

Panasonic

Programming Manual for Virtual SIP Trunk Card Pure IP-PBX

Model No. **KX-NCP500/KX-NCP1000**

Thank you for purchasing a Panasonic Pure IP-PBX.
Please read this manual carefully before using this product and save this manual for future use.

KX-NCP500/KX-NCP1000: PBMPR Software File Version 1.0000 or later

In this manual, the suffix of each model number (e.g., KX-NCP500**NE**) is omitted unless necessary.

Table of Contents

1	Overview	3
1.1	Information about IP Telephony Service	4
1.2	Important Notice for Subscription and Installation	7
1.2.1	Port Requirements	7
1.2.2	Firewall Requirements	7
1.2.3	Bandwidth Requirements	8
1.3	Specifications	10
2	Preparations for PC Programming	11
2.1	Maintenance Console Operating Instructions	12
2.1.1	Installing and Starting the Maintenance Console	12
2.2	Installing the V-SIPGW16 Card to the PBX	16
3	Programming the V-SIPGW16 Card	17
3.1	Programming the V-SIPGW16 Card—Port Property	18
3.1.1	Main	19
3.1.2	Account	20
3.1.3	Register	21
3.1.4	NAT	23
3.1.5	Option	24
3.1.6	Calling Party	25
3.1.7	Called Party	27
3.1.8	Voice/FAX	27
3.1.9	RTP/RTCP	30
3.1.10	T.38	31
3.1.11	T.38 Option	32
3.1.12	DSP	34
3.1.13	Supplementary Service	35
3.2	Programming the V-SIPGW16 Card—Card Property	37
3.2.1	Card Property	37
3.3	Programming the V-SIPGW16 Card—Shelf Property	40
3.3.1	Main	40
3.3.2	Timer	42
4	Maintenance	45
4.1	Configuration Management	46
4.1.1	Importing/Exporting V-SIPGW16 Provider Profiles	46
4.1.2	V-SIPGW16 Protocol Trace	47
	Index.....	49

Section 1

Overview

This section provides an overview of the V-SIPGW16 card.

1.1 Information about IP Telephony Service

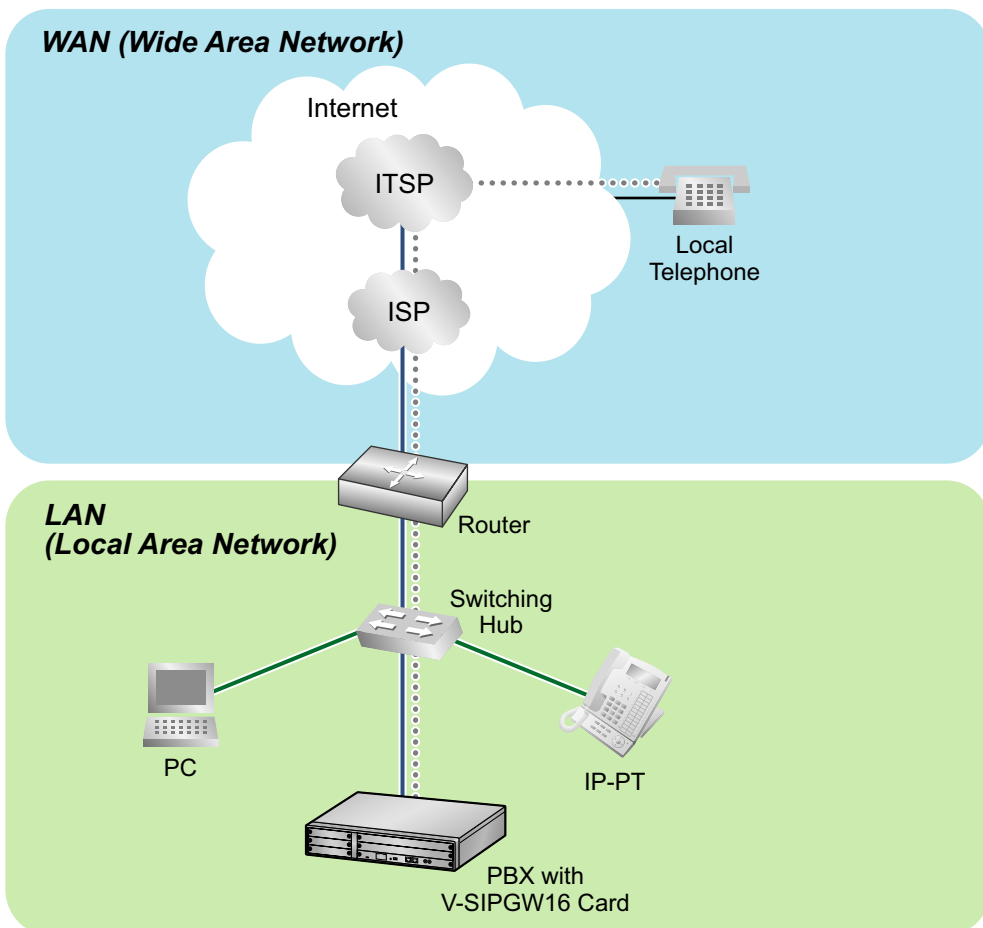
The Virtual 16-Channel SIP Trunk Card (V-SIPGW16) is a virtual trunk card which is designed to be easily integrated into an Internet Telephony Service provided by an ITSP (Internet Telephony Service Provider).

As a major SIP Provider, an ITSP provides its telephony service partly through the conventional telephone network (e.g., ISDN and Mobile), which is fee-based. An ISP (Internet Service Provider), another major SIP Provider, does not provide telephone connection itself. However, providing its users with Internet access, an ISP provides voice communication on the Internet for free. In this way, with VoIP technology based on the SIP protocol, the cost of voice communication can be much cheaper than conventional telephone networks.

The maximum number of V-SIPGW16 cards that can be installed in the virtual trunk slots of the PBX is one for the KX-NCP500 and two for the KX-NCP1000. The channel capacity of the card allows KX-NCP500 users to connect to up to 8 different ISP/ITSPs and allows KX-NCP1000 users to connect to up to 32 different ISP/ITSPs.

V-SIPGW16 Connection Outline

The following diagram illustrates a simple VoIP network connecting the V-SIPGW16 card to the Internet.



Requirements for Internet Telephony Service

- You need to subscribe with an ISP for Internet connection.
- You need to subscribe with an ITSP for telephone connection. The ISP and ITSP may be part of the same company.

Note

- VoIP communication using the V-SIPGW16 card may deteriorate depending on the ITSP being used.
- VoIP communication using the V-SIPGW16 card may deteriorate depending on the network conditions.

DNS (Domain Name System)

A DNS server normally provides the name resolution service for your computer. As domain names are alphabetic, they are easier to remember. The Internet, however, is based on IP addresses. Therefore, every time a domain name is used, a DNS server must translate the name into the corresponding IP address, and vice versa. For example, the domain name *www.example.com* may be translated to *192.0.34.166*. If one DNS server does not know how to translate a particular domain name, it asks another one, and so on, until the correct IP address is returned.

NAT (Network Address Translation) Traversal

When NAT/NAPT (Network Address Port Translation) is enabled, the router translates a local IP address from the PBX into a global IP address. However, the router with NAT enabled does not translate local IP addresses stored in SIP messages into global IP addresses.

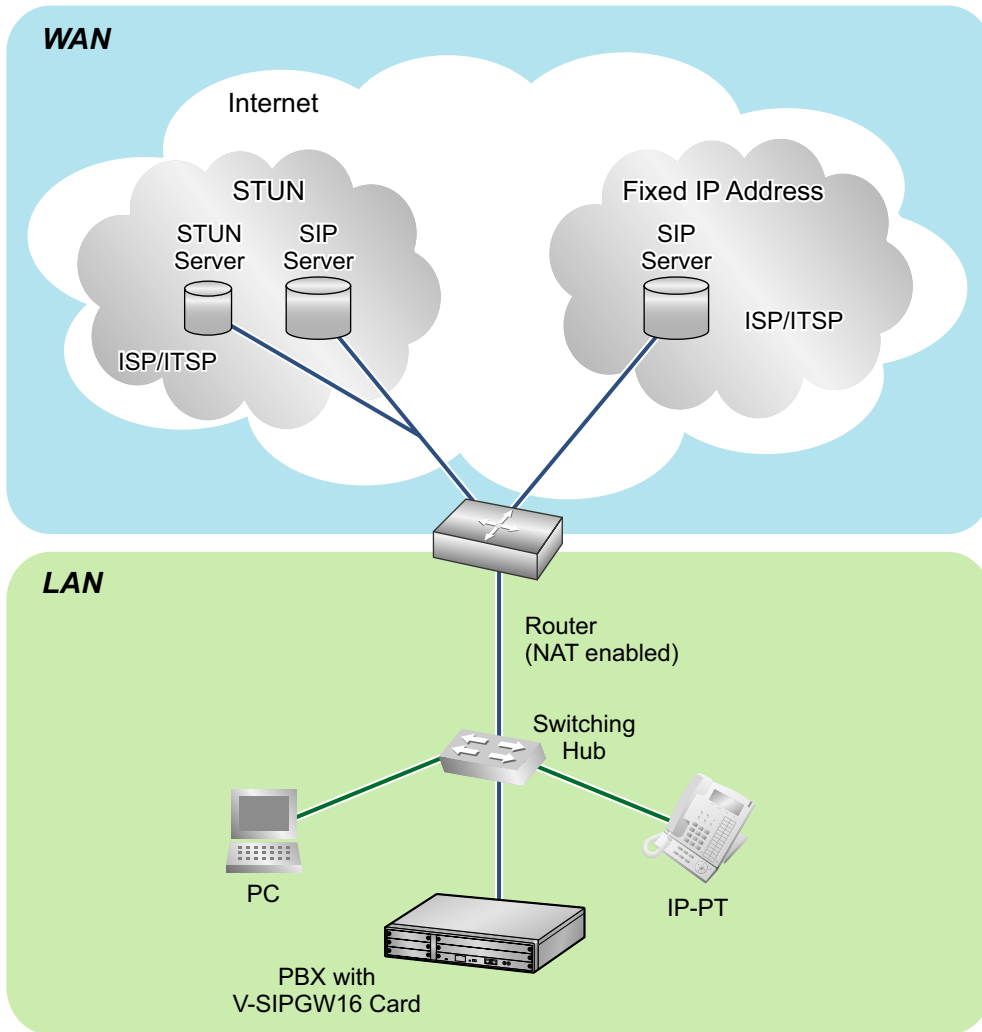
Therefore, the address which the SIP Server recognises as the destination IP address to reply to is actually the local IP address of the PBX, not the global IP address of the router. Therefore, if the SIP server receives a SIP message from the PBX and sends a message back to the PBX using the address stored in the SIP message, the packet information will not reach the PBX.

STUN Servers function to solve the global IP address problem under certain NAT conditions, for example, in case of full duplex communication. A STUN Server, used alongside the SIP Server, finds out the global IP address of the router with NAT enabled. With the STUN feature enabled, the packet information sent by the SIP Server is able to "traverse" NAT and reach the PBX.

The settings can be configured to specify whether to enable the NAT Traversal feature for each ISP/ITSP. In addition, the NAT Traversal method can be selected from "STUN" and "Fixed IP Address" (refer to 3.3.1 Main, NAT Traversal).

The V-SIPGW16 card may require the NAT Traversal feature to be enabled to connect to the WAN via a router. The following diagram illustrates how VoIP communication is enabled between the V-SIPGW16 card and the SIP Server (SIP Receiver) via a router with NAT enabled.

1.1 Information about IP Telephony Service



Note

- If an ISP/ITSP uses a device such as SBC (Session Border Controller), you may not have to enable the NAT Traversal feature.
- A STUN Server is supplied by an ISP/ITSP, and not included with the PBX.

1.2 Important Notice for Subscription and Installation

1.2.1 Port Requirements

Required Ports for Each Channel

When configuring a router with NAT enabled, you need to secure a certain number of ports for each SIP and RTP/RTCP channel. For RTP/RTCP, the number of required ports is double the number of activated SIP trunks (Ch). For SIP signalling, the number of required ports is always one regardless of the number of activated SIP trunks (Ch).

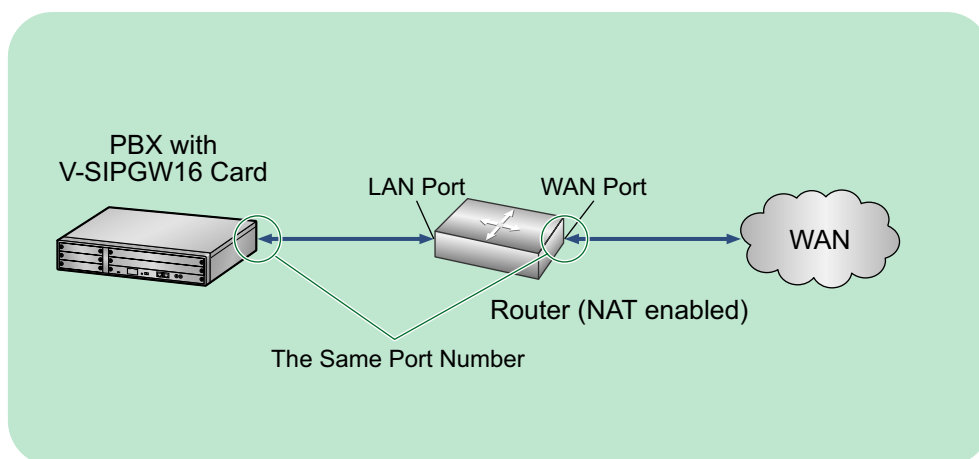
<Example>

If 4 SIP trunk channels are activated, you need the following number of ports:

Protocol Type	Required Port
RTP	4
RTCP	4
SIP	1
Total	9

Requirements for Port Number

Routers with NAT enabled do not translate port numbers stored in SIP messages. Therefore, it is necessary to configure the router with NAT enabled so that the PBX and the WAN port of the router have the same port number. Configuration is also required so that the router with NAT enabled does not change the port number. For details about configuration, refer to the documentation for your router.



1.2.2 Firewall Requirements

If the VoIP network contains a firewall, you must configure the firewall to allow VoIP packets to pass through certain ports of the ports listed below without being blocked by filtering. The ports for which you need to configure the firewall may vary depending on the network conditions.

For more information, consult your network administrator.

1.2.3 Bandwidth Requirements

[IP Packets to V-SIPGW16 Card]

Port	TCP/UDP	Default Port No.
SIP Client Port	UDP	35060
STUN Client Port	UDP	33478
NAT Voice (RTP) UDP Port	UDP	16000 to N
DNS (Client) Port	UDP	30053

[IP Packets from V-SIPGW16 Card]

Port	TCP/UDP	Default Port No.
SIP Server Port	UDP	5060
Registrar Server Port	UDP	5060
STUN Server Port	UDP	3478
Manual DNS Server Port	UDP	53

Router Requirements

- **Port Forwarding:**
When a V-SIPGW16 card is located under a NAT router, and when incoming packets are routed to a SIP Client port or NAT Voice (RTP) UDP port indicated in the **[IP Packets to V-SIPGW16 Card]** table above, it may be required to set the NAT router so that it forwards the incoming packets to the IP address of the V-SIPGW16 card.
- **SIP-NAT Feature:**
When a V-SIPGW16 card is located under a NAT router that supports the SIP-NAT feature^{*1}, it is recommended to disable this feature.

^{*1} When NAT is enabled, the router translates the IP address stored in the IP header and the port number stored in the UDP header. When SIP-NAT is enabled, the router also translates the IP address and port number stored in SIP messages.

1.2.3 Bandwidth Requirements

When using the V-SIPGW16 card, you must ensure that the WAN has enough bandwidth to support VoIP communications. Refer to the table below and ensure that the sum of the required bandwidth for each channel is smaller than the amount the WAN (e.g., ADSL network) can provide.

Note that the amount in the table is only a guide. Subscribe to a network that has enough bandwidth. If the amount of bandwidth required for VoIP communications is larger than what the network can accommodate, speech quality will be compromised.

Required Bandwidth for Each Channel

The required bandwidth depends on what combination of CODECs and packet sending interval is used. Keep in mind the following points about the type of CODEC and packet sending interval, in terms of the speech quality:

- The speech quality of the CODECs varies as follows: G.711 (High), G.729 (Low)
- The shorter the packet sending interval, the higher the speech quality.
- The higher the speech quality the V-SIPGW16 card provides, the more bandwidth the WAN requires.

CODEC	Packet Sending Interval					
	10 ms	20 ms	30 ms	40 ms	50 ms	60 ms
G.711	110.4 kbps	87.2 kbps	79.5 kbps	75.6 kbps	73.3 kbps	71.7 kbps
G.729	54.4 kbps	31.2 kbps	23.5 kbps	19.6 kbps	17.3 kbps	15.7 kbps

1.3 Specifications

For details about the RFCs and protocols for the V-SIPGW16 card, refer to the following specifications.

Items	Specification
SIP RFCs	RFC3261 (UDP only)
	RFC3262 (PRACK)
	RFC3264 (Offer/Answer)
	RFC3311 (UPDATE)
	RFC3581 (Symmetric Response Routing/rport)
	RFC4028 (Session Timer)
CODECs	G.711 (a-law and μ -law)
	G.729A
Voice Options	Echo Cancellation (48 ms)
	Jitter Buffer (200 ms)
	VAD (Voice Activity Detection)
	PLC (Packet Loss Concealment)
DTMF Relay	Inband/Outband (RFC2833)/Outband (INFO)
Fax Relay	G.711 Inband/T.38
Protocol/Function	RTP
	RTCP
	DNS (A/SRV)
	NAT Traversal (STUN)
	QoS (ToS field setting in IP header of RTP/RTCP)

Section 2

Preparations for PC Programming

This section serves as reference operating instructions when using the Maintenance Console software to programme the V-SIPGW16 card.

2.1 Maintenance Console Operating Instructions

2.1.1 Installing and Starting the Maintenance Console

System programming, diagnosis and administration for the V-SIPGW16 card can be performed with a PC using the Maintenance Console. This section describes how to install and start the Maintenance Console.

System Requirements

Required Operating System

- Microsoft® Windows® XP or Windows Vista® Business

Minimum Hardware Requirements

- CPU: 800 MHz Intel® Celeron® microprocessor
- HDD: 100 MB of available hard disk space
- RAM: 128 MB of available RAM

Recommended Display Settings

- Screen resolution: XGA (1024 × 768)
- DPI setting: Normal size (96 DPI)

Hardware Requirements

- Make sure that the PBX, IPCMPR card, and DSP card are installed.
- Make sure that the PBX is fully connected to your PC.

Note

For details about installation, refer to "Section 3 Installation" in the Installation Manual.

Trademarks

- Microsoft, Windows and Windows Vista are either registered trademarks or trademarks of Microsoft Corporation in the United States and/or other countries.
- Intel and Celeron are trademarks of Intel Corporation in the U.S. and other countries.
- All other trademarks identified herein are the property of their respective owners.
- Microsoft product screen shot(s) reprinted with permission from Microsoft Corporation.

Copyright for MD5

This software uses the Source Code of RSA Data Security, Inc. described in the RFC1321 (MD5Message-Digest Algorithm).

Copyright (C) 1991-2, RSA Data Security, Inc. Created 1991. All rights reserved.

Licence to copy and use this software is granted provided that it is identified as the "RSA Data Security, Inc. MD5 Message-Digest Algorithm" in all material mentioning or referencing this software or this function.

Licence is also granted to make and use derivative works provided that such works are identified as "derived from the RSA Data Security, Inc. MD5 Message-Digest Algorithm" in all material mentioning or referencing the derived work.

RSA Data Security, Inc. makes no representations concerning either the merchantability of this software or the suitability of this software for any particular purpose. It is provided "as is" without express or implied warranty of any kind.

These notices must be retained in any copies of any part of this documentation and/or software.

Installing the Maintenance Console

Note

- Make sure to install and use the latest version of the Maintenance Console.
 - To install or uninstall the software on a PC running Windows XP Professional, you must be logged in as a user in either the "Administrators" or "Power Users" group.
 - To install or uninstall the software on a PC running Windows Vista Business, you must be logged in as a user in the "Administrators" group.
 - Product specifications, including text displayed by the software, are subject to change without notice.
1. Copy the setup file of the Maintenance Console to your PC.
 2. Double-click the setup file to run the installer.
 3. Follow the on-screen instructions provided by the installation wizard.

Starting the Maintenance Console and Assigning the Basic Items (Quick Setup)

When you start the Maintenance Console with the Installer Level Programmer Code and connect to the PBX for the first time after initialisation (with the factory default setting), Quick Setup will launch automatically. During Quick Setup, you will set up the basic items. For details about the basic items, refer to "2.3.4 Quick Setup" in the Feature Guide.

1. Connect the PC to the PBX with an Ethernet straight cable or RS-232C cross cable.
2. Start the Maintenance Console from the Start menu.
3. "Information before programming" appears.
 - a. Carefully read this important additional information, which includes updates to this and other manuals.
 - b. Click **OK** to close this window.
4.
 - a. Enter the Installer Level Programmer Code (default: **INSTALLER**).
 - b. Click **OK**.
5. Click **Connect**.
6.
 - a. Select **KX-NCP500/1000** from **PBX Model**.
 - b. Select the **LAN** or **RS-232C** tab, depending on the type of PC connection with the PBX.
 - c. Specify the settings as required.

Note

When connecting to the PBX for the first time selecting **LAN**, the **IP Address** and **Port Number** must be set to **192.168.0.101** and **35300** respectively.

- d. Enter the system password for installer (default: **1234**).
- e. Click **Connect**.

7. When country/area data do not match:

- a. Click **OK** to replace the country/area data of the PBX. Replacement may take several minutes to complete.
- b. Follow the procedure described in Section "3.16.1 Starting the PBX" in the Installation Manual and restart the PBX.
- c. Repeat step **5** to reconnect the Maintenance Console to the PBX.

8. Follow the instructions of the Quick Setup wizard for the basic items in Quick Setup—Step 1 to 3.

9. In Quick Setup—Step 4, the IP addressing information for the IPCMPR card can be assigned automatically through a DHCP server or entered manually.

When using a DHCP server:

- a. Select **Enable** for the **DHCP Client** setting.
- b. Click **Apply**.

Note

The boxes will turn grey and the IP addresses will be assigned automatically after the PBX is reset.

When not using a DHCP server:

- a. Select **Disable** for the **DHCP Client** setting.
- b. In the **IP Address for IPCMPR Card** box, type the IP address of the IPCMPR card.¹
- c. In the **IP Address for VoIP-DSP** box, type the IP address of the DSP16 or DSP64 card.²
- d. In the **Subnet Mask** box, type the subnet mask address of the network.³
- e. In the **Default Gateway** box, type the IP address of the default gateway.⁴
- f. Click **Apply**.

When using a DHCP server to obtain a DNS server IP address automatically:

- a. Select **DHCP** for the **DNS Server Address Method** setting.
- b. Click **Apply**.

Note

The boxes will turn grey and the IP addresses will be assigned automatically after the PBX is reset.

When not using a DHCP server to obtain a DNS server IP address automatically:

- a. Select **Manual** for the **DNS Server Address Method** setting.
- b. In the **Manual Preferred DNS Server IP Address** box, type the IP address of the preferred DNS server.
- c. In the **Manual Alternate DNS Server IP Address** box, type the IP address of the alternate DNS server.
- d. Click **Apply**.

The system menu appears. You may now begin programming the PBX.

Notice

- Do not change the IP addresses of the IPCMPR and DSP cards once IP telephones are registered to the PBX using these IP addresses.
The IP telephones will not operate properly if these IP addresses are changed.
- A DHCP server must be able to use a "client identifier" option specified by RFC 2131.
- The PBX will not start properly if the IP addresses cannot be assigned automatically by the DHCP server when DHCP Client is set to Enable. In this case, you need to consult your network administrator because the DHCP server on your network may not be running or a network failure may have occurred. If the DHCP server cannot be available, change the DHCP Client setting to Disable and set fixed IP addresses, then restart the PBX.
To change the DHCP Client setting, connect the PC with an RS-232C cross cable or Ethernet straight cable. When connecting the PC with an Ethernet straight cable, make sure the PBX is disconnected from the LAN and then connect the PC with an Ethernet straight cable using 192.168.0.101 for the IP address of the IPCMPR card.

*1 Valid IP address range: "1.0.0.0" to "223.255.255.255"

*2 Valid IP address range: "1.0.0.0" to "223.255.255.255"

*3 Valid subnet mask address range: "0-255.0-255.0-255" (except 0.0.0.0 and 255.255.255.255)

*4 Valid IP address range: "1.0.0.0" to "223.255.255.255"

2.2 Installing the V-SIPGW16 Card to the PBX

1.
 - a. Under **Configuration**, click **Slot**.
 - b. Move the mouse pointer over the PBX image of **IPCMPR Virtual Slot** at the top of the screen.
 - c. Click **Select Shelf**.
2.
 - a. Click on the name of the **V-SIPGW16** card. An image of the card will be displayed.
 - b. Drag the image of the card to a **Trunk** slot and release it. The card will move into the slot space.
3. Click **Yes** to confirm.

Section 3

Programming the V-SIPGW16 Card

This section serves as reference operating instructions when using the Maintenance Console software to programme the V-SIPGW16 card.

3.1 Programming the V-SIPGW16 Card—Port Property

Various settings can be programmed for each virtual SIP gateway port. To change the status of virtual SIP gateway ports, click **Command**.

Accessing Port Properties




1. Under **Configuration**, click **Slot**.
2. Move the mouse pointer over the PBX image of **IPCMPR Virtual Slot** at the top of the screen.
3. Click **Select Shelf**.
4. Move the mouse pointer over the **V-SIPGW16** card.
5. Click **Port Property**.

Programming Port Property

Some of the parameters can be automatically programmed by selecting the desired SIP provider for each virtual SIP gateway port.

Icon Description

In this section, the following icons show the different parameter type.

Icon	Description for Parameter Type
	Programmed automatically by selecting the desired SIP provider in Select Provider .
	Manual programming is compulsory.
	Manual programming is compulsory under certain conditions.
Not marked	Manual programming is optional.

Automatic Programming

Some of the parameters on this screen can be automatically programmed by selecting the desired SIP provider for each virtual SIP gateway port. A maximum of 8 SIP providers can be programmed for the KX-NCP500 and a maximum of 32 SIP providers can be programmed for the KX-NCP1000. A different SIP provider can be assigned to each virtual SIP gateway port.

Follow the steps below to configure a SIP provider.

1. Click **Select Provider**.
A dialogue box will appear. Available virtual SIP gateway port numbers are displayed in the list.
2. From the **Provider** menu, select the desired SIP provider.
3. Highlight the desired port numbers or click **Select All** to select all the virtual SIP gateway port numbers to be assigned to the SIP provider selected in step 2.
4. Click **Execute**.
5. Click **OK**.
6. Click **Apply**.
Appropriate setting values designated by the SIP provider will be set in the parameters for the virtual SIP gateway ports.

Manual Programming

Follow the steps below to programme the parameters which are not automatically programmed by selecting a provider.

1. Click a desired tab.
2. Enter information or select from the list for each parameter.

3.1.1 Main

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

Channel Attribute

Specifies the channel property of each port to enable several sessions to be performed for one subscription with the SIP provider.

Value Range

Basic channel: The subscriber channel that is assigned the SIP registration information.

Additional channel for ChN (N=1–16): The subordinative channel that uses the same registration information as a Basic channel for SIP sessions. Select the Basic channel number to which the Additional channel belongs.

Not Used: The channel is not in use.

★ Provider Name

Specifies the name of the SIP provider.

Value Range

Max. 20 characters

★ SIP Server Location—Name

Specifies the domain name of the SIP proxy server.

Value Range

Max. 100 characters

☉ SIP Server Location—IP Address

Specifies the IP address of the SIP proxy server. This setting is compulsory when not using the DNS server.

Value Range

1.0.0.0–223.255.255.255

★ SIP Server Port Number

Specifies the port number of the SIP proxy server.

Value Range

1–65535

★ SIP Service Domain

Specifies the domain name provided by the SIP provider.

Value Range

Max. 100 characters

Subscriber Number

Specifies the number used as the CLIP number.

Value Range

Max. 16 digits (consisting of 0–9, *, and #)

3.1.2 Account

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

⊙ User Name

Specifies the user name (SIP Account) provided by the SIP provider.

Value Range

Max. 64 characters

⊙ Authentication ID

Specifies the authentication ID required for registration with the SIP server.

Value Range

Max. 64 characters

⊙ Authentication Password

Specifies the authentication password used for registration with the SIP provider.

Value Range

Max. 32 characters

3.1.3 Register

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

3.1.3 Register

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

★ **Register Ability**

Specifies whether to send the REGISTER message to the SIP server.

Value Range

Disable, Enable

★ **Register Sending Interval**

Specifies the maximum length of interval time after which the PBX sends the REGISTER message.

Value Range

10–86400 s

Un-Register Ability when port INS

Specifies whether to unregister the previous registration and send the REGISTER message to the SIP server when the port status is set back to INS.

Value Range

Disable, Enable

★ **Registrar Server—Name**

Specifies the domain name of the SIP registrar server.

Value Range

Max. 100 characters

⦿ **Registrar Server—IP Address**

Specifies the IP address of the SIP registrar server. This setting is compulsory when a register IP address is provided.

Value Range

1.0.0.0–223.255.255.255

★ **Registrar Server Port Number**

Specifies the port number of the SIP registrar server.

Value Range

1–65535

3.1.4 NAT

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

★ STUN Server—Name

Specifies the domain name of a STUN server.

Value Range

Max. 100 characters

☉ STUN Server—IP Address

Specifies the IP address of the STUN server. This setting is compulsory when the STUN method is selected and a DNS server is not used.

Value Range

1.0.0.0–223.255.255.255

★ STUN Server Port Number

Specifies the port number of the STUN server.

Value Range

1–65535

3.1.5 Option

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

★ Session Refresh (Session Timer)—Session Timer Ability

Enables the PBX to periodically refresh SIP sessions by sending repeated requests.

Value Range

Disable: This feature is not activated.

Enable (Active): Activates this feature only if the other device supports the feature.

Enable (Passive): Activates this feature only when requested by the other device.

★ Session Refresh (Session Timer)—Session Expire Timer

Specifies the length of time that the PBX waits before terminating SIP sessions when no reply to the repeated requests is received.

Value Range

90–3600 s

★ Session Refresh (Session Timer)—Refresh Method

Specifies the type of request that the PBX sends to periodically refresh SIP sessions.

Value Range

UPDATE, re-INVITE

Proxy-Require Option

Specifies option tags in a Proxy-Require header field so that the SIP server is notified that the client is behind a router with NAT enabled and firewall.

Value Range

Max. 100 characters

3.1.6 Calling Party

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

★ Header Type

Specifies the header of the SIP message in which the caller information is stored.

Value Range

From Header, P-Preferred-Identity Header

★ From Header—User Part

Specifies the value to be stored in the username part of the SIP-URI of the From header.

Value Range

User Name, Authentication ID, PBX-CLIP

From Header—SIP-URI

Specifies the complete SIP-URI address of the From header. The configuration in **From Header—User Part** will be invalid if this parameter is set.

Value Range

Max. 100 characters

★ P-Preferred-Identity Header—User Part

Specifies the value to be stored in the username part of the SIP-URI of the P-Preferred-Identity header.

Value Range

User Name, Authentication ID, PBX-CLIP

P-Preferred-Identity Header—SIP-URI

Specifies the complete SIP-URI address of the P-Preferred-Identity header. The configuration in **P-Preferred-Identity Header—User Part** will be invalid if this parameter is set.

Value Range

Max. 100 characters

★ Number Format

Selects the format of the CLIP number to be sent to the called party.

Value Range

International, +International, National

⦿ Remove Digit

Specifies the number of leading digits of the CLIP number to be removed. This setting is compulsory when **PBX-CLIP** is selected in **From Header—User Part** or **P-Preferred-Identity Header—User Part**.

Value Range

0–32

⦿ Additional Dial

Specifies the number to be added to the CLIP number in the place of the removed digits. This setting is compulsory when **PBX-CLIP** is selected in **From Header—User Part** or **P-Preferred-Identity Header—User Part**.

Value Range

Max. 20 digits (containing of 0–9, *, and #)

3.1.7 Called Party

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

★ Number Format

Specifies the format of an incoming call number.

Value Range

International, +International, National

★ Type

Specifies the header of the incoming SIP message in which the dialled number is stored.

Value Range

Request-URI, To header

3.1.8 Voice/FAX

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

IP Codec Priority—1st, 2nd, 3rd

Specifies the priority of the CODECs to be used. **None** is only available for 2nd and 3rd priorities.

Value Range

G.711A, G.711 μ , G.729A, None

Packet Sampling Time (G.711A)

Specifies the time interval between measurements (samples) of sound data during a conversation. The smaller this number, the higher the quality of the transmitted sound.

Value Range

10 ms, 20 ms, 30 ms, 40 ms, 50 ms, 60 ms

Packet Sampling Time (G.711 μ)

Specifies the time interval between measurements (samples) of sound data during a conversation. The smaller this number, the higher the quality of the transmitted sound.

Value Range

10 ms, 20 ms, 30 ms, 40 ms, 50 ms, 60 ms

Packet Sampling Time (G.729A)

Specifies the time interval between measurements (samples) of sound data during a conversation. The smaller this number, the higher the quality of the transmitted sound.

Value Range

10 ms, 20 ms, 30 ms, 40 ms, 50 ms, 60 ms

Voice Activity Detection for G.711

Enables the use of the Voice Activity Detection feature for the G.711 CODEC. This feature conserves bandwidth by detecting silent periods during a call and suppressing the packets of silence from being sent to the network.

Value Range

Disable, Enable

FAX Sending Method

Specifies the method of transporting the fax signal.

Value Range

G.711 Inband, T.38

Maximum Bit Rate

Specifies the maximum bit rate of the fax signal.

Value Range

No Speed Limit, 2400 bps, 4800 bps, 7200 bps, 9600 bps, 12000 bps, 14400 bps

FAX Detection Ability

Enables the use of the FAX Detection Ability feature. This feature enables end-to-end fax signal relay when the other party prefers a CODEC other than G.711. This feature functions only if the other party supports G.711.

Value Range

Disable, Enable

★ DTMF

Specifies the method to transport DTMF tones.

Value Range

Inband, Outband (RFC2833), Outband (SIP INFO)

Payload Type

Specifies the payload type of RFC2833 for DTMF tones.

Note

Manual programming is required if **DTMF** is set to **Outband (RFC2833)**.

Value Range

96–127

3.1.9 RTP/RTCP

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

RTP QoS Ability

Specifies the type of value to be stored in the ToS field in the IP header.

Value Range

ToS, DSCP, HEX

RTP QoS-ToS Priority

Specifies the priority level in the ToS field.

Value Range

0–7

RTP QoS-ToS Type

Specifies the ToS type in the ToS field.

Value Range

Normal, Monetary Cost, Reliability, Throughput, Delay

RTP QoS-DSCP

Specifies the value in the ToS field by a DSCP for DiffServ.

Value Range

0–63

RTP QoS-HEX

Specifies the value in the ToS field by a hexadecimal number.

Value Range

00–FF

RTCP Packet Sending Ability

Specifies whether to enable each port to send RTCP packets.

Value Range

Disable, Enable

RTCP Packet Interval

Specifies the interval time until the next RTCP packet is sent.

Value Range

5–60 s

3.1.10 T.38**Slot**

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

T.38 FAX Max Datagram

Specifies the maximum datagram size when using the T.38 protocol.

Value Range

272–512

T.38 FAX UDPTL Error Correction – Redundancy

Specifies whether to enable the redundancy feature when using the T.38 protocol.

Value Range

Disable, Enable

T.38 FAX UDPTL Redundancy count for T.30 messages

Specifies the redundancy count for T.30 messages when using the T.38 protocol.

Value Range

0–7

T.38 FAX UDPTL Redundancy count for data

Specifies the redundancy count for data when using the T.38 protocol.

Value Range

0–3

T.38 FAX Rate Management Method

Specifies the rate management method when using the T.38 protocol.

Value Range

Transferred TCF, Local TCF

3.1.11 T.38 Option

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

T.38 FAX QoS Available

Specifies the type of value to be stored in the ToS field in the IP header when using the T.38 protocol.

Value Range

ToS, DSCP, HEX

T.38 FAX QoS-ToS Priority

Specifies the priority level in the ToS field when using the T.38 protocol.

Value Range

0–7

T.38 FAX QoS-ToS Type

Specifies the ToS type in the ToS field when using the T.38 protocol.

Value Range

Normal, Monetary Cost, Reliability, Throughput, Delay

☉ T.38 FAX QoS-DSCP

Specifies the value in the ToS field by a DSCP for DiffServ when using the T.38 protocol. This setting is compulsory when **DSCP** is selected in **T.38 FAX QoS Available**.

Value Range

0–63

☉ T.38 FAX QoS-HEX

Specifies the value in the ToS field by a hexadecimal number when using the T.38 protocol. This setting is compulsory when **HEX** is selected in **T.38 FAX QoS Available**.

Value Range

00–FF

3.1.12 DSP

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

Echo Canceller Ability

Specifies the echo canceller ability time.

Value Range

OFF, 48 ms, 128 ms

DSP Digital Gain (Down)

Specifies the DSP Digital Gain for the down voice path.

Value Range

-14–6 dB

DSP Digital Gain (Up)

Specifies the DSP Digital Gain for the up voice path.

Value Range

-14–6 dB

EC Gain

Specifies the Echo Canceller Gain.

Value Range

-14–6 dB

NLP Setting

Specifies the NLP (Non-Linear Processor) setting to control echo sound quality.

Value Range

Disable, Weak, Normal, Strong

3.1.13 Supplementary Service

Slot

Indicates the slot position (reference only).

Value Range

Slot number

Port

Indicates the port number (reference only).

Value Range

Port number

Connection

Indicates the port status.

This column offers two ways to open the screen to select the port command:

- Click the desired cell in the column.
- Select the desired cell in the column, and then click **Command**.

Value Range

INS: The port is in service.

OUS: The port is out of service.

Fault: The port is not communicating with the network.

CLIR

Specifies whether to allow restriction of the display of the CLIP number on the called party's telephone when making a SIP trunk call.

Value Range

Yes, No

★ CNIP (Send)

Specifies whether to send the caller's name to be displayed on the called party's telephone when making a SIP trunk call.

Value Range

Yes, No

★ CNIP (Receive)

Specifies whether to receive the caller's name to be displayed on the called party's telephone when receiving a SIP trunk call.

Value Range

Yes, No

3.2 Programming the V-SIPGW16 Card—Card Property

3.2.1 Card Property

The properties of the V-SIPGW card can be specified. To programme the parameters, click **Common Settings**, and set the desired value.


Accessing Card Properties

1. Under **Configuration**, click **Slot**.
2. Move the mouse pointer over the PBX image of **IPCMPR Virtual Slot** at the top of the screen.
3. Click **Select Shelf**.
4. Move the mouse pointer over the **V-SIPGW16** card.
5. Click **Card Property**.

Programming Card Property

Icon Description

In this section, the following icons show the different parameter type.

Icon	Description for Parameter Type
	Manual programming is compulsory under certain conditions.
Not marked	Manual programming is optional.

DNS Server IP Address Method

Specifies the method of IP address assignment for a DNS server. When selecting **DHCP**, the DNS server IP address can be assigned automatically by a DHCP server. When selecting **Manual**, the DNS server IP address must be assigned manually.

Value Range

DHCP, Manual

DHCP Preferred DNS Server IP Address

Indicates the IP address of the preferred DNS server assigned automatically by the DHCP server (reference only).

Value Range

1.0.0.0–223.255.255.255

DHCP Alternate DNS Server IP Address

Indicates the IP address of the alternate DNS server assigned automatically by the DHCP server (reference only).

Value Range

1.0.0.0–223.255.255.255

☉ Manual Preferred DNS Server IP Address

Specifies the IP address of the preferred DNS server manually. This setting is compulsory if a DNS server IP address is not assigned automatically.

Value Range

1.0.0.0–223.255.255.255

Manual Alternate DNS Server IP Address

Specifies the IP address of the alternate DNS server manually.

Note

If **Manual Preferred DNS Server IP Address** is programmed, this setting is optional. However, if possible, it is recommended to specify **Manual Alternate DNS Server IP Address** as well.

Value Range

1.0.0.0–223.255.255.255

Manual DNS Server Port Number

Specifies the port number of the DNS server.

Value Range

1–65535

DNS Port Number

Specifies the port number of the PBX used for communication with the DNS server.

Value Range

1–65535

DNS Retry Interval Timer

Specifies the length of time until a query is retried when no response is received from the DNS server.

Value Range

1–10 s

DNS Retry Counter

Specifies the number of times that a query is retried when no response is received from the DNS server.

Value Range

1–10

DNS SRV Record Resolve Ability

Specifies whether to request that the DNS server translates domain names into IP addresses using the DNS SRV record.

Value Range

Disable, Enable

3.3 Programming the V-SIPGW16 Card—Shelf Property

Common settings for all of the V-SIPGW cards installed in the IPCMPR Virtual Shelf can be programmed.


Accessing Shelf Properties

1. Under **Configuration**, click **Slot**.
2. Move the mouse pointer over the PBX image of **IPCMPR Virtual Slot** at the top of the screen.
3. Click **Select Shelf**.
4. Move the mouse pointer over the **V-SIPGW16** card.
5. Click **Shelf Property**.

Programming Shelf Property

Icon Description

In this section, the following icons show the different parameter type.

Icon	Description for Parameter Type
	Manual programming is compulsory under certain conditions.
Not marked	Manual programming is optional.

3.3.1 Main

SIP Client Port Number

Specifies the port number of the PBX used for communications with the SIP server.

Value Range

1–65535

NAT Traversal

Specifies the NAT traversal method.

- STUN: A STUN Server, used alongside the SIP Server, finds out the global IP address of the router with NAT enabled.
- Fixed IP Address: The global IP address of the router with NAT enabled is fixed.

Note

Manual programming is optional except when programming is required depending on the network conditions.

Value Range

Off, Fixed IP Addr., STUN

NAT - Voice (RTP) UDP Port No.

Specifies the starting port number of the dynamic ports used for NAT Traversal.

Value Range

2–65000 (even number only)

☉ NAT - Keep Alive Packet Sending Ability

Specifies whether to send Keep Alive packets in order to maintain the NAT binding information. This setting may be compulsory depending on the network conditions.

Value Range

Disable, Enable

NAT - Keep Alive Packet Type

Specifies the type of Keep Alive packets to be sent out.

Value Range

Blank UDP, None

NAT - Keep Alive Packet Sending Interval

Specifies the interval time until the next Keep Alive packet is sent.

Note

It is required to set this interval shorter than the NAT binding time of the router. The default value is appropriate in most cases.

Value Range

1–60 s

☉ NAT - Fixed Global IP Address

Specifies the global IP address of the router with NAT enabled. This setting is compulsory if **Fixed IP Addr.** is selected in **NAT Traversal**.

Value Range

1.0.0.0–223.255.255.255

STUN Ability

Specifies whether to open a protocol port of the STUN server to enable STUN feature.

Value Range

Disable, Enable

STUN Client Port Number

Specifies the port number of the PBX used for communications with the STUN server.

Value Range

1–65535

STUN External Address Detection Retry Counter

Specifies the number of times that a query is retried when no response is received from the STUN server.

Value Range

0–8

STUN Resending Interval

Specifies the length of time until a query is retried when no response is received from the STUN server.

Value Range

0–1600 ms

SIP Called Party Number Check Ability

Specifies whether to receive a SIP trunk call when receiving the INVITE message with an incorrect target SIP-URI. The setting specified here is also applied when the request header is blank or contains characters that can not be modified to a receivable number.

Value Range

Disable (High->Low), Disable (Low->High), Enable

Symmetric Response Routing Ability

Enables this feature to request that the SIP server sends the response back to the source IP address and port from which the request originated.

Value Range

Disable, Enable

100rel Ability

Specifies whether to add the option tag 100rel to the header field of the INVITE message.

Value Range

Disable: This feature is not activated.

Enable (Active): Activates this feature only if the other device supports the feature.

Enable (Passive): Activates this feature only when requested by the other device.

Ringback Tone to Outside Caller

Enables the PBX to send a ringback tone to an outside caller when the network cannot send the tone.

Value Range

Disable, Enable

3.3.2 Timer

SIP T1

Specifies the retransmission interval time for INVITE requests and responses.

Value Range

5–255 × 100 ms

SIP T2

Specifies the retransmission interval time for INVITE responses and non-INVITE requests.

Value Range

40–255 × 100 ms

3.3.2 Timer

Section 4

Maintenance

This section serves as reference operating instructions for data management of the V-SIPGW16 card using the Maintenance Console software.

4.1 Configuration Management

4.1.1 Importing/Exporting V-SIPGW16 Provider Profiles

Provider profiles for the V-SIPGW16 card can be exported or imported (if updated provider profiles are available), using the Import/Export tool.

Follow the steps below to import or export the provider profiles:

Note

To access the following menu options, you must be logged in to the Maintenance Console in Batch or Interactive mode at Installer level. For details about the software mode and access level, refer to "2.1.1 Starting Maintenance Console and Software Modes" and "2.1.2 Access Levels" in the PC Programming Manual.

To import provider profiles

1. From the **Tool** menu, point to **Import**, and then click **V-SIPGW16 Provider** to import.
2. Navigate to the folder containing the provider profiles you want to open.
3. Select the file.
4. Click **Open** to open the file.
5. Click **OK**.

To export provider profiles

1. From the **Tool** menu, point to **Export**, and then click **V-SIPGW16 Provider** to export.
2. Navigate to the folder in which you want to save the file.
3. Enter a file name.
4. Click **Save**.
5. Click **OK**.

4.1.2 V-SIPGW16 Protocol Trace

The trace data of protocol activity can be collected from the V-SIPGW16 card and saved to the PC, using the utility, **V-SIPGW16 Protocol Trace**.

This option requires that the target V-SIPGW16 card be set to INS status.

Follow the steps below to trace and file V-SIPGW16 card protocol activity:

Note

To access the following menu options, you must be logged in to the Maintenance Console in Batch or Interactive mode at Installer level. For details about the software mode and access level, refer to "2.1.1 Starting Maintenance Console and Software Modes" and "2.1.2 Access Levels" in the PC Programming Manual.

To save trace data to the SD Memory Card

1. From the **Utility** menu, select **V-SIPGW16 Protocol Trace**.
When the progress bar disappears, the protocol trace is complete. The trace data has been saved to the SD Memory Card (file name: "PRTSIPC").

To transfer trace data to the PC

1. From the **Utility** menu, select **File Transfer PBX (SD Card) to PC**.
The list of trace data files will be displayed.
2. Select the desired trace data file.
3. Click **Transfer**.
4. Navigate to the folder in which you want to save the file.
5. Enter a file name.
6. Click **Save**.
7. Click **OK**.

4.1.2 V-SIPGW16 Protocol Trace

Index

Index

A

Accessing Card Properties 37
Accessing Port Properties 18
Accessing Shelf Properties 40
Automatic Programming, Port Property 18

B

Bandwidth Requirements 8

C

Card Property, Icon Description 37
Card Property, Programming 37
Configuration Management 46

D

DHCP Server 14
DNS (Domain Name System) 5
DNS Server 5

F

Firewall Requirements 7
Fixed IP Address 5

H

Hardware Requirements 12

I

Icon Description, Card Property 37
Icon Description, Port Property 18
Icon Description, Shelf Property 40
Important Notice for Subscription and Installation 7
Installing and Starting the Maintenance Console 12
Installing the V-SIPGW16 Card to the PBX 16
ISP (Internet Service Provider) 4
ITSP (Internet Telephony Service Provider) 4

M

Maintenance 45
Manual Programming, Port Property 18

N

NAT (Network Address Translation) Traversal 5

P

Port Property, Automatic Programming 18
Port Property, Icon Description 18
Port Property, Manual Programming 18
Port Requirements 7
Programming Card Property 37
Programming Port Property 18
Programming Shelf Property 40
Provider Profiles 46

Q

Quick Setup 13

R

Required Bandwidth for Each Channel 8
Router Requirements 8

S

SBC (Session Border Controller) 6
Shelf Property, Icon Description 40
Specifications 10
STUN 5
STUN Server 5
System Requirements 12

V

V-SIPGW16 Connection Outline 4
V-SIPGW16 Protocol Trace 47

Panasonic Communications Co., Ltd.

1-62, 4-chome, Minoshima, Hakata-ku, Fukuoka 812-8531, Japan

Copyright:

This material is copyrighted by Panasonic Communications Co., Ltd., and may be reproduced for internal use only. All other reproduction, in whole or in part, is prohibited without the written consent of Panasonic Communications Co., Ltd.

© 2008 Panasonic Communications Co., Ltd. All Rights Reserved.