replacing all IP Stations which have been set to Automatic Login mode with new ones, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMEC</div>	<p>Clear the authentication by MAC Address of all IP Stations.</p> <p style="text-align: center;">OFF LINE</p> <p>NOTE: <i>Execute the system data backup by CMEC Y=6>0: 0 after this data setting.</i></p>	<ul style="list-style-type: none"> • Y=7 (1) 00 (2) CCC
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>		

- When changing the way of ID registration of an IP Station from Automatic Login mode to Protected Login mode during the system operation, do the following programming.

START	DESCRIPTION	DATA
CM15	Provide the Protected Login Mode to Service Restriction Class C assigned by CM12 Y=07.	<ul style="list-style-type: none"> • Y=480 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0: Protected Login Mode
	<p>NOTE: <i>This data is effective only when CM08>513 is set to 1.</i></p>	
CM12	Clear the authentication by MAC Address of the IP Station.	<ul style="list-style-type: none"> • Y=90 (1) X-XXXXXXXX: Station No. (ID) (2) CCC
	<p>NOTE 1: <i>Execute the system data backup by CMEC Y=6>0: 0 after this data setting.</i></p>	
	<p>NOTE 2: <i>You can also clear the authentication by MAC Address of the IP Station by executing the logout operation, for logout operation, see “IP Station Logout”.</i></p> <p> Page 2-126</p>	
END		

(2) For IP Station Login (Protected Login Mode)

To register the IP Station ID for Protected Login mode, do the following programming.

NOTE 1: Register the IDs with Protected Login mode to the IP Stations that provide User Mobility feature.

NOTE 2: We recommend the Protected Login mode for the login method of D^{term}SP30.

NOTE 3: Only the Protected Login mode is available for the login method of SP350.

START	DESCRIPTION	DATA
CM08	Specify the ID registration method for Protected Login Mode.	(1) 513 (2) 0 : Protected Login Mode for All IP Stations 1◀: As per CM15 Y=480
CM12	Assign Service Restriction Class C to the required stations.	• Y=07 (1) X-XXXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Provide the Protected Login Mode to Service Restriction Class C assigned by CM12 Y=07. NOTE: This data is effective only when CM08>513 is set to 1.	• Y=480 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0: Protected Login Mode
CM42	Specify the number of digits for Password.	(1) 73 (2) 01-08 : 1 digit-8 digits NONE◀: 4 digits
CM2B	Assign the Password of the IP Station to the system.	• Y=00 (1) X-XXXXXXXXX: Station No. (2) Z-ZZZZZZZZ: Password Z : 0-9, A (*), B (#) NONE◀: No data
CM08	Specify if the system encodes the station number when the IP Stations login to the network. Specify if the system encodes the password when IP Stations login to the network.	(1) 514 (2) 0 : To encode by original method 1◀: Not encoded (1) 515 (2) 0 : Not encoded 1◀: To encode (As per CM08>517)
A		

A	DESCRIPTION	DATA
CM08	<p>Specify if the system overrides the IP Stations which have the same ID (station number) when the IP Stations login to the network.</p> <p>NOTE: <i>Set the second data to 0, when an IP Station user moves to visitor unit without the logout operation in User Mobility feature.</i></p>	<p>(1) 516 (2) 0 : To override 1◀: Not overridden</p>
	<p>Specify the encoding method of the password.</p> <p>NOTE 1: <i>Do not set the 2nd data 0, when the destination IP Station does not support the MD5 encoding.</i></p> <p>NOTE 2: <i>This data is effective when CM08>515 is set to 1.</i></p>	<p>(1) 517 (2) 0 : MD5 1◀: Original method</p>
CM20	<p>Assign the access code for Password Change.</p>	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A231: Password Change
<u>END</u>		

(3) For IP Station Login (Fixed Connection Mode)

To register the IP Station ID for Fixed Connection mode, do the following programming.

START	DESCRIPTION	DATA
CM08	Specify the ID registration method for Fixed Connection Mode.	(1) 513 (2) 1◀: As per CM15 Y=480
CM12	Assign Service Restriction Class C to the required stations.	• Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Provide the Fixed Connection Mode to Service Restriction Class C assigned by CM12 Y=07.	• Y=480 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 1: Fixed Connection Mode
CM42	Specify the number of digits for Password.	(1) 73 (2) 01-08 : 1 digit-8 digits NONE◀: 4 digits
CM2B	Assign the Password of the IP Station to the system.	• Y=00 (1) X-XXXXXXXX: Station No. (2) Z-ZZZZZZZZ: Password Z : 0-9, A (*), B (#) NONE◀: No data
CM08	Specify if the system encodes the station number when the IP Stations login to the network.	(1) 514 (2) 0 : To encode by original method 1◀: Not encoded
	Specify if the system encodes the password when IP Stations login to the network.	(1) 515 (2) 0 : Not encoded 1◀: To encode (As per CM08>517)
A		

A	DESCRIPTION	DATA
CM08	Specify if the system overrides the IP Stations which have the same ID (station number) when the IP Stations login to the network.	(1) 516 (2) 0 : To override 1◀: Not overridden
	NOTE: <i>Set the second data to 0, when an IP Station user moves to visitor unit without the logout operation in User Mobility feature.</i>	
	Specify the encoding method of the password.	(1) 517 (2) 0 : MD5 1◀: Original method
	NOTE 1: <i>Do not set the 2nd data 0, when the destination IP terminal does not support the MD5 encoding.</i>	
	NOTE 2: <i>This data is effective when CM08>515 is set to 1.</i>	
CM20	Assign the access code for Password Change.	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A231 : Password Change NONE◀: No data
<u>END</u>		

- To register the MAC Address for Fixed Connection Mode automatically, do the following programming after logging in from the terminal.

If this programming is performed after logging in from the terminal, the MAC address of the terminal is registered to the programming data memory of CM12 Y=92 automatically.

START	DESCRIPTION	DATA
CM12	Provide the MAC Address automatic registration in Fixed Connection Mode.	<ul style="list-style-type: none"> • Y=90 (1) X-XXXXXXXX: Station No. (2) 0: MAC Address automatic registration in Fixed Connection Mode
	<p>NOTE 1: <i>If you clear the registered IP Station MAC Address while the system is operating, IP Station is reset even if it is connected.</i></p>	
	<p>NOTE 2: <i>MAC Address automatic registration in Fixed Connection Mode should be executed during the terminal logging in.</i></p>	
	<p>NOTE 3: <i>A maximum of 256 MAC Addresses can be registered in Fixed Connection Mode.</i></p>	
	<p>NOTE 4: <i>This command has to be registered after assigning CM15 Y=480 2nd data "1" and CM2B Y=00 in Fixed Connection Mode. If you do not that, "DATA ERROR" is displayed.</i></p>	
	<p>NOTE 5: <i>Execute the system data backup by CMEC Y=6>0: 0 after this command registered. When changing this data of terminals accommodated in a Remote Unit, execute the system data copy by CMEC Y=8 to the Remote Unit.</i></p>	
	<p>NOTE 6: <i>Confirm the registered MAC Addresses by CM12 Y=92.</i></p>	
<u>END</u>		

- To register the MAC Address for Fixed Connection Mode manually, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM12</div>	<p>Assign the MAC Address in Fixed Connection Mode.</p>	<ul style="list-style-type: none"> • Y=92 (1) X-XXXXXXXX: Station No. (2) XXXXXXXXXXXXX: MAC Address (12 digits) <p>CCC : Clear NONE◀: No data</p> <p>NOTE 1: A maximum of 256 MAC Addresses can be registered in Fixed Connection Mode.</p> <p>NOTE 2: This command has to be registered after assigning CM15 Y=480 2nd data "1" and CM2B Y=00 in Fixed Connection Mode. If you do not that, "DATA ERROR" is displayed.</p> <p>NOTE 3: When a MAC Address is entered during the terminal logging in wrongly, "DATA ERROR" is displayed. When a MAC Address of the other terminals which is logging in is entered, "WAIT, BUSY NOW" is displayed.</p> <p>NOTE 4: Execute the system data backup by CMEC Y=6>0: 0 after this command is registered. When changing this data of terminals accommodated in a Remote Unit, execute the system data copy by CMEC Y=8 to the Remote Unit.</p>
<u>END</u>		

(4) Confirmation of IP Station status and terminal station number

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMFA</div>	<p>Read the IP Station status.</p> <p>NOTE: <i>This command is available for all login methods: Automatic Login Mode (Authentication by MAC Address), Protected Login Mode, and Fixed Connection Mode.</i></p> <p>Read the IP Station number registered in Fixed Connection Mode.</p> <p>NOTE 1: <i>This command can read the IP Station status and terminal station number assigned by CM12 Y=92.</i></p> <p>NOTE 2: <i>The MAC Address of the relevant terminal set in CM12 Y=92 is cleared by the 2nd data "CCC".</i></p>	<ul style="list-style-type: none"> • Y=02 (1) X-XXXXXXXX: Station No. (2) XX...X: Z <ul style="list-style-type: none"> XX...X : IP Address of IP Station Z : IP Station status (A/N) A: Login N: Disconnected FF : Except for IP Station NONE◀: Logout status/ Never been connected <ul style="list-style-type: none"> • Y=30 (1) 000-255: Block No. (2) X-XXXXXXXX: Station No. <ul style="list-style-type: none"> CCC : Clear NONE◀: No data
<u>END</u>		

(5) Reading the connecting Unit No. for IP Station

By the following programming, you can read the Unit No. from PCPro that the IP Station is currently logged in.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CMFA</div>	<p>Read the connecting Unit No. for IP Station (Only display).</p> <p>NOTE 1: <i>This command is used to confirm the Unit No. from PCPro that the IP Station is currently logged in.</i></p> <p>NOTE 2: <i>This feature is available only for IP Multiline Terminal/Standard SIP station.</i></p> <p>NOTE 3: <i>When PCPro is connected to Unit01, you can read all information of IP Stations logged in to the Unit during a normal mode. For IP Stations logged in to the Unit during a survival mode, "NONE" is displayed.</i></p> <p>NOTE 4: <i>When PCPro is connected to Unit02-50 during a normal mode, "CM CODE NOT ALLOWED" is displayed when entering the first data (information for IP Station cannot be read).</i></p> <p>NOTE 5: <i>When PCPro is connected to Unit02-50 during a survival mode, you can read only the information of IP Station logged in to the Unit.</i></p>	<ul style="list-style-type: none"> • Y=05 (1) X-XXXXXXXX: IP Station No. (2) 01-50 : Unit No. FF : Except for IP Station NONE◀: Logout Status/Disconnected
<u>END</u>		

- (6) For IP Station Logout (Automatic Login Mode, Protected Login Mode, Fixed Connection Mode)
For the IP Station user's logout operation, do the following programming.

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class A to the required IP Station.	<ul style="list-style-type: none"> Y=02 (1) X-XXXXXXXX: Station No. (2) XXZZ XX: 00-15◀: Service Restriction Class A
CM15	Allow the logout operation in Service Restriction Class A assigned by CM12 Y=02.	<ul style="list-style-type: none"> Y=143 (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 0: To allow
CM20	Assign the logout access code.	<ul style="list-style-type: none"> Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A239: IP Station Logout
CM90	Assign the logout key on the IP Station, if required.	<ul style="list-style-type: none"> Y=00 (1) My Line No. + <input type="text"/> + Key No. (2) F0B39: IP Station Logout
CM12	Provide the soft key feature to each IP Station.	<ul style="list-style-type: none"> Y=22 (1) X-XXXXXXXX: Station No. (2) 1◀ : Available
	Assign soft key pattern number to each IP Station.	<ul style="list-style-type: none"> Y=23 (1) X-XXXXXXXX: Station No. (2) 0 : Pattern No. 0 1 : Pattern No. 1 2 : Pattern No. 2 3◀: Pattern No. 3
A		

A	DESCRIPTION	DATA
CM9A	Assign the Logout function to the Soft Key of the IP Station.	<ul style="list-style-type: none"> • Y=00-03 Soft Key Pattern No. 0-3 assigned by CM12 Y=23 (1) 01 XX 01 : During dialing (Holding no call) XX: 00-15: Soft Key No. 00-03: Indicated on 1st display 04-07: Indicated on 2nd display 08-11: Indicated on 3rd display 12-15: Indicated on 4th display (2) F0B39: IP Station Logout
	Assign the characters indicated on the IP Station, corresponding to the Soft Key function assigned by CM9A Y=00-03.	<ul style="list-style-type: none"> • Y=10-13 Soft Key Pattern No. 0-3 assigned by CM12 Y=23 (1) Same as CM9A Y=00-03 (2) Soft Key name indicated on LCD (2-12 digits [maximum 6 characters]) 4C6F676F7574: Logout
	<p>NOTE 1: <i>The following characters can be registered; Alphabet upper case (A-Z), alphabet lower case (a-z), numeric (0-9), symbol (! " # \$ % & ' () * + , ; < = > ? @ [] ^ _ ' { } ~), Space, hyphen (-), period (.), slash (/), colon (:)</i></p>	
	<p>NOTE 2: <i>The character "CCC" cannot be registered.</i></p>	
END		

IP STATION SETUP

This section explains the setup procedure for IP Multiline Terminal (D^{term}85), IP Multiline Terminal (DT700/DT800/DT900 Series), Config Setting for XML Application, the Procedure for Registering Terminal License of DT820 and the setup procedure for SIP Wireless Terminal (MH240).

IP Multiline Terminal (D^{term}85) Setup

Data setting on the IP Multiline Terminal is required in addition to the system data programming when you start-up an IP Multiline Terminal first. Do the following setup after accommodation of IP Multiline Terminal to the LAN and the system data programming.

When the data server (such as System-based DRS, Network-based DRS on the LAN, DHCP Server, FTP/TFTP Server, etc.) is used, the IP Multiline Terminal must be connected to the LAN that accommodates the servers before you follow the procedure below.

WARNING

Do not allow regular (non-NEC trained/authorized) users to set “User Menu” items and “Administrator Menu” items.

IP Multiline Terminal may not work if information on these menus has been changed without network/administrator knowledge.

NOTE 1: *When using Preset Dialing, do the following setup after invalidating the HotDialPAD function once or pulling out the LAN cable from the IP Multiline Terminal.*

NOTE 2: *When you change only the setting on “User Menu” items after the IP Multiline Terminal setup has been completed, do the following STEP 2 through STEP 7, then press **Save** key (soft key).*

NOTE 3: *When you change only the setting on “Administrator Menu” items after the IP Multiline Terminal setup has been completed, do the following STEP 2, STEP 8 through STEP 11.*

STEP1: Connect the AC adapter or inline power patch panel to the IP Multiline Terminal.
– You must wait until either “Parameter Missing” or “DRS Connect Time Out” is indicated on the LCD.

STEP2: Press **Hold, Transfer, *, #** keys in that order.
– The “User Menu” appears on the LCD.
At this time, you can use Up and Down soft keys to scroll the menu indication.

STEP3: Press **Hold, #, 0** keys in that order.
– The “Administrator Menu” appears on the LCD.

- STEP4:** Press **7** key.
– The message of “Factory Value Set?” appears on the LCD.
- STEP5:** Press **OK** key (soft key).
– The “Administrator Menu” appears on the LCD.
- STEP6:** Press **Exit** key (soft key).
– The “User Menu” appears on the LCD.
- STEP7:** Select the setting item by pressing digit keys, and set the required data. Then press **Set** key (soft key). When you press the **Set** key, the LCD returns to the Menu indication. Repeat settings for the required items. The following table shows the setting items on the “User Menu”.

User Menu Items

MENU INDICATION		DESCRIPTION
1. DHCP Mode	1: Disable 2: Enable	Set when the DHCP is not used. Set when the DHCP is used. * Default: 2 (DHCP Enable)
2. DRS Address Primary	XXX.XXX.XXX.XXX	Enter the IP Address of System-based DRS. NOTE 1
3. BackUp-DRS Address		
1.DRS Address 2nd	XXX.XXX.XXX.XXX	Enter the IP Address of System-based DRS. When no secondary DRS is provided, this setting is not required. * Default: 0.0.0.0
4. IP Address	XXX.XXX.XXX.XXX	Enter the IP Address of the IP Multiline Terminal. *1 Default: 0.0.0.0 *2 When DHCP server is provided for the IP Multiline Terminal, this setting is not required.
5. Default Gateway	XXX.XXX.XXX.XXX	Enter the Default Gateway Address. *1 Default: 0.0.0.0 *2 When the DHCP server gives the address, or when no Gateway (no router) is provided, this setting is not required.
6. Subnet Mask	XXX.XXX.XXX.XXX	Enter the Subnet Mask. *1 Default: 0.0.0.0 *2 When the DHCP server gives the Subnet Mask, or when no Subnet Mask is required, this setting is not required.

Continued on next page

User Menu Items

MENU INDICATION		DESCRIPTION
7. DNS		
1. DNS Address	XXX.XXX.XXX.XXX	Enter the IP Address of DNS server when the DNS server is used. * Default: 0.0.0.0 NOTE 2
2. DNS Domain Name	DNS Domain Name	Enter the Domain Name of the DNS server when the DNS server is used.
3. DRS Name	DRS Name	Enter the Domain Name of the System-based DRS. * Default: NECDRS1
8. Ether Port Settings		
1. LAN Port Settings		
2. VLAN Mode (LAN)	1: Disable 2: Enable	Set when the VLAN is not used. Set when the VLAN is used. * Default: 1 (Disable) NOTE 3
3. VLAN ID (LAN)	ID 1-4094	Enter the VLAN ID. * Default: 7
4. VLAN Priority (LAN)	Priority 0-7	Enter the Priority of the VLAN. *1 When the HUB has no function to give the priority to the packet, no setting is required. *2 Default: 5
2. PC Port Settings		
2. VLAN Mode (PC)	1: Disable 2: Enable	Set when the VLAN is not used. Set when the VLAN is used. * Default: 1 (Disable) NOTE 3
3. VLAN ID (PC)	ID 1-4094	Enter the VLAN ID. * Default: 7
4. VLAN Priority (PC)	Priority 0-7	Enter the Priority of the VLAN. *1 When the HUB has no function to give the priority to the packet, no setting is required. *2 Default: 5
9. System Information		Select when displaying the list of current settings.

NOTE 1: In the Remote UNIT over IP system, when an IP Multiline Terminal is accommodated to a Remote Unit, input the IP address of the Remote Unit.

NOTE 2: For the IP Address of the DNS server, ask the network administrator.

NOTE 3: Only Tag VLAN is available.

STEP8: After you finish setting “User Menu” items, you can start setting “Administrator Menu” items. To display “Administrator Menu”, stay in the “User Menu” and press **Hold, #, 0** keys in that order.

– The “Administrator Menu” appears on the LCD.

At this time, you can use Up and Down soft keys to scroll the menu indication.

STEP9: Select the setting item by pressing digit keys, and set the required data. Then press **Set** key. When you press the **Set** key, the LCD returns to the Menu indication. Repeat settings for the required items. The following table shows the setting items on the “Administrator Menu”.

Administrator Menu Items

MENU INDICATION		DESCRIPTION
1. DtermIP Mode	1: Normal 2: Remote	Set when the IP Multiline Terminal is connected to the HUB/Router on the LAN. Set when the IP Multiline Terminal is connected to the HUB/Router via Internet/Intranet. * Default: 1 (Normal)
2. Download Address	XXX.XXX.XXX.XXX	Enter the TFTP/FTP Server Address when the server is provided on the LAN.
3. Area ID	0 : Automatic 1 : Japan 2 : America 3 : Australia 4 : Others 5 : Hong Kong 6 : Malaysia 7 : Singapore 8 : England 9 : Mexico 10: Taiwan 11: New Zealand 12: Korea 13: Brazil 14: China 15: Thailand 16: Thailand (EGAT) 17: Sri Lanka	Enter your country/area ID. * Default: 0 (Automatic)

Continued on next page

Administrator Menu Items

MENU INDICATION		DESCRIPTION
4. Port		
1. DRS Mate Port	1-65535	Enter the DRS Mate port number for System-based DRS. * Default: 3456 NOTE 1, NOTE 5
2. DRS Self Port	1024-65535	Enter the DRS Self port number. * Default: 3455 NOTE 1, NOTE 3
3. PROTIMS Self Port	1024-65534	Enter the PROTIMS Self port number. * Default: 3458 NOTE 1, NOTE 3
4. RTP Self Port	1024-65472	Enter the RTP (Real-time Transfer Protocol) Self port number. * Default: 3462 NOTE 1, NOTE 2, NOTE 3
5. Voice Ctrl Self Port	1024-65535	Enter the Voice Control Self port number. * Default: 4000 NOTE 1, NOTE 3
6. H.245 Self Port	0, 1024-65504	Enter the H.245 Self port number. * Default: 0 NOTE 1, NOTE 4
5. Download		
1. Program	File: ramhp2pw.tgz (Fixed)	Select TFTP or FTP by soft key, for downloading the IP Multiline Terminal program.
2. Config	File: config. ini (Fixed)	Select TFTP or FTP by soft key, for downloading the IP Multiline Terminal Configuration file.
3. Input File Name	File: XXXXXXXXX	Enter the file name of TFTP/FTP for downloading, and press its soft key.
6. Soft Reset		Select for resetting the IP Multiline Terminal software.
7. Factory Value		No setting is required.
8. Type of Service		No setting is required.
9. Language NOTE 6	0. English/Japanese 1. Portuguese 2. Spanish 3. French 4. German 5. Italian 6. Dutch 7. Norwegian 8. Danish 9. Swedish	Set the language of the screen. * Default: (Automatic)

NOTE 1: *Well-Known port (0-1023) cannot be assigned except for the DRS Mate Port.*

NOTE 2: *Odd port number cannot be assigned for the RTP Self Port.*

NOTE 3: *Do not assign the same port number for the DRS Self Port, PROTIMS Self Port, RTP Self Port and the Voice Ctrl Port. Also IP Multiline Terminal uses 64 ports from the port that is assigned for the RTP Self Port. For example, when the port number 3462 was assigned for the RTP Self Port, 64 ports (3462-3525) will be used. Do not assign the ports between 3462-3525 for DRS Self Port, PROTIMS Self Port and the Voice Ctrl Port. If you assign the wrong port number, the error message “Port Number Duplicate” will be displayed after the configuration.*

NOTE 4: *IP Multiline Terminal uses 32 ports from the port that is assigned for the H.245 Self Port.*

NOTE 5: *The DRS Mate port number for System-based DRS is set by CM0B Y=1XX>31.*

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NOTE 6: *D^{term}85 (Series i) provides the language of 0-9 when used as the IP Multiline Terminal.*

STEP10: After setting on the “Administrator Menu”, with displaying that menu, press **Exit** key (soft key).
– The “User Menu” appears on the LCD.



STEP11: Press **Save** key (soft key).
– The data is saved and IP Multiline Terminal is reset.
The LCD returns to normal condition.


End of setup

IP Multiline Terminal (DT700/DT800/DT900 Series) Setup

Data setting on the DT700/DT800/DT900 Series is required in addition to the system data programming when you start-up a DT700/DT800/DT900 Series first. Do the following setup after accommodation of the DT700/DT800/DT900 Series to the LAN and the system data programming.

There are two way to setup the DT700/DT800/DT900 Series.

- DT700/DT800/DT900 Series setup by the terminal  [Page 2-102](#)
- DT700/DT800/DT900 Series setup by Web programming  [Page 2-108](#)

NOTE 1: *The configuration menu list described in this section excerpts only necessary items for DT700/DT800/DT900 Series setup. As for the details of each setting items, refer to the “DT900 Setup Guide” for DT900 Series, and refer to the “APPENDIX D DT700/DT800 SERIES CONFIGURATION MENU LIST” for DT700/DT800 Series.  [Page D-1](#)*

NOTE 2: *Setup the terminal by Web programming when using DT710/DT770G (non LCD type).*

(1) DT700/DT800/DT900 Series setup by the terminal

STEP1: Connect the AC adapter or 0 power patch panel to the DT700/DT800/DT900 Series.

STEP2: Press **Menu** keys.
– The “Login” screen appears on the LCD.

STEP3: Enter the Login name and the Password.
– The “Config Menu” appears on the LCD.

STEP4: Select the setting item on next page by pressing digit keys, and set the required data. Then press **Set** key (soft key). When you press the **Set** key, the LCD returns to the Menu indication. Repeat settings for the required items. To change the data, press the **Set** key after **✕** key. Then enter the data again. The following table shows the setting items on the “Config Menu” screen.

NOTE: *Login by the following step when changing/confirming the configuration setting of the DT700/DT800/DT900 Series while the system is operating.*

Menu key → 0 → Set key (soft key) → Password → OK key (soft key)

Config Menu Items

MENU INDICATION		DESCRIPTION
1. Networking Settings		
1. DHCP Mode	1. Disable 2. Enable	Set when the DHCP server is not used. Set when the DHCP server is used. *1 Default: 2 (Enable) *2 Set this data according to your network condition.
2. IP Address	XXX.XXX.XXX.XXX	Enter the IP Address of terminal. * Default: 0.0.0.0
3. Default Gateway	XXX.XXX.XXX.XXX	Enter the Default Gateway Address. *1 Default: 0.0.0.0 *2 When the DHCP server gives the address, or when no Gateway (no router) is provided, this setting is not required.
4. Subnet Mask	XXX.XXX.XXX.XXX	Enter the Subnet Mask. *1 Default: 0.0.0.0 *2 When the DHCP server gives the Subnet Mask, or when no Subnet Mask is required, this setting is not required.

Continued on next page

Config Menu Items

MENU INDICATION	DESCRIPTION
1. Networking Settings (continued)	
6. Advanced Settings	
1. LAN Port Settings	
2. VLAN Mode	1: Disable 2: Enable
	Set when the VLAN is not used. Set when the VLAN is used. * Default: 1 (Disable)
3. VLAN ID	ID 1-4094
	Enter the VLAN ID. * Default: 7
4. VLAN Priority	Priority 0-7
	Enter the Priority of the VLAN. *1 When the HUB has no function to give the priority to the packet, no setting is required. *2 Default: 5
2. PC Port Settings	
2. Port VLAN Mode	1: Disable 2: Enable
	Set when the VLAN is not used. Set when the VLAN is used. * Default: 1 (Disable)
3. Port VLAN ID	ID 1-4094
	Enter the VLAN ID. *1 Default: 1 *2 Default for DT710 is "7".
4. Port VLAN Priority	Priority 0-7
	Enter the Priority of the VLAN. *1 When the HUB has no function to give the priority to the packet, no setting is required. *2 Default: 0 *3 Default for DT710 is "5".

Continued on next page

Config Menu Items

MENU INDICATION		DESCRIPTION
1. Networking Settings (continued)		
6. Advanced Settings (continued)		
6. LLDP Settings NOTE 2		
1. LLDP Mode	1. Disable 2. Enable	Set when terminal does not transmit/receive LLDP packets. Set when terminal transmits/receives LLDP packets. * Default: 2 (Enable)
2. Auto Setting Mode	1. Disable 2. Enable	Set when terminal does not perform data setting automatically when LLDP is received. Set when terminal performs data setting automatically when LLDP is received. * Default: 2 (Enable)
3. RX Waiting Mode	1-60 (second)	For Auto Setting, set the waiting time until terminal receives LLDP packets (1-60 seconds). * Default: 15 seconds
4. Transmit Interval	5-32768 (second)	Set intervals to transmit LLDP packets (5-32768 seconds). * Default: 30 seconds
5. Hold Multiplier	2-10 (multiple)	Set a multiplier determines the time to live for the LLDP packets to be transmitted (2-10 multiples). * Default: 4 multiples
6. Fast Start Count	1-10 (time)	Set the number of times to transmit LLDP packets at one-second intervals after terminal starts up (1-10 times). * Default: 3 times
7. Transparent Mode	1. Disable 2. Enable	Set when transmission of LLDP packets is restricted between the PC port and the LAN port. Set when transmission of LLDP packets is allowed between the PC port and the LAN port. * Default: 2 (Enable)
8. Asset ID		Set the Asset ID for LLDP packets to be transmitted (1-32 characters).

Continued on next page

Config Menu Items

MENU INDICATION		DESCRIPTION
2. SIP Settings		
1. SIP User	1. User ID 2. Password 3. Incoming Number	* Do not enter any data here.
2. Server Address & URI	1. 1st Server Address	Enter the IP address of VOIP port of the system. * Default: 0.0.0.0
4. SIP Server Port	1. 1st Server Port	Enter the UDP port number for terminal. *1 Default: 5060 *2 Match the Port No. to the setting of SIP Server Port for SV9300 side (example: change the Port No. 5060 to 5080 for default). NOTE 4
7. Encryption		
1. Authentication Mode	1. Disable 2. Enable	Set when not using encryption. Set when using encryption. * Default: 1 (Disable)
2. One Time Password NOTE 5	4-10 (digit)	Set the One Time Password. *1 Match the password set by CM2B Y=11>00. *2 Available characters: 0-9, *, #
3. Authentication Status	–	Confirmation of the authentication status. NOTE 6

NOTE 1: Do not set any data except for the data mentioned above.

NOTE 2: “6. LLDP Settings” is available for DT700 Series equipped with firmware version 4.0.0.0 or later. Set this service when you want to notify the terminal information (such as a terminal type and a port number in use) to the destination by the LLDP transmission/reception. Set them as necessary.

NOTE 3: When setting the IP address and the SIP Server Port for DT700/DT800/DT900 Series, those IP address and port number can automatically be allocated to the terminal by using DHCP Server (therefore, the setting for each terminal by Config menu is not required).

NOTE 4: On the Config setting of DT700/DT800/DT900 Series, when the SIP Server Port No. is used by default (5060) without changing its port number, set the UDP Port for DT700/DT800/DT900 Series and SIP trunk on the SV9300 side as follows (however, do the following setting only when not using the SIP trunk).

- Set the UDP port for DT700/DT800/DT900 Series to “5060” (set by CM0B Y=1XX>32)
- Set the UDP port for SIP trunk to other than “5060” (set by CM0B Y=1XX>36)

Continued on next page

NOTE 5: *The One Time Password is the temporary password used for the initial start up of the encryption terminal.*

A terminal and SV9300 use this password until the terminal normally starts up as the encryption terminal. After the completion of the terminal start up, a new password (T-PASS) is sent from SV9300 to the terminal. The encryption is performed by this password from then on. Also, as for the terminal that the start up is completed once, resetting the One Time Password is not required because the new password (T-PASS) is used even when restarting the terminal for the next time.

NOTE 6: *When the start up of the terminal is normally completed, “Acceptable” is displayed on the LCD of the terminal.*

STEP5: Press **Save** key (soft key) on “Config Menu” screen.
– The data is saved and the terminal is reset.

End of setup

(2) DT700/DT800/DT900 Series setup by Web programming

Web programming is the application to provide the Config setting of the terminal on the maintenance terminal using Web Browser.

STEP1: Start up the Web Browser on the maintenance terminal. Then, access to the IP address of the terminal.

NOTE 1: *With the Config menu of the terminal, confirm that the setting is Security Settings, Web Programming, Effective.*

NOTE 2: *When using DT710/DT770G (non LCD type), the IP address of the terminal must be obtained from DHCP server before the setup.*

STEP2: The login screen for Web programming is displayed. Login to the terminal to enter “User Name” and “Password”.

* The default of “User Name” and “Password” are as follows.

User Name: ADMIN

Password : 6633222

NOTE: *Change the password from the default password while the system is operating.*

STEP3: Select the setting item from the Config menu on next page, and set the required data.

Click the setting menu and select the desired setting item from the setting menu on the left side of the screen.

– The setting items are displayed on the right side of the screen. Click the **OK** key after the setting.

Config Menu Items

MENU INDICATION		DESCRIPTION
1. Networking Settings		
1. DHCP Mode	1. Disable 2. Enable	Set when the DHCP server is not used. Set when the DHCP server is used. *1 Default: 2 (Enable) *2 Set this data according to your network condition.
2. IP Address	XXX.XXX.XXX.XXX	Enter the IP Address of terminal. * Default: 0.0.0.0
3. Default Gateway	XXX.XXX.XXX.XXX	Enter the Default Gateway Address. *1 Default: 0.0.0.0 *2 When the DHCP server gives the address, or when no Gateway (no router) is provided, this setting is not required.
4. Subnet Mask	XXX.XXX.XXX.XXX	Enter the Subnet Mask. *1 Default: 0.0.0.0 *2 When the DHCP server gives the Subnet Mask, or when no Subnet Mask is required, this setting is not required.
6. Advanced Settings		
1. LAN Port Settings		
2. VLAN Mode	1: Disable 2: Enable	Set when the VLAN is not used. Set when the VLAN is used. * Default: 1 (Disable)
3. VLAN ID	ID 1-4094	Enter the VLAN ID. * Default: 7
4. VLAN Priority	Priority 0-7	Enter the Priority of the VLAN. *1 When the HUB has no function to give the priority to the packet, no setting is required. *2 Default: 5
2. PC Port Settings		
2. Port VLAN Mode	1: Disable 2: Enable	Set when the VLAN is not used. Set when the VLAN is used. * Default: 1 (Disable)
3. Port VLAN ID	ID 1-4094	Enter the VLAN ID. *1 Default: 1 *2 Default for DT710 is "7".

Continued on next page

Config Menu Items

MENU INDICATION		DESCRIPTION
1. Networking Settings (continued)		
6. Advanced Settings (continued)		
2. PC Port Settings (continued)		
4. Port VLAN Priority	Priority 0-7	Enter the Priority of the VLAN. *1 When the HUB has no function to give the priority to the packet, no setting is required. *2 Default: 0 *3 Default for DT710 is "5".
6. LLDP Settings NOTE 2		
1. LLDP Mode	1. Disable 2. Enable	Set when terminal does not transmit/receive LLDP packets. Set when terminal transmits/receives LLDP packets. * Default: 2 (Enable)
2. Auto Setting Mode	1. Disable 2. Enable	Set when terminal does not perform data setting automatically when LLDP is received. Set when terminal performs data setting automatically when LLDP is received. * Default: 2 (Enable)
3. RX Waiting Mode	1-60 (second)	For Auto Setting, set the waiting time until terminal receives LLDP packets (1-60 seconds). * Default: 15 seconds
4. Transmit Interval	5-32768 (second)	Set intervals to transmit LLDP packets (5-32768 seconds). * Default: 30 seconds
5. Hold Multiplier	2-10 (multiple)	Set a multiplier determines the time to live for the LLDP packets to be transmitted (2-10 multiples). * Default: 4 multiples
6. Fast Start Count	1-10 (time)	Set the number of times to transmit LLDP packets at one-second intervals after terminal starts up (1-10 times). * Default: 3 times
7. Transparent Mode	1. Disable 2. Enable	Set when transmission of LLDP packets is restricted between the PC port and the LAN port. Set when transmission of LLDP packets is allowed between the PC port and the LAN port. * Default: 2 (Enable)
8. Asset ID		Set the Asset ID for LLDP packets to be transmitted (1-32 characters).

Continued on next page

Config Menu Items

MENU INDICATION		DESCRIPTION
2. SIP Settings		
1. SIP User	1. User ID 2. Password 3. Incoming Number	* Do not enter any data here except DT710/DT770G (non LCD type). NOTE 4
2. Server Address & URI	1. 1st Server Address	Enter the IP address of VOIP port of the system. * Default: 0.0.0.0
4. SIP Server Port	1. 1st Server Port	Enter the UDP port number for terminal. *1 Default: 5060 *2 Match the Port No. to the setting of SIP Server Port for SV9300 side (example: change the Port No. 5060 to 5080 for default). NOTE 5
7. Encryption		
1. Authentication Mode	1. Disable 2. Enable	Set when not using encryption. Set when using encryption. * Default: 1 (Disable)
2. One Time Password NOTE 6	4-10 (digit)	Set the One Time Password. *1 Match the password set by CM2B Y=11>00. *2 Available characters: 0-9, *, #
3. Authentication Status	–	Confirmation of the authentication status. NOTE 7

NOTE 1: Do not set any data except for the data mentioned above.

NOTE 2: “6. LLDP Settings” is available for DT700 Series equipped with firmware version 4.0.0.0 or later. Set this service when you want to notify the terminal information (such as a terminal type and a port number in use) to the destination by the LLDP transmission/reception. Set them as necessary.

NOTE 3: When setting the IP address and the SIP Server Port for DT700/DT800/DT900 Series, those IP address and port number can automatically be allocated to the terminal by using DHCP Server (therefore, the setting for each terminal by Config menu is not required).

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NOTE 4: *To set the Login Mode of DT710/DT770G (non LCD type), be sure to set the User ID and Password. In addition, set the User ID and Password for each Login Mode as follows.*

(a) For Protected Login Mode (CM08>513: 0/1, CM15 Y=480: 0) and Fixed Connection Mode (CM08>513: 1, CM15 Y=480: 1)

- User ID : IP Station No. assigned by CM10 Y=01*
- Password: Password assigned by CM2B Y=00*

(b) For MAC Address Authentication (Automatic Login Mode) (CM08>513: 1, CM15 Y=480: 7)

- User ID : IP Station No. assigned by CM10 Y=01*
- Password: Password assigned by CM2B Y=10*

NOTE 5: *On the Config setting of DT700/DT800/DT900 Series, when the SIP Server Port No. is used by default (5060) without changing its port number, set the UDP Port for DT700/DT800/DT900 Series and SIP trunk on the SV9300 side as follows (however, do the following setting only when not using the SIP trunk).*

- Set the UDP port for DT700/DT800/DT900 Series to “5060” (set by CM0B Y=1XX>32)*
- Set the UDP port for SIP trunk to other than “5060” (set by CM0B Y=1XX>36)*

NOTE 6: *The One Time Password is the temporary password used for the initial start up of the encryption terminal.*

A terminal and SV9300 use this password until the terminal normally starts up as the encryption terminal. After the completion of the terminal start up, a new password (T-PASS) is sent from SV9300 to the terminal. The encryption is performed by this password from then on. Also, as for the terminal that the start up is completed once, resetting the One Time Password is not required because the new password (T-PASS) is used even when restarting the terminal for the next time.

NOTE 7: *When the start up of the terminal is normally completed, “Acceptable” is displayed on the LCD of the terminal.*

STEP4: Press the **Save** key after the setup is completed.

STEP5: Popup is displayed to confirm the saving. Press the **OK** key
– The data is saved and the terminal is reset.

End of setup

Config Setting for XML Application

XML applications are available with the terminals equipping XML Browser.

To use an XML application-based terminal, the following data setting on terminal is required.

- Config Menu Start up [☞ Page 2-113](#)
- XML Browser Setting [☞ Page 2-114](#)
- Push Server Registration [☞ Page 2-116](#)

NOTE 1: *When terminal is set by AutoConfig tool, do the setup similar to the following setup according to the Manual of AutoConfig tool.*

NOTE 2: *The following setup may be changed depending on the firmware version of terminal. For details, refer to the Manual of the terminal at the setup.*

NOTE 3: *Make sure the terminal setup is done before doing the following setup. For details, see “IP Multiline Terminal (DT700/DT800/DT900 Series) Setup”. [☞ Page 2-102](#)*

(1) Config Menu Start up

STEP1: Press **Menu** key of terminal.

STEP2: Press **Hold, Transfer**, *, # keys in that order.
– The “Menu” screen appears on the LCD.

NOTE: *The “Menu” screen can be displayed either before login or after login.*

STEP3: Select “Config” in the “Menu” screen.
– The “Login” screen appears on the LCD.

NOTE: *The “Login” screen is displayed only when **Menu** key is pressed in STEP1.*

STEP4: Enter the User Name and the Password, then press **OK** key.
– The “Config Menu” screen appears on the LCD.

Move on “(2) XML Browser Setting” next (only 8LDE and 8LDG).

(2) XML Browser Setting (only 8LDE and 8LDG)

There are the Fixed Mode and Normal Mode for an XML Browser. Select the mode depending on the applications to be used.

Fixed Mode : The XML Browser is automatically started as soon as login is done (unable to display the screen for a telephone).

Normal Mode: Available to select whether to start the XML Browser from the Menu of the terminal or automatically start it.

<Fixed Mode Setting>

STEP1: Select **[6]** Application Settings, **[1]** XML Settings, **[2]** XML Browser, **[1]** Home URL in that order on “Config Menu” screen.

STEP2: Enter URL with **Dial** key, then press **OK** key.
(Example: http://172.16.1.100/service/)

NOTE: *Changeover such as numeric, alphabet and symbol can be available by * key.*

STEP3: Press **↑** key (soft key) until the “Config Menu” screen appears on the LCD.

STEP4: Select **[6]** Application Settings, **[1]** XML Settings, **[4]** Fixed Mode in that order on “Config Menu” screen.

STEP5: Select Enable with **Dial** key, then press **OK** key.

STEP6: Press **↑** key (soft key) until the “Config Menu” screen appears on the LCD.

Move on “(3) Push Server Registration” next.

<Normal Mode Setting>

- Home URL setting of an XML Browser

This setup specifies the Home URL for Service start from the Menu and the Directory URL for Corporate Directory start from the Menu.

- (a) When using the XML Browser by Service start (Home URL registration) from the menu of the terminal

STEP1: Select Application Settings, XML Settings, XML Browser, Home URL in that order on “Config Menu” screen.

STEP2: Enter URL to access as a service with **Dial** key, then press **OK** key.
(Example: http://172.16.1.100/service/)

NOTE: *Changeover such as numeric, alphabet and symbol can be available by * key.*

STEP3: Press key (soft key) until the “Config Menu” screen appears on the LCD.

- (b) When using the XML Browser by Corporate Directory start from the menu of the terminal

STEP4: Select Application Settings, XML Settings, XML Browser, Directory URL in that order on “Config Menu” screen.

STEP5: Enter URL to access as a Corporate Directory with **Dial** key, then press **OK** key.
(Example: http://172.16.1.100/corporation/)

NOTE: *Changeover such as numeric, alphabet and symbol can be available by * key.*

STEP6: Press key (soft key) until the “Config Menu” screen appears on the LCD.

- Automatic start setting of an XML Browser

This setup allow the XML Browser to start automatically when turning on the telephone.

STEP7: Select Application Settings, XML Settings, XML Browser, Automatic Start in that order on “Config Menu” screen.

STEP8: Select Enable with **Dial** key, then press **OK** key.

STEP9: Press key (soft key) until the “Config Menu” screen appears on the LCD.

Move on “(3) Push Server Registration” next.

(3) Push Server Registration

- Push Server Access setting

This setup sets whether to display the push information from the Push Server (Application Server).

STEP1: Select [4] Security, [2] Push Server, [1] Push Server Access in that order on “Config Menu” screen.

STEP2: Select Enable or Disable with **Dial** key, then press **OK** key.

STEP3: Press ↑ key (soft key) until the “Config Menu” screen appears on the LCD.

- Client IP Address setting

This setup sets the IP address for Push Server (Application Server).

NOTE: *Be sure to set the Client IP Address when Push Server Access is set to “Enable”.*

STEP1: Select [4] Security, [2] Push Server, [2] Client IP Address in that order on “Config Menu” screen.

STEP2: Enter the IP Address with **Dial** key, then press **OK** key.

STEP3: Press ↑ key (soft key) until the “Config Menu” screen appears on the LCD.

- Push Server Port setting

This setup sets the port number (default: Port No. 85) of the Push Server (Application Server).

STEP1: Select [4] Security, [2] Push Server, [3] Push Server Port in that order on “Config Menu” screen.

STEP2: Enter the port number of Push Server with **Dial** key, then press **OK** key.

STEP3: Press ↑ key (soft key) until the “Config Menu” screen appears on the LCD, then press **Save** key when the “Config Menu” screen appeared.

– The data is saved and the terminal is reset.

End of setup

Procedure for Registering Terminal License of DT820

The procedure for registering the terminal license of DT820 is as follows.

STEP 1: Set up the required information (Location, Address, E-mail Address, Assigned License, etc.) to the License Management Server (LMS). For details, refer to the LMS manual.

STEP 2: Install the License Management Client (LMC) and set up the required information to get the license file from the LMS. For details, refer to the License Management Client Operations Guide.

NOTE: *The port setting of user's network equipment (such as the firewall, etc.) may have to be changed to accept a terminal license server port (default value is 6080).*

STEP 3: Set up the DT820 by terminal or Web programming.
For details, refer to “IP Multiline Terminal (DT700/DT800/DT900 Series) Setup” of the System Manual.

STEP 4: Select “7. License Setting” menu and set the required data to enable the additional functions from each terminal (Refer to Config Menu Items. [📄 Page 2-118](#)).

or

Download the setting files to multiple terminals by using IP Phone Manager in accordance with the following steps.

1. Collect the information from terminals.
2. Export the information to a CSV file.
3. Edit a CSV file.
4. Import a CSV file to IP Phone Manager.
5. Create setting file of each terminal.
6. Download setting file to each terminal.

– The DT820 requests the LMC to provide the terminal licenses after reboot. The LMC provides the terminal license and the DT820 sets it up itself.

Config Menu Items

MENU INDICATION		DESCRIPTION
7. License Settings		
1. License Server Settings NOTE 1		
1. Server Address	IP address 0.0.0.0-255.255.255.254	Enter the IP address of LMC. * Default: 0.0.0.0
2. Server Port	1024-65535	Enter the server port of LMC. * Default: 6080
2. License Request		
1. Gigabit Ethernet	1: Disable 2: Enable	Set when the Gigabit Ethernet license is not used. Set when the Gigabit Ethernet license is used. * Default: 1 (Disable)
2. Line Key NOTE 2	1: Disable 2: Line 16 3: Line 32	Set when the Line Key license is not used. Set when the 16 Line Key license is used. Set when the 32 Line Key license is used. * Default: 1 (Disable)
3. Adjust		
1. License User Prefix	0-16 character	Enter the License User Prefix. * Default: Blank

NOTE 1: When loading the terminal license to multiple terminals by the DHCP server, set the following data for “1. License Server Settings”.

- 1. Server Address: 43-171
- 2. Server Port : 43-172

NOTE 2: The Line Key license is only effective for the DT820 (8LDX).

STEP 5: Confirm the effective terminal license which is currently registered to the DT820 in accordance with the following steps.

1. Long-press the **Help** key.
2. Press **②** to select “2. System Information”.

```

1. QoS
2. System Information
   Up   Down           Exit
    
```

3. Press **⑦** to select “7. License Information”.

```

6. LLDP Receive Information
7. License Information
   Up   Down   Prev   Exit
    
```

4. To confirm “Gigabit Ethernet”, press soft key Up/Down.
 - Current Gigabit Ethernet license registration is shown on the 2nd line.

```

Gigabit Ethernet
                Assigned
   Up   Down   Prev
    
```

Disable : Use of license is disabled by terminal setting.
 Not Assigned: License registration failure.
 Assigned : License is registered and effective.

5. To confirm “Line Key”, press soft key Up/Down.
 - Current Line Key license registration is shown on the 2nd line.

```

Line Key
                Assigned (Line 32)
   Up   Down   Prev
    
```

Disable : Use of license is disabled by terminal setting.
 Not Assigned : License registration failure.
 Assigned (Line 16): 16 Keys License is registered and effective.
 Assigned (Line 32): 32 Keys License is registered and effective.
 Not Supported : For DT820 (6D), “Not Supported” is always displayed.
 For DT820 (8LDX), “Not Supported” is not displayed.

6. After confirming the terminal license registration status, press **Exit** key to return to idle Screen.

SIP Wireless Terminal (MH240) Setup

Data setting on the MH240 is required in addition to the system data programming when you start-up the MH240 first. Do the following setup after the system data programming.

NOTE 1: *The MH240 supports only for Automatic Login Mode (Authentication by MAC Address) assigned by CM15 Y=480: 7. For details, see “(1) For IP Station Login (Automatic Login Mode)”.* [☞ Page 2-83](#)

NOTE 2: *To accommodate the MH240, the Voice Call (CM15 Y=67) and Internal Zone Paging Access/All Zone Internal Paging (CM15 Y=49) features must be restricted. For details, see “SIP Wireless Terminal (MH240) Data Assignment”.* [☞ Page 2-31](#)

STEP1: Turn the power ON of the MH240.

STEP2: Press **Menu** keys.

– The “Menu” screen appears on the LCD.

STEP3: Select “5. Terminal Setting” on the “Menu” screen.

NOTE: *To select the menu, use the Dial Pad or **Up/Down** key of the MH240. Use the **Hold/Clear** key to return to the next menu up.*

STEP4: Select “3. Maintenance Setting” on the screen currently displayed.

– The “Maintenance Setting” screen appears on the LCD.

STEP5: Select “1. Config” on “Maintenance Setting” screen.

– The “Login” screen appears on the LCD.

STEP6: Enter the User ID and the User Password set to the MH240.

– The “Config Menu” screen appears on the LCD.

STEP7: Set the required data referring to the “Config Menu” items.

Set the required data on each menu. Then press **OK** key. When you press the **OK** key, the LCD returns to the Menu indication. Repeat settings for the required items. To change the data, press the **Menu** key. Then enter the data again. The following table shows the setting items on the “Config Menu” screen.

Config Menu Items

MENU INDICATION		DESCRIPTION
1. Network Setting		
1. Profile		
Prof. Select 1-4		
1. Wireless Setting		
1. Profile Name*	Profile Name	–
2. ESSID	1. Manual 2. AP Scan	Set the Access Point (AP) of the SSID. NOTE 1
3. Security	–	Set the items related to the security from the Security menu. NOTE 2
4. Advanced*	–	–
2. IP Setting NOTE 3		
1. Fixed IP	1. IP Address 2. Net Mask 3. Gateway 4. DNS Primary 5. DNS Secondary	Select “1. Fixed IP” from the IP Setting menu when manually entering the Fixed IP address. NOTE 4
2. DHCP	–	Select “2. DHCP” from the IP Setting menu when obtaining the IP address by DHCP.
3. Set SIP Prof*		–
2. Hand-off List*		–
3. 1st Connect*		–

*: Set each item depending on the system environment to be used.

Continued on next page

Config Menu Items

MENU INDICATION		DESCRIPTION
2. SIP Setting		
Prof. Select 1-4		
1. Profile Name*	Profile Name	–
2. SIP Proxy*	–	–
3. SIP Port*	–	–
4. Register	IP address	Set the IP address of the VOIP Port of SV9300.
5. Register Port*	–	–
6. User ID	User ID	Set the User ID at login.
7. User Password	User Password	Set the User Password at login.
8. Phone No.*	Phone No.	–
9. I-SIP Port	1024-65534	Set the Port No. of the VOIP Port of SV9300 (Follow the typical default setting [default: 5080]).
0. Advanced*	1. Register Timer 2. Codec 3. Payload Size 4. DTMF Mode 5. Jitter Buffer 6. VM Number	–
3. Config Password Setting*		–

*: Set each item depending on the system environment to be used.

NOTE 1: Set the ESSID as follows.

- Select "1. Manual" from the "2. ESSID" menu when manually entering the SSID.
- Select "2. AP Scan" from the "2. ESSID" menu when setting the SSID by searching Access Points (the list of Access Points which are available to access is displayed when selecting "2. AP Scan". Select the appropriate Access Point from the list).

NOTE 2: Select the appropriate setting from the Security menu corresponding to the Wireless LAN access point to connect.

NOTE 3: "2. IP Setting" is set by either "1. Fixed IP" or "2. DHCP".

NOTE 4: Enter the IP Address, Subnet Mask and Default Gateway respectively to the "1. IP Address", "2. Net Mask" and "3. Gateway" in Fixed IP menu.

NOTE 5: For details of the Config setting, refer to the Manual of the MH240.

STEP8: Press **OK** key on the “Config Menu” screen.
– The data is saved and the MH240 is reset.

End of setup

IP STATION LOGIN/LOGOUT OPERATION

To connect the IP Station to the network, registration of the ID for each IP Station is required.

To provide security, passwords for IP Station can be registered to the system that administers and identifies the password.

Follow the registration procedure as shown below.

NOTE: *Login operation is not supported with standard SIP terminal.*

IP Station Login

NOTE: *The way of ID registration is determined by CM08>513 and CM15 Y=480. Before registering the ID, program the system data. See “IP Station Login/Logout Data Assignment”. [Page 2-83](#)*

(1) Automatic Login Mode (Authentication by MAC Address)

Do the following operations only when performing initial setup to the SV9300. This operations are not necessary after initial setup.

NOTE 1: *To save the data registered by the following operation to your PC, do the following operations on the [Save/Load/Verify>Save] screen of the PCPro.*

- 1. Select “Registration data by MAC authentication” from the Area Name field.*
- 2. Enter the file name, then click the [Save] button.*

NOTE 2: *To load the registered data from your PC, reset the CPU blade after loading operation of the PCPro.*

It takes 5 minutes to start up the IP Station after CPU blade is reset.

When you start up the IP Station, the following appears on the LCD.

L	o	g	i	n	:					3	0	0	0	
P	a	s	s	w	d	:								
C	a	n	c	e	l		B	K		S	e	t	O	K

STEP1: Enter the station number of the IP Station to the first row behind “Login:”.

STEP2: Press **Set** key (soft key).

STEP3: Enter the password for reset setup to the second row behind “Passwd:”. **NOTE 3**

NOTE 3: *If no password is set by CM2B Y=10, skip this step.*

STEP4: Press **OK** key (soft key).
– Time and date appears on the LCD.

STEP5: Repeat the above steps on all IP Stations of Automatic Login Mode.

STEP6: Execute the system data backup by CMEC Y=6>0: 0.
– After the first registration, login operation is not required.

NOTE 4: *If you enter a wrong station number, logout the IP Station using the IP Station logout key assigned by CM90. After the login display appears on the LCD, enter the correct station number.*

NOTE 5: *When more than 2 DT700/DT800/DT900 Series (DESI-less) stations login simultaneously, it takes about 30 sec. to 15 minutes until the Line Key is displayed on the LCD (depends on the number of terminals).*

(2) Protected Login Mode/Fixed Connection Mode

When you start up the IP Station, the following appears on the LCD.

L O G I N	:	3 0 0 0
P a s s w d	:	
C a n c e l	B K	S e t O K

STEP1: Enter the station number of the IP Station to the first row behind “Login:”.

STEP2: Press **Set** key (soft key).

STEP3: Enter the password set by CM2B Y=00 for each IP Station to the second row behind “Passwd:”.

STEP4: Press **OK** key (soft key).
– Time and date appears on the LCD.

IP Station Logout

STEP1: Go off-hook or press Speaker key.

STEP2: Operate either of the following for logout.

- Enter the access code assigned by CM20
- Press the logout key assigned by CM90
- Press the soft key assigned by CM9A

NOTE: *Do not execute the logout operation from the IP Station in Automatic Login mode (authentication by MAC address) for normal operation, except when replacing the IP Station or changing the way of ID registration of the IP Station.*

IP-PAD DATA ASSIGNMENT

In addition to the system constitution data for VoIPDB, do the following programming.

The VoIPDB is required for the following connections.

- IP Station-to-Legacy terminal connection
- IP Station-to-Public Network/Tie Line Network connection
- External hold tone connection
- Conference Trunk (CFT) connection
- Voice Response System (VRS) connection

START	DESCRIPTION	DATA	
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM0B</div> <div style="border: 1px solid black; width: 100%; height: 100%; margin-top: 10px;"></div> <div style="border: 1px solid black; width: 100%; height: 100%; margin-top: 10px; text-align: center;">A</div>	Assign the IP address for the VoIPDB. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 5px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 00: IP Address (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data 	
	NOTE: <i>The second data must be entered including the periods (.).</i>	Assign the Subnet Mask for the VoIPDB. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 5px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 01 (2) XXX.XXX.XXX.XXX: 255.0.0.0-255.255.255.252: Subnet Mask (Maximum 15 digits) NONE◀: No data
	NOTE: <i>The second data must be entered including the periods (.).</i>	Assign the Default Gateway Address for the VoIPDB. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 5px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 02 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: Default Gateway Address (Maximum 15 digits) NONE◀: No data
	NOTE: <i>The second data must be entered including the periods (.).</i>	Assign the IP address (RTP) for the VoIPDB. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 5px; display: inline-block;">RESET</div>	<ul style="list-style-type: none"> • Y=2XX (VOIP port [2] + Unit No. [01-50]) (1) 00: IP Address (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data

A

CM0B

DESCRIPTION

DATA

Assign the number of voice channels (RTP) used for the VoIPDB.

- Y=2XX (VOIP port [2] + Unit No. [01-50])
- (1) 10
- (2) 16, 32, 48, 64, 80, 96, 112, 128 (Number of voice channels used)
- NONE◀: Use all channels of the VoIPDB

Assign the RTP Base port number for the VoIPDB.

RESET

- Y=2XX (VOIP port [2] + Unit No. [01-50])
- (1) 40: RTP Base Port for Voice Packet transmitting/receiving
- (2) 01024-65534: RTP Port No.
- NONE◀ : 10000 [10000-10255 (256 ports are used)]

NOTE 3, NOTE 4, NOTE 5

NOTE 1: Follow the typical default setting.

NOTE 2: Set this data when the router or firewall provides the restriction by TCP port.

NOTE 3: For this data, assign the RTP Base port number of voice packet transmitting/receiving ports (RTP ports) corresponding to the number of voice channels used for VoIPDB (assigned by CM0B Y=2XX>10).

NOTE 4: The RTP port is used at 2-port intervals from the RTP Base port number set by this command for each VoIPDB voice channel as shown in the example below.

Example)

When this data and the number of VoIPDB voice channels are set to default settings:

<u>Channel No.</u>	<u>RTP port</u>	
1	10000	← RTP Base port number
2	10002	
⋮	⋮	
128	10254	

Therefore, assign RTP Base port number so that it will not exceed the last RTP Base port number (65534).

NOTE 5: Be sure that the use range of each port assigned by CM0B Y=1XX>36, 37, 38, and 39 does not overlap the setting of the RTP Base port for voice packet transmitting/receiving.

B

B	DESCRIPTION	DATA
CM0B	Assign the UDP port number for the VoIPDB. <div style="text-align: right;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP port [1] + Unit No. [01-50]) (1) 39: UDP Port for voice control (2) 01024-65534: UDP Port No. NONE◀ : 56000 [56000 (1 port is used)]
	<p>NOTE 1: Follow the typical default setting.</p> <p>NOTE 2: Set this data when the router or fire-wall provides the restriction by TCP port.</p> <p>NOTE 3: Be sure that the use range of each port assigned by CM0B Y=2XX>40, Y=1XX>36, 37, and 38 does not overlap the setting of the UDP port for voice control.</p>	<p>NOTE 3</p>
	Assign the TCP server port number for the VoIPDB. <div style="text-align: right;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP port [1] + Unit No. [01-50]) (1) 37: TCP Server Port for P2P voice control (2) 01024-65534: TCP Port No. NONE◀ : 6100 [6100 (1 port is used)]
	<p>NOTE 1: Follow the typical default setting.</p> <p>NOTE 2: Set this data when the router or fire-wall provides the restriction by TCP port.</p> <p>NOTE 3: Be sure that the use range of each port assigned by CM0B Y=2XX>40, Y=1XX>36, 38, and 39 does not overlap the setting of the TCP server port for P2P voice control.</p>	<p>NOTE 3</p>
C		

C	DESCRIPTION	DATA
CM0B	Assign the TCP client port number for the VoIPDB.	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 38: TCP Client Port for P2P voice control (2) 01024-65534: TCP Base Port No. NONE ◀ : 6200 [6200-6327 (128 ports are used)]
	<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	NOTE 3, NOTE 4
	NOTE 1: Follow the typical default setting.	
	NOTE 2: Set this data when the router or fire-wall provides the restriction by TCP port.	
	<p>NOTE 3: 128 ports from the Base port are used for P2P voice control. All ports are not used simultaneously. Ports which are not used in the system, are periodically used in the following cases per calls (establish of TCP groups).</p> <ul style="list-style-type: none"> • Starting Peer-to-Peer connection • DTMF signals • Using TELNET (for maintenance) 	
	NOTE 4: Be sure that the use range of each port assigned by CM0B Y=2XX>40, Y=1XX>36, 37, and 39 does not overlap the setting of the TCP client port for P2P voice control.	
	Specify the DTMF sending method between IP station and VoIPDB.	<ul style="list-style-type: none"> • Y=2XX (VOIP Port [2] + Unit No. [01-50]) (1) 121 (2) 0 : In-band mode (Voice pass through) 1 ◀ : Out-band mode (with H.245 UII/RFC2833)
	NOTE: For the DTMF sending method between Standard SIP station and VoIPDB, set it according to the setting of Standard SIP station. Normally, Out-band mode (RFC2833) is recommended.	
D		

D	DESCRIPTION	DATA
CM0B	When using Standard SIP Terminal, assign the payload type of Out-band DTMF (RFC2833) between Standard SIP station and VoIPDB.	<ul style="list-style-type: none"> • Y=2XX (VOIP Port [1] + Unit No. [01-50]) (1) 150 (2) 001-127 : Payload type 1-127 NONE◀: 101
	Specify port number check for RTP packet.	<ul style="list-style-type: none"> • Y=2XX (VOIP Port [1] + Unit No. [01-50]) (1) 137 (2) 0 : Disable 1◀: Enable
CM41	Assign the DTMF Inter-digit Pause on system basis for the VoIPDB.	<ul style="list-style-type: none"> • Y=0 (1) 107 (2) 01 : 64 ms. 02 : 80 ms. 03 : 96 ms. 04 : 128 ms. 05 : 160 ms. 06 : 192 ms. 07 : 240 ms. NONE◀: No data
<u>END</u>		

Assign the required data to accommodate Single Line/Trunk by CM10 Y=00.

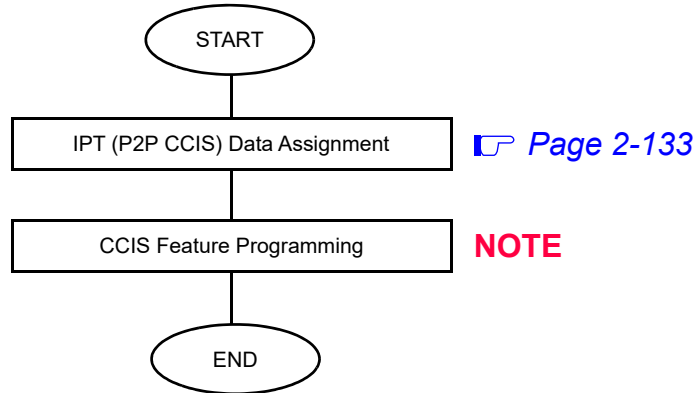
NOTE: When using 2-wired analog trunk (COT, LDT, ODT), set the call level to each trunk.

IPT (P2P CCIS) DATA ASSIGNMENT

Programming Procedure for IPT (P2P CCIS)

Program the system data according to the following procedure.

Programming Procedure for IPT (P2P CCIS)



NOTE: Refer to the “CCIS Features” in the *Networking Manual*.

IPT (P2P CCIS) Data Assignment

In addition to the system constitution data for IPT (P2P CCIS), do the following programming.

START	DESCRIPTION	DATA
CM10	Assign the trunk number for voice channel and control channel of IPT (P2P CCIS). <div style="text-align: right;">RESET</div> <p>NOTE: Set the trunk number for control channel of IPT (P2P CCIS) to one of the trunk number of IPT (P2P CCIS) (D000-D511).</p>	<ul style="list-style-type: none"> • Y=02 (1) 000-511: Virtual Port No. (2) D000-D511: Trunk No. of IPT (P2P CCIS) <p style="text-align: right;">NOTE</p>
CM08	Allow C.O. to C.O. transfer by station or attendant. Provide the Ring Transfer for Call Transfer-All Calls to a IPT (P2P CCIS) when a station holds another station. Specify the maximum number of dialed digits sent to IP network.	<ul style="list-style-type: none"> (1) 028 (2) 0: To allow (1) 253 (2) 0: Available (1) 379 (2) 0 : 15 digits 1◀: 24 digits
CM0B	Assign the IP address for the VoIPDB. <div style="text-align: right;">RESET</div> <p>NOTE: The second data must be entered including the periods (.).</p> Assign the Subnet Mask for the VoIPDB. <div style="text-align: right;">RESET</div> <p>NOTE: The second data must be entered including the periods (.).</p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 00 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 01 (2) XXX.XXX.XXX.XXX: 255.0.0.0-255.255.255.252: Subnet Mask (Maximum 15 digits) NONE◀: No data
A		

A	DESCRIPTION	DATA
CM0B	Assign the Default Gateway Address for the VoIPDB. <div style="text-align: right;">RESET</div>	<ul style="list-style-type: none"> Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 02 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: Default Gateway Address (Maximum 15 digits) NONE◀: No data
	NOTE: <i>The second data must be entered including the periods (.).</i>	
	Assign the IP address (RTP) for the VoIPDB. <div style="text-align: right;">RESET</div>	<ul style="list-style-type: none"> Y=2XX (VOIP port [2] + Unit No. [01-50]) (1) 00: IP Address (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data
	Assign the number of voice channels used for the VoIPDB.	<ul style="list-style-type: none"> Y=2XX (VOIP port [2] + Unit No. [01-50]) (1) 10 (2) 16, 32, 48, 64, 80, 96, 112, 128 (Number of voice channels used) NONE◀: Use all channels of the VoIPDB
CM30	Assign a trunk route number to each trunk number for voice channel of IPT (P2P CCIS). <div style="text-align: right;">IPT (P2P CCIS) RESET</div>	<ul style="list-style-type: none"> Y=00 (1) 000-511: Trunk No. assigned by CM10 Y=02 (2) 00-63 : Trunk Route No. NONE◀: No data
	NOTE 1: <i>It is recommended to assign the trunk route number within the range of 01 to 63.</i>	
	NOTE 2: <i>The trunk route number is not required to set for the IPT (P2P CCIS) trunk number for control channel assigned by CM10 Y=02.</i>	
	Assign a tenant number to each trunk number.	<ul style="list-style-type: none"> Y=01 (1) 000-511: Trunk No. assigned by CM10 Y=02 (2) 00, 01◀-63: Tenant No.
B		

B	DESCRIPTION	DATA
CM35	Assign the trunk route data for IPT (P2P CCIS) route.	<ul style="list-style-type: none"> • Y=000 (1) 00-63: Trunk Route No. (2) 04: Tie Line
	Specify the incoming connection signal as “Wink Start/CCIS”.	<ul style="list-style-type: none"> • Y=004 (1) 00-63: Trunk Route No. (2) 2: Answer signal arrives
	Specify the sender start condition as “Wink Start/CCIS”.	<ul style="list-style-type: none"> • Y=005 (1) 00-63: Trunk Route No. (2) 1◀: Release signal arrives
	Provide the IPT (P2P CCIS) route with No. 7 CCIS facilities.	<ul style="list-style-type: none"> • Y=009 (1) 00-63: Trunk Route No. (2) 03: Wink Start/CCIS
	Assign the IPT number to each IPT (P2P CCIS) route.	<ul style="list-style-type: none"> • Y=020 (1) 00-63: Trunk Route No. (2) 00: Wink Start/CCIS
	NOTE: <i>IPT (P2P CCIS) must be on IPT/ CCH00.</i>	<ul style="list-style-type: none"> • Y=090 (1) 00-63: Trunk Route No. (2) 0: No. 7 CCIS
	Assign the IPT number to each IPT (P2P CCIS) route.	<ul style="list-style-type: none"> • Y=091 (1) 00-63: Trunk Route No. (2) 00: IPT No.
	Specify the trunk seizure sequence for IPT (P2P CCIS) route.	<ul style="list-style-type: none"> • Y=083 (1) 00-63: Trunk Route No. (2) 0 : As per CM08>078 1◀: By allotter
CM08	Specify the trunk seizure sequence when the second data of CM35 Y=083 is set to 0.	<ul style="list-style-type: none"> (1) 078 (2) 0 : Highest available trunk 1◀: Lowest available trunk
C		

C

CM36

	DESCRIPTION	DATA
--	-------------	------

Allow Trunk-to-Trunk Connection with IPT (P2P CCIS) route.

- Y=0
- (1) XX ZZ
XX: 00-63: Incoming trunk route
ZZ : 00-63: Outgoing trunk route
- (2) 0: Allow

CMA7

Assign a trunk number for control channel of IPT (P2P CCIS) to the IPT number.

- Y=00
- (1) 00: IPT No.
- (2) 000-511: Trunk No. for control channel of IPT (P2P CCIS) assigned by CM10 Y=02

Assign an Original Point Code (OPC) to IPT number.

- Y=01
- (1) 00: IPT No.
- (2) 00001-16367: Original Point Code

RESET

NOTE 1: Do not change this data while the system is operating. If you do that, the operation of IP Stations will be unstable.

NOTE 2: If the own SV9300 connects the opposite 2400 IPX IPT (P2P CCIS) via CCIS when the own SV9300 is Remote UNIT over IP system, assign the Point Code of the own SV9300 larger than the point code of the opposite 2400 IPX. This restriction is not performed when the system is not in Remote UNIT over IP or the network is configured by only SV9300s.

Example of Point Code assignment

NETWORK CONFIGURATION	POINT CODE	REMARKS
Office A (SV9300)	00100	Remote UNIT over IP
Office B (SV9300)	00110	Remote UNIT over IP
Office C (SV9300)	00120	Remote UNIT over IP
Office D (2400 IPX)	00010	IPT (P2P CCIS) connection via CCIS
Office E (2400 IPX)	00020	IPT (P2P CCIS) connection via CCIS

D

D	DESCRIPTION	DATA
CMA7	<p>Assign an originating office number to each IPT number.</p> <p>NOTE: <i>This data must be assigned only when open numbering plan is adopted.</i></p> <p>Specify the Type of Service (TOS) field precedence for opposite IPT (P2P CCIS) control packets.</p> <p style="text-align: center;">IPT (P2P CCIS) RESET</p> <p>NOTE 1: <i>The priority of PRECEDENCE 0-7 is as follows.</i> <i>PRECEDENCE 0: Lowest priority</i> ∩ ∩ <i>PRECEDENCE 7: Highest priority</i> <i>PRECEDENCE 6 is recommended.</i></p> <p>NOTE 2: <i>By this data setting, the router recognizes the precedence of control packets and controls Weighted Fair Queuing (WFQ).</i></p> <p>NOTE 3: <i>When the router provides DiffServ QoS for the control packet, assign CMA7 Y=50 (this data setting is ineffective when CMA7 Y=50 is assigned). Page 2-138</i></p>	<ul style="list-style-type: none">• Y=06 <p>(1) 00: IPT No. (2) 0-9999 : Originating Office No. NONE◀: No data</p> <ul style="list-style-type: none">• Y=44 <p>(1) 00: IPT No. (2) 0-7 : PRECEDENCE 0-7 15◀: PRECEDENCE 0</p>
CMA8	<p>Assign the IPT number to the Destination Point Code (DPC).</p> <p>NOTE 1: <i>The Destination Point Code (DPC) is used to designate a terminating office in the No. 7 CCIS network.</i></p> <p>NOTE 2: <i>Assign the Destination Point Code (DPC) by CM8A Y=5000-5255> 168. Page 2-141</i></p> <p>NOTE 3: <i>IPT (P2P CCIS) must be on IPT/ CCH00.</i></p>	<ul style="list-style-type: none">• Y=0 <p>(1) 00001-16367: Destination Point Code (2) 00: IPT No.</p>
E		

E	DESCRIPTION	DATA
CMA8	<p>Assign the IP address of the destination IP trunk (IP address for VOIP port in opposite office) to the Destination Point Code (DPC).</p> <p>NOTE 1: <i>The Destination Point Code (DPC) is used to designate a terminating office in the No. 7 CCIS network.</i></p> <p>NOTE 2: <i>Assign the Destination Point Code (DPC) by CM8A Y=5000-5255> 168. Page 2-141</i></p>	<ul style="list-style-type: none"> • Y=1 (1) 00001-16367: Destination Point Code (2) XXXXXXXXXXXXX: 000000000001-255255255254: IP Address (Maximum 12 digits) NONE◀: No data
CMA7	<p>Specify the connection type for IPT (P2P CCIS) to “Point-to-Multipoint”.</p> <p style="text-align: center;">IPT (P2P CCIS) RESET</p> <p>Specify the release timer for IPT (P2P CCIS) in Point-to-Multipoint connection.</p> <p>When the router provides DiffServ QoS, specify the DiffServ code for IPT (P2P CCIS) control packet.</p> <p style="text-align: center;">IPT (P2P CCIS) RESET</p> <p>NOTE: <i>When this data is set, the TOS field precedence set by CMA7 Y=44 is ineffective. If you want to validate the precedence set by CMA7 Y=44, set “CCC” (data clear) for CMA7 Y=50.</i></p>	<ul style="list-style-type: none"> • Y=46 (1) 00: IPT No. (2) 0: Point-to-Multipoint <ul style="list-style-type: none"> • Y=45 (1) 00: IPT No. (2) 000 : 30 seconds 001-127 : 1-127 minutes (1 minute increment) NONE◀: Not released <ul style="list-style-type: none"> • Y=50 (1) 00: IPT No. (2) 00-3F: DiffServ code
CM20	<p>Assign the first digit of remote office station number to LCR Group 3.</p>	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X: 1st digit of remote office station No. (2) A129: LCR Group 3
F		

F	DESCRIPTION	DATA
CM8A	<p>Assign an Area Code Development Pattern number to LCR Group 3.</p> <p>Assign a Route Pattern number to the first digit of remote office station number for the Area Code Development Pattern number assigned by CM8A Y=A000.</p> <p>Assign the next Route Pattern number to the Route Pattern number assigned by CM8A Y=4000-4007.</p> <p>NOTE: <i>This data is required to expand the order of LCR selection that is assigned by CM8A Y=0000-0255. The order of LCR selection can be expand to the maximum seventh.</i></p>	<ul style="list-style-type: none"> • Y=A000 (1) 3: LCR Group 3 (2) 4000-4007: Area Code Development Pattern No. 0-7 <ul style="list-style-type: none"> • Y=4000-4007 Area Code Development Pattern No. 0-7 (1) X: 1st digit of remote office station No. (2) 0000-0255: Route Pattern No. 000-255 <ul style="list-style-type: none"> • Y=0000-0255 Route Pattern No. 000-255 (1) 0: Designation of next table (2) 0000-0255: Next Route Pattern No. 000-255
G		

G

CM8A

DESCRIPTION

DATA

Specify the order of LCR selection for the Route Pattern number assigned by CM8A Y=4000-4007.

- Y=0000-0255 Route Pattern No. 000-255
- (1) 1-4 : Order of LCR selection
 - 1 : 1st
 - 2 : 2nd
 - 3 : 3rd
 - 4 : 4th
- (2) XXX ZZ
 - XXX: 000-255: LCR Pattern No. 000-255
 - ZZ : 00-63: Trunk Route No. assigned by CM30 Y=00.

NOTE: You cannot assign CM8A Y=0000-0255 1st data 0 (Destination of next table) and 1st data 4 (4th LCR selection) simultaneously. The latest data assignment including system data memory clear is effective. The other assigned data is set as NONE.

Example:

Y	1ST DATA		2ND DATA	ORDER OF LCR SELECTION
0000	0	Destination of next table	0001	
	1	1st LCR selection	00000	1st LCR selection
	2	2nd LCR selection	00101	2nd LCR selection
	3	3rd LCR selection	00202	3rd LCR selection
	4	4th LCR selection	NONE (Unavailable)	
0001	0	Destination of next table	NONE	
	1	1st LCR selection	00303	4th LCR selection
	2	2nd LCR selection	00404	5th LCR selection
	3	3rd LCR selection	00505	6th LCR selection
	4	4th LCR selection	00606	7th LCR selection

H

H	DESCRIPTION	DATA
CM8A	Assign the Destination IP Address Pattern number to the LCR Pattern number.	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 167: Destination IP Address Pattern (2) 000-255: IP Address Pattern No.
	Assign the Destination Point Code (DPC) for IPT (P2P CCIS).	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 168: Destination Point Code for IPT (P2P CCIS) (2) 00001-16367: Destination Point Code
	<p>NOTE: Do not assign the same DPC to the plural LCR pattern numbers. If the same DPC is assigned to the plural LCR pattern numbers, the CCIS feature such as Outgoing Trunk Queuing-CCIS cannot be used.</p>	
	Specify the release timer for each opposite office respectively.	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 171: Release Timer for IPT (P2P CCIS) (2) 000 : 30 seconds 001-127 : 1-127 minutes 999 : Not released NONE◀: As per CMA7 Y=45
	<p>NOTE: When setting the release timer to each trunk route, set the second data to "NONE".</p>	
	Specify the DTMF sending method between IPT (P2P CCIS) and VoIPDB.	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 185: DTMF In-band mode for VoIPDB (2) 0 : In-band mode (Voice pass through) 1 : Out-band mode (with H.245 UII) NONE◀: As per CM0B Y=2XX>121
	<p>NOTE 1: This command is used to set the DTMF sending method for each opposite office.</p>	
	<p>NOTE 2: When connecting to stations/trunks in an own office by Call Transfer, this data setting follows the setting of CM0B Y=2XX>121.</p>	
I		

I

CM5B

DESCRIPTION

DATA

Assign the Destination IP Address for IPT (P2P CCIS).

- Y=01 Destination IP Address for IPT (P2P CCIS)
 - (1) XXX ZZ
 XXX: 000-255: IP Address Pattern No. assigned by CM8A
 Y=5000-5255
 ZZ : 00-07: IP Address No. 00-07 **NOTE 1**
 - (2) XXXXXXXXXXXXX:
 000000000001-255255255254:
 Destination IP Address of opposite IPT (P2P CCIS) (Maximum 12 digits) **NOTE 2**

NOTE 1: IP address number (00-07) is the number for specifying the IP address of opposite office which has plural IP addresses, when the offices are connected with Point-to-Multipoint. If the opposite office has plural IP addresses, set one or more IP address numbers. If the opposite office has only one IP address such as SV9300, set one IP address number.

Example: When the opposite office is 2400 IPX which has two IP addresses, set the data as shown below.

	1ST DATA	MEANING	2ND DATA	MEANING
CM5B Y=01	00000	IP Address Pattern No. assigned by CM8A + IP Address No. 00	10010015150	IP Address of opposite 2400 IPX
	00001	IP Address Pattern No. assigned by CM8A + IP Address No. 01	10010015151	IP Address of opposite 2400 IPX

NOTE 2: Destination IP Address of opposite IPT (P2P CCIS) is the IP Address of opposite SV9300 assigned by CM0B Y=0XX/1XX>00 or the IP Address of opposite 2400 IPX.

NOTE 3: This command should be assigned from the leading number of 1st data (IP Address Pattern No. + IP Address No.).

NOTE 4: When providing Alternate Routing for a fault occurrence for the IPT (P2P CCIS), the next IP address number of the last IP address number that failed in the connecting has the first priority to the next call.

J

J	DESCRIPTION	DATA
CM85	<p>Specify the maximum number of digits dialed by the calling party.</p> <p>NOTE: <i>The maximum number of digits (including the area codes) should be assigned to each area code.</i></p>	<ul style="list-style-type: none"> Y=0-7 Area Code Development Pattern No. 0-7 assigned by CM8A Y=A000 (1) X: 1st digit of remote office station No. (2) 01-24◀: 1-24 digits 25-79 : 25-79 digits
CM0B	<p>Assign the TCP server port number to the system.</p> <p style="text-align: center;">IPT (P2P CCIS) RESET</p> <p>Assign the TCP Client base port number to the system.</p> <p style="text-align: center;">IPT (P2P CCIS) RESET</p> <p>NOTE: <i>1024 ports from the base port number are assigned as TCP Client port number.</i></p>	<ul style="list-style-type: none"> Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 34 (2) 01024-65534: TCP Server Port No. 1024-65534 00000-01023: 57000 NONE◀ : 57000 Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 35 (2) 01024-64512: TCP Client Port No. 1024-64512 00000-01023: Controlled with 58000 64513-65534: Controlled with 58000 NONE◀ : 58000 [58000-59023 (1024 ports are used)]
CM08	<p>Provide the Link Reconnect for IP network.</p> <p>Provide the Link Reconnect for IPT (P2P CCIS).</p> <p>NOTE: <i>This command is effective when CM08>606 is set to 1.</i></p>	<ul style="list-style-type: none"> (1) 606 (2) 1◀: To provide (1) 628 (2) 1◀: To provide
END		

- To provide the Alternative Routing (another trunk route) for the IPT (P2P CCIS) when an outgoing call via the IPT (P2P CCIS) is not available due to a fault occurrence, in addition to “IPT (P2P CCIS) Data Assignment” do the following programming.

START	DESCRIPTION	DATA
CM35	Provide the Alternative Routing for IPT (P2P CCIS). When the outgoing trunk of the tandem office is IPT (P2P CCIS), provide the incoming trunk route with Alternative Routing for a tandem call.	<ul style="list-style-type: none"> Y=186 (1) 00-63: Trunk Route No. (2) 0: To provide <ul style="list-style-type: none"> Y=192 (1) 00-63: Trunk Route No. (2) 0: To provide
CM08	Provide the system with Alternative Routing of CCIS outgoing call when outgoing trunks of tandem office are all busy.	<ul style="list-style-type: none"> (1) 372 (2) 0: To provide
CM41	Assign the PBR timer when establishing tandem connection.	<ul style="list-style-type: none"> Y=0 (1) 104 (2) 02-99: 2-99 seconds (1 second increments) <p>If no data is set, the default setting is 7 seconds.</p> <p>NOTE 1: This data requires changing the timer value depending on the number of Alternative Routing. Set the timer value by the following calculating formula. $PBR \text{ timer value} = (\text{timer value of CMBA } Y=90 [\text{sec.}]) \times (\text{number of Alternative Routing } [\text{Max. 7 times}] + 1) + 1 (\text{sec.})$</p> <p>NOTE 2: Usually, it is recommended to set the second data to 33 (33 sec.).</p>
CM29	Assign a Numbering Plan Group to each tenant.	<ul style="list-style-type: none"> (1) 00-63: Tenant No. (2) 710-713 : Numbering Plan Group 0-3 NONE◀: Numbering Plan Group 0
CM20	Assign the access code for LCR Group 0-3.	<ul style="list-style-type: none"> Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A126: LCR Group 0 A127: LCR Group 1 A128: LCR Group 2 A129: LCR Group 3
A		

A	DESCRIPTION	DATA
CM90	Assign a service access key for LCR Group 0-2 to Multiline Terminal, if required.	<ul style="list-style-type: none"> • Y=00 (1) My Line No. + <input type="text"/> + Key No. (2) F0A26-F0A28: LCR Group 0-2
CM35	Assign the Area Code Development Pattern number for maximum digit analysis to each trunk route.	<ul style="list-style-type: none"> • Y=076 (1) 00-63: Trunk Route No. (2) 05-07: Area Code Development Pattern No. 5-7
CM8A	Assign an Area Code Development Pattern number to each LCR Group.	<ul style="list-style-type: none"> • Y=A000 (1) 0-3: LCR Group 0-3 (2) 4005-4007: Area Code Development Pattern No. 5-7
	Assign a Route Pattern number to each area code for the Area Code Development Pattern number assigned by CM8A Y=A000.	<ul style="list-style-type: none"> • Y=4005-4007 Area Code Development Pattern No. 5-7 (1) X...X: Area Code, Maximum 8 digits (2) 0000-0255: Route Pattern No. 000-255
B		

B

CM8A

DESCRIPTION

DATA

Specify the order of LCR selection for the Route Pattern number assigned by CM8A Y=4005-4007.

- Y=0000-0255 Route Pattern No. 000-255
- (1) 1-4: Order of LCR Selection
 - 1: 1st
 - 2: 2nd
 - 3: 3rd
 - 4: 4th
- (2) XXX ZZ
 - XXX: 000-255: LCR Pattern No.
 - ZZ : 00-63: Trunk Route No.

NOTE 1: You cannot assign CM8A Y=0000-0255 1st data 0 (Destination of next table) and 1st data 4 (4th LCR selection) simultaneously. The latest data assignment including system data memory clear is effective. The other assigned data is set as NONE.

Example:

Y		1ST DATA	2ND DATA	ORDER OF LCR SELECTION
0000	0	Destination of next table	0001	
	1	1st LCR selection	00000	1st LCR selection
	2	2nd LCR selection	00101	2nd LCR selection
	3	3rd LCR selection	00202	3rd LCR selection
	4	4th LCR selection	NONE (Unavailable)	
0001	0	Destination of next table	NONE	
	1	1st LCR selection	00303	4th LCR selection
	2	2nd LCR selection	00404	5th LCR selection
	3	3rd LCR selection	00505	6th LCR selection
	4	4th LCR selection	00606	7th LCR selection

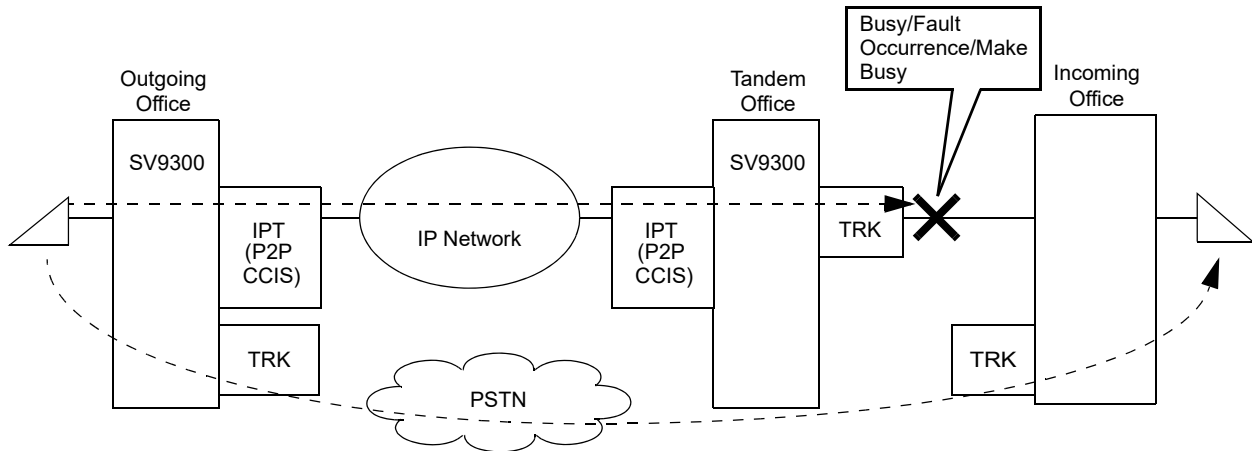
NOTE 2: The order of route selection can be expanded up to 7th order. To expand the order of route selection, assign CM8A Y=0000-0255>0 on next page without assigning the 4th order.

C

C	DESCRIPTION	DATA
CM8A	<p>Assign the next Route Pattern number to the Route Pattern number assigned by CM8A Y=4005-4007.</p> <p>NOTE: <i>This data is required to expand the order of LCR selection that is assigned by CM8A Y=0000-0255. The order of LCR selection can be expand to the maximum seventh.</i></p> <p>For area code deletion, designate the digits to be not deleted.</p>	<ul style="list-style-type: none"> • Y=0000-0255 Route Pattern No. 000-255 (1) 0: Designation of next table (2) 0000-0255: Next Route Pattern No. 000-255
CM85	<p>Specify the maximum number of digits to be dialed by calling party.</p> <p>NOTE: <i>The maximum number of digits including the area codes should be assigned to each area code.</i></p>	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 152: Deletion of all digits of the area code assigned by CM8A Y=4005-4007 (2) 1◀: Not deleted • Y=5-7 Area Code Development Pattern No. 5-7 assigned by CM8A Y=A000 (1) X-X...X: Area code dialed, Maximum 8 digits (2) 01-24◀: 1 digit-24 digits 25-79 : 25 digits-79 digits
CMA7	<p>Assign an ACM signal waiting timer after sending IAI signal in the outgoing calls via CCIS to IPT number.</p>	<ul style="list-style-type: none"> • Y=10 (1) 00: IPT No. (2) 01-14: 2-28 seconds (2 seconds increments) 15◀ : 6 seconds
<u>END</u>		

- When the offices connected with the CCIS via the IPT (P2P CCIS), the tandem office can send the busy/fault occurrence message to the outgoing office by the following conditions.
 - a) When all outgoing trunks of the tandem office are busy
 - b) When having a lack of available VoIPDB channels in the tandem office
 - c) When no answer timer of outgoing call (T1 timer) time-out occurs in the tandem connection (CCIS to CCIS)

The outgoing office which received the message can detour the call via another trunk route such as ISDN.



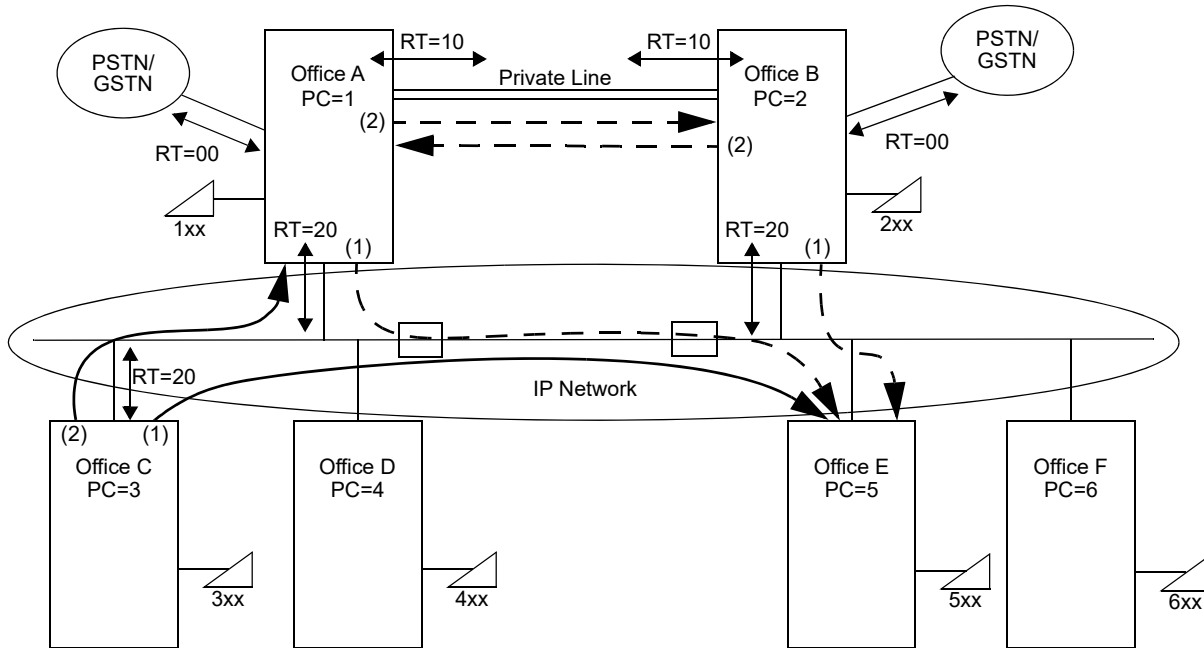
To provide this detouring feature for the systems, do the following programming to the incoming office.

START	DESCRIPTION	DATA
CM08	Provide the system with the alternative routing when lack of VoIPDB channel in tandem office.	(1) 624 (2) 0: To provide (CGC sending)
	Provide the system with the alternative routing when the no answer timer of outgoing call (T1 timer) time-out occurs in tandem connection (CCIS to CCIS).	(1) 666 (2) 0: To provide (CGC sending)
END		

■ Sample Programming of Alternate Routing

In the following network configuration, sample programming of each office A, B, and C to office E is shown below.

In the following network, the terminated call via the private line between office A and office B cannot make a detour through the private line.



Sample Programming of Office C

Data	Meaning
CM20 Y=0>5: A129	Access code for LCR Group (3)
CM8A Y=A000>3: 4000	Area Code Development Pattern No. (0)
CM8A Y=4000>5: 0005	Route Pattern No. (005) for the Area Code Development Pattern No.
CM8A Y=0005>1: 00520	Order of LCR selection (1: 1st) for the Route Pattern No.
CM8A Y=0005>2: 00120	Order of LCR selection (2: 2nd) for the Route Pattern No.
CM8A Y=5001>167: 001	Destination IP Address Pattern No. (001) for the LCR Pattern No.
CM8A Y=5001>168: 00001	DPC of Office A (00001)
CM8A Y=5005>167: 005	Destination IP Address Pattern No. (005) for the LCR Pattern No.
CM8A Y=5005>168: 00005	DPC of Office E (00005)
CM5B Y=01>00100: aaabbb-cccd	IP Address of Office A
CM5B Y=01>00500: aaabb-bcccd	IP Address of Office E
CM35 Y=186>20: 0	Providing Alternate Routing for the Trunk Route of IPT (P2P CCIS) (20)
CM35 Y=192>20: 0	Providing Alternate Routing for the Trunk Route of IN-side trunk (20)

Sample Programming of Office A

Data	Meaning
CM20 Y=0>5: A129	Access code for LCR Group (3)
CM8A Y=A000>3: 4000	Area Code Development Pattern No. (0)
CM8A Y=4000>5: 0005	Route Pattern No. (005) for the Area Code Development Pattern No.
CM8A Y=0005>1: 00520	Order of LCR selection (1: 1st) for the Route Pattern No.
CM8A Y=0005>2: 00010	Order of LCR selection (2: 2nd) for the Route Pattern No.
CM8A Y=5005>167: 005	Destination IP Address Pattern No. (005) for the LCR Pattern No.
CM8A Y=5005>168: 00005	DPC of Office E (00005)
CM5B Y=01>00500: aaabbb-ccdd	IP Address of Office E
CM35 Y=186>00: 1	Not providing Alternate Routing for the Trunk Route of IPT (P2P CCIS) (00)
CM35 Y=186>10: 1	Not providing Alternate Routing for the Trunk Route of IPT (P2P CCIS) (10)
CM35 Y=186>20: 0	Providing Alternate Routing for the Trunk Route of IPT (P2P CCIS) (20)
CM35 Y=192>00: 0	Providing Alternate Routing for the Trunk Route of IN-side trunk (00)
CM35 Y=192>10: 1	Not providing Alternate Routing for the Trunk Route of IN-side trunk (10)
CM35 Y=192>20: 0	Providing Alternate Routing for the Trunk Route of IN-side trunk (20)

Sample Programming of Office B

Data	Meaning
CM20 Y=0>5: A129	Access code for LCR Group (3)
CM8A Y=A000>3: 4000	Area Code Development Pattern No. (0)
CM8A Y=4000>5: 0005	Route Pattern No. (005) for the Area Code Development Pattern No.
CM8A Y=0005>1: 00520	Order of LCR selection (1: 1st) for the Route Pattern No.
CM8A Y=0005>2: 00010	Order of LCR selection (2: 2nd) for the Route Pattern No.
CM8A Y=5005>167: 005	Destination IP Address Pattern No. (005) for the LCR Pattern No.
CM8A Y=5005>168: 00005	DPC of Office E (00005)
CM5B Y=01>00500: aaabbb-ccdd	IP Address of Office E
CM35 Y=186>00: 1	Not providing Alternate Routing for the Trunk Route of IPT (P2P CCIS) (00)
CM35 Y=186>10: 1	Not providing Alternate Routing for the Trunk Route of IPT (P2P CCIS) (10)
CM35 Y=186>20: 0	Providing Alternate Routing for the Trunk Route of IPT (P2P CCIS) (20)
CM35 Y=192>00: 0	Providing Alternate Routing for the Trunk Route of IN-side trunk (00)
CM35 Y=192>10: 1	Not providing Alternate Routing for the Trunk Route of IN-side trunk (10)
CM35 Y=192>20: 0	Providing Alternate Routing for the Trunk Route of IN-side trunk (20)

- When the offices connected by CCIS via IPT (P2P CCIS) are disconnected due to a fault occurrence, the link down can be notified to the Multiline Terminals that are connected to the offices. To provide the link down notice with the Multiline Terminals, assign the following data.

START	DESCRIPTION	DATA
START		
CM90	Assign the Event Occurrence Notice button to the Multiline Terminal. NOTE: <i>Assign this data only to the station number of Multiline Terminal that the link down is notified.</i>	<ul style="list-style-type: none"> Y=00 (1) My Line No. + + Key No. (2) F1600-F1663: Event Occurrence Notice button No. 01-64 NONE◀ : No data
CM9B	Assign the link down notice for CCIS to the Event Occurrence Notice button.	<ul style="list-style-type: none"> Y=0 (1) XX ZZ XX: 01-64: Event Occurrence Notice button No. ZZ: 00: Link Down Notice for CCIS (2) 0 : To notify 1◀: Not notified
END		

LOCATION DATA ASSIGNMENT

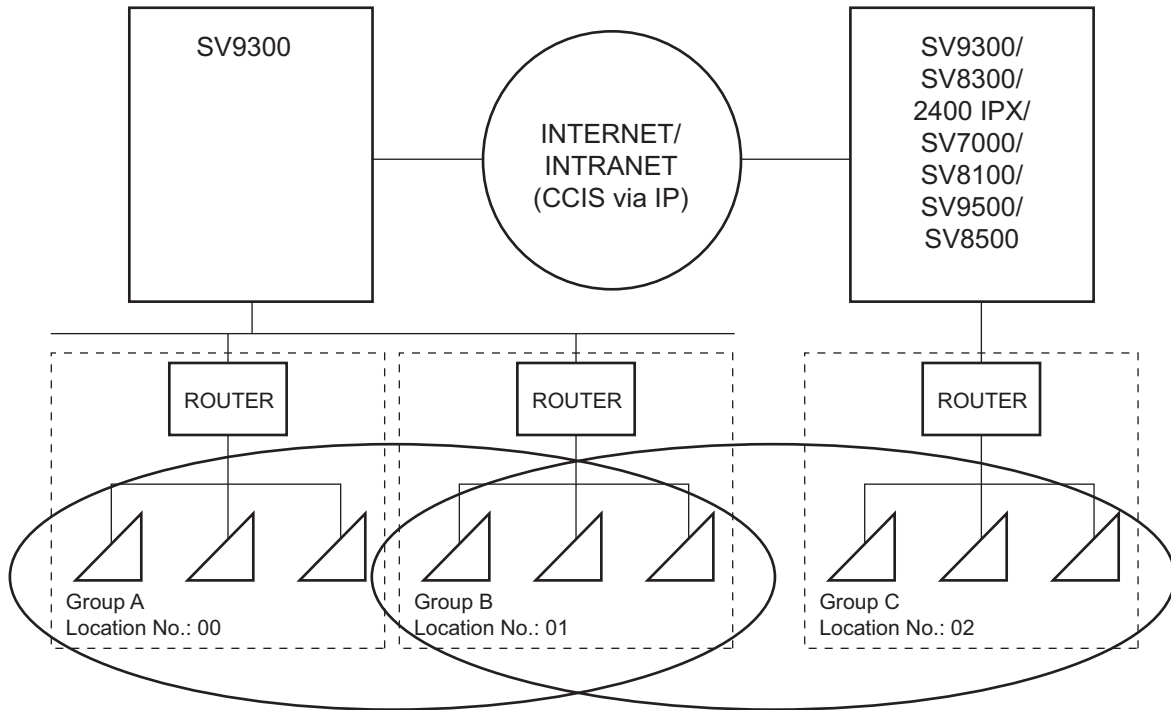
Location number is the data for administration of the group. Location number can differ from the tenant number, service class or trunk route.

For example, setting of the following network condition is possible with the combination of location numbers.

“The communication speed of the groups connected with station-to-station (group A and group B shown in the figure on next page) is 64 Kbps and the other groups connected via IP network (group B and group C shown in the figure on next page) is 8 Kbps.”

The figure on next page shows the sample location data for each group (location).

Sample Locations



Conditions

- Group A and Group B is connected by peer to peer in the system.
- Location number 00/01 has been set to each station in Group A and Group B by CM12 Y=39/50.

Sample Data

COMMAND	Y	FIRST DATA	SECOND DATA
67	00	0001	NONE
67	00	0100	NONE
67	01	0001	NONE
67	01	0100	NONE
67	02	0001	0000
67	02	0100	0000
67	03	0001	01
67	03	0100	01
67	14	00	5
67	14	01	5

Conditions

- Group B and Group C is connected by CCIS via IP.
- Location number 01 has been set to each station in Group B by CM12 Y=39/50.
- Location number 02 has been set to each LCR pattern by CM8A Y=5000-5255.

COMMAND	Y	FIRST DATA	SECOND DATA
67	00	0102	NONE
67	00	0201	NONE
67	01	0102	NONE
67	01	0201	NONE
67	02	0102	0103
67	02	0201	0103
67	03	0102	01
67	03	0201	01
67	14	01	5

NOTE: The first data of CM67 Y=00-06 must be set both on own office and opposite office.

START	DESCRIPTION	DATA
CM12	Assign the location number of the IP Station for local connection.	<ul style="list-style-type: none"> • Y=39 (1) X-XXXXXXXX: Station No. (2) 00-63 : Location No. 00-63 for Local Connection NONE◀: Location No. 00
	Assign the location number of the IP Station for remote connection.	<ul style="list-style-type: none"> • Y=50 (1) X-XXXXXXXX: Station No. (2) 00-63 : Location No. 00-63 for Remote Connection NONE◀: Location No. 00
CM8A	Assign the location number of the group to each LCR pattern number for the IPT (P2P CCIS).	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 173 (2) 00-63 : Location No. 00-63 NONE◀: Location No. 00
	<p>NOTE: <i>This data setting is not required for SP350/Standard SIP station.</i></p> <p>Provide the link reconnection between PC and the system for each LCR Pattern number for the IPT (P2P CCIS).</p>	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 174 (2) 0 : Not provided 3◀: To provide
CM0B	Assign the location number for stations/VoIPDB accommodated in the Unit.	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 10 (2) 00-63 : Location No. 00-63 NONE◀: Location No. 00
A	<p>NOTE 1: <i>This data is effective when the location number is not assigned by CM12 Y=39, 50.</i></p>	
	<p>NOTE 2: <i>When Standard SIP station is accommodated, assign the location number for VoIPDB by this command.</i></p>	

A

CM67

DESCRIPTION

Specify the CODEC list which is used to negotiate the CODEC type and payload size between locations to each location number assigned by CM12 Y=39, 50/CM0B Y=1XX>10/CM8A Y=5000-5255>173.

DATA

- Y=00
- (1) XXZZ
XX: 00-63: Location No.
ZZ : 00-63: Location No.
- (2) 0 : Programmable CODEC List 0
(As per CM42>100-103,
110-113)
- 1 : Programmable CODEC List 1
(As per CM42>120-123,
130-133)
- 2 : Programmable CODEC List 2
(As per CM42>140-143,
150-153)
- 3 : Programmable CODEC List 3
(As per CM42>160-163,
170-173)
- 4 : Fixed CODEC List 1
(Prioritize Tone Quality)
See the table below
- 5 : Fixed CODEC List 2
(Prioritize Band Width)
See the table below
- NONE◀: See the table below.

NOTE 1: This data setting is valid for packets that are sent from the location which is set by “XX” in the first data to the location which is set by “ZZ” in the first data.

NOTE 2: When using SIP trunks accommodated in different units, assign an inter-unit CODEC list by this command.

NOTE 3: The table below shows the setting of Fixed CODEC List1/2.

Priority	Fixed CODEC List 1 (CM67 Y=00:4)		Fixed CODEC List 2 (CM67 Y=00: 5)	
	CODEC Type	Payload Size	CODEC Type	Payload Size
1	G.722	40 ms.	G.729a	40 ms.
2	G.711 μ-law	40 ms.	G.711 μ-law	40 ms.
3	G.711 A-law	40 ms.	G.711 A-law	40 ms.
4	G.729a	40 ms.	G.722	40 ms.

*A-law/μ-law depends on CPU or CM04 Y=10-59.

B

B

DESCRIPTION

DATA

CM67

NOTE 4: *The table below shows the default setting of the CODEC List.*

Priority	Fixed CODEC List (CM67 Y=00: NONE) (Default)	
	CODEC Type	Payload Size
1	G.711 μ -law	40 ms.
2	G.711 A-law	40 ms.
3	G.729a	40 ms.
4	G.723.1	30 ms.

**A-law/ μ -law depends on CPU or CM04 Y=10-59.*

C

C

CM67

DESCRIPTION

DATA

NOTE 5: When assigning data for this command, take into account the CODEC specifications of IP communication terminals and the IP network bandwidth.
The table below shows the CODEC and payload size specification for each IP terminal.

×: Available/ –: Not available/Values in the brackets: Payload size (unit: ms)

IP Terminal		CODEC Specification			
		G.711	G.729a	G.723.1	G.722
VoIPDB*1		× (20/30/40)	× (20/30/40)	–	–*2
DT900 Series/DT800 Series/ DT700 Series(DT730/DT730CG/ DT730DG/DT750)		× (20/30/40)	× (20/30/40)	–	× (20/30/40)
DT700 Series	DT710	× (20/30/40)	× (20/30/40)	–	–
IP ^{term} 85	8D	× (20/30/40)	× (20/30/40)	–	–
	32D	× (20/30/40)	× (20/30/40)	× (30)	–
Soft Phone	D ^{term} SP30	× (20/40)	× (20/40)	–	–
	SP350	× (20/40)	× (20/40)	–	× (20/40)
Standard SIP Terminal		× (20/30/40)	× (20/30/40)	–	–

*1 Connections to a legacy extension/external terminal, IP trunk (P2P CCIS) and SIP trunk are subject to the communication using VoIP.

*2 G.722 (Wide Band Codec) is not available for VoIPDB. If the CODEC list includes only G.722, the system operates by substituting G.711 for G.722. In this situation, however, if the CODEC for the opposite office connected via IP Trunk (P2P CCIS) is also set to G.722 only, a call is not allowed because there is no available CODEC. Therefore, set this data including an available CODEC (G.711 or G.729a).

*3 For CODEC specification when using FAX over IP, see “FAX over IP”.

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NOTE 6: The setting of small payload size is effective for the voice quality measures in general, but the used bandwidth become bigger. Set the payload size considering the available bandwidth in the network.

D



CM67



DESCRIPTION	DATA
<p>Specify the Type of Service (TOS) field precedence of the control packet and voice packet for the IP Station/IPT (P2P CCIS)/VoIPDB to the location number assigned by CM12 Y=39, 50/CM8A Y=5000-5255>173.</p> <p>NOTE 1: <i>This data setting is valid for packets that are sent from the location which is set by “XX” in the first data to the location which is set by “ZZ” in the first data.</i></p> <p>NOTE 2: <i>The TOS between IP Stations or IP Station and VoIPDB is determined by this data setting.</i></p> <p>NOTE 3: <i>The priority of PRECEDENCE 0-7 is as follows. PRECEDENCE 0: Lowest priority PRECEDENCE 7: Highest priority PRECEDENCE 6 for control packet and PRECEDENCE 5 for voice packet are recommended.</i></p> <p>NOTE 4: <i>Follow the typical default setting.</i></p> <p>NOTE 5: <i>Assigning this command enables a router to recognize the precedence with WFQ (Weighed Fair Queuing) and to control the control packets and voice packets according to the precedence.</i></p> <p>NOTE 6: <i>When this data is set, DSCP set by CM67 Y=06 becomes invalid.</i></p> <p>NOTE 7: <i>This command assigns an inter-unit QoS (Quality of Service) when using SIP trunks that are accommodated in different units.</i></p> <p>NOTE 8: <i>This data setting is invalid for packets sent from a Standard SIP station.</i></p>	<ul style="list-style-type: none">• Y=01(1) XX ZZ XX: 00-63: Location No. ZZ : 00-63: Location No.(2) X Z X: 0-7 : PRECEDENCE 0-7 for control packet NOTE 3 Z: 0-7 : PRECEDENCE 0-7 for voice packet NOTE 3 NONE◀: 65

E

CM67

DESCRIPTION	DATA																				
<p>Specify the PAD data between locations to each location number assigned by CM12 Y=39, 50/CM0B Y=1XX>10/CM8A Y=5000-5255>173.</p> <p>NOTE 1: <i>When setting the PAD data for gaining to high value, the echo and howling may occur.</i></p> <p>NOTE 2: <i>This data setting is valid to the location that is set to "XX" in the first data.</i></p>	<ul style="list-style-type: none"> • Y=02 (1) XXZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) TTRR TT (Transmit): 00-08, 81-88 RR (Receive) : 00-08, 81-88 NONE◀ : 0000 <table style="margin-left: 40px;"> <thead> <tr> <th style="text-align: left;"><u>2nd Data</u></th> <th style="text-align: left;"><u>PAD Data [dB]</u></th> </tr> </thead> <tbody> <tr><td>00</td><td>0 dB</td></tr> <tr><td>01/81</td><td>+2 dB/-2 dB</td></tr> <tr><td>02/82</td><td>+4 dB/-4 dB</td></tr> <tr><td>03/83</td><td>+6 dB/-6 dB</td></tr> <tr><td>04/84</td><td>+8 dB/-8 dB</td></tr> <tr><td>05/85</td><td>+10 dB/-10 dB</td></tr> <tr><td>06/86</td><td>+12 dB/-12 dB</td></tr> <tr><td>07/87</td><td>+14 dB/-14 dB</td></tr> <tr><td>08/88</td><td>+16 dB/-16 dB</td></tr> </tbody> </table> <p style="text-align: right; margin-right: 20px;">+: Gain - : Loss</p>	<u>2nd Data</u>	<u>PAD Data [dB]</u>	00	0 dB	01/81	+2 dB/-2 dB	02/82	+4 dB/-4 dB	03/83	+6 dB/-6 dB	04/84	+8 dB/-8 dB	05/85	+10 dB/-10 dB	06/86	+12 dB/-12 dB	07/87	+14 dB/-14 dB	08/88	+16 dB/-16 dB
<u>2nd Data</u>	<u>PAD Data [dB]</u>																				
00	0 dB																				
01/81	+2 dB/-2 dB																				
02/82	+4 dB/-4 dB																				
03/83	+6 dB/-6 dB																				
04/84	+8 dB/-8 dB																				
05/85	+10 dB/-10 dB																				
06/86	+12 dB/-12 dB																				
07/87	+14 dB/-14 dB																				
08/88	+16 dB/-16 dB																				
<p>Provide the Echo Canceller between locations with each location number assigned by CM12 Y=39, 50/CM0B Y=1XX>10/CM8A Y=5000-5255>173, if required.</p> <p>NOTE: <i>This data setting is valid to the location that is set to "XX" in the first data.</i></p>	<ul style="list-style-type: none"> • Y=03 (1) XXZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) 00 : Echo Canceller OFF 01 : Echo Canceller ON NONE◀: Echo Canceller ON 																				

F

F	DESCRIPTION	DATA
CM67	<p>Specify the minimum value of jitter buffer between locations to each location number assigned by CM12 Y=39, 50/CM8A Y=5000-5255>173.</p> <p>NOTE 1: Assign the value which does not exceed the maximum value of jitter buffer set by CM67 Y=05.</p> <p>NOTE 2: This data setting is valid to the location that is set to "XX" in the first data.</p> <p>NOTE 3: Jitter buffer of GPZ-64IPLD/GPZ-128IPLD (VoIPDB) is controlled as 200 ms. on SV9300, even if the minimum value of jitter buffer is set to more than 210 ms.</p> <p>NOTE 4: When this data is set to other than default (NONE), ensure that CM67 Y=05 is also set to other than default (NONE).</p>	<ul style="list-style-type: none"> • Y=04 (1) XX ZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) 01-30 : 10 ms.-300 ms. (10 ms. increments) NONE◀: 10 ms.
G	<p>Specify the maximum value of jitter buffer between locations to each location number assigned by CM12 Y=39, 50/CM8A Y=5000-5255>173.</p> <p>NOTE 1: Assign the value which exceeds the minimum value of jitter buffer set by CM67 Y=04.</p> <p>NOTE 2: This data setting is valid to the location that is set to "XX" in the first data.</p> <p>NOTE 3: When this data is set to other than default (NONE), ensure that CM67 Y=04 is also set to other than default (NONE).</p>	<ul style="list-style-type: none"> • Y=05 (1) XX ZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) 01-30 : 10 ms.-300 ms. (10 ms. increments) NONE◀: 300 ms.

G	DESCRIPTION	DATA
CM67	<p>Specify the DiffServ Code Point (DSCP) of control packet and voice packet to each location number assigned by CM12 Y=39, 50/CM0B Y=1XX>10/CM8A Y=5000-5255>173.</p> <p>NOTE 1: <i>Set this data when the system is connected to the router that provides the DiffServ type QoS function.</i></p> <p>NOTE 2: <i>This data setting is valid for packets that are sent from the location which is set by "XX" in the first data to the location which is set by "ZZ" in the first data.</i></p> <p>NOTE 3: <i>Assign the default (C0) for a Standard SIP station.</i></p> <p>NOTE 4: <i>The TOS field precedence that is set by CM67 Y=01 becomes unavailable when this data is set. To make the TOS field precedence available again, reassign the data using CM67 Y=0 after assigning this data.</i></p> <p>NOTE 5: <i>This command assigns an inter-unit QoS (Quality of Service) when using SIP trunks that are accommodated in different units.</i></p>	<ul style="list-style-type: none"> • Y=06 (1) XXZZ XX: 00-63: Location No. ZZ: 00-63: Location No. (2) XXZZ XX : 00-FE: DSCP of control packet ZZ : 00-FE: DSCP of voice packet NONE◀: C0A0 <p style="text-align: right;">NOTE 3</p>
H		

H	DESCRIPTION	DATA
CM67	Assign the type of tone for each area/country where the location is set.	<ul style="list-style-type: none"> • Y=13 (1) 00-63: Location No. assigned by CM12 Y=39, 50 (2) 01 : Japan 02 : North America 03 : Australia 04 : A-law countries 05 : Hong Kong 06 : Malaysia 07 : Singapore 08 : UK 09 : Mexico 10 : Taiwan 11 : New Zealand 13 : China 14 : Thailand 15 : Brazil 16 : Netherlands 17 : Germany 18 : Italy 19 : Austria 20 : Belgium 21 : Spain 22 : Sweden 23 : UK 24 : Denmark 25 : Greece 26 : Switzerland 27 : South Africa NONE◀: Depends on Nation Code (CM31 Y=0>0)
I		

I	DESCRIPTION	DATA
CM67	<p>Specify the Type of Service (TOS) field precedence of the control packet for the IP Station-to-system to each location number assigned by CM12 Y=39, 50.</p>	<ul style="list-style-type: none"> • Y=14 (1) 00-63: Location No. (2) 0-7 : PRECEDENCE 0-7 for control packet <p>NONE ◀: 6</p>
	<p>RESET</p>	
	<p>NOTE 1: The TOS between the IP Station and the system is determined by this data setting.</p>	
	<p>NOTE 2: The DiffServ Code Point (DSCP) that is set by CM67 Y=15 is invalid when this data is set.</p>	
	<p>NOTE 3: A location number assigned by CM0B Y=1XX>10 is effective for DT700/DT800/DT900 Series/SP350. A location number of IP Station (assigned by CM12 Y=39, 50) cannot be assigned.</p>	
	<p>NOTE 4: The following reset is required for each terminal.</p> <ul style="list-style-type: none"> - When this data is set to the DT700/DT800/DT900 Series/SP350, a system reset is required. - When this data is set to the IP Station, a reset of terminal is required. 	
	<p>Specify the DiffServ Code Point (DSCP) of the control packet for IP Stations to each location number assigned by CM12 Y=39, 50.</p>	<ul style="list-style-type: none"> • Y=15 (1) 00-63: Location No. (2) 00-FE : DSCP of control packet for IP Station <p>NONE ◀: C0</p>
	<p>RESET</p>	
	<p>NOTE 1: The TOS field precedence that is set by CM67 Y=14 is invalid when this data is set.</p>	
	<p>NOTE 2: A location number assigned by CM0B Y=1XX>10 is effective for DT700/DT800/DT900 Series/SP350. A location number of IP Station (assigned by CM12 Y=39, 50) cannot be assigned.</p>	
	<p>NOTE 3: The following reset is required for each terminal.</p> <ul style="list-style-type: none"> - When this data is set to the DT700/DT800/DT900 Series/SP350, a system reset is required. - When this data is set to the IP Station, a reset of terminal is required. 	
J		

J

CM42

DESCRIPTION	DATA
<p>Specify the original CODEC list to the system, if required. To provide the original CODEC list, assign the CODEC type and payload size to CODEC list 0-3.</p>	<p>(1) 100-103: Priority 1-4 in CODEC list 0 120-123: Priority 1-4 in CODEC list 1 140-143: Priority 1-4 in CODEC list 2 160-163: Priority 1-4 in CODEC list 3</p> <p>(2) 01 : G.711 μ-law 64k 02 : G.711 A-law 64k 03 : G.723.1 (5.3/6.3k) 04 : G.729a 07 : G.722 (Wide Band Codec) NONE◀: No data</p> <p>(1) 110-113: Priority 1-4 in CODEC list 0 130-133: Priority 1-4 in CODEC list 1 150-153: Priority 1-4 in CODEC list 2 170-173: Priority 1-4 in CODEC list 3</p> <p>(2) 02-04 : 20 ms.-40 ms. NOTE NONE◀: No data</p>

NOTE: Regarding CODEC and payload size on each terminal, refer to [Page 2-157](#).

END

IP FEATURE PROGRAMMING

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Automatic Program Download for IP Multiline Terminal

General Description

This feature provides the method to download the latest firmware program of IP Station from the FTP/TFTP server automatically by system programming. The following three patterns of program download are available.

- Program Download at Appointed Time
At the appointed time set by CM43 Y=6>00, the system reads the firmware version of IP Station automatically and downloads the firmware program to the required IP Station.
- Program Download at Login Time
When the IP Stations log in to the system, the system reads the firmware version of them automatically and downloads the firmware program to the required IP Station.
- Program Download for Designated Terminal
The system reads the firmware version of the designated IP Stations automatically and downloads the firmware program to them if required.

Station Application

D^{term}85 (Series i) Terminals with LCD equipped for IP Adapter units (Peer-to-Peer version),
D^{term}IP INASET, D^{term}SP30, DT700/DT800/DT900 Series.

Service Conditions

1. For program download, the FTP or TFTP server is required.
2. This service is available for the IP Station with Peer-to-Peer connections. Terminals with IPELC connections are not supported.
3. Maximum of four terminals can be downloaded at the same time regardless of the type of download.
4. Firmware version filed in FTP/TFTP server is assigned by the system data (CM0C Y=00-07>02). When the latest firmware version is filed in the server, be sure to update the system data.
5. When the FTP/TFTP server is used for program download, login from the IP Station with a login name “**anonymous**” and password “**123456**” (authentication by FTP server is required).
6. For DT900 Series, use the DT900 Series terminals applied to the following firmware versions.
 - For ITK-XXXXX-1X TEL: firmware version 2.1.0.0 or later
 - For ITK-XXXXX-2X TEL: firmware version 3.0.0.0 or later

NOTE 1: For “ITK-XXXXX-2X TEL” DT900 Series, DT900 firmware V3.0.0.0 or later is applied to the terminal. DT900 firmware V1.0.0.0 to V2.X.X.X cannot be applied.

NOTE 2: The release schedule of “ITK-XXXXX-2X TEL” varies by the market. For details, please contact the NEC local subsidiary or dealer.

NOTE 3: For “ITK-XXXXX-1X TEL” DT900 Series, only DT900 firmware V1.0.0.0 to V2.X.X.X can be applied to the terminal.

7. The default directory/default file name/time for upgrading per terminal for each IP Station is as shown below. Leave the default directory/default file name/time for upgrading per terminal as it is.

Type of IP Station	TFTP Server (Default File Name)	FTP Server (Default Directory + Default File Name)	Time for Upgrading Per Terminal
DT930 (Touch Panel)	itkisipec.tgz	/DT930T/itkisiptc.tgz	Approx. 3 minutes
DT930 (For “ITK-XXXXX-2X”)	itk2isipvc.tgz	/DT930CG/itk2isipvc.tgz	Approx. 3 minutes
DT930 (For “ITK-XXXXX-1X”)	itkisipec.tgz	/DT930CG/itkisipvc.tgz	Approx. 3 minutes
DT920 (Self-Labeling) (For “ITK-XXXXX-2X”)	itk2isipec.tgz	/DT920D/itk2isipec.tgz	Approx. 3 minutes
DT920 (Self-Labeling) (For “ITK-XXXXX-1X”)	itkisipec.tgz	/DT920D/itkisipec.tgz	Approx. 3 minutes
DT920 (For “ITK-XXXXX-2X”)	itk2isipe.tgz	/DT920E/itk2isipe.tgz	Approx. 3 minutes
DT920 (For “ITK-XXXXX-1X”)	itkisipec.tgz	/DT920E/itkisipec.tgz	Approx. 3 minutes
DT830CG	itzisipvc.tgz	/DT830CG/itzisipvc.tgz	Approx. 3 minutes
DT830, DT830DG	itzisipvg.tgz	/DT830V/itzisipvg.tgz	Approx. 3 minutes
DT820 (Self-Labeling)	ityisipex.tgz	/DT820D/ityisipex.tgz	Approx. 3 minutes
DT820	ityisipe.tgz	/DT820E/ityisipe.tgz	Approx. 3 minutes
DT750	itlisips.tgz	/DT750S/itlisips.tgz	Approx. 20 minutes
DT730CG	itlisipvc.tgz	/DT730CG/itlisipvc.tgz	Approx. 3 minutes
DT730DG	itlisipvg.tgz	/DT730DG/itlisipvg.tgz	Approx. 3 minutes
DT730	itlisipv.tgz	/DT730V/itlisipv.tgz	Approx. 10 minutes
DT710	itlisipe.tgz	/DT710E/itlisipe.tgz	Approx. 10 minutes
D ^{term} 85 (Series i) (IP Bundled Type)	ramhp2pw.out	/IPRU/ramhp2pw.out	Approx. 3 minutes
D ^{term} 85 (Series i) (IP adapter Type)	ramhp2pr.hex	/IPR/ramhp2pr.hex	Approx. 3 minutes

Type of IP Station	TFTP Server (Default File Name)	FTP Server (Default Directory + Default File Name)	Time for Upgrading Per Terminal
D ^{term} IP INASET	dpPtoP1E.lnl	/INASET/dpPtoP1E.lnl	Approx. 3 minutes

8. Conditions on Program Download at Appointed Time
 - a. Program Download at Appointed Time service is available by the system data assignment (CM15 Y=482) on a station class of service basis.
 - b. By the same command mentioned above, program download can be retried just one time. When the download results in failure two times in succession, the IP Station operates with previous firmware.
 - c. When this service is executed, neither Program Download at Login Time nor Program Download for Designated Terminal can be executed.
 - d. PBX system checks the firmware version of each terminal starting from the terminal with the smallest Physical Port No., and execute this service if required.
 - e. This service is executed only for the idle terminals. This service is not executed for the terminals in Logout or busy status.
 - f. The actual starting time of the Program Download at Appointed Time may be varied with a range of +/-1 minute. For example, when 448 of IP Stations have to be upgraded, it takes about 225 minutes to be completed, without retry.
 - g. CMFA Y=50>01/02 counts the number of terminals that are successfully downloaded and the number of terminals that result in download failure only when the PBX sends the latest program download message to the terminals. During executing the download, “Downloading...” is displayed on the LCD of a terminal.
After the download is finished, the LCD of a terminal returns to the time display automatically.
 - h. Program Download at Appointed Time can be suspended by CMFA Y=50>00: CCC (clear) during executing the download. However, you cannot suspend the download of the terminal with “Downloading...” displayed.
9. Conditions on Program Download at Login Time
 - When upgrading the D^{term}85 (Series i) (IP Adapter Type/IP Bundle Type)
 - a. Program Download at Login Time service can be assigned on a system basis by CM0C Y=90>02.
 - b. When this service is assigned, retry in download failure status is not available.
 - c. During executing the download, “Downloading...” is displayed on the LCD of a terminal.
After the download is finished, the LCD of a terminal returns to the time display automatically.

NOTE: When the second data of CM15 Y=480 is set to “0” (Protected Login Mode), the “Login” screen is displayed.

- When upgrading the DT700/DT800/DT900 Series
 - a. Program Download at Login Time service can be assigned on a system basis by CM0C Y=90>02.
 - b. When this service is assigned, retry in download failure status is not available.
 - c. During executing the download, “Downloading...” is displayed on the LCD of a terminal.
After the download is finished, the LCD of a terminal returns to the time display automatically.
- 10. Conditions on Program Download for Designated Terminals
 - a. Program Download for Designated Terminals is assigned and executed on the Primary station number basis.
 - b. This service is executed only for idle terminals. This service is not executed for the terminals in Logout status.
 - c. When this service is assigned, retry in download failure status is not available.
 - d. During executing the download, “Downloading...” is displayed on the LCD of a terminal.
After the download is finished, the LCD of a terminal returns to the time display automatically.

Programming

(1) FTP/TFTP Server information setting

START	DESCRIPTION	DATA
CM0C	Specify the firmware in the FTP/TFTP server to update the IP Station's firmware.	<ul style="list-style-type: none"> • Y=00-07 Update Information Profile No. (1) 00: Type of Firmware for Update (2) 02 : D^{term} IP INASET 03 : D^{term}85 (Series i) (IP Adapter Type) 05 : D^{term}85 (Series i) (IP Bundled Type) 12 : DT710 13 : DT730/DT710 (Self-Labeling) 14 : DT750 17 : DT730DG 18 : DT730CG 19 : DT770G 21 : DT830/DT830DG 22 : DT830CG 23 : DT820 [9300V3 STEP2 software required] 24 : DT820 (Self-Labeling) [9300V3 STEP2 software required] 25 : DT930CG [9300V7 software required] 26 : DT920 [9300V7 software required] 27 : DT920 (Self-Labeling) [9300V7 software required] 28 : DT930 (Touch Panel) [9300V7 software required]
A		NONE◀: No data

A	DESCRIPTION	DATA
CM0C	<p>Specify the file version of the firmware in the FTP/TFTP server.</p> <p>NOTE: <i>If no data is set, the system does not update the firmware of IP Stations.</i></p>	<ul style="list-style-type: none"> • Y=00-07 Update Information Profile No. <ol style="list-style-type: none"> (1) 02: Firmware File Version for Update (2) AABBCDD (for DT700/DT800/DT900 Series) <ul style="list-style-type: none"> AA : 00-99: Integral No. of file version BB : 00-99: First decimal No. of file version CC : 00-99: Second decimal No. of file version DD : 00-99: Third decimal No. of file version <p>0000XXZZ (for D^{term}85)</p> <p>XX : 00-99: Integral No. of file version</p> <p>ZZ : 00-99: First decimal No. of file version</p> <p>NONE◀: No data</p>
	<p>Assign the IP Address for the FTP/TFTP server.</p>	<ul style="list-style-type: none"> • Y=00-07 Update Information Profile No. 00-07 <ol style="list-style-type: none"> (1) 04: IP Address for Server (2) XXXXXXXXXXXXX: <ul style="list-style-type: none"> 000000000001-255255255254: IP Address for the FTP/TFTP server (Maximum 12 digits) NONE◀: No data
B		

B

DESCRIPTION

DATA

CM0C

Specify the protocol of the server.

- Y=00-07 Update Information Profile No.
00-07
- (1) 05: Protocol of Server
- (2) 0 : FTP
- 1◀: TFTP

NOTE: *When using the TFTP/FTP server, allow the PBX to access the server with anonymity, and set the following information to the TFTP/FTP server if required.*

Login Name/Password	TFTP Server	FTP Server
Login Name	anonymous	anonymous
Password		123456

END

(2) Setting for automatic update

(a) To update the firmware of IP Stations automatically at the predetermined time:

START	DESCRIPTION	DATA
CM43	<p>Specify the start time for automatic update.</p> <p>NOTE: <i>This data is recommended to set to have two minutes or longer interval of time from the system clock.</i></p>	<ul style="list-style-type: none"> • Y=6 (1) 00: Start time for automatic update (2) YYYY MM DD HH mm <ul style="list-style-type: none"> YYYY : 2014-2099 (Year) MM : 01-12 (Month) DD : 01-31 (Date) HH : 00-23 (Hour) mm : 00-59 (Minutes) NONE◀: Not to update the firmware of IP Stations automatically at the predetermined time
CM12	<p>Assign Service Restriction Class C to the required stations.</p>	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	<p>Assign the way of automatic updating at the predetermined time in Service Restriction Class C assigned by CM12 Y=07.</p>	<ul style="list-style-type: none"> • Y=482 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0 : To update (One time retry) 1 : To update (No retry) 2◀: Not updated
END		

(b) To read the status of the automatic update at the predetermined time:

START	DESCRIPTION	DATA
CMFA	Read the status of the automatic update at the predetermined time.	<ul style="list-style-type: none"> • Y=50 (1) 00 (2) 00 : Not started 01 : Now updating 10 : Completed CCC: Interrupt updating/Reset count data
	NOTE: <i>If you type the second data CCC, the count data which can be read by CMFA Y=50>01/02 is cleared at the same time.</i>	
	Read the number of terminal that succeeded in updating.	<ul style="list-style-type: none"> • Y=50 (1) 01 (2) XXXX: Number of succeeded terminal in updating
	Read the number of terminal that failed in updating.	<ul style="list-style-type: none"> • Y=50 (1) 02 (2) XXXX: Number of failed terminal in updating
END		

(c) To update the firmware of IP Stations automatically when IP Stations login to the system:

START	DESCRIPTION	DATA
CM0C	Provide the automatic update when IP Stations login to the system.	<ul style="list-style-type: none"> • Y=90 (1) 02 (2) 0: To provide
	NOTE: <i>Set the second data to "1" (default), after the IP Stations firmware is updated.</i>	
END		

(d) To update the firmware of the IP Station which you specified:

START

DESCRIPTION

DATA

CM0C

Specify the IP Station to be updated, then execute the firmware updating.

- Y=50
- (1) X-XXXXXXXX: Station No.
- (2) 0: Start updating

After setting the first data and second data, PCPro/CAT displays the status of the IP Station. The table below shows the contents of the display and its meaning.

FIRST DATA	IP STATION STATUS		SECOND DATA	IP STATION STATUS	
	DISPLAY	MEANING		DISPLAY	MEANING
X-XXXXXXXX: IP Station No.	AABB CCDD	Current firmware version of the IP Station (for DT700/DT800/DT900 Series) AABBCCDD: AA : 00-99: Integral No. BB : 00-09: First decimal No. CC : 00-99: Second decimal No. DD : 00-99: Third decimal No.	0: Start updating	OK	Start updating
				DATA NOT FOUND	You cannot update the IP Station's firmware because the FTP/TFTP server information data has not been assigned NOTE 1
				WAIT, BUSY NOW	You cannot update the IP Station's firmware because other four IP Stations in the system are updated now NOTE 2
	DATA ERROR	The IP Station is logout status/The terminal is not IP terminal	You cannot update the IP Station's firmware		
WAIT, BUSY NOW	The IP Station is updated now/The IP Station is busy				

NOTE 1: Set this data after setting CM0C Y=00-07>00-05.

NOTE 2: Maximum four IP Stations can be updated at the same time in a system. Set this data after other four IP Stations are updated.

END

(3) To read the IP Station information:

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content;">CMFA</div>	Read the firmware version of the IP Station.	<ul style="list-style-type: none"> • Y=00 (1) X-XXXXXXXX: Station No. (2) AABCCDD (for DT700/DT800/DT900 Series) <ul style="list-style-type: none"> AA: 00-99: Integral No. of file version BB : 00-99: First decimal No. of file version CC : 00-99: Second decimal No. of file version DD: 00-99: Third decimal No. of file version 0000XXZZ (for D^{term}85) <ul style="list-style-type: none"> XX: 00-99: Integral No. of file version ZZ : 00-99: First decimal No. of file version
	Read the type of the IP Station.	<ul style="list-style-type: none"> • Y=01 (1) X-XXXXXXXX: Station No. (2) 02 : D^{term}IP INASET 03 : D^{term}85 (D^{term} Series i) (IP adapter type) 05 : D^{term}85 (D^{term} Series i) (IP Bundled type) 10 : D^{term}SP30 12 : DT710 13 : DT730/DT710 (Self-Labeling) 14 : DT750 15 : MH240 16 : SP350 17 : DT730DG 18 : DT730CG 19 : DT770G
<div style="border: 1px solid black; padding: 2px; width: fit-content;">A</div>		

A	DESCRIPTION	DATA
CMFA	Read the type of the IP Station.	21 : DT830/DT830DG
		22 : DT830CG
	NOTE 1: <i>The terminal type is read as "12" (DT710) when using 9300V3 software or before.</i>	23 : DT820 NOTE 1 [9300V3 STEP2 software required]
	NOTE 2: <i>The terminal type is read as "13" (DT730/DT710 (Self-Labeling)) when using 9300V3 software or before.</i>	24 : DT820 (Self-Labeling) NOTE 2 [9300V3 STEP2 software required]
	NOTE 3: <i>The terminal type is read as "22" (DT830CG) when using 9300V6 software or before.</i>	25 : DT930CG NOTE 3 [9300V7 software required]
	NOTE 4: <i>The terminal type is read as follows.</i> - When using 9300V3 STEP2 to 9300V6 software: "23" (DT820) - When using 9300V3 software or before: "12" (DT710)	26 : DT920 NOTE 4 [9300V7 software required]
	NOTE 5: <i>DT510 is available from 9300V9 software or later. The terminal type between DT510 and DT410 cannot be identified by the system data. Also, DT510 is identified as DT410 when using 9300V6 software or before (DT510 cannot be accommodated to SV9300 when using 9300V7/V8 software).</i>	27 : DT920 (Self-Labeling) [9300V7 software required]
		28 : DT930 (Touch Panel) [9300V7 software required]
		41 : D ^{term} 85 (D ^{term} Series i)
		42 : D ^{term} 85 (D ^{term} Series i) Russian
		43 : DT310
		44 : DT330
		45 : DT330 Chinese
		46 : DT510/DT410 NOTE 5
		47 : DT430
		48 : DT430 Chinese
		49 : DT530 [9300V7 software required]
		50 : DT530 Chinese [9300V7 software required]
		70 : Paging Adapter
		80 : Standard SIP station
		FF : Other
	NONE◀: Logout Status/Disconnected	

END

Operating Procedure

No manual operation is required.

Bandwidth Control

General Description

This feature allows to assign an available bandwidth threshold for VoIP traffic within a Location and between Locations, and to restrict outgoing/incoming calls when the VoIP traffic exceeds the threshold.

The Location is a group of VoIP devices (IP Enabled Digital Multiline Terminal, VoIPDB, or IPT (P2P CCIS), which the same VoIP communications parameters such as codec selection list and ToS field value are assigned.

When the VoIP traffic over CCIS exceeds the threshold, the call can be routed to legacy trunks (TDM network).

Station Application

Not applicable

Service Conditions

1. Two kinds of threshold can be assigned on a Location-to-Location basis by system programming.
 - Limit Threshold : Maximum bandwidth available for VoIP traffic.
 - When the traffic exceeds this threshold, the later calls are restricted within that Location or between those Locations.
 - In case of CCIS calls over IP network, the calls can be routed to the legacy trunks (TDM network).
 - Warning Threshold: This threshold should be assigned lower than the Limit Threshold.
2. Each threshold can be assigned on a Location-to-Location basis.
3. Maximum 65534 Kbps can be assigned for each threshold on a 1 Kbps increment. If no data is assigned, 100 Mbps is assigned (default data).
4. The following parameters can be recorded as peg count on a Location-to-Location basis.
 - a. The number exceeding the Limit Threshold
 - b. The number exceeding the Warning Threshold
 - c. Maximum bandwidth actually used
 - d. The bandwidth currently used
5. The bandwidth managed by this feature is that of voice packets for Peer-to-Peer connections only. The data packets for control signals of IP Enabled Digital Multiline Terminal, CCIS and PCPro are not supported by this feature. The bandwidth of the voice packets for each call is defined by the combination of voice encoding type and payload size as follows.

Bandwidth of voice packets for each call (Both way)

Type of Voice Encoding	Payload Size (ms.)		
	20	30	40
G.711 (64 Kbps)	160 Kbps	147.4 Kbps	144 Kbps
G.729a (8 Kbps)	48 Kbps	37.4 Kbps	32 Kbps
G.723.1 (6.3/5.3 Kbps)	–	34.2 Kbps	–

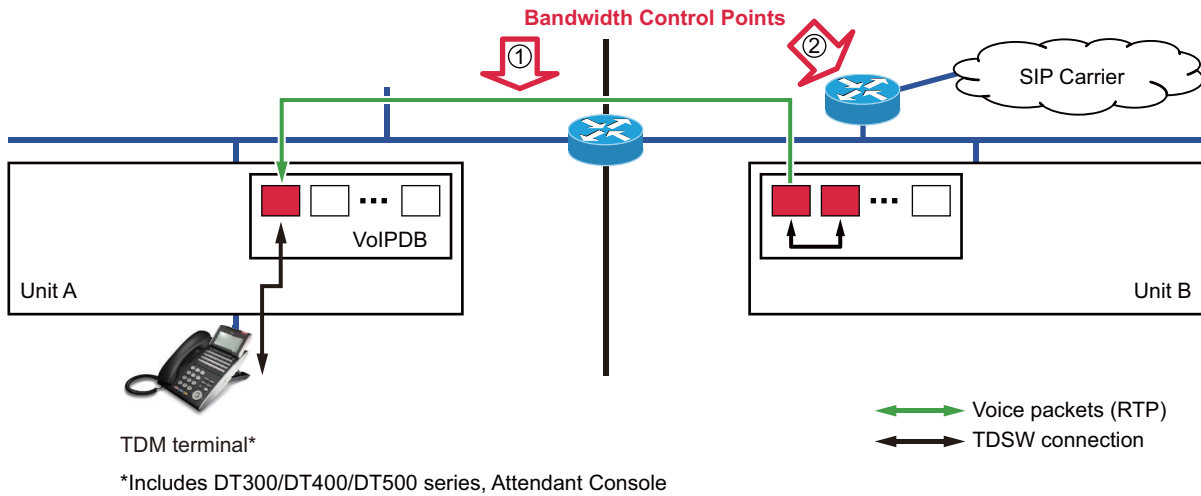
NOTE: *The bandwidth for G.723.1 is defined by which the voice encoding uses 6.3 Kbps.*

The system counts the bandwidth for each call based on the above table whenever the call is established, and the system deletes the bandwidth when the call is released.

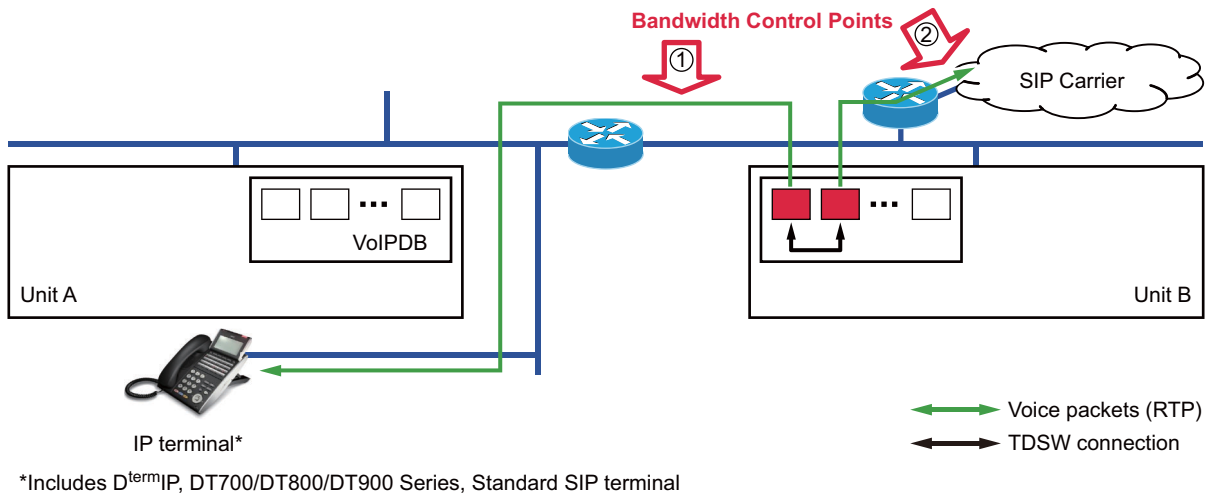
6. When an outgoing/incoming call is attempted, and the voice encoding type and payload size between the locations is not determined yet, the bandwidth for the call is temporarily determined based on the primary value in the CODEC list in the system. The system counts that temporary bandwidth.
7. When the call is established and the voice encoding type and payload size actually used are informed from the corresponding terminal, the system deletes the temporary bandwidth determined by #6 above and counts the bandwidth based on the voice encoding type and payload size actually used. In case of CCIS calls over IP network with Peer-to-Peer connections, the system maintains the bandwidth determined based on the primary value of the CODEC list in the system, as mentioned #6 above.
8. There are following conditions when the Limit Threshold is exceeded.
 - a. When an outgoing call is attempted and an answering party can be specified, the call is restricted when calling.
 - b. When an outgoing call is attempted but an answering party cannot be specified, the call is restricted after answered.
 - c. The bandwidth for a call on hold is not maintained.
 - d. When a held call is restricted from a different party from the party initiating hold and the Limit Threshold is exceeded, the retrieving the call is restricted and the both parties will receive reorder tone.
 - e. When a call is terminated to a Multiline group, the system will check the bandwidth for My Line only. Therefore, when the call is answered by Sub line and the Limit Threshold is exceeded, the call is restricted and the calling and called party will receive reorder tone.
9. Whether the call is restricted or not, when the Limit Threshold is exceeded, is assigned on a Location-to-Location basis by system programming.
10. When a call is restricted at originating the call, the calling party will hear reorder tone.
11. When a call is restricted at answering the call, both calling and called party will hear reorder tone.
12. When a call is restricted by this feature, no LCD indication is provided such as “RST”.

13. When an incoming CCIS call over IP network is restricted, the calling party will receive reorder tone.
14. Inter-unit connection has multiple IP segments according to the topology of the connection.
Control the bandwidth of each IP segment.

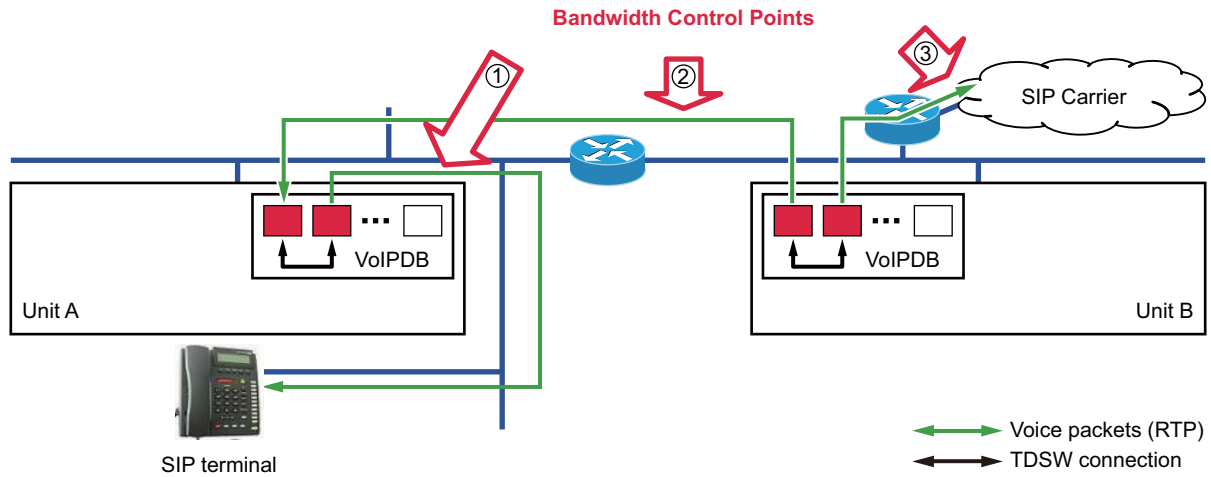
- TDM-SIP trunk



- IP terminal-SIP trunk



- SIP terminal-SIP trunk



Programming

START	DESCRIPTION	DATA
CM67	Specify the limit bandwidth for the traffic between locations to each location number assigned by CM12 Y=39, 50/CM8A Y=5000-5255>173.	<ul style="list-style-type: none"> • Y=90 (1) XX ZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) 00000-65534: 0 Kbps-65534 Kbps <p style="text-align: right;">NOTE 1</p> <p>NONE◀ : 100000 Kbps (100 Mbps)</p> <p>NOTE 1: Set the bandwidth for voice packets. The available bandwidth minus the bandwidth for control packets (40 Kbps) is the bandwidth for voice packets. If the reflection speed of terminals such as button reflection becomes slower by setting the value above-mentioned, set the bandwidth for voice packets to the value with the bandwidth for control packets supposed as more than 40 Kbps.</p> <p>NOTE 2: Assign the value that exceeds the warning bandwidth set by CM67 Y=92.</p>
A		

A	DESCRIPTION	DATA
CM67	<p>Specify the action when the traffic between locations exceeds the limit bandwidth assigned by CM67 Y=90.</p>	<ul style="list-style-type: none"> • Y=91 (1) XX ZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) 0 : Restrict the connection between location groups 3◀: Keep the connection between location groups
CMEA	<p>Specify the warning bandwidth for the traffic between locations to each location number assigned by CM12 Y=39, 50/CM8A Y=5000-5255>173.</p> <p>NOTE: Assign the value which does not exceed the limit bandwidth set by CM67 Y=90.</p>	<ul style="list-style-type: none"> • Y=92 (1) XX ZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) 00000-65534: 0 Kbps-65534 Kbps NONE◀ : 100000 Kbps (100 Mbps)
CMEA	<p>Specify if the fault information is stored into the memory, and the external alarm kind.</p>	<ul style="list-style-type: none"> • Y=2 (1) 040: Traffic of IP network exceeded limit bandwidth (Standard data: MJ) 041: Traffic of IP network exceeded warning bandwidth (Standard data: MN) 050: Traffic of IP network returned to normal condition from limit bandwidth excess (Standard data: --) 051: Traffic of IP network returned to normal condition from warning bandwidth excess (Standard data: --) (2) 0 : Fault memory store/No output of external alarm 1 : Fault memory store/External alarm is MN alarm 2 : Fault memory store/External alarm is MJ alarm 3◀: No fault memory store/No output of external alarm
END		

To read/clear the PEG count, do the following programming.

- When reading the PEG count between locations that exceeded the limit bandwidth
 $\boxed{\text{ST}} + \text{B400} + \boxed{\text{DE}} + 00\text{-}63$ (location number of sending side) +
 $00\text{-}63$ (location number of receiving side) + $\boxed{\text{DE}}$
- When clearing the PEG count between locations that exceeded the limit bandwidth
 $\boxed{\text{ST}} + \text{B400} + \boxed{\text{DE}} + 00\text{-}63$ (location number of sending side) +
 $00\text{-}63$ (location number of receiving side) + $\boxed{\text{DE}} + \text{CCC} + \boxed{\text{EXE}}$
- When clearing the PEG counts of all locations that exceeded the limit bandwidth
 $\boxed{\text{ST}} + \text{B400} + \boxed{\text{DE}} + 9999 + \boxed{\text{DE}} + \text{CCC} + \boxed{\text{EXE}}$
- When reading the PEG count between locations that exceeded the warning bandwidth
 $\boxed{\text{ST}} + \text{B401} + \boxed{\text{DE}} + 00\text{-}63$ (location number of sending side) +
 $00\text{-}63$ (location number of receiving side) + $\boxed{\text{DE}}$
- When clearing the PEG count between locations that exceeded the warning bandwidth
 $\boxed{\text{ST}} + \text{B401} + \boxed{\text{DE}} + 00\text{-}63$ (location number of sending side) +
 $00\text{-}63$ (location number of receiving side) + $\boxed{\text{DE}} + \text{CCC} + \boxed{\text{EXE}}$
- When clearing the PEG counts of all locations that exceeded the warning bandwidth
 $\boxed{\text{ST}} + \text{B401} + \boxed{\text{DE}} + 9999 + \boxed{\text{DE}} + \text{CCC} + \boxed{\text{EXE}}$

NOTE: *The PEG count of 00000-49999 can be stored to the system. If the PEG count exceeds 49999, the system counts it from 00000.*

To read/clear the bandwidth, do the following programming.

- When reading the maximum bandwidth between locations
 $\boxed{\text{ST}} + \text{B402} + \boxed{\text{DE}} + 00\text{-}63$ (location number of sending side) +
 $00\text{-}63$ (location number of receiving side) + $\boxed{\text{DE}}$
- When clearing the maximum bandwidth between locations
 $\boxed{\text{ST}} + \text{B402} + \boxed{\text{DE}} + 00\text{-}63$ (location number of sending side) +
 $00\text{-}63$ (location number of receiving side) + $\boxed{\text{DE}} + \text{CCC} + \boxed{\text{EXE}}$
- When clearing the maximum bandwidth of all locations in the system
 $\boxed{\text{ST}} + \text{B402} + \boxed{\text{DE}} + 9999 + \boxed{\text{DE}} + \text{CCC} + \boxed{\text{EXE}}$
- When reading the bandwidth that are used now
 $\boxed{\text{ST}} + \text{B403} + \boxed{\text{DE}} + 00\text{-}63$ (location number of sending side) +
 $00\text{-}63$ (location number of receiving side) + $\boxed{\text{DE}}$
- When clearing the bandwidth that are used now
 $\boxed{\text{ST}} + \text{B403} + \boxed{\text{DE}} + 00\text{-}63$ (location number of sending side) +
 $00\text{-}63$ (location number of receiving side) + $\boxed{\text{DE}} + \text{CCC} + \boxed{\text{EXE}}$
- When clearing all bandwidth that are used now
 $\boxed{\text{ST}} + \text{B403} + \boxed{\text{DE}} + 9999 + \boxed{\text{DE}} + \text{CCC} + \boxed{\text{EXE}}$

NOTE: *The bandwidth of 0000000-1677721 Kbps can be displayed on PCPro/CAT. Even if the bandwidth exceeds 1677721 Kbps, PCPro/CAT displays the bandwidth 1677721 Kbps.*

Operating Procedure

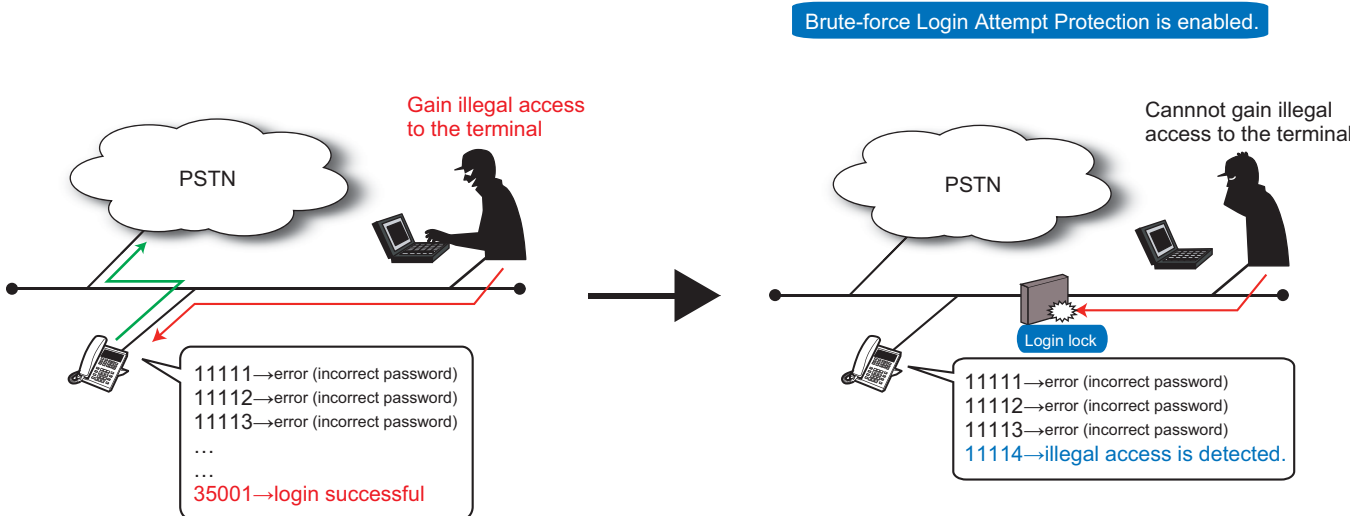
No manual operation is required.

Brute-force Login Attempt Protection

[9300V7 software required]

General Description

This feature prevents an unauthorized person from performing a brute-force login attack. If the number of failed login attempts counted by the system exceeds a certain number within the period of time, the system takes it as a brute force login attempt and prevents the person from logging into the station number.



Station Application

Standard SIP terminals (including smartphones)

Service Conditions

1. The **9300V7** or later System Version License is required for this feature.
2. When a station user fails to log into the terminal by using incorrect passwords several times (default setting: three times), the terminal will be locked.
3. The login operation to the locked station number is disabled.
4. If a retransmission of the login occurs due to the network factors or specifications of the terminal, the number of failures may be more than one time.
5. The connection status of the locked station is same as “Not connected.”
6. To enable this feature (CM08>1062:0), set Standard SIP station Authentication to “Allowed” (CM1D Y=32:15). The login operation to the station number assigned “Restricted” (CM1D Y=32:03) is disabled. For the details, refer to Authentication data assignment for Standard SIP station.

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NOTE: Even if this feature is effective (CM08>1062:0), Standard SIP stations without REGISTER (assigned by CMBC Y=15) can be used. It is not necessary to change the second data of CM1D Y=32 to 15 (Allowed).

7. The login lock state can be confirmed by the system data (CM12 Y=101) from PCPro/CAT.

8. The lockout state can be kept until it is canceled manually by using Time to Login lock (CM42 Y=221:00).
9. The login lock is canceled by the following cases:
 - The specified time (default setting: 10 minutes) has passed since locked. (CM42 Y=221)
 - The authorization code/password per station is changed or cleared with PCPro/CAT. (CM2B Y=00)
 - Canceling the login lock with PCPro/CAT (CM12 Y=101:CCC)
10. After system reset, the number of login failure is cleared and the login lock state is canceled. (The state before system reset cannot be kept.) It is the same as the following changeover:
 - System changeover occurs in dual CPU system.
 - Remote Unit changes over the operation mode (normal mode/survival mode) in Remote Unit over IP system.
 - Secondary Unit changes over from Normal Mode to Failover Mode in Failover system.
11. The number of login failure is cleared by the following cases:
 - The login succeeds.
 - The authorization code/password per station is changed or cleared with PCPro/CAT. (CM2B Y=00)
 - The login lock is forcibly canceled with PCPro/CAT. (CM12 Y=101: CCC)
12. When a login lock by Brute-force Login is attempted or the restriction is canceled, the following fault information is registered into the memory according to the system data (CMEA Y=2>134, 135):
 - Fault kind number 134 (Login lock by Brute-force Login Attempt)
 - Restoration kind number 135 (Login lock returned to normal condition)

Programming

(1) Assignment of Brute-force Login Attempt Protection

START	DESCRIPTION	DATA
CM08	Assign Brute-force login attempt protection.	(1) 1062 (2) 0 : Available 1◀ : Not Available
	NOTE: <i>To enable this feature, be sure to set CMID Y=32:15.</i>	
CM42	Assign the number of login failure to Login lock.	(1) 220 (2) 00 : No limit time(s) 01-99 : 1-99 time(s) NONE◀ : 3 times
	Assign time to Login lock.	(1) 221 (2) 00 : Fixed lockout 01-99 : 1-99 minute(s) NONE◀ : 10 minutes
END		

(2) Reading Login Lock State and Login Lock Forced Cancellation

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM12</div>	Assign Brute-force login attempt protection.	<ul style="list-style-type: none"> • Y=101 (1) X-XXXXXXXX: Station No. (2) 0 : Normal 1 : Login lock NONE◀: Not supported station CCC : To cancel (Forced) <p>NOTE 1: When a station is not locked, this data is set to “0” whether the station is connecting or not.</p> <p>NOTE 2: Setting “CCC” when the Login lock state is “0”, the number of login failure is cleared.</p>
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">END</div>		

(3) Assignment of Fault Information Collection

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CMEA</div>	Register the fault information into memory and the control of the external alarm for Brute-force login attempt protection.	<ul style="list-style-type: none"> • Y=2 (1) 134 (Login lock occurred) 135 (Login lock returned to normal condition) (2) 0 : No Fault Memory store/No output of External Alarm 1 : Fault Memory store/External Alarm is MN alarm 2 : Fault Memory store/External Alarm is MJ alarm 3◀ : Fault Memory store/No output of External Alarm
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">END</div>		

Operating Procedure

No manual operation is required.

Call Forwarding-Logout (IP Multiline Terminal)

General Description

This feature allows a call terminated to an IP Station in logout status to be forwarded to a predesignated station, outside number, Attendant Console or Voice Response System (VRS). This feature is also applicable to the IP Stations that the LAN cable is pulled out or the power is off.

Station Application

D^{term}85 (Series i) Terminals with LCD equipped for IP Adapter units (Peer-to-Peer version), D^{term}IP INASET, D^{term}SP30, DT700/DT800/DT900 Series.

Service Conditions

1. This feature is effective when an IP Station is in the following logout conditions.
 - When the IP Station is logged out by Logout Feature Access Code.
 - When Login Dialog is displayed on the IP Station.
 - When “Connecting.....” is displayed on an IP Station.
 - When the IP Station is in the Configuration mode for more than two minutes.
 - When the IP Station is reset by pulling out the LAN cable (When the LAN cable is pulled out for more than two minutes, the IP Station is automatically reset).
 - When the power of the IP Station is OFF.
2. When an IP Station is in a Logout status, calls cannot be terminated to the sublines on that IP Station.
3. When an IP Station in a Logout status is in a multiline group, a call to that IP Station cannot ring other stations in that multiline group.
4. A Call Forwarding destination can be a station, a Voice Mail System, an outside number, an Attendant Console and a Voice Response System (VRS).
5. When the Call Forwarding destination is a station, an outside number or an Attendant Console, this feature can be set by dialing Call Forwarding-Logout Set/Cancel Access Code or data assignment from a PCPro/CAT. Setting by Feature Access Key is not supported.

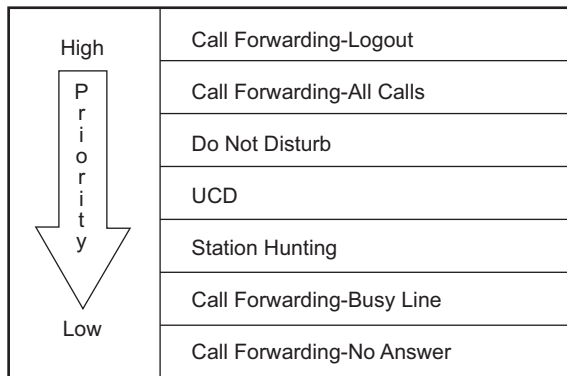
6. When the Call Forwarding destination is an outside number, there are following conditions.
 - a. When an IP Station in a logout status receives an incoming trunk call and Call Forwarding destination is an outside number (tandem connection), the call is forwarded or not in accordance with the restriction data for tandem connections (CM36). When the tandem connection between the incoming trunk route and outgoing trunk route is allowed by system programming, the call is forwarded to the outside number. When it is restricted, the caller will hear ROT.
 - b. There is no limitation for the number of registrations when a Call Forwarding destination is a station. You can verify the number of registrations by CMEC Y=2 when a Call Forwarding destination is an outside number. For details, refer to the Call Forwarding-All Calls feature in the “Programming Manual”.
 - c. Whether an IP Station is busy or not, the PCPro/CAT can set/change the Call Forwarding-Logout destination.
 - d. When a toll call number is assigned as the Call Forwarding destination, a calling station that is restricted to make a toll call can be forwarded to that toll call number.
7. When the Call Forwarding destination is a VRS, this feature can be assigned on a tenant basis by system programming. However, the Call Forwarding destination assigned by Feature Access Code or PCPro/CAT (station, outside number, Attendant Console) has priority over that assigned by the system data (VRS).

Up to 8 multi-connections are available per VRS. When the connection exceeds 8, the excess connections are restricted and the caller will hear ROT. Secondary callers cannot be connected to the beginning of the message. A message duration for the announcement is assigned by CM41 Y=0>102. When the message duration exceeds the time assigned by the system programming, the caller will hear ROT.
8. When the Call Forwarding destination is not assigned, a caller will hear ROT or RBT.

If the called IP Station sets Call Forwarding-All Calls, the caller will be forwarded to the destination of Call Forwarding-All Calls. Even if the called IP Station sets any of Call Forwarding-Busy Line, Call Forwarding-No Answer, Do Not Disturb and Station Hunting, such service will be ignored and the caller will hear ROT or RBT. ROT or RBT can be selected in station Class of Service assignment.

9. Figure below shows a service priority applied to an IP Station that sets more than one Call Forwarding or Hunting features at the same time. Call Forwarding-Logout, Call Forwarding-All Calls, Do Not Disturb, Uniform Call Distribution (UCD), Station Hunting, Call Forwarding-Busy Line, Call Forwarding-No Answer are listed in the order of priority. Note that when the IP Station in the logout status does not set both Call Forwarding-Logout and Call Forwarding-All Calls, the caller will not be forwarded by Do Not Disturb, UCD, Station Hunting, Call Forwarding-Busy Line, or Call Forwarding-No Answer and will hear ROT or RBT.

Service Priority in Logout Status



*When neither Call Forwarding-Logout nor Call Forwarding-All Calls is set, the caller will hear ROT/RBT.

10. Call Forwarding-Logout service can be allowed or denied in Class of Service assignment. Also, ROT/RBT, which calling party hears when Call Forwarding destination is not assigned, can be selected in Class of Service assignment.
11. Multiple Call Forwarding-Logout is available. Multiple Call Forwarding-Logout can be assigned by CM42>14.

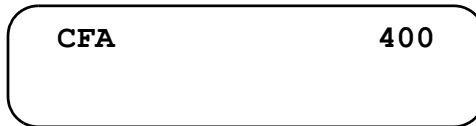
Multiple Call Forwarding-Logout can forward a maximum of five times when the called station sets Call Forwarding-Logout to a station that has set Call Forwarding-Logout, Call Forwarding-All Calls, or Call Forwarding-Busy Line. When Call Forwarding-Logout is repeated more than five times including Call Forwarding-All Calls and Call Forwarding-Busy Line, a calling party hears RBT from that point on.

When the call is forwarded, on the way of Multiple Call Forwarding-Logout feature, to a station that has set Call Forwarding-No Answer, the number of Call Forwardings so far is reset (=0) and Call Forwarding-Logout is carried out from the beginning.

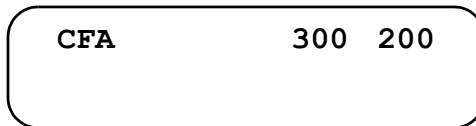
12. LCD displays of both calling party and Call Forwarding destination in Call Forwarding-Logout feature are illustrated below.

<Example> When Sta. 200 calls to Sta. 300 in logout status that sets Call Forwarding-Logout to Sta. 400:

- a. LCD display of calling party (Sta. 200)
(Called party: Sta. 300, Call Forwarding destination: Sta. 400)



- b. LCD Display of Call Forwarding destination (Sta. 400)
(Calling party: Sta. 200, Called party: Sta. 300)



Programming

To set a station, Voice Mail System, attendant console or outside party as the Call Forwarding-Logout destination:

NOTE: To set a Voice Mail System as the Call Forwarding-Logout destination, in addition to the following programming, do the programming of VOICE MAIL INTEGRATION.
For the programming of VOICE MAIL INTEGRAION, refer to the Programming Manual.

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required stations.	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Allow Call Forwarding-Logout in Service Restriction Class C assigned by CM12 Y=07. NOTE: To transfer calls to the VRS when Call Forwarding-Logout is not set to the station/C.O.line/Attendant Console, set the second data to "02".	<ul style="list-style-type: none"> • Y=481 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 02 :Allow (Send Announcement to VRS when no destination is set) 03◀:Allow (Send RBT when no destination is set)
CM29	Assign a Numbering Plan Group number to each tenant.	<ul style="list-style-type: none"> (1) 00-63: Tenant No. (2) 710-713 : Numbering Plan Group 0-3 NONE◀: Numbering Plan Group 0
CM20	Assign the access code for Call Forwarding-Logout Set/Cancel, respectively.	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A241: Call Forwarding-Logout Set A242: Call Forwarding-Logout Cancel
CM42	Specify the number of Call Forwarding for a call.	<ul style="list-style-type: none"> (1) 14 (2) 01-05 : 1 time-5 times NONE◀: 5 times
A		

A	DESCRIPTION	DATA
CME6	When setting the destination of Call Forwarding-Logout from PCPro/CAT, assign the destination No. to the required stations.	<ul style="list-style-type: none"> • Y=06 (1) X-XXXXXXXX: Station No. (2) Destination No.; <li style="padding-left: 20px;">X-XXXXXXXX: Station No. <li style="padding-left: 20px;">E000 : Attendant Console <li style="padding-left: 20px;">YY...Y : Outgoing Trunk Access Code <li style="padding-left: 40px;">(1-4 digits) + [] + Called No. <li style="padding-left: 40px;">(Maximum 26 digits) NONE◀: No data
CM35	To apply this feature to incoming calls, set the trunk route combination for Tandem connection.	<ul style="list-style-type: none"> • Y=005 (1) 00-63: Trunk Route No. (2) 1◀: Release signal arrives
CM36	Allow the connection between Incoming Trunk Route and Outgoing Trunk Route for Tandem connection.	<ul style="list-style-type: none"> • Y=0 (1) Incoming Trunk Route No. + Outgoing Trunk Route No. assigned by CM35 Y=005 (2) 0: Allow
<u>END</u>		

To set the Voice Response System (VRS) as the Call Forwarding-Logout destination:

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required stations.	<ul style="list-style-type: none"> Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Allow Call Forwarding-Logout in Service Restriction Class C assigned by CM12 Y=07.	<ul style="list-style-type: none"> Y=481 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 02: Allow (Send Announcement to VRS when no destination is set)
CM29	Assign a Numbering Plan Group number to each tenant.	(1) 00-63: Tenant No. (2) 710-713 : Numbering Plan Group 0-3 NONE◀: Numbering Plan Group 0
CM20	Assign the access code for Call Forwarding-Logout Set/Cancel, respectively.	<ul style="list-style-type: none"> Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A241: Call Forwarding-Logout Set A242: Call Forwarding-Logout Cancel
CM42	Specify the number of Call Forwarding for a call.	(1) 14 (2) 01-05 : 1 time-5 times NONE◀: 5 times
CM41	Specify the timing which an announcement for Call Forwarding-Logout is sent.	<ul style="list-style-type: none"> Y=0 (1) 102 (2) 01-99: 4-396 seconds (4 seconds increments) If no data is set, the default setting is 116-120 seconds.
CM49	Assign the announcement service for Call Forwarding-Logout to each Voice Response System accommodated into the system.	<ul style="list-style-type: none"> Y=00 (1) XXX: 000-015: VRS No. (2) 21XX: Announcement Service for Call Forwarding-Logout XX : Message Group No. (00-63)
A		

	DESCRIPTION	DATA
A		
CM49	Assign the message group number to the tenant number.	<ul style="list-style-type: none"> • Y=14 (1) 00-63: Tenant No. (2) 00-63: Message Group No. (Message Group No. assigned by CM49 Y=00)
CM51	When transferring calls to VRS (CM15 Y=481: 02), assign the VRS number for the destination of Call Forwarding-Logout to each tenant.	<ul style="list-style-type: none"> • Y=32 (1) 00-63: Tenant No. (2) EBXXX XXX : 000-015: VRS No. NONE◀: No data
END		

Operating Procedure

Call Forwarding to a station

■ To set by Call Forwarding-Logout Set Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Set Access Code and receive special dial tone.
3. Dial a desired station number.
4. “Set OK” is displayed and receive service set tone.
5. Press **Speaker** Key.

■ To cancel by Call Forwarding-Logout Cancel Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Cancel Access Code.
3. “Reset OK” is displayed and receive service set tone.
4. Press **Speaker** Key.

■ To cancel by Call Forwarding-Logout Set Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Set Access Code and receive special dial tone.
3. Call forwarding destination is displayed.
4. Press “*”.
5. “Reset OK” is displayed and receive service set tone.
6. Press **Speaker** Key.

Call Forwarding to an outside number

■ To set by Call Forwarding-Logout Set Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Set Access Code and receive special dial tone.
3. Dial trunk access code and desired outside number.
4. “Set OK” is displayed and receive service set tone.
5. Press **Speaker** Key.

■ To cancel by Call Forwarding-Logout Cancel Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Cancel Access Code.
3. “Reset OK” is displayed and receive service set tone.
4. Press **Speaker** Key.

■ To cancel by Call Forwarding-Logout Set Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Set Access Code and receive special dial tone.
3. Call forwarding destination is displayed.
4. Press “*”.
5. “Reset OK” is displayed and receive service set tone.
6. Press **Speaker** Key.

Call Forwarding to Attendant Console

■ To set by Call Forwarding-Logout Set Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Set Access Code and receive special dial tone.
3. Dial Access Code for Attendant Console.
4. “Set OK” is displayed and receive service set tone.
5. Press **Speaker** Key.

■ To cancel by Call Forwarding-Logout Cancel Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Cancel Access Code.
3. “Reset OK” is displayed and receive service set tone.
4. Press **Speaker** Key.

■ To cancel by Call Forwarding-Logout Set Access Code

1. Press **Speaker** Key and receive dial tone.
2. Dial Call Forwarding-Logout Set Access Code and receive special dial tone.
3. Attendant Console for Call forwarding destination is displayed.
4. Press “*”.
5. “Reset OK” is displayed and receive service set tone.
6. Press **Speaker** Key.

FAX over IP

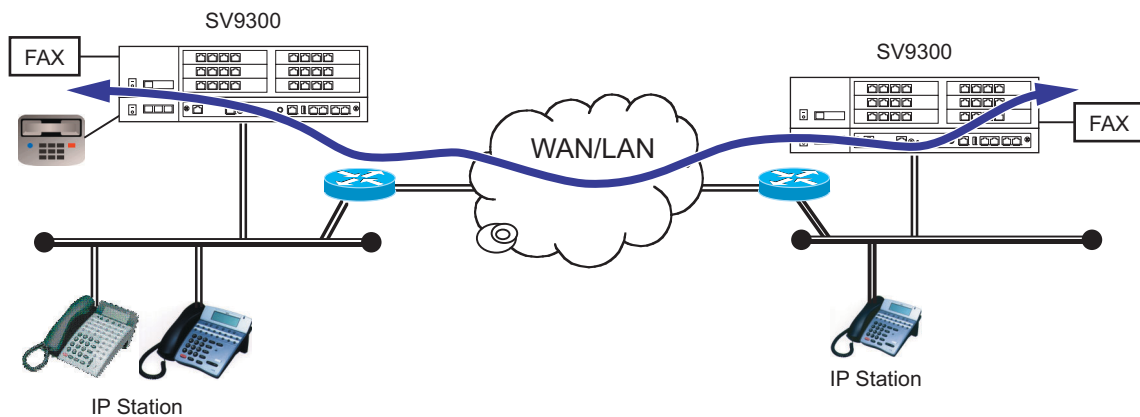
General Description

This feature allows the system to transmit facsimile communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN).

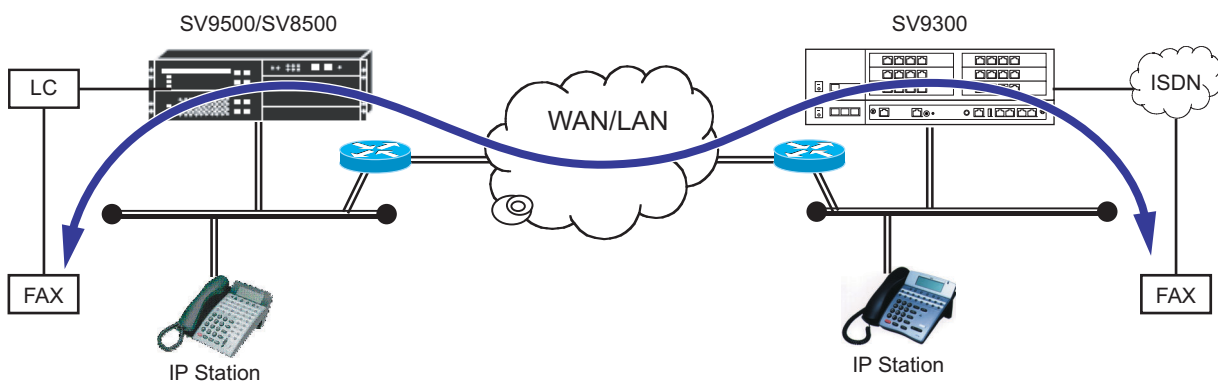
Since PBX regards facsimile equipment as one of ordinary telephones, VoIPDB is required for facsimile uses over IP network same as legacy stations. The facsimile transmission procedure (G.711/G.726 pass-through or T.38 (UDPTL)) is supported with VoIPDB/SIP.

The following figure shows a typical configuration of facsimile use on Peer-to-Peer CCIS network.

Example of FAX Use via Peer-to-Peer CCIS/SIP (SV9300 and SV9300)



Example of FAX Use via Peer-to-Peer CCIS/SIP (SV9500/SV8500 and SV9300)



Station Application

G3 facsimile stations

Service Conditions

Conditions on Overall System

1. VoIPDB is required for facsimile use on Peer-to-Peer CCIS network or Remote UNIT over IP function.
2. This feature supports FAX over IP with the following FAX protocol. The payload type is fixed to NEC's proprietary procedure, enhanced payload type (103) when CMBA Y=119: 01/02/03 is set.
 - G.711 Pass-through
 - G.726 Pass-through
 - T.38 (UDPTL)
3. The analog Media Converter used by 2400 IPX cannot be accommodated in SV9300.
4. A problem may occur such as "expected transmission speed is not obtained" or "not connectable", depending on a facsimile model.
5. If a Super G3 facsimile is used, the transmission speed will be equivalent to G3.
6. If a facsimile with Error Correction Mode (ECM) function is used, ECM does not work.
7. When the FAX station is specified by the system data (CM13 Y=07), the ringing interval to the FAX station shall be as per the setting of ringing signal for FAX (CM04 Y=00>09) (default: as per CM04 Y=00>05)
8. Connectable combinations between facsimile stations are shown below.

Connectable Combinations			Remarks
Source	Via	Destination	
SV9300 FAX station	Local	SV9300 FAX station	Station-to-Station connection
SV9300 FAX station	static (SIP)	SV9300 FAX station	
SV9300 FAX station	Peer-to-Peer CCIS	SV9300 FAX station	
SV9300 FAX station	Peer-to-Peer CCIS	2400 IPX FAX station	NOTE
SV9300 FAX station	Peer-to-Peer CCIS	2400 IPX MC FAX station	
SV9300 FAX station	Peer-to-Peer CCIS	LC under SV9500/ SV8500	

NOTE: *The connection with T.38 (UDPTL) protocol is not available.*

9. IP-FAX cannot be accommodated as a FAX station.
10. The table below shows the payload type for FAX communication via Peer-to-Peer CCIS between SV9500/SV8500 and SV9300.

Call Termination		MG(PRI), MG(BRI), MC&MG-COT under SV9500/ SV8500	MG(SIP) under SV9500/SV8500
Call Origination	LC under SV9500/SV8500		
FAX under SV9300	T.38 (UDP)	N/A	N/A
FAX via ISDN under SV9300	T.38 (UDP)	N/A	N/A

N/A: FAX communication is not available.

11. Required bandwidth for FAX connection (default setting) are shown below.

Connection Conditions	Required Bandwidth (One-way)
G.711 pass-through, Payload=40 ms. (No IP Header compression in Router)	72 Kbps (FAX Payload=320 byte)
G.711 pass-through, Payload=40 ms. (with IP Header compression in Router)	65 Kbps (FAX Payload=320 byte)
G.726 pass-through, Payload=40 ms. (No IP Header compression in Router)	40 Kbps (FAX Payload=160 byte)
G.726 pass-through, Payload=40 ms. (with IP Header compression in Router)	33 Kbps (FAX Payload=160 byte)

* This data does not include MAC Header.

12. FAX communication mode of VoIPDB is G.711 in default. To set the voice pass-through mode, FAX related data setting is required in system data programming.
13. The conditions for T.38 FAX communication are as follows.
- T.38 FAX supports only T.38 version = 0 (set by CM67 Y=21-24>19).
 - T.38 FAX supports up to 14400bps of FAX communication rate (set by CM67 Y=21-24>10).
 - T.38 FAX does not support T.38 FAX Fill Bit Removal.
 - T.38 FAX does not support T.38 FAX Transcoding MMR/JBIG.
 - T.38 FAX supports only T.38 (UDPTL) + Redundancy mode (does not support FEC).
 - T.38 FAX supports only UDP as the protocol to send T.38 packets (does not support TCP and RTP).
 - T.38 FAX supports up to 8 times of control data retransmission (set by CM67 Y=21-24>15).

- T.38 FAX supports up to 2 times of image data retransmission (set by CM67 Y=21-24>16).
- You cannot start speech communication after FAX communication. If you want to have a conversation on the phone after sending FAX data, you have to place a new call after the FAX communication is released.
- T.38 FAX does not support super G3 FAX and G4 FAX (other devices are supported).
- T.38 FAX can be mixed with G.711/G.726.
- The system data setting related to FAX protocol must be corresponded to the setting of the opposite PBX. When the setting of SV9300 is different from that of the opposite PBX, the call will be disconnected due to the FAX connection failure.
- VoIPDB changes the communication mode for FAX by CED/ANSam detection.
- The FAX terminal accommodated under SV9300 supports only the FAX connected under an LC blade.
- The LC blade is required for detecting revers signals when answering the FAX station accommodated under SV9300.

Conditions on SIP Trunk

1. The available CODEC for FAX communications via SIP trunk is G.711 pass-through and G.726 pass-through. The CODEC can be set by CMBA Y=119 for each profile.
2. When the FAX communication is detected during a call (mainly when using G.729a), CODEC can automatically be switched to G.711 pass-through or G.726 pass-through.
3. Available payload sizes for FAX communications are 20ms, 30ms and 40ms. Payload sizes can be set by system data for each profile.
4. Be sure to match CODEC and payload sizes of all devices in a SIP network.
5. With CMBA Y=119, you can set not to switch CODEC when FAX communications are detected. When not to switch CODEC, FAX communications are not available during a call using G.729a.
6. As for the FAX to be used, it is recommended to use the FAX equipped with ECM (Error Correction Mode) feature and make the setting effective.
7. As for FAX data in inter-unit connection, assign the FAX control data inter-unit by CM67 Y=20 and assign the FAX control data to SIP trunk carrier by CMBA Y=119.
8. Between SIP trunks, at tandem connection, FAX communication is not supported.
9. FAX communication may fail if delay or packet loss occurs in the IP network segment.
10. The conditions for T.38 FAX communications via SIP trunk are as follows.
 - The available CODEC is only T.38 (UDPTL) for T.38 FAX communications via SIP trunk.
 - T.38 FAX communications via SIP trunk support up to 14400 bps of FAX communication rate (set by CM67 Y=21-24>10).
 - T.38 FAX communications via SIP trunk support G3 FAX.
 - When answering from the analog G3 FAX station under SV9300, use an LC blade to detect answer signals.

- During T.38 FAX communications via SIP trunk, the operating conditions of Remote Units in Normal mode are as follows.
 - Remote Units communicate with the SIP network via SIP trunk of a Main Unit.
 - G.711 pass-through or G.726 pass-through is applied for the FAX communications between Remote Units and a Main Unit.
 - As for T.38 FAX communications from the FAX connected to the Remote Unit to the SIP network, the following number of VoIP channels are used.
 - Main Unit : 2 channels
 - Remote Unit: 1 channel
- The only “re-INVITE with SDP message” is supported to switch to T.38 FAX.
- Error responses for “re-INVITE with SDP message” when the switch to T.38 FAX is failed are as follows.
 - 488 Not Acceptable Here: The media negotiation for T.38 FAX failed.
 - 503 Service Unavailable : There is no license.
- When the media negotiation for T.38 FAX failed, the voice call is continued and FAX communications by G.711 pass-through or G.726 pass-through are performed.
 - *: Depending on the FAX specifications, there are cases that FAX communications by G.711 pass-through or G.726 pass-through are not available.
- It is impossible to switch to the calling state again after switching to FAX mode.
- To perform T.38 FAX communications, the SIP carrier which supports T.38 FAX for the opposite office is required.
- As for T.38 FAX communications via SIP trunk, Point-to-Point connection, Point-to-Multipoint connection and Multi-carrier connection are available.

Programming

(1) FAX station setting

START	DESCRIPTION	DATA
CM05	Assign the Unit number and Slot number for the LC blade. <div style="text-align: center; border: 1px solid black; border-radius: 15px; padding: 2px 10px;">BLADE RESET</div>	<ul style="list-style-type: none"> • Y=0 (1) XX ZZ XX: 01-50: Unit No. ZZ : 01-18: Slot No. (2) 20 : LC blade NONE◀: No data
CM10	Assign the station number for the FAX communication to the required Physical Port number.	<ul style="list-style-type: none"> • Y=00 (1) XX YY ZZ: Physical Port No. XX: 01-50: Unit No. YY: 01-18: Slot No. ZZ : 01-32: Circuit No. (2) X-XXXXXXXX: Station No. NONE◀: No data
CM13	Define the station number assigned by CM10 Y=00 as the FAX station.	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Station No. (2) 0: FAX Station
CM12	To provide the FAX station with the answering feature, assign Service Restriction Class A to the required FAX station number.	<ul style="list-style-type: none"> • Y=02 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class A
CM15	For the FAX station requiring Answer (reverse signal), restrict the Call Waiting Answer-Called Side in Service Restriction Class A assigned by CM12 Y=02.	<ul style="list-style-type: none"> • Y=044 (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 0: Restricted
CM13	Send the reverse signal to the FAX station requiring Answer. NOTE: <i>This command is effective when using the LC blade supports reverse signal.</i>	<ul style="list-style-type: none"> • Y=18 (1) X-XXXXXXXX: Station No. (2) 0: To send
A		

A	DESCRIPTION	DATA
CM04	Specify the ringing signal for FAX.	<ul style="list-style-type: none"> • Y=00 (1) 09 (2) 01 :ON 02 :2 seconds ON-4 seconds OFF 03 :1 second ON-2 seconds OFF 04 :0.5 seconds ON-0.5 seconds OFF 05 :0.25 seconds ON-0.25 seconds OFF 06 :0.5 seconds ON-0.5 seconds OFF 07 :0.25 seconds ON-0.25 seconds OFF 08 :0.375 seconds ON-0.25 seconds OFF- 09 :0.25 seconds ON-0.125 seconds OFF- 10 :1 second ON-4 seconds OFF 11 :0.25 seconds ON-0.25 seconds OFF 12 :1 second ON-3 seconds OFF 13 :0.25 seconds ON-0.25 seconds OFF 31 ◀:As per CM04 Y=00>05
<u>END</u>		

(2) FAX Protocol, FAX Payload size, Jitter Buffer settings for location via VoIPDB

To provide each location with the FAX over IP feature via VoIPDB, in addition to the “LOCATION DATA ASSIGNMENT”, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM0B</div>	Provide the FAX over IP feature to VoIPDB.	<ul style="list-style-type: none"> • Y=2XX (VOIP Port [2] + Unit No. [01-50]) (1) 54 (2) 0 : Not available 1 ◀: Available
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">A</div>	<p>NOTE: <i>The communication mode cannot be changed to the FAX mode when the second data is set “0” (not available).</i></p>	

A

CM67

DESCRIPTION

Assign the FAX control information list to each location number assigned by CM8A Y=5000-5255>173.

NOTE 1: Follow the typical default setting.

DATA

- Y=20
- (1) XXZZ
XX: 00-63: Location No.
ZZ : 00-63: Location No.
- (2) 1 : Fixed list 1 (See the table below)
2 : Fixed list 2 (See the table below)
3 : Fixed list 3 (See the table below)
4-7 : Programmable list 4-7 (depends on the setting of CM67 Y=21-24)
NONE◀: Fixed list 1 (See the table below)

	Fixed list 1	Fixed list 2	Fixed list 3
FAX Protocol	G.711	G.726	T.38 (UDPTL)
FAX Payload Size	40 ms.	40 ms.	20 ms.
Minimum Jitter Buffer	150 ms.	150 ms.	–
Maximum Jitter Buffer	150 ms.	150 ms.	–
Transmission Speed	–	–	14400 bps
The Number of Control Data Retransmission	–	–	3 times
The Number of Image Data Retransmission	–	–	0 time
T.38 Version	–	–	Version 0

NOTE 2: A control information list assigned by this command is valid for packets that are sent from the location which is set by “XX” in the first data to the location which is set by “ZZ” in the first data.

NOTE 3: When using SIP trunks accommodated in different units, assign an inter-unit FAX control information list between the units by this command.

B

B	DESCRIPTION	DATA
CM67	Assign the FAX protocol for the programmable list 4-7.	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 00: FAX Protocol Pattern No. (2) 00 : Not detect the FAX protocol NOTE 2 01 : G.711 μ-law 02 : G.711 A-law 03 : G.726 06 : T.38 (UDPTL) NONE◀: As per Fixed list 1 of CM67 Y=20
	<p>NOTE 1: <i>Follow the typical default setting.</i></p> <p>NOTE 2: <i>To avoid the false detection, set the second data to 00 when the locations (between locations) does not provide FAX over IP feature.</i></p> <p>NOTE 3: <i>When using SIP trunks accommodated in different units, assign an inter-unit FAX protocol information between the units by this command.</i></p>	*A-law/ μ -law depends on CM04 Y=10-59
C	Assign the FAX payload size for the programmable list 4-7.	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 01: FAX Payload Size Pattern No. (2) 02-04 : 20 ms.-40 ms. (10 ms. increments) NONE◀: As per Fixed list 1 of CM67 Y=20
	NOTE: <i>Follow the typical default setting.</i>	

C	DESCRIPTION	DATA
CM67	<p>Specify the maximum value of jitter buffer for the programmable list 4-7.</p> <p>NOTE 1: <i>Follow the typical default setting.</i></p> <p>NOTE 2: <i>Assign the value which exceeds the minimum value of jitter buffer set by CM67 Y=21/22/23/24>08.</i></p> <p>NOTE 3: <i>When this data is set to other than default (NONE), ensure that CM67 Y=21/22/23/24>08 is also set to other than default (NONE).</i></p> <p>NOTE 4: <i>Assign the value which is multiple of the payload size set by CM67 Y=21/22/23/24>01. If the setting value of the second data is not multiple of the payload size, the value remaining is omitted and assigned to the location number automatically.</i></p>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 09 (2) 02-30 : 20 ms.-300 ms. (10 ms. increments) NONE◀: 120 ms.
	<p>Assign the T.38 FAX transmission speed.</p>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 10 (2) 0 : 2400 bps 1 : 4800 bps 2 : 7200 bps 3 : 9600 bps 4 : 12000 bps 5 : 14400 bps NONE◀: 14400 bps
	<p>Assign the number of T.38 FAX control data retransmission.</p> <p>NOTE: <i>Follow the typical default setting.</i></p>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 15 (2) 0-8 : 0-8 times NONE◀: 3 times
	<p>Assign the number of T.38 FAX image data retransmission.</p> <p>NOTE: <i>Follow the typical default setting.</i></p>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 16 (2) 0-2 : 0-2 times NONE◀: 0 time
D		

D	DESCRIPTION	DATA
CM67	Assign the T.38 version. NOTE: <i>Follow the typical default setting.</i>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 19 (2) 0 : Version 0 (ASN.1 coding according to T.38 06-1998) 1 : Version 1 (1998 ASN.1 syntax, org doc: 11-2000) 2 : Version 2 (ASN.1 coding according to T.38 03-2002) 3 : Version 3 (2002 ASN.1 syntax extended, org doc: 04-2004) NONE◀: Version 0 (ASN.1 coding according to T.38 06-1998)
<u>END</u>		

Operating Procedure

No manual operation is required.

- (3) FAX Protocol, FAX Payload size, Jitter Buffer settings for location via SIP trunk
To provide each location with the FAX over IP feature via SIP trunk, in addition to the “LOCATION DATA ASSIGNMENT”, do the following programming.

NOTE: As for the settings of Multiple CODEC for the SIP trunk, refer to the “Multiple CODEC Selection”. [Page 2-330](#)

START	DESCRIPTION	DATA
CM0B	Provide the FAX over IP feature to VoIPDB. <div style="text-align: right; border: 1px solid black; border-radius: 10px; padding: 2px 10px;">RESET</div> <p>NOTE: The communication mode cannot be changed to the FAX mode when the second data is set “0” (not available).</p>	<ul style="list-style-type: none"> • Y=201 (VOIP Port [2] + Unit No. [01]) (1) 54 (2) 0 : Not available 1◀: Available
CMBA	Specify CODEC type of SIP trunk for FAX communication. <p>NOTE 1: This data is effective when the second data of CM0B Y=201>54 is set to “1”.</p> <p>NOTE 2: For the CODEC type when detecting FAX communication, the second data 01 (G.711 μ-law [Proprietary procedure, enhanced payload type]) is recommended.</p> <p>NOTE 3: When no data (NONE) is assigned to the second data, the changeover to the FAX communication from the voice communication using G.729a is not available.</p> <p>NOTE 4: Only 2nd data=06 is available for Point-to-Point connection (when CMA7 Y=46 is set to “1”).</p>	<ul style="list-style-type: none"> • Y=119 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 00 : No FAX mode detection 01 : G.711 μ-law (Proprietary procedure, enhanced payload type) 02 : G.711 A-law (Proprietary procedure, enhanced payload type) 03 : G.726 (Proprietary procedure, enhanced payload type) 06 : T.38 (UDPTL) 09 : G.711 μ-law (Proprietary procedure) 10 : G.711 A-law (Proprietary procedure) 11 : G.726 (Proprietary procedure) NONE◀: CODEC type is not changed for the FAX communication
A		

A	DESCRIPTION	DATA
CMBA	Specify the payload size for FAX communication from SIP trunk.	<ul style="list-style-type: none"> • Y=120 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 1 : 20 ms. 2 : 30 ms. 3 : 40 ms. NONE◀: Use payload size same as voice call's one
	<p>NOTE: <i>When the second data is set to NONE, the changeover to the FAX communication from the voice communication using G.729a is not allowed.</i></p> <p>Assign the SIP trunk location number.</p> <p>NOTE: <i>This data is effective only for T.38 FAX communication. Be sure to assign the SIP trunk location number by this data when communicating with T.38 FAX.</i></p>	<ul style="list-style-type: none"> • Y=138 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 00-63 : SIP Trunk Location No. NONE◀: As per CM0B Y=1XX>10
	Specify the FAX Jitter Buffer.	<ul style="list-style-type: none"> • Y=13 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 01-30 : 10 ms.-300 ms. (10 ms. increments) NONE◀: 150 ms.
B		

B

CM67

DESCRIPTION

DATA

Provide the T.38 FAX control information list to each location number assigned by CM8A Y=5000-5255>173.

- Y=20
- (1) XXZZ
XX: 00-63: Location No.
ZZ : 00-63: Location No.
- (2) 3 : Fixed list 3 (See the table below)
4-7 : Programmable list 4-7 (depends on the setting of CM67 Y=21-24)

	Fixed list 3
FAX Protocol	T.38 (UDPTL)
FAX Payload Size	20 ms.
Minimum Jitter Buffer	–
Maximum Jitter Buffer	–
Transmission Speed	14400 bps
The Number of Control Data Retransmission	3 times
The Number of Image Data Retransmission	0 time
T.38 Version	Version 0

NOTE 1: When the CODEC type of SIP trunk for FAX communication is set to T.38 (UDPTL) by CMBA Y=119, Fixed list 1 to 2 operate as Fixed list 3.

NOTE 2: A control information list assigned by this command is valid for packets that are sent from the location which is set by “XX” in the first data to the location which is set by “ZZ” in the first data.

NOTE 3: When using SIP trunks accommodated in different units, assign an inter-unit FAX control information list between the units by this command.

C

C	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM67</div>	<p>Assign the T.38 FAX transmission speed.</p>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 10 (2) 0 : 2400 bps 1 : 4800 bps 2 : 7200 bps 3 : 9600 bps 4 : 12000 bps 5 : 14400 bps NONE◀: 14400 bps
	<p>Assign the number of T.38 FAX control data retransmission.</p> <p>NOTE: <i>Follow the typical default setting.</i></p>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 15 (2) 0-8 : 0-8 times NONE◀: 3 times
	<p>Assign the number of T.38 FAX image data retransmission.</p> <p>NOTE: <i>Follow the typical default setting.</i></p>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 16 (2) 0-2 : 0-2 times NONE◀: 0 time
	<p>Assign the T.38 version.</p> <p>NOTE: <i>Follow the typical default setting.</i></p>	<ul style="list-style-type: none"> • Y=21/22/23/24 (1) 19 (2) 0 : Version 0 (ASN.1 coding according to T.38 06-1998) 1 : Version 1 (1998 ASN.1 syntax, org doc: 11-2000) 2 : Version 2 (ASN.1 coding according to T.38 03-2002) 3 : Version 3 (2002 ASN.1 syntax extended, org doc: 04-2004) NONE◀: Version 0 (ASN.1 coding according to T.38 06-1998)

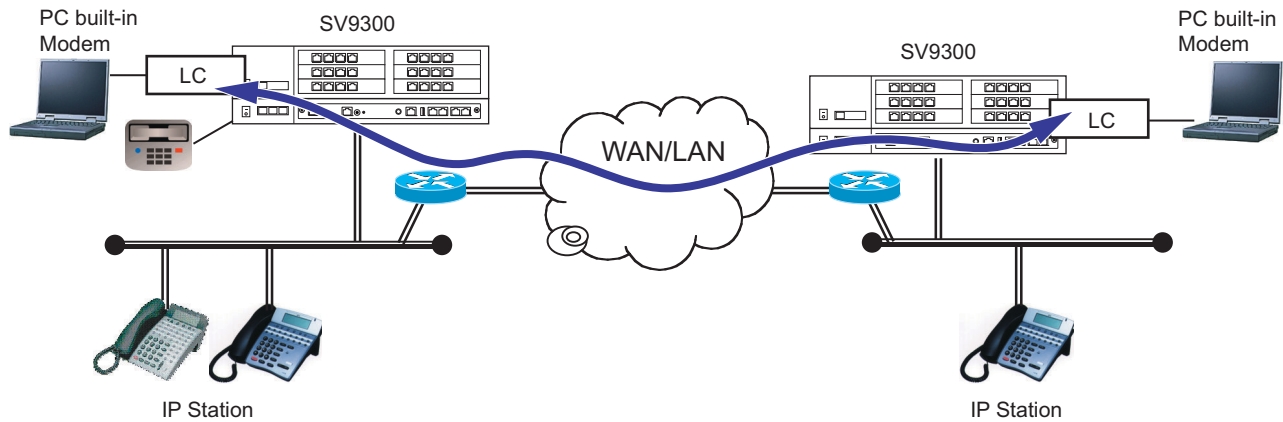
END

Modem over IP

This feature allows the system to transmit modem communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN).

The following figure shows a typical configuration of modem use on Peer-to-Peer CCIS network.

Example of the PC built-in modem Use via Peer-to-Peer CCIS



Station Application

Analog stations (modem)

Service Conditions

1. VoIPDB is required for Modem over IP use on Peer-to-Peer network or Remote UNIT over IP function.
2. Connectable combinations between modem stations are shown below.

Connectable Combinations			Remarks
Source	Via	Destination	
SV9300 MODEM station	Peer-to-Peer CCIS Local/Peer-to-Peer CCIS	SV9300 MODEM station	Peer-to-Peer CCIS Local (Remote UNIT over IP) Peer-to-Peer CCIS for stand alone system

3. The target value of the connection speed of modem communications over IP in SV9300 is shown in the table below.

NOTE: *The SV9300 does not guarantee the connection speed in the table that is a reference value. The actual speed varies depending on the connection configuration between modem, the conditions of network such as the number of hybrid circuits in the connection route, the quality of lines, or the type of modems. An evaluation test for the connectivity and the effective speed should be performed before actual operation.*

Connection Speed

Modem Protocol	Connection Speed
G.711 pass-through	14.4-24 Kbps
G.726 pass-through	9.6-14.4 Kbps

4. VoIPDB supports Modem over IP with the following modem protocol.
- G.711 Pass-through
 - G.726 Pass-through
5. Required bandwidth for modem connection (default setting) are shown below.

Connection Conditions	Required Bandwidth (One-way)
G.711 pass-through, Payload=40 ms. (No IP Header compression in Router)	72 Kbps (MODEM Payload=320 byte)
G.711 pass-through, Payload=40 ms. (with IP Header compression in Router)	65 Kbps (MODEM Payload=320 byte)
G.726 pass-through, Payload=40 ms. (No IP Header compression in Router)	40 Kbps (MODEM Payload=160 byte)
G.726 pass-through, Payload=40 ms. (with IP Header compression in Router)	33 Kbps (MODEM Payload=160 byte)

* This data does not include MAC Header.

Programming

(1) Modem station setting

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM05</div>	Assign the Unit number and Slot number for the LC blade. <div style="text-align: center; margin-top: 10px;"> <div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px;">BLADE RESET</div> </div>	<ul style="list-style-type: none"> • Y=0 (1) XX ZZ XX: 01-50: Unit No. ZZ : 01-18: Slot No. (2) 20 : LC blade NONE◀: No data
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM10</div>	Assign the station number for the MODEM communication to the required Physical Port number.	<ul style="list-style-type: none"> • Y=00 (1) XX YY ZZ: Physical Port No. XX: 01-50: Unit No. YY: 01-18: Slot No. ZZ : 01-32: Circuit No. (2) X-XXXXXXXX: Station No. NONE◀: No data
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>		

- (2) Modem Protocol, Modem Payload size, Jitter Buffer settings for location
To provide each location with the Modem over IP feature, in addition to the “LOCATION DATA ASSIGNMENT”, do the following programming.

START

DESCRIPTION

DATA

CM67

Assign the Modem control information list to each location number assigned by CM8A
Y=5000-5255>173.

NOTE 1: Follow the typical default setting.

- Y=20
- (1) XXZZ
XX: 00-63: Location No.
ZZ : 00-63: Location No.
- (2) 1 : Fixed list 1 (See the table below)
- 2 : Fixed list 2 (See the table below)
- 4-7 : Programmable list 4-7 (depends on the setting of CM67 Y=21-24)
- NONE◀: Fixed list 1 (See the table below)

	Fixed list 1	Fixed list 2
MODEM Protocol	G.711	G.726
MODEM Payload Size	40 ms.	40 ms.
Minimum Jitter Buffer	120 ms.	120 ms.
Maximum Jitter Buffer	120 ms.	120 ms.

NOTE 2: A control information list assigned by this command is valid for packets that are sent from the location which is set by “XX” in the first data to the location which is set by “ZZ” in the first data.

NOTE 3: When using SIP trunks accommodated in different units, assign an inter-unit modem control information list between the units by this command.

Assign the MODEM protocol for the programmable list 4-7.

NOTE 1: Follow the typical default setting.

NOTE 2: To avoid the false detection, set the second data 00 when the modem over IP feature is not provided between locations.

- Y=21/22/23/24
- (1) 00: Modem Protocol Pattern No.
- (2) 00: Not detect the Modem protocol
- 01 : G.711 μ-law
- 02 : G.711 A-law
- 03 : G.726
- NONE◀: As per Fixed list 1 by CM67
Y=20

NOTE 2

*A-law/μ-law depends on CM04 Y=10-59

A

A	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM67</div>	<p>Assign the Modem payload size for the programmable list 4-7.</p> <p>NOTE: <i>Follow the typical default setting.</i></p> <p>Specify the FAX jitter buffer for the programmable list 4-7.</p> <p>NOTE 1: <i>Follow the typical default setting.</i></p> <p>NOTE 2: <i>Assign the value which is multiple of the payload size set by CM67 Y=21/22/23/24>01. If the setting value of the second data is not multiple of the payload size, the value remaining is omitted and assigned to the location number automatically.</i></p>	<ul style="list-style-type: none"> • YY=21/22/23/24 (1) 01: Modem Payload Size Pattern No. (2) 02-04 : 20 ms.-40 ms. (10 ms. increments) NONE◀: As per Fixed list by CM67 Y=2 <ul style="list-style-type: none"> • Y=21/22/23/24 (1) 09 (2) 01-30 : 10 ms.-300 ms. (10 ms. increments) NONE◀: 150 ms.
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>		

Operating Procedure

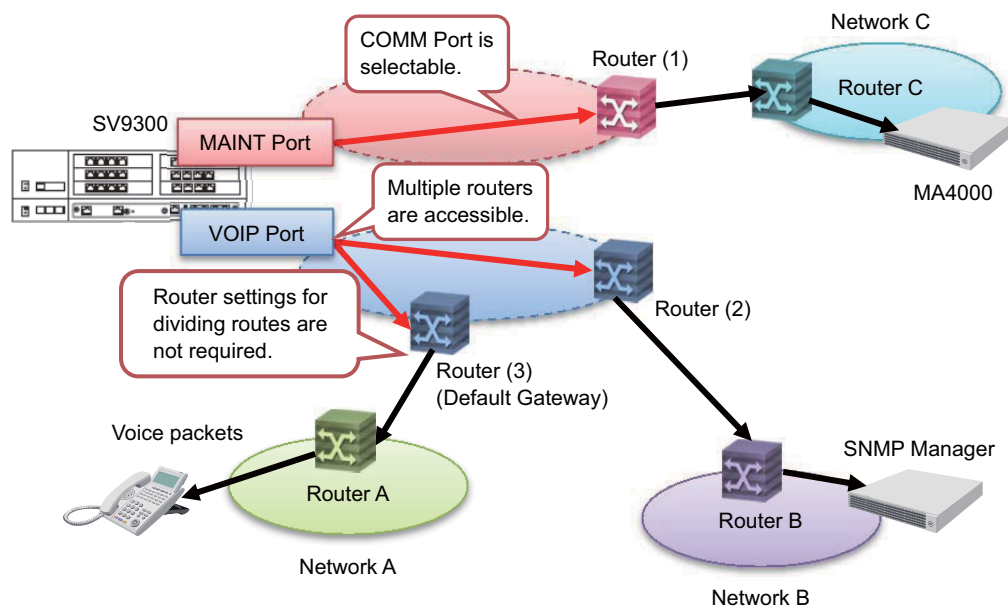
No manual operation is required.

Network Connection (For Multiple Routers)

[9300V4 software required]

General Description

Based on the built-in routing table of SV9300, the Network Connection (For Multiple Routers) feature divides the communications from application servers via routers to two or more gateway addresses. Therefore, this feature enables the communications beyond routers from both Maintenance port and VOIP port. Besides, by placing devices and applications in use to a segment different from the connected port and distributing traffic concentrated to a router for dividing routes (Default Gateway), this feature provides a stable communication.



NOTE: When using 9300V3 software or before, only the port where default gateway address (CM0B Y=0XX [Maintenance port] or CM0B Y=1XX [VOIP port]) is assigned can perform a communication beyond a router. The router that can be connected to the communication port is limited to one router.

Service Conditions

1. The setting of routing table (CM0D Y=00) can be registered only for Main Unit.
2. This feature is operated by the static routing method.
3. The communications beyond routers are performed that the IP address assigned to the default gateway (Maintenance port [CM0B Y=001]/VOIP port [CM0B Y=101]), or each router of routing table's default gateway assigned by CM0D Y=00 as a starting point.
4. A maximum of 20 routing settings can be registered in a routing table.
5. This feature allows the router connection for up to 21 routers.
Default gateway address (1 router) + Gateway addresses assigned to a routing table (up to 20 routers)
6. For the setting of a routing table, the registration of the following parameters is required by CM0D Y=00.

Parameters	Description
Destination address (0.0.0.1-255.255.255.254)	Assign the destination network address or the host address to which you want to communicate.
Net Mask (255.0.0.0-255.255.255.255)	Assign the destination Subnet Mask to which you want to communicate. * Assign "255.255.255.255" when the destination is a host address.
Gateway Address (0.0.0.1-255.255.255.254)	Assign the IP address of a router to be connected in a communication. * Default Gateway Address can be set only the address that allows the communication directory from the IP address of Maintenance port (CM0B Y=0XX)/VOIP port (CM0B Y=1XX) without a router (available to set only the address within the same network as the IP address of Maintenance port or VOIP port).

7. When the routing setting is changed by CM0D Y=00, the setting is immediately reflected. To reflect the setting when loading system data, a system reset is required.
8. The routing table of this feature does not support IPv6.
9. As for the features involving the VoIP packet communications (the communication between Units, IP telephone accommodation, IP trunk [CCIS/SIP] accommodation), set to communicate with the router assigned to the default gateway as usual.
10. Among the routing settings registered to a routing table, if there are two or more candidates for packets sending destination, the gateway address of packets sending destination is determined according to the longest match principle* .
* The routing setting that the address bit of (prefix long) a destination address which best matches is selected.

11. Target applications or non-target applications of the routing by this feature are as follows.

<Target applications>

- PCPro, MA4000
- OAI (such as BCT, OW5000, Presence Server, UCE) **NOTE**
- SMDR
- Hotel Server (an extended software for hotel features)
- SNMP manager

<Non-target applications>

- SIP Server (carrier connections of SIP trunk)
- Collaboration Server (3C)
- SNTP Server
- OAI (including IVR features) **NOTE**

NOTE: *IVR features (features which connect with PBX as an IP telephone and SIP trunk) do not support.*

Programming

(1) Routing Table Assignment

To assign the routing table, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM0D</div>	Assign the routing table.	<ul style="list-style-type: none"> • Y=00 (1) 01XX 01 : Fixed value XX: 00-19: Entry No. (2) 0/aaa...aaa/bbb...bbb/ccc...ccc/01 <p style="text-align: right; color: red; margin-top: 0;">NOTE 2 to NOTE 6</p> <p>CCC : Clear NONE◀: No data</p> <p>NOTE 1: <i>This command can be assigned only from PCPro (CAT mode is not available).</i></p> <p>NOTE 2: <i>The set contents of the second data (table data) are as follows.</i> 0D00>01XX: 0/aaa...aaa/bbb...bbb/ccc...ccc/01 0 : Fixed value aaa...aaa: Destination Address (0.0.0.1-255.255.255.254 [Maximum 15 digits]) bbb...bbb: Netmask (255.0.0.0-255.255.255.255 [Maximum 15 digits]) ccc...ccc : Gateway Address (0.0.0.1-255.255.255.254 [Maximum 15 digits]) 01 : Fixed value</p> <p>NOTE 3: <i>The second data cannot be overwritten. To change the set contents of the second data, clear the data by “CCC”, then register again. “SD CODE NOT ALLOWD” is displayed when overwriting.</i></p> <p>NOTE 4: <i>If the setting of routing table is failed. The following errors are displayed.</i> - “DATA ERROR”: The setting value is invalid. - “ASSIGNED ALREADY”: The same destination network address has already been registered.</p> <p>NOTE 5: <i>For a network specification (destination is a range specification), assign the destination network address to the destination address and assign the netmask to the range.</i> <An setting example> 0/192.168.2.0/255.255.255.0/192.168.1.254/01: Packets for 192.168.2.0-192.168.2.255 are sent to 192.168.1.254 (router).</p>
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">A</div>		

A	DESCRIPTION	DATA
CM0D	<p>NOTE 6: For a host specification (one router for destination), assign the destination host address to the destination address and assign "255.255.255.255" to the Netmask. <An setting example> 0/192.168.2.123/255.255.255.255/192.168.1.254/01: Only packets for 192.168.2.123 are sent to 192.168.1.254 (router).</p> <p>NOTE 7: Default Gateway Address can be set only the address that allows the communication directory from the IP address of Maintenance port (CM0B Y=0XX)/VOIP port (CM0B Y=IXX) without a router (available to set only the address within the same network as the IP address of Maintenance port or VOIP port).</p>	
END		

(2) Routing Table Status Display

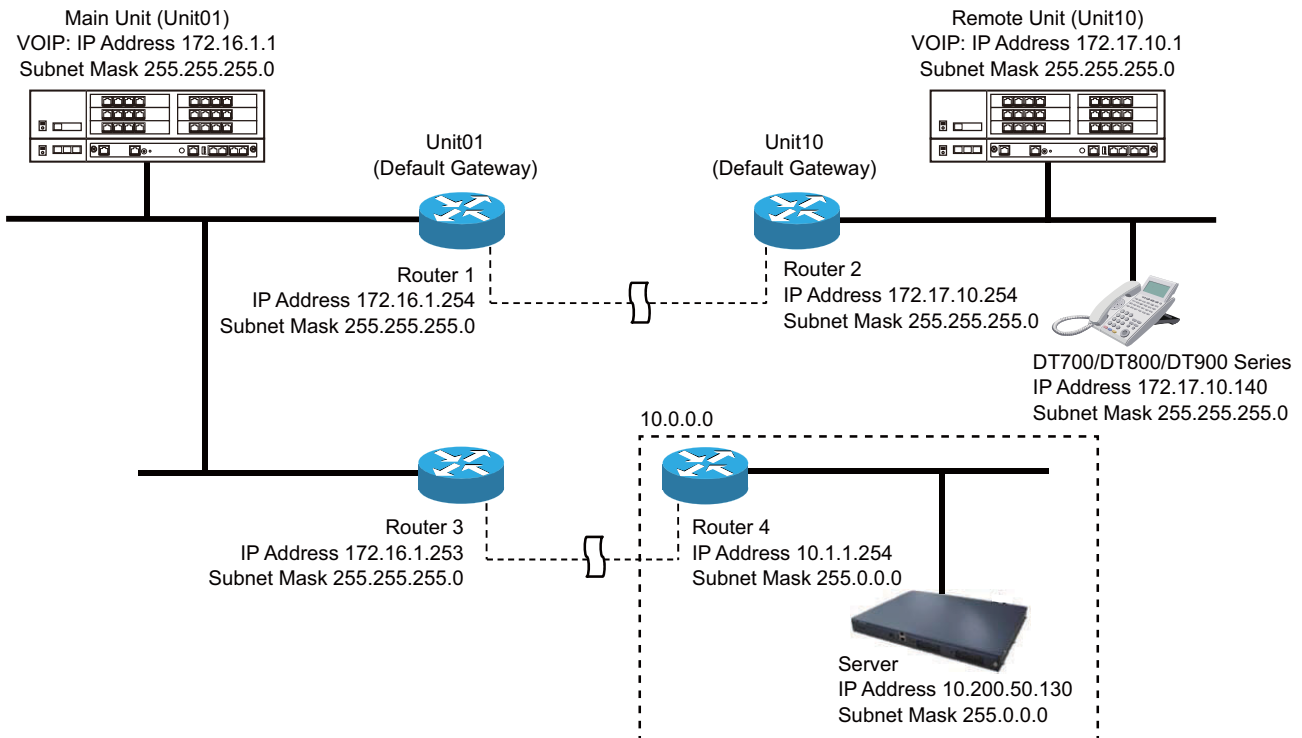
A Routing Table status can be displayed by the following system data.

START	DESCRIPTION	DATA
CM0D	Display the Routing Table status.	<ul style="list-style-type: none"> • Y=01 (1) 01XX 01 : Fixed value XX: 00-19: Entry No. (2) Z 0/aaa...aaa/bbb...bbb/ccc...ccc/01 <p style="text-align: right;">NOTE</p> <p style="text-align: right;">NONE◀: No data</p> <p>NOTE: The read contents of the second data (table data) are as follows. 0D01>01XX: Z 0/aaa...aaa/bbb...bbb/ccc...ccc/01 Z : Setting status of the Routing Table A/N: Routing table status applied/Not applied 0 : Fixed value aaa...aaa: Destination Address (0.0.0.1-255.255.255.254 [Maximum 15 digits]) bbb...bbb: Netmask (255.0.0.0-255.255.255.255 [Maximum 15 digits]) ccc...ccc : Gateway Address (0.0.0.1-255.255.255.254 [Maximum 15 digits]) 01 : Fixed value</p>
END		

■ Setting Example

To use multiple routers on the VOIP port side in the following system configuration:

- *1 The communication between the Main Unit and Remote Unit is performed via Default Gateway (Router 1, Router 2) from VOIP port (a Routing Table is not used).
- *2 For the communication to the server (10.200.50.130), set the Routing Table to use the Router 3 (172.16.1.253).



• Setting Example of Routing Table

CM	Y No.	1ST DATA	2ND DATA	REMARKS
0D	00	0100	0/10.0.0.0/255.0.0.0/ 172.16.1.253/01	To the Routing Table 00; (1) Assign the destination network address to be communicated to “10.0.0.0”. (2) Assign the destination Subnet Mask to be communicated to “255.0.0.0”. (3) Assign the IP address of the router to be connected at communication to “172.16.1.253 (Router 3)”.

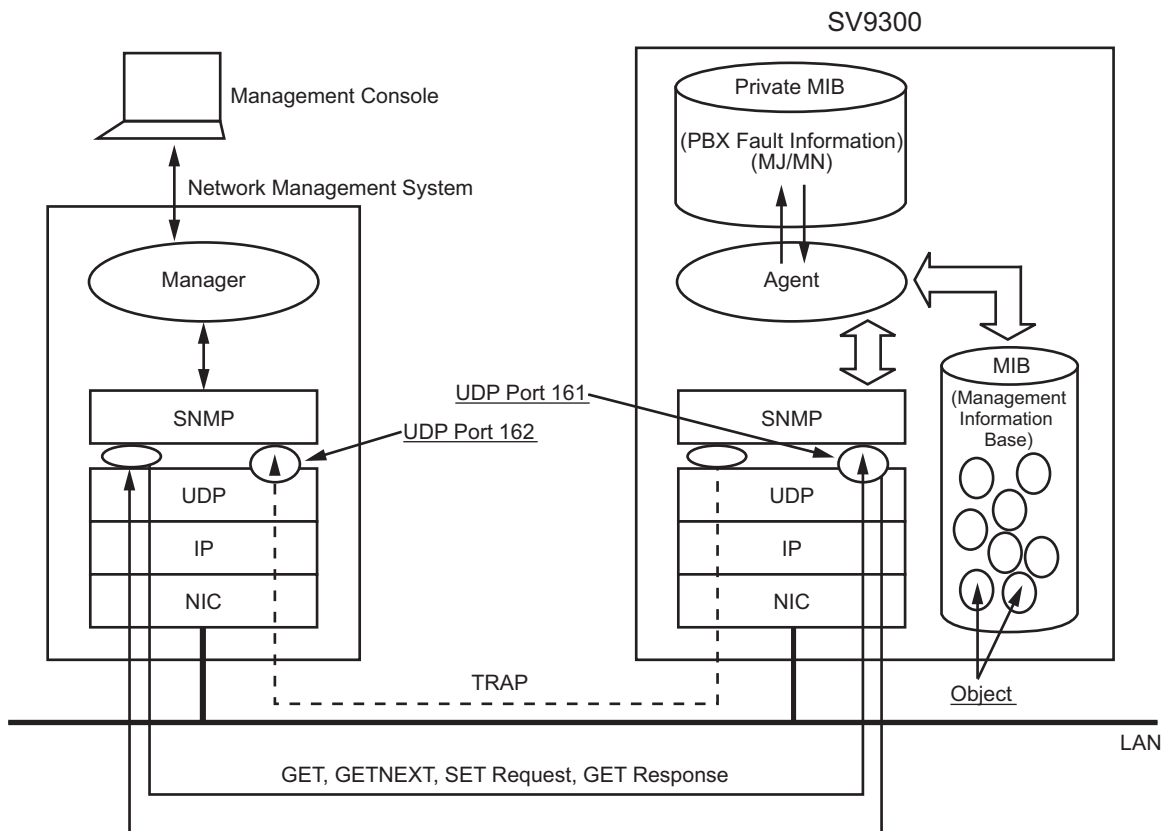
SNMP

General Description

Simple Network Management Protocol (SNMP) is a standard protocol for TCP/IP network management, which enables a network management application software to query a management agent (network device such as router, PC host, and hub) using a supported MIB (Management Information Base). The MIB is a database of network performance information that is stored on the network devices. The SV9300 can support the SNMP standard MIB (MIB-II, defined in IETF RFC 1213), SV9300-specific private MIB, and TRAP.

The SV9300 is also possible to correspond to the SNMP via the NAT.

SNMP System Configuration



Station Application

Not Applicable

Service Conditions

1. The SV9300 supports SNMP V1 and SNMP V2C.
2. The SV9300 supports the following standard MIB (standard MIB-II defined in IETF RFC 1213).
 - System Group
 - Interfaces Group
 - IP Group
 - ICMP Group
 - TCP Group
 - UDP Group
 - SNMP Group
3. There are following conditions to access the standard MIB information from the SNMP manager.
 - a) When “public” is assigned as a community name, the SNMP manager can always access all MIB information as “read-only”.
 - b) When “admin” is assigned as the community name, the SNMP manager can access the MIB information only when the PBX system data is assigned to allow using “admin”. In this case, the SNMP manager can read all MIB information, and write the following MIB information.
 - sysContact, sysName, sysLocation
 - c) When the community name uses a string with any character, the SNMP manager can read all MIB information.
4. Private MIB Extensions and Trap command are supported. The SV9300 provides alarm (MJ/MN) and fault information of the system as the MIB information and automatically sends the fault information to the SNMP manager by Trap command.
5. To set the IP address of Trap source by system data, the IP address to which the trap occurred on a SNMP manager can be managed and notified even via Network Address Translation (NAT).
6. Maximum four IP addresses of SNMP managers can be set in system data programming, to restrict access from any other addresses. The number of accessible SNMP managers can be expanded in system data programming. By masking one IP address with Subnet Mask, access to the system is allowed to all managers in the same network.
7. The system data setting related to SNMP is required only for Maintenance Port assigned to the Unit01. Therefore, standard MIB, private MIB and TRAP information can be obtained both from Maintenance Port and VOIP Port assigned to the Unit01. Furthermore, when setting up a SNMP manager to the network which is different from SV9300, the port that the Default Gateway address is assigned by CM0B Y=001>02/CM0B Y=101>02 is used. For 9300V4 software or later, the other port can be used when the routing table is assigned by CM0D Y=00.
8. This service can be operated on UDP/IP.
9. The UDP port number of the SV9300 for SNMP is 161 (fixed/standard).

10. The standard MIB objects assigned in the system group are as follows:

MIB object	Description	Value
sysDescr	System information	Setting from PCPro is available
sysObjectID	System object ID	Fixed as "1.3.6.1.4.1.119.1.76.3"
sysContact	Contact person for the system with information on how to contact this person	Assigned by ASCII characters (0-255 characters) from PCPro/SNMP manager.
sysName	System name administratively-assigned name for the system	Assigned by ASCII characters (0-255 characters) from PCPro/SNMP manager.
sysLocation	Physical location of the system	Assigned by ASCII characters (0-255 characters) from PCPro/SNMP manager.

NOTE 1: *Maximum of 64 characters can be assigned for each of standard MIB object "sysContact", "sysName", "sysLocation" from the PCPro. The standard MIB objects can be overwritten from the SNMP manager with maximum of 255 characters after they are set by PCPro.*

Be sure to execute the system data backup after the standard MIB objects are changed.

NOTE 2: *To register the standard MIB objects to the SV9300 from the SNMP manager, be sure to set the community name "admin" (CM0B Y=001>101) or optional community name (CM0B Y=001>110) beforehand.*

NOTE 3: *The above standard MIB objects are stored in the CPU as system data. Therefore, when the SNMP manager changes the MIB information after saving the system data by PCPro, verify error will occur between the saved system data and the system data in the CPU. There is no problem for the operation of SV9300 even if the verify error occurs.*

11. Private MIB objects (MJ/MN Lamp Read/Clear)

MJ/MN lamp of the SV9300 can be read or cleared by private MIB objects. The table below shows the notified information and each MIB object/ID.

SNMP Command Name	MIB Object/ID	Notified Information	Access to Object
GET/SET	IpsLampStatusClear/ 1.3.6.1.4.1.119.2.3.76.3.1.1	PBX Lamp status 1: MJ/MN Lamp Off 2: Either MJ or MN Lamp On	Read/Clear
GET	IpsMajorLampStatus/ 1.3.6.1.4.1.119.2.3.76.3.1.2	PBX MJ Lamp status 1: MJ Lamp Off 2: MJ Lamp On	Read only
GET	IpsMinorLampStatus/ 1.3.6.1.4.1.119.2.3.76.3.1.3	PBX MN Lamp status 1: MN Lamp Off 2: MN Lamp On	Read only

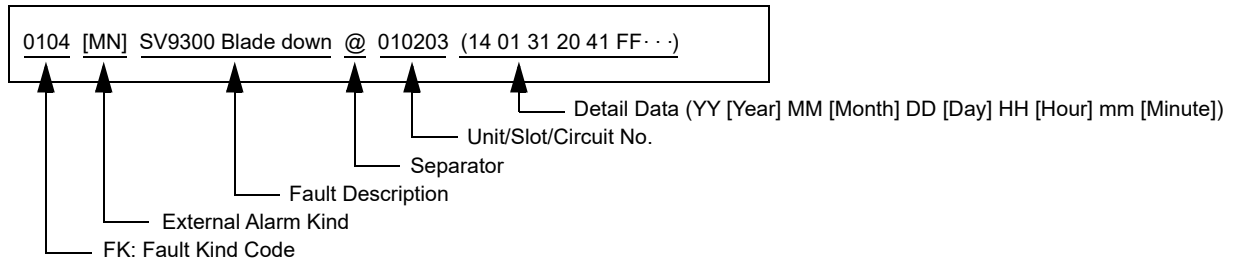
12. TRAP (Fault Message Notification)

SNMP-Trap automatically notifies the details of the fault message in a SNMP V1 format to the SNMP manager. The table below shows the notified fault message and each MIB object/ID.

SNMP Command Name	MIB Object/ID	Notified Information	Access to Object
–	IpsSystemMessageData/ 1.3.6.1.4.1.119.2.3.76.3.2.1	Contents of fault message (Trap) Fault message notification (Trap) Stored up to 255 bytes of alphanumeric characters (ASCII code) (See condition 13.) - FK: Fault Kind Code (4 characters) - External Alarm Kind: 0/1/2=[-]/[MN]/[MJ] - Fault Description (Max.197characters) - Unit/Slot/Circuit No. (5 characters) - Detail Data (43 characters)	Not accessible

13. SNMP-Trap Message Format

Fault message format of SNMP-Trap is as follows.



NOTE: The fault kind notified by SNMP-Trap is the same as the range of fault kind notified to PCPro. The fault messages in the table below (the list of “Fault Kind Code” and “Fault Description”) are not notified for default. To notify the fault messages, the registration of the fault message memory and the external alarm (set by CMEA Y=2) is required.

The table below shows the list of “Fault Kind Code” and “Fault Description”. For details, refer to Fault Display in PCPro.

Fault Kind Code	Fault Description
001	System reset
002	Serious failure B [9300V6 software required]
016	It is a day for periodic maintenance
017	Activation Code error occurred during the License Activation (Center Activation)
021	ISDN D-channel link connection failure
022	CCT link connection failure NOTE 1
	P2P-CCIS link connection failure NOTE 1
025	Number of lockout stations was more than predetermined number
026	Terminal Disconnected
028	SMDR output buffer (SRAM) exceeded 80% [9300V3 software required]
029	SMDR output buffer (SRAM) overflow [9300V3 software required]
02C	LAN application fault occurred
031	ISDN D-channel link connection returned to normal condition
032	CCT link connection returned to normal condition NOTE 2
	P2P-CCIS link connection returned to normal condition NOTE 2

Continued on next page

Fault Kind Code	Fault Description
035	Number of lockout stations restored to less than predetermined number
036	Terminal Connected
038	SMDR output buffer (SRAM) return to normal condition from 80 % condition [9300V3 software required]
039	SMDR output buffer (SRAM) return to normal condition from overflow [9300V3 software required]
03C	LAN application returned to normal condition
040	Traffic of IP network exceeded limit bandwidth
041	Traffic of IP network exceeded warning bandwidth
042	Communication error occurrence between Main Unit and Remote Unit
043	SIP trunk failure
048	Remote System Upgrade
04A	Long call duration
050	Traffic of IP network returned to normal condition from limit bandwidth excess
051	Traffic of IP network returned to normal condition from warning bandwidth excess
052	Communication error restoration between Main Unit and Remote Unit
053	SIP trunk returned to normal condition
061	VoIPDB Startup Process is failure
062	Firmware version un-matching between CPU blade and VoIPDB
064	User operation failure of User Web Portal
065	Web server failure
066	Malicious Call List Overflow [9300V3 software required]
100	Power failure
101	Power failure returned to normal condition
102	DTI line failure
103	DTI line returned to normal condition
104	Blade down
105	Blade returned to normal condition
106	Blade reset with CME03
107	Lack of option value
108	Lack of Highway Channel
109	Automatic Blade reset due to blade lockup

Continued on next page

Fault Kind Code	Fault Description
10A	CPU SRAM failure [9300V3 software required]
10B	Restricted blade [9300V3 software required]
10C	Firmware version un-matching between CPU blade and Line/Trunk blade [9300V3 software required]
10D	Lack of Highway Channel for data communication
10E	Illegal use of CPU blade for Remote Unit only
110	Communication error occurrence between active CPU and stand by CPU
111	Communication error restoration between active CPU and stand by CPU
112	Expansion Chassis (2U) failure
113	Expansion Chassis (2U) failure returned to normal condition
114	Automatic module reset due to Expansion Chassis (2U) lockup
115	Automatic system reset due to Expansion Chassis (2U) lockup
116	All blades lockup in Expansion Chassis (2U)
119	Remote Unit failure
11A	Communication error occurrence between Primary Unit and Secondary Unit
11B	Communication error restoration between Primary Unit and Secondary Unit
11C	Internal BUS failure on CPU Blade
11D	Power ON/OFF by Multiline Terminal Power Saving
11E	Restriction of Remote Maintenance via built-in modem
11F	Automatic system reset due to System Failure [9300V3 software required]
124	VoIPDB failure [9300V3 software required]
125	VoIPDB notification [9300V3 software required]
127	CPU failure [9300V6 software required]
12B	Standard SIP Terminal Disconnected [9300V3 STEP2 software required]
12C	Standard SIP Terminal Connected [9300V3 STEP2 software required]
12D	LAN Cable Disconnected [9300V3 STEP2 software required]

Continued on next page

Fault Kind Code	Fault Description
12E	LAN Cable Connected [9300V3 STEP2 software required]
130	System Resource failure [9300V3 software required]
132	IP Network failure [9300V6 software required]
133	IP Network failure returned to normal condition [9300V6 software required]
134	Login lock by Brute-force Login Attempt [9300V7 software required]
135	Login lock returned to normal condition [9300V7 software required]
136	Scam Call detected [9300V7 software required]
137	Scam Call detection returned to normal condition [9300V7 software required]

NOTE 1: *When the link connection failure to CCT occurs, the fault message “CCT link connection failure” is displayed. When the link connection failure to P2P-CCIS occurs, the fault message “P2P-CCIS link connection failure” is displayed.*

NOTE 2: *When the link connection failure to CCT is restored, the restoration message “CCT link connection returned to normal condition” is displayed. When the link connection failure to P2P-CCIS is restored, the restoration message “P2P-CCIS link connection returned to normal condition” is displayed.*

14. In following conditions, fault messages output by Trap might disappear. This is because SNMP is operated on UDP and communication is made without establishing the connection. Besides, since UDP does not resend messages, abandoned data cannot be retrieved again.
 - SNMP manager is not started
 - Network traffic is congested
 - Fault occurs on the line between PBX and SNMP manager
15. When entering data with characters, the following characters can be registered; Alphabet upper case (A-Z), alphabet lower case (a-z), numeric (0-9), symbol (! “ # \$ % & ’ () * + , ; < = > ? @ [] ^ _ ‘ { } ~), Space, hyphen (-), period (.), slash (/), colon (:)

NOTE: *The character “CCC” can not be registered.*

Programming

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM0B</div> <div style="border-left: 1px solid black; border-right: 1px solid black; border-bottom: 1px solid black; width: 100%; height: 400px; margin-top: 10px;"></div> <div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto; text-align: center;">A</div>	<p>To register the MIB system information from SNMP manager to the system, open the SNMP port of the system.</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 100 (2) 0 : Open SNMP port 1◀: Not open SNMP port
	<p>NOTE: <i>To avoid the unauthorized access to the network, keep the 2nd data to 1 except when registering the MIB system information.</i></p> <p>Allow the use of community name “admin”.</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 101 (2) 0 : Allow (admin) 1◀: Restrict (public)
	<p>NOTE: <i>Restrict the use of community name “admin” (set the community name to “public”) except the system construction.</i></p> <p>Set the community name with 1-25 characters.</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 110: 1-25 characters (2) A-Z, 0-9 : Character (Maximum 25 characters) NONE◀: No data

A	DESCRIPTION	DATA
CM0B	Assign the required IP Address for the SNMP manager (first place to fourth place). RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 124: First place 125: Second place 126: Third place 127: Fourth place (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address for the SNMP manager (Maximum 15 digits) NONE◀: No data NOTE 2
	NOTE 1: <i>The second data must be entered including the periods (.).</i>	
	NOTE 2: <i>The system allows the access only from the IP address that is set by this command. But if no IP address for the SNMP manager (first place to fourth place) are set, the access to the system is allowed to all SNMP managers. To avoid the unauthorized access to the system, this data setting is required.</i>	
	To check the IP address of SNMP manager from the network address, define the IP address of SNMP manager (fourth place) as the Subnet mask of the IP address for the SNMP manager (first place). RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 103 (2) 0 : Subnet Mask of the IP address for the SNMP manager (first place) 1◀: IP address for the SNMP manager (fourth place)
	Set the system information (sysDescr) with 1-128 characters. RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 111: 1-128 characters (2) A-Z, 0-9 : Character (Maximum 128 characters) NONE◀: No data
Set the system contact (sysContact) with 1-64 characters. RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 112: 1-64 characters (2) A-Z, 0-9 : Character (Maximum 64 characters) NONE◀: No data 	
B		

B	DESCRIPTION	DATA
CM0B	Set the system name (sysName) with 1-64 characters.	<ul style="list-style-type: none"> Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 113: 1-64 characters (2) A-Z, 0-9 : Character (Maximum 64 characters) NONE◀: No data
	RESET	
	Set the location of system (sysLocation) with 1-64 characters.	<ul style="list-style-type: none"> Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 114: 1-64 characters (2) A-Z, 0-9 : Character (Maximum 64 characters) NONE◀: No data
	RESET	
	Specify whether the trap is sent to the SNMP manager.	<ul style="list-style-type: none"> Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 102 (2) 0 : To send 1◀: Not sent
	RESET	
Specify the kind of trap message (Specific, object ID) sent to SNMP manager.	<ul style="list-style-type: none"> Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 104 (2) 0 : Variable Trap message by external alarm kind 1◀: Fixed Trap message 	
RESET		
[Australia Only]		
Assign the required IP Address for the destination of trap (first place to fourth place).	<ul style="list-style-type: none"> Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 120: First place 121: Second place 122: Third place 123: Fourth place (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address for the destination of trap (Maximum 15 digits) NONE◀: No data 	
RESET		
NOTE: <i>The second data must be entered including the periods (.).</i>		
C		

C	DESCRIPTION	DATA
CM0B	Assign the community name for the destination of trap (first place) with 1-25 characters. RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 115: 1-25 characters (2) A-Z, 0-9 : Character (Maximum 25 characters) NONE◀: No data
	Assign the community name for the destination of trap (second place) with 1-25 character. RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 116: 1-25 characters (2) A-Z, 0-9 : Character (Maximum 25 characters) NONE◀: No data
	Assign the community name for the destination of trap (third place) with 1-25 characters. RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 117: 1-25 characters (2) A-Z, 0-9 : Character (Maximum 25 characters) NONE◀: No data
	Assign the community name for the destination of trap (fourth place) with 1-25 characters. RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 118: 1-25 characters (2) A-Z, 0-9 : Character (Maximum 25 characters) NONE◀: No data
	Assign the IP Address for the trap source when using the NAT. RESET	<ul style="list-style-type: none"> • Y=001 (Maintenance Port [0] + Unit No. [01]) (1) 129 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address for the trap source (Maximum 15 digits) NONE◀: No data

NOTE 1: The second data must be entered including the periods (.).

NOTE 2: The IP address assigned by this data is set to the Agent address in “SNMP TRAP PDU”, and the system sends the IP address to the IP network. Wherever the system is located on the LAN, system administrator can manage it easily by setting of the convenient IP address.

END

Operating Procedure

To make available the SNMP for SV9300, register the following SNMP data to the SV9300 using the PC-Pro or SNMP manager.

Once SNMP data is registered, the operation of SV9300 is not required.

Also, a part of MIB system information can be changed from the SNMP manager.

(1) PCPro Operation

STEP1: Open the SNMP port of the system by CM0B Y=001>100.

STEP2: Set the community name (for system password) by CM0B Y=001>110.

There are three community names as follows.

“public”: SNMP manager can read All MIB information

“admin”: SNMP manager can read All MIB information, and can write a part of MIB information.

“Optional community name”: SNMP manager can read/write the MIB information such as a general SNMP

Usually, set the “Optional community name” by CM0B Y=001>110.

Community name “public” is always available.

To avoid the unauthorized access to the network, use the community name “admin” only for the system construction (CM0B Y=001>101).

STEP3: Set the IP address for SNMP manager by CM0B Y=001>124-127.

STEP4: Set the system information (sysDescr) by CM0B Y=001>111.

STEP5: Set the system contact (sysContact) by CM0B Y=001>112.

STEP6: Set the system name (sysName) by CM0B Y=001>113.

STEP7: Set the system location (sysLocation) by CM0B Y=001>114.

STEP8: Specify whether the trap is sent or not by CM0B Y=001>102.

STEP9: Set the IP address for the destination of trap by CM0B Y=001>120-123.

STEP10: Set the community name for the destination of trap by CM0B Y=001>115-118.

STEP11: When using the NAT, set the IP address for the trap source by CM0B Y=001>129.

STEP12: Execute the system data backup by CMEC Y=6>0: 0.

STEP13: Press the RESET switch of CPU blade.

(2) SNMP Manager Operation

The following MIB system information can be changed from the SNMP manager. In addition, when the following information is set from the PCPro, SNMP manager can overwrite them.

- System Contact (sysContact)
- System Name (sysName)
- System Location (sysLocation)

NOTE: *When the above information is overwritten by the SNMP manager, be sure to execute the system data backup from the PCPro.*

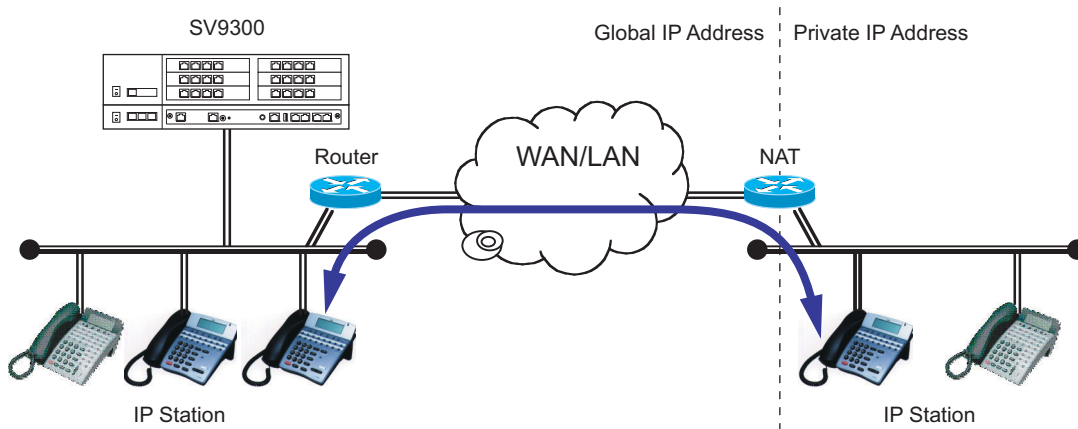
Terminal Login via NAT

General Description

This feature allows the system to transmit Network Address Translation (NAT) over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN).

The following figure shows a typical configuration of NAT use on IP network.

Example of NAT Use on IP Network



Station Application

Not Applicable

Service Conditions

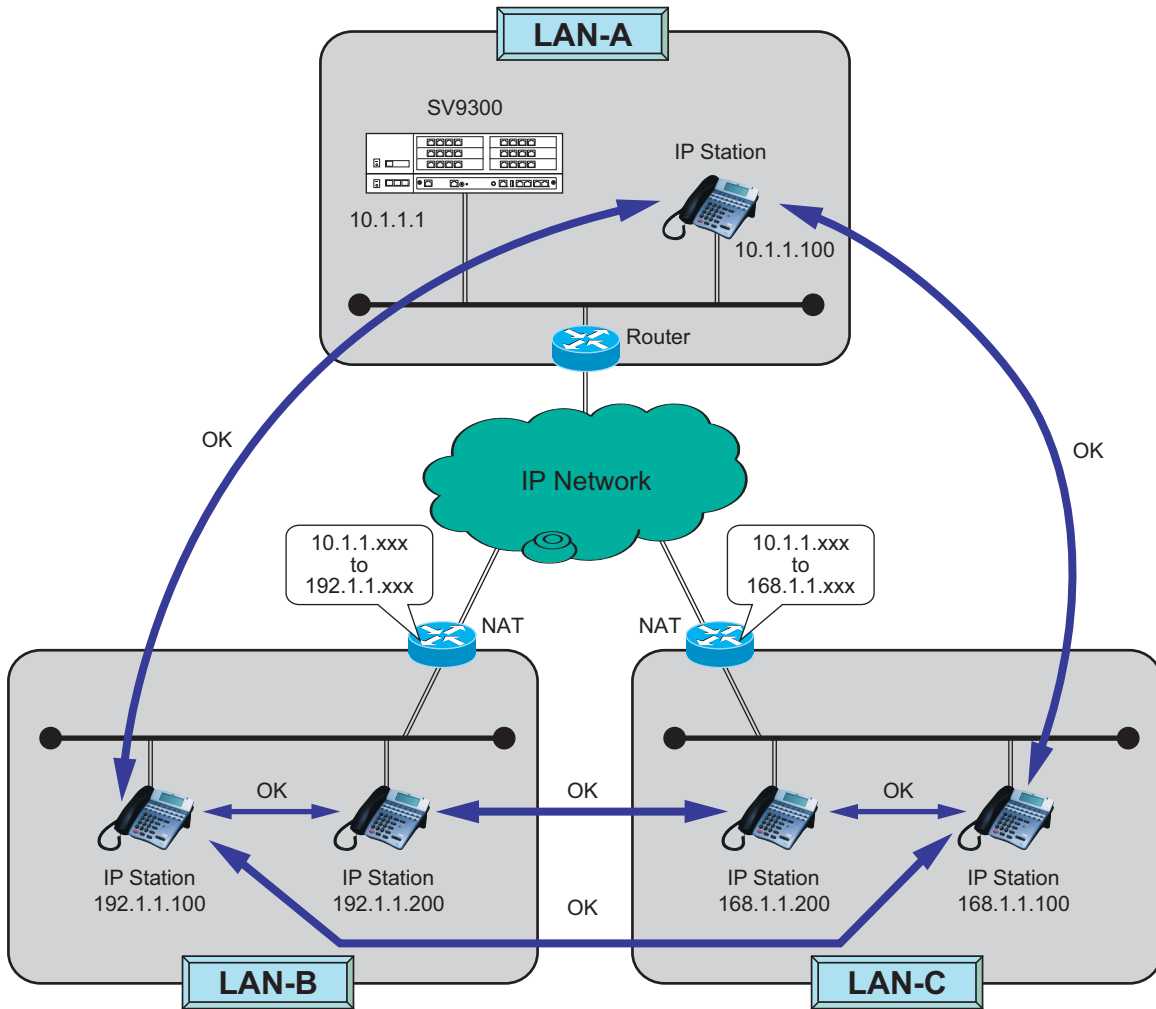
Conditions on Overall System

1. As an address translation method, Basic NAT is used. NATP (IP masquerade) is not supported.
2. Global IP address is assigned to an IP Station/VoIPDB placed in stand-alone system or Main Unit for Remote UNIT over IP when NAT is performed. Only single Private IP address can be translated to single Global IP address. Private IP address can be assigned to IP Stations under the NAT only.
3. Global IP address that is assigned to IP Station does not change when the terminal has logged in.
4. Equipment that supports NAT is as follows. (all of the equipment that is related to communications and supporting NAT are required. If there is equipment that does not support NAT, one-way speech or no tone might occur.)

EQUIPMENT		Available : × Not available: -
Terminals	D ^{term} 85 (Series i) (IP Adapter Type)	×
	D ^{term} 85 (Series i) (IP Bundled Type)	×
	D ^{term} IP INASET (ITR-240G-1)	-
	Soft Phone (D ^{term} SP30)	×
	Soft Phone (SP350)	-
	DT700/DT800/DT900 Series	-
	D ^{term} IP INASET (ITR-LC-1)	-
Daughter Board	VoIPDB	×

Conditions on IP Station

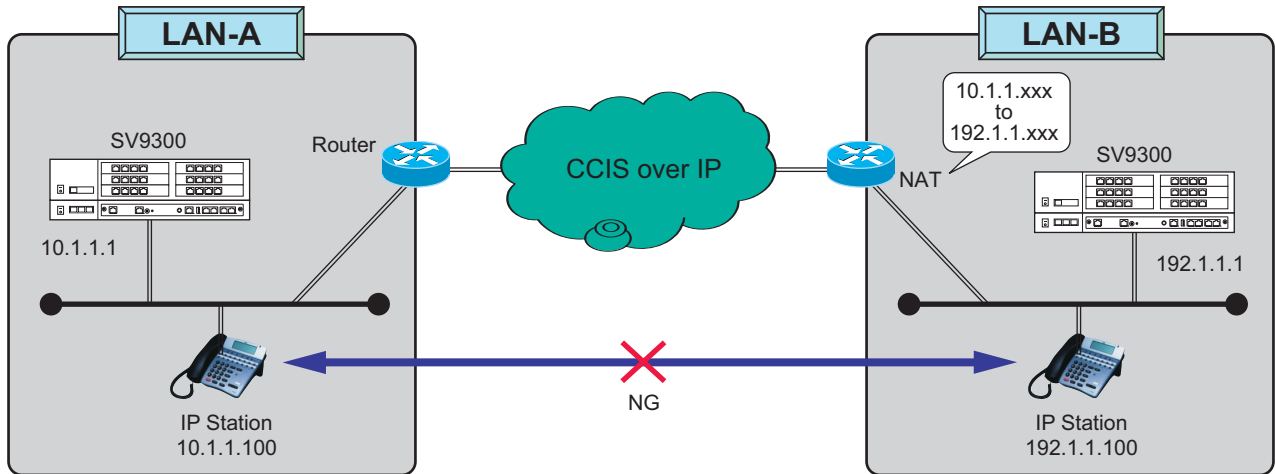
1. IP Station can be installed at remote location from SV9300 via NAT.
 - IP Station can communicate each other under the same NAT.
 - IP Station can communicate each other under different NATs.
 - IP Station cannot be connected via multiple NATs.



2. IP Stations used between networks via NAT can be set different location numbers since it is necessary to specify the connection status of NAT between locations in system data programming (CM67 Y=08).
3. Be sure to set NAT, as the assigned IP address at login of IP Station is not changed before logout.

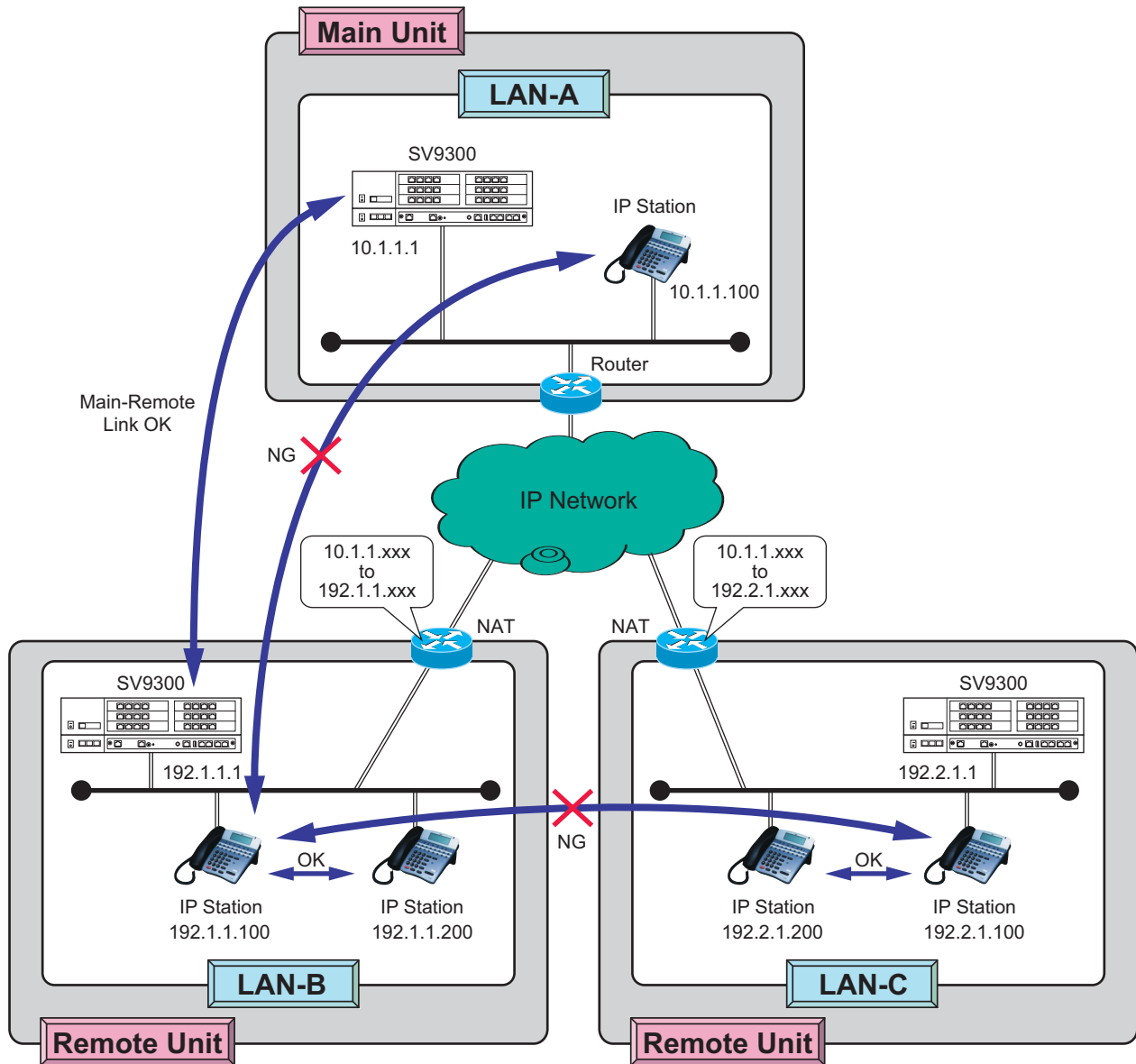
Conditions on CCIS Connection (Peer to Peer Connection)

1. Communication in CCIS connection via NAT is not available.
 - IP Stations accommodated in the same office can communicate each other.
 - IP Stations accommodated in the different offices via NAT cannot communicate each other.



Conditions on Remote UNIT over IP

1. Remote Unit can operate in normal remote mode, which is subordinate operation to the Main Unit, when NAT is used between the Main Unit and Remote Unit.
 - IP Stations accommodated in the same unit can communicate each other. This condition is applied to normal remote operation and survival operation.
 - IP Stations accommodated in the different units via NAT cannot communicate each other.



2. Be sure to set NAT, as the assigned IP address for the unit is not changed during normal operation mode.

Conditions on VoIPDB

- When the VoIPDB is connected via NAT, a Global IP address is assigned to the VoIPDB of Remote Unit for Remote UNIT over IP by system data (CM0B Y=1XX>70). Therefore, in NAT setting, the IP address of VoIPDB should be assigned statically.

Programming

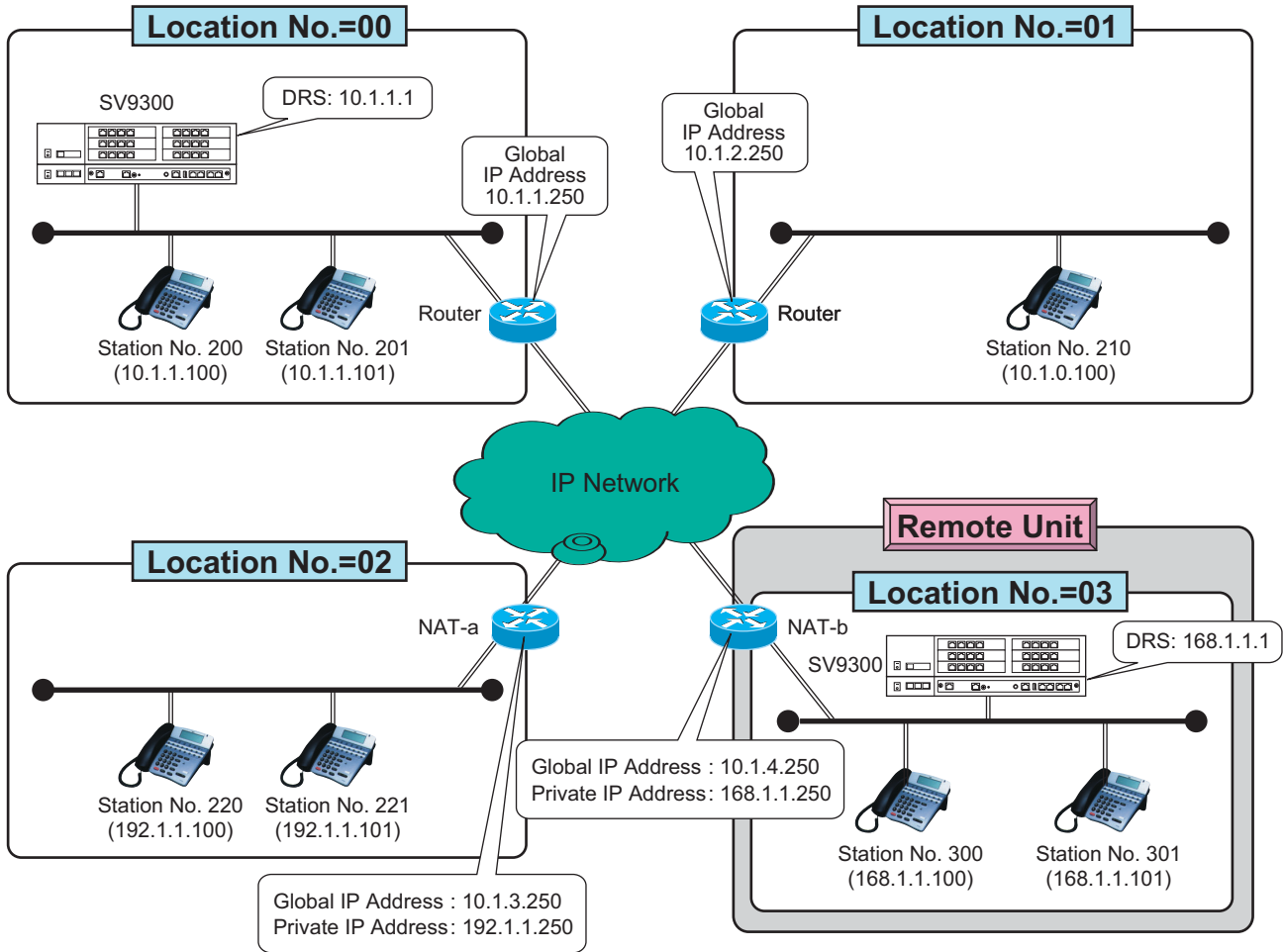
START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM67</div>	<p>Specify whether the IP Station at Remote Unit location from IPS via NAT can communicate each other under the same NAT or not.</p> <p>NOTE: <i>The same location number must not be assigned to IP Station that is not accommodated under the same NAT. However, multiple location numbers can be assigned to IP Station that is accommodated under the same NAT.</i></p>	<ul style="list-style-type: none"> • Y=07 (1) XX ZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) 0 : Under the same NAT 1◀: Under the different NAT or Not used NAT
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM0B</div>	<p>Specify whether the connection between locations is restricted or not.</p> <p>NOTE: <i>Restrict the following connection by this command when NAT is used.</i></p> <ul style="list-style-type: none"> - Connection via CCIS (Peer to Peer connection) - Connection between Main Unit and Remote Units - Connection between locations that are restricted the communications 	<ul style="list-style-type: none"> • Y=08 (1) XX ZZ XX: 00-63: Location No. ZZ : 00-63: Location No. (2) 0 : To restrict 1◀: Not restricted
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>	<p>Assign the Global IP address for VoIPDB of Remote Unit of Remote UNIT over IP when VoIPDB is controlled by NAT.</p> <div style="text-align: right; margin-right: 50px;"> <div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div> </div> <p>NOTE: <i>The second data must be entered including the periods (.).</i></p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 70 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: Global IP Address for VoIPDB (Maximum 15 digits) NONE◀: No data

Example of Programming for Terminal Login via NAT

The following shows example of data programming for Terminal Login via NAT.

– Network configuration

Network Configuration



– Programming of Terminal Login via NAT

Programming of Terminal Login via NAT

COMMAND	Y	FIRST DATA	SECOND DATA
67	07	0202	0
67	07	0303	0
67	08	0003	0
67	08	0300	0
67	08	0103	0
67	08	0301	0
67	08	0203	0
67	08	0302	0

– Programming of IP Station

Programming of IP Station

STATION NUMBER	IP ADDRESS	DRS	GATEWAY ADDRESS
200	10.1.1.100 (Global IP Address)	10.1.1.1	10.1.1.250 (Global IP Address)
201	10.1.1.101 (Global IP Address)	10.1.1.1	10.1.1.250 (Global IP Address)
210	10.1.2.100 (Global IP Address)	10.1.1.1	10.1.2.250 (Global IP Address)
220	192.1.1.100 (Private IP Address)	10.1.1.1	192.1.1.250 (Private IP Address)
221	192.1.1.101 (Private IP Address)	10.1.1.1	192.1.1.250 (Private IP Address)
300	168.1.1.100 (Private IP Address)	168.1.1.1	192.1.1.250 (Private IP Address)
301	168.1.1.101 (Private IP Address)	168.1.1.1	192.1.1.250 (Private IP Address)

– Setting of Router (NAT-a [Cisco router])

- > ip nat pool POOL1 10.1.3.1 10.1.3.15 netmask 255.255.255.240
[Register the range of Global IP address 10.1.3.1-10.1.3.15 for NAT as POOL1.]
* Set the number of Global IP address to exceed the number of IP Station.
- > access-list 1 permit any
[Register all packets of private network as list 1.]
- > ip nat inside source list 1 pool POOL1
[Find the matched packets to list 1 in private network and translate within range of POOL1.]
- > interface Ethernet0
[Port of LAN side]
- >> ip nat inside
[Designate private network of NAT]
- > interface Ethernet1
[Port of WAN side]
- >> ip nat outside
[Designate global network of NAT]

Communication Availability

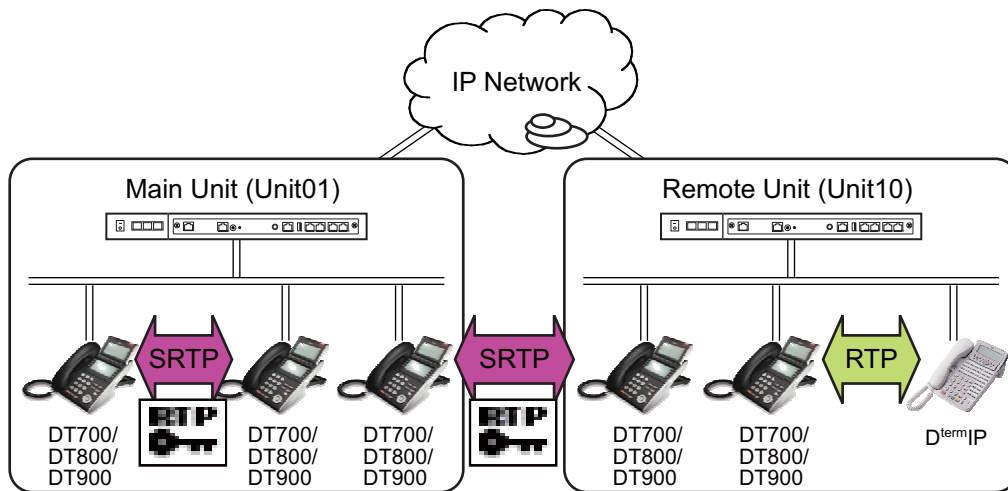
STATION NUMBER	Available : × Not available: -
200-201	×
200-210	×
200-220	×
200-300	-
210-220	×
220-221	×
220-300	-
300-301	×

VoIP Encryption

General Description

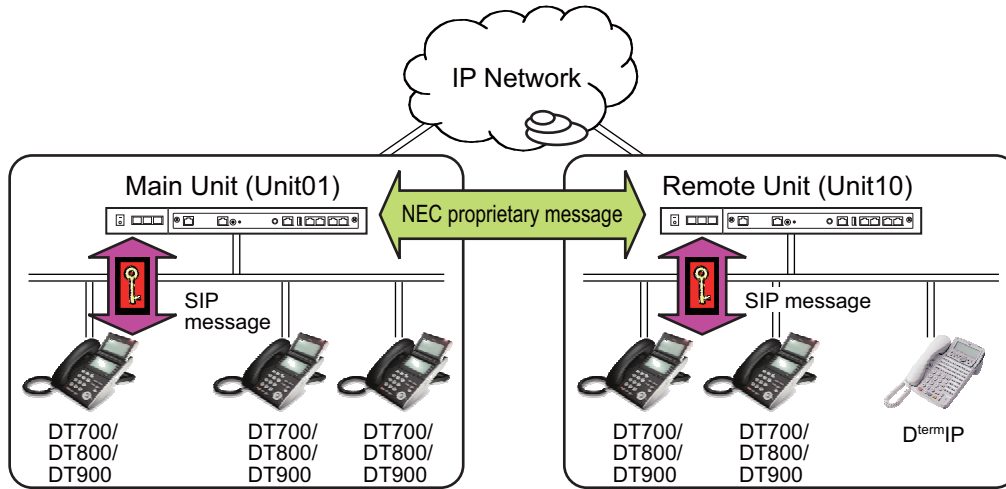
In the voice calls and the communications with DT700/DT800/DT900 Series, this feature provides a secure communications by enabling the following encryptions.

- Voice encryption (SRTP) for Station-to-Station call between DT700/DT800/DT900 Series:
Encrypts the voice data (SRTP) for Station-to-Station call between DT700/DT800/DT900 Series within a same Unit or between multi Units.



NOTE: When using this feature while a call is in progress, the icon indicating an encryption is displayed on the terminal LCD until the call ends.

- Signaling message (SIP message) encryption for DT700/DT800/DT900 Series:
Encrypts the signaling messages (SIP messages) between DT700/DT800/DT900 Series and SV9300.



NOTE 1: Since the SIP messages are not sent/received between Main and Remote Units (NEC proprietary messages are used), the signaling messages are not encrypted.

NOTE 2: SIP messages are encrypted even if the Remote Unit is in survival mode.

Service Conditions

General Conditions

1. To use this feature, the following licenses are required.
 - System Version License
2. This feature is supported only for DT700/DT800/DT900 Series only.
 - Encryption of control packets: DT700/DT800/DT900 Series only
 - Encryption of voice packets: Station-to-Station call between the DT700/DT800/DT900 Series only

NOTE: For the equipment that mirror voice packets and record the telephone conversation (such as @log), the conversation cannot be replayed because the recorded voice packets are mirrored with encryption.

3. The license check will be performed on the terminal start up. “Exceed Capability” is displayed on the terminal LCD if there is not enough license.

4. Up to 1000 encryption enabled DT700/DT800/DT900 Series terminals can be accommodated in a system (up to 500 terminals are available for simultaneous call). “Exceed Capability” is displayed on the terminal LCD as in the case of General Conditions (3) when more than 1000 DT700/DT800/DT900 Series terminals are accommodated.
5. The following encryption methods are supported.
 - Control packets (SIP packets): AES-ECB 128bit
 - Voice packets (RTP packets): AES-CTR 128bit
6. For calls between DT700/DT800/DT900 Series with encryption enabled and one without encryption enabled, voice packets are not encrypted.
7. To improve security, it is recommended to set the second data of CM15 Y=486 to “0” (encrypt both control packets and voice packets).
8. For peg count of encrypted calls, the number of times of encrypted calls connected is counted. However, when the call is placed on hold and released, it is counted as two calls, that is, a call before Hold and a call after Hold Release.
9. To improve security, it is recommended to clear the One Time Password (OTP) manually after the terminal set up ends (an initial or add-on installation). The One Time Password (OTP) must be set again if cleared.
10. When the encryption feature is not set on the terminal, or the firmware version of the terminal is 2.1.1.0 or before, the terminal starts as one without encryption enabled even if the encryption feature is enabled (CM15 Y=486: 0) in system data. Then, “No Encryption” is displayed on PCPro when confirming by CMFA Y=04.
11. When setting the One Time Password (OTP) by the command (CM2B Y=11), it must be input into the terminal by Config menu.
12. Even if you make the encryption feature for DT700/DT800/DT900 Series enable, the voice data between VoIPDB-DT700/DT800/DT900 Series is not encrypted when External Hold Tone source is sent. The voice data will be encrypted again when a call is returned to the two-party call between DT700/DT800/DT900 Series terminals by Hold Release.

Terminal Conditions

1. To provide the encryption feature to DT700 Series, the terminal firmware version 2.1.1.0 or later is required (firmware version 4.0.3.0 or later is recommended).
2. When the encryption setting (CM15 Y=486) is changed, the reset of the terminal is required.
3. To enable the encryption feature, set the Authentication Mode to “Enable” on DT700/DT800/DT900 Series terminal setting.
4. The encryption status of DT700/DT800/DT900 Series can be confirmed by the following LCD display.
 - (a) When confirming by the “Authentication Status” on the Config menu of the DT700/DT800/DT900 Series: When the start up of the terminal is normally succeeded, “Acceptable” is displayed on the LCD of the terminal. And when the start up of the terminal is failed, “Not Acceptable” is displayed on the LCD of the terminal.

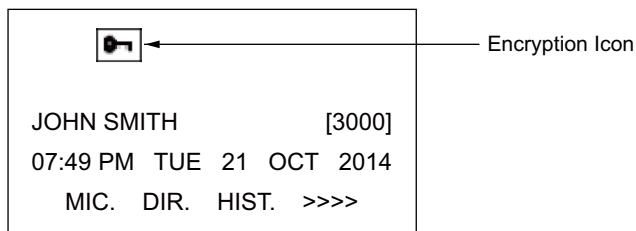
- (b) When confirming the encryption status by the LCD display before DT700/DT800/DT900 Series start up:

The following messages are displayed by the terminal status.

Status	LCD Display	Meaning/Corrective Action
One Time Password setting NG	OTP Not Include	Confirm whether the One Time Password (OTP) is assigned by Config menu.
Authentication NG to terminal	Failed (server)	Confirm whether the One Time Password (OTP) assigned to the terminal matches the one set by CM2B Y=11 again.
SIP messages are not sent	SIP Server Not Found	Confirm whether the network between SV9300 and terminal is not disconnected.
During acquiring T-PASS	Authenticating	Wait for about two minutes for during the terminal start up.
Acquiring T-PASS succeeded	Authentication Accepted	Wait for about two minutes for during the terminal start up.
During registration	Connecting	Wait for about 30 seconds for during the terminal start up.

5. While an encrypted call is in progress, the icon indicating that voice packets are encrypted is displayed on the terminal LCD.

- A display example



NOTE: *The encryption icon is not displayed for DT710.*

Programming

- (1) VoIP encryption setting for DT700/DT800/DT900 Series Station-to-Station call
To encrypt the DT700/DT800/DT900 Series Station-to-Station call, do the following programming.

START	DESCRIPTION	DATA
CM12	Assign Service Restriction Class C to the required stations.	<ul style="list-style-type: none"> • Y=07 (1) X-XXXXXXXX: Station No. (2) 00-15◀: Service Restriction Class C
CM15	Specify the VoIP encryption method for DT700/DT800/DT900 Series in Service Restriction Class C assigned by CM12 Y=07. NOTE: <i>A reset of the terminal is required when this data is set.</i>	<ul style="list-style-type: none"> • Y=486 (1) 00-15: Service Restriction Class C assigned by CM12 Y=07 (2) 0 : Control Packets and Voice Packets 1 : Control Packets only 2 : Voice Packets only 7◀: No Encryption
CM2B	Set the One Time Password (OTP) for VoIP encryption. NOTE 1: <i>Set the password assigned in this data to One Time Password (OTP) for the terminal.</i> NOTE 2: <i>It is recommended to clear this data after the terminal set up ends. When setting the password again after clearing this data, assign this data again.</i>	<ul style="list-style-type: none"> • Y=11 (1) 00 (2) XXXX-XXXXXXXXXXXX: One Time Password (OTP) (4-10 digits) X : 0-9, A (*), B (#)
END		

(2) DT700/DT800/DT900 Series encryption status confirmation

To confirm the encryption status on DT700/DT800/DT900 Series, do the following programming.

START	DESCRIPTION	DATA
<p>START</p> <div style="border: 1px solid black; width: 60px; height: 30px; margin: 0 auto; text-align: center; line-height: 30px;">CMFA</div> <p>END</p>	<p>Read the VoIP Encryption status on DT700/DT800/DT900 Series (Only display).</p>	<ul style="list-style-type: none"> • Y=04 (1) X-XXXXXXXX: Station No. (2) 0 : Encryption <li style="padding-left: 20px;">1◀: No Encryption

Operating Procedure

Set the VoIP encryption according to the following procedures.

1. Assign the system data for VoIP encryption to SV9300 (refer to “Programming” in this feature).
[📄 Page 2-252](#)
2. Set the encryption setting to “Enable” in the Config menu of DT700/DT800/DT900 Series.
For details of the encryption setting, refer to the “Config Menu Items”. [📄 Page 2-106, Page 2-111](#)
3. Confirm the terminal status from PCPro or DT700/DT800/DT900 Series.

NOTE: *You cannot confirm the encryption status from DT700 Series without LCD (ITL-2E-1(BK) TEL).*

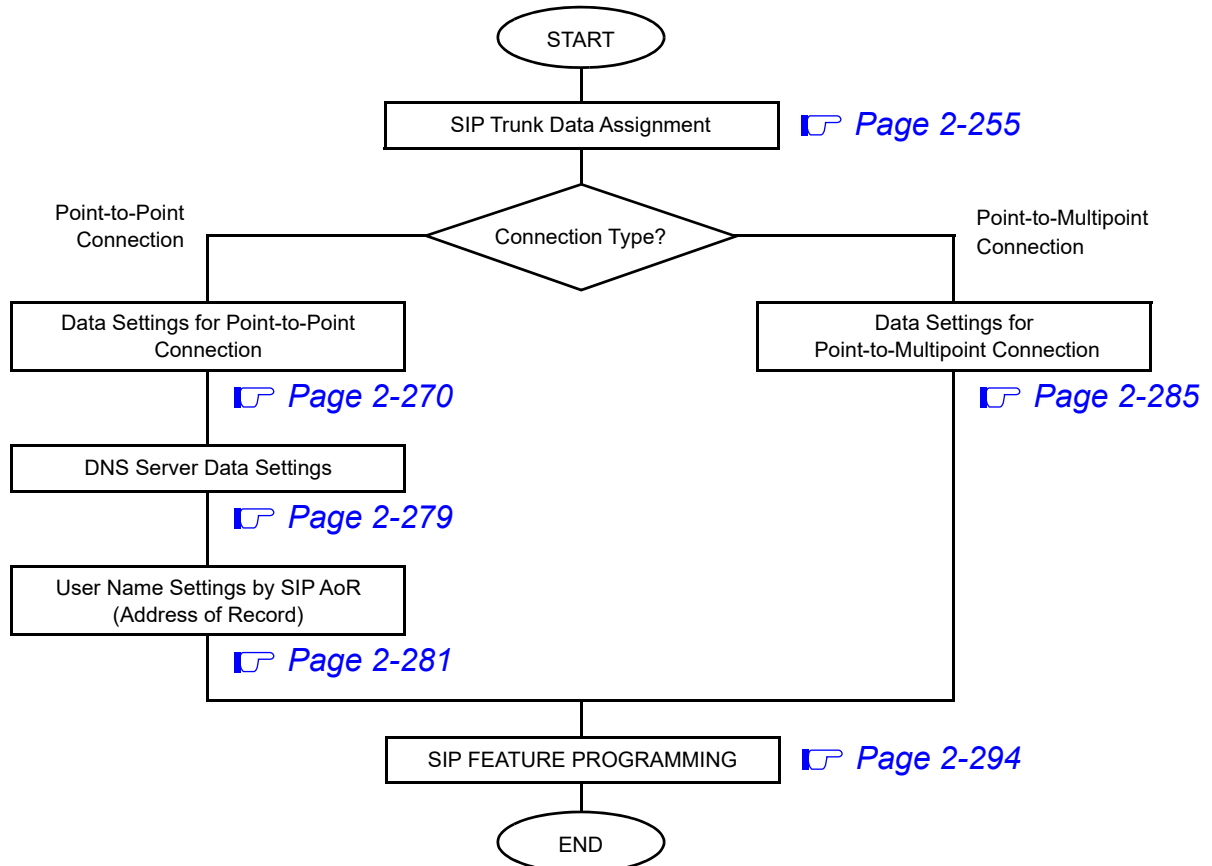
4. Delete the One Time Password (OTP) on SV9300 by the system data (CM2B Y=11) after start up of the all terminals.
* The One Time Password (OTP) on the terminal will be automatically deleted after start up of the terminal.

SIP TRUNK DATA ASSIGNMENT

Programming Procedure for SIP Trunk

Program the system data according to the following procedure.

Programming Procedure for SIP Trunk



SIP Trunk Data Assignment

This section describes the required system data settings for using SIP trunk.

NOTE: When entering data with characters, the following characters can be registered; Alphabet upper case (A-Z), alphabet lower case (a-z), numeric (0-9), symbol (! " # \$ % & ' () * + , ; < = > ? @ [] ^ _ ' { } ~), Space, hyphen (-), period (.), slash (/), colon (:)

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM10</div>	<p>Assign a trunk number to Virtual Port number of the SIP trunk.</p> <p style="text-align: right;">RESET</p> <p>NOTE 1: In case of Point-to-Point connection, SIP trunk can be accommodated in all units. In case of Point-to-Multipoint connection, SIP trunk can be accommodated in Unit01 only.</p> <p>NOTE 2: For effective use of ports, specify the continuous trunk numbers.</p> <p>NOTE 3: Every port assigned by this command is used as a speech channel.</p> <p>NOTE 4: Up to 127 voice channels (100 channels for 9300V3 software or before) can be used per unit. Up to 512 channels can be used in the whole system.</p>	<ul style="list-style-type: none"> • Y=03 (1) XX ZZZ: Virtual Port No. XX : 01-50 : Unit No. ZZZ: 000-127: Virtual Port No. of the SIP Trunk (2) D000-D511: Trunk No.
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM08</div>	<p>Allow C.O. to C.O. transfer by station or attendant.</p> <p>Provide the Ring Transfer for Call Transfer-All Calls to a SIP when a station holds another station.</p>	<ul style="list-style-type: none"> (1) 028 (2) 0: To allow (1) 253 (2) 0: Available
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">A</div>		

A	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM0B</div>	Assign the IP address for the SIP trunk. <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 00: IP Address for the SIP trunk (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address for the SIP trunk (Maximum 15 digits) NONE◀: No data
	<p>NOTE: <i>The second data must be entered including the periods (.).</i></p>	
	Assign the Subnet Mask for the SIP trunk. <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 01 (2) XXX.XXX.XXX.XXX: 255.0.0.0-255.255.255.252: Subnet Mask for the SIP trunk (Maximum 15 digits) NONE◀: No data
	<p>NOTE: <i>The second data must be entered including the periods (.).</i></p>	
	Assign the Default Gateway Address for the SIP trunk. <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px;">RESET</div>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 02 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: Default Gateway Address for the SIP trunk (Maximum 15 digits) NONE◀: No data
<p>NOTE: <i>The second data must be entered including the periods (.).</i></p>		
Assign the IP address (RTP) for the VoIPDB. <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px;">RESET</div>	<ul style="list-style-type: none"> • Y=2XX (VOIP port [2] + Unit No. [01-50]) (1) 00: IP Address (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address (Maximum 15 digits) NONE◀: No data 	
<p>NOTE: <i>The second data must be entered including the periods (.).</i></p>		
Assign the number of voice channels used for the SIP trunk.	<ul style="list-style-type: none"> • Y=2XX (VOIP Port [2] + Unit No. [01-50]) (1) 10 (2) 16, 32, 48, 64, 80, 96, 112, 128 (Number of voice channels used) NONE◀: Use all channels of the VoIPDB 	
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">B</div>		

B	DESCRIPTION	DATA
CM0B	Assign the RTP Base port number for the SIP trunk. <div style="text-align: center; border: 1px solid black; border-radius: 15px; padding: 2px 10px;">RESET</div>	<ul style="list-style-type: none"> • Y=2XX (VOIP port [2] + Unit No. [01-50]) (1) 40: RTP Base Port for Voice Packet transmitting/receiving (2) 01024-65534: RTP Port No. NONE ◀ : 10000 [10000-10255 (256 ports are used among ports)] <p style="color: red; margin-top: 0;">NOTE 3, NOTE 4, NOTE 5</p>

NOTE 1: Follow the typical default setting.

NOTE 2: Set this data when the router or firewall provides the restriction by TCP port.

NOTE 3: For this data, assign the RTP Base port number of voice packet transmitting/receiving ports (RTP ports) corresponding to the number of voice channels used for VoIPDB (assigned by CM0B Y=2XX>10).

NOTE 4: The RTP port is used at 2-port intervals from the RTP Base port number set by this command for each VoIPDB voice channel as shown in the example below.

Example)

When this data and the number of VoIPDB voice channels are set to default settings:

Channel No.	RTP port	
1	10000	← RTP Base port number
2	10002	
⋮	⋮	
128	10254	

Therefore, assign RTP Base port number so that it will not exceed the last RTP Base port number (65534).

NOTE 5: Be sure that the use range of each port assigned by CM0B Y=1XX>36, 37, 38, 39 does not overlap the setting of the RTP Base port for voice packet transmitting/receiving.

C

C	DESCRIPTION	DATA
CM0B	<p>Assign the UDP port number for the SIP trunk.</p> <p style="text-align: right;">(RESET)</p> <p>NOTE 1: Follow the typical default setting.</p> <p>NOTE 2: Set this data when the router or fire-wall provides the restriction by TCP port.</p> <p>NOTE 3: Be sure that the use range of each port assigned by CM0B Y=1XX>37, 38, 39, Y=2XX>40 does not overlap the setting of the UDP port for SIP control packet.</p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 36: UDP Port for SIP control packet (2) 01024-65534: UDP No. NONE◀ : 5060 [5060 (1port is used)] <p style="text-align: right;">NOTE 3</p>
	<p>Specify SIP trunk source IP address check.</p>	<ul style="list-style-type: none"> • Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 53 (2) 0 : Available 1◀: Not available
CM30	<p>Assign a trunk route number to each trunk number for voice channel of SIP trunk.</p> <p style="text-align: right;">(RESET)</p> <p>Assign a tenant number to each trunk number.</p> <p>NOTE: When the opposite PBX is IPMaster/MG3000/APEX3600i/SV8300, assign 02 or more for tenant number.</p>	<ul style="list-style-type: none"> • Y=00 (1) 000-511: Trunk No. assigned by CM10 (2) 00-63: Trunk Route No. <ul style="list-style-type: none"> • Y=01 (1) 000-511: Trunk No. assigned by CM10 (2) 00, 01◀-63: Tenant No.
CM35	<p>Provide the voice channel route with SIP trunk facilities.</p> <p>Assign the SIP number to each voice channel trunk route.</p>	<ul style="list-style-type: none"> • Y=090 (1) 00-63: Trunk Route No. (2) 0: SIP trunk <ul style="list-style-type: none"> • Y=091 (1) 00-63: Trunk Route No. (2) 00-63 : SIP trunk No. NONE◀: No data
D		

D	DESCRIPTION	DATA
CMA7	Specify the connection type for SIP trunk. <input type="button" value="RESET"/>	<ul style="list-style-type: none"> • Y=46 (1) 00-63: SIP Trunk No. (2) 0 : Point-to-Multipoint 1 ◀: Point-to-Point
CM30	Assign a Circuit Identification Code (CIC) to each trunk number for voice channel of SIP trunk route. <input type="button" value="RESET"/>	<ul style="list-style-type: none"> • Y=35 (1) 000-511: Trunk No. assigned by CM10 (2) 001-127 : CIC NONE ◀: No data
	<p>NOTE 1: Assign this data after setting CM35 Y=090, 091.</p> <p>NOTE 2: For Point-to-Multipoint connection (when CMA7 Y=46 is set to "0"), be sure that the trunk route No. assigned by this data does not overlap the trunk route No. used for IPT (P2P CCIS) within a same office.</p>	
CM35	Assign the trunk route data for voice channel of SIP trunk route.	<ul style="list-style-type: none"> • Y=000 (1) 00-63: Trunk Route No. (2) 00: C.O. line [when using a point-to-point connection with the SIP carrier] 04: Tie Line [when using a point-to-multipoint connection as a SIP-dedicated line]
	Specify the answer signal for voice channel of SIP trunk route.	<ul style="list-style-type: none"> • Y=004 (1) 00-63: Trunk Route No. (2) 2: Answer signal arrives
E		

E	DESCRIPTION	DATA
CM35	Specify the release signal for voice channel of SIP trunk route.	<ul style="list-style-type: none"> • Y=005 (1) 00-63: Trunk Route No. (2) 1◀: Release signal arrives
	Specify the incoming connection signal for voice channel of SIP trunk route.	<ul style="list-style-type: none"> • Y=009 (1) 00-63: Trunk Route No. (2) 03: SIP
	NOTE: <i>For a SIP Trunk incoming connection signaling type specification, be sure to set this data to 03 (Wink Start/CCIS/SIP).</i>	
	Specify the outgoing start condition for voice channel of SIP trunk route.	<ul style="list-style-type: none"> • Y=020 (1) 00-63: Trunk Route No. (2) 00: SIP
	NOTE: <i>For a SIP Trunk sender start condition specification, be sure to set this data to 00 (Wink Start/CCIS/SIP).</i>	
	Specify the Type of Service (TOS) field precedence for SIP trunk voice packet.	<ul style="list-style-type: none"> • Y=134 (1) 00-63: Trunk Route No. (2) 00-07: PRECEDENCE 0-7 15◀: PRECEDENCE 0
<p>NOTE 1: <i>The priority of PRECEDENCE 0-7 is as follows.</i></p> <p style="margin-left: 40px;"><i>PRECEDENCE 0: Lowest priority</i></p> <p style="margin-left: 100px;">}</p> <p style="margin-left: 100px;">}</p> <p style="margin-left: 40px;"><i>PRECEDENCE 7: Highest priority</i></p> <p style="margin-left: 40px;"><i>PRECEDENCE 5 is recommended.</i></p>		
NOTE 2: <i>By this data setting, the router recognizes the precedence of voice packets and controls Weighted Fair Queuing (WFQ).</i>		
NOTE 3: <i>This command assigns QoS for packets that are sent from a unit which accommodates a SIP trunk to another device to which the SIP trunk is connected.</i>		
F		

F	DESCRIPTION	DATA
CM35	<p>When the router provides DiffServ QoS, specify the DS code point for SIP trunk voice packet.</p> <p>NOTE 1: <i>For a setting value for this command, consult with a network manager. If no specific instruction is given from the manager, do not modify this setting.</i></p> <p>NOTE 2: <i>When this data is set, the TOS field precedence set by CM35 Y=134 is ineffective. If you want to validate the precedence set by CM35 Y=134, set "CCC" (data clear) for CM35 Y=161.</i></p> <p>NOTE 3: <i>Assigning this command enables a router to recognize the precedence with Diffserv (Differentiated Service) and to control voice packets according to the precedence.</i></p> <p>NOTE 4: <i>This command assigns QoS for packets that are sent from a unit which accommodates a SIP trunk to another device to which the SIP trunk is connected.</i></p>	<ul style="list-style-type: none"> • Y=161 (1) 00-63: Trunk Route No. (2) 00-3F: DS code point
	<p>Assign the pattern number for adding an access code for outgoing call to the calling number stored by Message Reminder when terminating a tandem call via SIP trunk.</p>	<ul style="list-style-type: none"> • Y=279 (1) 00-63: Trunk Route No. (2) 0-7 : Pattern No. 0-7 NONE◀: No data
CM50	<p>Assign the Pattern number for adding an access code for outgoing call to the calling number stored by Message Reminder when terminating a tandem call via SIP trunk.</p>	<ul style="list-style-type: none"> • Y=11 (1) 0-7: Pattern No. 0-7 assigned by CM35 Y=279 (2) X-XXXXXX: Access Code for outgoing call (1-6 digits) X : 0-9, A (*), B (#) NONE◀: No data
G		

G	DESCRIPTION	DATA
CM41	Assign the PBR timer when establishing tandem connection to CCIS.	<ul style="list-style-type: none"> • Y=0 (1) 104: PBR timer (2) 02-99 : 2-99 seconds (1 second increments) NONE◀: 7 seconds
CMA7	Assign an Originating Point Code (OPC) to each SIP trunk number.	<ul style="list-style-type: none"> • Y=01 (1) 00-63: SIP trunk No. (2) 00001-16367: Originating Point Code
	RESET	
	NOTE 1: <i>Originating Point Code (OPC) is not informed to SIP servers.</i>	
	NOTE 2: <i>The Originating Point Code (OPC) is used to designate an originating office in the No. 7 CCIS network. A single Originating Point Code (OPC) should be assigned to all SIP trunk numbers provided in the same system.</i>	
	NOTE 3: <i>One OPC cannot be assigned to multiple stations.</i>	
H		

H	DESCRIPTION	DATA
CMA7	<p>Assign a Destination Point Code (DPC) to each SIP trunk number.</p> <p>NOTE 1: <i>A DPC assigned by this command is used only within the Main Unit, and will not be sent to a SIP Server and others.</i></p> <p>NOTE 2: <i>For this DPC, assign an arbitrary value which is different from a value set by CMA7 Y=1.</i></p> <p>NOTE 3: <i>When a SIP Trunk is used in a Point-to-Multipoint connection (i.e. the second data=0 for CMA7 Y=46) and the voice encoding is switched by LCR Pattern number, assign both of CM8A Y=5000-5255>168 (DPC) and this command.</i></p> <p>Assign the Profile number for control packet to each SIP trunk number.</p> <p>NOTE: <i>When a SIP Trunk is used in a Point-to-Multipoint connection (i.e. the second data=0 for CMA7 Y=46) and the voice encoding is switched by LCR Pattern number, assign both of CM8A Y=5000-5255>179 (SIP Profile number) and this command.</i></p> <p>Specify the accommodated unit per SIP trunk number.</p>	<ul style="list-style-type: none"> • Y=02 (1) 00-63: SIP trunk No. (2) 00001-16367: Destination Point Code <ul style="list-style-type: none"> • Y=71 (1) 00-63: SIP trunk No. (2) 00-63: Profile No. for control packet <ul style="list-style-type: none"> • Y=78 (1) 00-63: SIP trunk No. (2) 01-50 : Unit number NONE◀: No data
I	<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	

I	DESCRIPTION	DATA
CMA8	Assign the SIP number to the Destination Point Code (DPC) for SIP assigned by CMA7 Y=02. <div style="text-align: right;">(RESET)</div>	<ul style="list-style-type: none"> • Y=0 (1) 00001-16367: Destination Point Code assigned by CMA7 Y=02 (2) 00-63: SIP trunk No.
<p>NOTE: <i>Set the SIP trunk number of the SIP trunk to the Destination Point Code (DPC) of the SIP server assigned by CMA7 Y=02.</i></p>		
CMBA	Specify the Type of Service (TOS) field precedence for SIP trunk control packet. <div style="text-align: right;">(RESET)</div>	<ul style="list-style-type: none"> • Y=04 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 00-07: PRECEDENCE 0-7 15◀ : PRECEDENCE 0
<p>NOTE 1: <i>The higher number has higher priority.</i></p>		
<p>NOTE 2: <i>For a setting value for this command, consult with a network manager. If no specific instruction is given from the manager, set the second data to “06”.</i></p>		
<p>NOTE 3: <i>Assigning this command enables a router to recognize the precedence with WFQ (Weighed Fair Queuing) and to control the control packets according to the precedence.</i></p>		
<p>NOTE 4: <i>When SIP message 481 “Call/Transaction Does Not Exist, 100 trying” is sent, the profile number (by CMA7 Y=71) is followed.</i></p>		
<p>NOTE 5: <i>This command assigns QoS for packets that are sent from a unit which accommodates a SIP trunk to another device to which the SIP trunk is connected.</i></p>		
J		

J	DESCRIPTION	DATA
CMBA	<p>When the route provides DiffServ QoS, specify the TOS field Precedence for SIP trunk control packet.</p>	<ul style="list-style-type: none"> • Y=10 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 00-3F: TOS field Precedence
	<p>RESET</p>	
	<p>NOTE 1: For a setting value for this command, consult with a network manager. If no specific instruction is given from the manager, do not assign any data to this command.</p>	
	<p>NOTE 2: When this data is set, the TOS field precedence set by CMBA Y=04 becomes unavailable. To make the TOS field precedence available again set “CCC” (data clear) for CMBA Y=10.</p>	
	<p>NOTE 3: Assigning this command enables a router to recognize the precedence with Diffserv (Differentiated Service) and to control the control packets according to the precedence.</p>	
	<p>NOTE 4: When SIP message 481 “Call/Transaction Does Not Exist, 100 Trying” is sent, the profile number (by CMA7 Y=71) is followed.</p>	
	<p>NOTE 5: This command assigns QoS for packets that are sent from a unit which accommodates a SIP trunk to another device to which the SIP trunk is connected.</p>	
K		

K

CMBA

DESCRIPTION

DATA

Specify voice encoding selection precedence for SIP trunk.

- Y=21
- (1) 00-63: Profile No. assigned by CMA7
- (2) 0 : Programmable List
- 1 : Standard Mode 2
- 2 : Standard Mode 3
- 3 : Tone Quality Mode 2
- 4 : Band Mode 2
- 5 : Tone Quality Mode 1
- 6 : Band Mode 1
- 7◀: Standard Mode 1

NOTE 1: The meanings of the second data are shown below.

SETTING DATA	MODE	HIGH ← SELECTION PRECEDENCE → LOW		
		1	2	3
0	Programmable List	As per CMBA Y=121	As per CMBA Y=122	As per CMBA Y=123
1	Standard Mode 2	G.711 μ -law (64 Kbps)		
2	Standard Mode 3	G.711 μ -law (64Kbps)	G.711A-law (64Kbps)	–
3	Tone Quality Mode 2	G.711 μ -law (64Kbps)	G.711A-law (64Kbps)	G.729a (8Kbps)
4	Band Mode 2	G.729a (8Kbps)	G.711A-law (64Kbps)	G.711A-law (64Kbps)
5	Tone Quality Mode 1	G.711 μ -law (64Kbps)	G.729a (8Kbps)	–
6	Band Mode 1	G.729a (8Kbps)	G.711 μ -law (64Kbps)	–
7◀	Standard Mode 1	G.711 μ -law (64Kbps, 20ms fixed)		

NOTE 2: When the voice encoding selection setting differs from that for the opposite SIP trunk, the voice encoding selection may differ in the user's usual SIP trunk setting according to the negotiation when the SIP session is made.

NOTE 3: Payload size is set as per CMBA Y=22 when the second data of CMBA Y=21 is set to a value other than "7" (Standard Mode 1).

L

L	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	Specify payload size for SIP trunk.	<ul style="list-style-type: none"> • Y=22 (1) 00-63: Profile No. assigned by CMA7 1 : 20 ms. (2) 2 : 30 ms. 3◀: 40 ms.
	Assign the Port number for the SIP Server.	<ul style="list-style-type: none"> • Y=31 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 00000-65534: SIP Server Port No. NONE◀ : 05060
	(RESET)	
	<p>NOTE: <i>Because the standard payload size for SIP is 20 ms., it is strongly recommended to set the second data to "1".</i></p> <p>NOTE 1: <i>The common Port number is 05060.</i></p> <p>NOTE 2: <i>Set the same port numbers to the all ports in a network for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p>	
	Specify the provisional response code when the system receives the incoming call and starts to call the stations.	<ul style="list-style-type: none"> • Y=91 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : 183 Session Progress (with SDP) 1 : 180 Ringing (with SDP) 7◀: 180 Ringing (without SDP)
(RESET)		
Specify the error response code when all incoming SIP trunks are busy.	<ul style="list-style-type: none"> • Y=97 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : 480 Temporarily Unavailable 1 : 486 Busy Here 7◀: 503 Service Unavailable 	
(RESET)		
<p>NOTE: <i>Set the second data to "7" for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p>		
M		

M	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	Specify the periodical resending of provisional response when receiving to SIP trunk.	<ul style="list-style-type: none"> • Y=104 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To provide 1 : To provide only when 100rel is provided 3◀: Not provided
	Specify whether to request provisional responses with reliability (100rel) when sending from SIP trunk.	<ul style="list-style-type: none"> • Y=105 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Available (Supported header and Require header) 1 : Available (Supported header) 3◀: Not available
	Specify response code when the terminating terminal/trunk is busy.	<ul style="list-style-type: none"> • Y=106 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : 503 7◀: 486
	Specify whether to require provisional responses with reliability (100rel) when receiving.	<ul style="list-style-type: none"> • Y=114 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Not required 1◀: Auto
	Specify whether to add “+” for calling number/delete “+” for called number.	<ul style="list-style-type: none"> • Y=117 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To provide 1◀: Not provided
	Specify the CODEC type of SIP Trunk (First priority).	<ul style="list-style-type: none"> • Y=121 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 01 : G.711 μ-law (64 Kbps) 02 : G.711 A-law (64 Kbps) 04 : G.729a (8 Kbps) NONE◀: No data
	<p>NOTE: This command is effective when CMBA Y=21 is set to “0” (programmable list).</p>	
<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>		
<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>		
N		

N	DESCRIPTION	DATA
CMBA	<p>Specify the CODEC type of SIP Trunk (Second priority).</p> <p style="text-align: right;">(RESET)</p> <p>NOTE: <i>This command is effective when CMBA Y=21 is set to "0" (programmable list).</i></p> <p>Specify the CODEC type of SIP Trunk (Third priority).</p> <p style="text-align: right;">(RESET)</p> <p>NOTE: <i>This command is effective when CMBA Y=21 is set to "0" (programmable list).</i></p> <p>Select how to reply for re-INVITE SDP which has unsupported media only.</p> <p style="text-align: right;">(RESET)</p>	<ul style="list-style-type: none"> • Y=122 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 01 : G.711 μ-law (64 Kbps) 02 : G.711 A-law (64 Kbps) 04 : G.729a (8 Kbps) NONE◀: No data <ul style="list-style-type: none"> • Y=123 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 01 : G.711 μ-law (64 Kbps) 02 : G.711 A-law (64 Kbps) 04 : G.729a (8 Kbps) NONE◀: No data <ul style="list-style-type: none"> • Y=127 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Reply with SIP status 488 and maintain session 3◀: Reply with SIP status 200OK which contains the answer for each media type
END		

Data Settings for Point-to-Point Connection

When the connection type of SIP trunk is Point-to-Point, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	<p>Assign the IP Address for the SIP server.</p> <p style="text-align: right;">RESET</p> <p>NOTE 1: <i>This data setting is not required when the IP address for SIP server is determined by DNS resolution.</i></p> <p>NOTE 2: <i>For Point-to-Multipoint connection (when CMA7 Y=46 is set to "0"), the IP address assigned by this data depends on the IP address pattern assigned by CM8A Y=5000-5255> 167. Page 2-285</i></p>	<ul style="list-style-type: none"> • Y=30 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXXXXXXXXXXXX: 000000000001-255255255254: SIP Server IP Address (12 digits) NONE◀: No data
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">A</div>	<p>Assign Representative number (Registration number) for the SIP trunk.</p> <p>NOTE 1: <i>When the second data of CMBA Y=44 is set to "0" or "1", the Representative number (Registration number) assigned by this command is used as the calling number.</i></p> <p>NOTE 2: <i>For Representative number (Registration number), 0-9, A (*) and B (#) can be used.</i></p> <p>NOTE 3: <i>When connecting to a SIP carrier, assign a Contract Number, specified by the carrier, as the Representative number with this command.</i></p> <p>NOTE 4: <i>This setting is not required when connecting to carrier with SIP AoR (Address of Record) user name.</i></p>	<ul style="list-style-type: none"> • Y=32 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) X-XX...XX: Representative number (Registration number) (1-16 digits) X: 0-9, A (*), B (#) NONE◀: No data

A	DESCRIPTION	DATA
CMBA	Specify the SIP trunk registration method to the SIP server.	<ul style="list-style-type: none"> • Y=70 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To register the time set by CMBA Y=71 3◀: Not registered
	RESET	
	<p>NOTE 1: When connecting a SIP carrier, set the second data to “0”, because a registration to the SIP server is generally required.</p>	
	<p>NOTE 2: SIP trunk is re-registered half the time set by CMBA Y=71 to SIP server periodically when the second data is set to “0”.</p>	
	<p>NOTE 3: This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to “0”).</p>	
	Assign the SIP trunk registration expiry time to the SIP server.	<ul style="list-style-type: none"> • Y=71 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 120-8553600: 120 seconds-8553600 seconds (99 days) NONE◀ : 3600 seconds (1 hour)
	RESET	
	<p>NOTE 1: When no registration expiry time is specified by the SIP carrier, assigning this data is not required.</p>	
	<p>NOTE 2: This data setting is effective only when CMBA Y=70 is set to “0”.</p>	
	<p>NOTE 3: Set the time to cancel the registration after registering SIP trunk with this command to SIP server.</p>	
	<p>NOTE 4: When the registration has been canceled by SIP server, re-register to SIP server for half the period of time set by this command (in case of 3600 seconds, set 1800 seconds).</p>	
	<p>NOTE 5: When re-registration from SIP server is not executed during the period of time set by this command after the registration has been canceled by SIP server, call reception from the network to SIP is restricted.</p>	
	<p>NOTE 6: This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to “0”).</p>	
B		

B	DESCRIPTION	DATA
CMBA	Assign the Authentication user name when registering to/sending from the SIP server with character code.	<ul style="list-style-type: none"> • Y=72 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) X-XX...XX: User name (Maximum 32 digits) NONE◀: No data
	<p style="text-align: right;">RESET</p> <p>NOTE: <i>You can confirm the user name set by this command with CMBA Y=73.</i></p>	
	Assign the Authentication user name when registering to/sending from the SIP server with character code (First 12 characters).	<ul style="list-style-type: none"> • Y=73 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: User name (24 digits, 12 characters fixed) See Character Code Table. Page 2-293 NONE◀: No data
	<p style="text-align: right;">RESET</p> <p>NOTE 1: <i>A character string into which characters assigned by CMBA Y=73, 100 and 101 are concatenated is used as an Authentication user name.</i></p>	
	<p>NOTE 2: <i>When the character code to be set is less than the number of digits necessary, add the character code FF.</i></p>	
	<p>NOTE 3: <i>You can confirm the user name set by this command with CMBA Y=72.</i></p>	
C		

C	DESCRIPTION	DATA
CMBA	<p>Assign the Authentication user name when registering to/sending from the SIP server with character code (Middle 12 characters).</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=100 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: User name (24 digits, 12 characters fixed) See Character Code Table. Page 2-293 <p>NONE◀: No data</p>
	<p>NOTE 1: A character string into which characters assigned by CMBA Y=73, 100 and 101 are concatenated is used as an Authentication user name.</p>	
	<p>NOTE 2: When the character code to be set is less than the number of digits necessary, add the character code FF.</p>	
	<p>NOTE 3: You can confirm the user name set by this command with CMBA Y=72.</p>	
	<p>Assign the Authentication user name when registering to/sending from the SIP server with character code (Last 8 characters).</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=101 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: User name (16 digits, 8 characters fixed) See Character Code Table. Page 2-293 <p>NONE◀: No data</p>
	<p>NOTE 1: A character string into which characters assigned by CMBA Y=73, 100 and 101 are concatenated is used as an Authentication user name.</p>	
	<p>NOTE 2: When the character code to be set is less than the number of digits necessary, add the character code FF.</p>	
	<p>NOTE 3: You can confirm the user name set by this command with CMBA Y=72.</p>	
D		

D	DESCRIPTION	DATA
CMBA	<p>Assign the Authentication password when registering to/sending from the SIP server with character code.</p> <p style="text-align: right;">RESET</p> <p>NOTE: <i>You can confirm the password set by CMBA Y=75 with this command.</i></p> <p>Assign the Authentication password when registering to/sending from the SIP server with character code (First 12 characters).</p> <p style="text-align: right;">RESET</p> <p>NOTE 1: <i>A character string into which characters assigned by CMBA Y=75, 102 and 103 are concatenated is used as an Authentication password.</i></p> <p>NOTE 2: <i>When the character code to be set is less than the number of digits necessary, add the character code FF.</i></p> <p>NOTE 3: <i>You can confirm the password set by this command with CMBA Y=74.</i></p> <p>NOTE 4: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p>	<ul style="list-style-type: none"> • Y=74 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: Password (Maximum 32 digits) NONE◀: No data <ul style="list-style-type: none"> • Y=75 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: Password (24 digits, 12 characters fixed) See Character Code Table. Page 2-293 NONE◀: No data
E		

E	DESCRIPTION	DATA
CMBA	Assign the Authentication password when registering to/sending from the SIP server with character code (Middle 12 characters).	<ul style="list-style-type: none"> • Y=102 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: Password (24 digits, 12 characters fixed) See Character Code Table. Page 2-293 NONE◀: No data
	<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	
	<p>NOTE 1: <i>A character string into which characters assigned by CMBA Y=75, 102 and 103 are concatenated is used as an Authentication password.</i></p>	
	<p>NOTE 2: <i>When the character code to be set is less than the number of digits necessary, add the character code FF.</i></p>	
	<p>NOTE 3: <i>You can confirm the password set by this command with CMBA Y=74.</i></p>	
	<p>NOTE 4: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p>	
F		

F	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	<p>Assign the Authentication password when registering to/sending from the SIP server with character code (Last 8 characters).</p> <p style="text-align: right;">RESET</p> <p>NOTE 1: <i>A character string into which characters assigned by CMBA Y=75, 102 and 103 are concatenated is used as an Authentication user name.</i></p> <p>NOTE 2: <i>When the character code to be set is less than the number of digits necessary, add the character code FF.</i></p> <p>NOTE 3: <i>You can confirm the password set by this command with CMBA Y=74.</i></p> <p>NOTE 4: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>Specify the IP address used for SIP-URI.</p> <p style="text-align: right;">RESET</p> <p>NOTE: <i>Assign this data when an IP address is used for SIP URI of SIP Trunk (i.e. when no domain name is assigned to the second data of CMBA Y=76).</i></p> <p>Specify the SIP trunk identity header.</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=103 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: Password (16 digits, 8 characters fixed) See Character Code Table. Page 2-293 <p>NONE◀: No data</p> <ul style="list-style-type: none"> • Y=85 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Address of SIP interface set by CM0B Y=101>00 1◀: IP address of SIP server set by CMBA Y=30 <ul style="list-style-type: none"> • Y=86 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : P-Asserted-Identity 1◀: P-Preferred-Identity
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">G</div>		

G	DESCRIPTION	DATA
CMBA	<p>Specify the Stop No-Answer timer (T1 timer) when receiving first response (100Trying).</p> <p>NOTE: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>Specify the SIP trunk identity header.</p> <p style="text-align: right;">(RESET)</p> <p>NOTE: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>Specify whether to provide Session Refreshment when 18X: provisional response is received again.</p> <p>NOTE: <i>Set the second data to "1" for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>Specify whether to perform re-registration unconditionally when receiving "authentication error" on the registration.</p> <p style="text-align: right;">(RESET)</p>	<ul style="list-style-type: none"> • Y=87 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Not stopped 1◀: To stop <ul style="list-style-type: none"> • Y=55 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : SIP-URL + tel-URL 1 : SIP-URL 2 : tel-URL 3 : SIP-URL + tel-URL only when the calling number is not informed 4 : SIP-URL only when the calling number is not informed 5 : tel-URL only when the calling number is not informed 6 : Remote-Party-ID only when the calling number is not informed 7◀: No identity header <ul style="list-style-type: none"> • Y=27 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Not provided 1◀: Provide <ul style="list-style-type: none"> • Y=107 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To provide unconditionally 1◀: To provide only when nonce value does not match
H		

H	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	<p>Specify whether to perform registration when also receiving “subscriber error”, “authentication error” or “timeout error” during the registration.</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=108 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To provide 1◀: Not provided
	<p>Specify the way of response when receiving NON-INVITE in IDLE state.</p>	<ul style="list-style-type: none"> • Y=109 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Not answered 1◀: To answer with 481
	<p>Specify whether to send a signal to require a deletion during a reset setting registration.</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=110 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To provide 1◀: Not provided
	<p>NOTE: <i>When this data is set to “0”, if REGISTER registration request is sent by CMA7 Y=77, the registration of the SIP trunk is disconnected from the SIP server for a moment.</i></p>	
	<p>Specify the logic to be used for contact header user field.</p> <p style="text-align: right;">RESET</p>	<ul style="list-style-type: none"> • Y=112 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Random 1◀: As per the setting of Username field by CMBA Y=92

END

DNS Server Data Settings

(1) When DNS is to be used in Point-to-Point connection, do the following programming.

START	DESCRIPTION	DATA
CM0B	Assign the Primary IP address for DNS server. RESET	<ul style="list-style-type: none"> Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 60 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address for the DNS server (Maximum 15 digits) NONE◀: No data
	<p>NOTE 1: The second data must be entered including the periods (.).</p> <p>NOTE 2: Set the IP address of DNS server only when using the domain name to describe URL.</p>	
	Assign the Secondary IP address for DNS server. RESET	<ul style="list-style-type: none"> Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 61 (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: IP Address for the DNS server (Maximum 15 digits) NONE◀: No data
	<p>NOTE 1: The second data must be entered including the periods (.).</p> <p>NOTE 2: Set the IP address of DNS server only when using the domain name to describe URL.</p>	
CMBA	Assign the SIP trunk domain name for SIP-URI with character. RESET	<ul style="list-style-type: none"> Y=76 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) X-XX...XX: Domain name (Maximum 128 characters) NONE◀: No data
	<p>NOTE 1: When SIP-URI is "1234@nec.com", the part of "nec.com" is set by this command. When NONE is set, IP address such as "1234@192.168.1.1" is used.</p> <p>NOTE 2: This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</p>	
A		

A	DESCRIPTION	DATA
CMBA	Specify whether to query a DNS server to get the IP address or not.	<ul style="list-style-type: none"> • Y=25 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To provide 1 ◀: Not provided
	<p style="text-align: center;">RESET</p> <p>NOTE 1: <i>When the second data is set to “0”, only the IP address replied from to a DNS server is used. When the second data is set to “1”, the SIP server IP address assigned by CMBA Y=30 is used.</i></p> <p>NOTE 2: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to “0”).</i></p>	
	Assign the Fully Qualified Domain Name (FQDN) for SIP server with character.	<ul style="list-style-type: none"> • Y=93 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) X-XX...XX: Domain name (Maximum 128 characters) NONE ◀: No data
	<p style="text-align: center;">RESET</p> <p>NOTE 1: <i>“CCC” cannot be registered.</i></p> <p>NOTE 2: <i>When the setting of this command is changed, reset the DNS cache table should be cleared by CMBA Y=99.</i></p>	
END		

(2) To clear the DNS cache table, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CMBA</div>	Clear the cache table. <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px; margin-top: 10px;">RESET</div>	<ul style="list-style-type: none"> • Y=99 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) CCC: DNS cache table clearance
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">END</div>	<p>NOTE: When an IP address has been cached in the DNS cache table, the IP address cached is displayed in the second data.</p>	

User Name Settings by SIP AoR (Address of Record)

For setting the user name by using SIP AoR (Address of Record), do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CMBA</div>	Assign the SIP AoR user name with character. <div style="text-align: right; border: 1px solid black; border-radius: 15px; padding: 2px 10px; margin-top: 10px;">RESET</div>	<ul style="list-style-type: none"> • Y=45 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) X-XX...XX: SIP AoR user name (Maximum 32 characters) NONE◀: No data
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">A</div>	<p>NOTE 1: You can also confirm the SIP AoR user name set by CMBA Y=46/47/54 with this command.</p> <p>NOTE 2: This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</p>	

A	DESCRIPTION	DATA
CMBA	Assign the SIP AoR user name with character code (First 12 characters).	<ul style="list-style-type: none"> • Y=46 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: SIP AoR user name (Maximum 24 digits: 12 characters fixed) See Character Code Table. Page 2-293 NONE◀: No data
	<p>NOTE 1: A character string into which characters assigned by CMBA Y=46, 47 and 54 are concatenated is used as a SIP AoR user name.</p> <p>NOTE 2: When the character code to be set is less than the number of digits necessary, add the character code FF.</p> <p>NOTE 3: You can confirm the user name set by this command with CMBA Y=45.</p> <p>NOTE 4: This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</p>	
	Assign the SIP AoR user name with character code (Middle 12 characters).	<ul style="list-style-type: none"> • Y=47 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: SIP AoR user name (Maximum 24 digits: 12 characters fixed) See Character Code Table. Page 2-293 NONE◀: No data
B	<p>NOTE 1: A character string into which characters assigned by CMBA Y=46, 47 and 54 are concatenated is used as a SIP AoR user name.</p> <p>NOTE 2: When the character code to be set is less than the number of digits necessary, add the character code FF.</p> <p>NOTE 3: You can confirm the user name set by this command with CMBA Y=45.</p> <p>NOTE 4: This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</p>	

B	DESCRIPTION	DATA
CMBA	Assign the SIP AoR user name with character code (Last 8 characters).	<ul style="list-style-type: none"> • Y=54 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXX...X: SIP AoR user name (Maximum 16 digits: 8 characters fixed) See Character Code Table. Page 2-293 <p>NONE◀: No data</p>
	RESET	
	<p>NOTE 1: A character string into which characters assigned by CMBA Y=46, 47 and 54 are concatenated is used as a SIP AoR user name.</p>	
	<p>NOTE 2: When the character code to be set is less than the number of digits necessary, add the character code FF.</p>	
	<p>NOTE 3: You can confirm the user name set by this command with CMBA Y=45.</p>	
	<p>NOTE 4: This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</p>	
	Specify the display name/user name for From Header.	<ul style="list-style-type: none"> • Y=92 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Display name: SIP AoR User name following CMBA Y=45 : User name : SIP AoR User name following CMBA Y=45 2 : Display name: Caller ID following CMBA Y=44 : User name : SIP AoR User name following CMBA Y=45 3◀: Display name: Caller ID following CMBA Y=44 : User name : Caller ID following CMBA Y=44
	<p>NOTE 1: When a calling party wants to inform a called party of the calling party name, be sure to set the second data of this command to "2" or "3".</p>	
	<p>NOTE 2: For a carrier connection using an AoR User name, be sure to set the second data to "0" or "2".</p>	
C		



CMBA

END

DESCRIPTION

DATA

Specify the username for To header of REGISTER message.
[9300V5 software required]

- Y=177
- (1) 00-63: Profile No. assigned by CMA7
Y=71
- (2) 0 : As per CMBA Y=45
- 1 : As per CMBA Y=32
- 7◀: As per CMBA Y=92

NOTE: The username for To header of REGISTER message is set as the following table depending on the combination of CMBA Y=92 and CMBA Y=177.

◀: Default

		Second Data of CMBA Y=92		
		0	2	3◀
Second Data of CMBA Y=177 [9300V5 software required]	0	As per CMBA Y=45		
	1	As per CMBA Y=32		
	7◀	As per CMBA Y=45	As per CMBA Y=45	As per CMBA Y=32

Specify the from header/contact header for REGISTER message.

- Y=157
- (1) 00-63: Profile No. assigned by CMA7
Y=71
- (2) 0 : As per CMBA Y=45
- 1 : As per CMBA Y=32
- 7◀: As per CMBA Y=92

Data Settings for Point-to-Multipoint Connection

In case the SIP trunk connection method is Point-to-Multipoint, do the following programming.

START	DESCRIPTION	DATA
CM8A	<p>Assign the Destination IP Address Pattern number to the LCR Pattern number.</p> <p>NOTE: <i>Set this command only for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p>	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 167: Destination IP Address Pattern (2) 000-255: IP Address Pattern No.
	<p>Assign the Destination Point Code (DPC) for SIP Point-to-Multipoint connection.</p> <p>NOTE: <i>Set this command only for Point-to-Multipoint connection when Voice encoding is switched by LCR Pattern number.</i></p>	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 168: Destination Point Code for SIP Point-to-Multipoint connection (2) 00001-16367: Destination Point Code
	<p>Assign the Profile number for the LCR Pattern number.</p> <p>NOTE: <i>Set this command only for Point-to-Multipoint connection when Voice encoding is switched by LCR Pattern number.</i></p>	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 179: Profile Number for SIP Point-to-Multipoint connection (2) 00-63 : Profile Number NONE◀: As per CMA7 Y=71
A		

A	DESCRIPTION	DATA
CM5B	Assign the Destination IP Address for SIP trunk.	<ul style="list-style-type: none"> • Y=01 Destination IP Address for SIP Trunk XXX ZZ (1) XXX: 000-255: IP Address Pattern No. assigned by CM8A Y=5000-5255>167 ZZ : 00-07: IP Address No. 00-07 <li style="text-align: right;">NOTE 2 (2) XXXXXXXXXXXXX: 000000000001-255255255254: Destination IP Address of opposite SIP Trunk (Maximum 12 digits)
	NOTE 1: Set this command only for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").	
	NOTE 2: IP address number (00-07) is the number for specifying the IP addresses of the SIP trunks of opposite office (SV9300/SV8300/NEAX2000IPS) which has plural SIP trunks, when the offices are connected with Point-to-Multipoint.	
	NOTE 3: This command should be assigned from the leading number of 1st data (IP Address Pattern No. + IP Address No.).	
<u>END</u>		

Ringing Tone

To specify the interval of ringing tones and the ringer tone pattern to each DID number:

START	DESCRIPTION	DATA
CM08	Provide Ringer Tone Pattern assigned by CM76 Y=22/23 for a DID incoming call from the SIP trunk.	(1) 649 (2) 0: To provide
CM76	Specify the ringing tone interval of SLT or Multiline Terminal on DID calls. NOTE 1: Do not set the second data to “3” (as per CM35 Y=033 (Multiline Terminal) or as per CM04 Y=00>06). [North America Only] NOTE 2: For Multiline Terminal, Special Ringing; 0.5 seconds ON-0.5 seconds OFF [For Australia/Asia/Africa/Europe/Latin America/Middle East/Russia] or 0.25 seconds ON-0.25 seconds OFF-0.25 seconds ON-0.25 seconds OFF [For EMEA] is applied.	<ul style="list-style-type: none"> • Y=22 (1) 000-999: Number Conversion Block No. assigned by CM76 Y=90 (2) For Multiline Terminal: 0: 0.5 seconds ON-0.5 seconds OFF 1: 0.5 seconds ON-0.5 seconds OFF -0.5 seconds ON-1.5 seconds OFF 2: 1 second ON-2 seconds OFF For Single Line Telephone: 0: As per CM04 Y=00>05 1: As per CM04 Y=00>07 2: As per CM04 Y=00>05 (See Interval of Ringing Tones for Single Line Telephone by CM04 Y=00>05/06/07) Page 2-292 [North America Only] For Multiline Terminal: 0: Rering 1: Special Ringing 2: Internal Ringing (See Interval of Ringing Tones for Multiline Terminal by CM08>392/396/397) Page 2-291 For Single Line Telephone: 0: As per CM04 Y=00>05 1: As per CM04 Y=00>07 2: As per CM04 Y=00>05 (See Interval of Ringing Tones for Single Line Telephone by CM04 Y=00>05/06/07) Page 2-292 [Other than North America]
A		

A	DESCRIPTION	DATA
CM76	Specify the Multiline Terminal Ringer Tone Pattern on DID calls. For this assignment, do not set CM76 Y=23 to 7 (As per CM35 Y=034/164).	<ul style="list-style-type: none"> • Y=23 (1) 000-999: Number Conversion Block No. assigned by CM76 Y=00/90, CM2A Y=50-52 (2) 0 : Ringer Tone Pattern 0 1 : Ringer Tone Pattern 1 2 : Ringer Tone Pattern 2 3 : Ringer Tone Pattern 3 4 : Ringer Tone Pattern 4 5 : Ringer Tone Pattern 5 6 : Ringer Tone Pattern 6 7◀: As per CM35 Y=034/164
CMBA	Specify the Distinctive Ringing Pattern for incoming calls from SIP trunk.	<ul style="list-style-type: none"> • Y=113 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : EXTERNAL 1◀: INTERNAL
<u>END</u>		

To specify the interval of ringing tones and the ringer tone pattern to each trunk route:

START	DESCRIPTION	DATA
CM08	Provide Ringer Tone Pattern assigned by CM35 Y=033/034/164 for an incoming trunk call from the SIP trunk.	(1) 649 (2) 0: To provide
CM35	Specify the ringing tone interval of SLT or Multiline Terminal to an incoming call for voice channel of SIP trunk route. NOTE 1: <i>For incoming calls to Trunk Line Appearance key on Multiline Terminal, the special ringing; 0.2 seconds ON-0.2 seconds OFF will be applied. [North America Only]</i> NOTE 2: <i>For Multiline Terminal, Special Ringing; 0.5 seconds ON-0.5 seconds OFF [For Australia/Asia/Africa/Europe/Latin America/Middle East/Russia] or 0.25 seconds ON-0.25 seconds OFF-0.25 seconds ON-0.25 seconds OFF [For EMEA] is applied.</i>	<ul style="list-style-type: none"> • Y=033 (1) 00-63: Trunk Route No. 00-63 (2) For Multiline Terminal: NOTE 1 0 : 0.4 seconds ON-0.2 seconds OFF -0.4 seconds ON-2 seconds OFF 1 : 0.4 seconds ON-0.2 seconds OFF -0.4 seconds ON-2 seconds OFF 2 : 1 second ON-2 seconds OFF 3◀: 2 seconds ON-4 seconds OFF For Single Line Telephone: 0 : As per CM04 Y=00>05 1 : As per CM04 Y=00>07 2 : As per CM04 Y=00>05 3◀: As per CM04 Y=00>06 (See Interval of Ringing Tones for Single Line Telephone by CM04 Y=00>05/06/07) Page 2-292 [North America Only] For Multiline Terminal: 0 : Ringing 1 : Special Ringing 2 : Internal Ringing 3◀: External Ringing (See Interval of Ringing Tones for Multiline Terminal by CM08>392/396/397) Page 2-291 NOTE 2 For Single Line Telephone: 0 : As per CM04 Y=00>05 1 : As per CM04 Y=00>07 2 : As per CM04 Y=00>05 3◀: As per CM04 Y=00>06 (See Interval of Ringing Tones for Single Line Telephone by CM04 Y=00>05/06/07) Page 2-292 [Other than North America]
A		

A

DESCRIPTION

DATA

CM35

Specify the Multiline Terminal Ringer Tone Pattern on incoming calls.

- Y=034
- Y=164
- (1) 00-63: Trunk Route No. 00-63
- (2) See the table below.

◀: Default

CM35 Y=034	CM35 Y=164: 0	CM35 Y=164: 1 ▶
0	Ringer Tone Pattern 3	Ringer Tone Pattern 0
1	Ringer Tone Pattern 6	Ringer Tone Pattern 1
2	Ringer Tone Pattern 5	Ringer Tone Pattern 2
3 ▶	Ringer Tone Pattern 4	Ringer Tone Pattern 7

CM76

Specify the interval of ringing tones on DID calls.

- Y=22
- (1) 000-999: Number Conversion Block No. assigned by CM76 Y=90
- (2) 3 ▶: As per CM35 Y=033

Specify the Multiline Terminal Ringer Tone Pattern on DID calls.

- Y=23
- (1) 000-999: Number Conversion Block No. assigned by CM76 Y=90
- (2) 7 ▶: As per CM35 Y=034/164

CMBA

Specify the Distinctive Ringing Pattern for incoming calls from SIP trunk.

- Y=113
- (1) 00-63: Profile No. assigned by CMA7 Y=71
- (2) 0 : EXTERNAL
- 1 ▶: INTERNAL

END

**Interval of Ringing Tones for Multiline Terminal
by CM08>392/396/397**

Unit: seconds ◀: Default

Pattern	1	2	3	4	5	6	7	8
CM08>392	1◀	0	1◀	0	1◀	0	0	1◀
CM08>396	1◀	1◀	0	0	1◀	0	1◀	0
CM08>397	1◀	1◀	1◀	1◀	0	0	0	0
Internal Ringing	1 ON -2 OFF	1 ON -2 OFF	2 ON -4 OFF	2 ON -4 OFF	0.375 ON -0.25 OFF -0.375 ON -2 OFF	0.375 ON -0.25 OFF -0.375 ON -2 OFF	1 ON -4 OFF	0.25 ON -0.25 OFF -0.25 ON -4.25 OFF
External Ringing	0.375 ON -0.25 OFF -0.375 ON -2 OFF	2 ON -4 OFF	0.375 ON -0.25 OFF -0.375 ON -2 OFF	2 ON -4 OFF	2 ON -4 OFF	2 ON -4 OFF	0.25 ON -0.25 OFF -0.25 ON -4.25 OFF	1 ON -4 OFF
Special Ringing	0.25 ON -0.125 OFF -0.25 ON -0.125 OFF -0.25 ON -2 OFF	0.5 ON -0.5 OFF -0.5 ON -1.5 OFF	0.25 ON -0.125 OFF -0.25 ON -0.125 OFF -0.25 ON -2 OFF	0.5 ON -0.5 OFF -0.5 ON -1.5 OFF	0.25 ON -0.125 OFF -0.25 ON -0.125 OFF -0.25 ON -2 OFF	0.25 ON -0.125 OFF -0.25 ON -0.125 OFF -0.25 ON -2 OFF	0.25 ON -0.125 OFF -0.25 ON -0.125 OFF -0.25 ON -2 OFF	0.25 ON -0.125 OFF -0.25 ON -0.125 OFF -0.25 ON -2 OFF

NOTE 1: The above ringer patterns (5-8) are effective only when CM31 Y=0>0: 04, 15.

NOTE 2: Pattern 5 is standard setting for Brazil.

NOTE 3: Pattern 6 is standard setting for France.

NOTE 4: Pattern 7 and 8 are standard setting for EMEA.

**Interval of Ringing Tones for Single Line Telephone
by CM04 Y=00>05/06/07**

◀: Default

2nd Data	CM04 Y=00		
	1st Data=05 (Single Line Telephone ringing signal for Station-to-Station connection)	1st Data=06 (Single Line Telephone ringing signal from a trunk)	1st Data=07 (Special ringing signal for Single Line Telephone ring)
01	ON	ON	ON
02	2 seconds ON-4 seconds OFF	2 seconds ON-4 seconds OFF	2 seconds ON-4 seconds OFF
03	1 second ON-2 seconds OFF	1 second ON-2 seconds OFF	1 second ON-2 seconds OFF
04	0.5 seconds ON-0.5 seconds OFF	0.5 seconds ON-0.5 seconds OFF	0.5 seconds ON-0.5 seconds OFF
05	0.25 seconds ON-0.25 seconds OFF	0.25 seconds ON-0.25 seconds OFF	0.25 seconds ON-0.25 seconds OFF
06	0.5 seconds ON-0.5 seconds OFF -0.5seconds ON-1.5 seconds OFF	0.5 seconds ON-0.5 seconds OFF -0.5 seconds ON-1.5 seconds OFF	0.5 seconds ON-0.5 seconds OFF -0.5seconds ON-1.5 seconds OFF
07	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-5.25 seconds OFF	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-5.25 seconds OFF	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-5.25 seconds OFF
08	0.375 seconds ON-0.25 seconds OFF -0.375 seconds ON-2 seconds OFF	0.375 seconds ON-0.25 seconds OFF -0.375 seconds ON-2 seconds OFF	0.375 seconds ON-0.25 seconds OFF -0.375 seconds ON-2 seconds OFF
09	0.25 seconds ON-0.125 seconds OFF -0.25 seconds ON-0.125 seconds OFF -0.25 seconds ON-2 seconds OFF	0.25 seconds ON-0.125 seconds OFF -0.25 seconds ON-0.125 seconds OFF -0.25 seconds ON-2 seconds OFF	0.25 seconds ON-0.125 seconds OFF -0.25 seconds ON-0.125 seconds OFF -0.25 seconds ON-2 seconds OFF
10	1 second ON-4 seconds OFF	1 second ON-4 seconds OFF	1 second ON-4 seconds OFF
11	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-4.25 seconds OFF	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-4.25 seconds OFF	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-4.25 seconds OFF
12	1 second ON-3 seconds OFF	1 second ON-3 seconds OFF	1 second ON-3 seconds OFF
13	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-2.25 seconds OFF	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-2.25 seconds OFF	0.25 seconds ON-0.25 seconds OFF -0.25 seconds ON-2.25 seconds OFF
31◀	1 second ON-2 seconds OFF	2 seconds ON-4 seconds OFF	0.375 seconds ON-0.25 seconds OFF -0.375 seconds ON-2 seconds OFF

Character Code Table

X: Upper digit Y: Lower digit

Y \ X	2	3	4	5	6	7
0		0	@	P	\	p
1	!	1	A	Q	a	q
2	”	2	B	R	b	r
3	#	3	C	S	c	s
4	\$	4	D	T	d	t
5	%	5	E	U	e	u
6	&	6	F	V	f	v
7	'	7	G	W	g	w
8	(8	H	X	h	x
9)	9	I	Y	i	y
A	*	:	J	Z	j	z
B	+	;	K	[k	{
C	,	<	L	¥	l	
D	-	=	M]	m	}
E	.	>	N	^	n	~
F	/	?	O	_	o	←

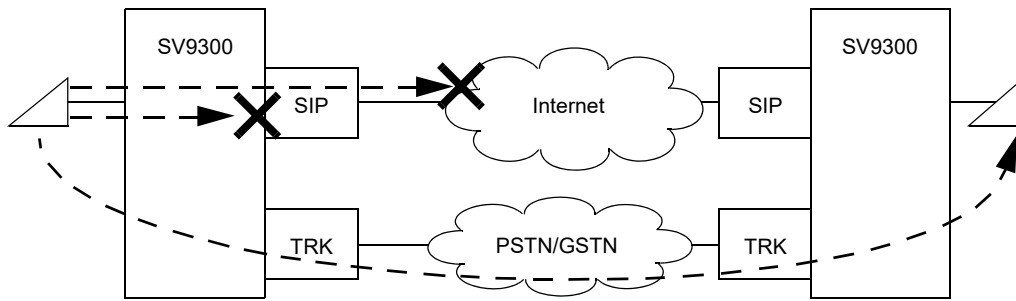
SIP FEATURE PROGRAMMING

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- Congestion Control for SIP Trunk [Page 2-321](#)
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- Fault Registration [Page 2-329](#)
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[9300V5 software required]

Alternative Routing for SIP Trunk

General Description

This feature provides the Alternative Routing (another trunk route) for the SIP trunk when the connected destinations on the IP network are all busy, or an outgoing call via the SIP trunk is not available due to a fault occurrence.



Service Conditions

- Overall Conditions

1. The routing process is executed in the following situations.
 - Time out occurrence
 - SIP error response receive
 - Make busy for IP address **[9300V3 STEP2 software required]**
 - Make busy for trunk route **[9300V3 STEP2 software required]**
2. The tables below shows the routing methods for Alternative Routing.

×: Available –: Not available

Conditions	Connecting Method	P-P Connection	P-MP Connection	
	Alternative Routing Method	Pilot	Pilot	Circular
Routing Process of Alternative Routing	Time out occurrence	×	×	×
	SIP error response receive	×	×	×
	Make busy for IP address	×	×	×
	Make busy for trunk route	–	×	×

3. The maximum number of Alternative Routing for per call origination is six times (seven times including the first call origination).
4. The following trunks can be specified as the Alternative Routing destination:
COT, LDT, ODT, DTI, CCT, PRT, BRT, SIP, IPT (P2P CCIS)
5. SIP trunk call originated from ISDN terminal cannot be rerouted.
6. When using DNS function, the rerouting process does not work.

• Conditions on Time Out Occurrence

1. Specify the response waiting timer when a call is originated (INVITE transaction time-out timer) by CMBA Y=90. Alternative Routing cannot be performed without time-out.
2. With CM41 Y=0>104, set the PBR timer value larger than the response waiting timer set by CMBA Y=90 when a call is originated (INVITE transaction time-out timer) and SV9300 is used as a Gateway.
3. Pilot method or Circular method is available in this Alternative Routing.

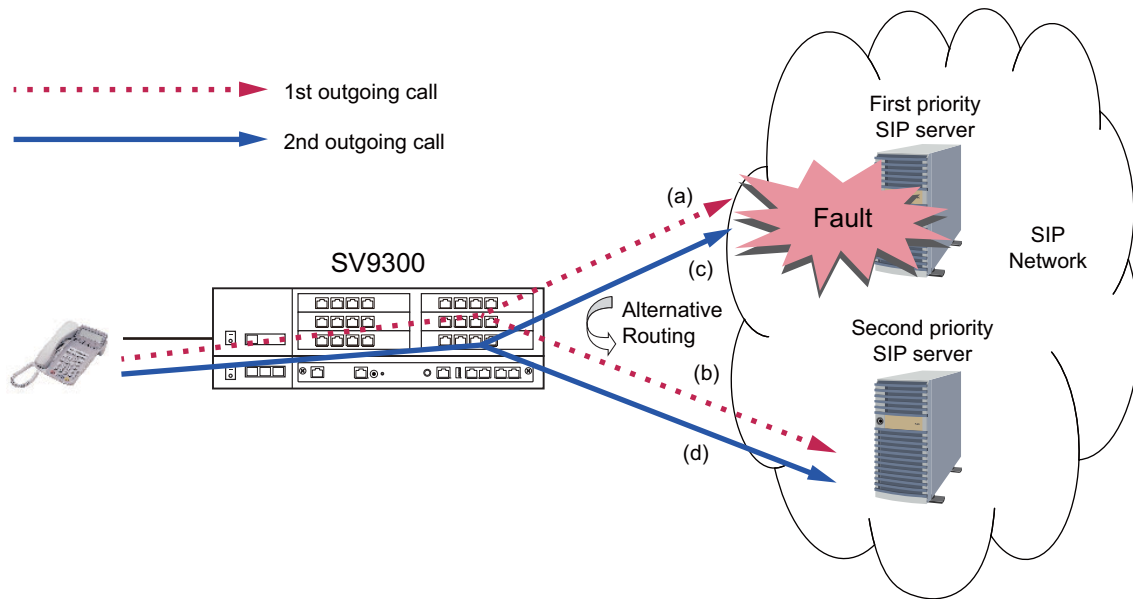
(1) In case of Pilot Method:

The call origination is done to the first priority destination constantly. And the Alternative Routing works to the second priority destination when the fault occurs in the first priority destination.

<An Example of Outgoing Call to SIP Network>

- (a) At the 1st outgoing call, originate a call to the first priority SIP server.
- (b) Due to a fault occurrence, the Alternative Routing works to the second priority SIP server.
- (c) At the 2nd outgoing call, originate a call to the second priority SIP server.
- (d) Due to a fault occurrence, the Alternative Routing works to the second priority SIP server.

The Image of Alternative Routing by Pilot Method



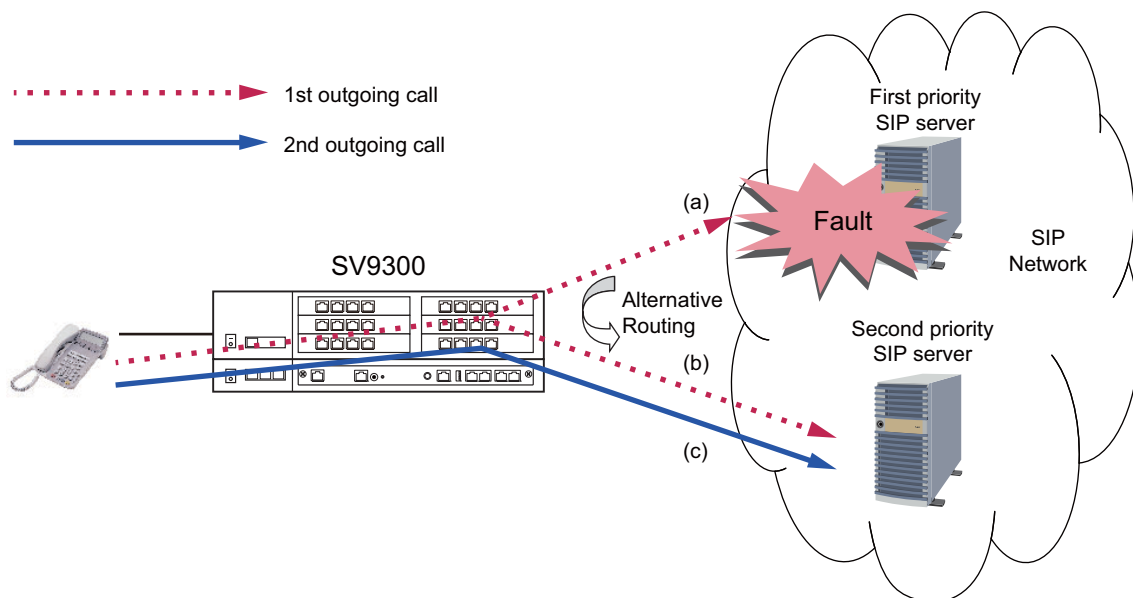
(2) In case of Circular Method:

After working the Alternative Routing, the next outgoing call is done to the rerouted destination. And when a fault occurs in the rerouted destination, the Alternative Routing works to the next priority destination.

<An Example of Outgoing Call to SIP Network>

- (a) At the 1st outgoing call, originate a call to the first priority SIP server.
- (b) Due to a fault occurrence, the Alternative Routing works to the second priority SIP server.
- (c) At the 2nd outgoing call, originate a call to the second priority SIP server.

The Image of Alternative Routing by Circular Method



- Conditions on SIP Error Response Receive

1. Alternative Routing is available when the following error response code is received.
 - 408 Request Timeout (CMBA Y=115: 0)
 - 480 Temporarily Unavailable (CMBA Y=132: 1)
 - 486 Busy Here (CM08>372: 0, CMBA Y=111: 1)
 - 503 Service Unavailable (CM08>372: 0, CMBA Y=148: 1)
2. The Alternative Routing does not work when receiving an SIP error response before receiving 180 Ringing or 183 Session Progress.
3. Pilot method or Circular method is available in this Alternative Routing.
(As for the details of Pilot method or Circular method, refer to the item 3 of Conditions on Time Out Occurrence.)

- Conditions on Make busy of Destination IP Address **[9300V3 STEP2 software required]**
 1. The trunk which enables to make busy of the destination IP address (assigned by CM5B Y=01) is a SIP trunk only.
 2. Although outgoing calls for the trunk route or destination IP address in a make busy status are restricted, incoming calls are not restricted.
 3. When a call is originated to a SIP server of the destination IP address which is made busy, the Alternative Routing works if the Alternative Routing setting is assigned.

<An Example of Outgoing Call to SIP Network>

* It is assumed that the setting of Alternative Routing is provided.

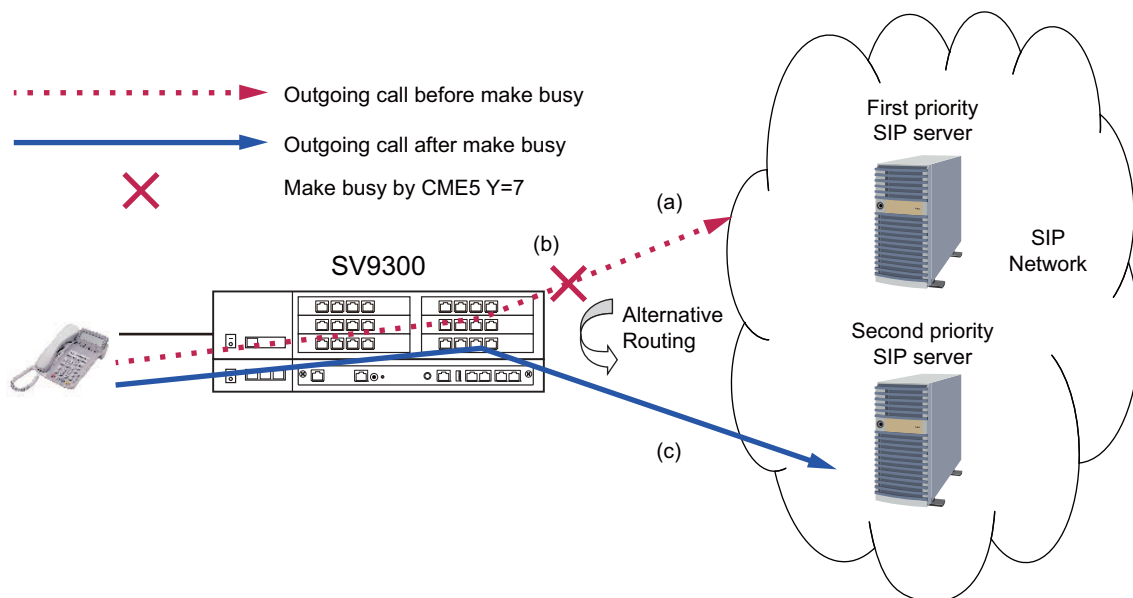
(a) At the outgoing call before make busy, originate a call to the first priority SIP server.

(b) Make busy the destination IP address to the first priority SIP server.

(c) At the outgoing call after make busy, originate a call to the second priority SIP server.

* When the make busy is released, originate a call to the first priority SIP server.

The Image of Alternative Routing in Make Busy



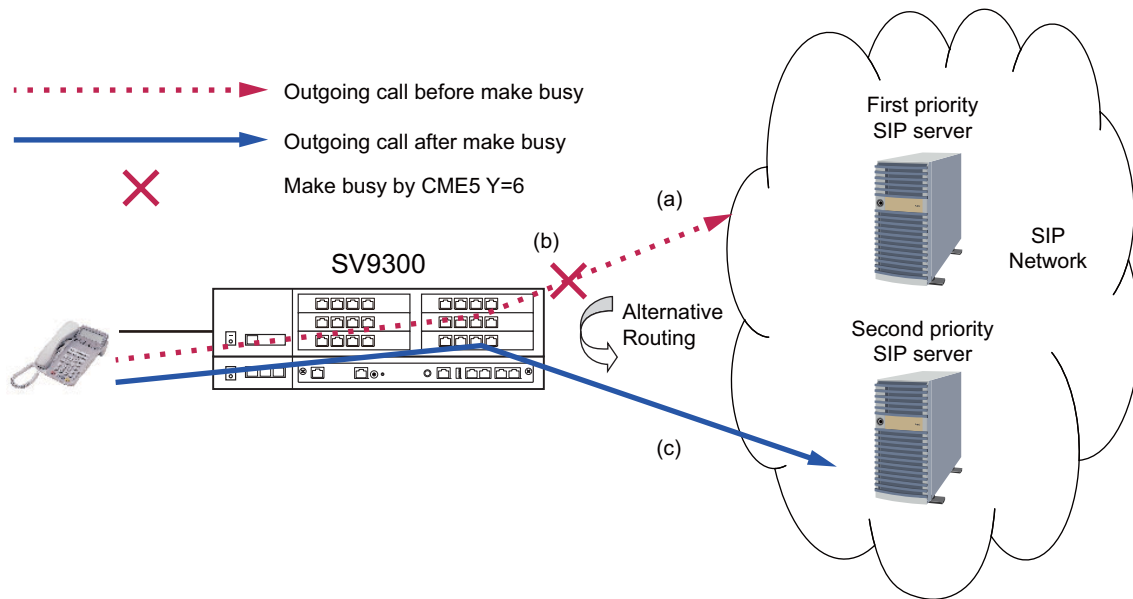
- Conditions on Make busy of Outgoing Trunk Route **[9300V3 STEP2 software required]**
 1. Make busy of outgoing trunk routes is available for all trunk types.
 2. Although outgoing call for an outgoing trunk route in a make busy status is restricted, an incoming calls is not restricted.
 3. When a call is originated to the trunk route which is made busy, the Alternative Routing works if the Alternative Routing setting is assigned.

<An Example of Outgoing Call to SIP Network>

* It is assumed that the setting of Alternative Routing is provided.

- (a) At the outgoing call before make busy, originate a call to the first priority SIP server.
 - (b) Make busy the outgoing trunk route to the first priority SIP server.
 - (c) At the outgoing call after make busy, originate a call to the second priority SIP server.
- * When the make busy is released, originate a call to the first priority SIP server.

The Image of Alternative Routing in Make Busy



4. PEG count (the number of outgoing calls/busy calls) to the trunk route which is made busy is not performed.

Programming

This section shows the programming of Alternative Routing for SIP trunk.

- Time Out Occurrence [Page 2-304](#)
- SIP Error Response Receive [Page 2-305](#)
- Make Busy of Destination IP Address [Page 2-306](#)
- Make Busy of Outgoing Trunk Route [Page 2-306](#)

(1) Basic Data Assignment for Alternative Routing

START	DESCRIPTION	DATA
CM35	<p>Provide the Alternative Routing for SIP.</p> <p>For a tandem connection, provide the Alternative Routing for a tandem call.</p> <p>NOTE: <i>Set this data for the incoming trunk route.</i></p>	<ul style="list-style-type: none"> • Y=186 (1) 00-63: Trunk Route No. (2) 0: To provide <ul style="list-style-type: none"> • Y=192 (1) 00-63: Trunk Route No. (2) 0: To provide
CM08	<p>Provide the system with Alternative Routing when lack of VoIPDB channel in tandem office.</p> <p>Provide the system with Alternative Routing when no answer timer of outgoing call (T1 timer) time-out occurs in tandem connection.</p>	<ul style="list-style-type: none"> (1) 624 (2) 0: To provide (CGC setting) <ul style="list-style-type: none"> (1) 666 (2) 0: To provide
CM29	Assign a Numbering Plan Group to each tenant.	<ul style="list-style-type: none"> (1) 00-63: Tenant No. (2) 710-713 : Numbering Plan Group 0-3 NONE◀: Numbering Plan Group 0
CM20	Assign the access code for LCR Group 0-3.	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A126: LCR Group 0 A127: LCR Group 1 A128: LCR Group 2 A129: LCR Group 3
CM90	Assign the service access key for LCR Group 0-3 to Multiline Terminal, if required.	<ul style="list-style-type: none"> • Y=00 (1) My Line No. + <input type="text"/> + Key No. (2) F0A26-F0A28: LCR Group 0-3
A		

A

CM8A

DESCRIPTION

DATA

Assign an Area Code Development Pattern number to each LCR Group.

- Y=A000
- (1) 0-3: LCR Group 0-3
- (2) 4005-4007: Area Code Development Pattern No. 5-7

Assign a Route Pattern number to each Area Code for the Area Code Development Pattern number assigned by CM8A Y=A000.

- Y=4005-4007 Area Code Development Pattern No. 5-7
- (1) X...X: Area Code, Maximum 8 digits
- (2) 0000-0255: Route Pattern No. 000-255

Specify the order of LCR selection for the Route Pattern number assigned by CM8A Y=4005-4007.

- Y=0000-0255 Route Pattern No. 000-255
- (1) 1-4: Order of LCR Selection
 - 1: 1st
 - 2: 2nd
 - 3: 3rd
 - 4: 4th
- (2) XXX ZZ
 - XXX: 000-255: LCR Pattern No.
 - ZZ : 00-63: Trunk Route No.

NOTE 1: When providing Alternative Routing, the order of route selection for secondary SIP server should be assigned to 2nd order.

NOTE 2: You cannot assign CM8A Y=0000-0255 1st data 0 (Destination of next table) and 1st data 4 (4th LCR selection) simultaneously. The latest data assignment including system data memory clear is effective. The other assigned data is set as NONE.

Example:

Y	1ST DATA		2ND DATA	ORDER OF LCR SELECTION
0000	0	Destination of next table	0001	
	1	1st LCR selection	00000	1st LCR selection
	2	2nd LCR selection	00101	2nd LCR selection
	3	3rd LCR selection	00202	3rd LCR selection
	4	4th LCR selection	NONE (Unavailable)	
0001	0	Destination of next table	NONE	
	1	1st LCR selection	00303	4th LCR selection
	2	2nd LCR selection	00404	5th LCR selection
	3	3rd LCR selection	00505	6th LCR selection
	4	4th LCR selection	00606	7th LCR selection

B

B	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: 40px; margin: 0 auto;">CM8A</div>	<p>NOTE 3: <i>The order of route selection can be expanded up to 7th order. To expand the order of route selection, assign CM8A Y=0000-0255>0 on next page without assigning the 4th order.</i></p> <p>NOTE 4: <i>When providing the Alternative Routing by Pilot method in Point-to-Multipoint Connection, assign the different LCR Pattern No. to each order of LCR selection.</i></p> <p>NOTE 5: <i>When providing the Alternative Routing by Circular method in Point-to-Multipoint Connection, assign the same LCR Pattern No. to each order of LCR selection.</i></p> <p>Assign the next Route Pattern number to the Route Pattern number assigned by CM8A Y=4005-4007.</p> <p>NOTE: <i>This data is required to expand the order of LCR selection that is assigned by CM8A Y=0000-0255. The order of LCR selection can be expand to the maximum seventh.</i></p>	<ul style="list-style-type: none"> • Y=0000-0255 Route Pattern No. 000-255 (1) 0: Designation of next table (2) 0000-0255: Next Route Pattern No. 000-255
<div style="border: 1px solid black; padding: 2px; width: 40px; margin: 0 auto;">C</div> <p>When providing the Alternative Routing by Pilot method in Point-to-Point Connection</p>	<div style="border: 1px solid black; padding: 2px; width: 40px; margin: 0 auto;">D</div> <p>When providing the Alternative Routing by Pilot/ Circular method in Point-to-Multipoint Connection</p>	

To provide the Alternative Routing by Pilot method in Point-to-Point Connection:

C	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: 40px; margin: 0 auto;">CMBA</div>	<p>Assign the IP Address for the SIP server.</p> <p>NOTE: <i>When assigning this command, clear the data of CM8A Y=5XXX>167 (CM8A Y=5XXX>167: CCC).</i></p>	<ul style="list-style-type: none"> • Y=30 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXXXXXXXXXXXX: 000000000001-255255255254: SIP Server IP Address (12 digits) NONE◀: No data
<div style="border: 1px solid black; padding: 2px; width: 40px; margin: 0 auto;">END</div>		

To provide the Alternative Routing by Pilot/Circular method in Point-to-Multipoint Connection:

D	DESCRIPTION	DATA
CM8A	Assign the Destination IP Address Pattern number to the LCR Pattern number.	<ul style="list-style-type: none"> • Y=5000-5255 Pattern No. 000-255 (1) 167: Destination IP Address Pattern (2) 000-255: IP Address Pattern No.
	NOTE 1: Set this command only for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").	
	NOTE 2: When assigning this command, clear the data of CMBA Y=30 (CMBA Y=30: CCC).	
CM5B	Assign the Destination IP Address for SIP trunk.	<ul style="list-style-type: none"> • Y=01 Destination IP Address for SIP Trunk (1) XXX ZZ XXX: 000-255: IP Address Pattern No. assigned by CM8A Y=5000-5255> 167 ZZ : 00-07: IP Address No. 00-07 (2) XXXXXXXXXXXXX: 000000000001-255255255254: Destination IP Address of opposite SIP Trunk (Maximum 12 digits)
	NOTE 1: Set this command only for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").	
	NOTE 2: IP address number (00-07) is the number for specifying the IP addresses of the SIP trunks of opposite office (SV9300/SV8300/NEAX 2000 IPS) which has plural SIP trunks, when the offices are connected with Point-to-Multipoint.	
	NOTE 3: This command should be assigned from the leading number of 1st data (IP Address Pattern No. + IP Address No.).	
END		

(2) Time Out Occurrence Data Assignment

In addition to the (1) Basic Data Assignment for Alternative Routing, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	<p>Specify the response waiting timer when a call is originated (INVITE transaction timeout timer).</p> <p>NOTE 1: <i>Set the timer value until receiving the response from the called party in a call origination. If the time-out occurs, the system regards it as the network fault occurrence, and executes the Alternative Routing.</i></p> <p>NOTE 2: <i>To provide a tandem connection to CCIS, set the value of second data that does not exceed the value of PBR timer assigned by CM41 Y=0>104.</i></p> <p>NOTE 3: <i>It is recommended to set the second data to 03 for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>Specify the resend INVITE message to a call before alternative routing because of sending time-out.</p>	<ul style="list-style-type: none"> • Y=90 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 00 : No Time Out 01-30: 1 second-60 seconds (2 second increments) 31◀ : 32 seconds <ul style="list-style-type: none"> • Y=146 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To stop 1◀ : To continue
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM41</div>	<p>Assign the PBR timer when establishing tandem connection.</p> <p>NOTE 1: <i>This data requires changing the timer value depending on the number of Alternative Routing. Set the timer value by the following calculating formula. PBR timer value = (timer value of CMBA Y=90 [sec.] × (number of Alternative Routing [Max. 7 times] + 1) + 1 (sec.)</i></p> <p>NOTE 2: <i>Usually, it is recommended to set the second data to 33 (33 sec.).</i></p>	<ul style="list-style-type: none"> • Y=0 (1) 104 (2) 02-99: 2-99 seconds (1 second increments) If no data is set, the default setting is 7 seconds.
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>		

(3) SIP Error Response Receive Data Assignment

In addition to the (1) Basic Data Assignment for Alternative Routing, do the following programming.

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM08</div>	<p>Provide the system with Alternative Routing of CCIS outgoing call when outgoing trunks of tandem office are all busy or when receiving an error response from an opposing device after a station originates a SIP Trunk call.</p> <p>NOTE 1: <i>When 0 (Available) is set by this command, an alternative routing is performed at receiving an error response “486 Busy Here” or “503 Service Unavailable”.</i></p> <p>NOTE 2: <i>Whether to perform an alternative routing at receiving an error response “486 Busy Here” can be by CMBA Y=111.</i></p> <p>NOTE 3: <i>Whether to perform an alternative routing at receiving an error response “503 Busy Here” can be by CMBA Y=148.</i></p>	<p>(1) 372 (2) 0: To provide</p>
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	<p>Specify whether to provide an alternative routing when receiving the 486 Busy Here response.</p> <p>Provide an alternative routing when receiving the 408 Request time-out response.</p> <p>NOTE: <i>This data is not available for Point-Multipoint connection (when CMA7 Y=46 is set to “0”).</i></p> <p>Provide an alternative routing when receiving the 480 Temporarily Unavailable.</p> <p>Provide alternative routing when receiving the 503 Service Unavailable.</p>	<ul style="list-style-type: none"> • Y=111 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : No provided 1◀: To provide <ul style="list-style-type: none"> • Y=115 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : No provided 1◀: To provide <ul style="list-style-type: none"> • Y=132 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 1◀: To provide <ul style="list-style-type: none"> • Y=148 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 1◀: To provide
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">END</div>		

(4) Make Busy of Destination IP Address Data Assignment
[9300V3 STEP2 software required]

To set make busy to the destination IP address pattern number assigned by CM5B Y=01, do the following programming.

NOTE: *To provide Alternative Routing when a call is originated to a SIP server of the IP address which is made busy, do the programming of “(1) Basic Data Assignment for Alternative Routing”.* [☞ Page 2-300](#)

START	DESCRIPTION	DATA
START <div style="border: 1px solid black; padding: 2px; display: inline-block;">CME5</div> END	Set make busy to the IP address pattern number assigned by CM5B Y=01.	<ul style="list-style-type: none"> • Y=7 (1) XXX ZZ XXX: 000-255: IP Address Pattern No. ZZ : 00-07: IP Address No. (2) 0: Make busy set

(5) Make Busy of Outgoing Trunk Route Data Assignment
[9300V3 STEP2 software required]

To set make busy to the outgoing trunk route number assigned by CM30 Y=00, do the following programming.

NOTE: *To provide Alternative Routing when a call is originated to a SIP server of the IP address which is made busy, do the programming of “(1) Basic Data Assignment for Alternative Routing”.* [☞ Page 2-300](#)

START	DESCRIPTION	DATA
START <div style="border: 1px solid black; padding: 2px; display: inline-block;">CME5</div> END	Set make busy to the outgoing trunk route number assigned by CM30 Y=00.	<ul style="list-style-type: none"> • Y=6 (1) 00-63: Trunk Route No. (2) 0: Make busy set

Caller ID Display

General Description

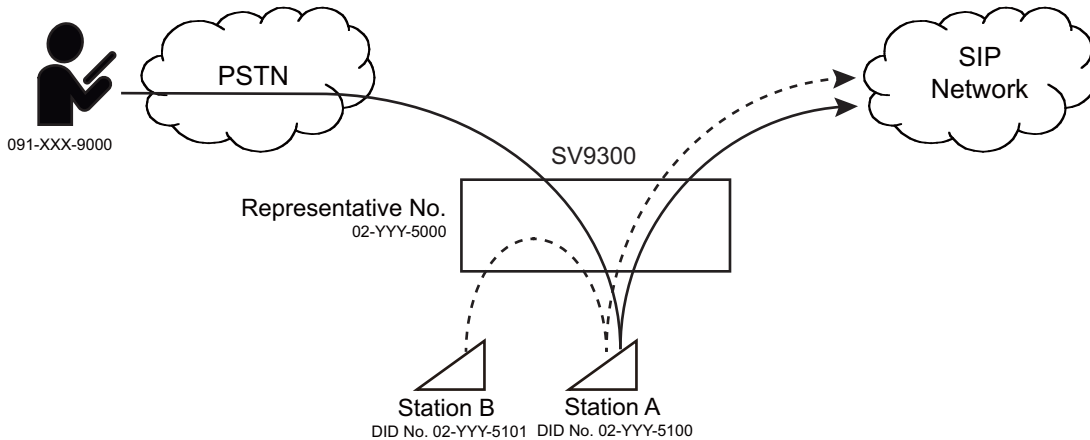
This feature provides Caller ID Display to the called station. By this feature, a caller number informed from the network can be displayed on the LCD of Multiline Terminal.

Service Conditions

- Overall Conditions
 1. The number of the dial number to possess and display is the number of the dial number when Direct Inward Dialing (DID) is contracted.
 2. Only the contracted pilot number can be displayed.
 3. Optional number or pilot number is sent to the called station.
 4. SIP message with the following header is sent for Caller ID Display.
E.G. Caller number + 0123456
From : "+0123456" <sip:+0123456@195.2.174.100:5060>;tag=XXXXXXXXXXXXXXXX-X
Contact: <sip:+0123456@221.186.89.197:5060>
 5. Only 0-9, *, # can be displayed. The calling number is not informed when alphabet is informed. However, 0-9, *, # is displayed with converting by the function of PBX.
 6. Set CM08>379 to "1".

• Conditions on Caller ID Notification of Incoming Call Forwarded to SIP Network

1. When an incoming call terminated to the station which sets Call Forwarding-Outside to the SIP network, the Caller ID is transferred to the SIP network as follows.



Incoming Call from		Caller ID Notified to SIP Network
PSTN	Station A sets Call Forwarding-Outside to SIP Network	091-XXX-9000 (Original Caller ID)
Station B		02-YYY-5000 (Representative No.) 02-YYY-5100 (DID No. of Station A) NOTE
		02-YYY-5101 (DID No. of Station B) 02-YYY-5000 (Representative No.) 02-YYY-5100 (DID No. of Station A) NOTE

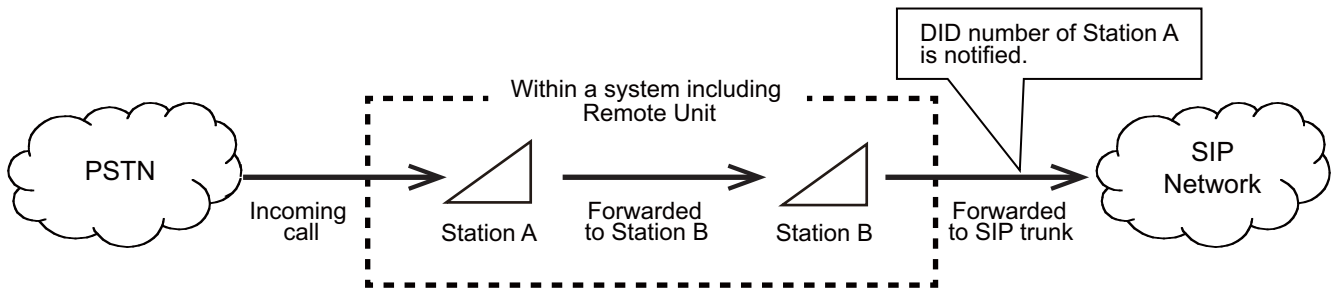
NOTE: Notifying the DID number of Station A is available from 9300V7 or later software.

2. Conditions for notifying the DID Number of forwarding station to SIP Network are as follows.

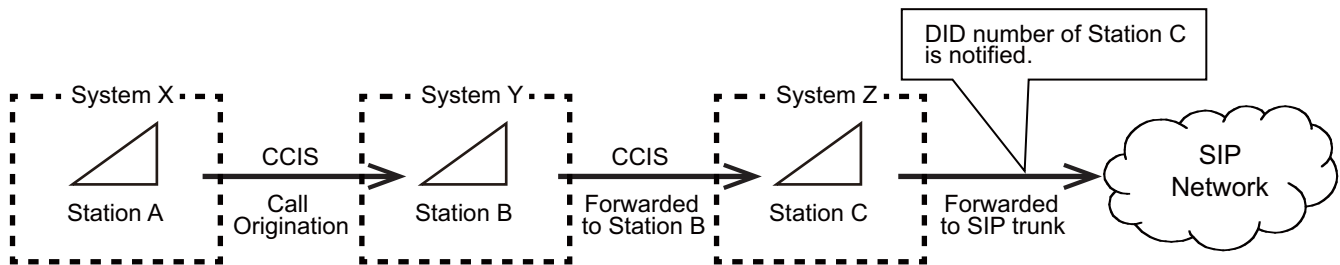
[9300V7 software required]

- This feature is available for all types of incoming trunk calls.
- The outgoing trunk is available for SIP trunk only.
- The types of available Call Forwarding features are as follows.
Call Forwarding-All Calls/Busy Line/No Answer
- The Call Forwarding-Outside can be set to the types of station including virtual station except for ISDN terminal and Attendant Console.

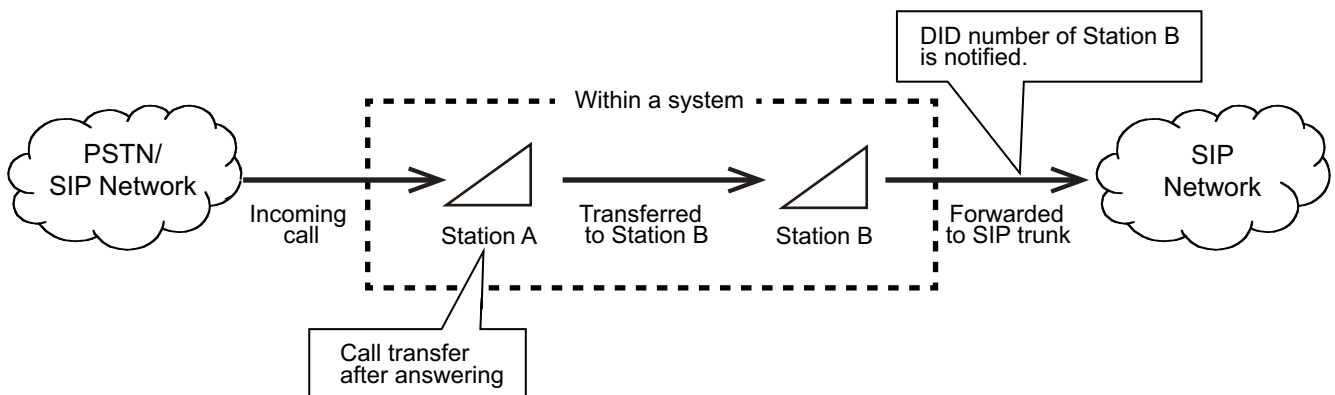
- When a call is forwarded multiple times in a system (including Remote Unit) and then forwarded to SIP network, the DID number of first forwarding station is notified to the SIP network.



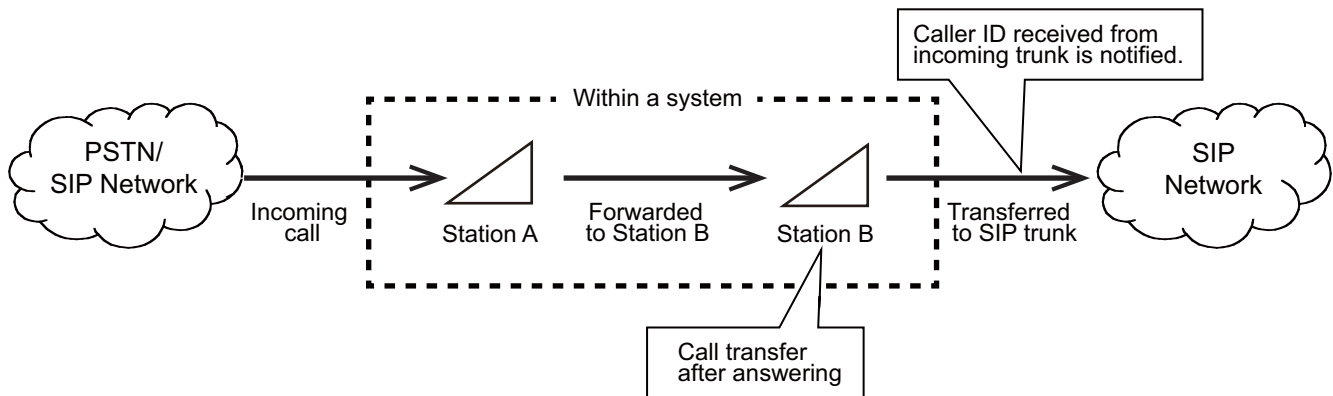
- When a call is forwarded multiple times between systems via CCIS network and then forwarded to SIP network, the DID number of last forwarding station is notified to the SIP network.



- When this feature is set in combination with the feature “CLI Transparency by transferring”, there are the following conditions.
 - When a call is automatically forwarded to SIP trunk by Call Forwarding-Outside, this feature has priority.



- When a call is manually transferred to SIP trunk by Call Transfer, the feature “CLI Transparency by transferring” has priority.



- The DID number notified to SIP network can be selected by the following system data.
 - CM08>1242: When a call is terminated from a trunk to a station and forwarded to SIP trunk by Call Forwarding-All Calls /Busy Line /No Answer /Logout
 - CM08>1243: When a call is originated from Station A to Station B and forwarded to SIP trunk by Call Forwarding-All Calls /Busy Line /No Answer /Logout

Programming

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM08</div>	<p>Provide the system with Caller ID Display service.</p>	<p>(1) 379 (2) 1◀: To provide</p>
	<p>Provide the sending of calling station number or calling party number to the analog telephone for Caller ID-Station when an incoming call is terminated via SIP Trunk.</p>	<p>(1) 603 (2) 0: To provide</p>
	<p>NOTE: <i>The sending of a calling station number to the analog telephone for Caller ID-Station is effective when CM08>507 is set to 1.</i></p>	
<p>Select the reference to Caller ID when an incoming call is received to SIP trunk and “anonymous” is set to From header Displayname field or Username field.</p>	<p>(1) 1202 (2) 0 : As per CMBA Y=126 1◀: Caller ID is not informed</p>	
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM12</div>	<p>Assign the ISDN/SIP Subscriber number 1-4 for each line to be used.</p>	<ul style="list-style-type: none"> • Y=12 • Y=46 (1) X-XXXXXXXX: Station No. (2) X-XXXX: ISDN/SIP Subscriber No. (In-dial No. [1-4 digits])
	<p>NOTE: <i>Assign the Local Office Code + Subscriber No. (1-16 digits) for the second data.</i></p>	<ul style="list-style-type: none"> • Y=51 • Y=52 (1) X-XXXXXXXX: Station No. (2) X-XXXX: ISDN/SIP Subscriber No. (In-dial No. [1-16 digits]) NOTE
	<p>Assign the ISDN/SIP Local Office Code Table number 1/2 when CM12 Y=12/46 is assigned.</p>	<ul style="list-style-type: none"> • Y=13 • Y=47 (1) X-XXXXXXXX: Station No. (2) 00-14: ISDN/SIP Local Office Code Table number 00-14
<p>NOTE: <i>This command is assigned in combination with CM12 Y=12/46 as follows.</i> CM12 Y=12 - CM12 Y=13 CM12 Y=46 - CM12 Y=47</p>		
<div style="border: 1px solid black; padding: 2px; width: 20px; height: 20px; margin: 0 auto; display: flex; align-items: center; justify-content: center;">A</div>		

A	DESCRIPTION	DATA
CM12	Specify Caller ID method when CM12 Y=12, 46, 51, 52 has not been set up.	<ul style="list-style-type: none"> (1) Y=56 (2) X-XXXXXXXXX: Station No. 0 : Representative Number assigned by CMBA Y=32 1◀: Not informed
CM50	Assign the ISDN/SIP Local Office Code Table 2. NOTE: <i>A toll number can be included in a ISDN/SIP Local Office Code.</i>	<ul style="list-style-type: none"> • Y=05 (1) 00-14: ISDN/SIP Local Office Code Table number 00-14 (2) X-XXXXXXXXXXXXX: ISDN/SIP Local Office Code NONE◀: No data
CM8A	Assign an Area Code Development Pattern number. Assign a Route Pattern number to each area code for the Area Code Development Pattern number assigned by CM8A Y=A000. Specify the order of LCR selection for the Route Pattern number assigned by CM8A Y=4000-4007.	<ul style="list-style-type: none"> • Y=A000 (1) 0-3: LCR Group 0-3 (2) 4000-4007: Area Code Development Pattern No. 0-7 • Y=4000-4007 Area Code Development Pattern No. 0-7 (1) X-XXXXXXXXX: Area Code, Maximum 8 digits (2) 0000-0255: Route Pattern No. 000-255 • Y=0000-0255 Route Pattern No. 000-255 (1) 1-4: Order of LCR Selection 1: 1st 2: 2nd 3: 3rd 4: 4th (2) XXX ZZ XXX: 000-255: LCR Pattern No. ZZ : 00-63: Trunk Route No.
B		

B	DESCRIPTION	DATA
CMA7	Provide the SIP trunk number with Caller ID Display service.	<ul style="list-style-type: none"> • Y=26 (1) 00-63: SIP Trunk No. 00-63 (2) 01◀: To provide
	Provide the SIP trunk number with Calling Party Information transferring service.	<ul style="list-style-type: none"> • Y=28 (1) 00-63: SIP Trunk No. 00-63 (2) 01◀: To provide
	Specify Originating Office number for SIP Trunk number.	<ul style="list-style-type: none"> • Y=06 (1) 00-63: SIP Trunk No. 00-63 (2) X-XXXX: Originating Office number
CM08	Specify Caller ID method for SIP Trunk.	<ul style="list-style-type: none"> (1) 1220 (2) 0 : Old method (As per CM8A Y=5XXX>176/CMBA Y=44) 1◀: New method (As per CM8A Y=5XXX>186/CMBA Y=160)
	<p>NOTE: <i>When using the same Caller ID method as that of SV8300, set the second data to "0".</i></p>	
C		

C	DESCRIPTION	DATA
CM8A	<p>Specify Calling party number sent from SIP Trunk.</p> <p>(NEW METHOD)</p> <p>NOTE 1: <i>This data is effective only when CM08>1220 is set to "1". If CM08>1220 is set to "0", the data assigned by CMBA Y=5000-5255>176 shall be effective.</i></p> <p>NOTE 2: <i>When the second data is set to "01"- "04" and no In-dial number is assigned for the notification, the second data assigned by CM12 Y=56 (0: Representative No./1: Not informed) shall be effective.</i></p> <p>NOTE 3: <i>Originating Office number is assigned by CMA7 Y=06.</i></p> <p>NOTE 4: <i>"Not informed" is set when CMA7 Y=46 2nd data=1 (Point-to-Point connection) is assigned, and "Station number" is set when CMA7 Y=46 2nd data=0 (Point-to-Multipoint connection).</i></p> <p>NOTE 5: <i>"Not informed" is set when CMA7 Y=46 2nd data =1" (Point-to-Point connection) is assigned, and "Station number (including Originating Office number)" is set when CMA7 Y=46 2nd data=0 (Point-to-Multipoint connection) is assigned.</i></p>	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-256 (1) 186 (2) 00 : Not informed 01 : As per CM12 Y=12/13 NOTE 2 02 : As per CM12 Y=46/47 NOTE 2 03 : As per CM12 Y=51 NOTE 2 04 : As per CM12 Y=52 NOTE 2 08 : Representative Number assigned by CMBA Y=32 12 : Station number 13 : Station number (including Originating Office number) NOTE 3 14 : Not informed/Station Number NOTE 4 15◀ : Not informed/Station number (including Originating Office number) NOTE 3, NOTE 5
D		

D	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CM8A</div>	<p>(OLD METHOD)</p> <p>NOTE 1: <i>As for the calling number informed by the combination of CM8A Y=5000-5255>176 and CMBA Y=44, see NOTE 2 of CMBA Y=44.</i> Page 2-318</p> <p>NOTE 2: <i>This data is effective only when CM08>1220 is set to "0". If CM08>1220 is set to "1", the data assigned by CMBA Y=5000-5255>186 shall be effective.</i></p>	<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 176 (2) 00 : As per Caller ID conversion mode assigned by CMBA Y=44 01 : As per Caller ID conversion mode assigned by CMBA Y=44 02 : As per Caller ID conversion mode assigned by CMBA Y=44 08 : To send Representative No. assigned by CMBA Y=32 14 : To send Station No. without Originating Office No. assigned by CMA7 Y=06. 15◀ : To send Station No. with Originating Office No. assigned by CMA7 Y=06.
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	<p>Specify Caller ID conversion in SIP trunk tandem connection.</p> <p>(NEW METHOD)</p> <p>NOTE 1: <i>This data is effective when CM08>1220: 1 is assigned.</i></p> <p>NOTE 2: <i>Display of a calling party number sent from SIP Trunk depends on the Caller ID method assigned by CM08>1220 and the data combination for the method as shown below.</i></p>	<ul style="list-style-type: none"> • Y=160 (1) 00-63: Profile number (2) 00 : Caller ID conversion mode 0 01 : Caller ID conversion mode 1 02 : Caller ID conversion mode 2 03 : Caller ID conversion mode 3 15◀ : Not informed
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">E</div>		

E

DESCRIPTION

DATA

CMBA

When the call is originated from a station accommodated in SV9300

◀: Default

CM8A Y=5000-5255>186	CM12 Y=12/46/51/52	CM12 Y=56 CMA7 Y=46	Calling Party Number to be informed
00	–	–	Not informed
01	Subscriber number is assigned by CM12 Y=12	–	As per CM12 Y=12/13
	No Subscriber number is assigned by CM12 Y=12	CM12 Y=56: 0	As per CMBA Y=44
		CM12 Y=56: 1	Not informed
02	Subscriber number is assigned by CM12 Y=46	–	As per CM12 Y=46, 47
	No Subscriber number is assigned by CM12 Y=46	CM12 Y=56: 0	As per CMBA Y=44
		CM12 Y=56: 1	Not informed
03	Subscriber number is assigned by CM12 Y=51	–	As per CM12 Y=51
	No Subscriber number is assigned by CM12 Y=51	CM12 Y=56: 0	As per CMBA Y=44
		CM12 Y=56: 1	Not informed
04	Subscriber number is assigned by CM12 Y=52	–	As per CM12 Y=52
	No Subscriber number is assigned by CM12 Y=52	CM12 Y=56: 0	As per CMBA YY=44
		CM12 Y=56: 1	Not informed
08	–	–	Representative Number assigned by CMBA Y=32
12	–	–	Station number
13	–	–	Station number (CMA7 Y= 06 is included)
14	–	CMA7 Y=46: 0	Station number
		CMA7 Y=46: 1	Not informed
15◀	–	CMA7 Y=46: 0	Station number (CMA7 Y= 06 is included)
		CMA7 Y=46: 1	Not informed

F

F

DESCRIPTION

DATA

CMBA

When using SV9300 as a gateway (Tandem connection)

◀: Default

INCOMING CALLING NUMBER	CMBA Y=160					REMARKS
	00 (Caller ID conversion mode 0)	01 (Caller ID conversion mode 1)	02 (Caller ID conversion mode 2)	03 (Caller ID conversion mode 3)	15◀ (Not informed)	
Not informed	Representative Number assigned by CMBA Y=32	Representative Number assigned by CMBA Y=32	Not informed	Not informed	Not informed	
Informed	Representative Number assigned by CMBA Y=32	Original caller ID/ DID Number NOTE	Original caller ID/ DID Number NOTE	Representative Number assigned by CMBA Y=32	Not informed	

NOTE: When an Incoming Calling is FMC, a prefix code is added to the calling number (as set by CM35 Y=308, 311).

(OLD METHOD)

RESET

NOTE 1: This data is effective when CM08> 1220: 0 is assigned.

- Y=44
- (1) 00-63: Profile number 00-63
- (2) 00 : Caller ID conversion mode 0
- 01 : Caller ID conversion mode 1
- 02 : Caller ID conversion mode 2
- 03 : Caller ID conversion mode 3
- 15◀: Not provided

G

G

CMBA

DESCRIPTION

DATA

NOTE 2: *Display of Calling Party Number Sent from SIP Trunk*

Display of a calling party number sent from SIP Trunk depends on the Caller ID method assigned by CM08>1220 and the data combination for the method as shown below.

When the call is originated from a station accommodated in SV9300

◀: Default

CMBA Y=5XXX> 176	CMBA Y=44					REMARKS
	00 (Caller ID conversion mode 0)	01 (Caller ID conversion mode 1)	02 (Caller ID conversion mode 2)	03 (Caller ID conversion mode 3)	15◀ (Not provided)	
00	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Not informed	Representative Number by CMBA Y=32	Not informed	
01	Representative Number by CMBA Y=32	DID Number 1	DID Number 1	DID Number 1	DID Number 1	Subscriber number set to CM12 Y=12, 13
	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Station Number	Representative Number by CMBA Y=32	Station Number	No Subscriber number set to CM12 Y=12, 13
02	Representative Number by CMBA Y=32	DID Number 2	DID Number 2	DID Number 2	DID Number 2	Subscriber number set to CM12 Y=46, 47/51, 52
	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Station Number	Representative Number by CMBA Y=32	Station Number	No Subscriber number set to CM12 Y=46, 47/51, 52
08	Representative Number by CMBA Y=32					
14	Station Number (without Originating Office number by CMA7 Y=06)					
15◀	Station Number (with Originating Office number by CMA7 Y=06)					

H

H

DESCRIPTION

DATA

CMBA

When using SV9300 as a gateway (Tandem connection)

◀: Default

INCOMING CALLING NUMBER	CMBA Y=44				
	00 (Caller ID conversion mode 0)	01 (Caller ID conversion mode 1)	02 (Caller ID conversion mode 2)	03 (Caller ID conversion mode 3)	15◀ (Not provided)
Not informed	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Not informed
Informed	Representative Number by CMBA Y=32	Original caller ID/DID Number	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Original caller ID/DID Number

When CM08>1202 is set to "0" (as per CMBA Y=126), select the reference to Caller ID.

NOTE: Set this data to "1" to display a calling number and a calling name at the same time.

- Y=126
 - (1) 00-63: Profile No. assigned by CMA7 Y=71
 - (2) 0 : Get caller ID from the Username field if the Displayname field of the From header of initial INVITE message is blank
 - 1 : Get caller ID from the Username field of the From header of initial INVITE message only.
 - 3◀: Get caller ID from the Displayname field of the From header of initial INVITE message only

CM35

Set Calling Name information transfer to ISDN on tandem call from CCIS for voice channel of SIP trunk route.

- Y=145
 - (1) 00-63: Trunk Route No. 00-63
 - (2) 0 : To provide
 - 1◀: Not provided

I

I

CM08

DESCRIPTION

DATA

Specify Caller ID sent to SIP trunk, when a call is terminated from a trunk to a station and forwarded to SIP trunk by Call Forwarding-All Calls /Busy Line/No Answer/Logout features.
[9300V7 software required]

- (1) 1242
- (2) 0 :SIP subscriber number of Station (forwarding station)
- 1◀: Caller ID notified from incoming trunk or Representative Number

NOTE 1: When the second data of this command is set to “0”, the Caller ID sent to the SIP trunk is as follows.

- For a new method (assigned by CM08>1220>1): SIP subscriber numbers assigned by CM8A Y=5XXX>186: 01-04 are sent.
- For an old method (assigned by CM08>1220>0): SIP subscriber numbers assigned by CM8A Y=5XXX>176: 01-02 are sent.

NOTE 2: When the second data of this command is set to “1”, the Caller ID sent to SIP trunk depends on the settings of CM08>1220 and CMBA Y=160, or CM08>1220 and CMBA Y=44.

Specify Caller ID sent to SIP trunk, when a call is originated from Station A to Station B and forwarded to SIP trunk by Call Forwarding-All Calls /Busy Line/No Answer/Logout features.
[9300V7 software required]

- (1) 1243
- (2) 0 :SIP subscriber number of Station (forwarding station)
- 1◀: SIP subscriber number of Station A (originating station) or Representative Number

NOTE 1: When the second data of this command is set to “0”, the Caller ID sent to the SIP trunk is as follows.

- For a new method (assigned by CM08>1220>1): SIP subscriber numbers assigned by CM8A Y=5XXX>186: 01-04 are sent.
- For an old method (assigned by CM08>1220>0): SIP subscriber numbers assigned by CM8A Y=5XXX>176: 01-02 are sent.

NOTE 2: When the second data of this command is set to “1”, the Caller ID sent to SIP trunk depends on the settings of CM08>1220 and CM8A Y=5XXX>186, or CM08>1220 and CM8A Y=5XXX>176.

END

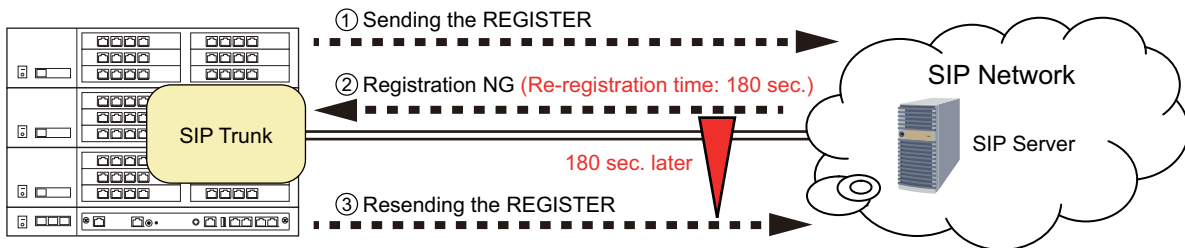
Congestion Control for SIP Trunk

General Description

This feature provides the following congestion control features for SIP trunk when the SIP network congestion occurs.

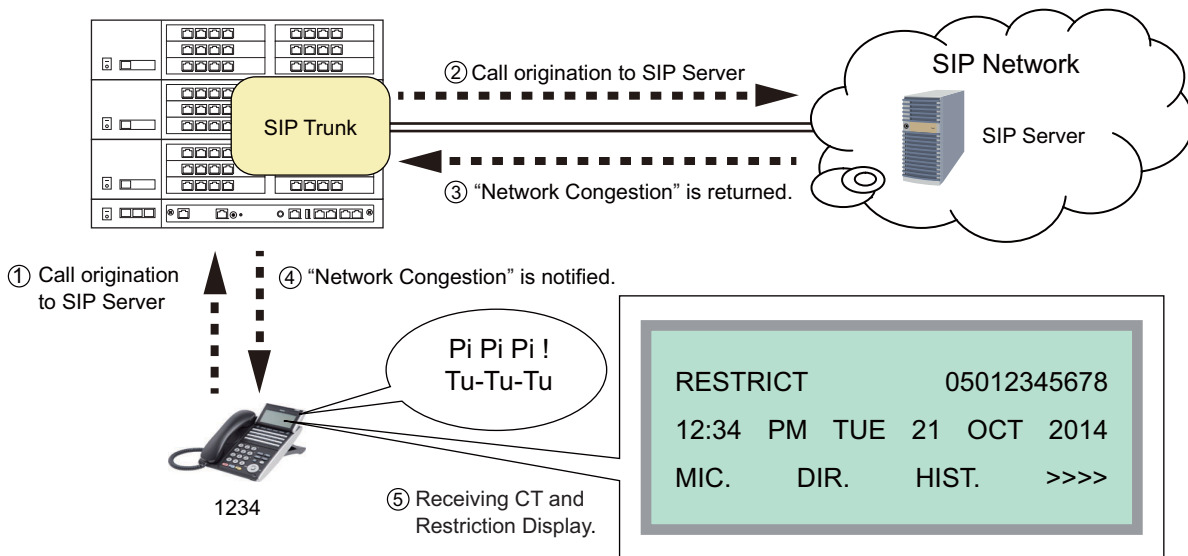
- Re-registration to SIP server

This feature allows the system to resend the REGISTER when the REGISTER sent to SIP server upon SV9300 start up is rejected due to SIP network congestion.



- Congestion Notification on Call Origination

This feature allows the system to notify congestion using a Restriction Display and Congestion Tone (CT) when a response signal “Network Congestion” is received from the SIP network for outgoing calls via SIP trunk.



Service Conditions

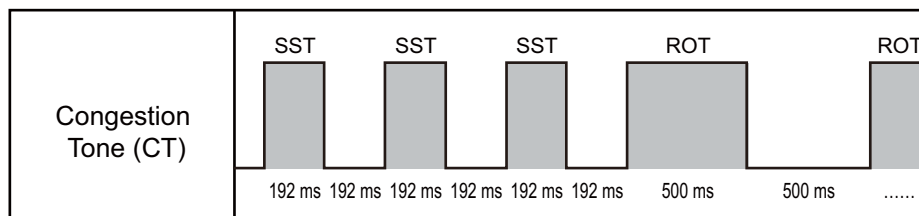
Conditions of Re-registration to SIP Server

- When a SIP trunk detects network congestion, the REGISTER is resent according to the time instructed from the SIP network. In addition, when no response is received from the SIP network, the REGISTER is resent after the elapse of the time set by CMBA Y=158.
- By the fault information display, you can verify the Fault/Restoration status when registering the REGISTER sent from SV9300 (the settings of fault information display are required by CMEA Y=2>043, 053). In additions, the following fault details are registered depending on the SIP network condition.

Fault Kind Code	Fault Description	SIP Network Condition	Fault Details
043	SIP trunk failure	The registration is rejected by the SIP network congestion	SIP network congestion occurred
		No answer from SIP network	Registration error
053	SIP trunk returned to normal condition	SIP network condition is restored and the registration is completed	Restoration from registration failure

Conditions of Congestion Notification on Call Origination

- When a response signal for “Network Congestion” is received against a call originated to SIP Network, the congestion is notified to the calling party using the Congestion Tone and Restriction Display as long as the calling party is a station.
- When the call is originated from a trunk (tandem connection), this Congestion Notification is not provided.
- As a Restriction Display, RESTRICT is displayed on LCD.
- To receive the Congestion Tone (CT), set the second data of CM08>1214 to 0 (CT).
- The sending intervals of Congestion Tone (CT) are as follows (SST of 192 msec ON and OFF is repeated three times, after that ROT is received).



6. The usable terminals for Congestion Notification are as follows.

×: Available –: Not available

Terminal Type	Availability	
	Restriction Display	Congestion Tone (CT)
Single Line Telephone	–	×
DT300/DT400/DT500 Series	×	×
D ^{term}	×	×
D ^{term} IP	×	×
DT700/DT800/DT900 Series	×	×
Soft Phone	×	–
Standard SIP Terminal	–	–
SIP Wireless Terminal (MH240)	×	–
ISDN Terminal	–	–
Attendant Console	–	–
Mobility Access	–	–

Programming

(1) REGISTER resend timer settings when REGISTER sending fails.

To set the REGISTER resend timer when the REGISTER sent from SV9300 is rejected due to the SIP network congestion, do the following programming.

START	DESCRIPTION	DATA
CMBA	Set the REGISTER resend timer when REGISTER sending fails.	<ul style="list-style-type: none"> Y=118 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 1 : 320 sec. 2 : As per CMBA Y=158 NONE◀: As per CM0B Y=1XX>10
	<p style="text-align: center;">RESET</p> <p>Set the REGISTER resend variable timer when REGISTER sending fails.</p> <p>NOTE 1: <i>This data is effective when the second data of CMBA Y=118 is set to 2 (as per CMBA Y=158).</i></p> <p>NOTE 2: <i>The REGISTER resend variable timer is sent by the period added random sec. (0-30 sec.) from the value set by this data.</i></p>	<ul style="list-style-type: none"> Y=158 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 001-254 : 1-254 minutes NONE◀: 3 minutes
END		

(2) Congestion notification settings on call origination

To set the Tone when a response signal “Network Congestion” is received from a SIP network on call origination, do the following programming.

START	DESCRIPTION	DATA
CM08	Set the Tone to “Congestion Tone (CT)” when a response signal “Network Congestion” is received from a SIP network.	<ul style="list-style-type: none"> (1) 1214 (2) 0: Congestion Tone (CT)
END		

Direct Inward Dialing (DID)

General Description

This feature provide incoming calls with Direct Inward Dialing (DID) number.

However, DID Digit Conversion is required when the DID numbers differ from the station numbers set on PBX.

Service Conditions

1. The number of digits to be developed on the DID number can be specified 1-8 digits by CM35 Y=171.
2. The maximum number to be developed is 1000 (assigned by CM76 Y=90).
3. 0-9, *, # can be used for DID Digit Conversion.

Programming

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM35</div>	<p>Provide DID Digit Conversion to the voice channel trunk route number assigned by CM30 Y=00.</p> <p>Specify the Development Table for DID Digit Conversion for voice channel of trunk route.</p> <p>NOTE: <i>Development Table 1 is available for SIP trunk.</i></p> <p>Specify the number of digits to be converted on DID for voice channel of trunk route.</p> <p>NOTE: <i>In the overall system, when a station number which is developed on the DID number corresponds to the another DID number; set the number of digits developed on the DID number (assigned by CM35 Y=171) one digits larger than the station number. Also, set the first data (DID number) of CM76 Y=90 one digit larger (in the case, the maximum number of the station digits is seven).</i></p>	<ul style="list-style-type: none"> • Y=018 <ol style="list-style-type: none"> (1) 00-63: Trunk Route No. (2) 0: To provide • Y=170 <ol style="list-style-type: none"> (1) 00-63: Trunk Route No. (2) 0 : Development Table 1 3◀: Development Table 0 • Y=171 <ol style="list-style-type: none"> (1) 00-63: Trunk Route No. (2) 01-08: 1-8 digits 15◀ : 4 digits
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">A</div>		

A	DESCRIPTION	DATA
CM76	Assign the Number Conversion Block number for Development Table 1.	<ul style="list-style-type: none"> • Y=90 (1) X-XXXXXXXX: DID No. (2) 000-999 : Number Conversion Block No. NONE◀: No data
	<p>NOTE 1: <i>With the number of digits to be converted on DID by this command, the number conversion (CM76 Y=90) is executed. DID number conversion is started from the last digit of the received station number. Example: the received station number is 050-1234-5678 When the second data is set to NONE: 5678 is developed by CM76. When the second data is set to 08: 12345678 is developed.</i></p> <p>NOTE 2: <i>This data is effective only when CM35 Y=18 is set to “0” and CM35 Y=170 is set to “0”.</i></p> <p>NOTE 3: <i>When DID number is displayed on the LCD of Multiline Terminal or Attendant Console, only the last 4 digits are displayed.</i></p> <p>NOTE 4: <i>When DID number is sent to the CS report, only the last 4 digits are sent.</i></p> <p>NOTE 5: <i>In the overall system, when a station number which is developed on the DID number corresponds to the another DID number, set the number of digits developed on the DID number (assigned by CM35 Y=171) one digit larger than the station number. Also, set the first data (DID number) of CM76 Y=90 one digit larger (in the case, the maximum number of the station digits is seven).</i></p>	
B		

B	DESCRIPTION	DATA
CM76	Assign the data for interpreting the digits received.	<ul style="list-style-type: none"> • Y=01 Day Mode • Y=02 Night Mode • Y=03 Mode A • Y=04 Mode B <p>(1) 000-999 : Number Conversion Block No. assigned by CM76 Y=90</p> <p>(2) X-XXXXXXXXX: Station No. to be terminated DXX: Change Terminating System to:</p> <p>D01 : TAS</p> <p>D02 : Trunk Line (Direct) Appearance</p> <p>D03 : Trunk Line (Direct) Appearance + TAS</p> <p>D09 : Automated Attendant</p> <p>D10 : Attendant Console + TAS</p> <p>D11 : Attendant Console + Trunk Line (Direct) Appearance</p> <p>D12 : Attendant Console + Trunk Line Appearance + TAS</p> <p>D14 : Attendant Console</p> <p>D16 : Remote Access to System (DISA)</p> <p>NONE◀: No data</p>
<u>END</u>		

Registration Number “+” Addition/Deletion

General Description

This feature provides call originating with “+” number addition and call terminating with “+” number deletion.

Service Conditions

- All the SIP message header numbers are added “+”.
 - <Before setting the system data>


```
INVITE sip: 987654@195.2.174.100:5060 sip/2.0
From      : “123456” <sip:123456@195.2.174.100:5060>;tag=34e442234a970-63
To        : <sip:987654@195.2.174.100:5060>
Contact   : <sip:123456@221.186.89.197:5060>
```
 - <After setting the system data>


```
INVITE sip: +987654@195.2.174.100:5060 sip/2.0
From      : “+123456” <sip:123456@195.2.174.100:5060>;tag=34e442234a970-63
To        : <sip:+987654@195.2.174.100:5060>
Contact   : <sip:+123456@221.186.89.197:5060>
```
- The number on From header display name following “+” is informed as the calling number when a call is originated.
- The number is not informed when the number on From header display name following “+” is not 0-9, *, #.

Programming

START	DESCRIPTION	DATA
START <div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div> END	Specify whether to provide the “+” number addition/deletion feature.	<ul style="list-style-type: none"> Y=117 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To provide 1 ◀: Not provided

Fault Registration

General Description

This feature provides fault occurrence date, time, and fault content registration on CPU blade when the SIP trunk failure occurs. This feature also informs fault occurrence to the external alarm.

Service Conditions

1. The contents to register are as follows:
 - PBX internal fault: System reset occurrence
 - Line failure : Link failure/Registration failure/DNS query failure
 - Call failure : Out of session timer

Programming

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM08</div>	Enable the fault information storage feature.	(1) 450 (2) 0 : Not stored 1 ◀: To store
END		

Fragmented IP Packet Receiving

General Description

This feature provides receiving fragmented IP packets.

Service Conditions

The data size of fragmented IP packets is a maximum of 2000 bytes (total bytes in the IP packet data field).

Multiple CODEC Selection

General Description

This feature provides CODEC selection customized the voice bandwidth of connected network.

Service Conditions

1. CODEC type G.711 μ -law/A-law and G.729a are available.
2. The precedence of the voice encoding selection for a SIP trunk depends on the setting of CMBA Y=21.
3. Recvonly, sendonly, inactive on SDP parameter Attribute (a) are not supported.
4. The system returns “488 Not Acceptable Here” when the system receives a CODEC which cannot be allowed.
5. The system does not allow to receive the Comfort Noise on SDP parameter attribute. When connecting 3rd party devices or SIP trunk, configure them so that the Comfort Noise is not sent.

Programming

START

DESCRIPTION

DATA

CMBA

Specify the voice encoding selection precedence for SIP.

- Y=21
- (1) 00-63: Profile No. assigned by CMA7
Y=71/CM8A Y=5000-5255>179
- (2) 0-7◀: See table below.

◀: Default

SETTING DATA	MODE	HIGH←SELECTION PRECEDENCE→LOW		
		1	2	3
0	Programmable List	As per CMBA Y=121	As per CMBA Y=122	As per CMBA Y=123
1	Standard Mode 2	G.711 μ-law (64 Kbps)		
2	Standard Mode 3	G.711 μ-law (64 Kbps)	G.711 A-law (64 Kbps)	–
3	Tone Quality Mode 2	G.711 μ-law (64 Kbps)	G.711 A-law (64 Kbps)	G.729a (8 Kbps)
4	Band Mode 2	G.729a (8 Kbps)	G.711 A-law (64 Kbps)	G.711 A-law (64 Kbps)
5	Tone Quality Mode 1	G.711 μ-law (64 Kbps)	G.729a (8 Kbps)	–
6	Band Mode 1	G.729a (8 Kbps)	G.711 μ-law (64 Kbps)	–
7◀	Standard Mode 1	G.711 μ-law (64 Kbps, 20 ms fixed)		

NOTE 1: When the voice encoding selection setting differs from that for the opposite SIP trunk, the voice encoding selection may differ in the user's usual SIP trunk setting according to the negotiation when the SIP session is made.

NOTE 2: Payload size is set as per CMBA Y=22 when the second data of CMBA Y=21 is set to a value other than "7" (Standard Mode 1).

Specify the payload size for SIP trunk.

- Y=22
- (1) 00-63: Profile No. assigned by CMA7
Y=71
- (2) 1 : 20 ms.
2 : 30 ms.
3◀: 40 ms.

NOTE 1: This data is not available when the second data of CMBA Y=21 is set to "7" (default).

NOTE 2: Because the standard payload size for SIP is 20 ms., it is strongly recommended to set the second data to "1".

END

Session Timer

General Description

This feature confirms whether the call status between the originating terminal and the communicated terminal is normal or not by sending/receiving a signal at a constant interval.

Service Conditions

1. The session timer method is decided by the receiving SIP message from the communicated terminal set by CMBA Y=56.
2. Session timer does not start when the following definitions are not contained in SIP message (200OK or INVITE).
 - Require : timer
 - Supported: timer

Programming

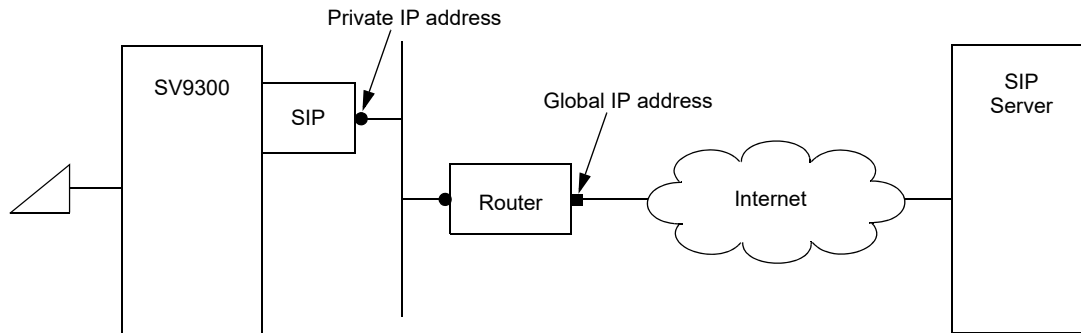
START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	<p>Specify the session timer refresher.</p> <p>NOTE: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>Specify the session timer method.</p> <p>NOTE 1: <i>When the second data is set to 3, the session timer method is decided by the receiving message from the communicated terminal.</i></p> <p>NOTE 2: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>NOTE 3: <i>For a call termination, follow the profile No. assigned by CMA7 Y=71.</i></p>	<ul style="list-style-type: none"> • Y=29 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : uas 1◀: uac <ul style="list-style-type: none"> • Y=56 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Update 1 : Invite 3◀: Auto
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">A</div>		

A	DESCRIPTION	DATA
CMBA	Provide the session timer.	<ul style="list-style-type: none"> • Y=83 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 1◀: To provide
	<p>NOTE 1: <i>The second data should be set to "1" for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>NOTE 2: <i>For Point-to-Multipoint connection (when CMA7 Y=46 is set to "0"), unify whether to use the session timer method in a network.</i></p>	
	Specify the session timer value.	<ul style="list-style-type: none"> • Y=88 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 1-8553600: Session timer value 1 second - 8553600 seconds (99 days) NONE◀ : 1800 seconds
	<p>NOTE 1: <i>This command is effective only when CMBA Y=29 is set to "0".</i></p> <p>NOTE 2: <i>This data should be set to NONE for Point-to-Multipoint connection (when CMA7 Y=46 is set to "0").</i></p> <p>NOTE 3: <i>For a terminating call, follow the profile number assigned by CMA7 Y=71.</i></p> <p>NOTE 4: <i>When the second data of CMBA Y=29 is set to "0" (uas), follow the session timer value of the profile number assigned by CMA7 Y=71.</i></p>	
<u>END</u>		

NAT Support in SIP Trunk

General Description

This feature provides the connection to the Internet with the NAT/NAPT functions of a router.



Service Conditions

1. This feature is available only for Point-to-Point connection (not available for Point-to-Multipoint connection).
2. DHCP cannot be used because the private IP address of the router side must be fixed.
3. VLAN and this feature cannot be used at the same time.
4. A router must be capable of NAT/NAPT processing and band processing to be used in accordance with the number of accommodated channels of SIP.

Programming

- When operating a system assigning one SIP trunk number (CM35 Y=091) (for Single-Carrier connection):

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin-bottom: 5px;">CM0B</div> <div style="text-align: center; border: 1px solid black; border-radius: 10px; padding: 2px; margin-top: 10px;">RESET</div>	Assign the Global IP address of NAT for the SIP. NOTE 1: <i>When assigning an IP address, the second data must be entered including the periods (.).</i> NOTE 2: <i>Set the private IP addresses for the SIP with CM0B Y=1XX>00, 01, 02.</i>	<ul style="list-style-type: none"> Y=1XX (VOIP Port [1] + Unit No. [01-50]) (1) 70: Global IP Address (2) XXX.XXX.XXX.XXX: 0.0.0.1-255.255.255.254: Global IP Address (Maximum 15 digits) NONE◀: No data
<u>END</u>		

- When operating a system assigning two or more SIP trunk numbers (CM35 Y=091) (for Multi-Carrier connection):

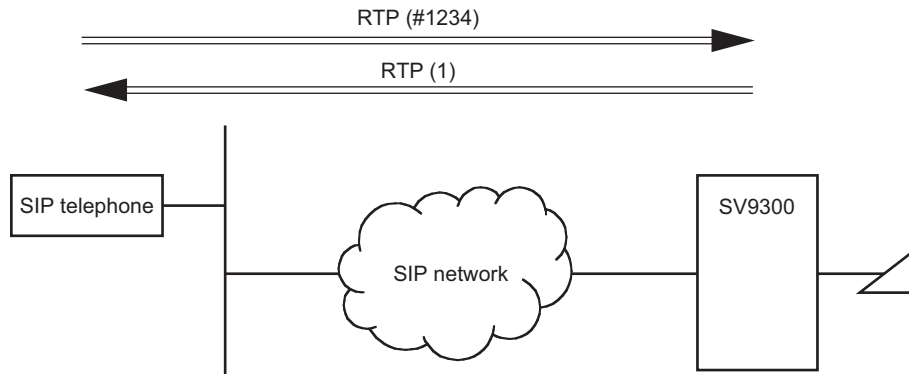
START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin-bottom: 5px;">CMBA</div> <div style="text-align: center; border: 1px solid black; border-radius: 10px; padding: 2px; margin-top: 10px;">RESET</div>	Assign the Global IP address of NAT for SIP.	<ul style="list-style-type: none"> Y=139 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) XXXXXXXXXXXXX: 000000000000-255255255254: Global IP Address of SIP Server (Maximum 12 digits) NONE◀: No data
<u>END</u>		

Out-band DTMF

General Description

This feature provides the DTMF relay supports RFC 2833 in SIP network.

When receiving a DTMF signal from the SIP network, the SV9300 retrieves the DTMF information from RTP and sends to the terminal under the SV9300. When sending the DTMF signal from the terminal under the SV9300 to the network, the SV9300 analyzes the DTMF signal, and sends out on RTP.



Service Conditions

- The following DTMF relay method can be used in the SIP network (assigned by CMBA Y=52).
 - Out-band DTMF (RFC2833)
 - In-band DTMF (Voice pass through)
- Be sure to match the DTMF relay method in a network. The DTMF relay method of the own office is assigned by the setting of system data (CMBA Y=52) and the DTMF relay method of the opposite office. The table below shows the combination of the DTMF relay method.

SETTING OF OWN OFFICE \ SETTING OF OPPOSITE OFFICE	RFC2833 METHOD		IN-BAND METHOD	
	RECEIVING AT OWN OFFICE	RECEIVING AT OPPOSITE OFFICE	RECEIVING AT OWN OFFICE	RECEIVING AT OPPOSITE OFFICE
RFC2833 method	RFC2833	RFC2833	In-band	In-band
In-band method	In-band	In-band	In-band	In-band

3. The sending of a DTMF is available only for the DTMF relay method assigned by CMBA Y=52. For receiving a DTMF, The DTMF may be replayed even if the DTMF is received by the DTMF relay method other than assigned one. The table below shows the DTMF relay methods that can be replayed by the DTMF settings.

×: Available –: Not available

RECEIVING DTMF SETTING OF DTMF RELAY METHOD	DTMF (RFC2833)	IN-BAND DTMF
RFC2833 method	×	×
In-band method	–	×

4. DTMF (RFC2833) that can be sent/received are “0-9”, *, and “#”.
“A”, “B”, “C” and “D” are not supported.

KINDS OF DTMF	NOTATION IN RTP MESSAGE
0–9	0, 1, 2, 3, 4, 5, 6, 7, 8, 9
*	10
#	11

5. If the payload type is unknown when the DTMF (RFC2833) is received from the opposite office, the own office operates as “In-band”.

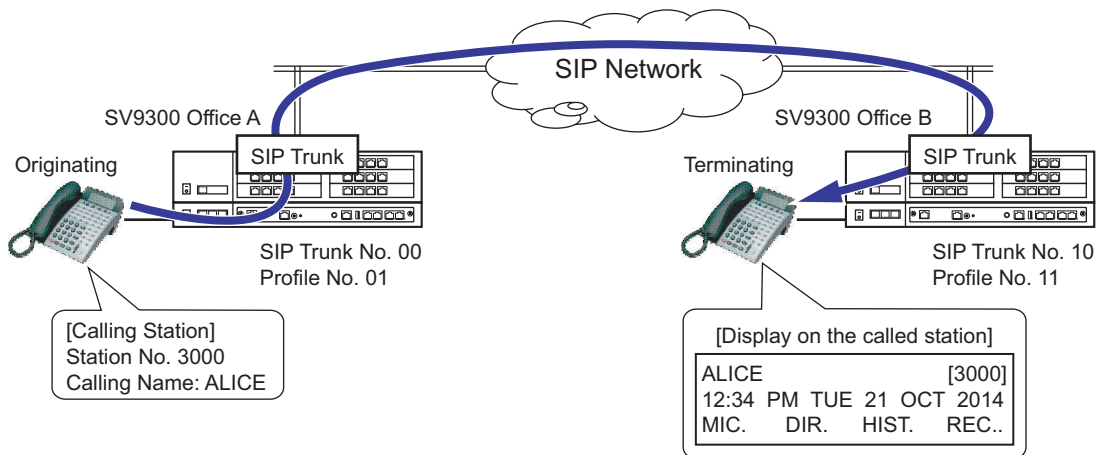
Programming

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 2px; width: fit-content; margin: 0 auto;">CMBA</div>	<p>Assign the DTMF relay method.</p> <p>NOTE: <i>This command assigns DTMF mode for sending from a unit in which a SIP Trunk is accommodated to another device to which the SIP trunk is connected. For Intra-unit DTMF, refer to CM67 Y=26.</i></p>	<ul style="list-style-type: none"> • Y=52 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 03 : Out-band DTMF (RFC2833) NONE◀: In-band DTMF (Voice pass through)
<p>Assign the payload type of Out-band DTMF (RFC2833).</p>	<ul style="list-style-type: none"> • Y=128 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 001-127 : Payload type 001-127 NONE◀: Payload type 101 	
<u>END</u>		

Calling Number and Name Display

General Description

This feature provides Calling Number and Name Display to the called station when terminating a call from a SIP network. By this feature, the calling number and name informed from the SIP network can be displayed on the LCD of Multiline Terminal.



Service Conditions

1. This feature is available only when SIP trunk is used.
2. This feature is available even in the system with a Point-to-Multipoint connection.
3. The conditions when originating calls are as follows.
 - Calling names assigned for each station number can be informed.
 - With ASCII code, the maximum of 16 characters can be informed.
 - An alphabet of up to 16 characters (one byte character) can be registered for one station number.
 - When a calling name has a space on the top, the space is deleted when informed.
 - When using this feature with Multiline Terminals, set the calling name to the my line number. The name set to the my line is informed to the called station when a call is originated from the sub line.
 - Calling names can be also assigned for station numbers of the terminal without LCD.
 - Calling names can be registered/changed/deleted by access codes. However, this setting is limited to the my line numbers.
 - For outgoing calls from SIP trunk, a calling name of each station number on a called side is not displayed on the terminal on a calling side.

- The table below shows the conditions for calling name notifications in Tandem Connection.

◀: Default

Incoming Calling Number	Incoming Calling Name	CMBA Y=44 (Calling Name informed to SIP Trunk)				
		00	01	02	03	15◀
Not informed	Not informed	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Not informed
-	Informed	Representative Number by CMBA Y=32	Name informed from Incoming Trunk	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Name informed from Incoming Trunk
Informed	-	Representative Number by CMBA Y=32	Original caller ID/ DID Number	Representative Number by CMBA Y=32	Representative Number by CMBA Y=32	Original caller ID/ DID Number

- The conditions when terminating calls are as follows.
 - For D^{term} series, the character code to be displayed is determined by the display language order from the system and also depends on the language display feature of each terminal. As for the Standard SIP Terminal, the character code to be displayed depends on the language display feature of each terminal.
 - The maximum of 16 characters (one byte character) can be displayed as a name. When a received name has more than 16 characters, 16 characters from the top are displayed (17th character and later cannot be displayed).
 - When a received name has “+” on the top, only the “+” is not displayed.
 - When a space is included on the top of a received name area, the spaced is deleted when displayed.
 - When the parameter ID is assigned to the Privacy header, the name is not displayed.
 - When this feature is effective, and an incoming call with no calling name information on the From header is terminated, the calling name is not displayed.
 - If “Anonymous” is indicated on the Displayname field of the From header, the calling name and the calling number are not displayed.
 - The calling party number acquisition field can be specified by CMBA Y=126.

- When calling number acquisition field is set to Username field and “calling name in Displayname field/calling number in Username field” is written in INVITE From header, the calling number and name can be simultaneously displayed. However, system data assignment (CM15 Y=400) is required to display them together.

<Example>

From : “ABCDE” <123456@195.168.1.2>; tag = 1234t2tXXXX

Calling Name: ABCDE, and Calling Number: 123456 are displayed.

- In tandem connection, when a name is written in Displayname field of INVITE From header received from SIP trunk, the name information is sent to outgoing trunk as it is. In this case, note that name notification may not be available because of outgoing trunk specifications or settings.
5. The system data conditions for outgoing calls are as follows.
- Whether to provide calling name notification for each station when originating calls can be assigned by CM8A Y=5000-5255>183. For conditions, see the table below.

◀: Default

	CM8A Y=5000-5255>183	
	2nd Data=0	2nd Data=15◀
Displayname Field	The name assigned for each station is informed (set by CM77 Y=0/1). NOTE 1, NOTE 2	The calling number is informed. NOTE 2, NOTE 3

NOTE 1: The calling number is informed when CM77 Y=0/1 is not set.

NOTE 2: The way of calling number notification follows the setting of “Caller ID Display”.

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NOTE 3: The calling number is written in the Displayname field when CM8A Y=5000-5255>183: 15 (calling party name is not sent) is assigned. In this case, the calling number to be sent depends on the setting of “Caller ID Display”. [Page 2-307](#)

- To provide the calling name notification, set the second data of CMBA Y=92 (display name/user name for From Header) to “3”. The calling name cannot be informed when the second data of CMBA Y=92 is set to “0”.
- Calling names can be assigned by CM77 Y=0/1 (station name assignment with character code/character). Calling names can be also assigned by the access code (CM20 Y=0-3: A110)

Programming

START	DESCRIPTION	DATA
CM08	Provide Caller ID Display/Name Display (Attendant Called/Calling Name Display) to the called station when a call is terminated via SIP.	(1) 379 (2) 1◀: To provide
	<p>NOTE: <i>To make this feature effective, be sure to set the second data to "1".</i></p> <p>Provide the sending of calling station number or calling party number to the analog telephone for Caller ID-Station when an incoming call is terminated via SIP Trunk.</p> <p>NOTE: <i>The sending of a calling station number to the analog telephone for Caller ID-Station is effective when CM08>507 is set to 1.</i></p>	(1) 603 (2) 0: To provide
CM77	Enter the desired station user's name to each station number by CM77 Y=0 or CM77 Y=1.	<ul style="list-style-type: none"> • Y=0 By Character Code (1) X-XXXXXXXX: Station No. (2) Character Code 20-7F (Maximum 32 digits) See Character Code Table. Page 2-293
CM8A	Specify whether to inform the calling name from SIP trunk.	<ul style="list-style-type: none"> • Y=1 By Character (1) X-XXXXXXXX: Station No. (2) A-Z, 0-9: Character (Maximum 16 characters)
		<ul style="list-style-type: none"> • Y=5000-5255 LCR Pattern No. 000-255 (1) 183: Calling Name Information from SIP Trunk (2) 00 : To inform the calling name assigned for each station (to inform the calling name assigned by CM77 Y=0/CM77 Y=1) 15◀: Not informed the calling name
A		

A	DESCRIPTION	DATA
CMA7	Provide the SIP trunk number with Caller ID Display service.	<ul style="list-style-type: none"> • Y=26 (1) 00-63: SIP Trunk No. 00-63 (2) 1◀: To provide
	NOTE 1: <i>Effective when CM08>379 is set to "1".</i>	
	NOTE 2: <i>To provide Calling Name Display-CCIS, set the second data to "1" in both own office and opposite office.</i>	
	NOTE 3: <i>To make this feature effective, be sure to set the second data to "1".</i>	
CMBA	Specify the display name/user name for From Header.	<ul style="list-style-type: none"> • Y=92 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Display name: SIP AoR User name following CMBA Y=45 : User name : SIP AoR User name following CMBA Y=45 2 : Display name: Caller ID following CMBA Y=44 : User name : SIP AoR User name following CMBA Y=45 3◀: Display name: Caller ID following CMBA Y=44 : User name : Caller ID following CMBA Y=44
	RESET	
	NOTE 1: <i>When a calling party wants to inform a called party of the calling party name, be sure to set the second data of this command to "2" or "3".</i>	
	NOTE 2: <i>For a carrier connection using an AoR User name, be sure to set the second data to "0" or "2".</i>	
B		

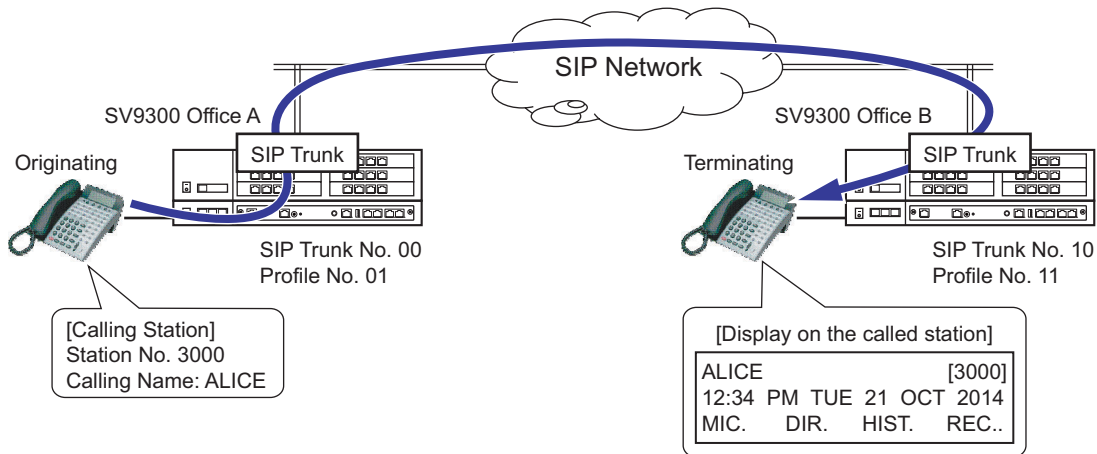
B	DESCRIPTION	DATA
CMBA	<p>Set the reference to Caller ID.</p> <p>NOTE: <i>To display the calling number and name at the same time, be sure to set the second data to “1”.</i></p>	<ul style="list-style-type: none"> • Y=126 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : Get caller ID from the Username field if the Displayname field of the From header of initial INVITE message is blank. 1 : Get caller ID from the Username field of the From header of initial INVITE message only. 3◀: Get caller ID from the Displayname field of the From header of initial INVITE message only.
CM20	<p>Assign the logout access code.</p> <p>NOTE: <i>For details of CM20, refer to the Command Manual.</i></p>	<ul style="list-style-type: none"> • Y=0-3 Numbering Plan Group 0-3 (1) X-XXXX: Access Code (2) A110: Name Display
CM15	<p>Specify the displaying pattern of Caller ID on the LCD of Multiline Terminal before answering or after answering a trunk call.</p> <p>NOTE 1: <i>To display the calling number and name at the same time, set the second data of CM15 Y=400 to “1” or “7”.</i></p> <p>NOTE 2: <i>When the second data of CM15 Y=400 is set to “7”, set the second data of CM15 Y=136 to “1” (Calling Name Display).</i></p> <p>NOTE 3: <i>When the second data of CM15 Y=400 is set to “1”, set the second data of CM15 Y=136 to “0” (Calling Number Display).</i></p>	<ul style="list-style-type: none"> • Y=400 (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 0 : Not displayed calling number and calling name simultaneously 1 : To display Calling Name on upper line of LCD, Calling No. on middle line of LCD 7◀: To display calling number on upper line of LCD, calling name on middle line of LCD
<u>END</u>	<p>Allow Calling Name Display for incoming trunk calls in Service Restriction Class A assigned by CM12 Y=02.</p>	<ul style="list-style-type: none"> • Y=136 (1) 00-15: Service Restriction Class A assigned by CM12 Y=02 (2) 1◀: Calling Name Display

Operating Procedure

No manual operation is required.

■ Sample Programming of Calling Number and Name Display

To provide Calling Number and Name Display for both Office A and B in the illustration below, the following (a), (b) and (c) must be assigned.



(a) Sample Programming of name assignment for the station in originating Office A

Command	1st Data	2nd Data	Remarks
CM77 Y=1	3000	ALICE	Assign "ALICE" as a name for station 3000 from PCPro.

(b) Sample Programming for informing Calling Name when a call is originating from Office A

Command	1st Data	2nd Data	Remarks
CMA7 Y=26	00	1	Provide Name Display-CCIS for SIP trunk number 00.
CM8A Y=5000-5255	183	00	Inform the calling name assigned for each station by CM77 Y=1 when calls are originated from SIP trunk.
CMBA Y=92	01	3	Set from header Displayname/User-name for profile number 01 as below. - Displayname: Calling number - Username : Calling number

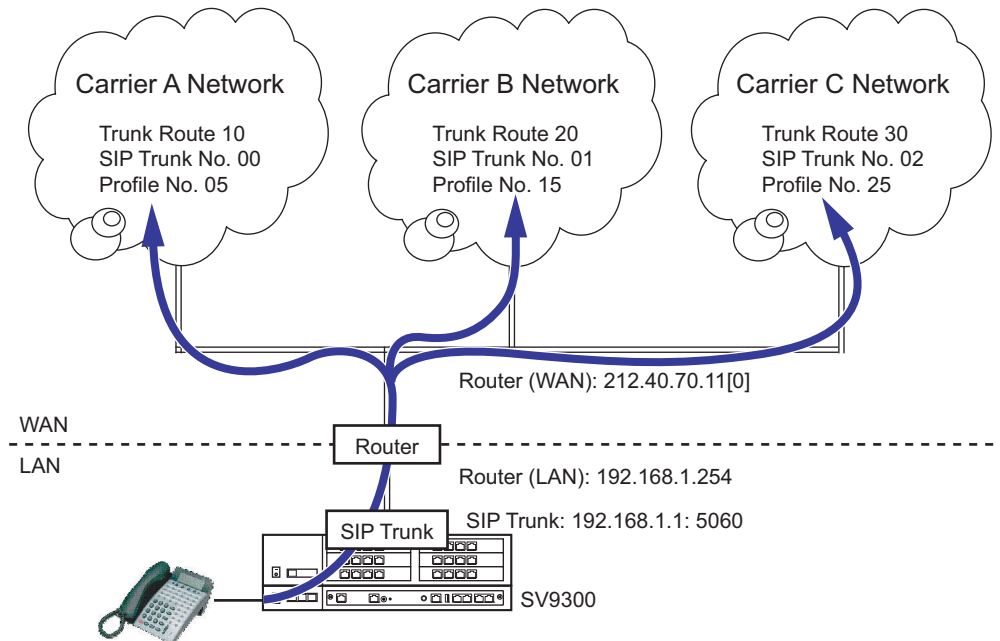
(c) Sample Programming for displaying Calling Name when a call is terminated to Office B

Command	1st Data	2nd Data	Remarks
CM08	379	1	Provide the Caller ID/Name Display when calls are terminated from SIP trunk.
CMA7 Y=26	10	1	Provide Name Display-CCIS for SIP trunk number 10.
CMBA Y=126	11	1	Set the reference to Caller ID to User-name for displaying the calling number and name at the same time.

Multi-Carrier Connection

General Description

This feature allows one system to use more than one carrier. Therefore, by alternate routing using Multi-Carrier connection and secondary server on some failure occurrence, high reliable network can be established.



Service Conditions

1. This feature is available only when SIP trunk is used.
2. Up to 512 trunks for voice channels can be registered.
3. The conditions for carriers with this feature are as follows.
 - The maximum of 64 carriers can be used (The maximum of 8 carriers per unit).
 - For the conditions of each carrier, such as destination server IP/Port/SIP, set them for each profile number (set by CMA7 Y=71).
 - The port number of SIP trunk cannot be prepared for each carrier (one port for one system).
4. The number of DNS servers that can be registered are only primary and secondary (As per CM0B Y=1XX>60/61). This is common to all carriers.
5. The conditions for each service with this feature are as follows.
 - Ringing Tone can be specified for each carrier (trunk route).
 - The carrier (=SIP trunk number) whose REGISTER fails can be checked by the fault Information of PCPro.

- When an outgoing call to SIP network fails due to some troubles in the carrier side, the call can be rerouted through up to 6 patterns with 3 carriers by CM8A development. When providing this alternate routing, confirm the conditions of carriers.
 - Link down notification is sent to all the carriers by a line/trunk feature key. It is not possible to send it to each carrier.
 - When FAX communication is rerouted by some reason, if the CODEC of reroute destination carrier is G.729a, the FAX communication will fail.
 - When Multi-Carrier connection is used, Static connection service cannot be used together.
 - Up to two kinds of calling number (CM12 Y=12/46) including ISDN can be sent. When more than two kinds of calling number are handled, notify the calling number with the representative number (CMBA Y=32).
 - The IP address matching feature (CMBA Y=137) can be set only when Single Carrier connection is used.
6. The system data conditions are as follows.
- When Multi-Carrier connection is used, set other IP trunks to Point-to-Point connection (CMA7 Y=46: 1). Point-to-Multipoint connection cannot be used together.
 - In Multi-Carrier connection, assign a Global IP address for NAT by CMBA Y=139. When using Single-Carrier connection, assign a Global IP address for NAT by CM0B Y=1XX>70.
7. To use Multi-Carrier connection, the following licenses are required.
- System Version License
 - SIP Trunk Channels License

Programming

STEP1: Do the programming of “SIP Trunk Data Assignment”. see “SIP Trunk Data Assignment”

[Page 2-255](#)

NOTE: *In addition to the “SIP Trunk Data Assignment”, to provide the system with Multi-Carrier connection service, the system data setting in the following table are required.*

Command	Meaning	Required Settings for Multi-Carrier Connection
CM0B Y=1XX>60	Primary IP address for DNS server	Assign DNS server (primary) available for all the carriers.
CM0B Y=1XX>61	Secondary IP address for DNS server	Assign DNS server (secondary) available for all the carriers.
CM30 Y=00	Trunk route allocation	Assign the trunk route to the voice channel trunk for each carrier.
CM35 Y=091	SIP trunk number for SIP	Set the SIP trunk number for each carrier not to overlap each other.
CMA7 Y=01	Originating Point Code (OPC)	Set the same OPC for all the SIP trunk numbers.
CMA7 Y=02	Destination Point Code (DPC)	Set the different DPCs respectively for each SIP trunk number.
CMA7 Y=46	Connection Method for SIP	Set the second data to “1” (Point-to-Point connection).
CMA7 Y=71	SIP profile number for control packet	Set the different profile number respectively for each carrier.

STEP2: Assign the Global IP address for NAT by CMBA Y=139. See “When operating a system assigning two or more SIP trunk numbers (CM35 Y=091) (for Multi-Carrier connection):”

[Page 2-335](#)

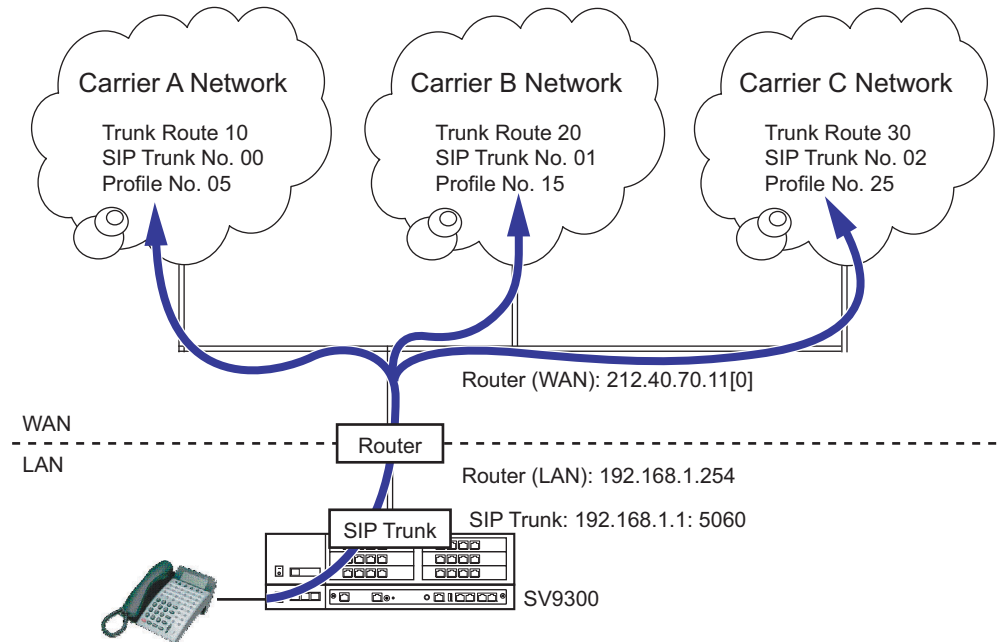
Operating Procedure

No manual operation is required.

Sample Programming for Multi-Carrier Connection

■ Sample Programming for Multi-Carrier Connection Network Setting

In the following Multi-Carrier connection configuration, sample programming for Carrier A, B and C are shown below.



*1 Carrier A and B use the IP address of SIP server specified by the carrier.

*2 Carrier C uses the IP address of SIP server domain name resolved by DNS.

The setting values of the sample programming for Carrier A, B and C are based on the following setting examples.

Setting of SV9300:

- IP address for the system (CM0B Y=101>00) : 192.168.1.1
- Subnet Mask for the system (CM0B Y=101>01) : 255.255.0.0
- UDP Port for SIP control packet (CM0B Y=101>36) : 5060
- IP address (RTP) for VoIPDB (CM0B Y=201>00) : 192.168.1.10
- IP address of the router (LAN) (CM0B Y=101>02) : 192.168.1.254
- IP address of the router (WAN) : 212.40.70.11 [0]

Settings specified from Carrier A, B and C:

Setting specified from Carriers	Setting Data		
	Carrier A	Carrier B	Carrier C
Query to DNS server	Not Provided	Not Provided	Provided
Primary IP address of DNS server	–	–	10.176.83.130
Secondary IP address of DNS server	–	–	10.176.83.135
SIP server domain name	–	–	nec.ne.jp
IP address of SIP server	61.200.166.137	61.200.166.138	
Port number of SIP server	5060	5060	5060
Global IP Address for NAT	–	–	135.205.50.30
Authentication user name	v987987987	w654654654	z321321321
Authentication password	11112222	33334444	55556666
Representative number	050-1111-2222	050-3333-4444	050-5555-6666
SIP-URL use name	a11223344	b55667788	c99001122
SIP-URL domain name	pha3.nec.ne.jp	rib4.nec.ne.jp	sjc5.neci.ne.jp
SIP trunk registration method to SIP server	To register the time set by CMBA Y=71	To register the time set by CMBA Y=71	To register the time set by CMBA Y=71
SIP trunk registration expiry time to the SIP server	3600 sec.	1800 sec.	3600 sec.
From header Username	SIP-URL use name	SIP-URL use name	SIP-URL use name
Re-sending period of REGISTER after REGISTER sending	320 sec.	60 sec.	60 sec.
Setting logic for contact header user field	Random	Random	Random
Voice encoding	G.711/20 ms.	G.711/20 ms.	G.711/20 ms.

Setting for connecting with Carrier A, B and C:

Setting for connecting with Carrier A, B and C	Setting Data		
	Carrier A	Carrier B	Carrier C
The number of voice channels	32	16	8
Trunk route number	10	20	30
Circuit Identification Code (CIC)	001-032	033-048	049-056
SIP trunk number (CCH number)	00	01	02
Originating Point Code	00100	00100	00100
Destination Point Code	00200	00300	00400
Profile number	05	15	25

Sample Programming for Carrier A

Command	1st Data	2nd Data	Remarks
CM10 Y=03	01032-01063	D032-D063	Virtual port number for voice channel
CM30 Y=00	032-063	10	Trunk route number
CM35 Y=090	10	0	Protocol Type: CCIS/SIP
CM35 Y=091	10	00	SIP trunk number
CM30 Y=35	032-063	001-032	Circuit Identification Code (CIC number)
CM35 Y=000	10	00	Kind of trunk: C.O.line (When using Point-to-Point connection to SIP carrier)
CM35 Y=004	10	2	Answer signal: Answer signal arrives
CM35 Y=005	10	1	Release signal: Release signal arrive
CM35 Y=009	10	03	Incoming connection signaling: Wink start/SIP
CM35 Y=020	10	00	Sender start condition: Wink start/SIP
CMA7 Y=01	00	00100	Originating Point Code
CMA7 Y=02	00	00200	Destination Point Code
CMA7 Y=46	00	1	Point-to-Point connection
CMA7 Y=71	00	05	Profile number

Continued on next page

Command	1st Data	2nd Data	Remarks
CMA8 Y=0	00200	00	SIP trunk number for Destination Point Code
CMBA Y=21	05	7	CODEC: G.711 only, Payload size: 20 ms.
CMBA Y=30	05	061200166137	SIP server IP address
CMBA Y=31	05	5060	SIP server port number
CMBA Y=32	05	05011112222	Representative number
CMBA Y=45	05	a11223344	SIP-URL user name
CMBA Y=70	05	0	SIP trunk registration method to the SIP server: To register the time set by CMBA Y=71
CMBA Y=71	05	3600	SIP trunk registration expiry time to the SIP server: 3600 sec.
CMBA Y=72	05	v987987987	Authentication user name
CMBA Y=74	05	11112222	Authentication password
CMBA Y=76	05	pha3.nec.ne.jp	SIP-URL domain name
CMBA Y=92	05	0	From header Username field: SIP-URL user name
CMBA Y=139	05	212040070011	Global IP address for NAT

Sample Programming for Carrier B

Command	1st Data	2nd Data	Remarks
CM10 Y=03	01033-01048	D064-D078	Virtual port number for voice channel
CM30 Y=00	033-048	20	Trunk route number
CM35 Y=090	20	0	Protocol Type: CCIS/SIP
CM35 Y=091	20	01	SIP trunk number
CM30 Y=35	033-048	001-016	Circuit Identification Code (CIC number)
CM35 Y=000	20	00	Kind of trunk: C.O.line (When Point-to-Point connection to SIP carrier)
CM35 Y=004	20	2	Answer signal: Answer signal arrives
CM35 Y=005	20	1	Release signal: Release signal arrives
CM35 Y=009	20	03	Incoming connection signaling: Wink start/SIP

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Command	1st Data	2nd Data	Remarks
CM35 Y=020	20	00	Sender start condition: Wink start/SIP
CMA7 Y=01	01	00100	Originating Point Code
CMA7 Y=02	01	00300	Destination Point Code
CMA7 Y=46	01	1	Point-to-Point connection
CMA7 Y=71	01	15	Profile number
CMA8 Y=0	00300	01	SIP trunk number for Destination Point Code
CMBA Y=21	15	7	CODEC: G.711 only, Payload size: 20 ms.
CMBA Y=30	15	061200166138	SIP server IP address
CMBA Y=31	15	5060	SIP server port number
CMBA Y=32	15	05033334444	Representative number
CMBA Y=45	15	b55667788	SIP-URL user name
CMBA Y=70	15	0	SIP trunk registration method to the SIP server: To register the time set by CMBA Y=71
CMBA Y=71	15	1800	SIP trunk registration expiry time to the SIP server: 1800 sec.
CMBA Y=72	15	w654654654	Authentication user name
CMBA Y=74	15	33334444	Authentication password
CMBA Y=76	15	rib4.nec.ne.jp	SIP-URL domain name
CMBA Y=92	15	0	From header Username field: SIP-URL user name
CMBA Y=139	15	212040070011	Global IP address for NAT

Sample Programming for Carrier C

Command	1st Data	2nd Data	Remarks
CM0B Y=101	60	10.176.83.130	Primary IP address of DNS server
CM0B Y=101	61	10.176.83.135	Secondary IP address of DNS server
CM10 Y=03	01049-01056	D049-D056	Virtual port number for voice channel
CM30 Y=00	049-056	30	Trunk route number
CM35 Y=090	30	0	Protocol Type: CCIS/SIP
CM35 Y=091	30	02	SIP trunk number
CM30 Y=35	049-056	001-008	Circuit Identification Code (CIC number)
CM35 Y=000	30	00	Kind of trunk: C.O.line (When Point-to-Point connection to SIP carrier)
CM35 Y=004	30	2	Answer signal: Answer signal arrives
CM35 Y=005	30	1	Release signal: Release signal arrives
CM35 Y=009	30	03	Incoming connection signaling: Wink start/SIP
CM35 Y=020	30	00	Sender start condition: Wink start/SIP
CMA7 Y=01	02	00100	Originating Point Code
CMA7 Y=02	02	00400	Destination Point Code
CMA7 Y=46	02	1	Point-to-Point connection
CMA7 Y=71	02	25	Profile number
CMA8 Y=0	00400	02	SIP trunk number for Destination Point Code
CMBA Y=21	25	7	CODEC: G.711 only, Payload size: 20 ms.
CMBA Y=25	25	0	Query a DNS server: Provide
CMBA Y=31	25	5060	SIP server port number
CMBA Y=32	25	05055556666	Representative number
CMBA Y=45	25	C99001122	SIP-URL user name
CMBA Y=70	25	0	SIP trunk registration method to the SIP server: To register the time set by CMBA Y=71
CMBA Y=71	25	3600	SIP trunk registration expiry time to the SIP server: 3600 sec.
CMBA Y=72	25	z321321321	Authentication user name

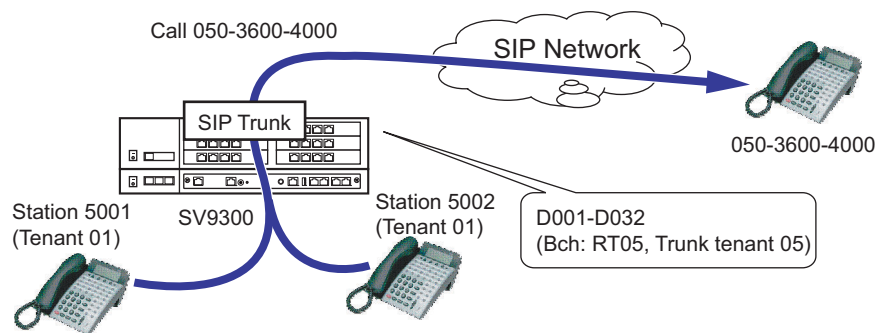
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Command	1st Data	2nd Data	Remarks
CMBA Y=74	25	55556666	Authentication password
CMBA Y=76	25	sjc5.nec.ne.jp	SIP-URL domain name
CMBA Y=92	25	0	From header Username field: SIP-URL user name
CMBA Y=93	25	nec.ne.jp	SIP server domain name
CMBA Y=139	25	212040070011	Global IP address for NAT

■ Sample Programming for Call Origination by Multi-Carrier Connection

Conditions of sample programming:

- 050-3600-4000: destination (called party) number
- Trunk route number 05 (for SIP trunk)

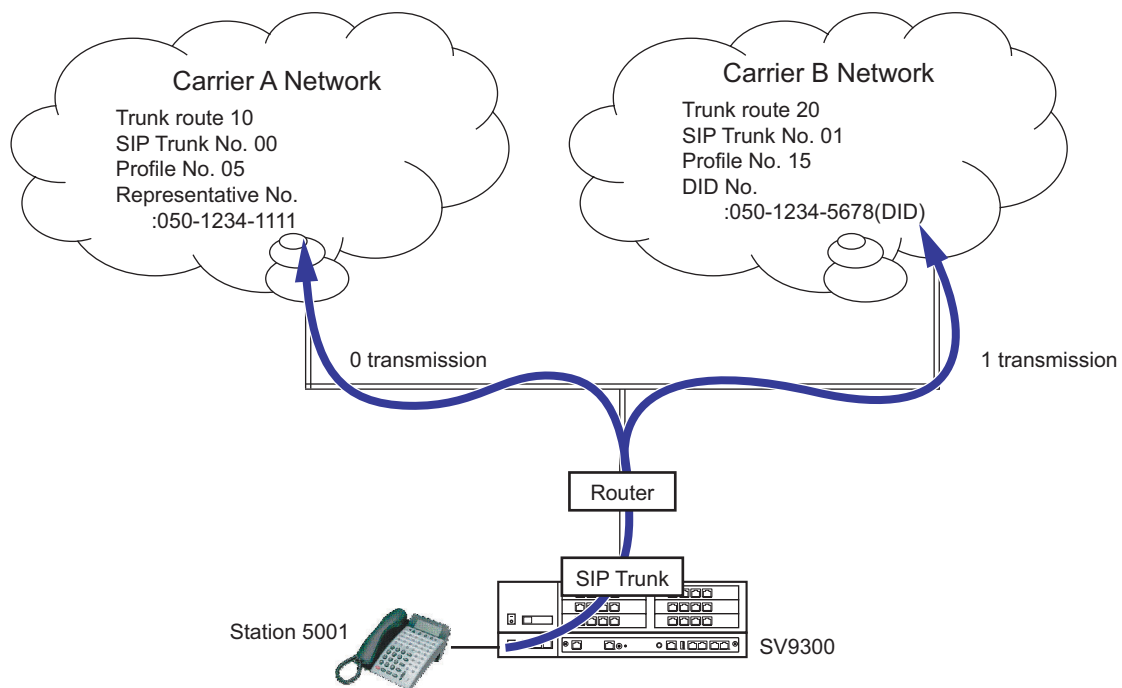


NOTE: The following example is assuming that the startup of SIP trunk and the programming for Station-to-Station call are completed.

Command	1st Data	2nd Data	Remarks
CM29	01	710	Tenant group 0 (related command: CM20 Y=0)
CM20 Y=0	0	A126	LCR group 0
CM8A Y=A000	0	4000	Area Code Development pattern No. 0
CM8A Y=4000	0	0005	Route pattern No. 005
CM8A Y=0005	1	00505	LCR pattern No. 005 + Trunk route No. 05
CM85 Y=0	0	11	11 digits (because 05036004000 is 11 digits.)

Command	1st Data	2nd Data	Remarks
CM8A Y=A000	3	4007	Area code development pattern 7
CM8A Y=4007	0	8000	Inter-Office Termination In order to start DID termination for incoming number with 0XX on the top

■ Sample Programming for Calling Number Notification by Multi-Carrier Connection



Command	1st Data	2nd Data	Remarks
CM20 Y=0	0	A126	LCR group 0
CM20 Y=0	1	A127	LCR group 1
CM8A Y=A000	0	4000	Area code development pattern 0
CM8A Y=A000	1	4001	Area code development pattern 1

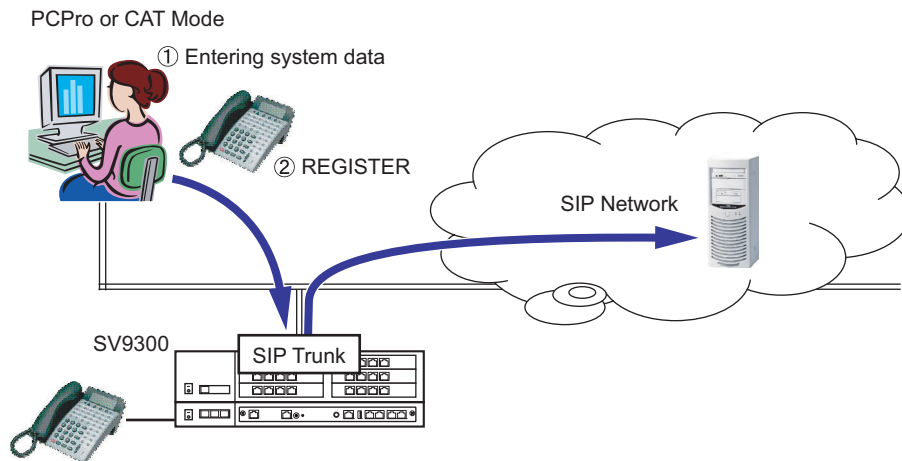
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Command	1st Data	2nd Data	Remarks
CM8A Y=4000	0-9	00010	Route pattern number 010
CM8A Y=4001	0-9	00020	Route pattern number 020
CM8A Y=0010	1	01010	LCR pattern No. 010 + Trunk route No. 10
CM8A Y=0020	1	02020	LCR pattern No. 010 + Trunk route No. 20
CM8A Y=5010	176	08	To send the representative number
CM8A Y=5020	176	02	SIP subscriber number assigned by CM12 Y=46/47 (when no data is set to CM 12 Y=46/47, the calling party number is not sent)
CMBA Y=44	15	15	Not converted
CM12 Y=46	5001	5678	SIP subscriber number setting
CM12 Y=47	5001	00	SIP Local Office Code table number setting
CM50 Y=05	00	0501234	SIP Local Office Code 00
CMBA Y=32	05	05012341111	Representative number setting

REGISTER Sending by System Data

General Description

This feature allows the system to send REGISTER message (message for registering or deleting SIP account) to SIP server by entering commands to SV9300 with PCPro or CAT mode. Therefore, this feature provides more quick recovery from temporary fault.



Service Conditions

1. This feature is available only when SIP trunk is used.
2. Regardless of whether the SIP trunk is already registered in the SIP server or not, REGISTER registration or deletion request is sent to the SIP server by the system data setting of this feature.
3. The system data conditions are as follows.
 - The registration status can be read out by CMA7 Y=77.
 - After reading the registration status by CMA7 Y=77, REGISTER deletion request can be sent by entering "0" to the second data.
 - After reading the registration record by CMA7 Y=77, REGISTER registration request can be sent by entering "1" to the second data.

Programming

START	DESCRIPTION	DATA
CMA7	Specify the Registration status/Manual registration sending.	<ul style="list-style-type: none"> • Y=77 (1) 00-63: SIP trunk No. (2) 0 : Not registered/REGISTER deletion request is sent 1◀: Registration is completed/REGISTER is sent again
CMBA	Assign the SIP trunk registration expiry time to the SIP server.	<ul style="list-style-type: none"> • Y=71 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 1-8553600: 1 second-8553600 seconds (99 days) NONE◀ : 3600 seconds (1 hour)
RESET		
<p>NOTE 1: When no registration expiry time is specified by the SIP carrier, assigning this data is not required.</p>		
<p>NOTE 2: This data setting is effective only when CMBA Y=70 is set to “0”.</p>		
<p>NOTE 3: Set the time to cancel the registration after registering SIP trunk with this command to SIP server.</p>		
<p>NOTE 4: When the registration has been canceled by SIP server, re-register to SIP server for half the period of time set by this command (in case of 3600 seconds, set 1800 seconds).</p>		
<p>NOTE 5: When re-registration from SIP server is not executed during the period of time set by this command after the registration has been canceled by SIP server, call reception from the network to SIP is restricted.</p>		
<p>NOTE 6: This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to “0”).</p>		
A		

A	DESCRIPTION	DATA
CMBA	Specify whether to send a signal to require a deletion during a reset setting registration.	<ul style="list-style-type: none"> • Y=110 (1) 00-63: Profile No. assigned by CMA7 Y=71 (2) 0 : To provide 1 ◀: Not provided
	<div style="border: 1px solid black; border-radius: 15px; padding: 2px 10px; display: inline-block;">RESET</div>	
	<p>NOTE 1: <i>When this data is set to “0”, if REGISTER registration request is sent by CMA7 Y=77, the registration of the SIP trunk is disconnected from the SIP server for a moment.</i></p>	
	<p>NOTE 2: <i>This data is not available when using Point-to-Multipoint connection (when CMA7 Y=46 is set to “0”).</i></p>	
<u>END</u>		

Operating Procedure

For details of the CAT mode operations, refer to the “Command Manual”.

For details of the operations of PCPro, refer to the “PC Programming Manual”.

System Operation after Receiving an Unsupported Response Message by SIP Trunk

[9300V3 STEP2 software required]

General Description

This feature releases the trunk when the response for INVITE sent from a SIP trunk is an unsupported response.

Service Conditions

- (1) Conditions for releasing a trunk are as follows.
- When unsupported response is received as the response for INVITE on a SIP trunk.
 - When the second data of CM08>1231 is set to 0.
 - * When the second data of CM08>1231 is set to 1, a received response is ignored.

Programming

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM08</div>	Specify the system operation after receiving an unsupported response message by SIP trunk.	(1) 1231 (2) 0 : Disconnect call 1 ◀: Ignore the response
END		

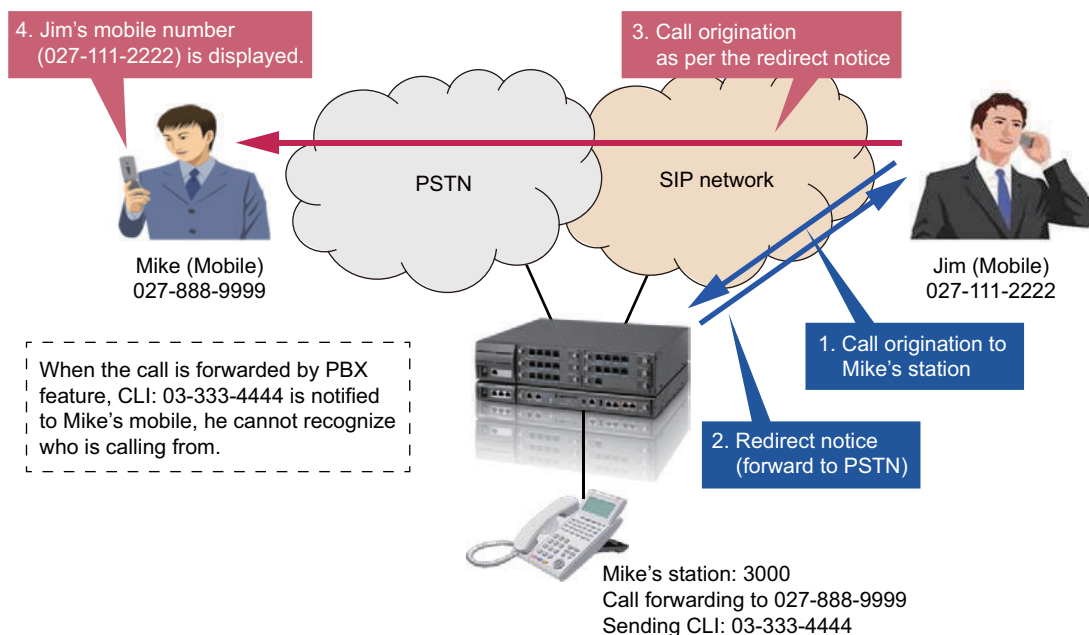
CLI Pass Through by SIP Redirect Notice “302 Moved Temporarily”

[9300V5 software required] [For New Zealand]

General Description

In order to meet the New Zealand SIP carrier specifications, SV9300 sends redirection message to the network. When the station which is set Call Forwarding-Outside and receives a call from SIP carrier network, SV9300 sends “302 Moved temporarily” to the network. After receiving this message, SIP carrier redirects the call to the Call Forwarding-Outside target number. The called party (Call Forwarding-Outside target number) can confirm caller ID information.

(This feature is available in New Zealand only.)



Service Conditions

1. The called station shall be set Call Forwarding to the outside party.
2. The following type of trunks can be used for Call Forwarding-Outside with this feature.
 - COT
 - DID
 - ISDN (BRI/PRI)
 - Q-SIG
 - T1
 - MFC
 - SIP which is set as C.O. line

3. This feature is not available when the call is answered and transferred to the outside party by using Automated Attendant/Direct Inward System Access (DISA).
4. In case of Multiple Call Forwarding is set, this feature is available if the final destination is outside party.
Ex: This feature is available when the incoming call is arrived to the Station-A in the following conditions.
 - * Station-A sets Call Forwarding to Station B
 - * Station-B sets Call Forwarding to outside party through COT
5. To enable this feature, the second data of “CM35 Y=366” shall be set as 0 for the incoming SIP trunk. If the SIP carrier network does not support redirection, the second data shall be set as 1.
6. This feature is accomplished even if no trunk line resources are available for Call Forwarding-Outside.
7. This feature does not depend on SIP trunk assignment or multiple SIP carrier environment and can be used.
8. If the above “1-5” conditions are not satisfied, this feature is not available and Call Forwarding feature is initiated.
9. The service conditions on Call Forwarding target number are as follows.
 - (a) SV9300 sends “302 Moved temporarily” message to the SIP carrier network including Call Forwarding-Outside target number which is added/deleted prefix number.
 - (b) The target number of Call Forwarding-Outside shall be set with trunk access code and phone numbers.
10. The following Call Forwarding services can be used with this feature.
 - Call Forwarding-All Calls
 - Call Forwarding-Busy line
 - Call Forwarding-No Answer
 - Call Forwarding-Logout (IP station)
 - Call Forwarding-Not Available (Standard SIP station)
 - Split Call Forwarding-All Calls
 - Split Call Forwarding-Busy Line
 - Split Call Forwarding-No Answer

11. When this feature is operated, the SMDR and OAI SMFN are executed as following conditions.

Service Feature	Type of Call Forwarding set by Station	SMDR/OAI SMFN Executions
SMDR	Call Forwarding-All Calls Call Forwarding-Busy Line Call Forwarding-Logout	Same conditions as Do Not Disturb/ Intercept Announcement (Received station number is not assigned).
	Call Forwarding-No Answer Call Forwarding-Not Available (Standard SIP station)	Same conditions as abandoned call/ station busy status.
OAI SMFN	Call Forwarding-All Calls Call Forwarding-Busy Line Call Forwarding-Logout	Same conditions as Do Not Disturb/ Intercept Announcement (Received station number is not assigned)/station busy status.
	Call Forwarding-No Answer Call Forwarding-Not Available (Standard SIP station)	Same conditions as abandoned call.

12. This feature is not available for MA transfer (MA on SIP trunk).

Programming

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM35</div>	Provide the SIP 302 Redirect. NOTE: <i>Set this data to the incoming trunk route of the SIP trunk.</i>	<ul style="list-style-type: none"> • Y=366 (1) 00-63: Trunk Route No. (2) 0: To provide
END		

State Distinction When Terminating a Call to Standard SIP Station via SIP Trunk

[9300V5 software required]

General Description

By setting the second data of CM35 Y=368 to 0 (To provide), this feature provides the state distinction function when terminating a call to a Standard SIP station via SIP trunk.

Service Conditions

1. Terminal types that the state distinction to a called terminal is provided are as follows.
 - Standard SIP Terminal
2. An error response is sent when the called terminal is in states such as Not Available, Power Off, or not logged in.
3. A state distinction by a trunk is not supported.

Programming

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM35</div>	<p>Provide the state distinction when terminating a call to Standard SIP station via SIP trunk.</p> <p>NOTE 1: <i>Set this command to the trunk route of a SIP trunk.</i></p> <p>NOTE 2: <i>This command is valid only when terminating a call via SIP trunk.</i></p>	<ul style="list-style-type: none"> • Y=368 (1) 00-63: Trunk Route No. (2) 0: To provide
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">END</div>		

SYSTEM DATA BACKUP

CAUTION

- If you operate as follows without system data backup after system data setting or service memory setting (registration of the features such as “Call Forwarding” and “Speed Dialing” from a station), the data that has been set is invalid.

You must execute the system data backup before the following operations.

- Turning Off the system
- System Reset (reset of CPU blade)
- Changing the CPU blade to Off-Line Mode
- Changing the CPU blade to On-Line Mode after system data setting under Off-Line Mode

In addition, the VRS data backup is also required when any data is recorded or changed in the VRS.

- You can execute the system data backup by the following two ways.
 - Executing the system data backup once a day at the time set by CM43 Y=5>00
(If no data is set, the default setting is 3:00 a.m.)
As the system data backup is performed, the VRS data backup is also performed at the same time.
 - Executing the system data backup from PCPro/CAT by CMEC Y=6>0: 0
- Do not reset the CPU blade while “SYSD” LED on the CPU blade is flashing.

(1) Regular system data backup

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CM43</div>	Assign the time for saving the system data once a day.	<ul style="list-style-type: none"> • Y=5 (1) 00: Regular Backup Time (2) HH MM <ul style="list-style-type: none"> HH : 00-23 (Hour) MM : 00-59 (Minute) 9999 : No backup the system data NONE◀: 0300 (3:00 a.m.)
END		

(2) Manual system data backup

START	DESCRIPTION	DATA
<div style="border: 1px solid black; padding: 5px; width: fit-content; margin: 0 auto;">CMEC</div>	Save the system data for backup to flash memory on the CPU blade.	<ul style="list-style-type: none"> • Y=6 (1) 0: System Data Backup (2) 0: Start to save <ul style="list-style-type: none"> 1: Now saving 3: Stand by
	<p>NOTE 1: You can assign only “0” to the second data. “1” is displayed while the system data is being copied.</p> <p>NOTE 2: Backup takes about 5 minutes on On-line/Off-line mode. While saving the system data to flash memory, “SYSD” LED on the CPU blade flashes.</p>	
END		

SYSTEM RESET

When you set the system data under Off-Line mode, system reset is required after data setting.

- (1) Set the SENSE switch to “0-4” (On Line) position and press the RESET switch on the CPU blade.
- (2) The lamps light in the following order.
 - “RUN” LED lights momentarily.
 - “S1” or “S2” LED flashes while copying the CPU program from the Flash Memory to the SDRAM.
 - “S1” or “S2” LED goes off.
 - “SYSD” LED lights while copying the system data and the voice data of Voice Response System from the Flash Memory to the SDRAM.
 - “S2” LED flashes while downloading the DSP firmware program.
 - “S2” LED goes off.
 - “SYSD” LED flashes while copying the voice data of Voice Response System from the Flash Memory to the SDRAM.
 - “RUN” LED flashes.
- (3) The operating mode has been changed to On-Line mode.
The backup data has been restored to the system.

SYSTEM SPECIFICATIONS

This appendix explains the specifications of the system.

Appendix

A

SYSTEM SPECIFICATIONS

System Specifications

ITEM	SPECIFICATIONS	REMARKS
Available Terminals	IP Station IP Multiline Terminal Soft Phone Standard SIP Terminal All Legacy Terminals Digital Multiline Terminal Single Line Telephone DSS/BLF Console SN716 DESKCON	
Number of Terminals	Maximum 1500 total IP Station : 1024 Legacy terminal: 1296	IP Station + Legacy terminal \leq 1500
Available Line/ Trunk Blade	All LT blades used in SV9300	288 ports per unit 1500 ports per system
OAI	Built-in OAI of CPU blade (LAN connection)	Connection via RS-232C is not available.
SMDR	CPU built-in SMDR on RS-232C CPU built-in SMDR on IP	
PCPro	Direct Connection (RS-232C) Modem Connection IP Connection	IP connection is available for System Upgrade.

IP SPECIFICATIONS

IP Specifications

ITEM	SPECIFICATIONS	REMARKS
Voice Encoding	G.711 G.722 G.723.1 (5.3k/6.3k) G.729a	64 Kbps PCM 7/16 kHz UEMCLIP CPU-MLQ/ACELP 8 Kbps CS-ACELP
VoIPDB	64 × 128 channels per blade Automatically seized per call	
FAX Communication Feature	G.711 pass-through G.726 pass-through T.38 (UDPTL)	VoIPDB/SIP
MODEM Communication Feature	G.711 pass-through G.726 pass-through	VoIPDB
DTMF Signal	In-band mode (voice pass through) Out-band mode (with H.245/ RFC2833)	VoIPDB/IP Station/SIP
Inter-office/Intra-office Signaling	H.245	IP Station-to-IP Station connection IP Station-to-VoIPDB connection
	PROTIMS over IP	IP Station-to-SV9300 connection
	CCIS over IP	Point-to-Multipoint connection
	SIP	Point-to-Point connection/ Point-to-Multipoint connection
Intra-office Connection Method	Point-to-Multipoint connection	IPT (P2P CCIS)
	Point-to-Point connection or Point-to-Multipoint connection	SIP trunk
Jitter Control	Dynamic Jitter Buffer/Static (300 ms.)	
Quality of Service (QoS)	<ul style="list-style-type: none"> • TOS, IP Precedence • DiffServ 	

Continued on next page

IP Specifications

ITEM		SPECIFICATIONS	REMARKS
LAN Interface		10BASE-T/100BASE-TX (ETHER-NET port) (TIA/EIA category 5) 10BASE-T/100BASE-TX/ 1000BASE-T (VOIP port) (TIA/EIA category 5 or larger)	Auto Negotiation is available. 100BASE-TX is recommended. For using 1000BASE-T, the cable needs to be larger than category 5e.
Echo Canceller (VoIPDB)		G.168 (128 ms. [Max. Consumption Time])	
Payload Size	IP Station/ IPT (P2P CCIS)	20 ms.-40 ms. (10 ms. increments)	
	SIP Trunk	20 ms.-40 ms. (G.711 μ -law/A-law, G.729a)	

VOICE QUALITY MEASURES IN VoIPDB

This appendix describes how to reduce echoes in SV9300 and the setup method.

In SV9300, the VoIPDB is used to relay the voice data on the IP network (on the IP packet) to the signals (PCM data) on TDM. The VoIPDB is equipped with an echo canceller circuit to suppress the echoes that influence calls. However, the echo canceller by itself cannot diminish echoes completely in some hardware configuration that contains lines on the TDM side.

This appendix describes how to reduce echoes in SV9300 and the setup method.

COUNTERMEASURES FOR ECHO

VoIPDB Channel Automatic Selection Procedure

(1) About VoIPDB Channel Automatic Selection

The echo canceller installed in the VoIPDB has a function that automatically learns (trains) the characteristics of echoes that were generated, corrects the echoes, and stores the information. However, one to two seconds are required for training and when the “actual call status” and “the echo characteristics that were stored” do not match, echoes may occur for one to two seconds from the start of a call. SV9300 reduces echoes by “storing the status of the echo characteristics and using the result as the key for selecting the next channel” for each VoIPDB channel, using the echo canceller circuit function.

In practice, the system classifies echo canceller training results into the models (levels) as indicated STEP2 of “(2) Setting Up Procedure for VoIPDB Channel Automatic Selection” according to the “connection destination on the TDM side” and stores the information for each VoIPDB channel. These levels are organized in the sequence starting from the level with the characteristics closest to those of the channel.

After the levels of all the VoIPDB channels are set, the VoIPDB channel storing the level matching with that of the connection destination is used in subsequent connection processing. If there is an VoIPDB channel whose level has not been set, the channel is captured first and trained.

(2) Setting Up Procedure for VoIPDB Channel Automatic Selection

[Cautions and Reference Information]

- When the setup described in this section is not carried out or the channel setting distribution ratio is incorrectly set, customers may experience echoes for 1 to 2 seconds until all the VoIPDB channels are used. However, by continuing the operation, the value is set to the optimum value automatically under the customer's use environment (Actually, if the number of VoIPDB channels is greater, the level matching probability increases so that the number of channels also influences the echo occurrence frequency).
- The training does not become valid unless the caller carries out a conversation.
- If a digital trunk such as ISDN is used for the main trunk line, the influence of the echo is less than that of the analog trunk.

Carry out training together with the VoIPDB canalization test at the final stage of construction.

STEP1: Determine the total number of VoIPDB channel.

STEP2: Assume the connection ratio (number of channels allocated) of IP station and IPT (P2P CCIS) to terminals that communicate on TDM side of VoIPDB. Match this total value and the total value that was determined in STEP1.

Level	Destination of VoIPDB (TDM side)	Connection Ratio	Number of VoIPDB Channel
00	COT/LD/OD (2W)	%	
01	LC/LLC	%	
02	BRT/PRT/CCT/OD (4W)	%	
03	DLC/BRT (when connecting ISDN Terminal)	%	
04	UM8000/CFT/VRS	%	
05	VoIPDB loopback (when two channels are used by SIP trunk/Standard SIP Terminal)	%	
06	Specific Service Restriction Class/Trunk Route	%	
07	Specific Service Restriction Class/Trunk Route	%	
08	Specific Service Restriction Class/Trunk Route	%	
09	Specific Service Restriction Class/Trunk Route	%	
	Total (total number of VoIPDB channel determined by STEP1)	100 %	

STEP3: Press the RESET switch on the CPU blade.

STEP4: Carry out “calls” with the call destination of each level as many times as the number of channels that were allocated.

NOTE: *Connection with a line that sends a voice announcement, music, or tones continuously from the call destination such as time information and weather information from a main trunk line is ineffective. The call destination could be silent. A call of several seconds is sufficient.*

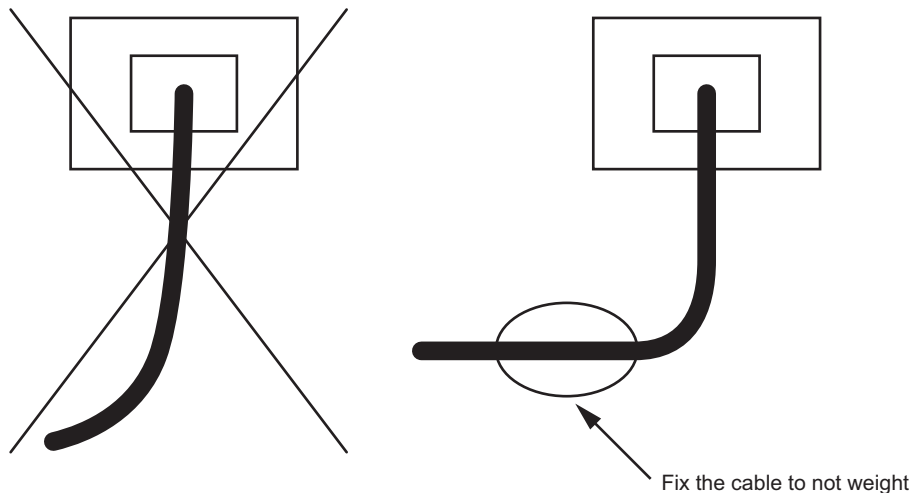
COUNTERMEASURES FOR OTHERS

Mixed Use of Voice and Data

When voice and data exist in the same LAN in mixed mode, the transmission priority order of the voice packet must be increased on the LAN to transmit a voice packet such as QoS setting and fragment certainly. See the related manual and router instruction manual for the setting method. Otherwise, voice may be interrupted or the voice quality deteriorates.

Connection with Hubs

Since the category 5 cable used for LAN is heavy and the cable sags if it simply inserted in a modular jack under its own weight, a load is applied to the modular jack, causing a contact fault. Fix the cable to prevent the weight from being applied to the modular jack directly.



TCP/UDP/RTP PORT NUMBER CONDI- TIONS

This appendix explains the default TCP/UDP/RTP port number and range of the available TCP/UDP/RTP port numbers used in SV9300.

TCP/UDP/RTP PORT NUMBER CONDITIONS

Tables below show the default TCP/UDP/RTP port number and range of the available TCP/UDP/RTP port numbers used in SV9300.

For IP Station

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for Login to DRS (UDP)	SV9300 VoIPDB (IP-PAD)	3456	Base Port: 1024-65534 Assigned by CM0B Y=1XX>31	1	Page 2-17
	IP Multiline Terminal	3455	Base Port: 1024-65535 Assigned by IP Station set up (Administrator Menu)	1	Page 2-101
Port for IP Multiline Terminal (D ^{term} 85) (PROTIMS) (UDP)	SV9300 VoIPDB (IP-PAD)	50000-52047 (For IP Multiline Terminal)	Base Port: 1024-65534 Assigned by CM0B Y=1XX>30 (Default: 50000)	1/terminal	Page 2-16
	IP Multiline Terminal	3458	Base Port: 1024-65534 Assigned by IP Station set up (Administrator Menu)	1	Page 2-101

Continued on next page

For IP Station

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for IP Multiline Terminal (DT700/DT800/DT900 Series, MH240) (UDP)	SV9300 VoIPDB (IP-PAD)	5080	Base Port: 1024-65534 Assigned by CM0B Y=1XX>32	1	Page 2-17
	IP Multiline Terminal (DT700/DT800/DT900 Series)	5060	Base Port: 1024-65534 Assigned by IP Station set up (Administrator Menu)	1	Page 2-106 Page 2-111
	SIP Wireless Terminal (MH240)	5080	Base Port: 1024-65534 Assigned by IP Station set up (Administrator Menu)	1	Page 2-122
Port for Standard SIP Station (UDP)	SV9300 VoIPDB (IP-PAD)	5070	Base Port: 00001-65534 Assigned by CM0B Y=1XX>33	1	Page 2-35
	Standard SIP Terminal	–	As per the terminal setting	1	–
Port for DNS for PUSH Notification (UDP)	SV9300 VoIPDB (IP-PAD)	Random	An idle port is selected automatically.	1	Page 2-71
Port for PUSH Notification Request (TCP)	SV9300 VoIPDB (IP-PAD)	55000-55127	Base Port: 1024-65407 Assigned by CM0B Y=1XX>170	128	Page 2-72

Continued on next page

For IP Station

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for PUSH Notification Request (for a Representative Call Notification)	SV9300 VoIPDB (IP-PAD)	55500-55627	Base Port: 1024-65407 Assigned by CM0B Y=1XX>182	128	<i>Page 2-81</i>
Port for Soft Phone (SP350) (UDP)	SV9300 VoIPDB (IP-PAD)	3456 (For Data Conference/ Instant Message between the offices)	Base Port: 1024-65534 Assigned by CM0B Y=1XX>31	1	<i>Page 2-19</i>
		5080 (For control)	Base Port: 1024-65534 Assigned by CM0B Y=1XX>32	1	<i>Page 2-17</i>
		5082 (For Presence service)	Base Port: 1024-65534 Assigned by CM0B Y=1XX>43	1	<i>Page 2-18</i>
	SP350	5060	As per the setting on SP350 side	1	–
	OW5000	5082 (For Presence service)	As per the setting on OW5000 side	1	–

For IPT (P2P CCIS)

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
P2P CCIS TCP Server Port	SV9300 VoIPDB (IP-PAD)	57000	Base Port: 1024-65534 Assigned by CM0B Y=1XX>34	1	Page 2-143
P2P CCIS TCP Client Port (TCP)	SV9300 VoIPDB (IP-PAD)	58000- 59023	Base Port: 1024-65534 Assigned by CM0B Y=1XX>35 Select by Round Robin method (Use from the lowest in blank port)	1024	Page 2-143

For SIP Trunk

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for SIP control (UDP)	SV9300 VoIPDB (IP-PAD)	5060	Base Port: 1024-65534 Assigned by CM0B Y=1XX>36	1	Page 2-258
Port for DNS (UDP)	SV9300 VoIPDB (IP-PAD)	Random	An idle port is selected automatically.	1	–

For Voice Control/Voice Packet

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for Voice Control (UDP)	SV9300 VoIPDB (IP-PAD)	56000	Base Port: 1024-65534 Assigned by CM0B Y=1XX>39	1	Page 2-129
	IP Multiline Terminal	4000	Base Port: 1024-65535 Assigned by IP Station Set up (Administrator Menu)	1	Page 2-101
Port for P2P Voice Control (TCP)	SV9300 VoIPDB (IP-PAD)	6100 (For Server)	Base Port: 1024-65534 Assigned by CM0B Y=1XX>37	1	Page 2-129
		6200-6327 (For Client)	Base Port: 1024-65534 Use 128 ports in Round Robin method assigned by CM0B Y=1XX>38	129	Page 2-130
	IP Multiline Terminal	1024-65535	Base Port: 1024-65535 Assigned by Round Robin method (Use from the lowest in blank port) Change by configuration	1	Page 2-101
Port for Voice Packet (RTP)	SV9300 VoIPDB (IP-PAD)	10000-10255	Base Port: 1024-65534 Assigned by CM0B Y=2XX>40 * Assign an even number to the base port (default: 10000).	256 (2/channel)	Page 2-128
	Communication party	—	As per the communication party	—	—

For Communication Between Units

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for Communication between Units (UDP)	SV9300 VoIPDB (IP-PAD)	3300-3401	Base Port: 1024-65534 Assigned by CM0B Y=1XX=40	102	—
Port for Communication between Units for Standard SIP Station Control (UDP)	SV9300 VoIPDB (IP-PAD)	5076	Fixed	1	—
Port for Communication between Units for SIP Trunk Control (UDP)	SV9300 VoIPDB (IP-PAD)	5060	Communication between units uses the same port as the SIP trunk carrier uses. Base Port: 1024-65534 Assigned by CM0B Y=1XX>36	1	—

For Maintenance/Others

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for SNMP (UDP)	SV9300 VoIPDB (IP-PAD)/ Maintenance Port	161	Fixed	1	—
	SNMP Manager	161-162	Fixed	2	—

Continued on next page

For Maintenance/Others

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for Program Download (TCP)	SV9300 VoIPDB (IP-PAD)/ Maintenance Port	3000-3019 (Data)	Fixed	20	—
	FTP Server	20-21	As per the FTP server setting Base Port: 1024-65534 Assigned by CM0C Y=11>XX01	2	—
Port for communication between ACT-STBY for Dual CPU System (TCP)	SV9300 Maintenance Port	3410	Base Port: 1024-65534 Assigned by CM0B Y=0XX>42	1	—
Port for PCPro Connection (TCP)	SV9300 VoIPDB (IP-PAD)/ Maintenance Port	60000	Fixed	1	—
Port for SMDR (TCP)	SV9300 VoIPDB (IP-PAD)/ Maintenance Port	60010	Enable to select the VoIPDB (IP-PAD)/ Maintenance Port by CM0B Y=0XX>92	1	—

Continued on next page

For Maintenance/Others

PORT TYPE	DESTINATION	DEFAULT PORT No.	REMARKS	NUMBER OF PORT USED	REFERENCE PAGE
Port for OAI (TCP)	SV9300	1024	Enable to select the OAI Port by CM0B Y=001>41 (default: 60030)	1	-
	VoIPDB	1025			
	(IP-PAD)/	1039			
	Maintenance Port	60030			
Port for PMS (TCP)	SV9300	60050	Enable to select the VoIPDB (IP-PAD)/ Maintenance Port by CM0B Y=0XX>93	1	-
Port for SNTP Client (TCP)	SV9300	49152-	Fixed	1	-
	VoIPDB (IP-PAD)	65534			
Port for VoIPDB control (UDP)	NTP Server	123	Fixed	1	-
	SV9300	512	Fixed	1	-
Port for Web Server (TCP)	VoIPDB	513	Fixed	1	-
	VoIPDB/ Maintenance port	80/443 NOTE	Base port: 1024-65534 Assigned by CM0B Y=001>22 (VoIPDB/Maintenance port)	1	-

NOTE: The default Port No. varies depending on the connection mode to use.

- For HTTP (CM0B Y=001>23:1◀): Port No. = 80

- For HTTPS (CM0B Y=001>23:0): Port No. = 443*

* Port No. 443 is available for 9300V8 software or later.

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DT700/DT800 SERIES CONFIGURATION MENU LIST

This appendix explains the configuration menu used for DT700/DT800 Series configuration settings.

As for the configuration menu used for DT900 Series configuration settings, refer to the DT900 Setup Guide.

DT700 SERIES CONFIGURATION MENU LIST

This section briefly explains about the each parameter and the meaning of DT700 Series configuration menus.

* This section describes only the configuration menu list of the following latest firmware version.

- DT730G Series Configuration Menu List (FW 1.0.0.0 or later) [Page D-3](#)
- DT750/DT730/DT710 Series Configuration Menu List (FW 5.0.0.0 or later) [Page D-24](#)
- DT770G Configuration Menu List [Page D-50](#)

DT730G Series Configuration Menu List (FW 1.0.0.0 or later)

This section lists the configuration menu used for DT730G Series configuration setting. DT730G (12/24CG), and DT730G (12/24DG) columns on the tables below show whether the menu is available or not for each terminal type.

LEGEND	X: Available	N: Not available	-: Not applicable
---------------	--------------	------------------	-------------------

NOTE 1: Up to four Service menus are displayed. Service menu is not displayed when a service URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL].

NOTE 2: Phone menu is displayed when a URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL]. Usually, "Application Name" is displayed when a URL is registered.

NOTE 3: The Config menu is not displayed on the menu window. To open the Config menu, press digit key "0". In addition, by pressing "Hold", "Transfer", "*", "#" key in this order in an idle state of telephone, you can access to the login screen of Configuration Menu.

1. History

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
1. History	Lists/deletes the history of outgoing/incoming calls	-	X	X

When "1. History" is selected, the "History menu" of SV9300 is displayed.

[History]	
NEC ICHIRO	[300]
3:09 PM	TUE 18 OCT 2016
Outgoing	Incoming

NOTE: In SV9300, the built-in Call History of a telephone cannot be used. Call History on SV9300 side is used.

2. Directory

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
2. Directory	Displays Terminal Phonebook	-	X	X

When “2. Directory” is selected, the “Directory service” screen of SV9300 is displayed.

[Directory]

Name>>

Set Common Personal

NOTE: In SV9300, the built-in Directory of a telephone cannot be used. The Directory service on SV9300 side is used.

3. Service

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
3. Service	Starts up XML browser.	-	X	X
1. Service 1	Starts up XML browser.	-	X	X
2. Service 2		-	X	X
3. Service 3		-	X	X
4. Service 4		-	X	X

NOTE: At most four services are displayed on the menu. A service is not displayed when a service URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL].

4. Setting

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
1. User Setting	Sets user interface related.	-	X	X
1. Incoming Call	Sets ringing and lamp indication data for incoming call.	-	X	X
1. Ring Volume (Setting prohibited)	-	-	-	-
2. Offhook Ring (Setting prohibited)	-	Enable	-	-
1. Disable (Setting prohibited)	-			
2. Enable	-			
3. Headset Ring	Selects whether to use headset ringing.	Disable	X	X
1. Disable	Headset ringing is OFF.			
2. Enable	Headset ringing is ON.			
4. Ring Tone	Sets Ringing Tone for incoming call termination.	-	X	X
1. External Call NOTE 1	Sets Ringing Tone for external call termination.	Automatic	X	X
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.			
2. Internal Call NOTE 1	Sets Ringing Tone for station call termination.			
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.			
3. IM (Setting prohibited)	Not used.	-	-	-

4. Setting (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
5. Illumination NOTE 2	Sets lamp indication pattern for incoming call termination.	-	X	X
1. External Call NOTE 3	Sets Illumination for External Call.	-	X	X
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.	Automatic	X	X
1. Disable	Lamp indication is OFF.			
2. Red	Specifies the color of lamp indication for incoming call termination.			
3. Green				
4. Blue				
5. Yellow				
6. Purple				
7. Light Blue				
8. White				
9. Rotation				
2. Internal Call NOTE 3		Sets Illumination for Internal Call.	-	X
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.	Automatic	X	X
1. Disable	Lamp indication is OFF.			
2. Red	Specifies the color of lamp indication for incoming call termination.			
3. Green				
4. Blue				
5. Yellow				
6. Purple				
7. Light Blue				
8. White				
9. Rotation				
3. IM (Setting prohibited)		-	-	-
6. Headset Ring Volume (Setting prohibited)	-	-	-	-

4. Setting (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
2. Talk	Sets call related data.	-	X	X
1. RTP Alarm NOTE 4	Sets whether to allow the alarm for detecting packet loss in receiving speech packets.	Automatic	X	X
0. Automatic	Applies RTP alarm setting on the SV9300.			
1. Disable	Alarm is OFF.			
2. Enable	Alarm is ON.			
2. DTMF Tone (Setting prohibited)	-	Automatic	-	-
0. Automatic	-			
1. Disable (Setting prohibited)	-			
2. Enable (Setting prohibited)	-			
3. Key Touch Tone	Sets touch tone for pressing digit key/touch-sensitive panel.	Automatic NOTE 5	X	X
0. Automatic	Sets Automatic to key touch tone.			
1. No Tone	Sets key touch tone disabled.			
2. Tone	Sets key touch tone enabled.			
3. Voice	Enables the key touch tone by voice recognition.			
4. Hold Music	Sets Music-on-Hold.	Default	X	X
1. Default	Applies the default music (Minuet).			
2. Download	Applies the downloaded music.			
5. Prefix (Setting prohibited)	-	-	-	-
1. Prefix 1 ~ 3. Prefix 3 (Setting prohibited)	-	-	-	-

4. Setting (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
3. Display	Sets display related data.	-	X	X
1. Time Format (Setting prohibited)	-	Automatic NOTE 6	-	-
0. Automatic	-			
1. 12 hour Indication (Setting prohibited)	-			
2. 24 hour Indication (Setting prohibited)	-			
2. Local Volume	Selects whether to display volume gage when adjusting the volume level.	Automatic NOTE 7	X	X
0. Automatic	Sets Automatic for volume gage display.			
1. Disable	Volume gage is not displayed.			
2. Enable	Volume gage is displayed.			
3. Screen Saver	-	-	-	-
1. Screen Saver Mode	Selects whether to use Screen Saver.	Disable	X	X
1. Disable	Screen Saver is not used.			
2. Enable	Screen Saver is used.			
2. Wait Time	Sets wait time until Screen Saver activation (1-999 min.).	120	X	X
4. Back Light	Switches ON/OFF the backlight on DT730G (12/24DG).	Enable NOTE 8	-	X
1. Disable	Backlight is OFF.			
2. Enable	Backlight is ON.			
4. Back Light	Turns up/down the backlight on DT730G (12/24CG).	Level 2 NOTE 8	X	-
1. Level 1 (Bright) - 4. Level 4 (Dark)	Select the bright level of backlight.			
5. Font Size NOTE 9	Switches the font size on configuration menu.	Standard	X	X
1. Standard (28 dots (12/24CG)/16 dots (12/24DG))	Selects Standard or Small of font size.			
2. Small (24 dots (12/24CG)/12 dots (12/24DG))				

4. Setting (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
6. Language	Selects the language on date and time display.			
0. Automatic	Sets Automatic for language setting on date and time display.			
1. Japanese	Selected language is displayed on date and time display.	Automatic NOTE 10	X	X
2. NEC English				
3. NEC Portuguese				
4. NEC Spanish				
5. NEC French				
6. German				
7. Italian				
8. Dutch				
9. Norwegian				
10. Danish				
11. Swedish				
12. Greek				
13. English				
14. Portuguese				
15. Spanish				
16. French				
17. Russian				
18. Turkish				
19. Romania				
20. Polish				
21. Catalan				
22. Korean				
23. Download 1	Selected download language is displayed on date and time display. NOTE 11			
24. Download 2				
25. Download 3				
26. Download 4				
7. Advanced	Sets other data for display.	-	X	X
1. Backlight Fade Control	Sets whether Backlight Fade Control is in service, when turning on/off the backlight of the display.	Enable	-	X
1. Disable	Disables Backlight Fade Control.			
2. Enable	Enables Backlight Fade Control.			
1. Wallpaper	Sets the Home screen background.	Default	X	-
1. Default	Sets the default screen background.			
2. Download	Sets the download data file.			
2. Shortcut Icon (Setting prohibited)	-	-	-	-
3. Font Color	Sets the font color for Time Display/Feature Information Display area. * You can check the color while selecting.	Color 2	X	-
1~16. Color 1 ~ Color 16	Sets Font Color 1 ~ Font Color 16.			

4. Setting (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
4. Turn Off Display	Sets the data when turning off the backlight of the display automatically.	-		
1. Turn Off Display Mode	Sets whether Turn Off Display Mode is in service.	App Priority	X	-
1. Disable	Disables Turn Off Display Mode.			
2. Enable	Enables Turn Off Display Mode.			
3. App Priority	Sets to depend on an application.			
2. Wait Time	Sets the time until turning off the display (1-999 min.).	180		
5. Backlight Fade Control	Sets whether Backlight Fade Control is in service, when turning on/off the backlight of the display.	Enable	X	-
1. Disable	Disables Backlight Fade Control.			
2. Enable	Enables Backlight Fade Control.			
4. Change Password NOTE 12	Assigns the password of the terminal, used for data clear, security, and login of user mode.	0000	X	X
5. Security (Setting prohibited)	-	-	-	-
6. Peripherals	Setting for using peripheral equipment of terminal.	-	X	X
1. USB Memory	Setting for safely removing USB memory from the terminal.			
1. Remove				
7. Usability	Sets the operation of Help key while pop-up window is displayed.	-	X	X
1. Help Key Mode	Setting for Help Key Mode (SubMenu/Popup Window)	SubMenu	X	X
1. SubMenu	Sets the Help key which displays sub menu		X	X
2. Popup Window	Sets the Help key which displays pop-up window		X	X
0. Setting Reset	Initializes all the assigned personal information on the terminal to factory default.	-	X	X

4. Setting (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
2. Download	Sets download related data.			
1. Download Files	Downloads files of Music-on-Hold, Ringing Tone, and Phonebook.	-	X	X
1. Hold Music	Downloads a sound file used for Music-on-Hold.	MOH.wav	X	X
2. Ring Tone	Downloads a sound file used for Ringing Tone.	-		
1. Download 1	Downloads Melody 1 ~ Melody 3.	Melody1.wav	X	X
2. Download 2		Melody2.wav		
3. Download 3		Melody3.wav		
3. Directory (Setting prohibited)	-	-	-	-
4. Wallpaper	Downloads a file used for wallpaper.	WallPaper.bmp	X	-
2. Download Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X
3. Protocol	Specifies the protocol for file downloading.			
1. FTP	FTP is used.	FTP	X	X
2. TFTP	TFTP is used.			
4. FTP Settings	Sets data for accessing FTP server.			
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.			
3. Folder	Specifies the directory to save the download file.			
3. Data Backup/Restore	Makes backup of terminal data or restores from backup file.	-	X	X
1. Data Backup	Saves the terminal setting to backup file.	PersonalData.tgz		
2. Data Restore	Restores the terminal data by reading out the saved backup file.	PersonalData.tgz	X	X
3. Server Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X
4. Protocol	Specifies the protocol for file downloading.			
1. FTP	FTP is used.	FTP	X	X
2. TFTP	TFTP is used.			
5. FTP Settings	Sets data for accessing FTP server.			
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.			
3. Folder	Specifies the directory to save backup file.			

NOTE 1: To provide the Distinctive Ringing of external call/internal call by SV9300, be sure to use with default (Automatic). When "Automatic" is set, by the system data setting (CM08>390) of SV9300, Ringing Tone can be changed by the system data (CM15 Y=491, CM35 Y=034, 164) or terminal operations ("Feature" + "3") (default: as per CM15 Y=491, CM35 Y=034, 164).

NOTE 2: Distinctive lamp indication for an external/internal call is available in SV9300.

NOTE 3: When "Automatic" is set, by the system data setting (CM08>144) of SV9300, the illumination color can be changed to Red or Green (default: Red).
Set the "0. Automatic" to use color coding by 7-color LED.

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- NOTE 4:** *When “Automatic” is set, RTP Alarm depends on the system data setting (CM08>1004) of SV9300 (default: To send).*
- NOTE 5:** *Follows the setting (CM15 Y=291) of SV9300 side when “Automatic” is set. In all other cases, follows the setting of a terminal side.*
- NOTE 6:** *Time Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.*
- NOTE 7:** *A setting for Enable is applied when Automatic has been set.*
- NOTE 8:** *Back Light menu becomes effective when the system data (CM15 Y=485) of SV9300 is set to other than 0 (Always off).
Back Light of ten key cannot be used when terminals and option equipment for power failure are implemented.*
- NOTE 9:** *When a font size is changed, press the Cursor key and return to the previous menu, then the font size becomes effective.*
- NOTE 10:** *The setting of display language on the Clock Display screen depends on the system data settings (CM12 Y=63, CM04 Y=00) of SV9300.*
- NOTE 11:** *If “Language Pack” files are not downloaded, these parameters are not displayed.*
- NOTE 12:** *Change Password is used for password for User Mode of Web programming and security unlock. Be sure to change the default password.*

5. Phone

Depending on the Home URL setting by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL], this menu is displayed as follows.

- When Home URL is not registered, home screen is displayed and the menu name is displayed as “Phone”.
- When Home URL is registered, XML application screen is displayed, and the menu name is displayed as the application name to be described in the XML application.

0. Config

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
1. Network Settings	Sets network related data.	-	X	X
1. DHCP Mode	Selects whether to use DHCP protocol.	Enable	X	X
1. Disable	DHCP is not used.			
2. Enable	DHCP is used.			
2. IP Address	Sets DT700 Series terminal's IP address.	0.0.0.0	X	X
3. Default Gateway	Sets default router's IP address.	0.0.0.0	X	X
4. Subnet Mask	Sets subnet mask.	0.0.0.0	X	X
5. DNS Address	Sets DNS server's IP address.	0.0.0.0	X	X
6. Advanced Settings	Sets network expanded data.	-	X	X
1. LAN Port Settings	Sets LAN port.	-	X	X
1. Speed&Duplex	Sets communication speed of Ether port.	Autonego	X	X
1. Autonego	Autonegotiation			
2. 100M Full	100Mbps in full-duplex mode			
3. 100M Half	100Mbps in half-duplex mode			
4. 10M Full	10Mbps in full-duplex mode			
5. 10M Half	10Mbps in half-duplex mode			
2. VLAN Mode	Selects whether to use VLAN on terminal.	Disable	X	X
1. Disable	VLAN is not used.			
2. Enable	VLAN is used.			
3. VLAN ID	Sets VLAN ID of terminal [1-4094].	7	X	X
4. VLAN Priority	Sets VLAN Priority of terminal [0-7].	5	X	X
2. PC Port Settings	Sets PC port.	-	X	X
1. Speed&Duplex	Sets communication speed of PC port.	Autonego	X	X
1. Autonego	Autonegotiation			
2. 100M Full	100Mbps in full-duplex mode			
3. 100M Half	100Mbps in half-duplex mode			
4. 10M Full	10Mbps in full-duplex mode			
5. 10M Half	10Mbps in half-duplex mode			
2. Port VLAN Mode	Selects whether to use port VLAN of PC port.	Disable	X	X
1. Disable	Port VLAN of PC port is not used.			
2. Enable	Port VLAN of PC port is used.			
3. Port VLAN ID	Sets port VLAN ID of PC port [1-4094].	1	X	X
4. Port VLAN Priority	Sets port VLAN priority of PC port [0-7].	0	X	X
5. Port Available	Selects whether to use PC port.	Automatic	X	X
0. Automatic	Sets Automatic for PC port.			
1. Disable	PC port is not used.			
6. EAPOL Forwarding	Select whether to forward EAPOL packet from PC port.	Disable	X	X
1. Disable	Not forward EAPOL packet from PC port.			
2. Enable	Forwards EAPOL packet from PC port.			
7. PC Port Security	Select whether to clear authentication status of the terminal which is connected to PC port for authentication switch when the link of PC port has been down.	Enable	X	X
1. Disable	Not clear the authentication status.			
2. Enable	Clear the authentication status.			

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
3. CDP Mode (Setting prohibited)	-	-	-	-
4. Type of Service	Sets ToS value.	-	X	X
1. RTP	Sets ToS value for RTP in Hex.	A0		
2. SIP	Sets ToS value for SIP in Hex.	C0		
5. Self Port Settings	Sets each port for DT700 Series.	-	-	-
1. RTP Self Port	Sets RTP port for DT700 Series [1024-65520 (except 59986-60000)]. NOTE 1 * Odd port numbers cannot be set.	3462	X	X
2. SIP Self Port	Sets SIP port for DT700 Series [1024-65534]. NOTE 1	5060	X	X
6. LLDP Settings	Sets information related to LLDP.	-	X	X
1. LLDP Mode	Sets whether to transmit/receive LLDP packets.	Enable	X	X
1. Disable	DT700 Series does not transmit/receive LLDP packets.			
2. Enable	DT700 Series transmits/receives LLDP packets.			
2. Auto Setting Mode	Sets whether to perform data setting automatically when LLDP packets are received.	Enable	X	X
1. Disable	DT700 Series does not perform data setting automatically when LLDP packets are received.			
2. Enable	DT700 Series performs data setting automatically when LLDP packets are received.			
3. RX Waiting Time	For Auto Setting, sets the waiting time until DT700 Series receives LLDP packets (1-60 seconds).	15	X	X
4. Transmit Interval	Sets intervals to transmit LLDP packets (5-32768 seconds).	30	X	X
5. Hold Multiplier NOTE 2	Sets a multiplier determining the time to live for the LLDP packets to be transmitted (2-10 multiples).	4	X	X
6. Fast Start Count	Sets the number of times to transmit LLDP packets at one-second intervals after DT700 Series starts up (1-10 times).	3	X	X
7. Transparent Mode	Sets whether Transparent Mode of LLDP packets is enabled between the PC port and the LAN port.	Enable	X	X
1. Disable	Transparent Mode of LLDP packets is disabled.			
2. Enable	Transparent Mode of LLDP packets is enabled.			
8. Asset ID	Sets the Asset ID for LLDP packets to be transmitted. * “-” is displayed in default settings.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
7. Spare IP Settings (Setting prohibited)	-	-	-	-
1. Spare/Backup IP Mode (Setting prohibited)	-	Disable	-	-
1. Disable	-			
2. Spare IP (Setting prohibited)	-			
3. Backup IP (Setting prohibited)	-			
2. Network Settings (Setting prohibited)	-	-	-	-
1. IP Address (Setting prohibited)	-	0.0.0.0		
2. Default Gateway (Setting prohibited)	-	0.0.0.0		
3. Subnet Mask (Setting prohibited)	-	0.0.0.0		
4. DNS Address (Setting prohibited)	-	0.0.0.0		
3. SIP Settings (Setting prohibited)	-	-	-	-
1. 2nd Server Address (Setting prohibited)	-	0.0.0.0	-	-
2. 3rd Server Address (Setting prohibited)	-	0.0.0.0	-	-
3. 4th Server Address (Setting prohibited)	-	0.0.0.0	-	-
4. 2nd Server Port (Setting prohibited)	-	0	-	-
5. 3rd Server Port (Setting prohibited)	-	0	-	-
6. 4th Server Port (Setting prohibited)	-	0	-	-
2. SIP Settings	Sets SIP related data.	-	X	X
1. SIP User (Setting prohibited)	-	-	-	-
1. User ID (Setting prohibited)	-	-	-	-
2. Password (Setting prohibited)	-	-	-	-
3. Extension Number (Setting prohibited)	-	-	-	-
4. Backup Logon (Setting prohibited)	-	Disable	-	-

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
2. Server Address & URI	Sets IP address or URI of SIP server. NOTE 3	-	X	X
1. 1st Server Address	Sets IP address or fully qualified domain name (FQDN) of 1st ~ 4th SIP server. NOTE 4	0.0.0.0	X	X
2. 2nd Server Address				
3. 3rd Server Address				
4. 4th Server Address				
5. Registrar Destination (Setting prohibited)	-	0.0.0.0	-	-
1. 1st Register Address (Setting prohibited)	-		-	-
2. 2nd Register Address (Setting prohibited)				
3. 3rd Register Address (Setting prohibited)				
4. 4th Register Address (Setting prohibited)				
3. Access Mode	Sets access mode for SIP server.	Normal	X	X
1. Normal	Normal access			
2. Remote	Remote access			
4. SIP Server Port	Sets SIP server port [1024-65535].	5060	X	X
1. 1st Server Port	Specifies the destination port number of sending packet to 1st ~ 4th SIP server. NOTE 5			
2. 2nd Server Port				
3. 3rd Server Port				
4. 4th Server Port				
5. Registrar Destination (Setting prohibited)	-		-	-
1. 1st Register Port (Setting prohibited)	-			
2. 2nd Register Port (Setting prohibited)				
3. 3rd Register Port (Setting prohibited)				
4. 4th Register Port (Setting prohibited)				

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
5. Area ID	Sets the area (country) using DT700 Series.			
0. Automatic	The SV9300 side setting is applied when Automatic is selected.			
1. Japan	Selects the area (country) NOTE: Area IDs allowed on the SV9300 can be selected.	Automatic	X	X
2. America (Canada and North America)				
3. Australia				
4. Others				
5. Hong Kong				
6. Malaysia				
7. Singapore				
8. Mexico				
9. Taiwan				
10. New Zealand				
11. Korea				
12. Brazil				
13. China				
14. Thailand				
15. Thailand (EGAT)				
16. Sri Lanka				
17. Germany				
18. Italy				
19. Netherlands				
20. Catalonia				
21. Denmark				
22. Sweden				
23. Poland				
24. Czech				
25. Norway				
26. France				
27. Portugal				
28. Spain				
29. Austria				
30. Belgium				
31. UK				
32. Greece				
33. Switzerland				
34. South Africa				
35. Brazil (EMEA)				

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
6. Extra Function (Setting prohibited)	-	-	-	-
1. Retransmit Interval (Setting prohibited)	-	Standard	-	-
1. Standard	-			
2. 6 Seconds (Setting prohibited)	-			
2. Keeping Call (Setting prohibited)	-	Enable	-	-
1. Disable (Setting prohibited)	-			
2. Enable	-			
3. Howler Tone (Setting prohibited)	Selects whether to send howler tone. NOTE 6	Disable	-	-
1. Disable	Howler tone sending is OFF.			
2. Enable (Setting prohibited)	Howler tone sending is ON.			
7. Encryption	Sets encryption data.	-	X	X
1. Authentication Mode	Selects whether to use authentication.	Disable	X	X
1. Disable	Authentication is not used.			
2. Enable	Authentication is used.			
2. One Time Password	Sets One Time Password.	-	X	X
3. Authentication Status	Displays authentication status.	-	X	X
8. NAT Traversal (Setting prohibited)	-	-	-	-
1. NAT Traverse Mode (Setting prohibited)	-	Disable	-	-
1. Disable				
2. Dynamic (Setting prohibited)				
3. Static (Setting prohibited)				
2. Network Area Name (Setting prohibited)	-	-	-	-
3. WAN Settings (Setting prohibited)	-	0.0.0.0	-	-
1. WAN Mate IP Address (Setting prohibited)				
2. WAN SIP Mate Port (Setting prohibited)		5060		
3. WAN Self IP Address (Setting prohibited)		0.0.0.0		

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
3. Maintenance Settings	Sets maintenance related data.	-	X	X
1. Download Menu	Downloads specified file to DT700 Series.	-	X	X
1. Download Files	Specifies download file name for each file type.	-	X	X
1. Boot & Program	Downloads boot and program.	12/24CG: itlispvc.tgz 12/24DG: itlispvg.tgz	X	X
2. Config	Downloads configuration file.	config.tgz	X	X
3. Voice File	Downloads voice file.	voice.tgz	X	X
4. Language Pack	Downloads translation data file.	LangPack. tgz	X	X
2. Download Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X
3. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
4. FTP Settings	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.	-	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X
2. Hard Reset	Resets DT700 Series without data saving.	-	X	X
3. Data Clear	Initializes DT700 Series selecting the initialization mode.	-	X	X
1. Factory Value	Initializes DT700 Series to factory default.	-	X	X
2. Terminal Data	Clears terminal data on DT700 Series.	-	X	X
3. Personal Data	Clears personal data on DT700 Series.	-	X	X
4. Adjust	Adjusts LCD, audio, and button kit.	-	X	X
1. Default Audio	Sets audio data.	-	X	X
1. Sidetone Volume	Sets Side Tone volume [0-7].	4	X	X
2. Send Gain	Sets sending gain of handset [0-31].	24	X	X
3. Receive Bias	Sets receiving bias of handset [0-7].	0	X	X
4. Receive Gain	Sets receiving gain of handset [0-31].	22	X	X
5. Talk Characteristics	Sets talk characteristics of handset.	-	X	X
1. Noise Suppression	Sets whether to enable the Noise Suppressor. (Set "1. Disable" in normal operation.)	Disable	X	X
1. Disable	Disables the Noise Suppressor.			
2. Enable	Enables the Noise Suppressor.			
2. Handsfree Mode	Sets a send gain mode of handsfree call.	-	X	X
1. Retro Mode	Sets to Retro mode.	Retro Mode		
2. New Mode	Sets to New mode.			

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
2. Key Kit Type	Sets additional button kit data.	-	X	X
1~5. Kit 1, 2, 8, 9, 10 NOTE 7	Key 1, 2, 8, 9, 10 on button kit are used.	Kit 1	X	X
1. Line Key Type	(Do not change the factory default value of this parameter.)	-	X	X
1. Line 12	Sets to 12-line of DT730G.		X	X
2~7. Line 24, 32, 8, 6, 2, 0	Not used.		-	-
3. LCD Contrast	Adjusts LCD Contrast.	Area Contrast (Narrow)	-	X
0. Area Contrast				
1. Wide				
2. Narrow				
5. Auto config	Selects whether to use auto-config function.	Enable	X	X
1. Disable	Auto config is not used.			
2. Enable	Auto config is used.			
6. Data Maintenance	Sets data maintenance related.	-	X	X
1. Backup	Executes data backup.	-	X	X
1. All Data	Executes backup for all the configuration setting and data.	AllData.tgz	X	X
2. Terminal Data	Executes backup for terminal setting and data.	Terminal Data.tgz	X	X
3. Personal Data	Executes backup for personal setting and data.	Personal Data.tgz	X	X
2. Restore	Executes restoration.	-	X	X
1. All Data	Restores all the configuration setting and data.	AllData.tgz	X	X
2. Terminal Data	Restores terminal setting and data.	Terminal Data.tgz	X	X
3. Personal Data	Restores personal setting and data.	Personal Data.tgz	X	X
3. Server Address	Sets IP address of the server that executes data backup/restoration.	0.0.0.0	X	X
4. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
5. FTP Settings	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.	-	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
7. Simple Test NOTE 8	Changes the terminal to test mode.	-	X	X
1. Call Test	Call Test	-	X	X
2. Call Test Address	Sets an IP address of the opposite terminal to perform the call test.	10.136.20.188	X	X
3. Key Test 1 (Random)	Tests to press the random button.	-	X	X
4. Key Test 2 (Sequential)	Tests to press the sequential button.	-	X	X
5. Loop Back Test	Tests the voice loop back.	-	X	X
1. Handset - Hand Speaker	Tests to speak and receive on the hand speaker.	-	X	X
2. Handset - Speaker	Tests to speak on the handset and receive on the speaker.	-	X	X
3. Headset - Head Speaker	Tests to speak and receive on the head speaker.	-	X	X
4. Headset - Speaker	Tests to speak on the headset and receive on the speaker.	-	X	X
5. MIC - Hand Speaker	Tests to speak on the MIC and receive on the hand speaker.	-	X	X
6. MIC - Head Speaker	Tests to speak on the MIC and receive on the head speaker.	-	X	X
7. MIC - Speaker	Tests to speak on the MIC and receive on the speaker.	-	X	X
6. Speaker Test	Tests the speaker.	-	X	X
1. Japan	A DT for Japan is heard in maximum volume.	-	X	X
2. America	A DT for North America is heard in maximum volume.	-	X	X
7. Receiver Test	Receiver Test for a handset	-	X	X
4. Security	Sets security related data.	-	X	X
1. Web Programming	Selects whether to use Web Programming.			
1. Disable	Web Programming is disabled.	Enable	X	X
2. Enable	Web Programming is enabled.			
2. Push Server	Sets XML's Push server related data.	-	X	X
1. Push Server Access	Sets whether to access XML's Push server.			
1. Disable	Access is disabled.			
2. Enable	Access is enabled.	Enable	X	X
3. Enable (All Client) NOTE 9	Access is enabled. Select this item to set more than one XML's Push server.			
2. Client IP Address	IP address of XML's Push server.			
1. Client 1	Sets IP address of 1st XML's Push server.			
2. Client 2	Sets IP address of 2nd XML's Push server.	0.0.0.0	X	X
3. Client 3	Sets IP address of 3rd XML's Push server.			
4. Client 4	Sets IP address of 4th XML's Push server.			
3. Push Server Port	Port number of XML's Push server [82, 1024-65535]. NOTE 1	82	X	X
3. Admin Password	Sets administrator's password. * Be sure to change the default password.	6633222	X	X

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
4. 802.1X Supplicant	Sets 802.1X authentication related parameter.	-	X	X
1. Supplicant	Sets whether to use supplicant on DT700 Series.	Disable	X	X
1. Disable	Supplicant is disabled on DT700 Series.			
2. Enable	Supplicant is enabled on DT700 Series.			
2. EAP Method (Setting prohibited)	-	-	-	-
3. Account	Sets user ID for authentication.	-	X	X
4. Password	Sets password for authentication.	-	X	X
5. EAP VLAN Mode	Selects whether to add VLAN tag to EAP packet.	Enable	X	X
1. Disable	VLAN tag is not added to EAP packet.			
2. Enable	VLAN tag is added to EAP packet.			
5. VPN Client (Setting prohibited)	-	-	-	-
6. USB	Sets USB related parameter.	Enable	X	X
1. Power	Sets whether to supply power to USB devices.			
1. Disable	Power is not supplied to USB devices.			
2. Enable	Power is supplied to USB devices.			
5. Terminal Information	Displays terminal information.	-	X	X
Hardware Version	Displays hardware version of DT700 Series.	-	X	X
MAC Address	Displays MAC address of DT700 Series.	-	X	X
Firmware Version	Displays firmware version of DT700 Series.	-	X	X
6. Application Settings	Sets application data.	-	X	X
1. XML Settings	Specifies XML browser function.	-	X	X
1. Terminal ID	Login number for accessing XML server.	0000	X	X
2. XML Browser	Sets XML browser related data.	-	X	X
1. Home URL NOTE 10	Sets the address of the contents to access when starting up XML browser.	-	X	X
2. Proxy Settings	Sets proxy server related data.	-	X	X
1. Proxy IP	Sets proxy IP address of XML browser.	0.0.0.0	X	X
2. Proxy Port	Sets proxy port of XML browser [1024-65535].	8080	X	X
3. Directory URL	Sets URL of Corporate Phonebook.	-	X	X
4. Service URL NOTE 11	Sets URL of XML application.	-	X	X
2. IPM Self Port	Sets the port for connecting to IP Phone Manager [1024-65535]. NOTE 1	3530	X	X
3. Popup	Sets the popup window display.	-	X	X
1. Popup Mode	Sets the mode of the popup window display.	App Priority	X	X
1. Enable	Sets the popup window display enabled.			
2. App Priority	Sets to depend on a definition of XML tag for Popup window.			
4. Screen Saver URL	Sets reference URL for Screen Saver activation.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT730G (12/24CG)	DT730G (12/24DG)
5. Emergency Call	Sets the destination number of emergency call which a terminal user can originate even while the Security Mode is in service.	-	X	X
Emergency Dial 1	Up to three destination numbers can be specified.			
Emergency Dial 2				
Emergency Dial 3				
6. Menu Key Mode	Changes to menu key mode.	Enable	X	X
1. Disable	Sets the menu key mode disabled.			
2. Enable	Sets the menu key mode enabled.			

NOTE 1: A unique number must be assigned to each of the following ports. For RTP Self Port, 16 ports starting from an assigned port are used.

- RTP Self Port
- SIP Self Port
- Push Server Port
- IPM Self Port

NOTE 2: $LLDP\ time\ to\ live = LLDP\ packet\ transmission\ interval \times the\ set\ value$ (65535 seconds at most)

NOTE 3: "SIP server" used here refers to the SV9300. The SV9300 acts as SIP server where DT730G Series terminal is registered.

NOTE 4: Follow the IP address setting (CM0B Y=1XX>00) of VOIP port on SV9300.

NOTE 5: Follow the SIP Server port setting (CM0B Y=1XX>32) on SV9300 (example: change 5060 to 5080 when default is set).

NOTE 6: Howler Tone to be sent depends on the system data setting (CM08>153, CM13 Y=04) of SV9300 (default: To send).

NOTE 7: Do not set the Key Kit 8, 9, 10 because these Kits are not supported.

NOTE 8: This service is available to perform various tests such as Key Test, LED, LCD, and Call Test.

NOTE 9: Be careful when you specify "3. Enable (All Client)" because security services are disabled for Push when "3. Enable (All Client)" is set.

NOTE 10: When you register the Home URL, the indication on the Menu screen turns "Application Name" from "Phone". Set the Lock Mode to Disable and Disable the Security key.

To connect a phone and IP Phone Manager, the port number to be used needs to match. Set the port number on the phone side from [0. Config] - [6. Application Setting] - [2. IPM Self Port].

NOTE 11: When you select the parameter of Service URL, a Menu for 1st - 4th Service URL is displayed. URL of up to four XML applications can be registered.

DT750/DT730/DT710 Series Configuration Menu List (FW 5.0.0.0 or later)

This section lists the configuration menu used for DT700 Series configuration setting. DT750 Series, DT730 Series, and DT710 Series columns on the tables below show whether the menu is available or not for each terminal type.

LEGEND	X: Available	N: Not available	-: Not applicable
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NOTE 1: *DT710 Series supports Tool and Config menu only.*

NOTE 2: *The supported parameter information for DT710 DESI-less is same as DT730/DT730G Series.*

NOTE 3: *Service menu is not displayed when a service URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL]. For DT710 Series, Tool menu is displayed instead of Service menu.*

NOTE 4: *Phone menu is displayed when a URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL]. Usually, "Application Name" is displayed when a URL is registered.*

NOTE 5: *The Config menu is not displayed on the menu window. To open the Config menu, press digit key "0". In addition, for DT730 Series, by pressing "Hold", "Transfer", "*", "#" key in this order in an idle state of telephone, you can access to the login screen of Configuration Menu.*

1. History (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series
1. History	Lists/deletes the history of outgoing/incoming calls.	-	X	X

When "1. History" is selected, the "History menu" of SV9300 is displayed.

[History]	
NEC ICHIRO	[300]
3:09 PM	TUE 18 OCT 2016
Outgoing	Incoming

NOTE: *In SV9300, the built-in Call History of a telephone cannot be used. Call History on SV9300 side is used.*

2. Directory (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series
2. Directory	Displays Terminal Phonebook	-	X	X

When “2. Directory” is selected, the “Directory service” screen of SV9300 is displayed.

[Directory]

Name>>

Set Common Personal

NOTE: In SV9300, the built-in Directory of a telephone cannot be used. The Directory service on SV9300 side is used.

3. Service (for DT750 Series and DT730Series) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
3. Service	Starts up XML browser.	-	X	X	-
1. Service	Starts up XML browser.	-	X	X	-

NOTE: Service menu is displayed only when the URL is set by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL].

3. Tool (for DT710 Series) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
3. Tool	Starts up XML browser.	-	-	-	X
1. Service	Starts up XML browser.	-	-	-	X

4. Setting (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
1. User Setting	Sets user interface related.	-	X	X	X NOTE 13
1. Incoming Call	Sets ringing and lamp indication data for incoming call.	-	X	X NOTE 11	X
1. Ring Volume	-	-	-	-	-
2. Offhook Ring	Selects whether to use offhook ringing (ringing for incoming call in busy state).	Enable	-	-	X
1. Disable	Offhook ringing is OFF.				
2. Enable	Offhook ringing is ON.				
3. Headset Ring	Selects whether to use headset ringing.	Disable	X	X	-
1. Disable	Headset ringing is OFF.				
2. Enable	Headset ringing is ON.				
4. Ring Tone	Sets Ringing Tone for incoming call termination.	-	X	X	X
1. External Call NOTE 1	Sets Ringing Tone for external call termination.	Automatic	X	X	X
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.				X
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.				
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.				N
2. Internal Call NOTE 1	Sets Ringing Tone for station call termination.	Automatic	X	X	X
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.				X
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.				
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.				N
3. IM (Setting prohibited)	-	-	-	-	-
5. Illumination	Sets lamp indication pattern for incoming call termination.	-	X	X	-
1. External Call NOTE 2	Sets Illumination for External Call.	-	X	X	-
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.	Automatic	X	X	-
1. Disable	Lamp indication is OFF.				
2. Red					
3. Green					
4. Blue					
5. Yellow					
6. Purple					
7. Light Blue					
8. White					
9. Rotation					
	Specifies the color of lamp indication for incoming call termination.				

4. Setting (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
2. Internal Call NOTE 2	Sets Illumination for Internal Call.	-	X	X	-
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.	Automatic	X	X	-
1. Disable	Lamp indication is OFF.				
2. Red	Specifies the color of lamp indication for incoming call termination.				
3. Green					
4. Blue					
5. Yellow					
6. Purple					
7. Light Blue					
8. White					
9. Rotation					
3. IM (Setting prohibited)	-	-	-	-	-
6. Headset Ring Volume (Setting prohibited)	-	-	-	-	-
2. Talk	Sets call related data.	-	X	X	X
1. RTP Alarm NOTE 3	Sets whether to allow the alarm for detecting packet loss in receiving speech packets.	Automatic	X	X	X
0. Automatic	Applies RTP alarm setting on the SV9300.				
1. Disable	Alarm is OFF.				
2. Enable	Alarm is ON.				
2. DTMF Tone (Setting prohibited)	-	Automatic	-	-	-
0. Automatic	-				
1. Disable (Setting prohibited)	-				
2. Enable (Setting prohibited)	-				
3. Key Touch Tone	Sets touch tone for pressing digit key/touch-sensitive panel.	Automatic NOTE 4	X	X	X
0. Automatic	Sets Automatic to key touch tone.				
1. No Tone	Sets key touch tone disabled.				
2. Tone	Sets key touch tone enabled.				
3. Voice	Enables the key touch tone by voice recognition.				
4. Hold Music	Sets Music-on-Hold.	Default	X	X	X
1. Default	Applies the default music (Minuet).				
2. Download	Applies the downloaded music.				
5. Prefix (Setting prohibited)	-	-	-	-	-
1. Prefix 1 ~ 3. Prefix 3 (Setting prohibited)	-	-	-	-	-

4. Setting (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
3. Display	Sets display related data.	-	X	X NOTE 12	X NOTE 14
1. Time Format (Setting prohibited)	Switches 12-hour/24-hour indication.	Automatic NOTE 5	-	-	-
0. Automatic	-				
1. 12 hour Indication (Setting prohibited)	-				
2. 24 hour Indication (Setting prohibited)	-				
2. Local Volume	Selects whether to display volume gage when adjusting the volume level.	Automatic NOTE 6	X	X	X
0. Automatic	Sets Automatic for volume gage display.				
1. Disable	Volume gage is not displayed.				
2. Enable	Volume gage is displayed.				
3. Screen Saver	Sets Screen Saver related data.	-	X	X	X
1. Screen Saver Mode	Selects whether to use Screen Saver.	Disable	X	X	X
1. Disable	Screen Saver is not used.				
2. Enable	Screen Saver is used.				
2. Wait Time	Sets wait time until Screen Saver activation (1-999 min.).	120	X	X	X
4. Back Light	Switches ON/OFF the backlight on DT730.	Enable NOTE 7	-	X	-
1. Disable	Backlight is OFF.				
2. Enable	Backlight is ON.				
4. Back Light	Turns up/down the backlight on DT750.	Level 3 NOTE 7	X	-	-
1. Level 1 (Bright) ~ 4. Level 4 (Dark)	Select the bright level of backlight.				
5. Font Size NOTE 8	Switches the font size on configuration menu.	Standard (16dot)	X	X	-
1. Standard (16dot)	Selects Standard or Small of font size.				
2. Small (12dot)					

4. Setting (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
6. Language	Selects the language on date and time display.				
0. Automatic	Sets Automatic for language setting on date and time display.				
1. Japanese	Selected language is displayed on date and time display.	Automatic NOTE 9	X	X	X
2. NEC English					
3. NEC Portuguese					
4. NEC Spanish					
5. NEC French					
6. German					
7. Italian					
8. Dutch					
9. Norwegian					
10. Danish					
11. Swedish					
12. Greek					
13. English					
14. Portuguese					
15. Spanish					
16. French					
17. Russian					
18. Turkish					
19. Romania					
20. Polish					
21. Catalan					
7. Advanced	Sets other data for display.	-	X	-	-
1. Wallpaper	Sets the Home screen background.	Default	X	-	-
1. Default	Sets the default screen background.				
2. Download	Sets the download data file.				
2. Touch Panel	Selects whether to use touch-sensitive panel function.	Enable	X	-	-
1. Disable	Touch-sensitive panel is OFF.				
2. Enable	Touch-sensitive panel is ON.				
3. Shortcut Icon (Setting prohibited)	-	-	-	-	-
4. Font Color	Sets the font color for Time Display/Feature Information Display area. * You can check the color while selecting.	Color 2	X	-	-
1~16. Color 1 ~ Color 16	Sets Font Color 1 ~ Font Color 16.				

4. Setting (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
5. Turn Off Display	Sets the data when turning off the backlight of the display automatically.	-			
1. Turn Off Display Mode	Sets whether Turn Off Display Mode is in service.	App Priority	X	-	-
1. Disable	Disables Turn Off Display Mode.				
2. Enable	Enables Turn Off Display Mode.				
3. App Priority	Sets to depend on an application.				
2. Wait Time	Sets the time until turning off the display (1-999 min.).	180			
4. Change Password NOTE 10	Assigns the password of the terminal, used for data clear, security, and login of user mode.	0000	X	X	X
5. Security (Setting prohibited)	-	-	-	-	-
6. Peripherals (Setting prohibited)	-	-	-	-	-
7. Usability	Sets the operation of Help key while pop-up window is displayed.	-	X	X	-
1. Help Key Mode	Setting for Help Key Mode (SubMenu/Popup Window)	SubMenu	X	X	-
1. SubMenu	Sets the Help key which displays sub menu		X	X	-
2. Popup Window	Sets the Help key which displays pop-up window		X	X	-
0. Setting Reset	Initializes all the assigned personal information on the terminal to factory default.	-	X	X	X
2. Download	Sets download related data.	-	X	X	-
1. Download Files	Downloads files of Music-on-Hold, Ringing Tone, and Phonebook.	-	X	X	-
1. Hold Music	Downloads a sound file used for Music-on-Hold.	MOH.wav	X	X	X
2. Ring Tone	Downloads a sound file used for Ringing Tone.	-	X	X	-
1. Download 1	Downloads Melody 1 ~ Melody 3.	Melody1.wav			
2. Download 2		Melody2.wav			
3. Download 3		Melody3.wav			
3. Directory (Setting prohibited)	-	-	-	-	-
4. Wallpaper	Downloads a file used for wallpaper.	WallPaper.bmp	X	-	-
2. Download Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X	X
3. Protocol	Specifies the protocol for file downloading.	FTP	X	X	X
1. FTP	FTP is used.				
2. TFTP	TFTP is used.				
4. FTP Settings	Sets data for accessing FTP server.	-	X	X	X
1. User ID	Sets user ID for login to FTP server.				
2. Password	Sets password for login to FTP server.				
3. Folder	Specifies the directory to save the download file.				

4. Setting (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
3. Data Backup/Restore	Makes backup of terminal data or restores from backup file.	-	X	X	X
1. Data Backup	Saves the terminal setting to backup file.	Personal Data.tgz	X	X	X
2. Data Restore	Restores the terminal data by reading out the saved backup file.	Personal Data.tgz	X	X	X
3. Server Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X	X
4. Protocol	Specifies the protocol for file downloading.	FTP	X	X	X
1. FTP	FTP is used.				
2. TFTP	TFTP is used.				
5. FTP Settings	Sets data for accessing FTP server.	-	X	X	X
1. User ID	Sets user ID for login to FTP server.				
2. Password	Sets password for login to FTP server.				
3. Folder	Specifies the directory to save backup file.				

NOTE 1: To provide the Distinctive Ringing of external call/internal call by SV9300, be sure to use with default (Automatic). When “Automatic” is set, by the system data setting (CM08>390) of SV9300, Ringing Tone can be changed by the system data (CM15 Y=491, CM35 Y=034, 164) or terminal operations (“Feature” + “3”) (default: as per CM15 Y=491, CM35 Y=034, 164).

NOTE 2: When “Automatic” is set, by the system data setting (CM08>144) of SV9300, the illumination color can be changed to Red or Green (default: Red).
Set the “0. Automatic” to use color coding by 7-color LED.

NOTE 3: When “Automatic” is set, RTP Alarm depends on the system data setting (CM08>1004) of SV9300 (default: To send).

NOTE 4: Follows the setting (CM15 Y=291) of SV9300 side when “Automatic” is set. In all other cases, follows the setting of a terminal side.

NOTE 5: Time Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.

NOTE 6: A setting for Enable is applied when Automatic has been set.

NOTE 7: Back Light menu becomes effective when the system data (CM15 Y=485) of SV9300 is set to other than 0 (Always off).

Back Light of ten key cannot be used when terminals and option equipment for power failure are implemented.

NOTE 8: When a font size is changed, press the Cursor key and return to the previous menu, then the font size becomes effective.

NOTE 9: The setting of display language on the Clock Display screen depends on the system data settings (CM12 Y=63, CM04 Y=00) of SV9300.

NOTE 10: Change Password is used for password for User Mode of Web programming and security unlock. Be sure to change the default password.

NOTE 11: For one screen DESI-less type, the order of menus ([4. Terminal Settings] - [1. User Setting] - [1. Incoming Call]) is as follows.

Parameter	Description	Default	Remarks	
1. User Setting	Sets user interface related.	-		
1. Incoming Call	Sets ringing and lamp indication data for incoming call.	-		
1. Ring Volume (Setting prohibited)	-	-		
2. Offhook Ring (Setting prohibited)	-	Enable		
1. Disable (Setting prohibited)	-			
2. Enable	-			
3. Ring Tone	Sets Ringing Tone for incoming call termination.	-		
1. External Call	Sets Ringing Tone for external call termination.	Automatic	To provide the Distinctive Ringing of external call/ internal call by SV9300, be sure to use with default (Automatic). When "Automatic" is set, by the system data setting (CM08>390) of SV9300, Ringing Tone can be changed by the system data (CM15 Y=491, CM35 Y=034, 164) or terminal operations ("Feature" + "3") (default: as per CM15 Y=491, CM35 Y=034, 164).	
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.			
2. Internal Call	Sets Ringing Tone for station call termination.	Automatic		
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.			
4. Illumination	Sets lamp indication pattern for incoming call termination.	-		
1. External Call	Sets Illumination for External Call.	Automatic		When "Automatic" is set, by the system data setting (CM08>144) of SV9300, the illumination color can be changed to Red or Green (default: Red). Set the "0. Automatic" to use color coding by 7-color LED.
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.			
1. Disable	Lamp indication is OFF.			
2. Red	Specifies the color of lamp indication for incoming call termination.			
3. Green				
4. Orange				
5. Rotation				
2. Internal Call	Sets Illumination for Internal Call.	Automatic		
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.			
1. Disable	Lamp indication is OFF.			
2. Red	Specifies the color of lamp indication for incoming call termination.			
3. Green				
4. Orange				
5. Rotation				

NOTE 12: For one screen DESI-less type, the order of menus ([4. Terminal Settings] - [1. User Setting] - [3. Display]) is as follows.

Parameter	Description	Default	Remarks
1. Display	Sets display related data.	-	
1. Time Format (Setting prohibited)	-	Automatic	Time Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.
0. Automatic	-		
1. 12 hour Indication (Setting prohibited)	-		
2. 24 hour Indication (Setting prohibited)	-		
2. Local Volume	Selects whether to display volume gage when adjusting the volume level.	Automatic	A setting for Enable is applied when Automatic has been set.
0. Automatic	Sets Automatic for volume gage display.		
1. Disable	Volume gage is not displayed.		
2. Enable	Volume gage is displayed.		
3. Screen Saver	-	-	
1. Screen Saver Mode	Selects whether to use Screen Saver.	Disable	
1. Disable	Screen Saver is not used.		
2. Enable	Screen Saver is used.		
2. Wait Time	Sets wait time until Screen Saver activation (1-999 min.).	120	
4. Back Light	Switches ON/OFF the backlight on DT730.	Enable	<ul style="list-style-type: none"> Back Light menu becomes effective when the system data (CM15 Y=485) of SV9300 is set to other than 0 (Always off). Back Light of ten key cannot be used when terminals and option equipment for power failure are implemented.
1. Disable	Backlight is OFF.		
2. Enable	Backlight is ON.		
5. Font Size	Switches the font size on configuration menu.	Standard (16dot)	When a font size is changed, the font size will not be become effective until the menu is ended once.
1. Standard (16dot)	Selects Standard or Small of font size.		
2. Small (12dot)			

Parameter	Description	Default	Remarks
6. Language	Selects the language on date and time display.		
0. Automatic	Sets Automatic for language setting on date and time display.		
1. Japanese	Selected language is displayed on date and time display.	Automatic	The setting of display language on the Clock Display screen depends on the system data settings (CM12 Y=63, CM04 Y=00) of SV9300.
2. NEC English			
3. NEC Portuguese			
4. NEC Spanish			
5. NEC French			
6. German			
7. Italian			
8. Dutch			
9. Norwegian			
10. Danish			
11. Swedish			
12. Greek			
13. English			
14. Portuguese			
15. Spanish			
16. French			
17. Russian			
18. Turkish			
19. Romania			
20. Polish			
21. Catalan			
7. Home Display Mode	Switches the screen display between the home screen and the DESI-less screen.		
0. Automatic	Sets automatic switching for screen display.	Automatic	
1. Fixed Display Mode	Sets Fixed Display (DESI-less screen).		

NOTE 13: For DT710 DESI-less terminal, the order of menus ([4. Terminal Settings] - [1. User Setting] - [1. Incoming Call]) is as follows.

Parameter	Description	Default	Remarks
1. User Setting	Sets user interface related.	-	
1. Incoming Call	Sets ringing and lamp indication data for incoming call.	-	
1. Offhook Ring	Selects whether to use offhook ringing (ringing for incoming call in busy state).	Enable	
1. Disable	Offhook ringing is OFF.		
2. Enable	Offhook ringing is ON.		
2. Ring Tone	Sets Ringing Tone for incoming call termination.	-	
1. External Call	Sets Ringing Tone for external call termination.	Automatic	
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.		
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.		
2. Internal Call	Sets Ringing Tone for station call termination.		
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.		
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.		

NOTE 14: For DT710 DESI-less terminal, the order of menus ([4. Terminal Settings] - [1. User Setting] - [3. Display]) is as follows.

Parameter	Description	Default	Remarks
3. Display	Sets display related data.	-	
1. Time Format (Setting prohibited)	Switches 12-hour/24-hour indication.	Automatic	Same as other terminals.
0. Automatic	-		
1. 12 hour Indication (Setting prohibited)	-		
2. 24 hour Indication (Setting prohibited)	-		
2. Local Volume	Selects whether to display volume gage when adjusting the volume level.		
0. Automatic	Selects whether to display volume gage when adjusting the volume level.		
1. Disable	Volume gage is not displayed.		
2. Enable	Volume gage is displayed.		
3. Screen Saver	Sets Screen Saver related data.	-	
1. Screen Saver Mode	Selects whether to use Screen Saver.	Disable	
1. Disable	Screen Saver is not used.		
2. Enable	Screen Saver is used.		
2. Wait Time	Sets wait time until Screen Saver activation (1-999 min.).	120	
4. Language	Selects the language on date and time display.	Automatic	
0. Automatic	Selected language is displayed on date and time display.		
1. Japanese			
2. NEC English			

5. Phone

Depending on the Home URL setting by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL], this menu is displayed as follows.

- When Home URL is not registered, home screen is displayed and the menu name is displayed as “Phone”.
- When Home URL is registered, XML application screen is displayed, and the menu name is displayed as the application name to be described in the XML application.

0. Config (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
1. Network Settings	Sets network related data.	-	X	X	X
1. DHCP Mode	Selects whether to use DHCP protocol.	Enable	X	X	X
1. Disable	DHCP is not used.				
2. Enable	DHCP is used.				
2. IP Address	Sets DT700 Series terminal's IP address.	0.0.0.0	X	X	X
3. Default Gateway	Sets default router's IP address.	0.0.0.0	X	X	X
4. Subnet Mask	Sets subnet mask.	0.0.0.0	X	X	X
5. DNS Address	Sets DNS server's IP address.	0.0.0.0	X	X	X
6. Advanced Settings	Sets network expanded data.	-	X	X	X
1. LAN Port Settings	Sets LAN port.	-	X	X	X
1. Speed&Duplex	Sets communication speed of Ether port.	Autonego	X	X	X
1. Autonego	Autonegotiation				
2. 100M Full	100Mbps in full-duplex mode				
3. 100M Half	100Mbps in half-duplex mode				
4. 10M Full	10Mbps in full-duplex mode				
5. 10M Half	10Mbps in half-duplex mode				
2. VLAN Mode	Selects whether to use VLAN on terminal.	Disable	X	X	X
1. Disable	VLAN is not used.				
2. Enable	VLAN is used.				
3. VLAN ID	Sets VLAN ID of terminal [1-4094].	7	X	X	X
4. VLAN Priority	Sets VLAN Priority of terminal [0-7].	5	X	X	X

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
2. PC Port Settings	Sets PC port.	-	X	X	X
1. Speed&Duplex	Sets communication speed of PC port.	Autonego	X	X	X
1. Autonego	Autonegotiation				
2. 100M Full	100Mbps in full-duplex mode				
3. 100M Half	100Mbps in half-duplex mode				
4. 10M Full	10Mbps in full-duplex mode				
5. 10M Half	10Mbps in half-duplex mode				
2. Port VLAN Mode	Selects whether to use port VLAN of PC port.	Disable	X	X	X
1. Disable	Port VLAN of PC port is not used.				
2. Enable	Port VLAN of PC port is used.				
3. Port VLAN ID	Sets port VLAN ID of PC port [1-4094].	1	X	X	-
		7	-	-	X
4. Port VLAN Priority	Sets port VLAN priority of PC port [0-7].	0	X	X	-
		5	-	-	X
5. Port Available	Selects whether to use PC port.	Automatic	X	X	X
0. Automatic	Sets Automatic for PC port.				
1. Disable	PC port is not used.				
6. EAPOL Forwarding	Select whether to forward EAPOL packet from PC port.	Disable	X	X	X
1. Disable	Not forward EAPOL packet from PC port.				
2. Enable	Forwards EAPOL packet from PC port.				
7. PC Port Security	Select whether to clear authentication status of the terminal which is connected to PC port for authentication switch when the link of PC port has been down.	Enable	X	X	X
1. Disable	Not clear the authentication status.				
2. Enable	Clear the authentication status.				
3. CDP Mode	Selects whether to notify HUB of power consuming information.	Disable	X	X	X
1. Disable	CDP is not used (Not send CDP to HUB).				
2. Enable	CDP is used (Sends CDP to HUB).				
4. Type of Service	Sets ToS value.	-	X	X	X
1. RTP	Sets ToS value for RTP in Hex.	A0			
2. SIP	Sets ToS value for SIP in Hex.	C0			
5. Self Port Settings	Sets each port for DT700 Series.	-	X	X	-
1. RTP Self Port	Sets RTP port for DT700 Series [1024-65520 (except 59986-60000)]. NOTE 1 * Odd port numbers cannot be set.	3462	X	X	X
2. SIP Self Port	Sets SIP port for DT700 Series [1024-65534]. NOTE 1	5060	X	X	X
6. LLDP Settings	Sets information related to LLDP.	-	-	-	-
1. LLDP Mode	Sets whether to transmit/receive LLDP packets.	Enable	X	X	X
1. Disable	DT700 Series does not transmit/receive LLDP packets.				
2. Enable	DT700 Series transmits/receives LLDP packets.				

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
2. Auto Setting Mode	Sets whether to perform data setting automatically when LLDP packets are received.				
1. Disable	DT700 Series does not perform data setting automatically when LLDP packets are received.	Enable	X	X	X
2. Enable	DT700 Series performs data setting automatically when LLDP packets are received.				
3. RX Waiting Time	For Auto Setting, sets the waiting time until DT700 Series receives LLDP packets (1-60 seconds).	15	X	X	X
4. Transmit Interval	Sets intervals to transmit LLDP packets (5-32768 seconds).	30	X	X	X
5. Hold Multiplier NOTE 2	Sets a multiplier determining the time to live for the LLDP packets to be transmitted (2-10 multiples).	4	X	X	X
6. Fast Start Count	Sets the number of times to transmit LLDP packets at one-second intervals after DT700 Series starts up (1-10 times).	3	X	X	X
7. Transparent Mode	Sets whether Transparent Mode of LLDP packets is enabled between the PC port and the LAN port.				
1. Disable	Transparent Mode of LLDP packets is disabled.	Enable	X	X	X
2. Enable	Transparent Mode of LLDP packets is enabled.				
8. Asset ID	Sets the Asset ID for LLDP packets to be transmitted. * “-” is displayed in default settings.	-	X	X	X

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
7. Spare IP Settings (Setting prohibited)	-	-	-	-	-
1. Spare/Backup IP Mode (Setting prohibited)	-	Disable	-	-	-
1. Disable	-				
2. Spare IP (Setting prohibited)	-				
3. Backup IP (Setting prohibited)	-				
2. Network Settings (Setting prohibited)	-	-	-	-	-
1. IP Address (Setting prohibited)	-	0.0.0.0			
2. Default Gateway (Setting prohibited)	-	0.0.0.0			
3. Subnet Mask (Setting prohibited)	-	0.0.0.0			
4. DNS Address (Setting prohibited)	-	0.0.0.0			
3. SIP Settings (Setting prohibited)	-	-	-	-	-
1. 2nd Server Address (Setting prohibited)	-	0.0.0.0	-	-	-
2. 3rd Server Address (Setting prohibited)	-	0.0.0.0	-	-	-
3. 4th Server Address (Setting prohibited)	-	0.0.0.0	-	-	-
4. 2nd Server Port (Setting prohibited)	-	0	-	-	-
5. 3rd Server Port (Setting prohibited)	-	0	-	-	-
6. 4th Server Port (Setting prohibited)	-	0	-	-	-
2. SIP Settings	Sets SIP related data.	-	X	X	X
1. SIP User (Setting prohibited)	-	-	-	-	-
1. User ID (Setting prohibited)	Sets user ID for digest authentication.	-	-	-	- NOTE 12
2. Password (Setting prohibited)	Sets password for digest authentication.	-	-	-	- NOTE 12
3. Extension Number (Setting prohibited)	-	-	-	-	-
4. Backup Logon (Setting prohibited)	-	Disable	-	-	-

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
2. Server Address & URI	Sets IP address or URI of SIP server. NOTE 3	-	X	X	X
1. 1st Server Address	Sets IP address or fully qualified domain name (FQDN) of 1st ~ 4th SIP server. NOTE 4	0.0.0.0	X	X	X
2. 2nd Server Address					
3. 3rd Server Address					
4. 4th Server Address					
5. Registrar Destination (Setting prohibited)	-	0.0.0.0	-	-	-
1. 1st Register Address (Setting prohibited)	-		-	-	-
2. 2nd Register Address (Setting prohibited)					
3. 3rd Register Address (Setting prohibited)					
4. 4th Register Address (Setting prohibited)					
3. Access Mode	Sets access mode for SIP server.	Normal	X	X	X
1. Normal	Normal access				
2. Remote	Remote access				
4. SIP Server Port	Sets SIP server port [1024-65535].	5060	X	X	X
1. 1st Server Port	Specifies the destination port number of sending packet to 1st ~ 4th SIP server. NOTE 5				
2. 2nd Server Port					
3. 3rd Server Port					
4. 4th Server Port					
5. Registrar Destination (Setting prohibited)	-		-	-	-
1. 1st Register Port (Setting prohibited)	-				
2. 2nd Register Port (Setting prohibited)					
3. 3rd Register Port (Setting prohibited)					
4. 4th Register Port (Setting prohibited)					

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
5. Area ID	Sets the area (country) using DT700 Series.				
0. Automatic	The SV9300 side setting is applied when Automatic is selected.				
1. Japan	Selects the area (country) NOTE: Area IDs allowed on the SV9300 can be selected.	Automatic	X	X	X
2. America (Canada and North America)					
3. Australia					
4. Others					
5. Hong Kong					
6. Malaysia					
7. Singapore					
8. Mexico					
9. Taiwan					
10. New Zealand					
11. Korea					
12. Brazil					
13. China					
14. Thailand					
15. Thailand (EGAT)					
16. Sri Lanka					
17. Germany					
18. Italy					
19. Netherlands					
20. Catalonia					
21. Denmark					
22. Sweden					
23. Poland					
24. Czech					
25. Norway					
26. France					
27. Portugal					
28. Spain					
29. Austria					
30. Belgium					
31. UK					
32. Greece					
33. Switzerland					
34. South Africa					
35. Brazil (NECU)					

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
6. Extra Function (Setting prohibited)	-	-	-	-	-
1. Retransmit Interval (Setting prohibited)	-	Standard	-	-	-
1. Standard	-				
2. 6 Seconds (Setting prohibited)	-				
2. Keeping Call (Setting prohibited)	-	Enable	-	-	-
1. Disable (Setting prohibited)	-				
2. Enable	-				
3. Howler Tone (Setting prohibited)	Selects whether to send howler tone. NOTE 7	Disable	-	-	-
1. Disable	Howler tone sending is OFF.				
2. Enable (Setting prohibited)	Howler tone sending is ON.				
7. Encryption	Sets encryption data.	-	X	X	X
1. Authentication Mode	Selects whether to use authentication.	Disable	X	X	X
1. Disable	Authentication is not used.				
2. Enable	Authentication is used.				
2. One Time Password	Sets One Time Password.	-	X	X	X
3. Authentication Status	Displays authentication status.	-	X	X	X
8. NAT Traversal (Setting prohibited)	-	-	-	-	-
1. NAT Traverse Mode	-	Disable	-	-	-
1. Disable					
2. Dynamic (Setting prohibited)					
3. Static (Setting prohibited)					
2. Network Area Name (Setting prohibited)	-	-	-	-	-
3. WAN Settings (Setting prohibited)	-	-	-	-	-
1. WAN Mate IP Address (Setting prohibited)					
2. WAN SIP Mate Port (Setting prohibited)					
3. WAN Self IP Address (Setting prohibited)					

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
3. Maintenance Settings	Sets maintenance related data.	-	X	X	X
1. Download Menu	Downloads specified file to DT700 Series.	-	X	X	X
1. Download Files	Specifies download file name for each file type.	-	X	X	X
1. Config	Downloads configuration file.	config.tgz	X	X	X
2. Certificate (Setting prohibited)	-	-	-	-	-
3. Boot & Program	Downloads boot and program.	DT750: itlisips.tgz DT730: itlisipv.tgz DT710: itlisper.tgz	X	X	X
4. Flash Memory Image	Downloads flash memory image.	imgsip.tgz	X	X	X
5. Presence Icon (Setting prohibited)	-	-	-	-	-
6. Voice File	Downloads voice file.	voice.tgz	X	X	N
7. Language Pack	Downloads translation data file.	LangPack.tgz	X	X	-
2. Download Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X	X
3. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X	X	X
1. FTP	FTP is used.				
2. TFTP	TFTP is used.				
4. FTP Settings	Sets data for accessing FTP server.	-	X	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X	X
2. Password	Sets password for login to FTP server.	-	X	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X	X
2. Hard Reset	Resets DT700 Series without data saving.	-	X	X	X
3. Data Clear	Initializes DT700 Series selecting the initialization mode.	-	X	X	X
1. Factory Value	Initializes DT700 Series to factory default.	-	X	X	X
2. Terminal Data	Clears terminal data on DT700 Series.	-	X	X	X
3. Personal Data	Clears personal data on DT700 Series.	-	X	X	N
4. Adjust	Adjusts LCD, audio, and button kit.	-	X	X	X
1. Default Audio	Sets audio data.	-	X	X	X
1. Sidetone Volume	Sets Side Tone volume [0-7].	4	X	X	X
2. Send Gain	Sets sending gain of handset [0-31].	24	X	X	X
3. Receive Bias	Sets receiving bias of handset [0-7].	0	X	X	X
4. Receive Gain	Sets receiving gain of handset [0-31].	22	X	X	X
5. Noise Suppression	Sets whether to enable the Noise Suppressor. (Set "1. Disable" in normal operation.)	Disable	X	X	X
1. Disable	Sets to disable the Noise Suppressor.				
2. Enable	Sets to enable the Noise Suppressor.				

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
2. Key Kit Type	Sets additional button kit data.	-	X	X	X
1-16. Kit 1-17 (5-7, 13-15 setting prohibited) (Kit 1-17) NOTE 7	Key 1 ~ 4, 8 ~ 12, 17 on button kit are used (5-7, 13-15 setting prohibited).	DT750: Kit 1 DT730: Kit 1 DT710: Kit 1, 9	X	X	X
1. Line Key Type	(Do not change the factory default value of this parameter.)	-	X	X	X
1. Line 12	Sets to 12-line of DT750/DT730/DT710.	-	X	X	X
2~7. Line 24, 32, 8, 6, 2, 0	Not used.	-	-	-	-
3. Calibration	Adjusts touch-sensitive panel location.	-	X	-	-
4. LCD Contrast	Adjusts LCD Contrast.	Wide	-	X	-
0. Area Contrast					
1. Wide					
2. Narrow					
5. Auto config	Selects whether to use auto-config function.	Enable	X	X	X
1. Disable	Auto config is not used.				
2. Enable	Auto config is used.				
6. Data Maintenance	Sets data maintenance related.	-	X	X	X
1. Backup	Executes data backup.	-	X	X	X
1. All Data	Executes backup for all the configuration setting and data.	AllData.tgz	X	X	X
2. Terminal Data	Executes backup for terminal setting and data.	TerminalData.tgz	X	X	X
3. Personal Data	Executes backup for personal setting and data.	PersonalData.tgz	X	X	X
2. Restore	Executes restoration.	-	X	X	X
1. All Data	Restores all the configuration setting and data.	AllData.tgz	X	X	X
2. Terminal Data	Restores terminal setting and data.	TerminalData.tgz	X	X	X
3. Personal Data	Restores personal setting and data.	PersonalData.tgz	X	X	X
3. Server Address	Sets IP address of the server that executes data backup/restoration.	0.0.0.0	X	X	X
4. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X	X	X
1. FTP	FTP is used.				
2. TFTP	TFTP is used.				
5. FTP Settings	Sets data for accessing FTP server.	-	X	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X	X
2. Password	Sets password for login to FTP server.	-	X	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X	X

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
7. Simple Test NOTE 8	Changes the terminal to test mode.	-	X	X	X
1. Call Test	Call Test	-	X	X	X
2. Call Test Address	Sets an IP address of the opposite terminal to perform the call test.	10.136.20.188	X	X	X
3. Key Test 1 (Random)	Tests to press the random button.	-	X	X	X
4. Key Test 2 (Sequential)	Tests to press the sequential button.	-	X	X	X
5. Loop Back Test	Tests the voice loop back.	-	X	X	X
1. Handset - Hand Speaker	Tests to speak and receive on the hand speaker.	-	X	X	X
2. Handset - Speaker	Tests to speak on the handset and receive on the speaker.	-	X	X	X
3. Headset - Head Speaker	Tests to speak and receive on the head speaker.	-	X	X	X
4. Headset - Speaker	Tests to speak on the headset and receive on the speaker.	-	X	X	X
5. MIC - Hand Speaker	Tests to speak on the MIC and receive on the hand speaker.	-	X	X	X
6. MIC - Head Speaker	Tests to speak on the MIC and receive on the head speaker.	-	X	X	X
7. MIC - Speaker	Tests to speak on the MIC and receive on the speaker.	-	X	X	X
6. Speaker Test	Tests the speaker.	-	X	X	X
1. Japan	A DT for Japan is heard in maximum volume.	-	X	X	X
2. America	A DT for North America is heard in maximum volume.	-	X	X	X
7. Receiver Test	Receiver Test for a handset	-	X	X	X
4. Security	Sets security related data.	-	X	X	X
1. Web Programming	Selects whether to use Web Programming.				
1. Disable	Web Programming is disabled.	Enable	X	X	N
2. Enable	Web Programming is enabled.				
2. Push Server	Sets XML's Push server related data.	-	X	X	N
1. Push Server Access	Sets whether to access XML's Push server.				
1. Disable	Access is disabled.				
2. Enable	Access is enabled.	Enable	X	X	X
3. Enable (All Client) NOTE 9	Access is enabled. Select this item to set more than one XML's Push server.				
2. Client IP Address	IP address of XML's Push server.				
1. Client 1	Sets IP address of 1st XML's Push server.				
2. Client 2	Sets IP address of 2nd XML's Push server.	0.0.0.0	X	X	X
3. Client 3	Sets IP address of 3rd XML's Push server.				
4. Client 4	Sets IP address of 4th XML's Push server.				
3. Push Server Port	Port number of XML's Push server [82, 1024-65535]. NOTE 1	82	X	X	N
3. Admin Password	Sets administrator's password. * Be sure to change the default password.	6633222	X	X	X

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
4. 802.1X Supplicant	Sets 802.1X authentication related parameter.	-	X	X	X
1. Supplicant	Sets whether to use supplicant on DT700 Series.	Disable	X	X	X
1. Disable	Supplicant is disabled on DT700 Series.				
2. Enable	Supplicant is enabled on DT700 Series.				
2. EAP Method (Setting prohibited)	-	-	-	-	-
3. Account	Sets user ID for authentication.	-	X	X	X
4. Password	Sets password for authentication.	-	X	X	X
5. EAP VLAN Mode	Selects whether to add VLAN tag to EAP packet.	Enable	X	X	X
1. Disable	VLAN tag is not added to EAP packet.				
2. Enable	VLAN tag is added to EAP packet.				
5. VPN Client (Setting prohibited)	-	-	-	-	-
5. Terminal Information	Displays terminal information.	-	X	X	X
Hardware Version	Displays hardware version of DT700 Series.	-	X	X	X
MAC Address	Displays MAC address of DT700 Series.	-	X	X	X
Firmware Version	Displays firmware version of DT700 Series.	-	X	X	X
6. Application Settings	Sets application data.	-	X	X	X
1. XML Settings	Specifies XML browser function.	-	X	X	X
1. Terminal ID	Login number for accessing XML server.	0000	X	X	X
2. XML Browser	Sets XML browser related data.	-	X	X	X
1. Home URL NOTE 10	Sets the address of the contents to access when starting up XML browser.	-	X	X	X
2. Proxy Settings	Sets proxy server related data.	-	X	X	X
1. Proxy IP	Sets proxy IP address of XML browser.	0.0.0.0	X	X	X
2. Proxy Port	Sets proxy port of XML browser [1024-65535].	8080	X	X	X
3. Directory URL	Sets URL of Corporate Phonebook.	-	X	X	N
4. Service URL	Sets URL of XML application.	-	X	X	N
5. Conf Key URL	Sets URL when the Conf Key is used. *This data can be set only when Key Kit 17 is set.	-	X	N	N
2. IPM Self Port	Sets the port for connecting to IP Phone Manager [1024-65535 (except 17185, 60000)]. NOTE 1	3530	X	X	X

0. Config (Continued) (FW 5.0.0.0 or later)

Parameter	Description	Default	DT750 Series	DT730 Series	DT710 Series
3. Popup	Sets the popup window display.	-	X	X	X
1. Popup Mode	Sets the mode of the popup window display.	App Priority	X	X	X
1. Enable	Sets the popup window display enabled.				
2. App Priority	Sets to depend on a definition of XML tag for Popup window.				
2. Close Button	Sets the Close Button display for closing the popup window.	Enable	X	N	N
1. Disable	Sets the Close Button display disabled.				
2. Enable	Sets the Close Button display enabled.				
3. Window Size Button	Sets the Window Size Button display for changing the size of the popup window.	Enable	X	N	N
1. Disable	Sets the Window Size Button display disabled.				
2. Enable	Sets the Window Size Button display enabled.				
4. Screen Saver URL	Sets reference URL for Screen Saver activation.	-	X	X	X
5. Emergency Call NOTE 11	Sets the destination number of emergency call which a terminal user can originate even while the Security Mode is in service.	-	X	X	N
Emergency Dial 1	Up to three destination numbers can be specified.				
Emergency Dial 2					
Emergency Dial 3					
6. Menu Key Mode	Changes to menu key mode.	Enable	X	X	N
1. Disable	Sets the menu key mode disabled.				
2. Enable	Sets the menu key mode enabled.				
7. DESI-less Mode	Sets the DESI-less screen view.	DESI-less Mode 2	X	N	N
1. DESI-less Mode 1	Switches the LCD to DESI-less mode off-hook.				
2. DESI-less Mode 2	Does not change the LCD to DESI-less mode.				

NOTE 1: A unique number must be assigned to each of the following ports. For RTP Self Port, 16 ports starting from an assigned port are used.

- RTP Self Port
- SIP Self Port
- Push Server Port
- IPM Self Port

NOTE 2: $LLDP \text{ time to live} = LLDP \text{ packet transmission interval} \times \text{the set value}$ (65535 seconds at most)

NOTE 3: "SIP server" used here refers to the SV9300. The SV9300 acts as SIP server where DT700 Series terminal is registered.

NOTE 4: Follow the IP address setting (CM0B Y=1XX>00) of VOIP port on SV9300.

NOTE 5: Follow the SIP Server port setting (CM0B Y=1XX>32) on SV9300 (example: change 5060 to 5080 when default is set).

NOTE 6: Howler Tone to be sent depends on the system data setting (CM08>153, CM13 Y=04) of SV9300 (default: To send).

NOTE 7: The Key Kit 16 is not displayed.

The Key Kit 17 is for hotels.

The Key Kit 17 is not displayed in the value type.

- NOTE 8:** *This service is available to perform various tests such as Key Test, LED, LCD, and Call Test.*
- NOTE 9:** *Be careful when you specify “3. Enable (All Client)” because security services are disabled for Push when “3. Enable (All Client)” is set.*
- NOTE 10:** *When you register the Home URL, the indication on the Menu screen turns “Application Name” from “Phone”. Set the Lock Mode to Disable and Disable the Security key.
To connect a phone and IP Phone Manager, the port number to be used needs to match. Set the port number on the phone side from [0. Config] - [6. Application Setting] - [2. IPM Self Port].*
- NOTE 11:** *For DT710 DESI-less, this menu is displayed as “5. Menu Key Mode”.*
- NOTE 12:** *To set the login method of DT710, be sure to set the User ID and the password. Also, set the User ID and the password for each login method as follows.*
- (a) For Protected Login Mode (CM08>513: 0/1, CM15 Y=480: 0) or Fixed Connection Mode*
 - User ID: IP Station No. assigned by CM10 Y=01*
 - Password: Password assigned by CM2B Y=00*
 - (b) For MAC address authentication (Automatic Login Mode) (CM08>513: 1, CM15 Y=480: 7)*
 - User ID: IP Station No. assigned by CM10 Y=01*
 - Password: Password assigned by CM2B Y=10*

DT770G Configuration Menu List

This section lists the configuration menu used for DT770G configuration setting.

On the login window for Web Programming, enter the User ID (default value: ADMIN) and the password (default value: 6633222) to login.

Config

Parameter	Description	Default
Network Settings	Sets network related data.	-
DHCP Mode	Selects whether to use DHCP protocol.	Enable
1. Disable	DHCP is not used.	
2. Enable	DHCP is used.	
IP Address	Sets DT770G terminal's IP address.	0.0.0.0
Default Gateway	Sets default router's IP address.	0.0.0.0
Subnet Mask	Sets subnet mask.	0.0.0.0
DNS Address	Sets DNS server's IP address.	0.0.0.0
Advanced Settings	Sets network expanded data.	-
LAN Port Settings	Sets LAN port.	-
Speed&Duplex	Sets communication speed of Ether port.	Autonego
1. Autonego	Autonegotiation	
2. 100M Full	100Mbps in full-duplex mode	
3. 100M Half	100Mbps in half-duplex mode	
4. 10M Full	10Mbps in full-duplex mode	
5. 10M Half	10Mbps in half-duplex mode	
VLAN Mode	Selects whether to use VLAN on terminal.	Disable
1. Disable	VLAN is not used.	
2. Enable	VLAN is used.	
VLAN ID	Sets VLAN ID of terminal [1-4094].	7
VLAN Priority	Sets VLAN Priority of terminal [0-7].	5
PC Port Settings	Sets PC port.	-
Speed&Duplex	Sets communication speed of PC port.	Autonego
1. Autonego	Autonegotiation	
2. 100M Full	100Mbps in full-duplex mode	
3. 100M Half	100Mbps in half-duplex mode	
4. 10M Full	10Mbps in full-duplex mode	
5. 10M Half	10Mbps in half-duplex mode	
Port VLAN Mode	Selects whether to use port VLAN of PC port.	Disable
1. Disable	Port VLAN of PC port is not used.	
2. Enable	Port VLAN of PC port is used.	
Port VLAN ID	Sets port VLAN ID of PC port [1-4094].	1
Port VLAN Priority	Sets port VLAN priority of PC port [0-7].	0
Port Available	Selects whether to use PC port.	Automatic
0. Automatic	Sets Automatic for PC port.	
1. Disable	PC port is not used.	
EAPOL Forwarding	Select whether to forward EAPOL packet from PC port.	Disable
1. Disable	Not forward EAPOL packet from PC port.	
2. Enable	Forwards EAPOL packet from PC port.	

Config (Continued)

Parameter	Description	Default
PC Port Security	Select whether to clear authentication status of the terminal which is connected to PC port for authentication switch when the link of PC port has been down.	Enable
1. Disable	Not clear the authentication status.	
2. Enable	Clear the authentication status.	
Type Of Service	Sets ToS value.	-
1. RTP	Sets ToS value for RTP in Hex.	A0
2. SIP	Sets ToS value for SIP in Hex.	C0
Self Port Settings	Sets each port for DT770G.	-
1.RTP Self Port	Sets RTP port for DT770G [1024-65520 (except 59986-60000)]. NOTE 1 * Odd port numbers cannot be set.	3462
2.SIP Self Port	Sets SIP port for DT770G [1024-65534 (except 17185-60000)]. NOTE 1	5060
LLDP Settings	Sets information related to LLDP.	-
LLDP Mode	Sets whether to transmit/receive LLDP packets.	Enable
1. Disable	DT770G does not transmit/receive LLDP packets.	
2. Enable	DT770G transmits/receives LLDP packets.	
Auto Setting Mode	Sets whether to perform data setting automatically when LLDP packets are received.	Enable
1. Disable	DT770G does not perform data setting automatically when LLDP packets are received.	
2. Enable	DT770G performs data setting automatically when LLDP packets are received.	
RX Waiting Time	For Auto Setting, sets the waiting time until DT770G receives LLDP packets (1-60 seconds).	15
Transmit Interval	Sets intervals to transmit LLDP packets (5-32768 seconds).	30
Hold Multiplier NOTE 2	Sets a multiplier determining the time to live for the LLDP packets to be transmitted (2-10 multiples).	4
Fast Start Count	Sets the number of times to transmit LLDP packets at one-second intervals after DT770G starts up (1-10 times).	3
Transparent Mode	Sets whether Transparent Mode of LLDP packets is enabled between the PC port and the LAN port.	Enable
1. Disable	Transparent Mode of LLDP packets is disabled.	
2. Enable	Transparent Mode of LLDP packets is enabled.	
Asset ID	Sets the Asset ID for LLDP packets to be transmitted. * “-” is displayed in default settings.	-

Config (Continued)

Parameter	Description	Default
Spare IP Settings (Setting prohibited)	-	-
Spare/Backup IP Mode (Setting prohibited)	-	Disable
1. Disable	-	
2. Spare IP (Setting prohibited)	-	
3. Backup IP (Setting prohibited)	-	
Network Settings (Setting prohibited)	-	-
IP Address (Setting prohibited)	-	0.0.0.0
Default Gateway (Setting prohibited)	-	0.0.0.0
Subnet Mask (Setting prohibited)	-	0.0.0.0
DNS Address (Setting prohibited)	-	0.0.0.0
SIP Settings (Setting prohibited)	-	-
2nd Server Address (Setting prohibited)	-	0.0.0.0
3rd Server Address (Setting prohibited)	-	0.0.0.0
4th Server Address (Setting prohibited)	-	0.0.0.0
2nd Server Port (Setting prohibited)	-	0
3rd Server Port (Setting prohibited)	-	0
4th Server Port (Setting prohibited)	-	0
SIP Settings	Sets SIP related data.	-
SIP User (Setting prohibited)	Sets user data.	-
User ID (Setting prohibited)	Sets user ID for digest authentication. NOTE 3	-
Password (Setting prohibited)	Sets password for digest authentication. NOTE 3	-
Extension Number (Setting prohibited)	-	-
Backup Logon (Setting prohibited)	-	Disable

Config (Continued)

Parameter	Description	Default
Server Address	Sets IP address or URI of SIP server. NOTE 4	-
1. 1st Server Address	Sets IP address or fully qualified domain name (FQDN) of 1st ~ 4th SIP server. NOTE 5	0.0.0.0
2. 2nd Server Address		
3. 3rd Server Address		
4. 4th Server Address		
5. Registrar Destination (Setting prohibited)	-	0.0.0.0
1. 1st Register Address (Setting prohibited)	-	
2. 2nd Register Address (Setting prohibited)		
3. 3rd Register Address (Setting prohibited)		
4. 4th Register Address (Setting prohibited)		
Access Mode	Sets access mode for SIP server.	Normal
1. Normal	Normal access	
2. Remote	Remote access	
SIP Server Port	Sets SIP server port [1024-65535].	5060
1. 1st Server Port	Specifies the destination port number of sending packet to 1st ~ 4th SIP server. NOTE 6	
2. 2nd Server Port		
3. 3rd Server Port		
4. 4th Server Port		
5. Registrar Destination (Setting prohibited)	-	
1. 1st Register Port (Setting prohibited)	-	
2. 2nd Register Port (Setting prohibited)		
3. 3rd Register Port (Setting prohibited)		
4. 4th Register Port (Setting prohibited)		

Config (Continued)

Parameter	Description	Default
Area ID	Sets the area (country) using DT770G.	
0. Automatic	The SV9300 side setting is applied when Automatic is selected.	
1. Japan		
2. America (Canada and North America)		
3. Australia		
4. Others		
5. Hong Kong		
6. Malaysia		
7. Singapore		
8. Mexico		
9. Taiwan		
10. New Zealand		
11. Korea		
12. Brazil		
13. China		
14. Thailand		
15. Thailand (EGAT)		
16. Sri Lanka		
17. Germany	Selects the area (country)	Automatic
18. Italy	NOTE: Area IDs allowed on the SV9300 can be selected.	
19. Netherlands		
20. Catalonia		
21. Denmark		
22. Sweden		
23. Poland		
24. Czech		
25. Norway		
26. France		
27. Portugal		
28. Spain		
29. Austria		
30. Belgium		
31. UK		
32. Greece		
33. Switzerland		
34. South Africa		
35. Brazil (NECU)		

Config (Continued)

Parameter	Description	Default
Extra Function (Setting prohibited)	-	-
Retransmit Interval (Setting prohibited)	-	Standard
1. Standard	-	
2. 6 Seconds (Setting prohibited)	-	
Keeping Call (Setting prohibited)	-	Enable
1. Disable (Setting prohibited)	-	
2. Enable	-	
Howler Tone (Setting prohibited)	Selects whether to send howler tone. NOTE 7	Disable
1. Disable	Howler tone sending is OFF.	
2. Enable (Setting prohibited)	Howler tone sending is ON.	
Encryption	Sets encryption data.	-
Authentication Mode	Selects whether to use authentication.	Disable
1. Disable	Authentication is not used.	
2. Enable	Authentication is used.	
One Time Password	Sets One Time Password.	-
Authentication Status	Displays authentication status.	-
NAT Traversal (Setting prohibited)	-	-
NAT Traverse Mode	-	Disable
1. Disable		
2. Dynamic (Setting prohibited)		
3. Static (Setting prohibited)		
Network Area Name (Setting prohibited)	-	-
WAN Settings (Setting prohibited)	-	-
1. WAN Mate IP Address (Setting prohibited)		
2. WAN SIP Mate Port (Setting prohibited)		
3. WAN Self IP Address (Setting prohibited)		

Config (Continued)

Parameter	Description	Default
Maintenance	Sets maintenance related data.	-
Download Menu	Downloads specified file to DT770G.	-
Download Files	Specifies download file name for each file type.	-
Boot & Program	Downloads boot and program.	itlisipcr.tgz
Config	Downloads configuration file.	config.tgz
Download Address	Sets IP address of FTP/TFTP server.	0.0.0.0
Protocol	Selects FTP or TFTP for file downloading protocol.	FTP
1. FTP	FTP is used.	
2. TFTP	TFTP is used.	
FTP Settings	Sets data for accessing FTP server.	-
1. User ID	Sets user ID for login to FTP server.	-
2. Password	Sets password for login to FTP server.	-
3. Folder	Specifies the directory to save the download file.	-
Hard Reset	Resets DT770G without data saving.	-
Data Clear	Initializes DT770G selecting the initialization mode.	-
1. Factory Value	Initializes DT770G to factory default.	-
2. Terminal Data	Clears terminal data on DT770G.	-
3. Personal Data	Clears personal data on DT770G.	-
Adjust	Adjusts audio and button kit.	-
Default Audio	Sets audio data.	-
Sidetone Volume NOTE 8	Sets Side Tone volume [0-7].	4
Send Gain NOTE 8	Sets sending gain of handset [0-31].	24
Receive Bias NOTE 8	Sets receiving bias of handset [0-7].	0
Receive Gain NOTE 8	Sets receiving gain of handset [0-31].	22
Talk Characteristics	Sets talk characteristics of handset.	-
Noise Suppression	Sets whether to enable the Noise Suppressor. (Set "1. Disable" in normal operation.)	Disable
Disable	Disables the Noise Suppressor.	
Enable	Enables the Noise Suppressor.	
Handsfree Mode	Sets a send gain mode of handsfree call.	Retro Mode
Retro Mode	Sets to Retro mode.	
New Mode	Sets to New mode.	
Key Kit Type	Sets additional button kit data.	-
1. Kit 1 ~ 2. Kit 2, 8. Kit 8 ~ 10. Kit 10, 17. Kit 17	Key 1 ~ 2, 8 ~ 10, 17 on button kit are used.	Kit 9
1. Line Key Type	(Do not change the factory default value of this parameter.)	Line 2
1~ 4. Line 12, 24, 32, 8	Not used.	
5. Line 6	Sets to 2-line of DT770G.	
7. Line 0	Not used.	
Auto config	Selects whether to use auto-config function.	Enable
1. Disable	Auto config is not used.	
2. Enable	Auto config is used.	

Config (Continued)

Parameter	Description	Default
Data Maintenance	Sets data maintenance related.	-
Backup	Executes data backup.	-
1. All Data	Executes backup for all the configuration setting and data.	AllData.tgz
2. Terminal Data	Executes backup for terminal setting and data.	Terminal Data.tgz
3. Personal Data	Executes backup for personal setting and data.	Personal Data.tgz
Restore	Executes restoration.	-
1. All Data	Restores all the configuration setting and data.	AllData.tgz
2. Terminal Data	Restores terminal setting and data.	Terminal Data.tgz
3. Personal Data	Restores personal setting and data.	Personal Data.tgz
Server Address	Sets IP address of the server that executes data backup/restoration.	0.0.0.0
Protocol	Selects FTP or TFTP for file downloading protocol.	FTP
1. FTP	FTP is used.	
2. TFTP	TFTP is used.	
FTP Settings	Sets data for accessing FTP server.	-
1. User ID	Sets user ID for login to FTP server.	-
2. Password	Sets password for login to FTP server.	-
3. Folder	Specifies the directory to save the download file.	-
4. Security	Sets security related data.	-
Admin Password	Sets administrator's password. * Be sure to change the default password.	6633222
802.1X Supplicant	Sets 802.1X authentication related parameter.	-
Supplicant	Sets whether to use supplicant on DT770G.	Disable
1. Disable	Supplicant is disabled on DT770G.	
2. Enable	Supplicant is enabled on DT770G.	
Account	Sets user ID for EAP-MD5.	-
Password	Sets password for EAP-MD5.	-
EAP VLAN Mode	Selects whether to add VLAN tag to EAP packet.	Enable
1. Disable	VLAN tag is not added to EAP packet.	
2. Enable	VLAN tag is added to EAP packet.	
VPN Client (Setting prohibited)	-	-
USB	Sets USB related parameter.	Enable
Power	Sets whether to supply power to USB devices.	
1. Disable	Power is not supplied to USB devices.	
2. Enable	Power is supplied to USB devices.	
6. Application Settings	Sets application data.	-
IPM Self Port	Sets the port for connecting to IP Phone Manager [1024-65535 (except 17185, 60000)]. NOTE 1	3530

-
-
- NOTE 1:** *A unique number must be assigned to each of the following ports. For RTP Self Port, 16 ports starting from an assigned port are used.*
- RTP Self Port
 - SIP Self Port
 - Push Server Port
 - IPM Self Port
- NOTE 2:** *LLDP time to live = LLDP packet transmission interval × the set value (65535 seconds at most).*
- NOTE 3:** *To set the login method of DT710, be sure to set the User ID and the password. Also, set the User ID and the password for each login method as follows.*
- (a) *For Protected Login Mode (CM08>513: 0/1, CM15 Y=480: 0) or Fixed Connection Mode*
 - User ID: IP Station No. assigned by CM10 Y=01
 - Password: Password assigned by CM2B Y=00
 - (b) *For MAC address authentication (Automatic Login Mode) (CM08>513: 1, CM15 Y=480: 7)*
 - User ID: IP Station No. assigned by CM10 Y=01
 - Password: Password assigned by CM2B Y=10
- NOTE 4:** *“SIP server” used here refers to the SV9300. The SV9300 acts as SIP server where DT700 Series terminal is registered.*
- NOTE 5:** *Follow the IP address setting (CM0B Y=1XX>00) of VOIP port on SV9300.*
- NOTE 6:** *Follow the SIP Server port setting (CM0B Y=1XX>32) on SV9300 (example: change 5060 to 5080 when default is set).*
- NOTE 7:** *Howler Tone to be sent depends on the system data setting (CM08>153, CM13 Y=04) of SV9300 (default: To send).*
- NOTE 8:** *Be sure not to change from default value to the others.*

On the login window for Web Programming, enter the User ID (default value: USER) and the password (default value: 0000) to login.

Setting

Parameter	Description	Default
User Setting	Sets user interface related.	-
Incoming Call	Sets ringing and lamp indication data for incoming call.	-
Offhook Ring	Selects whether to use offhook ringing (ringing for incoming call in busy state).	Enable
1. Disable	Offhook ringing is OFF.	
2. Enable	Offhook ringing is ON.	
Ring Tone	Sets Ringing Tone for incoming call termination.	-
External Call NOTE 1	Sets Ringing Tone for external call termination.	Automatic
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.	
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.	
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.	
Internal Call NOTE 1	Sets Ringing Tone for station call termination.	
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.	
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.	
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.	
Talk	Sets call related data.	-
RTP Alarm NOTE 2	Sets whether to allow the alarm for detecting packet loss in receiving speech packets.	Automatic
0. Automatic	Applies RTP alarm setting on the SV9300.	
1. Disable	Alarm is OFF.	
2. Enable	Alarm is ON.	
DTMF Tone (Setting prohibited)	-	Automatic
0. Automatic	-	
1. Disable (Setting prohibited)	-	
2. Enable (Setting prohibited)	-	
Key Touch Tone	Sets touch tone for pressing digit key/touch-sensitive panel.	Automatic NOTE 3
0. Automatic	Sets Automatic to key touch tone.	
1. No Tone	Sets key touch tone disabled.	
2. Tone	Sets key touch tone enabled.	
Hold Music	Sets Music-on-Hold.	Default
1. Default	Applies the default music (Minuet).	
2. Download	Applies the downloaded music.	
Display	Sets display related data.	-
Language	Selects the language on date and time display of Web page.	Automatic NOTE 4
0. Automatic	Sets Automatic for language setting on date and time display of Web page.	
1. Japanese	Selected language is displayed on date and time display of Web page.	
2. NEC English		

Setting (Continued)

Parameter	Description	Default
Change Password NOTE 5	Assigns the password of the terminal, used for data clear, security, and login of user mode.	0000
Peripherals	Setting for using peripheral equipment of terminal.	-
USB Memory	Setting for safely removing USB memory from the terminal.	-
1. Remove		
BCA Option	Setting for BCA (Bluetooth cradle phone adapter)	-
BT Connection Settings	Setting for Bluetooth Connection.	-
Auto Connection	Selects whether to use auto connection.	Enable
Disable	Auto connection is not used.	
Enable	Auto connection is used.	
Connection Cycle	Sets Bluetooth connection cycle.	10 Seconds
3 Seconds	Bluetooth connection cycle is 3 seconds.	
10 Seconds	Bluetooth connection cycle is 10 seconds.	
1 Minute	Bluetooth connection cycle is 1 minute.	
10 Minutes	Bluetooth connection cycle is 10 minutes.	
Interrupt Ringing	Selects whether to use Interrupt Ringing.	Enable
Disable	Interrupt Ringing is OFF.	
Enable	Interrupt Ringing is ON.	
Mobile Answer Mode	Specifies mobile answer mode.	Auto Detect
Auto Detect	Auto detect is used.	
Redirect	Redirect is used.	
Acoustic Setting	Sets acoustic type and audio.	-
Acoustic Type Select	Selects Acoustic Type.	Type 1
Handset TX Gain	Sets sending gain of handset [-15dB~15dB].	0
Handset RX Gain	Sets receiving gain of handset [-18dB~18dB].	
HF TX Gain	Sets sending gain of hands free [-15dB~15dB].	
HF RX Gain	Sets receiving gain of hands free [-18dB~18dB].	
Setting Reset	Initializes all the assigned personal information on the terminal to factory default.	-
Download	Sets download related data.	-
Download Files	Downloads files of Music-on-Hold, Ringing Tone, and Phonebook.	-
Hold Music	Downloads a sound file used for Music-on-Hold.	MOH.wav
Ring Tone	Downloads a sound file used for Ringing Tone.	-
1. Download 1		Melody1.wav
2. Download 2	Downloads Melody 1 ~ Melody 3.	Melody2.wav
3. Download 3		Melody3.wav
Download Address	Sets IP address of FTP/TFTP server.	0.0.0.0
Protocol	Specifies the protocol for file downloading.	FTP
1. FTP	FTP is used.	
2. TFTP	TFTP is used.	
FTP Settings	Sets data for accessing FTP server.	-
1. User ID	Sets user ID for login to FTP server.	
2. Password	Sets password for login to FTP server.	
3. Folder	Specifies the directory to save the download file.	

Setting (Continued)

Parameter	Description	Default
Data Backup/Restore	Makes backup of terminal data or restores from backup file.	-
Data Backup	Saves the terminal setting to backup file.	-
Data Restore	Restores the terminal data by reading out the saved backup file.	-
Server Address	Sets IP address of FTP/TFTP server.	0.0.0.0
Protocol	Specifies the protocol for file downloading.	FTP
1. FTP	FTP is used.	
2. TFTP	TFTP is used.	
FTP Settings	Sets data for accessing FTP server.	-
1. User ID	Sets user ID for login to FTP server.	
2. Password	Sets password for login to FTP server.	
3. Folder	Specifies the directory to save backup file.	

- NOTE 1:** To provide the Distinctive Ringing of external call/internal call by SV9300, be sure to use with default (Automatic). When “Automatic” is set, by the system data setting (CM08>390) of SV9300, Ringing Tone can be changed by the system data (CM15 Y=491, CM35 Y=034, 164) or terminal operations (“Feature” + “3”) (default: as per CM15 Y=491, CM35 Y=034, 164).
- NOTE 2:** When “Automatic” is set, RTP Alarm depends on the system data setting (CM08>1004) of SV9300 (default: To send).
- NOTE 3:** Follows the setting (CM15 Y=291) of SV9300 side when “Automatic” is set. In all other cases, follows the setting of a terminal side.
- NOTE 4:** The setting of display language on the Clock Display screen depends on the system data settings (CM12 Y=63, CM04 Y=00) of SV9300.
- NOTE 5:** Change Password is used for password for User Mode of Web programming and security unlock. Be sure to change the default password.

DT800 SERIES CONFIGURATION MENU LIST

This section briefly explains about the each parameter and the meaning of DT800 Series configuration menus.

* Configuration menus listed in this section correspond to the following firmware version. Except DT820 Series Configuration Menu List (FW 2.0.0.0), if there are additional parameters by the firmware upgrade, corresponding firmware version is listed in an each field.

- DT830G Series Configuration Menu List (FW 3.0.3.0 or later) [☞ Page D-63](#)
- DT830 Series Configuration Menu List (FW 3.0.3.0 or later) [☞ Page D-90](#)
- DT820 Series Configuration Menu List (FW 2.0.0.0) [☞ Page D-118](#)
- DT820 Series Configuration Menu List (FW 3.0.0.0 or later) [☞ Page D-136](#)

DT830G Series Configuration Menu List (FW 3.0.3.0 or later)

This section lists the configuration menu used for DT830G Series configuration settings. DT830G (12/24CG), and DT830G (12/24DG) columns on the tables below show whether the menu is available or not for each terminal type.

LEGEND	X: Available	N: Not available	-: Not applicable
---------------	--------------	------------------	-------------------

NOTE 1: Up to four Service menus are displayed. Service menu is not displayed when a service URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL].

NOTE 2: Phone menu is displayed when a URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL]. Usually, "Application Name" is displayed when a URL is registered.

NOTE 3: The Config menu is not displayed on the menu window. To open the Config menu, press digit key "0". In addition, by pressing "Hold", "Transfer", "*", "#" key in this order in an idle state of telephone, you can access to the login screen of Configuration Menu.

1. History

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
1. History	Lists/deletes the history of outgoing/incoming call	-	X	X

When "1. History" is selected, the "History menu" of SV9300 is displayed.

[History]	
NEC ICHIRO	[300]
3:09 PM	TUE 18 OCT 2016
Outgoing	Incoming

NOTE: In SV9300, the built-in Call History of a telephone cannot be used. Call History on SV9300 side is used.

2. Directory

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
2. Directory	Displays Terminal Phonebook	-	X	X

When “2. Directory” is selected, the “Directory service” screen of SV9300 is displayed.

[Directory]

Name>>

Set Common Personal

NOTE: In SV9300, the built-in Directory of a telephone cannot be used. The Directory service on SV9300 side is used.

3. Service

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
3. Service	Starts up XML browser.	-	X	X
1. Service 1	Starts up XML browser.	-	X	X
2. Service 2		-	X	X
3. Service 3		-	X	X
4. Service 4		-	X	X

NOTE: At most four services are displayed on the menu. A service is not displayed when a service URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL].

4. Setting

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
1. User Setting	Sets user interface related.	-	X	X
1. Incoming Call	Sets ringing and lamp indication data for incoming call.	-	X	X
1. Ring Volume (Setting prohibited)	-	-	-	-
2. Offhook Ring (Setting prohibited)	-	Enable	-	-
1. Disable (Setting prohibited)	-			
2. Enable	-			
3. Headset Ring	Selects whether to use headset ringing.	Disable	X	X
1. Disable	Headset ringing is OFF.			
2. Enable	Headset ringing is ON.			

4. Setting (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. Ring Tone	Sets Ringing Tone for incoming call termination.	-	X	X
1. External Call NOTE 1	Sets Ringing Tone for external call termination.	Automatic	X	X
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3 NOTE 2	Sets Download 1 ~ Download 3 for Ringing Tone.			
2. Internal Call NOTE 1	Sets Ringing Tone for station call termination.			
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3 NOTE 2	Sets Download 1 ~ Download 3 for Ringing Tone.			
3. IM	Not used.	-	-	-
5. Illumination	Sets lamp indication pattern for incoming call termination.	-	X	X
1. External Call NOTE 3	Sets Illumination for External Call.	-	X	X
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.	Automatic	X	X
1. Disable	Lamp indication is OFF.			
2. Red	Specifies the color of lamp indication for incoming call termination.			
3. Green				
4. Blue				
5. Yellow				
6. Purple				
7. Light Blue				
8. White				
9. Rotation				

4. Setting (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
2. Internal Call NOTE 3	Sets Illumination for Internal Call.	-	X	X
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.	Automatic	X	X
1. Disable	Lamp indication is OFF.			
2. Red	Specifies the color of lamp indication for incoming call termination.			
3. Green				
4. Blue				
5. Yellow				
6. Purple				
7. Light Blue				
8. White				
9. Rotation				
3. IM (Setting prohibited)	-	-	-	-
6. Headset Ring Volume (Setting prohibited)	-	-	-	-
2. Talk	Sets call related data.	-	X	X
1. RTP Alarm NOTE 4	Sets whether to allow the alarm for detecting packet loss in receiving speech packets.	Automatic	X	X
0. Automatic	Applies RTP alarm setting on the SV9300.			
1. Disable	Alarm is OFF.			
2. Enable	Alarm is ON.			
2. DTMF Tone (Setting prohibited)	-	Automatic	X	X
0. Automatic	-			
1. Disable (Setting prohibited)	-			
2. Enable (Setting prohibited)	-			
3. Key Touch Tone	Sets touch tone for pressing digit key/touch-sensitive panel.	Automatic NOTE 5	X	X
0. Automatic	Sets Automatic to key touch tone.			
1. No Tone	Sets key touch tone disabled.			
2. Tone	Sets key touch tone enabled.			
3. Voice	Enables the key touch tone by voice recognition.			
4. Hold Music	Sets Music-on-Hold.	Default	X	X
1. Default	Applies the default music (Minuet).			
2. Download	Applies the downloaded music.			
5. Prefix (Setting prohibited)	-	-	X	X
1. Prefix 1 ~ 3. Prefix 3 (Setting prohibited)	-	-	X	X

4. Setting (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
3. Display	Sets display related data.	-	-	-
1. Calendar Format	-	Automatic NOTE 6	-	-
1. Calendar Format (Setting prohibited)	-			
0. Automatic	-			
1. Type 1 (Setting prohibited)	-			
2. Type 2 (Setting prohibited)				
3. Type 3 (Setting prohibited)				
4. Type 4 (Setting prohibited)				
5. Type 5 (Setting prohibited)				
2. Time Format (Setting prohibited)	-	Automatic NOTE 6	-	-
0. Automatic	-			
1. 12 hour Indication (Setting prohibited)	-			
2. 24 hour Indication (Setting prohibited)	-			
2. Local Volume	Selects whether to display volume gage when adjusting the volume level.	Automatic NOTE 8	X	X
0. Automatic	Sets Automatic for volume gage display.			
1. Disable	Volume gage is not displayed.			
2. Enable	Volume gage is displayed.			
3. Screen Saver	-	-	-	-
1. Screen Saver Mode	Selects whether to use Screen Saver.	Disable	X	X
1. Disable	Screen Saver is not used.			
2. Enable	Screen Saver is used.			
2. Wait Time	Sets wait time until Screen Saver activation (1- 999 min.).	120	X	X
4. Back Light	Switches ON/OFF the backlight on DT830G (12/24DG).	Enable NOTE 9	-	X
1. Disable	Backlight is OFF.			
2. Enable	Backlight is ON.			
4. Back Light	Turns up/down the backlight on DT830G (12/ 24CG).	Level 3	X	-
1. Level 1 (Bright) - 4. Level 4 (Dark)	Select the bright level of backlight.			
5. Font Size NOTE 10	Switches the font size on configuration menu.	Standard	X	X
1. Standard (28 dots (12/ 24CG)/16 dots (12/ 24DG))	Selects Standard or Small of font size.			
2. Small (24 dots (12/ 24CG)/12 dots (12/ 24DG))				

4. Setting (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)	
6. Language	Selects the language on date and time display.				
0. Automatic	Sets Automatic for language setting on date and time display.				
1. Japanese	Selected language is displayed on date and time display. * For FW4.0.0.0 or later, each parameter in this menu is displayed in their native language.	Automatic NOTE 11	X	X	
2. NEC English					
3. NEC Portuguese					
4. NEC Spanish					
5. NEC French					
6. German					
7. Italian					
8. Dutch					
9. Norwegian					
10. Danish					
11. Swedish					
12. Greek					
13. English					
14. Portuguese					
15. Spanish					
16. French					
17. Russian					
18. Turkish					
19. Azərbaycanca [FW5.1.0.0 or later]					
20. Romania					
21. Polish					
22. Catalan					
23. Korean					
24. Chinese (Simplified Chinese) [FW4.0.0.0 or later]				-	
25. Chinese (Traditional Chinese) [FW4.0.0.0 or later]				-	
7. Advanced	Sets other data for display.	-	X	X	
1. Backlight Fade Control	Sets whether Backlight Fade Control is in service, when turning on/off the backlight of the display.	Enable	-	X	
1. Disable	Disables Backlight Fade Control.				
2. Enable	Enables Backlight Fade Control.				
1. Wallpaper	Sets the Home screen background.	Default	X	-	
1. Default	Sets the default screen background.				
2. Download	Sets the download data file.				
2. Shortcut Icon (Setting prohibited)	-	-	-	-	

4. Setting (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
3. Font Color	Sets the font color for Time Display/Feature Information Display area. * You can check the color while selecting.	Color 2	X	-
1~16. Color 1 ~ Color 16	Sets Font Color 1 ~ Font Color 16.			
4. Turn Off Display	Sets the data when turning off the backlight of the display automatically.	-		
1. Turn Off Display Mode	Sets whether Turn Off Display Mode is in service.	App Priority	X	-
1. Disable	Disables Turn Off Display Mode.			
2. Enable	Enables Turn Off Display Mode.			
3. App Priority	Sets to depend on an application.			
2. Wait Time	Sets the time until turning off the display (1-999 min.).	180		
5. Backlight Fade Control	Sets whether Backlight Fade Control is in service, when turning on/off the backlight of the display.	Enable	X	X
1. Disable	Disables Backlight Fade Control.			
2. Enable	Enables Backlight Fade Control.			
4. Change Password	Assigns the password of the terminal, used for data clear, security, and login of user mode. NOTE 12	0000	X	X
5. Security	Setting for security.	-	-	-
1. VPN [FW5.2.0.0 or later]	Setting of VPN for security.	0.0.0.0	N	N
1. IP Address (PC)	Sets the IP address of PC connected to a PC port when VPN is enabled.			
6. Peripherals	Setting for using peripheral equipment of terminal.	-		
1. USB Memory	Setting for safely removing USB memory from the terminal.	-		
1. Remove				
2. BCA Option	Setting for the configuration of BCA.	-		
1. BCA Connection Settings	Setting for the BCA connection.	-		
1. Auto Connection	Sets whether Auto Connection is in service.	Enable	X	X
1. Disable	Disables Auto Connection.			
2. Enable	Enables Auto Connection.			
2. Connection Cycle	Selects connection cycle of Auto Connection.	10 Seconds		
1. 3 Seconds	Connection cycle is 3 seconds.			
2. 10 Seconds	Connection cycle is 10 seconds.			
3. 1 Minutes	Connection cycle is 1 minute.			
4. 10 Minutes	Connection cycle is 10 minute.			
2. Interrupt Ringing	Sets whether Interrupt Ringing is in service.	Enable		
1. Disable	Disables Interrupt Ringing.			
2. Enable	Enables Interrupt Ringing.			
3. Smart Device Type	Sets the type of Smart Device.	Type 1		
1. Type 1	Sets the Smart Device type as Type 1.			
2. Type 2	Sets the Smart Device type as Type 2.			

4. Setting (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. Auto Path Switch	Sets whether Auto Path Switch is in service.	Enable		
1. Disable	Disables Auto Path Switch.			
2. Enable	Enables Auto Path Switch.			
5. Illumination	Sets lamp indication pattern for incoming call termination.	Blue	X	X
1. Disable	Specifies the color of lamp indication for incoming call termination.			
2. Red				
3. Green				
4. Blue				
5. Yellow				
6. Purple				
7. Light Blue				
8. White				
9. Rotation				
7. Usability	Sets the operation of Help key while pop-up window is displayed.	-	X	X
1. Help Key Mode	Setting for Help Key Mode (SubMenu/Popup Window)	SubMenu	X	X
1. SubMenu	Sets the Help key which displays sub menu		X	X
2. Popup Window	Sets the Help key which displays pop-up window		X	X
0. Setting Reset	Initializes all the assigned personal information on the terminal to factory default.	-	X	X
2. Download	Sets download related data.	-	X	X
1. Download Files	Downloads files of Music-on-Hold, Ringing Tone, and Phonebook.	-	X	X
1. Hold Music	Downloads a sound file used for Music-on-Hold.	MOH.wav	X	X
2. Ring Tone	Downloads a sound file used for Ringing Tone.	-	X	X
1. Download 1	Downloads Melody 1 ~ Melody 3.	Melody1.wav		
2. Download 2		Melody2.wav		
3. Download 3		Melody3.wav		
3. Directory (Setting prohibited)	-	-	-	-
4. Wallpaper	Downloads a file used for wallpaper.	WallPaper.bmp	X	-
2. Download Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X	X
3. Protocol	Specifies the protocol for file downloading.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.			

4. Setting (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. FTP Settings NOTE 13	Sets data for accessing FTP server.			
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.			
3. Folder	Specifies the directory to save the download file.			
3. Data Backup/Restore	Makes backup of terminal data or restores from backup file.	-		
1. Data Backup	Saves the terminal setting to backup file. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
2. Data Restore	Restores the terminal data by reading out the saved backup file. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
3. Server Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X	X
4. Protocol	Specifies the protocol for file downloading.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.			
5. FTP Settings NOTE 14	Sets data for accessing FTP server.			
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.			
3. Folder	Specifies the directory to save backup file.			

NOTE 1: To provide the Distinctive Ringing of external call/internal call by SV9300, be sure to use with default (Automatic). When "Automatic" is set, by the system data setting (CM08>390) of SV9300, Ringing Tone can be changed by the system data (CM15 Y=491, CM35 Y=034, 164) or terminal operations ("Feature" + "3") (default: as per CM15 Y=491, CM35 Y=034, 164).

NOTE 2: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes "15. Music Ring 1~17. Music Ring 3" (formerly "15. Download 1~17. Download 3"). The setting details are as follows.

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
15. Music Ring 1~17. Music Ring 3	Sets Music Ring 1 ~ Music Ring 3.	Automatic	X	X

NOTE 3: When "Automatic" is set, by the system data setting (CM08>144) of SV9300, the illumination color can be changed to Red or Green (default: Red).

Set the "0. Automatic" to use color coding by 7-color LED.

NOTE 4: When "Automatic" is set, RTP Alarm depends on the system data setting (CM08>1004) of SV9300 (default: To send).

NOTE 5: Follows the setting (CM15 Y=291) of SV9300 side when "Automatic" is set. In all other cases, follows the setting of a terminal side.

NOTE 6: Calendar Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.

- NOTE 7:** Time Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.
- NOTE 8:** A setting for Enable is applied when Automatic has been set.
- NOTE 9:** Back Light menu becomes effective when the system data (CM15 Y=485) of SV9300 is set to other than 0 (Always off).
Back Light of ten key cannot be used when terminals and option equipment for power failure are implemented.
- NOTE 10:** When a font size is changed, press the Cursor key and return to the previous menu, then the font size becomes effective.
- NOTE 11:** The setting of display language on the Clock Display screen depends on the system data settings (CM12 Y=63, CM04 Y=00) of SV9300.
- NOTE 12:** Change Password is used for password for User Mode of Web programming and security unlock. Be sure to change the default password.
- NOTE 13:** If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “4. Account Settings” (formerly “4. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. Account Settings	Account settings of download server (FTP/HTTPS).			
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).			
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).			

- NOTE 14:** If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “5. Account Settings” (formerly “5. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
5. Account Settings	Account settings of backup/restore server (FTP/HTTPS).			
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).			
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).			

5. Phone

Depending on the Home URL setting by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL], this menu is displayed as follows.

- When Home URL is not registered, home screen is displayed and the menu name is displayed as “Phone”.
- When Home URL is registered, XML application screen is displayed, and the menu name is displayed as the application name to be described in the XML application.

0. Config

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
1. Network Settings	Sets network related data.	-	X	X
1. DHCP Mode	Selects whether to use DHCP protocol.	Enable	X	X
1. Disable	DHCP is not used.			
2. Enable	DHCP is used.			
2. IP Address	Sets DT800 Series terminal's IP address.	0.0.0.0	X	X
3. Default Gateway	Sets default router's IP address.	0.0.0.0	X	X
4. Subnet Mask	Sets subnet mask.	0.0.0.0	X	X
5. DNS Address	Sets DNS server's IP address.	0.0.0.0	X	X
6. Advanced Settings	Sets network expanded data.	-	X	X
1. LAN Port Settings	Sets LAN port.	-	X	X
1. Speed&Duplex	Sets communication speed of Ether port.	Autonego	X	X
1. Autonego	Autonegotiation			
2. 100M Full	100Mbps in full-duplex mode			
3. 100M Half	100Mbps in half-duplex mode			
4. 10M Full	10Mbps in full-duplex mode			
5. 10M Half	10Mbps in half-duplex mode			
2. VLAN Mode	Selects whether to use VLAN on terminal.	Disable	X	X
1. Disable	VLAN is not used.			
2. Enable	VLAN is used.			
3. VLAN ID	Sets VLAN ID of terminal [1-4094].	7	X	X
4. VLAN Priority	Sets VLAN Priority of terminal [0-7].	5	X	X
5. Auto-MDIX	Selects whether to use Auto-MDIX.	Enable	X	X
1. Disable	Auto-MDIX is not used.			
2. Enable	Auto-MDIX is used.			
2. PC Port Settings	Sets PC port.	-	X	X
1. Speed&Duplex	Sets communication speed of PC port.	Autonego	X	X
1. Autonego	Autonegotiation			
2. 100M Full	100Mbps in full-duplex mode			
3. 100M Half	100Mbps in half-duplex mode			
4. 10M Full	10Mbps in full-duplex mode			
5. 10M Half	10Mbps in half-duplex mode			
2. VLAN Mode	Selects whether to use port VLAN of PC port.	Disable	X	X
1. Disable	Port VLAN of PC port is not used.			
2. Enable	Port VLAN of PC port is used.			
3. Port VLAN ID	Sets port VLAN ID of PC port [1-4094].	1	X	X
4. Port VLAN Priority	Sets port VLAN priority of PC port [0-7].	0	X	X
5. Port Available	Selects whether to use PC port.	Automatic	X	X
0. Automatic	Sets Automatic for PC port.			
1. Disable	PC port is not used.			
6. EAPOL	Select whether to forward EAPOL packet from PC port.	Disable	X	X
1. Disable	Not forward EAPOL packet from PC port.			
2. Enable	Forwards EAPOL packet from PC port.			

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
7. PC Port Security	Select whether to clear authentication status of the terminal which is connected to PC port for authentication switch when the link of PC port has been down.	Enable	X	X
1. Disable	Not clear the authentication status.			
2. Enable	Clear the authentication status.			
8. Auto-MDIX	Selects whether to use Auto-MDIX.	Enable	X	X
1. Disable	Auto-MDIX is not used.			
2. Enable	Auto-MDIX is used.			
3. CDP Mode (Setting prohibited)	-	-	-	-
4. Type of Service	Sets ToS value.	-		
1. RTP	Sets ToS value for RTP in Hex.	A0	X	X
2. SIP	Sets ToS value for SIP in Hex.	C0		
5. Self Port Settings	Sets each port for DT800 Series.	-		
1. RTP Self Port	Sets RTP port for DT800 Series [1024-65520 (except 59986-60000)]. NOTE 1 * Odd port numbers cannot be set.	3462	X	X
2. SIP Self Port	Sets SIP port for DT800 Series [1024-65534]. NOTE 1	5060	X	X
6. LLDP Settings	Sets information related to LLDP.	-		
1. LLDP Mode	Sets whether to transmit/receive LLDP packets.	Enable	X	X
1. Disable	DT800 Series does not transmit/receive LLDP packets.			
2. Enable	DT800 Series transmits/receives LLDP packets.			
2. Auto Setting Mode	Sets whether to perform data setting automatically when LLDP packets are received.	Enable	X	X
1. Disable	DT800 Series does not perform data setting automatically when LLDP packets are received.			
2. Enable	DT800 Series performs data setting automatically when LLDP packets are received.			
3. RX Waiting Time	For Auto Setting, sets the waiting time until DT800 Series receives LLDP packets (1-60 seconds).	15	X	X
4. Transmit Interval	Sets intervals to transmit LLDP packets (5-32768 seconds).	30	X	X
5. Hold Multiplier NOTE 2	Sets a multiplier determining the time to live for the LLDP packets to be transmitted (2-10 multiples).	4	X	X
6. Fast Start Count	Sets the number of times to transmit LLDP packets at one-second intervals after DT800 Series starts up (1-10 times).	3	X	X

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
7. Transparent Mode	Sets whether Transparent Mode of LLDP packets is enabled between the PC port and the LAN port.	Enable	X	X
1. Disable	Transparent Mode of LLDP packets is disabled.			
2. Enable	Transparent Mode of LLDP packets is enabled.			
8. Asset ID	Sets the Asset ID for LLDP packets to be transmitted. * "-" is displayed in default settings.	-	X	X
7. Spare IP Settings (Setting prohibited)	-	-	-	-
1. Spare/Backup IP Mode (Setting prohibited)	-	Disable	-	-
1. Disable	-			
2. Spare IP (Setting prohibited)	-			
3. Backup IP (Setting prohibited)	-			
2. Network Settings (Setting prohibited)	-	-	-	-
1. IP Address (Setting prohibited)	-	0.0.0.0		
2. Default Gateway (Setting prohibited)	-	0.0.0.0		
3. Subnet Mask (Setting prohibited)	-	0.0.0.0		
4. DNS Address (Setting prohibited)	-	0.0.0.0		
3. SIP Settings (Setting prohibited)	-	-	-	-
1. 2nd Server Address (Setting prohibited)	-	0.0.0.0	-	-
2. 3rd Server Address (Setting prohibited)	-	0.0.0.0	-	-
3. 4th Server Address (Setting prohibited)	-	0.0.0.0	-	-
4. 2nd Server Port (Setting prohibited)	-	0	-	-
5. 3rd Server Port (Setting prohibited)	-	0	-	-
6. 4th Server Port (Setting prohibited)	-	0	-	-
8. TTL Settings	Setting for multicast TTL (Time To Live)	-	X	X
1. RTP (Multicast)	Sets the value of multicast TTL (0-255)	32		
9. Priority Queue	Sets whether to enable Priority Queue of the internal switching HUB.	Disable	X	X
1. Disable	Priority Queue is disabled.			
2. Enable	Priority Queue is enabled.			

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)	
2. SIP Settings	Sets SIP related data.	-	X	X	
1. SIP User	-	-	-	-	
1. User ID (Setting prohibited)	-	-	-	-	
2. Password (Setting prohibited)	-	-	-	-	
3. Extension Number (Setting prohibited)	-	-	-	-	
4. Backup Logon (Setting prohibited)	-	Disable	-	-	
2. Server Address & URI	Sets IP address or URI of SIP server. NOTE 3	-	X	X	
1. 1st Server Address	Sets IP address or fully qualified domain name (FQDN) of 1st ~ 4th SIP server. NOTE 4	0.0.0.0	X	X	
2. 2nd Server Address					
3. 3rd Server Address					
4. 4th Server Address					
5. Registrar Destination (Setting prohibited)	-	0.0.0.0	-	-	
1. 1st Register Address (Setting prohibited)	-	0.0.0.0	-	-	
2. 2nd Register Address (Setting prohibited)					
3. 3rd Register Address (Setting prohibited)					
4. 4th Register Address (Setting prohibited)					
3. Access Mode	Sets access mode for SIP server.	Normal	X	X	
1. Normal	Normal access				
2. Remote	Remote access				
4. SIP Server Port	Sets SIP server port [1024-65535].	5060	X	X	
1. 1st Server Port	Specifies the destination port number of sending packet to 1st ~ 4th SIP server. NOTE 5				
2. 2nd Server Port					
3. 3rd Server Port					
4. 4th Server Port					
5. Registrar Destination (Setting prohibited)	-				
1. 1st Register Port (Setting prohibited)	-			-	-
2. 2nd Register Port (Setting prohibited)					
3. 3rd Register Port (Setting prohibited)					
4. 4th Register Port (Setting prohibited)					

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
5. Area ID	Sets the area (country) using DT800 Series.			
0. Automatic	The SV9300 side setting is applied when Automatic is selected.			
1. Japan	Selects the area (country) NOTE: Area IDs allowed on the SV9300 can be selected.	Automatic	X	X
2. America (Canada and North America)				
3. Australia				
4. Others				
5. Hong Kong				
6. Malaysia				
7. Singapore				
8. Mexico				
9. Taiwan				
10. New Zealand				
11. Korea				
12. Brazil				
13. China				
14. Thailand				
15. Thailand (EGAT)				
16. Sri Lanka				
17. Germany				
18. Italy				
19. Netherlands				
20. Catalonia				
21. Denmark				
22. Sweden				
23. Poland				
24. Czech				
25. Norway				
26. France				
27. Portugal				
28. Spain				
29. Austria				
30. Belgium				
31. UK				
32. Greece				
33. Switzerland				
34. South Africa				
35. Brazil (EMEA)				

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
6. Extra Function (Setting prohibited)	-	-	-	-
1. Retransmit Interval (Setting prohibited)	-	Standard	-	-
1. Standard	-			
2. 6 Seconds (Setting prohibited)	-			
2. Keeping Call (Setting prohibited) NOTE 6	-	-	-	-
1. Disable (Setting prohibited)	-	Enable	-	-
2. Enable	-			
3. Howler Tone (Setting prohibited) NOTE 7	-	Disable	-	-
1. Disable	-			
2. Enable (Setting prohibited)	-			
4. Random Wait Time [FW5.0.0.0 or later]	Sets random wait time of when the connection between SV9300 and terminal starts (5-3600 seconds). * Do not change the factory default value of this parameter.	5	X	X
5. Connect Wait Time [FW5.0.0.0 or later]	Sets the time to wait until the connection between SV9300 and terminal is complete (30-300 seconds). * Do not change the factory default value of this parameter.	30	X	X
7. Encryption	Sets encryption data.	-	X	X
1. Authentication Mode	Selects whether to use authentication.	Disable	X	X
1. Disable	Authentication is not used.			
2. Enable	Authentication is used.			
2. One Time Password	Sets One Time Password.	-	X	X
3. Authentication Status	Displays authentication status.	-	X	X
8. NAT Traversal (Setting prohibited)	-	-	-	-
1. NAT Traverse Mode (Setting prohibited)	-	Disable	-	-
1. Disable				
2. Dynamic (Setting prohibited)				
3. Static (Setting prohibited)				
2. Network Area Name (Setting prohibited)	-	-	-	-

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
3. WAN Settings (Setting prohibited)		-	-	-
1. WAN Mate IP Address (Setting prohibited)		0.0.0.0	-	-
2. WAN SIP Mate Port (Setting prohibited)		5060	-	-
3. WAN Self IP Address (Setting prohibited)		0.0.0.0	-	-
3. Maintenance Settings	Sets maintenance related data.	-	X	X
1. Download Menu	Downloads specified file to DT800 Series.	-	X	X
1. Download Files	Specifies download file name for each file type.	-	X	X
1. Boot & Program	Downloads boot and program.	12/24CG: itzisipvc.tgz 12/24DG: itzisipvg.tgz	X	X
2. Config	Downloads configuration file.	config.tgz	X	X
3. Voice File	Downloads voice file.	voice.tgz	X	X
4. Language Pack	Downloads translation data file.	LangPack.tgz	X	X
5. Certificate [FW5.0.0.0 or later]	Downloads certificate files.	-		
1. Client Certificate	Downloads client certificate file.	clientcert.p12	X	X
2. Root Certificate	Downloads root certificate file.	rootcert.der		
2. Download Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X	X
3. Protocol	Selects FTP or TFTP for file downloading protocol.			
1. FTP	FTP is used.	FTP	X	X
2. TFTP	TFTP is used.			
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.			
4. FTP Settings NOTE 8	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.	-	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X
2. Hard Reset	Resets DT800 Series without data saving.	-	X	X
3. Data Clear	Initializes DT800 Series selecting the initialization mode.	-	X	X
1. Factory Value	Initializes DT800 Series to factory default.	-	X	X
2. Terminal Data	Clears terminal data on DT800 Series.	-	X	X
3. Personal Data	Clears personal data on DT800 Series.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. Adjust	Adjusts LCD, audio, and button kit.	-	X	X
1. Default Audio	Sets audio data.	-	X	X
1. Sidetone Volume	Sets Side Tone volume [0-7].	4	X	X
2. Send Gain	Sets sending gain of handset [0-31].	24	X	X
3. Receive Bias	Sets receiving bias of handset [0-7].	0	X	X
4. Receive Gain	Sets receiving gain of handset [0-31].	22	X	X
5. Talk Characteristics	Sets talk characteristics of handset.	-	X	X
1. Noise Suppression	Sets whether to enable the Noise Suppressor. (Set "1. Disable" in normal operation.)	Disable	X	X
1. Disable	Disables the Noise Suppressor.			
2. Enable	Enables the Noise Suppressor.			
2. Handsfree Mode	Sets a send gain mode of handsfree call.	Retro Mode	X	X
1. Retro Mode	Sets to Retro mode.			
2. New Mode	Sets to New mode.			
6. DTMF Send Level	Specifies the DTMF Send Level [0-48] (level 0 (maximum) ~ level 48 (minimum))	32	X	X
2. Key Kit Type	Sets additional button kit data.	-	X	X
1-5 Kit 1, 2, 9, 10 NOTE 9	Key 1, 2, 9, 10 on button kit are used.	Kit 1	X	X
1. Line Key Type	(Do not change the factory default value of this parameter.)	-	X	X
1. Line 12	Sets to 12-line of DT830G.		X	X
2~7. Line 24, 32, 8, 6, 2, 0	Not used.		-	-
3. LCD Contrast	Adjusts LCD Contrast.	Area Contrast (Narrow)	-	X
0. Area Contrast				
1. Wide				
2. Narrow				
4. BCA Audio	Setting for the acoustic type and audio of BCA.	-	X	X
1. Acoustic Type	Selects acoustic type.	Type 1		
1. Type 1 - 5. Type 5				
2. Handset TX Gain	Sets sending gain of handset [0dB~30dB].	15		
3. Handset RX Gain	Sets receiving gain of handset [0dB~36dB].	18		
4. HF TX Gain	Sets sending gain of hands free [0dB~30dB].	15		
5. HF RX Gain	Sets receiving gain of hands free [0dB~36dB].	18		
6. Noise Suppression	Setting for the Noise Suppressor.	-		
1. TX	Sets the Noise Suppression level for TX.	Disable		
2. RX	Sets the Noise Suppression level for RX.	Disable		
5. ADA Audio [FW4.0.0.0 or later]	-	Disable	-	-
1. Echo Canceller (Setting prohibited)				
6. EHS Cable NOTE 10 [FW5.0.0.0 or later]	Selects whether to use the cable for wireless headset connection.	None	X	X
1. None	Wireless headset is not used (instead of it, headset is used).			
2. APN-91	Wireless headset is used.			

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
7. Character Code (Setting prohibited) [FW5.1.0.0 or later]	-	Automatic	-	-
1. Automatic	-		-	-
2. Eastern Europe (Setting prohibited)	-		-	-
8. Headset	Adjust headset	0 dB	X	X
1. Send Level	Sets voice sending level of headset.			
1. 0 dB	Not change the sending voice level from default.			
2. -5 dB	Set -5dB lower voice level than default.			
3. -10 dB	Set -10dB lower voice level than default.			
5. Auto config	Selects whether to use auto-config function.	Enable	X	X
1. Disable	Auto config is not used.			
2. Enable	Auto config is used.			
6. Data Maintenance	Sets data maintenance related.	-	X	X
1. Backup	Executes data backup.	-	X	X
1. All Data	Executes backup for all the configuration settings and data. * For encryption, "AllData.tgz.ef" is applied as a file name.	AllData.tgz	X	X
2. Terminal Data	Executes backup for terminal settings and data. * For encryption, "TerminalData.tgz.ef" is applied as a file name.	Terminal Data.tgz	X	X
3. Personal Data	Executes backup for personal settings and data. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
2. Restore	Executes restoration.	-	X	X
1. All Data	Restores all the configuration settings and data. * For encryption, "AllData.tgz.ef" is applied as a file name.	AllData.tgz	X	X
2. Terminal Data	Restores terminal settings and data. * For encryption, "TerminalData.tgz.ef" is applied as a file name.	Terminal Data.tgz	X	X
3. Personal Data	Restores personal settings and data. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
3. Server Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X	X
4. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.			
5. FTP Settings NOTE 11	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.	-	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
7. Simple Test NOTE 12	Changes the terminal to test mode.	-	X	X
1. Call Test	Call Test	-	X	X
2. Call Test Address	Sets an IP address of the opposite terminal to perform the call test.	10.136.20. 188	X	X
3. Key Test 1 (Random)	Tests to press the random button.	-	X	X
4. Key Test 2 (Sequential)	Tests to press the sequential button.	-	X	X
5. Loop Back Test	Tests the voice loop back.	-	X	X
1. Handset - Hand Speaker	Tests to speak and receive on the hand speaker.	-	X	X
2. Handset - Speaker	Tests to speak on the handset and receive on the speaker.	-	X	X
3. Headset - Head Speaker	Tests to speak and receive on the head speaker.	-	X	X
4. Headset - Speaker	Tests to speak on the headset and receive on the speaker.	-	X	X
5. MIC - Hand Speaker	Tests to speak on the MIC and receive on the hand speaker.	-	X	X
6. MIC - Head Speaker	Tests to speak on the MIC and receive on the head speaker.	-	X	X
7. MIC - Speaker	Tests to speak on the MIC and receive on the speaker.	-	X	X
6. Speaker Test	Tests the speaker.	-	X	X
1. Japan	A DT for Japan is heard in maximum volume.	-	X	X
2. America	A DT for North America is heard in maximum volume.	-	X	X
7. Receiver Test	Receiver Test for a handset	-	X	X
4. Security	Sets security related data.	-	X	X
1. Web Programming	Selects whether to use Web Programming.			
1. Disable	Web Programming is disabled.	Enable	X	X
2. Enable	Web Programming is enabled.			
2. Push Server	Sets XML's Push server related data.	-	X	X
1. Push Server Access	Sets whether to access XML's Push server.			
1. Disable	Access is disabled.			
2. Enable	Access is enabled.	Enable	X	X
3. Enable (All Client) NOTE 13	Access is enabled. Select this item to set more than one XML's Push server.			
2. Client IP Address NOTE 14	IP address of XML's Push server.			
1. Client 1	Sets IP address of 1st XML's Push server.	0.0.0.0	X	X
2. Client 2	Sets IP address of 2nd XML's Push server.			
3. Client 3	Sets IP address of 3rd XML's Push server.			
4. Client 4	Sets IP address of 4th XML's Push server.			
3. Push Server Port	Port number of XML's Push server [82, 1024-65535]. NOTE 1	82	X	X
3. Admin Password	Sets administrator's password. * Be sure to change the default password.	6633222	X	X

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. 802.1X Supplicant	Sets 802.1X authentication related parameter.	-	X	X
1. Supplicant	Sets whether to use supplicant on DT800 Series.	Disable	X	X
1. Disable	Supplicant is disabled on DT800 Series.			
2. Enable	Supplicant is enabled on DT800 Series.			
2. EAP Method [FW5.0.0.0 or later]	Selects EAP-MD5 or EAP-TLS for the EAP Method.	EAP-MD5	X	X
1. EAP-MD5	EAP-MD5 is used.			
2. EAP-TLS	EAP-TLS is used.			
3. Account	Sets user ID for authentication.	-	X	X
4. Password	Sets password for authentication.	-	X	X
5. EAPOL VLAN Mode	Selects whether to add VLAN tag to EAP packet.	Enable	X	X
1. Disable	VLAN tag is not added to EAP packet.			
2. Enable	VLAN tag is added to EAP packet.			
5. VPN Client NOTE 15 [FW5.0.0.0 to FW5.1.x.x] For FW5.2.0.0 or later, see NOTE 25.	Sets VPN Client related parameter.	-	-	-
1. VPN Mode	Sets whether to enable VPN Mode.	Disable	X	X
1. Disable	VPN Mode is disabled.			
2. Enable	VPN Mode is enabled.			
2. VPN Gateway Address	Sets VPN Gateway address.	0.0.0.0	X	X
3. VPN Network Settings	Sets VPN network related data.	-	X	X
1. VPN IP Address	Sets VPN IP Address.	0.0.0.0		
4. IKE Settings	Sets IKE protocol related data.	-	X	X
1. IKE Version NOTE 17	Selects IKE protocol version.	V1		
1. V1 NOTE 18	IKE protocol version is specified to V1.			
2. V2	IKE protocol version is specified to V2.			
2. Authentication Mode	Selects authentication mode.	PSK		
1. PSK NOTE 19	Authentication mode is specified to PSK.			
2. RSA NOTE 20	Authentication mode is specified to RSA.			
3. PSK Setting	Sets the parameter of PSK (up to 64 characters).	-		
4. IKE ID	Sets the parameter of IKE ID (up to 127 characters).	-		
5. SA Validity	Sets the expiration of SA validity when the IPsec connection has been established (3600-86400 seconds).	86400		
5. Child SA Validity	Sets the expiration of Child SA validity when the IPsec connection has been established (3600-86400 seconds).	28800	X	X
6. NAT Traversal	Sets IPsec NAT traversal related data.	-	X	X
1. Keep Alive	Sets keepalive for NAT traversal (0, 1-3600 seconds). NOTE 21	20		
2. NAT Traversal Port	Sets port for NAT traversal (1024-65535).	4500		

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
7. L2TP Settings	Sets L2TP connection related data.	-		
1. Cert ID	Sets certificate ID for L2TP (up to 127 characters).	-	X	X
2. Cert Password	Sets certificate password for L2TP (up to 127 characters).	-		
6. USB	Sets USB related parameter.	-	X	X
1. Power	Sets whether to supply power to USB devices.			
1. Disable	Power is not supplied to USB devices.	Enable	X	X
2. Enable	Power is supplied to USB devices.			
7. User Password	-	-	X	X
1. Expiration Date	Sets the expiration of user password (0-90 days). * When "0" is specified, the warning indication will not be indicated.	0	X	X
2. Warning Interval	Sets the interval to regenerate the warning message which prompts a password resetting (1-24 hours).	1	X	X
8. Certificate [FW5.0.0.0 or later]	Sets certificate related data.	-		
1. Install	Sets install related data of certificate.	-	X	X
1. PKCS#12 Password	Sets PKCS#12 password.			
2. Select	Selects the client certificate.			
1. 802.1X	802.1X is selected as a client certificate. NOTE 16	-	X	X
2. VPN NOTE 15	VPN connection is selected as a client certificate. NOTE 16			
3. Delete	Sets delete method of certificate.	-		
1. Delete Data	Selects the type of certificate data to be deleted.			
1. Client Certificate	Deletes client certificate data.		X	X
2. Root Certificate	Deletes root certificate data.	-		
3. Expired Certificates	Deletes expired certificate data.			
4. All Certificates	Deletes all certificates data.			
2. Auto Delete Mode	Sets whether to delete the certificate data automatically.			
1. Disable	Certificate data is not deleted automatically.	Disable	X	X
2. Enable	Certificate data is deleted automatically.			

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. Validity	Sets the expiration of certificate validity.	-	X	X
1. Warning Period	Sets the time to display the warning notice for the end of certificate validity (0-90 days). NOTE 22	7	X	X
2. Installation time	Sets the time of when the certificate is installed. * This setting will be applied to authenticate a certificate until the time information from the SV9300 is received.	-		
1. Year	Year (2016-2037)	2016	X	X
2. Month	Month (1-12)	1		
3. Day	Day (1-31)	1		
4. Hour	Hour (0-23)	0		
5. Minute	Minute (0-59)	0		
6. Second	Second (0-59)	0		
9. HTTPS Settings [FW5.1.0.0 or later]	Setting of HTTPS communication.	-	X	X
1. Server Authentication	Setting of a server certificate authentication.	Disable	X	X
1. Disable	Sets the server authentication disabled.			
2. Enable	Sets the server authentication enabled.			
5. Terminal Information	Displays terminal information.	-	X	X
Hardware Version	Displays hardware version of DT800 Series.	-	X	X
MAC Address	Displays MAC address of DT800 Series.	-	X	X
Firmware Version	Displays firmware version of DT800 Series.	-	X	X
6. Application Settings	Sets application data.	-	X	X
1. XML Settings	Specifies XML browser function.	-	X	X
1. Terminal ID	Login number for accessing XML server.	0000	X	X
2. XML Browser	Sets XML browser related data.	-	X	X
1. Home URL NOTE 23	Sets the address of the contents to access when starting up XML browser.	-	X	X
2. Proxy Settings	Sets proxy server related data.	-	X	X
1. Proxy IP	Sets proxy IP address of XML browser.	0.0.0.0	X	X
2. Proxy Port	Sets proxy port of XML browser [1024-65535].	8080	X	X
3. Directory URL	Sets URL of Corporate Phonebook.	-	X	X
4. Service URL NOTE 24	Sets URL of XML application.	-	X	X
1. 1st Service URL	URL of up to four XML applications can be specified.			
2. 2nd Service URL				
3. 3rd Service URL				
4. 4th Service URL				
2. IPM Self Port	Sets the port for connecting to IP Phone Manager [1024-65535]. NOTE 1	3530	X	X
3. Popup	Sets the popup window display.	-	X	X
1. Popup Mode	Sets the mode of the popup window display.	App Priority	X	X
1. Enable	Sets the popup window display enabled.			
2. App Priority	Sets to depend on a definition of XML tag for Popup window.			

0. Config (Continued)

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. Screen Saver URL	Sets reference URL for Screen Saver activation.	-	X	X
5. Emergency Call	Sets the destination number of emergency call which a terminal user can originate even while the Security Mode is in service.	-	X	X
Emergency Dial 1	Up to three destination numbers can be specified.	-	X	X
Emergency Dial 2				
Emergency Dial 3				
6. Menu Key Mode	Changes to menu key mode.	Enable	X	X
1. Disable	Sets the menu key mode disabled.			
2. Enable	Sets the menu key mode enabled.			
7. Voice Rec Settings (Setting prohibited)	-	-	-	-

NOTE 1: A unique number must be assigned to each of the following ports. For RTP Self Port, 16 ports starting from an assigned port are used.

- RTP Self Port
- SIP Self Port
- Push Server Port
- IPM Self Port

NOTE 2: $LLDP\ time\ to\ live = LLDP\ packet\ transmission\ interval \times the\ set\ value$ (65535 seconds at most)

NOTE 3: "SIP server" used here refers to the SV9300. The SV9300 acts as SIP server where DT830G Series terminal is registered.

NOTE 4: Follow the IP address setting (CM0B Y=1XX>00) of VOIP port on SV9300.

NOTE 5: Follow the SIP Server port setting (CM0B Y=1XX>32) on SV9300 (example: change 5060 to 5080 when default is set).

NOTE 6: If the firmware version of a terminal is FW5.1.0.0 or later, "2. Keeping Call" menu is shown as follows.

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
2. Keeping Call (Setting prohibited)	-	-	-	-
1. Keeping Call Mode (Setting prohibited)	-	Enable	-	-
1. Disable (Setting prohibited)	-			
2. Enable	-			
2. No RTP Time for H.C. (Setting prohibited)	-	0	-	-

NOTE 7: Howler Tone to be sent depends on the system data setting (CM08>153, CM13 Y=04) of SV9300 (default: To send).

NOTE 8: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “4. Account Settings” (formerly “4. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
4. Account Settings	Account settings of download server (FTP/HTTPS).			
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).			
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).			

NOTE 9: Do not set the Key Kit 9, 10 because these Kits are not supported.

NOTE 10: The following table shows the usable headset type depending on the setting of “EHS Cable”.

Setting of EHS Cable	Usable Headsets (Cables)
None	HW510 (A10) HW251 (A10)
APN-91	WT100 (APN-91) Voyager Legend CS (APN-91)

NOTE 11: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “5. Account Settings” (formerly “5. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
5. Account Settings	Account settings of backup/restore server (FTP/HTTPS).			
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).			
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).			

NOTE 12: This service is available to perform various tests such as Key Test, LED, LCD, and Call Test.

NOTE 13: Be careful when you specify “3. Enable (All Client)” because security services are disabled for Push when “3. Enable (All Client)” is set.

NOTE 14: To use Photo Frame Application for DT830G (12/24CG), configure one of the four Push servers (1st to 4th Push servers) with the IP address for Photo Frame.

NOTE 15: For North America and China, this setting is not available. Therefore, this setting cannot be selected on the terminal.

NOTE 16: Up to 15 certificates can be selected.

NOTE 17: The following table shows the supporting authentication mode and IKE version.

X: Available N: Not available

Authentication Mode	IKE Version	
	IKEv1	IKEv2
PSK	X	X
RSA	N	X

NOTE 18: For IKE version “V1”, the supported authentication mode is “PSK” only. Therefore, when “1. V1” is specified, be sure to specify “1. PSK” on “2. Authentication Mode”. When “2. RSA” is specified, a communication error occurs.

NOTE 19: IKE authentication is executed with “PSK” which has been assigned in [0. Config] - [4. Security] - [5. VPN Client] - [4. IKE Settings] - [3. PSK Setting].

NOTE 20: IKE authentication is executed with “RSA” which has been assigned in [0. Config] - [4. Security] - [8. Certificate] - [2. Select] - [2. VPN].

NOTE 21: When “0” is specified, this setting will be disabled.

NOTE 22: When “0” is specified, the warning indication will not be indicated.

NOTE 23: When you register the Home URL, the indication on the Menu screen turns “Application Name” from “Phone”. Set the Lock Mode to Disable and Disable the Security key. To connect a phone and IP Phone Manager, the port number to be used needs to match. Set the port number on the phone side from [0. Config] - [6. Application Setting] - [2. IPM Self Port].

NOTE 24: When you select the parameter of Service URL, a Menu for 1st - 4th Service URL is displayed. URL of up to four XML applications can be registered.

NOTE 25: If the firmware version of a terminal is FW5.2.0.0 or later, this menu becomes “5. VPN Settings” (formerly “5. VPN Client”). The setting details are as follows.

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
5. VPN Settings [FW5.2.0.0 or later]	Sets VPN.	-	-	-
1. VPN Mode	Sets whether to enable VPN Feature.	Disable	X	X
1. Disable	VPN Mode is disabled.			
2. Enable	VPN Mode is enabled.			
2. VPN Network Settings	Sets VPN network related data.	-	X	X
1. VPN Client Address	Sets the terminal IP address to be specified by VPN center site	0.0.0.0	X	X
2. VPN Gateway Address	Sets VPN Gateway address.	0.0.0.0	X	X
3. L2TP/IPsec Settings	Sets L2TP/IPsec related data.	-	X	X
1. IKE Settings	Sets IKE protocol related data.	-	X	X
1. IKE Version NOTE 17	Selects IKE protocol version.	V1	X	X
1. V1 NOTE 18	IKE protocol version is specified to V1.			
2. V2	IKE protocol version is specified to V2.			

Parameter	Description	Default	DT830G (12/24CG)	DT830G (12/24DG)
2. Authentication Mode	Selects IKE authentication mode.	PSK	X	X
1. PSK NOTE 19	Authentication mode is specified to PSK.			
2. RSA NOTE 20	Authentication mode is specified to RSA.			
3. PSK Setting	Sets the parameter of PSK (up to 64 characters).	-	X	X
4. IKE ID	Sets the parameter of IKE ID (up to 127 characters).	-	X	X
5. SA Validity	Sets the expiration of SA validity when the IPsec connection has been established (3600-86400 seconds).	86400	X	X
2. Child SA Validity	Sets the expiration of Child SA validity when the IPsec connection has been established (3600-86400 seconds).	28800	X	X
3. NAT Traversal	Sets IPsec NAT traversal related data.	-	X	X
1. Keep Alive	Sets keep alive for NAT traversal (0, 1-3600 seconds). NOTE 21	20		
2. NAT Traversal Port	Sets port for NAT traversal (1024-65535).	4500		
4. Cert ID	Sets authentication ID for L2TP (up to 127 characters).	-	X	X
5. Cert Password	Sets authentication password for L2TP (up to 127 characters).	-	X	X

DT830 Series Configuration Menu List (FW 3.0.3.0 or later)

This section lists the configuration menu used for DT830 Series configuration settings. DT830 Series column on the tables below show whether the menu is available or not for each terminal type.

LEGEND	X: Available	N: Not available	-: Not applicable
---------------	--------------	------------------	-------------------

NOTE 1: Up to four Service menus are displayed. Service menu is not displayed when a service URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL].

NOTE 2: Phone menu is displayed when a URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL]. Usually, "Application Name" is displayed when a URL is registered.

NOTE 3: The Config menu is not displayed on the menu window. To open the Config menu, press digit key "0". In addition, by pressing "Hold", "Transfer", "*", "#" key in this order in an idle state of telephone, you can access to the login screen of Configuration Menu.

1. History

Parameter	Description	Default	DT830 Series
1. History	Lists/deletes the history of outgoing/incoming call	-	X

When "1. History" is selected, the "History menu" of SV9300 is displayed.

[History]	
NEC ICHIRO	[300]
3:09 PM	TUE 18 OCT 2016
Outgoing	Incoming

NOTE: In SV9300, the built-in Call History of a telephone cannot be used. Call History on SV9300 side is used.

2. Directory

Parameter	Description	Default	DT830 Series
2. Directory	Displays Terminal Phonebook	-	X

When “2. Directory” is selected, the “Directory service” screen of SV9300 is displayed.

[Directory]
Name>>
Set Common Personal

NOTE: In SV9300, the built-in Directory of a telephone cannot be used. The Directory service on SV9300 side is used.

3. Service

Parameter	Description	Default	DT830 Series
3. Service	Starts up XML browser.	-	X
1. Service	Starts up XML browser.	-	X
2. Service		-	X
3. Service		-	X
4. Service		-	X

NOTE: At most four services are displayed on the menu. A service is not displayed when a service URL is not registered by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL].

4. Setting

Parameter	Description	Default	DT830 Series
1. User Setting	Sets user interface related.	-	X
1. Incoming Call	Sets ringing and lamp indication data for incoming call.	-	X
1. Ring Volume (Setting prohibited)	-	-	-
2. Offhook Ring (Setting prohibited)	-	Enable	-
1. Disable	-		
2. Enable	-		
3. Headset Ring	Selects whether to use headset ringing.	Disable	X
1. Disable	Headset ringing is OFF.		
2. Enable	Headset ringing is ON.		

4. Setting (Continued)

Parameter	Description	Default	DT830 Series
4. Ring Tone	Sets Ringing Tone for incoming call termination.	-	X
1. External Call NOTE 1	Sets Ringing Tone for external call termination.	Automatic	X
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.		
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.		
15. Download 1 ~ 17. Download 3 NOTE 2	Sets Download 1 ~ Download 3 for Ringing Tone.		
2. Internal Call NOTE 1	Sets Ringing Tone for station call termination.		
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.		
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.		
15. Download 1 ~ 17. Download 3 NOTE 2	Sets Download 1 ~ Download 3 for Ringing Tone.		
3. IM	Not used.	-	-
5. Illumination	Sets lamp indication pattern for incoming call termination.	-	X
1. External Call NOTE 3	Sets Illumination for External Call.	-	X
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.	Automatic	X
1. Disable	Lamp indication is OFF.		
2. Red	Specifies the color of lamp indication for incoming call termination.		
3. Green			
4. Blue			
5. Yellow			
6. Purple			
7. Light Blue			
8. White			
9. Rotation			

4. Setting (Continued)

Parameter	Description	Default	DT830 Series
2. Internal Call NOTE 3	Sets Illumination for Internal Call.	-	X
0. Automatic	Sets Automatic for the lamp indication of incoming call termination.	Automatic	X
1. Disable	Lamp indication is OFF.		
2. Red	Specifies the color of lamp indication for incoming call termination.		
3. Green			
4. Blue			
5. Yellow			
6. Purple			
7. Light Blue			
8. White			
9. Rotation			
3. IM (Setting prohibited)	-	-	-
6. Headset Ring Volume (Setting prohibited)	-	-	-
2. Talk	Sets call related data.	-	X
1. RTP Alarm NOTE 4	Sets whether to allow the alarm for detecting packet loss in receiving speech packets.	Automatic	X
0. Automatic	Applies RTP alarm setting on the SV9300.		
1. Disable	Alarm is OFF.		
2. Enable	Alarm is ON.		
2. DTMF Tone (Setting prohibited)	-	Automatic	X
0. Automatic	-		
1. Disable (Setting prohibited)	-		
2. Enable (Setting prohibited)	-		
3. Key Touch Tone	Sets touch tone for pressing digit key.	Automatic NOTE 5	X
0. Automatic	Sets Automatic to key touch tone.		
1. No Tone	Sets key touch tone disabled.		
2. Tone	Sets key touch tone enabled.		
3. Voice	Enables the key touch tone by voice recognition.		
4. Hold Music	Sets Music-on-Hold.	Default	X
1. Default	Applies the default music (Minuet).		
2. Download	Applies the downloaded music.		
5. Prefix (Setting prohibited)	-	-	X
1. Prefix 1 ~ 3. Prefix 3 (Setting prohibited)	-	-	-

4. Setting (Continued)

Parameter	Description	Default	DT830 Series
3. Display	Sets display related data.	-	-
1. Calendar Format	-	Automatic NOTE 6	-
1. Calendar Format (Setting prohibited)	-		
0. Automatic	-		
1. Type 1	-		
2. Type 2			
3. Type 3			
4. Type 4			
5. Type 5			
2. Time Format (Setting prohibited)	-	Automatic NOTE 7	-
0. Automatic	-		
1. 12 hour Indication (Setting prohibited)	-		
2. 24 hour Indication (Setting prohibited)	-		
2. Local Volume	Selects whether to display volume gage when adjusting the volume level.	Automatic NOTE 8	X
0. Automatic	Sets Automatic for volume gage display.		
1. Disable	Volume gage is not displayed.		
2. Enable	Volume gage is displayed.		
3. Screen Saver	-	-	-
1. Screen Saver Mode	Selects whether to use Screen Saver.	Disable	X
1. Disable	Screen Saver is not used.		
2. Enable	Screen Saver is used.		
2. Wait Time	Sets wait time until Screen Saver activation (1-999 min.).	120	X
4. Back Light	Switches ON/OFF the backlight on DT830.	Enable NOTE 9	X
1. Disable	Backlight is OFF.		
2. Enable	Backlight is ON.		
5. Font Size NOTE 10	Switches the font size on configuration menu.	Standard (16dot)	X
1. Standard (16 dot)	Selects Standard or Small of font size.		
2. Small (12 dot)			

4. Setting (Continued)

Parameter	Description	Default	DT830 Series
6. Language	Selects the language on date and time display.		
0. Automatic	Sets Automatic for language setting on date and time display.		
1. Japanese	Selected language is displayed on date and time display. * For FW4.0.0.0 or later, each parameter in this menu is displayed in their native language.	Automatic NOTE 11	X
2. NEC English			
3. NEC Portuguese			
4. NEC Spanish			
5. NEC French			
6. German			
7. Italian			
8. Dutch			
9. Norwegian			
10. Danish			
11. Swedish			
12. Greek			
13. English			
14. Portuguese			
15. Spanish			
16. French			
17. Russian			
18. Turkish			
19. Azərbaycanca [FW5.1.0.0 or later]			
20. Romania			
21. Polish			
22. Catalan			
23. Korean [FW4.0.0.0 or later]			
24. Chinese (Simplified Chinese) [FW4.0.0.0 or later]			
25. Chinese (Traditional Chinese) [FW4.0.0.0 or later]			
7. Advanced	Sets other data for display.	-	-
1. Backlight Fade Control	Sets whether Backlight Fade Control is in service, when turning on/off the backlight of the display.	Enable	X
1. Disable	Disables Backlight Fade Control.		
2. Enable	Enables Backlight Fade Control.		
4. Change Password NOTE 12	Assigns the password of the terminal, used for data clear, security, and login of user mode.	0000	X
5. Security	Setting for security.	-	-
1. VPN [FW5.2.0.0 or later]	Setting of VPN for security.	0.0.0.0	N
1. IP Address (PC)	Sets the IP address of PC connected to a PC port when VPN is enabled.		

4. Setting (Continued)

Parameter	Description	Default	DT830 Series
6. Peripherals	Setting for using peripheral equipment of terminal.	-	
1. USB Memory	Not used.	-	
2. BCA Option	Setting for BCA (Bluetooth cradle phone adapter)	-	
1. BCA Connection Settings	Setting for Bluetooth Connection.	-	
1. Auto Connection	Selects whether to use Auto Connection.	Enable	X
1. Disable	Auto Connection is not used.		
2. Enable	Auto Connection is used.		
2. Connection Cycle	Sets Bluetooth connection cycle.	10 Seconds	
1. 3 Seconds	Bluetooth connection cycle is 3 seconds.		
2. 10 Seconds	Bluetooth connection cycle is 10 seconds.		
3. 1 Minute	Bluetooth connection cycle is 1 minute.		
4. 10 Minutes	Bluetooth connection cycle is 10 minutes.		
2. Interrupt Ringing	Selects whether to use Interrupt Ringing.	Enable	
1. Disable	Interrupt Ringing is OFF.		
2. Enable	Interrupt Ringing is ON.		
3. Smart Device Type	Setting for Smart Device Type.	Type 1	
1. Type 1	Sets Smart Device Type for type 1.		
2. Type 2	Sets Smart Device Type for type 2.		
4. Auto Path Switch	Sets whether Auto Path Switch is in service.	Enable	
1. Disable	Disables Auto Path Switch.		
2. Enable	Enables Auto Path Switch.		
5. Illumination	Sets lamp indication pattern for incoming call termination.	Blue	
1. Disable	Specifies the color of lamp indication for incoming call termination.		
2. Red			
3. Green			
4. Blue			
5. Yellow			
6. Purple			
7. Light Blue			
8. White			
9. Rotation			
7. Usability	Sets the operation of Help key while pop-up window is displayed.	-	X
1. Help Key Mode	Setting for Help Key Mode (SubMenu/Popup Window)	SubMenu	X
1. SubMenu	Sets the Help key which displays sub menu		X
2. Popup Window	Sets the Help key which displays pop-up window		X
0. Setting Reset	Initializes all the assigned personal information on the terminal to factory default.	-	X

4. Setting (Continued)

Parameter	Description	Default	DT830 Series
2. Download	Sets download related data.	-	X
1. Download Files	Downloads files of Music-on-Hold, Ringing Tone, and Phonebook.	-	X
1. Hold Music	Downloads a sound file used for Music-on-Hold.	MOH.wav	X
2. Ring Tone	Downloads a sound file used for Ringing Tone.	-	X
1. Download 1	Downloads Melody 1 ~ Melody 3.	Melody1.wav	
2. Download 2		Melody2.wav	
3. Download 3		Melody3.wav	
3. Directory (Setting prohibited)	-	-	-
2. Download Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X
3. Protocol	Specifies the protocol for file downloading.	FTP	X
1. FTP	FTP is used.		
2. TFTP	TFTP is used.		
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.		
4. FTP Settings NOTE 13	Sets data for accessing FTP server.	-	X
1. User ID	Sets user ID for login to FTP server.		
2. Password	Sets password for login to FTP server.		
3. Folder	Specifies the directory to save the download file.		
3. Data Backup/Restore	Makes backup of terminal data or restores from backup file.	-	X
1. Data Backup	Saves the terminal setting to backup file. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	
2. Data Restore	Restores the terminal data by reading out the saved backup file. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X
3. Server Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X
4. Protocol	Specifies the protocol for file downloading.	FTP	X
1. FTP	FTP is used.		
2. TFTP	TFTP is used.		
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.		
5. FTP Settings NOTE 14	Sets data for accessing FTP server.	-	X
1. User ID	Sets user ID for login to FTP server.		
2. Password	Sets password for login to FTP server.		
3. Folder	Specifies the directory to save backup file.		

NOTE 1: To provide the Distinctive Ringing of external call/internal call by SV9300, be sure to use with default (Automatic). When “Automatic” is set, by the system data setting (CM08>390) of SV9300, Ringing Tone can be changed by the system data (CM15 Y=491, CM35 Y=034, 164) or terminal operations (“Feature” + “3”) (default: as per CM15 Y=491, CM35 Y=034, 164).

NOTE 2: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “15. Music Ring 1~17. Music Ring 3” (formerly “15. Download 1~17. Download 3”). The setting details are as follows.

Parameter	Description	Default	DT830 Series
15. Music Ring 1~17. Music Ring 3	Sets Music Ring 1 ~ Music Ring 3.	Automatic	X

NOTE 3: When “Automatic” is set, by the system data setting (CM08>144) of SV9300, the illumination color can be changed to Red or Green (default: Red).

Set the “0. Automatic” to use color coding by 7-color LED.

NOTE 4: When “Automatic” is set, RTP Alarm depends on the system data setting (CM08>1004) of SV9300 (default: To send).

NOTE 5: Follows the setting (CM15 Y=291) of SV9300 side when “Automatic” is set. In all other cases, follows the setting of a terminal side.

NOTE 6: Calendar Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.

NOTE 7: Time Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.

NOTE 8: A setting for Enable is applied when Automatic has been set.

NOTE 9: Back Light menu becomes effective when the system data (CM15 Y=485) of SV9300 is set to other than 0 (Always off).

Back Light of ten key cannot be used when terminals and option equipment for power failure are implemented.

NOTE 10: When a font size is changed, press the Cursor key and return to the previous menu, then the font size becomes effective.

NOTE 11: The setting of display language on the Clock Display screen depends on the system data settings (CM12 Y=63, CM04 Y=00) of SV9300.

NOTE 12: Change Password is used for password for User Mode of Web programming and security unlock. Be sure to change the default password.

NOTE 13: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “4. Account Settings” (formerly “4. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT830 Series
4. Account Settings	Account settings of download server (FTP/HTTPS).		
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).		
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).		

NOTE 14: *If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “5. Account Settings” (formerly “5. FTP Settings”). The setting details are as follows.*

Parameter	Description	Default	DT830 Series
5. Account Settings	Account settings of backup/restore server (FTP/HTTPS).		
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).		
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).		

5. Phone

Depending on the Home URL setting by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL], this menu is displayed as follows.

- When Home URL is not registered, home screen is displayed and the menu name is displayed as “Phone”.
- When Home URL is registered, XML application screen is displayed, and the menu name is displayed as the application name to be described in the XML application.

0. Config

Parameter	Description	Default	DT830 Series
1. Network Settings	Sets network related data.	-	X
1. DHCP Mode	Selects whether to use DHCP protocol.	Enable	X
1. Disable	DHCP is not used.		
2. Enable	DHCP is used.		
2. IP Address	Sets DT830 Series terminal's IP address.	0.0.0.0	X
3. Default Gateway	Sets default router's IP address.	0.0.0.0	X
4. Subnet Mask	Sets subnet mask.	0.0.0.0	X
5. DNS Address	Sets DNS server's IP address.	0.0.0.0	X
6. Advanced Settings	Sets network expanded data.	-	X
1. LAN Port Settings	Sets LAN port.	-	X
1. Speed&Duplex	Sets communication speed of Ether port.	Autonego	X
1. Autonego	Autonegotiation		
2. 100M Full	100Mbps in full-duplex mode		
3. 100M Half	100Mbps in half-duplex mode		
4. 10M Full	10Mbps in full-duplex mode		
5. 10M Half	10Mbps in half-duplex mode		
2. VLAN Mode	Selects whether to use VLAN on terminal.	Disable	X
1. Disable	VLAN is not used.		
2. Enable	VLAN is used.		
3. VLAN ID	Sets VLAN ID of terminal [1-4094].	7	X
4. VLAN Priority	Sets VLAN Priority of terminal [0-7].	5	X
5. Auto-MDIX	Selects whether to use Auto-MDIX.	Enable	X
1. Disable	Auto-MDIX is not used.		
2. Enable	Auto-MDIX is used.		

0. Config (Continued)

Parameter	Description	Default	DT830 Series
2. PC Port Settings	Sets PC port.	-	X
1. Speed&Duplex	Sets communication speed of PC port.	Autonego	X
1. Autonego	Autonegotiation		
2. 100M Full	100Mbps in full-duplex mode		
3. 100M Half	100Mbps in half-duplex mode		
4. 10M Full	10Mbps in full-duplex mode		
5. 10M Half	10Mbps in half-duplex mode		
2. VLAN Mode	Selects whether to use port VLAN of PC port.	Disable	X
1. Disable	Port VLAN of PC port is not used.		
2. Enable	Port VLAN of PC port is used.		
3. Port VLAN ID	Sets port VLAN ID of PC port [1-4094].	1	X
4. Port VLAN Priority	Sets port VLAN priority of PC port [0-7].	0	X
5. Port Available	Selects whether to use PC port.	Automatic	X
0. Automatic	Sets Automatic for PC port.		
1. Disable	PC port is not used.		
6. EAPOL	Select whether to forward EAPOL packet from PC port.	Disable	X
1. Disable	Not forward EAPOL packet from PC port.		
2. Enable	Forwards EAPOL packet from PC port.		
7. PC Port Security	Select whether to clear authentication status of the terminal which is connected to PC port for authentication switch when the link of PC port has been down.	Enable	X
1. Disable	Not clear the authentication status.		
2. Enable	Clear the authentication status.		
8. Auto-MDIX	Selects whether to use Auto-MDIX.	Enable	X
1. Disable	Auto-MDIX is not used.		
2. Enable	Auto-MDIX is used.		
3. CDP Mode (Setting prohibited)	-	-	-
4. Type of Service	Sets ToS value.	-	X
1. RTP	Sets ToS value for RTP in Hex.	A0	
2. SIP	Sets ToS value for SIP in Hex.	C0	
5. Self Port Settings	Sets each port for DT830 Series.	-	-
1. RTP Self Port	Sets RTP port for DT830 Series [1024-65520 (except 59986-60000)]. NOTE 1 * Odd port numbers cannot be set.	3462	X
2. SIP Self Port	Sets SIP port for DT830 Series [1024-65534]. NOTE 1	5060	X
6. LLDP Settings	Sets information related to LLDP.	-	X
1. LLDP Mode	Sets whether to transmit/receive LLDP packets.	Enable	X
1. Disable	DT830 Series does not transmit/receive LLDP packets.		
2. Enable	DT830 Series transmits/receives LLDP packets.		

0. Config (Continued)

Parameter	Description	Default	DT830 Series
2. Auto Setting Mode	Sets whether to perform data setting automatically when LLDP packets are received.		
1. Disable	DT830 Series does not perform data setting automatically when LLDP packets are received.	Enable	X
2. Enable	DT830 Series performs data setting automatically when LLDP packets are received.		
3. RX Waiting Time	For Auto Setting, sets the waiting time until DT830 Series receives LLDP packets (1-60 seconds).	15	X
4. Transmit Interval	Sets intervals to transmit LLDP packets (5-32768 seconds).	30	X
5. Hold Multiplier NOTE 2	Sets a multiplier determining the time to live for the LLDP packets to be transmitted (2-10 multiples).	4	X
6. Fast Start Count	Sets the number of times to transmit LLDP packets at one-second intervals after DT830 Series starts up (1-10 times).	3	X
7. Transparent Mode	Sets whether Transparent Mode of LLDP packets is enabled between the PC port and the LAN port.		
1. Disable	Transparent Mode of LLDP packets is disabled.	Enable	X
2. Enable	Transparent Mode of LLDP packets is enabled.		
8. Asset ID	Sets the Asset ID for LLDP packets to be transmitted. * “-” is displayed in default settings.	-	X

0. Config (Continued)

Parameter	Description	Default	DT830 Series
7. Spare IP Settings (Setting prohibited)	-	-	X
1. Spare/Backup IP Mode (Setting prohibited)	-	Disable	-
1. Disable	-		
2. Spare IP (Setting prohibited)	-		
3. Backup IP (Setting prohibited)	-		
2. Network Settings (Setting prohibited)	-	-	-
1. IP Address (Setting prohibited)	-	0.0.0.0	
2. Default Gateway (Setting prohibited)	-	0.0.0.0	
3. Subnet Mask (Setting prohibited)	-	0.0.0.0	
4. DNS Address (Setting prohibited)	-	0.0.0.0	
3. SIP Settings (Setting prohibited)	-	-	-
1. 2nd Server Address (Setting prohibited)	-	0.0.0.0	-
2. 3rd Server Address (Setting prohibited)	-	0.0.0.0	-
3. 4th Server Address (Setting prohibited)	-	0.0.0.0	-
4. 2nd Server Port (Setting prohibited)	-	0	-
5. 3rd Server Port (Setting prohibited)	-	0	-
6. 4th Server Port (Setting prohibited)	-	0	-
8. TTL Settings	Setting for multicast TTL (Time To Live)	32	X
1. RTP (Multicast)	Sets the value of multicast TTL (0-255)		X
9. Priority Queue	Sets whether to enable Priority Queue of the internal switching HUB.	Disable	X
1. Disable	Priority Queue is disabled.		X
2. Enable	Priority Queue is enabled.		X
2. SIP Settings	Sets SIP related data.	-	X
1. SIP User	-	-	-
1. User ID (Setting prohibited)	-	-	-
2. Password (Setting prohibited)	-	-	-
3. Extension Number (Setting prohibited)	-	-	-
4. Backup Logon (Setting prohibited)	-	Disable	-

0. Config (Continued)

Parameter	Description	Default	DT830 Series
2. Server Address & URI	Sets IP address or URI of SIP server. NOTE 3	-	X
1. 1st Server Address	Sets IP address or fully qualified domain name (FQDN) of 1st ~ 4th SIP server. NOTE 4	0.0.0.0	X
2. 2nd Server Address			
3. 3rd Server Address			
4. 4th Server Address			
5. Registrar Destination (Setting prohibited)	-	-	-
1. 1st Register Address (Setting prohibited)	-	0.0.0.0	-
2. 2nd Register Address (Setting prohibited)			
3. 3rd Register Address (Setting prohibited)			
4. 4th Register Address (Setting prohibited)			
3. Access Mode	Sets access mode for SIP server.	Normal	X
1. Normal	Normal access		
2. Remote	Remote access		
4. SIP Server Port	Sets SIP server port [1024-65535].	5060	X
1. 1st Server Port	Specifies the destination port number of sending packet to 1st ~ 4th SIP server. NOTE 5		
2. 2nd Server Port			
3. 3rd Server Port			
4. 4th Server Port			
5. Registrar Destination (Setting prohibited)	-		-
1. 1st Register Port (Setting prohibited)			
2. 2nd Register Port (Setting prohibited)			
3. 3rd Register Port (Setting prohibited)			
4. 4th Register Port (Setting prohibited)	-		

0. Config (Continued)

Parameter	Description	Default	DT830 Series
5. Area ID	Sets the area (country) using DT830 Series.		
0. Automatic	The SV9300 side setting is applied when Automatic is selected.		
1. Japan	Selects the area (country) NOTE: Area IDs allowed on the SV9300 can be selected.	Automatic	X
2. America (Canada and North America)			
3. Australia			
4. Others			
5. Hong Kong			
6. Malaysia			
7. Singapore			
8. Mexico			
9. Taiwan			
10. New Zealand			
11. Korea			
12. Brazil			
13. China			
14. Thailand			
15. Thailand (EGAT)			
16. Sri Lanka			
17. Germany			
18. Italy			
19. Netherlands			
20. Catalonia			
21. Denmark			
22. Sweden			
23. Poland			
24. Czech			
25. Norway			
26. France			
27. Portugal			
28. Spain			
29. Austria			
30. Belgium			
31. UK			
32. Greece			
33. Switzerland			
34. South Africa			
35. Brazil (EMEA)			

0. Config (Continued)

Parameter	Description	Default	DT830 Series
6. Extra Function (Setting prohibited)	-	-	-
1. Retransmit Interval (Setting prohibited)	-	Standard	-
1. Standard	-		
2. 6 Seconds (Setting prohibited)	-		
2. Keeping Call (Setting prohibited) NOTE 6	-	-	-
1. Disable (Setting prohibited)	-	Enable	-
2. Enable	-		
3. Howler Tone (Setting prohibited) NOTE 7	-	Disable	-
1. Disable	-		
2. Enable (Setting prohibited)	-		
4. Random Wait Time [FW5.0.0.0 or later]	Sets random wait time of when the connection between SV9300 and terminal starts (5-3600 seconds). * Do not change the factory default value of this parameter.	5	X
5. Connect Wait Time [FW5.0.0.0 or later]	Sets the time to wait until the connection between SV9300 and terminal is complete (30-300 seconds). * Do not change the factory default value of this parameter.	30	X
7. Encryption	Sets encryption data.	-	X
1. Authentication Mode	Selects whether to use authentication.	Disable	X
1. Disable	Authentication is not used.		
2. Enable	Authentication is used.		
2. One Time Password	Sets One Time Password.	-	X
3. Authentication Status	Displays authentication status.	-	X
8. NAT Traversal (Setting prohibited)	-	-	-
1. NAT Traverse Mode (Setting prohibited)	-	Disable	-
1. Disable (Setting prohibited)			
2. Dynamic (Setting prohibited)			
3. Static (Setting prohibited)			
2. Network Area Name (Setting prohibited)	-	-	-

0. Config (Continued)

Parameter	Description	Default	DT830 Series
3. WAN Settings (Setting prohibited)	-	-	-
1. WAN Mate IP Address (Setting prohibited)		0.0.0.0	
2. WAN SIP Mate Port (Setting prohibited)		5060	
3. WAN Self IP Address (Setting prohibited)		0.0.0.0	
3. Maintenance Settings	Sets maintenance related data.	-	X
1. Download Menu	Downloads specified file to DT830 Series.	-	X
1. Download Files	Specifies download file name for each file type.	-	X
1. Boot & Program	Downloads boot and program.	itzisipvg.tgz	X
2. Config	Downloads configuration file.	config.tgz	X
3. Voice File	Downloads voice file.	voice.tgz	X
4. Language Pack	Downloads translation data file.	LangPack.tgz	X
5. Certificate [FW5.0.0.0 or later]	Downloads certificate files.	-	X
1. Client Certificate	Downloads client certificate file.	clientcert.p12	
2. Root Certificate	Downloads root certificate file.	rootcert.der	
2. Download Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X
3. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X
1. FTP	FTP is used.		
2. TFTP	TFTP is used.		
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.		
4. FTP Settings NOTE 8	Sets data for accessing FTP server.	-	X
1. User ID	Sets user ID for login to FTP server.	-	X
2. Password	Sets password for login to FTP server.	-	X
3. Folder	Specifies the directory to save the download file.	-	X
2. Hard Reset	Resets DT830 Series without data saving.	-	X
3. Data Clear	Initializes DT830 Series selecting the initialization mode.	-	X
1. Factory Value	Initializes DT830 Series to factory default.	-	X
2. Terminal Data	Clears terminal data on DT830 Series.	-	X
3. Personal Data	Clears personal data on DT830 Series.	-	X

0. Config (Continued)

Parameter	Description	Default	DT830 Series
4. Adjust	Adjusts LCD, audio, and button kit.	-	X
1. Default Audio	Sets audio data.	-	X
1. Sidetone Volume	Sets Side Tone volume [0-7].	4	X
2. Send Gain	Sets sending gain of handset [0-31].	24	X
3. Receive Bias	Sets receiving bias of handset [0-7].	0	X
4. Receive Gain	Sets receiving gain of handset [0-31].	22	X
5. Talk Characteristics	Sets talk characteristics of handset.	-	X
1.Noise Suppression	Sets whether to enable the Noise Suppressor. (Set "1. Disable" in normal operation.)	Disable	X
1. Disable	Disables the Noise Suppressor.		
2. Enable	Enables the Noise Suppressor.		
2. Handsfree Mode	Sets a send gain mode of handsfree call.	Retro mode	
1. Retro Mode	Sets to Retro mode.		
2. New Mode	Sets to New mode.		
6. DTMF Send Level	Sets DTMF Send Level [0-48]. (level 0 (maximum) ~ level 48 (minimum))	32	
2. Key Kit Type	Sets additional button kit data.	-	
1 ~ 5. Kit 1, 2, 8, 9, 10 NOTE 9	key kit 1, 2, 8, 9, 10 on button kit are used.	Kit 1	X
1. Line Key Type	(Do not change the factory default value of this parameter.)	-	X
1. Line 12	Sets to 12-line of DT830.		X
2~7. Line 24, 32, 8, 6, 2, 0	Not used.		-
3. LCD Contrast	Adjusts LCD Contrast.	Area Contrast (Narrow)	X
0. Area Contrast			
1. Wide 2. Narrow			
4. BCA Audio	Setting for the acoustic type and audio of BCA.	-	X
1. Acoustic Type	Selects acoustic type.	Type 1	X
1. Type 1 - 5. Type 5			
2. Handset TX Gain	Sets sending gain of handset [0dB~30dB].	15	X
3. Handset RX Gain	Sets receiving gain of handset [0dB~36dB].	18	X
4. HF TX Gain	Sets sending gain of hands free [0dB~30dB].	15	X
5. HF RX Gain	Sets receiving gain of hands free [0dB~36dB].	18	X
6. Noise Suppression	Setting for the Noise Suppressor.	-	X
1. TX	Sets the Noise Suppression level for TX.	Disable	X
2. RX	Sets the Noise Suppression level for RX.	Disable	X
5. ADA Audio [FW4.0.0.0 or later]	-	Disable	-
1. Echo Canceller (Setting prohibited)			

0. Config (Continued)

Parameter	Description	Default	DT830 Series
6. EHS Cable NOTE 10 [FW5.0.0.0 or later]	Selects whether to use the cable for wireless headset connection.		
1. None	Wireless headset is not used (instead of it, None wired headset is used).	None	X
2. APD-80	Wireless headset with APD-80 is used.		
3. APN-91	Wireless headset with APN-91 is used.		
7. Character Code (Setting prohibited) [FW5.1.0.0 or later]	-	Automatic	-
1. Automatic	-		-
2. Eastern Europe (Setting prohibited)	-		-
8. Headset	Adjust headset		
1. Send Level	Sets voice sending level of headset.		
1. 0 dB	Not change the sending voice level from default.	0 dB	X
2. -5 dB	Set -5dB lower voice level than default.		
3. -10 dB	Set -10dB lower voice level than default.		
5. Auto config	Selects whether to use auto-config function.	Enable	X
1. Disable	Auto config is not used.		
2. Enable	Auto config is used.		
6. Data Maintenance	Sets data maintenance related.	-	X
1. Backup	Executes data backup.	-	X
1. All Data	Executes backup for all the configuration setting and data. * For encryption, "AllData.tgz.ef" is applied as a file name.	AllData.tgz	X
2. Terminal Data	Executes backup for terminal setting and data. * For encryption, "TerminalData.tgz.ef" is applied as a file name.	Terminal Data.tgz	X
3. Personal Data	Executes backup for personal setting and data. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X
2. Restore	Executes restoration.	-	X
1. All Data	Restores all the configuration setting and data. * For encryption, "AllData.tgz.ef" is applied as a file name.	AllData.tgz	X
2. Terminal Data	Restores terminal setting and data. * For encryption, "TerminalData.tgz.ef" is applied as a file name.	Terminal Data.tgz	X
3. Personal Data	Restores personal setting and data. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X

0. Config (Continued)

Parameter	Description	Default	DT830 Series
3. Server Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X
4. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X
1. FTP	FTP is used.		
2. TFTP	TFTP is used.		
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.		
5. FTP Settings NOTE 11	Sets data for accessing FTP server.	-	X
1. User ID	Sets user ID for login to FTP server.	-	X
2. Password	Sets password for login to FTP server.	-	X
3. Folder	Specifies the directory to save the download file.	-	X
7. Simple Test NOTE 12	Changes the terminal to test mode.	-	X
1. Call Test	Call Test	-	X
2. Call Test Address	Sets an IP address of the opposite terminal to perform the call test.	10.136.20.188	X
3. Key Test 1 (Random)	Tests to press the random button.	-	X
4. Key Test 2 (Sequential)	Tests to press the sequential button.	-	X
5. Loop Back Test	Tests the voice loop back.	-	X
1. Handset - Hand Speaker	Tests to speak and receive on the hand speaker.	-	X
2. Handset - Speaker	Tests to speak on the handset and receive on the speaker.	-	X
3. Headset - Head Speaker	Tests to speak and receive on the head speaker.	-	X
4. Headset - Speaker	Tests to speak on the headset and receive on the speaker.	-	X
5. MIC - Hand Speaker	Tests to speak on the MIC and receive on the hand speaker.	-	X
6. MIC - Head Speaker	Tests to speak on the MIC and receive on the head speaker.	-	X
7. MIC - Speaker	Tests to speak on the MIC and receive on the speaker.	-	X
6. Speaker Test	Tests the speaker.	-	X
1. Japan	A DT for Japan is heard in maximum volume.	-	X
2. America	A DT for North America is heard in maximum volume.	-	X
7. Receiver Test	Receiver Test for a handset	-	X

0. Config (Continued)

Parameter	Description	Default	DT830 Series
4. Security	Sets security related data.	-	X
1. Web Programming	Selects whether to use Web Programming.	Enable	X
1. Disable	Web Programming is disabled.		
2. Enable	Web Programming is enabled.		
2. Push Server	Sets XML's Push server related data.	-	X
1. Push Server Access	Sets whether to access XML's Push server.	Enable	X
1. Disable	Access is disabled.		
2. Enable	Access is enabled.		
3. Enable (All Client) NOTE 13	Access is enabled. Select this item to set more than one XML's Push server.		
2. Client IP Address	IP address of XML's Push server.	0.0.0.0	X
1. Client 1	Sets IP address of 1st XML's Push server.		
2. Client 2	Sets IP address of 2nd XML's Push server.		
3. Client 3	Sets IP address of 3rd XML's Push server.		
4. Client 4	Sets IP address of 4th XML's Push server.		
3. Push Server Port	Port number of XML's Push server [82, 1024-65535]. NOTE 1	82	X
3. Admin Password	Sets administrator's password. * Be sure to change the default password.	6633222	X
4. 802.1X Supplicant	Sets 802.1X authentication related parameter.	-	X
1. Supplicant	Sets whether to use supplicant on DT830 Series.	Disable	X
1. Disable	Supplicant is disabled on DT830 Series.		
2. Enable	Supplicant is enabled on DT830 Series.		
2. EAP Method [FW5.0.0.0 or later]	Selects EAP-MD5 or EAP-TLS for the EAP Method.	EAP-MD5	X
1. EAP-MD5	EAP-MD5 is used.		
2. EAP-TLS	EAP-TLS is used.		
3. Account	Sets user ID for authentication.	-	X
4. Password	Sets password for authentication.	-	X
5. EAPOL VLAN Mode	Selects whether to add VLAN tag to EAP packet.	Enable	X
1. Disable	VLAN tag is not added to EAP packet.		
2. Enable	VLAN tag is added to EAP packet.		

0. Config (Continued)

Parameter	Description	Default	DT830 Series
5. VPN Client NOTE 14 [FW5.0.0.0 to FW5.1.x.x] For FW5.2.0.0 or later, see NOTE 24.	Sets VPN Client related parameter.	-	-
1. VPN Mode	Sets whether to enable VPN Mode.	Disable	X
1. Disable	VPN Mode is disabled.		
2. Enable	VPN Mode is enabled.		
2. VPN Gateway Address	Sets VPN Gateway address.	0.0.0.0	X
3. VPN Network Settings	Sets VPN network related data.	-	X
1. VPN IP Address	Sets VPN IP Address.	0.0.0.0	
4. IKE Settings	Sets IKE protocol related data.	-	X
1. IKE Version NOTE 17	Selects IKE protocol version.	V1	
1. V1 NOTE 18	IKE protocol version is specified to V1.		
2. V2	IKE protocol version is specified to V2.		
2. Authentication Mode	Selects authentication mode.	PSK	
1. PSK NOTE 19	Authentication mode is specified to PSK.		
2. RSA NOTE 20	Authentication mode is specified to RSA.		
3. PSK Setting	Sets the parameter of PSK (up to 64 characters).	-	
4. IKE ID	Sets the parameter of IKE ID (up to 127 characters).	-	
5. SA Validity	Sets the expiration of SA validity when the IPsec connection has been established (3600-86400 seconds).	86400	
5. Child SA Validity	Sets the expiration of Child SA validity when the IPsec connection has been established (3600-86400 seconds).	28800	X
6. NAT Traversal	Sets IPsec NAT traversal related data.	-	X
1. Keep Alive	Sets keepalive for NAT traversal (0, 1-3600 seconds). NOTE 21	20	
2. NAT Traversal Port	Sets port for NAT traversal (1024-65535).	4500	
7. L2TP Settings	Sets L2TP connection related data.	-	X
1. Cert ID	Sets certificate ID for L2TP (up to 127 characters).	-	
2. Cert Password	Sets certificate password for L2TP (up to 127 characters).	-	
6. USB (Setting prohibited)	-	Enable	-
7. User Password	Setting for User Password.	-	X
1. Expiration Date	Sets the expiration of user password (0-90 days). * When "0" is specified, the warning indication will not be indicated.	0	X
2. Warning Interval	Sets the interval to regenerate the warning message which prompts a password resetting (1-24 hours).	1	X

0. Config (Continued)

Parameter	Description	Default	DT830 Series
8. Certificate [FW5.0.0.0 or later]	Sets certificate related data.	-	X
1. Install	Sets install related data of certificate.	-	
1. PKCS#12 Password	Sets PKCS#12 password.	-	X
2. Select	Selects the client certificate.	-	
1. 802.1X	802.1X is selected as a client certificate. NOTE 15	-	
2. VPN NOTE 14	VPN connection is selected as a client certificate. NOTE 15	-	X
3. Delete	Sets delete method of certificate.	-	
1. Delete Data	Selects the type of certificate data to be deleted.	-	
1. Client Certificate	Deletes client certificate data.	-	
2. Root Certificate	Deletes root certificate data.	-	X
3. Expired Certificates	Deletes expired certificate data.	-	
4. All Certificates	Deletes all certificates data.	-	
2. Auto Delete Mode	Sets whether to delete the certificate data automatically.	Disable	X
1. Disable	Certificate data is not deleted automatically.	-	
2. Enable	Certificate data is deleted automatically.	-	
4. Validity	Sets the expiration of certificate validity.	-	X
1. Warning Period	Sets the time to display the warning notice for the end of certificate validity (0-90 days). NOTE 16	7	
2. Installation time	Sets the time of when the certificate is installed. * This setting will be applied to authenticate a certificate until the time information from the SV9300 is received.	-	X
1. Year	Year (2016-2037)	2016	
2. Month	Month (1-12)	1	
3. Day	Day (1-31)	1	
4. Hour	Hour (0-23)	0	
5. Minute	Minute (0-59)	0	
6. Second	Second (0-59)	0	
9. HTTPS Settings [FW5.1.0.0 or later]	Setting of HTTPS communication.	-	X
1. Server Authentication	Setting of a server certificate authentication.	Disable	X
1. Disable	Sets the server authentication disabled.		
2. Enable	Sets the server authentication enabled.		
5. Terminal Information	Displays terminal information.	-	X
Hardware Version	Displays hardware version of DT830 Series.	-	X
MAC Address	Displays MAC address of DT830 Series.	-	X
Firmware Version	Displays firmware version of DT830 Series.	-	X

0. Config (Continued)

Parameter	Description	Default	DT830 Series
6. Application Settings	Sets application data.	-	X
1. XML Settings	Specifies XML browser function.	-	X
1. Terminal ID	Login number for accessing XML server.	0000	X
2. XML Browser	Sets XML browser related data.	-	X
1. Home URL NOTE 22	Sets the address of the contents to access when starting up XML browser.	-	X
2. Proxy Settings	Sets proxy server related data.	-	X
1. Proxy IP	Sets proxy IP address of XML browser.	0.0.0.0	X
2. Proxy Port	Sets proxy port of XML browser [1024-65535].	8080	X
3. Directory URL	Sets URL of Corporate Phonebook.	-	X
4. Service URL NOTE 23	Sets URL of XML application.		
1. 1st Service URL	URL of up to four XML applications can be specified.	-	X
2. 2nd Service URL			
3. 3rd Service URL			
4. 4th Service URL			
2. IPM Self Port	Sets the port for connecting to IP Phone Manager [1024-65535 (except 17185, 60000)]. NOTE 1	3530	X
3. Popup	Sets the popup window display.	-	X
1. Popup Mode	Sets the mode of the popup window display.	App Priority	X
1. Enable	Sets the popup window display enabled.		
2. App Priority	Sets to depend on a definition of XML tag for Popup window.		
4. Screen Saver URL	Sets reference URL for Screen Saver activation.	-	X
5. Emergency Call	Sets the destination number of emergency call which a terminal user can originate even while the Security Mode is in service.	-	X
Emergency Dial 1	Up to three destination numbers can be specified.	-	X
Emergency Dial 2			
Emergency Dial 3			
6. Menu Key Mode	Changes to menu key mode.	Enable	X
1. Disable	Sets the menu key mode disabled.		
2. Enable	Sets the menu key mode enabled.		

NOTE 1: A unique number must be assigned to each of the following ports. For RTP Self Port, 16 ports starting from an assigned port are used.

- RTP Self Port
- SIP Self Port
- Push Server Port
- IPM Self Port

NOTE 2: LLDP time to live = LLDP packet transmission interval × the set value (65535 seconds at most)

NOTE 3: “SIP server” used here refers to the SV9300. The SV9300 acts as SIP server where DT830G Series terminal is registered.

NOTE 4: Follow the IP address setting (CM0B Y=1XX>00) of VOIP port on SV9300.

NOTE 5: Follow the SIP Server port setting (CM0B Y=1XX>32) on SV9300 (example: change 5060 to 5080 when default is set).

NOTE 6: If the firmware version of a terminal is FW5.1.0.0 or later, “2. Keeping Call” menu is shown as follows.

Parameter	Description	Default	DT830 Series
2. Keeping Call (Setting prohibited)	-	-	-
1. Keeping Call Mode (Setting prohibited)	-	Enable	-
1. Disable (Setting prohibited)	-		
2. Enable	-		
2. No RTP Time for H.C. (Setting prohibited)	-	0	-

NOTE 7: Howler Tone to be sent depends on the system data setting (CM08>153, CM13 Y=04) of SV9300 (default: To send).

NOTE 8: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “4. Account Settings” (formerly “4. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT830 Series
4. Account Settings	Account Settings of download server (FTP/HTTPS).	-	X
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).		
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).		
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).		

NOTE 9: Do not set the Key Kit 8, 9, 10 because these Kits are not supported.

NOTE 10: The following table shows the usable headset type depending on the setting of “EHS Cable”.

Setting of EHS Cable	Usable Headsets (Cables)
None	HW510 (A10) HW251 (A10)
APD-80	W710 (APD-80 cable (Savi 710))
APN-91	WT100 (APN-91) Voyager Legend CS (APN-91)

NOTE 11: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “5. Account Settings” (formerly “5. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT830 Series
5. Account Settings	Account settings of backup/restore server (FTP/HTTPS).		
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).		
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).		

NOTE 12: This service is available to perform various tests such as Key Test, LED, LCD, and Call Test.

NOTE 13: Be careful when you specify “3. Enable (All Client)” because security services are disabled for Push when “3. Enable (All Client)” is set.

NOTE 14: For North America and China, this setting is not available. Therefore, this setting cannot be selected on the terminal.

NOTE 15: Up to 15 certificates can be selected.

NOTE 16: When “0” is specified, the warning indication will not be indicated.

NOTE 17: The following table shows the supporting authentication mode and IKE version.

X: Available N: Not available

Authentication Mode	IKE Version	
	IKEv1	IKEv2
PSK	X	X
RSA	N	X

NOTE 18: For IKE version “V1”, the supported authentication mode is “PSK” only. Therefore, when “1. V1” is specified, be sure to specify “1. PSK” on “2. Authentication Mode”. When “2. RSA” is specified, a communication error occurs.

NOTE 19: IKE authentication is executed with “PSK” which has been assigned in [0. Config] - [4. Security] - [5. VPN Client] - [4. IKE Settings] - [3. PSK Setting].

NOTE 20: IKE authentication is executed with “RSA” which has been assigned in [0. Config] - [4. Security] - [8. Certificate] - [2. Select] - [2. VPN].

NOTE 21: When “0” is specified, this setting will be disabled.

NOTE 22: When you register the Home URL, the indication on the Menu screen turns “Application Name” from “Phone”. Set the Lock Mode to Disable and Disable the Security key.

To connect a phone and IP Phone Manager, the port number to be used needs to match. Set the port number on the phone side from [0. Config] - [6. Application Setting] - [2. IPM Self Port].

NOTE 23: When you select the parameter of Service URL, a Menu for 1st - 4th Service URL is displayed. URL of up to four XML applications can be registered.

NOTE 24: If the firmware version of a terminal is FW5.2.0.0 or later, this menu becomes “5. VPN Settings” (formerly “5. VPN Client”). The setting details are as follows.

Parameter	Description	Default	DT830 Series
5. VPN Settings [FW5.2.0.0 or later]	Sets VPN.	-	-
1. VPN Mode	Sets whether to enable VPN Feature.	Disable	X
1. Disable	VPN Mode is disabled.		
2. Enable	VPN Mode is enabled.		
2. VPN Network Settings	Sets VPN network related data.	-	X
1. VPN Client Address	Sets the terminal IP address to be specified by VPN center site	0.0.0.0	X
2. VPN Gateway Address	Sets VPN Gateway address.	0.0.0.0	X
3. L2TP/IPsec Settings	Sets L2TP/IPsec related data.	-	X
1. IKE Settings	Sets IKE protocol related data.	-	X
1. IKE Version NOTE 17	Selects IKE protocol version.	V1	X
1. V1 NOTE 18	IKE protocol version is specified to V1.		
2. V2	IKE protocol version is specified to V2.		
2. Authentication Mode	Selects IKE authentication mode.	PSK	X
1. PSK NOTE 19	Authentication mode is specified to PSK.		
2. RSA NOTE 20	Authentication mode is specified to RSA.		
3. PSK Setting	Sets the parameter of PSK (up to 64 characters).	-	X
4. IKE ID	Sets the parameter of IKE ID (up to 127 characters).	-	X
5. SA Validity	Sets the expiration of SA validity when the IPsec connection has been established (3600-86400 seconds).	86400	X
2. Child SA Validity	Sets the expiration of Child SA validity when the IPsec connection has been established (3600-86400 seconds).	28800	X
3. NAT Traversal	Sets IPsec NAT traversal related data.	-	X
1. Keep Alive	Sets keep alive for NAT traversal (0, 1-3600 seconds). NOTE 21	20	
2. NAT Traversal Port	Sets port for NAT traversal (1024-65535).	4500	
4. Cert ID	Sets authentication ID for L2TP (up to 127 characters).	-	X
5. Cert Password	Sets authentication password for L2TP (up to 127 characters).	-	X

DT820 Series Configuration Menu List (FW 2.0.0.0)

This section lists the configuration menu used for DT820 Series configuration settings. DT820 (8LDX) and DT820 (6D) columns on the tables from the next page show whether the menu is available or not for each terminal type.

LEGEND	X: Available	N: Not available	-: Not applicable
---------------	--------------	------------------	-------------------

Starting Up

STEP1: When the terminal starting up, it displays the Phone Screen.

```
MARY SMITH           [200]
 4:26 PM MON 11 MAY 2015
MIC   DIR   HIST.   >>>>
```

STEP2: Press the Menu key to display the Menu Screen.

```
1 Setting
0 Config
  ↓      Back   OK
```


1. Setting

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
1. User Setting	Sets user interface related.	-	X	X
1. Incoming Call	Sets ringing and lamp indication data for incoming call.	-	X	X
1. Offhook Ring	Selects whether to use offhook ringing (ringing for incoming call in busy state).	Enable	X	X
1. Disable	Offhook ringing is OFF.			
2. Enable	Offhook ringing is ON.			
2. Headset Ring	Selects whether to use headset ringing.	Disable	X	X
1. Disable	Headset ringing is OFF.			
2. Enable	Headset ringing is ON.			
3. Ring Tone	Sets Ringing Tone for incoming call termination.	-	X	X
1. External Call NOTE 1	Sets Ringing Tone for external call termination.	Automatic	X	X
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.		X	-
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.			
2. Internal Call NOTE 1	Sets Ringing Tone for station call termination.		X	X
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3	Sets Download 1 ~ Download 3 for Ringing Tone.			
2. Talk	Sets call related data.	-	X	X
1. RTP Alarm NOTE 2	Sets whether to allow the alarm for detecting packet loss in receiving speech packets.	Automatic	X	X
0. Automatic	Applies RTP alarm setting on the SV9300.			
1. Disable	Alarm is OFF.			
2. Enable	Alarm is ON.	Automatic	X	X
2. DTMF Tone	Sets whether to hear DTMF (PB) tone on receiving DTMF signal.			
0. Automatic	Applies DTMF tone setting on the SV9300.			
1. Disable	DTMF tone is OFF.			
2. Enable	DTMF tone is ON.	Automatic NOTE 3	X	-
3. Key Touch Tone	Sets touch tone for pressing digit key.			
0. Automatic	Sets Automatic to key touch tone.			
1. No Tone	Sets key touch tone disabled.			
2. Tone	Sets key touch tone enabled.	Default	X	X
3. Voice	Enables the key touch tone by voice recognition.			
4. Hold Music	Sets Music-on-Hold.			
1. Default	Applies the default music (Minuet).	Default	X	X
2. Download	Applies the downloaded music.			

1. Setting (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
3. Display	Sets display related data.	-	X	X
1. Calendar Format	-	Automatic NOTE 4	-	-
1. Calendar Format (Setting prohibited)	-			
0. Automatic	-			
1. Type 1	-			
2. Type 2				
3. Type 3				
4. Type 4				
5. Type 5	-			
2. Time Format (Setting prohibited)	-	Automatic NOTE 5	-	-
0. Automatic (Setting prohibited)	-			
1. 12 hour Indication (Setting prohibited)	-			
2. 24 hour Indication (Setting prohibited)	-			
2. Local Volume	Selects whether to display volume gage when adjusting the volume level.	Automatic NOTE 6	X	X
0. Automatic	Sets Automatic for volume gage display.			
1. Disable	Volume gage is not displayed.			
	2. Enable	Volume gage is displayed.		
3. Screen Saver	-	-	-	-
1. Screen Saver Mode	Selects whether to use Screen Saver.	Disable	X	X
1. Disable	Screen Saver is not used.			
2. Enable	Screen Saver is used.			
2. Wait Time	Sets wait time until Screen Saver activation (1-999 min.).	120	X	X
4. Back Light	Switches ON/OFF the backlight on DT820.	Enable NOTE 7	X	X
1. Disable	Backlight is OFF.			
2. Enable	Backlight is ON.			

1. Setting (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
5. Language	Selects the language on date and time display.			
0. Automatic	Sets Automatic for language setting on date and time display.			
1. Japanese	Selected language is displayed on date and time display. * Each parameter in this menu is displayed in their native language.	Automatic NOTE 8	X	X
2. NEC English				
3. NEC Portuguese				
4. NEC Spanish				
5. NEC French				
6. German				
7. Italian				
8. Dutch				
9. Norwegian				
10. Danish				
11. Swedish				
12. Greek				
13. English				
14. Portuguese				
15. Spanish				
16. French				
17. Russian				
18. Turkish				
19. Romania				
20. Polish				
21. Catalan				
22. Korean				
23. Chinese (Simplified Chinese) NOTE 9				
24. Chinese (Traditional Chinese) NOTE 9				
6. Advanced	Sets other data for display.	-	-	-
1. Backlight Fade Control	Sets whether Backlight Fade Control is in service, when turning on/off the backlight of the display.	Enable	X	X
1. Disable	Disables Backlight Fade Control.			
2. Enable	Enables Backlight Fade Control.			
4. Change Password NOTE 10	Assigns the password of the terminal, used for data clear, security, and login of user mode.	0000	X	X
0. Setting Reset	Initializes all the assigned personal information on the terminal to factory default.	-	X	X

1. Setting (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. Download	Sets download related data.			
1. Download Files	Downloads files of Music-on-Hold, Ringing Tone, and Phonebook.	-	X	X
1. Hold Music	Downloads a sound file used for Music-on-Hold.	MOH.wav	X	X
2. Ring Tone	Downloads a sound file used for Ringing Tone.	-		
1. Download 1	Downloads Melody 1 ~ Melody 3.	Melody1. wav	X	-
2. Download 2		Melody2. wav		
3. Download 3		Melody3. wav		
2. Download Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X
3. Protocol	Specifies the protocol for file downloading.			
1. FTP	FTP is used.	FTP	X	X
2. TFTP	TFTP is used.			
4. FTP Settings	Sets data for accessing FTP server.			
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.			
3. Folder	Specifies the directory to save the download file.			
3. Data Backup/Restore	Makes backup of terminal data or restores from backup file.	-		
1. Data Backup	Saves the terminal setting to backup file. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
2. Data Restore	Restores the terminal data by reading out the saved backup file. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
3. Server Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X
4. Protocol	Specifies the protocol for file downloading.			
1. FTP	FTP is used.	FTP	X	X
2. TFTP	TFTP is used.			
5. FTP Settings	Sets data for accessing FTP server.			
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.			
3. Folder	Specifies the directory to save backup file.			

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- NOTE 1:** *To provide the Distinctive Ringing of external call/internal call by SV9300, be sure to use with default (Automatic). When “Automatic” is set, by the system data setting (CM08>390) of SV9300, Ringing Tone can be changed by the system data (CM15 Y=491, CM35 Y=034, 164) or terminal operations (“Feature” + “3”) (default: as per CM15 Y=491, CM35 Y=034, 164).*
- NOTE 2:** *When “Automatic” is set, RTP Alarm depends on the system data setting (CM08>1004) of SV9300 (default: To send).*
- NOTE 3:** *Follows the setting (CM15 Y=291) of SV9300 side when “Automatic” is set. In all other cases, follows the setting of a terminal side.*
- NOTE 4:** *Calendar Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.*
- NOTE 5:** *Time Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.*
- NOTE 6:** *A setting for Enable is applied when Automatic has been set.*
- NOTE 7:** *Back Light menu becomes effective when the system data (CM15 Y=485) of SV9300 is set to other than 0 (Always off).
Back Light of ten key cannot be used when terminals and option equipment for power failure are implemented.*
- NOTE 8:** *The setting of display language on the Clock Display screen depends on the system data settings (CM12 Y=63, CM04 Y=00) of SV9300.*
- NOTE 9:** *For the language setting 23 and 24, there are following conditions.*
- *When the language is selected from the language setting 1-22, or when SV9300 specifies the language other than Chinese while the language setting 0 (Automatic) is selected, the standard font is displayed.*
 - *When the language setting 23 is selected, or when SV9300 specifies the Chinese (Simplified Chinese) while the language setting 0 (Automatic) is selected, the Chinese (Simplified Chinese) font is displayed.*
 - *When the language setting 24 is selected, or when SV9300 specifies the Chinese (Traditional Chinese) while the language setting 0 (Automatic) is selected, the Chinese (Traditional Chinese) font is displayed.*
- NOTE 10:** *Change Password is used for password for User Mode of Web programming and security unlock. Be sure to change the default password.*
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0. Config

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
1. Network Settings	Sets network related data.	-	X	X
1. DHCP Mode	Selects whether to use DHCP protocol.	Enable	X	X
1. Disable	DHCP is not used.			
2. Enable	DHCP is used.			
2. IP Address	Sets DT820 Series terminal's IP address.	0.0.0.0	X	X
3. Default Gateway	Sets default router's IP address.	0.0.0.0	X	X
4. Subnet Mask	Sets subnet mask.	0.0.0.0	X	X
5. DNS Address	Sets DNS server's IP address.	0.0.0.0	X	X
6. Advanced Settings	Sets network expanded data.	-	X	X
1. LAN Port Settings	Sets LAN port.	-	X	X
1. Speed&Duplex	Sets communication speed of Ether port.	Autonego	X	X
1. Autonego	Autonegotiation			
2. 100M Full	100Mbps in full-duplex mode			
3. 100M Half	100Mbps in half-duplex mode			
4. 10M Full	10Mbps in full-duplex mode			
5. 10M Half	10Mbps in half-duplex mode			
2. VLAN Mode	Selects whether to use VLAN on terminal.	Disable	X	X
1. Disable	VLAN is not used.			
2. Enable	VLAN is used.			
3. VLAN ID	Sets VLAN ID of terminal [1-4094].	7	X	X
4. VLAN Priority	Sets VLAN Priority of terminal [0-7].	5	X	X
5. Auto-MDIX	Selects whether to use Auto-MDIX.	Enable	X	X
1. Disable	Auto-MDIX is not used.			
2. Enable	Auto-MDIX is used.			

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. PC Port Settings	Sets PC port.	-	X	X
1. Speed&Duplex	Sets communication speed of PC port.	Autonego	X	X
1. Autonego	Autonegotiation			
2. 100M Full	100Mbps in full-duplex mode			
3. 100M Half	100Mbps in half-duplex mode			
4. 10M Full	10Mbps in full-duplex mode			
5. 10M Half	10Mbps in half-duplex mode			
2. VLAN Mode	Selects whether to use port VLAN of PC port.	Disable	X	X
1. Disable	Port VLAN of PC port is not used.			
2. Enable	Port VLAN of PC port is used.			
3. Port VLAN ID	Sets port VLAN ID of PC port [1-4094].	1	X	X
4. Port VLAN Priority	Sets port VLAN priority of PC port [0-7].	0	X	X
5. Port Available	Selects whether to use PC port.	Automatic	X	X
0. Automatic	Sets Automatic for PC port.			
1. Disable	PC port is not used.			
6. EAPOL	Select whether to forward EAPOL packet from PC port.	Disable	X	X
1. Disable	Not forward EAPOL packet from PC port.			
2. Enable	Forwards EAPOL packet from PC port.			
7. PC Port Security	Select whether to clear authentication status of the terminal which is connected to PC port for authentication switch when the link of PC port has been down.	Enable	X	X
1. Disable	Not clear the authentication status.			
2. Enable	Clear the authentication status.			
8. Auto-MDIX	Selects whether to use Auto-MDIX.	Enable	X	X
1. Disable	Auto-MDIX is not used.			
2. Enable	Auto-MDIX is used.			
3. CDP Mode (Setting prohibited)	-	-	-	-
4. Type of Service	Sets ToS value.	-	X	X
1. RTP	Sets ToS value for RTP in Hex.	A0		
2. SIP	Sets ToS value for SIP in Hex.	C0		
5. Self Port Settings	Sets each port for DT820 Series.	-	-	-
1. RTP Self Port	Sets RTP port for DT820 Series [1024-65520 (except 59986-60000)]. NOTE 1 * Odd port numbers cannot be set.	3462	X	X
2. SIP Self Port	Sets SIP port for DT820 Series [1024-65534]. NOTE 1	5060	X	X
6. LLDP Settings	Sets information related to LLDP.	-	X	X
1. LLDP Mode	Sets whether to transmit/receive LLDP packets.	Enable	X	X
1. Disable	DT820 Series does not transmit/receive LLDP packets.			
2. Enable	DT820 Series transmits/receives LLDP packets.			

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. Auto Setting Mode	Sets whether to perform data setting automatically when LLDP packets are received.			
1. Disable	DT820 Series does not perform data setting automatically when LLDP packets are received.	Enable	X	X
2. Enable	DT820 Series performs data setting automatically when LLDP packets are received.			
3. RX Waiting Time	For Auto Setting, sets the waiting time until DT820 Series receives LLDP packets (1-60 seconds).	15	X	X
4. Transmit Interval	Sets intervals to transmit LLDP packets (5-32768 seconds).	30	X	X
5. Hold Multiplier NOTE 2	Sets a multiplier determining the time to live for the LLDP packets to be transmitted (2-10 multiples).	4	X	X
6. Fast Start Count	Sets the number of times to transmit LLDP packets at one-second intervals after DT820 Series starts up (1-10 times).	3	X	X
7. Transparent Mode	Sets whether Transparent Mode of LLDP packets is enabled between the PC port and the LAN port.			
1. Disable	Transparent Mode of LLDP packets is disabled.	Enable	X	X
2. Enable	Transparent Mode of LLDP packets is enabled.			
8. Asset ID	Sets the Asset ID for LLDP packets to be transmitted. * “-” is displayed in default settings.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
7. Spare IP Settings (Setting prohibited)	-	-	X	X
1. Spare/Backup IP Mode (Setting prohibited)	-	Disable	-	-
1. Disable	-			
2. Spare IP (Setting prohibited)	-			
3. Backup IP (Setting prohibited)	-			
2. Network Settings (Setting prohibited)	-	-	-	-
1. IP Address (Setting prohibited)	-	0.0.0.0		
2. Default Gateway (Setting prohibited)	-	0.0.0.0		
3. Subnet Mask (Setting prohibited)	-	0.0.0.0		
4. DNS Address (Setting prohibited)	-	0.0.0.0		
3. SIP Settings (Setting prohibited)	-	-	-	-
1. 2nd Server Address (Setting prohibited)	-	0.0.0.0	-	-
2. 3rd Server Address (Setting prohibited)	-	0.0.0.0	-	-
3. 4th Server Address (Setting prohibited)	-	0.0.0.0	-	-
4. 2nd Server Port (Setting prohibited)	-	0	-	-
5. 3rd Server Port (Setting prohibited)	-	0	-	-
6. 4th Server Port (Setting prohibited)	-	0	-	-
8. TTL Settings	Setting for multicast TTL (Time To Live)	32	X	X
1. RTP (Multicast)	Sets the value of multicast TTL (0-255)		X	X
9. Priority Queue	Sets whether to enable Priority Queue of the internal switching HUB.	Disable	X	X
1. Disable	Priority Queue is disabled.		X	X
2. Enable	Priority Queue is enabled.		X	X
2. SIP Settings	Sets SIP related data.	-	X	X
1. SIP User	-	-	-	-
1. User ID (Setting prohibited)	-	-	-	-
2. Password (Setting prohibited)	-	-	-	-
3. Extension Number (Setting prohibited)	-	-	-	-
4. Backup Logon (Setting prohibited)	-	Disable	-	-

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)	
2. Server Address & URI	Sets IP address or URI of SIP server. NOTE 3	-	X	X	
1. 1st Server Address	Sets IP address or fully qualified domain name (FQDN) of 1st ~ 4th SIP server. NOTE 4	0.0.0.0	X	X	
2. 2nd Server Address					
3. 3rd Server Address					
4. 4th Server Address					
5. Registrar Destination (Setting prohibited)	-	-	-	-	
1. 1st Register Address (Setting prohibited)	-	0.0.0.0	-	-	
2. 2nd Register Address (Setting prohibited)					
3. 3rd Register Address (Setting prohibited)					
4. 4th Register Address (Setting prohibited)					
3. Access Mode	Sets access mode for SIP server.	Normal	X	X	
1. Normal	Normal access				
2. Remote	Remote access				
4. SIP Server Port	Sets SIP server port [1024-65535].	5060	X	X	
1. 1st Server Port	Specifies the destination port number of sending packet to 1st ~ 4th SIP server. NOTE 5				
2. 2nd Server Port					
3. 3rd Server Port					
4. 4th Server Port					
5. Registrar Destination (Setting prohibited)	-				
1. 1st Register Port (Setting prohibited)	-			-	-
2. 2nd Register Port (Setting prohibited)					
3. 3rd Register Port (Setting prohibited)					
4. 4th Register Port (Setting prohibited)					

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
5. Area ID	Sets the area (country) using DT820 Series.			
0. Automatic	The SV9300 side setting is applied when Automatic is selected.			
1. Japan	Selects the area (country) NOTE: Area IDs allowed on the SV9300 can be selected.	Automatic	X	X
2. America (Canada and North America)				
3. Australia				
4. Others				
5. Hong Kong				
6. Malaysia				
7. Singapore				
8. Mexico				
9. Taiwan				
10. New Zealand				
11. Korea				
12. Brazil				
13. China				
14. Thailand				
15. Thailand (EGAT)				
16. Sri Lanka				
17. Germany				
18. Italy				
19. Netherlands				
20. Catalonia				
21. Denmark				
22. Sweden				
23. Poland				
24. Czech				
25. Norway				
26. France				
27. Portugal				
28. Spain				
29. Austria				
30. Belgium				
31. UK				
32. Greece				
33. Switzerland				
34. South Africa				
35. Brazil (EMEA)				

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
6. Extra Function (Setting prohibited)	-	-	-	-
1. Retransmit Interval (Setting prohibited)	-	Standard	-	-
1. Standard	-			
2. 6 Seconds (Setting prohibited)	-			
2. Keeping Call (Setting prohibited)	-	Enable	-	-
1. Disable (Setting prohibited)	-			
2. Enable	-			
3. Howler Tone (Setting prohibited) NOTE 6	-	Disable	-	-
1. Disable	-			
2. Enable (Setting prohibited)	-			
7. Encryption	Sets encryption data.	-	X	X
1. Authentication Mode	Selects whether to use authentication.	Disable	X	X
1. Disable	Authentication is not used.			
2. Enable	Authentication is used.			
2. One Time Password	Sets One Time Password.	-	X	X
3. Authentication Status	Displays authentication status.	-	X	X
8. NAT Traversal (Setting prohibited)	-	-	-	-
1. NAT Traverse Mode (Setting prohibited)	-	Disable	-	-
1. Disable (Setting prohibited)				
2. Dynamic (Setting prohibited)				
3. Static (Setting prohibited)				
2. Network Area Name (Setting prohibited)	-	-	-	-
3. WAN Settings (Setting prohibited)	-	-	-	-
1. WAN Mate IP Address (Setting prohibited)		0.0.0.0		
2. WAN SIP Mate Port (Setting prohibited)		5060		
3. WAN Self IP Address (Setting prohibited)		0.0.0.0		

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
3. Maintenance Settings	Sets maintenance related data.	-	X	X
1. Download Menu	Downloads specified file to DT820 Series.	-	X	X
1. Download Files	Specifies download file name for each file type.	-	X	X
1. Boot & Program	Downloads boot and program.	DT820 (6D): ityisipe.tgz DT820 (8LDX): ityisipex.tgz	X	X
2. Config	Downloads configuration file.	config.tgz	X	X
3. Voice File	Downloads voice file.	voice.tgz	X	-
4. Language Pack	Downloads translation data file.	LangPack.tgz	X	X
2. Download Address	Sets IP address of FTP/TFTP server.	0.0.0.0	X	X
3. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
4. FTP Settings	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.	-	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X
2. Hard Reset	Resets DT820 Series without data saving.	-	X	X
3. Data Clear	Initializes DT820 Series selecting the initialization mode.	-	X	X
1. Factory Value	Initializes DT820 Series to factory default.	-	X	X
2. Terminal Data	Clears terminal data on DT820 Series.	-	X	X
3. Personal Data	Clears personal data on DT820 Series.	-	X	X
4. Adjust	Adjusts LCD, audio, and button kit.	-	X	X
1. Default Audio	Sets audio data.	-	X	X
1. Sidetone Volume	Sets Side Tone volume [0-7].	4	X	X
2. Send Gain	Sets sending gain of handset [0-31].	24	X	X
3. Receive Bias	Sets receiving bias of handset [0-7].	0	X	X
4. Receive Gain	Sets receiving gain of handset [0-31].	22	X	X
5. Talk Characteristics	Sets talk characteristics of handset.	-	X	X
1.Noise Suppression	Sets whether to enable the Noise Suppressor. (Set "1. Disable" in normal operation.)	Disable	X	X
1. Disable	Disables the Noise Suppressor.			
2. Enable	Enables the Noise Suppressor.			
2. Handsfree Mode	Sets a send gain mode of handsfree call.	Retro mode	X	X
1. Retro Mode	Sets to Retro mode.			
2. New Mode	Sets to New mode.			
6. DTMF Send Level	Sets DTMF Send Level [0-48]. (level 0 (maximum) ~ level 48 (minimum))	32		

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. EHS Headset	Sets EHS Headset data.			
1. Disable	Disable the Headset Jack connection.	Disable	X	X
2. Enable	Enable the Headset Jack connection.			
5. Auto config	Selects whether to use auto-config function.			
1. Disable	Auto config is not used.	Enable	X	X
2. Enable	Auto config is used.			
6. Data Maintenance	Sets data maintenance related.	-	X	X
1. Backup	Executes data backup.	-	X	X
1. All Data	Executes backup for all the configuration setting and data. * For encryption, "AllData.tgz.ef" is applied as a file name.	AllData.tgz	X	X
2. Terminal Data	Executes backup for terminal setting and data. * For encryption, "TerminalData.tgz.ef" is applied as a file name.	Terminal Data.tgz	X	X
3. Personal Data	Executes backup for personal setting and data. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
2. Restore	Executes restoration.	-	X	X
1. All Data	Restores all the configuration setting and data. * For encryption, "AllData.tgz.ef" is applied as a file name.	AllData.tgz	X	X
2. Terminal Data	Restores terminal setting and data. * For encryption, "TerminalData.tgz.ef" is applied as a file name.	Terminal Data.tgz	X	X
3. Personal Data	Restores personal setting and data. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
3. Server Address	Sets IP address of the server that executes data backup/restoration.	0.0.0.0	X	X
4. Protocol	Selects FTP or TFTP for file downloading protocol.			
1. FTP	FTP is used.	FTP	X	X
2. TFTP	TFTP is used.			
5. FTP Settings	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.	-	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
7. Simple Test NOTE 7	Changes the terminal to test mode.	-	X	X
1. Call Test	Call Test	-	X	X
2. Call Test Address	Sets an IP address of the opposite terminal to perform the call test.	10.136.20.188	X	X
3. Key Test 1 (Random)	Tests to press the random button.	-	X	X
4. Key Test 2 (Sequential)	Tests to press the sequential button.	-	X	X
5. Loop Back Test	Tests the voice loop back.	-	X	X
1. Handset - Hand Speaker	Tests to speak and receive on the hand speaker.	-	X	X
2. Handset - Speaker	Tests to speak on the handset and receive on the speaker.	-	X	X
3. Headset - Head Speaker	Tests to speak and receive on the head speaker.	-	X	X
4. Headset - Speaker	Tests to speak on the headset and receive on the speaker.	-	X	X
5. MIC - Hand Speaker	Tests to speak on the MIC and receive on the hand speaker.	-	X	X
6. MIC - Head Speaker	Tests to speak on the MIC and receive on the head speaker.	-	X	X
7. MIC - Speaker	Tests to speak on the MIC and receive on the speaker.	-	X	X
6. Speaker Test	Tests the speaker.	-	X	X
1. Japan	A DT for Japan is heard in maximum volume.	-	X	X
2. America	A DT for North America is heard in maximum volume.	-	X	X
7. Receiver Test	Receiver Test for a handset	-	X	X
4. Security	Sets security related data.	-	X	X
1. Web Programming	Selects whether to use Web Programming.			
1. Disable	Web Programming is disabled.	Enable	X	X
2. Enable	Web Programming is enabled.			
2. Admin Password	Sets administrator's password. * Be sure to change the default password.	6633222	X	X
3. 802.1X Supplicant	Sets 802.1X authentication related parameter.	-	X	X
1. Supplicant	Sets whether to use supplicant on DT820 Series.			
1. Disable	Supplicant is disabled on DT820 Series.	Disable	X	X
2. Enable	Supplicant is enabled on DT820 Series.			
2. EAP Method (Setting prohibited)	-	-	-	-
3. Account	Sets user ID for authentication.	-	X	X
4. Password	Sets password for authentication.	-	X	X
5. EAPOL VLAN Mode	Selects whether to add VLAN tag to EAP packet.			
1. Disable	VLAN tag is not added to EAP packet.	Enable	X	X
2. Enable	VLAN tag is added to EAP packet.			

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
4. User Password	Setting for User Password.	Enable	X	X
1. Expiration Date	Sets the expiration of user password (0-90 days). * When "0" is specified, the warning indication will not be indicated.	0	X	X
2. Warning Interval	Sets the interval to regenerate the warning message which prompts a password resetting (1-24 hours).	1	X	X
5. Terminal Information	Displays terminal information.	-	X	X
Hardware Version	Displays hardware version of DT820 Series.	-	X	X
MAC Address	Displays MAC address of DT820 Series.	-	X	X
Firmware Version	Displays firmware version of DT820 Series.	-	X	X
6. Application Settings	Sets application data.	-	X	X
1. IPM Self Port	Sets the port for connecting to IP Phone Manager [1024-65535 (except 17185, 60000)]. NOTE 1	3530	X	X
2. Voice Rec Settings (Setting prohibited)	-	-	-	-
7. License Settings	-	-	X	X
1. License Server Settings	-	-	X	X
1. Server Address	Enter the IP address of LMC [0.0.0.0-255.255.255.254].	0.0.0.0	X	X
2. Server Port	Enter the server port of LMC [1024-65535].	6080	X	X
2. License Request	-	-	X	X
1. Gigabit Ethernet	-	-	X	X
1. Disable	Set when the Gigabit Ethernet license is not used.	Disable	X	X
2. Enable	Set when the Gigabit Ethernet license is used.		X	X
2. Line Key	-	-	X	X
1. Disable	Set when the Line Key license is not used.	Disable	X	-
2. Line 16	Set when the 16 Line Key license is used.		X	-
3. Line 32	Set when the 32 Line Key license is used.		X	-
3. Adjust	-	-	X	X
1. License User Prefix	Enter the License User Prefix.	Blank	X	X

- NOTE 1:** *A unique number must be assigned to each of the following ports. For RTP Self Port, 16 ports starting from an assigned port are used.*
- RTP Self Port
 - SIP Self Port
 - Push Server Port
 - IPM Self Port
- NOTE 2:** *LLDP time to live = LLDP packet transmission interval × the set value (65535 seconds at most)*
- NOTE 3:** *“SIP server” used here refers to the SV9300. The SV9300 acts as SIP server where DT820 Series terminal is registered.*
- NOTE 4:** *Follow the IP address setting (CM0B Y=1XX>00) of VOIP port on SV9300.*
- NOTE 5:** *Follow the SIP Server port setting (CM0B Y=1XX>32) on SV9300 (example: change 5060 to 5080 when default is set).*
- NOTE 6:** *Howler Tone to be sent depends on the system data setting (CM08>153, CM13 Y=04) of SV9300 (default: To send).*
- NOTE 7:** *This service is available to perform various tests such as Key Test, LED, LCD, and Call Test.*

DT820 Series Configuration Menu List (FW 3.0.0.0 or later)

This section lists the configuration menu used for DT820 Series configuration settings. DT820 (8LDX) and DT820 (6D) columns on the tables below show whether the menu is available or not for each terminal type.

NOTE: *History and Directory are displayed only for DT820 (8LDX). However, Terminal Call History and Terminal Phone Book are not supported. Therefore, when you select “History” on the main menu, Call History at the SV9300 side will be activated.*

LEGEND	X: Available	N: Not available	-: Not applicable
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3. Service

Parameter	Description	Default
3. Service	Starts up XML browser.	-
1. Service NOTE 1	Starts up XML browser.	-

NOTE 1: *For DT820 (8LDX), service menu is displayed only when the URL is set by [0. Config] - [6. Application Settings] - [1. XML Settings] - [4. Service URL].*

NOTE 2: *For DT820 (6D), service menu is displayed only when the URL is set by [0. Config] - [6. Application Settings] - [1. XML Settings] - [3. Service URL].*

NOTE 3: *At most four services for DT820 (8LDX)/two services for DT820 (6D) are displayed on the menu. Even though [1. Service] isn't displayed on the menu when only one URL is set, selection of [3. Service] starts up XML browser.*

4. Setting

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
1. User Setting	Sets user interface related.	-	X	X
1. Incoming Call	Sets ringing and lamp indication data for incoming call.	-	X	X
1. Offhook Ring	Selects whether to use offhook ringing (ringing for incoming call in busy state).	Enable	X	X
1. Disable	Offhook ringing is OFF.			
2. Enable	Offhook ringing is ON.			
2. Headset Ring	Selects whether to use headset ringing.	Disable	X	X
1. Disable	Headset ringing is OFF.			
2. Enable	Headset ringing is ON.			

4. Setting (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
3. Ring Tone	Sets Ringing Tone for incoming call termination.	-	X	X
1. External Call NOTE 1	Sets Ringing Tone for external call termination.	Automatic	X	X
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3 NOTE 2	Sets Download 1 ~ Download 3 for Ringing Tone.			
2. Internal Call NOTE 1	Sets Ringing Tone for station call termination.			
0. Automatic	Sets Automatic for the Ringing Tone of incoming call termination.			
1. Tone Type 1 ~ 14. Tone Type 14	Sets Tone Type 1 ~ Tone Type 14 for Ringing Tone.			
15. Download 1 ~ 17. Download 3 NOTE 2	Sets Download 1 ~ Download 3 for Ringing Tone.			
2. Talk	Sets call related data.	-	X	X
1. RTP Alarm NOTE 3	Sets whether to allow the alarm for detecting packet loss in receiving speech packets.	Automatic	X	X
0. Automatic	Applies RTP alarm setting on the SV9300.			
1. Disable	Alarm is OFF.			
2. Enable	Alarm is ON.			
2. DTMF Tone	Sets whether to hear DTMF (PB) tone on receiving DTMF signal.	Automatic	X	X
0. Automatic	Applies DTMF tone setting on the SV9300.			
1. Disable	DTMF tone is OFF.			
2. Enable	DTMF tone is ON.			
3. Key Touch Tone	Sets touch tone for pressing digit key.	Automatic NOTE 4	X	X
0. Automatic	Sets Automatic to key touch tone.			
1. No Tone	Sets key touch tone disabled.			
2. Tone	Sets key touch tone enabled.			
3. Voice	Enables the key touch tone by voice recognition.			
4. Hold Music	Sets Music-on-Hold.	Default	X	X
1. Default	Applies the default music (Minuet).			
2. Download	Applies the downloaded music.			

4. Setting (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
3. Display	Sets display related data.	-	X	X
1. Calendar Format	-	Automatic NOTE 5	-	-
1. Calendar Format (Setting prohibited)	-			
0. Automatic	-			
1. Type 1	-			
2. Type 2				
3. Type 3				
4. Type 4				
5. Type 5	-			
2. Time Format (Setting prohibited)	-	Automatic NOTE 6	-	-
0. Automatic (Setting prohibited)	-			
1. 12 hour Indication (Setting prohibited)	-			
2. 24 hour Indication (Setting prohibited)	-			
2. Local Volume	Selects whether to display volume gage when adjusting the volume level.	Automatic NOTE 7	X	X
0. Automatic	Sets Automatic for volume gage display.			
1. Disable	Volume gage is not displayed.			
	2. Enable	Volume gage is displayed.		
3. Screen Saver	-	-	-	-
1. Screen Saver Mode	Selects whether to use Screen Saver.	Disable	X	X
1. Disable	Screen Saver is not used.			
2. Enable	Screen Saver is used.			
2. Wait Time	Sets wait time until Screen Saver activation (1-999 min.).	120	X	X
4. Back Light	Switches ON/OFF the backlight on DT820.	Enable NOTE 8	X	X
1. Disable	Backlight is OFF.			
2. Enable	Backlight is ON.			

4. Setting (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
5. Language	Selects the language on date and time display.			
0. Automatic	Sets Automatic for language setting on date and time display.			
1. Japanese	<p>Selected language is displayed on date and time display. * Each parameter in this menu is displayed in their native language.</p>	Automatic NOTE 9	X	X
2. NEC English				
3. NEC Portuguese				
4. NEC Spanish				
5. NEC French				
6. German				
7. Italian				
8. Dutch				
9. Norwegian				
10. Danish				
11. Swedish				
12. Greek				
13. English				
14. Portuguese				
15. Spanish				
16. French				
17. Russian				
18. Turkish				
19. Azərbaycanca [FW5.1.0.0 or later]				
20. Romania				
21. Polish				
22. Catalan				
23. Korean				
24. Chinese (Simplified Chinese) NOTE 10				
25. Chinese (Traditional Chinese) NOTE 10				
6. Advanced	Sets other data for display.	-	-	-
1. Backlight Fade Control	Sets whether Backlight Fade Control is in service, when turning on/off the backlight of the display.	Enable	X	X
1. Disable	Disables Backlight Fade Control.			
2. Enable	Enables Backlight Fade Control.			
4. Change Password NOTE 11	Assigns the password of the terminal, used for data clear, security, and login of user mode.	0000	X	X

4. Setting (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
6. Usability	Sets the operation of Help key while popup window is displayed.	-	X	X
1. Help Key Mode	Setting for Help Key Mode (SubMenu/Popup Window).	SubMenu	X	X
1. SubMenu	Sets the Help key which displays submenu.			
2. Popup Window	Sets the Help key which displays pop-up window.			
0. Setting Reset	Initializes all the assigned personal information on the terminal to factory default.	-	X	X
2. Download	Sets download related data.	-	X	X
1. Download Files	Downloads files of Music-on-Hold, Ringing Tone, and Phonebook.	-	X	X
1. Hold Music	Downloads a sound file used for Music-on-Hold.	MOH.wav	X	X
2. Ring Tone	Downloads a sound file used for Ringing Tone.	-	X	-
1. Download 1	Downloads Melody 1 ~ Melody 3.	Melody1.wav		
2. Download 2		Melody2.wav		
3. Download 3		Melody3.wav		
2. Download Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X	X
3. Protocol	Specifies the protocol for file downloading.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.			
4. FTP Settings NOTE 12	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.			
2. Password	Sets password for login to FTP server.			
3. Folder	Specifies the directory to save the download file.			
3. Data Backup/Restore	Makes backup of terminal data or restores from backup file.	-	X	X
1. Data Backup	Saves the terminal setting to backup file. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz		
2. Data Restore	Restores the terminal data by reading out the saved backup file. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
3. Server Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X	X
4. Protocol	Specifies the protocol for file downloading.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.			

4. Setting (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
5. FTP Settings NOTE 13	Sets data for accessing FTP server.			
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.			
3. Folder	Specifies the directory to save backup file.			

NOTE 1: To provide the Distinctive Ringing of external call/internal call by SV9300, be sure to use with default (Automatic). When “Automatic” is set, by the system data setting (CM08>390) of SV9300, Ringing Tone can be changed by the system data (CM15 Y=491, CM35 Y=034, 164) or terminal operations (“Feature” + “3”) (default: as per CM15 Y=491, CM35 Y=034, 164).

NOTE 2: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “15. Music Ring 1~17. Music Ring 3” (formerly “15. Download 1~17. Download 3”). The setting details are as follows.

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
15. Music Ring 1~17. Music Ring 3	Sets Music Ring 1 ~ Music Ring 3.	Automatic	X	X

NOTE 3: When “Automatic” is set, RTP Alarm depends on the system data setting (CM08>1004) of SV9300 (default: To send).

NOTE 4: Follows the setting (CM15 Y=291) of SV9300 side when “Automatic” is set. In all other cases, follows the setting of a terminal side.

NOTE 5: Calendar Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.

NOTE 6: Time Format follows the setting (CM08>2001/CM12 Y=58) of SV9300.

NOTE 7: A setting for Enable is applied when Automatic has been set.

NOTE 8: Back Light menu becomes effective when the system data (CM15 Y=485) of SV9300 is set to other than 0 (Always off).

Back Light of ten key cannot be used when terminals and option equipment for power failure are implemented.

NOTE 9: The setting of display language on the Clock Display screen depends on the system data settings (CM12 Y=63, CM04 Y=00) of SV9300.

NOTE 10: For the language setting 23 and 24, there are following conditions.

- When the language is selected from the language setting 1-22, or when SV9300 specifies the language other than Chinese while the language setting 0 (Automatic) is selected, the standard font is displayed.
- When the language setting 23 is selected, or when SV9300 specifies the Chinese (Simplified Chinese) while the language setting 0 (Automatic) is selected, the Chinese (Simplified Chinese) font is displayed.
- When the language setting 24 is selected, or when SV9300 specifies the Chinese (Traditional Chinese) while the language setting 0 (Automatic) is selected, the Chinese (Traditional Chinese) font is displayed.

NOTE 11: Change Password is used for password for User Mode of Web programming and security unlock. Be sure to change the default password.

NOTE 12: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “4. Account Settings” (formerly “4. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
4. Account Settings	Account settings of download server (FTP/HTTPS).			
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).			
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).			

NOTE 13: If the firmware version of a terminal is FW5.1.0.0 or later, this menu becomes “5. Account Settings” (formerly “5. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
5. Account Settings	Account settings of backup/restore server (FTP/HTTPS).			
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).			
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).			

5. Phone

Depending on the Home URL setting by [0. Config] - [6. Application Settings] - [1. XML Settings] - [2. XML Browser] - [1. Home URL], this menu is displayed as follows.

- When Home URL is not registered, home screen is displayed and the menu name is displayed as “Phone”.
- When Home URL is registered, XML application screen is displayed, and the menu name is displayed as the application name to be described in the XML application.

0. Config

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
1. Network Settings	Sets network related data.	-	X	X
1. DHCP Mode	Selects whether to use DHCP protocol.	Enable	X	X
1. Disable	DHCP is not used.			
2. Enable	DHCP is used.			
2. IP Address	Sets DT820 Series terminal's IP address.	0.0.0.0	X	X
3. Default Gateway	Sets default router's IP address.	0.0.0.0	X	X
4. Subnet Mask	Sets subnet mask.	0.0.0.0	X	X
5. DNS Address	Sets DNS server's IP address.	0.0.0.0	X	X
6. Advanced Settings	Sets network expanded data.	-	X	X
1. LAN Port Settings	Sets LAN port.	-	X	X
1. Speed&Duplex	Sets communication speed of Ether port.	Autonego	X	X
1. Autonego	Autonegotiation			
2. 100M Full	100Mbps in full-duplex mode			
3. 100M Half	100Mbps in half-duplex mode			
4. 10M Full	10Mbps in full-duplex mode			
5. 10M Half	10Mbps in half-duplex mode			
2. VLAN Mode	Selects whether to use VLAN on terminal.	Disable	X	X
1. Disable	VLAN is not used.			
2. Enable	VLAN is used.			
3. VLAN ID	Sets VLAN ID of terminal [1-4094].	7	X	X
4. VLAN Priority	Sets VLAN Priority of terminal [0-7].	5	X	X
5. Auto-MDIX	Selects whether to use Auto-MDIX.	Enable	X	X
1. Disable	Auto-MDIX is not used.			
2. Enable	Auto-MDIX is used.			

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. PC Port Settings	Sets PC port.	-	X	X
1. Speed&Duplex	Sets communication speed of PC port.	Autonego	X	X
1. Autonego	Autonegotiation			
2. 100M Full	100Mbps in full-duplex mode			
3. 100M Half	100Mbps in half-duplex mode			
4. 10M Full	10Mbps in full-duplex mode			
5. 10M Half	10Mbps in half-duplex mode			
2. VLAN Mode	Selects whether to use port VLAN of PC port.	Disable	X	X
1. Disable	Port VLAN of PC port is not used.			
2. Enable	Port VLAN of PC port is used.			
3. Port VLAN ID	Sets port VLAN ID of PC port [1-4094].	1	X	X
4. Port VLAN Priority	Sets port VLAN priority of PC port [0-7].	0	X	X
5. Port Available	Selects whether to use PC port.	Automatic	X	X
0. Automatic	Sets Automatic for PC port.			
1. Disable	PC port is not used.			
6. EAPOL	Select whether to forward EAPOL packet from PC port.	Disable	X	X
1. Disable	Not forward EAPOL packet from PC port.			
2. Enable	Forwards EAPOL packet from PC port.			
7. PC Port Security	Select whether to clear authentication status of the terminal which is connected to PC port for authentication switch when the link of PC port has been down.	Enable	X	X
1. Disable	Not clear the authentication status.			
2. Enable	Clear the authentication status.			
8. Auto-MDIX	Selects whether to use Auto-MDIX.	Enable	X	X
1. Disable	Auto-MDIX is not used.			
2. Enable	Auto-MDIX is used.			
3. CDP Mode (Setting prohibited)	-	-	-	-
4. Type of Service	Sets ToS value.	-	X	X
1. RTP	Sets ToS value for RTP in Hex.	A0		
2. SIP	Sets ToS value for SIP in Hex.	C0		
5. Self Port Settings	Sets each port for DT820 Series.	-	-	-
1. RTP Self Port	Sets RTP port for DT820 Series [1024-65520 (except 59986-60000)]. NOTE 1 * Odd port numbers cannot be set.	3462	X	X
2. SIP Self Port	Sets SIP port for DT820 Series [1024-65534]. NOTE 1	5060	X	X
6. LLDP Settings	Sets information related to LLDP.	-	X	X
1. LLDP Mode	Sets whether to transmit/receive LLDP packets.	Enable	X	X
1. Disable	DT820 Series does not transmit/receive LLDP packets.			
2. Enable	DT820 Series transmits/receives LLDP packets.			

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. Auto Setting Mode	Sets whether to perform data setting automatically when LLDP packets are received.			
1. Disable	DT820 Series does not perform data setting automatically when LLDP packets are received.	Enable	X	X
2. Enable	DT820 Series performs data setting automatically when LLDP packets are received.			
3. RX Waiting Time	For Auto Setting, sets the waiting time until DT820 Series receives LLDP packets (1-60 seconds).	15	X	X
4. Transmit Interval	Sets intervals to transmit LLDP packets (5-32768 seconds).	30	X	X
5. Hold Multiplier NOTE 2	Sets a multiplier determining the time to live for the LLDP packets to be transmitted (2-10 multiples).	4	X	X
6. Fast Start Count	Sets the number of times to transmit LLDP packets at one-second intervals after DT820 Series starts up (1-10 times).	3	X	X
7. Transparent Mode	Sets whether Transparent Mode of LLDP packets is enabled between the PC port and the LAN port.			
1. Disable	Transparent Mode of LLDP packets is disabled.	Enable	X	X
2. Enable	Transparent Mode of LLDP packets is enabled.			
8. Asset ID	Sets the Asset ID for LLDP packets to be transmitted. * “-” is displayed in default settings.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
7. Spare IP Settings (Setting prohibited)	-	-	X	X
1. Spare/Backup IP Mode (Setting prohibited)	-	Disable	-	-
1. Disable	-			
2. Spare IP (Setting prohibited)	-			
3. Backup IP (Setting prohibited)	-			
2. Network Settings (Setting prohibited)	-	-	-	-
1. IP Address (Setting prohibited)	-	0.0.0.0		
2. Default Gateway (Setting prohibited)	-	0.0.0.0		
3. Subnet Mask (Setting prohibited)	-	0.0.0.0		
4. DNS Address (Setting prohibited)	-	0.0.0.0		
3. SIP Settings (Setting prohibited)	-	-	-	-
1. 2nd Server Address (Setting prohibited)	-	0.0.0.0	-	-
2. 3rd Server Address (Setting prohibited)	-	0.0.0.0	-	-
3. 4th Server Address (Setting prohibited)	-	0.0.0.0	-	-
4. 2nd Server Port (Setting prohibited)	-	0	-	-
5. 3rd Server Port (Setting prohibited)	-	0	-	-
6. 4th Server Port (Setting prohibited)	-	0	-	-
8. TTL Settings	Setting for multicast TTL (Time To Live)	32	X	X
1. RTP (Multicast)	Sets the value of multicast TTL (0-255)		X	X
9. Priority Queue	Sets whether to enable Priority Queue of the internal switching HUB.	Disable	X	X
1. Disable	Priority Queue is disabled.		X	X
2. Enable	Priority Queue is enabled.		X	X
2. SIP Settings	Sets SIP related data.	-	X	X
1. SIP User	-	-	-	-
1. User ID (Setting prohibited)	-	-	-	-
2. Password (Setting prohibited)	-	-	-	-
3. Extension Number (Setting prohibited)	-	-	-	-
4. Backup Logon (Setting prohibited)	-	Disable	-	-

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. Server Address & URI	Sets IP address or URI of SIP server. NOTE 3	-	X	X
1. 1st Server Address	Sets IP address or fully qualified domain name (FQDN) of 1st ~ 4th SIP server. NOTE 4	0.0.0.0	X	X
2. 2nd Server Address				
3. 3rd Server Address				
4. 4th Server Address				
5. Registrar Destination (Setting prohibited)	-	-	-	-
1. 1st Register Address (Setting prohibited)	-	0.0.0.0	-	-
2. 2nd Register Address (Setting prohibited)				
3. 3rd Register Address (Setting prohibited)				
4. 4th Register Address (Setting prohibited)				
3. Access Mode	Sets access mode for SIP server.	Normal	X	X
1. Normal	Normal access			
2. Remote	Remote access			
4. SIP Server Port	Sets SIP server port [1024-65535].	5060	X	X
1. 1st Server Port	Specifies the destination port number of sending packet to 1st ~ 4th SIP server. NOTE 5			
2. 2nd Server Port				
3. 3rd Server Port				
4. 4th Server Port				
5. Registrar Destination (Setting prohibited)	-			
1. 1st Register Port (Setting prohibited)	-		-	-
2. 2nd Register Port (Setting prohibited)				
3. 3rd Register Port (Setting prohibited)				
4. 4th Register Port (Setting prohibited)				

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
5. Area ID	Sets the area (country) using DT820 Series.			
0. Automatic	The SV9300 side setting is applied when Automatic is selected.			
1. Japan	Selects the area (country) NOTE: Area IDs allowed on the SV9300 can be selected.	Automatic	X	X
2. America (Canada and North America)				
3. Australia				
4. Others				
5. Hong Kong				
6. Malaysia				
7. Singapore				
8. Mexico				
9. Taiwan				
10. New Zealand				
11. Korea				
12. Brazil				
13. China				
14. Thailand				
15. Thailand (EGAT)				
16. Sri Lanka				
17. Germany				
18. Italy				
19. Netherlands				
20. Catalonia				
21. Denmark				
22. Sweden				
23. Poland				
24. Czech				
25. Norway				
26. France				
27. Portugal				
28. Spain				
29. Austria				
30. Belgium				
31. UK				
32. Greece				
33. Switzerland				
34. South Africa				
35. Brazil (EMEA)				

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
6. Extra Function (Setting prohibited)	-	-	-	-
1. Retransmit Interval (Setting prohibited)	-	Standard	-	-
1. Standard	-			
2. 6 Seconds (Setting prohibited)	-			
2. Keeping Call (Setting prohibited) NOTE 6	-	-	-	-
1. Disable (Setting prohibited)	-	Enable	-	-
2. Enable	-			
3. Howler Tone (Setting prohibited) NOTE 7	-	Disable	-	-
1. Disable	-			
2. Enable (Setting prohibited)	-			
4. Random Wait Time	Sets random wait time of when the connection between SV9300 and terminal starts (5-3600 seconds). * Do not change the factory default value of this parameter.	5	X	X
5. Connect Wait Time	Sets the time to wait until the connection between SV9300 and terminal is complete (30-300 seconds). * Do not change the factory default value of this parameter.	30	X	X
7. Encryption	Sets encryption data.	-	X	X
1. Authentication Mode	Selects whether to use authentication.	Disable	X	X
1. Disable	Authentication is not used.			
2. Enable	Authentication is used.			
2. One Time Password	Sets One Time Password.	-	X	X
3. Authentication Status	Displays authentication status.	-	X	X
4. Icon Indication (Setting prohibited) [FW5.1.0.0 or later]	Select whether to indicate an icon on a DT820 terminal during RTP encryption.	Disable	-	-
1. Disable	Icon indication is disabled during RTP encryption.			
2. Enable	Icon indication is enabled during RTP encryption.			
8. NAT Traversal (Setting prohibited)	-	-	-	-
1. NAT Traverse Mode (Setting prohibited)	-	Disable	-	-
1. Disable (Setting prohibited)	-			
2. Dynamic (Setting prohibited)	-			
3. Static (Setting prohibited)	-			

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. Network Area Name (Setting prohibited)	-	-	-	-
3. WAN Settings (Setting prohibited)	-	-	-	-
1. WAN Mate IP Address (Setting prohibited)		0.0.0.0		
2. WAN SIP Mate Port (Setting prohibited)		5060		
3. WAN Self IP Address (Setting prohibited)		0.0.0.0		
3. Maintenance Settings	Sets maintenance related data.	-	X	X
1. Download Menu	Downloads specified file to DT820 Series.	-	X	X
1. Download Files	Specifies download file name for each file type.	-	X	X
1. Boot & Program	Downloads boot and program.	DT820 (6D): ityisipe.tgz DT820 (8LDX): ityisipex.tgz	X	X
2. Config	Downloads configuration file.	config.tgz	X	X
3. Voice File	Downloads voice file.	voice.tgz	X	-
4. Language Pack	Downloads translation data file.	LangPack.tgz	X	X
5. Certificate	Downloads certificate files.	-	X	X
1. Client Certificate	Downloads client certificate file.	clientcert.p12		
2. Root Certificate	Downloads root certificate file.	rootcert.der		
2. Download Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X	X
3. Protocol	Selects FTP or TFTP for file downloading protocol.	FTP	X	X
1. FTP	FTP is used.			
2. TFTP	TFTP is used.			
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.			
4. FTP Settings NOTE 8	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.	-	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X
2. Hard Reset	Resets DT820 Series without data saving.	-	X	X
3. Data Clear	Initializes DT820 Series selecting the initialization mode.	-	X	X
1. Factory Value	Initializes DT820 Series to factory default.	-	X	X
2. Terminal Data	Clears terminal data on DT820 Series.	-	X	X
3. Personal Data	Clears personal data on DT820 Series.	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
4. Adjust	Adjusts LCD, audio, and button kit.	-	X	X
1. Default Audio	Sets audio data.	-	X	X
1. Sidetone Volume	Sets Side Tone volume [0-7].	4	X	X
2. Send Gain	Sets sending gain of handset [0-31].	24	X	X
3. Receive Bias	Sets receiving bias of handset [0-7].	0	X	X
4. Receive Gain	Sets receiving gain of handset [0-31].	22	X	X
5. Talk Characteristics	Sets talk characteristics of handset.	-	X	X
1.Noise Suppression	Sets whether to enable the Noise Suppressor. (Set "1. Disable" in normal operation.)	Disable		
1. Disable	Disables the Noise Suppressor.			
2. Enable	Enables the Noise Suppressor.			
2. Handsfree Mode	Sets a send gain mode of handsfree call.	Retro mode	X	X
1. Retro Mode	Sets to Retro mode.			
2. New Mode	Sets to New mode.			
6. DTMF Send Level	Sets DTMF Send Level [0-48]. (level 0 (maximum) ~ level 48 (minimum))	32		
2. EHS Cable NOTE 9	Selects whether to use the cable for wireless headset connection.	None	X	X
1. None	Wireless headset is not used (instead of it, wired headset is used).			
2. APD-80	Wireless headset with APD-80 is used.			
3. Character Code (Setting prohibited) [FW5.1.0.0 or later]	-	Automatic	-	-
1. Automatic	-		-	-
2. Eastern Europe (Setting prohibited)	-		-	-
4. Headset	Adjust headset	0 dB	X	
1. Send Level	Sets voice sending level of headset.			
1. 0 dB	Not change the sending voice level from default.			
2. -5 dB	Set -5dB lower voice level than default.			
3. -10 dB	Set -10dB lower voice level than default.			
5. Auto config	Selects whether to use auto-config function.	Enable	X	X
1. Disable	Auto config is not used.			
2. Enable	Auto config is used.			
6. Data Maintenance	Sets data maintenance related.	-	X	X
1. Backup	Executes data backup.	-	X	X
1. All Data	Executes backup for all the configuration setting and data. * For encryption, "AllData.tgz.ef" is applied as a file name.	AllData.tgz	X	X
2. Terminal Data	Executes backup for terminal setting and data. * For encryption, "TerminalData.tgz.ef" is applied as a file name.	Terminal Data.tgz	X	X
3. Personal Data	Executes backup for personal setting and data. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. Restore	Executes restoration.	-	X	X
1. All Data	Restores all the configuration setting and data. * For encryption, "AllData.tgz.ef" is applied as a file name.	AllData.tgz	X	X
2. Terminal Data	Restores terminal setting and data. * For encryption, "TerminalData.tgz.ef" is applied as a file name.	Terminal Data.tgz	X	X
3. Personal Data	Restores personal setting and data. * For encryption, "PersonalData.tgz.ef" is applied as a file name.	Personal Data.tgz	X	X
3. Server Address	Enter an IP address of server/URI. * URI input is available for FW5.1.0.0 or later.	0.0.0.0	X	X
4. Protocol	Selects FTP or TFTP for file downloading protocol.			
1. FTP	FTP is used.	FTP	X	X
2. TFTP	TFTP is used.			
3. HTTPS [FW5.1.0.0 or later]	HTTPS is used.			
5. FTP Settings NOTE 10	Sets data for accessing FTP server.	-	X	X
1. User ID	Sets user ID for login to FTP server.	-	X	X
2. Password	Sets password for login to FTP server.	-	X	X
3. Folder	Specifies the directory to save the download file.	-	X	X
7. Simple Test NOTE 11	Changes the terminal to test mode.	-	X	X
1. Call Test	Call Test	-	X	X
2. Call Test Address	Sets an IP address of the opposite terminal to perform the call test.	10.136.20.188	X	X
3. Key Test 1 (Random)	Tests to press the random button.	-	X	X
4. Key Test 2 (Sequential)	Tests to press the sequential button.	-	X	X
5. Loop Back Test	Tests the voice loop back.	-	X	X
1. Handset - Hand Speaker	Tests to speak and receive on the hand speaker.	-	X	X
2. Handset - Speaker	Tests to speak on the handset and receive on the speaker.	-	X	X
3. Headset - Head Speaker	Tests to speak and receive on the head speaker.	-	X	X
4. Headset - Speaker	Tests to speak on the headset and receive on the speaker.	-	X	X
5. MIC - Hand Speaker	Tests to speak on the MIC and receive on the hand speaker.	-	X	X
6. MIC - Head Speaker	Tests to speak on the MIC and receive on the head speaker.	-	X	X
7. MIC - Speaker	Tests to speak on the MIC and receive on the speaker.	-	X	X
6. Speaker Test	Tests the speaker.	-	X	X
1. Japan	A DT for Japan is heard in maximum volume.	-	X	X
2. America	A DT for North America is heard in maximum volume.	-	X	X
7. Receiver Test	Receiver Test for a handset	-	X	X

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
4. Security	Sets security related data.	-	X	X
1. Web Programming	Selects whether to use Web Programming.	Enable	X	X
1. Disable	Web Programming is disabled.			
2. Enable	Web Programming is enabled.			
2. Push Server	Sets XML's Push server related data.	-	X	X
1. Push Server Access	Sets whether to access XML's Push server.	Enable		
1. Disable	Access is disabled.			
2. Enable	Access is enabled.			
3. Enable (All Client) NOTE 12	Access is enabled. Select this item to set more than one XML's Push server.			
2. Client IP Address	IP address of XML's Push server.	0.0.0.0		
1. Client 1	Sets IP address of 1st XML's Push server.			
2. Client 2	Sets IP address of 2nd XML's Push server.			
3. Client 3	Sets IP address of 3rd XML's Push server.			
4. Client 4	Sets IP address of 4th XML's Push server.			
3. Push Server Port	Port number of XML's Push server [82, 1024-65535]. NOTE 1	82		
3. Admin Password	Sets administrator's password. * Be sure to change the default password.	6633222		
4. 802.1X Supplicant	Sets 802.1X authentication related parameter.	-	X	X
1. Supplicant	Sets whether to use supplicant on DT820 Series.	Disable	X	X
1. Disable	Supplicant is disabled on DT820 Series.			
2. Enable	Supplicant is enabled on DT820 Series.			
2. EAP Method	Selects EAP-MD5 or EAP-TLS for the EAP Method.	EAP-MD5	X	X
1. EAP-MD5	EAP-MD5 is used.			
2. EAP-TLS	EAP-TLS is used.			
3. Account	Sets user ID for authentication.	-	X	X
4. Password	Sets password for authentication.	-	X	X
5. EAPOL VLAN Mode	Selects whether to add VLAN tag to EAP packet.	Enable	X	X
1. Disable	VLAN tag is not added to EAP packet.			
2. Enable	VLAN tag is added to EAP packet.			
5. VPN Client NOTE 13 [FW3.0.0.0 to FW3.1.x.x] For FW3.2.0.0 or later, see NOTE 24.	Sets VPN Client related parameter.	-	-	-
1. VPN Mode	Sets whether to enable VPN Mode.	Disable	X	X
1. Disable	VPN Mode is disabled.			
2. Enable	VPN Mode is enabled.			
2. VPN Gateway Address	Sets VPN Gateway address.	0.0.0.0	X	X
3. VPN Network Settings	Sets VPN network related data.	-	X	X
1. VPN IP Address	Sets VPN IP Address.	0.0.0.0		

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
4. IKE Settings	Sets IKE protocol related data.	-		
1. IKE Version NOTE 15	Selects IKE protocol version.	V1	X	X
1. V1 NOTE 16	IKE protocol version is specified to V1.			
2. V2	IKE protocol version is specified to V2.			
2. Authentication Mode	Selects authentication mode.	PSK		
1. PSK NOTE 17	Authentication mode is specified to PSK.			
2. RSA NOTE 18	Authentication mode is specified to RSA.			
3. PSK Setting	Sets the parameter of PSK (up to 64 characters).	-		
4. IKE ID	Sets the parameter of IKE ID (up to 127 characters).	-		
5. SA Validity	Sets the expiration of SA validity when the IPsec connection has been established (3600-86400 seconds).	86400		
5. Child SA Validity	Sets the expiration of Child SA validity when the IPsec connection has been established (3600-86400 seconds).	28800		
6. NAT Traversal	Sets IPsec NAT traversal related data.	-	X	X
1. Keep Alive	Sets keepalive for NAT traversal (0, 1-3600 seconds). NOTE 19	20		
2. NAT Traversal Port	Sets port for NAT traversal (1024-65535).	4500		
7. L2TP Settings	Sets L2TP connection related data.	-	X	X
1. Cert ID	Sets certificate ID for L2TP (up to 127 characters).	-		
2. Cert Password	Sets certificate password for L2TP (up to 127 characters).	-		
6. User Password	Setting for User Password.	Enable	X	X
1. Expiration Date	Sets the expiration of user password (0-90 days). * When "0" is specified, the warning indication will not be indicated.	0	X	X
2. Warning Interval	Sets the interval to regenerate the warning message which prompts a password resetting (1-24 hours).	1	X	X
7. Certificate	Sets certificate related data.	-	X	X
1. Install	Sets install related data of certificate.	-		
1. PKCS#12 Password	Sets PKCS#12 password.	-		
2. Select	Selects the client certificate.	-	X	X
1. 802.1X	802.1X is selected as a client certificate. NOTE 14			
2. VPN NOTE 13	VPN connection is selected as a client certificate. NOTE 14			

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
3. Delete	Sets delete method of certificate.	-		
1. Delete Data	Selects the type of certificate data to be deleted.	-	X	X
1. Client Certificate	Deletes client certificate data.			
2. Root Certificate	Deletes root certificate data.			
3. Expired Certificates	Deletes expired certificate data.			
4. All Certificates	Deletes all certificates data.			
2. Auto Delete Mode	Sets whether to delete the certificate data automatically.	Disable	X	X
1. Disable	Certificate data is not deleted automatically.			
2. Enable	Certificate data is deleted automatically.			
4. Validity	Sets the expiration of certificate validity.	-	X	X
1. Warning Period	Sets the time to display the warning notice for the end of certificate validity (0-90 days). NOTE 20	7	X	X
2. Installation time	Sets the time of when the certificate is installed. * This setting will be applied to authenticate a certificate until the time information from the SV9300 is received.	-	X	X
1. Year	Year (2016-2037)	2016		
2. Month	Month (1-12)	1		
3. Day	Day (1-31)	1		
4. Hour	Hour (0-23)	0		
5. Minute	Minute (0-59)	0		
6. Second	Second (0-59)	0		
9. HTTPS Settings [FW3.1.0.0 or later]	Setting of HTTPS communication.	-	X	X
1. Server Authentication	Setting of a server certificate authentication.	Disable	X	X
1. Disable	Sets the server authentication disabled.			
2. Enable	Sets the server authentication enabled.			
5. Terminal Information	Displays terminal information.	-	X	X
Hardware Version	Displays hardware version of DT820 Series.	-	X	X
MAC Address	Displays MAC address of DT820 Series.	-	X	X
Firmware Version	Displays firmware version of DT820 Series.	-	X	X
6. Application Settings	Sets application data.	-	X	X
1. XML Settings	Specifies XML browser function.	-	X	X
1. Terminal ID	Login number for accessing XML server.	0000	X	X
2. XML Browser	Sets XML browser related data.	-	X	X
1. Home URL	Sets the address of the contents to access when starting up XML browser.	-	X	X
2. Proxy Settings	Sets proxy server related data.	-	X	X
1. Proxy IP	Sets proxy IP address of XML browser.	0.0.0.0	X	X
2. Proxy Port	Sets proxy port of XML browser [1024-65535].	8080	X	X

0. Config (Continued)

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
3. Directory URL	Sets URL of Corporate Phonebook.	-	X	-
3. Service URL	Sets URL of XML application.			
1. 1st Service URL	URL of up to four XML applications can be specified.	-	-	X
2. 2nd Service URL				
3. 3rd Service URL				
4. 4th Service URL				
4. Service URL	Sets URL of XML application.			
1. 1st Service URL	URL of up to four XML applications can be specified.	-	X	-
2. 2nd Service URL				
3. 3rd Service URL				
4. 4th Service URL				
2. IPM Self Port	Sets the port for connecting to IP Phone Manager [1024-65535 (except 17185, 60000)]. NOTE 1	3530	X	X
3. Popup	Sets the popup window display.	-		
1. Popup Mode	Sets the mode of the popup window display.	App Priority	X	X
1. Enable	Sets the popup window display enabled.			
2. App Priority	Sets to depend on a definition of XML tag for Popup window.			
4. Screen Saver URL	Sets reference URL for Screen Saver activation.	-	X	X
5. Menu Key Mode	Changes to menu key mode.	Enable	X	X
1. Disable	Sets the menu key mode disabled.		X	X
2. Enable	Sets the menu key mode enabled.		X	X
7. License Settings	-	-	X	X
1. License Server Settings	-	-	X	X
1. Server Address	Enter the IP address of LMC [0.0.0.0-255.255.255.254].	0.0.0.0	X	X
2. Server Port	Enter the server port of LMC [1024-65535].	6080	X	X
2. License Request	-	-	X	X
1. Gigabit Ethernet	-	-	X	X
1. Disable	Set when the Gigabit Ethernet license is not used.	Disable	X	X
2. Enable	Set when the Gigabit Ethernet license is used.		X	X
2. Line Key	-	-	X	X
1. Disable	Set when the Line Key license is not used.	Disable	X	-
2. Line 16	Set when the 16 Line Key license is used.		X	-
3. Line 32	Set when the 32 Line Key license is used.		X	-
3. Adjust	-	-	X	X
1. License User Prefix	Enter the License User Prefix.	Blank	X	X

NOTE 1: A unique number must be assigned to each of the following ports. For RTP Self Port, 16 ports starting from an assigned port are used.

- RTP Self Port
- SIP Self Port
- Push Server Port
- IPM Self Port

NOTE 2: LLDP time to live = LLDP packet transmission interval × the set value (65535 seconds at most)

NOTE 3: “SIP server” used here refers to the SV9300. The SV9300 acts as SIP server where DT820 Series terminal is registered.

NOTE 4: Follow the IP address setting (CM0B Y=1XX>00) of VOIP port on SV9300.

NOTE 5: Follow the SIP Server port setting (CM0B Y=1XX>32) on SV9300 (example: change 5060 to 5080 when default is set).

NOTE 6: If the firmware version of a terminal is FW3.1.0.0 or later, “2. Keeping Call” menu is shown as follows.

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
2. Keeping Call (Setting prohibited)	-	-	-	-
1. Keeping Call Mode (Setting prohibited)	-	Enable	-	-
1. Disable (Setting prohibited)	-			
2. Enable	-			
2. No RTP Time for H.C. (Setting prohibited)	-	0	-	-

NOTE 7: Howler Tone to be sent depends on the system data setting (CM08>153, CM13 Y=04) of SV9300 (default: To send).

NOTE 8: If the firmware version of a terminal is FW3.1.0.0 or later, this menu becomes “4. Account Settings” (formerly “4. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
4. Account Settings	Account settings of download server (FTP/HTTPS).	-	X	X
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).			
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).			
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).			

NOTE 9: The following table shows the usable headset type depending on the setting of “EHS Cable”.

Setting of EHS Cable	Usable Headsets (Cables)
None	HW510 (A10) HW251 (A10)
APD-80	W710 (APD-80 cable (Savi 710))

NOTE 10: If the firmware version of a terminal is FW3.1.0.0 or later, this menu becomes “5. Account Settings” (formerly “5. FTP Settings”). The setting details are as follows.

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
5. Account Settings	Account settings of backup/restore server (FTP/HTTPS).			
1. User ID	Sets the User ID of FTP/HTTPS server (up to 64 characters).	-	X	X
2. Password	Sets the password of FTP/HTTPS server (up to 64 characters).			
3. Folder	Sets the folder name of FTP/HTTPS server (up to 64 characters).			

NOTE 11: This service is available to perform various tests such as Key Test, LED, LCD, and Call Test.

NOTE 12: Be careful when you specify “3. Enable (All Client)” because security service are disabled for Push when “3. Enable (All Client)” is set.

NOTE 13: For North America and China, this setting is not available. Therefore, this setting cannot be selected on the terminal.

NOTE 14: Up to 15 certificates can be selected.

NOTE 15: The following table shows the supporting authentication mode and IKE version.

X: Available N: Not available

Authentication Mode	IKE Version	
	IKEv1	IKEv2
PSK	X	X
RSA	N	X

NOTE 16: For IKE version “V1”, the supported authentication mode is “PSK” only. Therefore, when “1. V1” is specified, be sure to specify “1. PSK” on “2. Authentication Mode”. When “2. RSA” is specified, a communication error occurs.

NOTE 17: IKE authentication is executed with “PSK” which has been assigned in [0. Config] - [4. Security] - [5. VPN Client] - [4. IKE Settings] - [3. PSK Setting].

NOTE 18: IKE authentication is executed with “RSA” which has been assigned in [0. Config] - [4. Security] - [8. Certificate] - [2. Select] - [2. VPN].

NOTE 19: When “0” is specified, this setting will be disabled.

NOTE 20: When “0” is specified, the warning indication will not be indicated.

NOTE 21: If the firmware version of a terminal is FW3.2.0.0 or later, this menu becomes “5. VPN Settings” (formerly “5. VPN Client”). The setting details are as follows.

Parameter	Description	Default	DT820 (8LDX)	DT820 (6D)
5. VPN Settings [FW3.2.0.0 or later]	Sets VPN.	-	-	-
1. VPN Mode	Sets whether to enable VPN Feature.	Disable	X	X
1. Disable	VPN Mode is disabled.			
2. Enable	VPN Mode is enabled.			
2. VPN Network Settings	Sets VPN network related data.	-	X	X
1. VPN Client Address	Sets the terminal IP address to be specified by VPN center site	0.0.0.0	X	X
2. VPN Gateway Address	Sets VPN Gateway address.	0.0.0.0	X	X
3. L2TP/IPsec Settings	Sets L2TP/IPsec related data.	-	X	X
1. IKE Settings	Sets IKE protocol related data.	-	X	X
1. IKE Version NOTE 15	Selects IKE protocol version.	V1	X	X
1. V1 NOTE 16	IKE protocol version is specified to V1.			
2. V2	IKE protocol version is specified to V2.			
2. Authentication Mode	Selects IKE authentication mode.	PSK	X	X
1. PSK NOTE 17	Authentication mode is specified to PSK.			
2. RSA NOTE 18	Authentication mode is specified to RSA.			
3. PSK Setting	Sets the parameter of PSK (up to 64 characters).	-	X	X
4. IKE ID	Sets the parameter of IKE ID (up to 127 characters).	-	X	X
5. SA Validity	Sets the expiration of SA validity when the IPsec connection has been established (3600-86400 seconds).	86400	X	X
2. Child SA Validity	Sets the expiration of Child SA validity when the IPsec connection has been established (3600-86400 seconds).	28800	X	X
3. NAT Traversal	Sets IPsec NAT traversal related data.	-	X	X
1. Keep Alive	Sets keep alive for NAT traversal (0, 1-3600 seconds). NOTE 19	20		
2. NAT Traversal Port	Sets port for NAT traversal (1024-65535).	4500		
4. Cert ID	Sets authentication ID for L2TP (up to 127 characters).	-	X	X
5. Cert Password	Sets authentication password for L2TP (up to 127 characters).	-	X	X

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