

Setup Reference guide for KX-NS Series
(Tested with NS1000 Ver4.42)
“I-Net Connect” SIP Trunk service
with External Router



Panasonic

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◆ SUMMARY

This document is a reference for configuring “**I-Net Connect**” SIP trunks onto KX-NS Series systems and includes the settings required for Incoming Call DDI routing and Outgoing Call CLI presentation. SIP trunk specific account details are provided to you by **I-Net Connect**.

◆ Attention:

This document was created based on the results of test environment accounts.

Panasonic cannot guarantee SIP Trunk operation in all environments, however as a result of completing this Inter-Operability Test Panasonic will provide technical support for any issues experienced an assist as far as possible in providing a resolution.

Please obtain relevant information from Service provider before configuration of SIP trunks.

Panasonic will not be held liable for any information provided in this guidance document.

Information used in this document is for interoperability testing.

Information and Specifications in this document are subject to change without notice.

◆ Note

When using multiple lines

It is not possible to assign two or more lines to one ID. Make multiple line agreement and obtain multiple IDs.

Busy

If there are incoming calls when all lines are busy, the caller hears ROT instead of Busy.

Transfer / Forward

When the Transfer/forwarding destination is PSTN, the CLI is not call originator but PBX's CLI.

FAX

Fax that send to PSTN (Call to Japan PSTN) had failures by signaling filtered.

Ringling time

When receiving a call, the bell rings for 1 minute, but on the calling side it keeps calling for about 17 seconds longer.

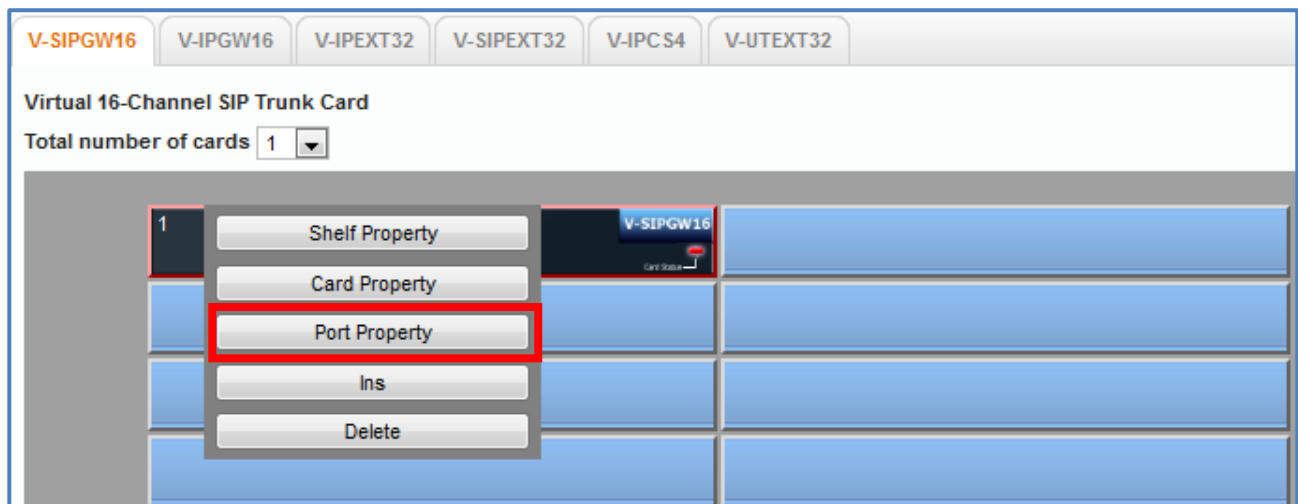
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(1) Provisioning the SIP Trunk

SIP Trunk – Port Property

Set the [V-SIPGW16] card to [OUS] and then select [Port Property]



[Main] Tab

1. Channel Attribute: *Basic Channel*
2. Provider Name: *Enter a name – reference only*
3. SIP Server Location – Name: *ipbx.inetplc.org – (I-Net Connect provided)*
4. SIP Server Location – IP Address: *Not required*
5. SIP Server port Number: *Leave at 5060*
6. SIP Service Domain: *Not required*
7. Subscriber Number: *Not required*

Port Property - Virtual SIP Gateway													
Select Provider Add Provider Trunk Adaptor													
«	Main	Account	Register	NAT	Option	Calling Party	Called Party	Voice/FAX	RTP/RTCP	T.38	T.38 Option	»	
Port	Connect	Connectio Attribute	Trunk Property	Channel Attribute	Provider Name (20 characters)	SIP Server Name (100 characters)	SIP Server IP Address	SIP Server IP Address for Failover	SIP Server Port Number	SIP Service Domain (100 characters)	Subscriber Number	B	
1	ALL	ALL	ALL	ALL									
1	OUS	SIP Provider	Public	Basic channel	I-Net 1	ipbx.inetplc.org			5060				
2	OUS	SIP Provider	Public	Basic channel	I-Net 2	ipbx.inetplc.org			5060				
3	OUS	SIP Provider	Public	Basic channel	I-Net 3	ipbx.inetplc.org			5060				
4	OUS	SIP Provider	Public	Not Used					5060				

SIP Trunk – Port Property continued

[Account] Tab

1. User name: **Enter the *Username as supplied by I-Net Connect*.**
(Note this is username without @ipbx.inetplc.org)
For example: Username = 12345670
Enter: 12345670
2. Authentication ID: **Enter the *Authentication ID as supplied by I-Net Connect*.**
(Note this is Authentication ID without @ipbx.inetplc.org)
For example: Authentication ID = 12345670
Enter: 12345670
3. Authentication Password: **Enter the *password as supplied by I-Net Connect*.**
For example: password = passwordABCD
Enter: passwordABCD

« Main Account Register NAT Option Calling Party Called Party Voice/FAX »							
No.	Shelf	Slot	Port	Connect	User Name (64 characters)	Authentication ID (64 characters)	Authentication Password (32 characters)
1	Virtual	1	1	OUS	12345670	12345670	passwordABCD
2	Virtual	1	2	OUS	12345671	12345671	passwordEFGH
3	Virtual	1	3	OUS	12345672	12345672	passwordJKLM
4	Virtual	1	4	OUS			

[Register] Tab

1. Register Ability: *Leave enabled*
2. Register Interval: **Enter: 120**
3. Un-Register Ability: *Leave enabled*
4. Registrar Server – Name: *Not required*
5. Registrar Server – IP Address: *Not required*
6. Registrar Server port number: *Leave at 5060*

« Main Account Register NAT Option Calling Party Called Party Voice/FAX RTP/RTCP T.38 T.38 Option »												
No.	Shelf	Slot	Port	Connect	Register Ability	Register Sending Interval (s)	Un-Register Ability when port INS	Registrar Server Name (100 characters)	Registrar Server IP Address	Registrar Server IP Address for Failover	Registrar Server Port Number	
1	Virtual	1	1	OUS	Enable	120	Enable				5060	
2	Virtual	1	2	OUS	Enable	120	Enable				5060	
3	Virtual	1	3	OUS	Enable	120	Enable				5060	

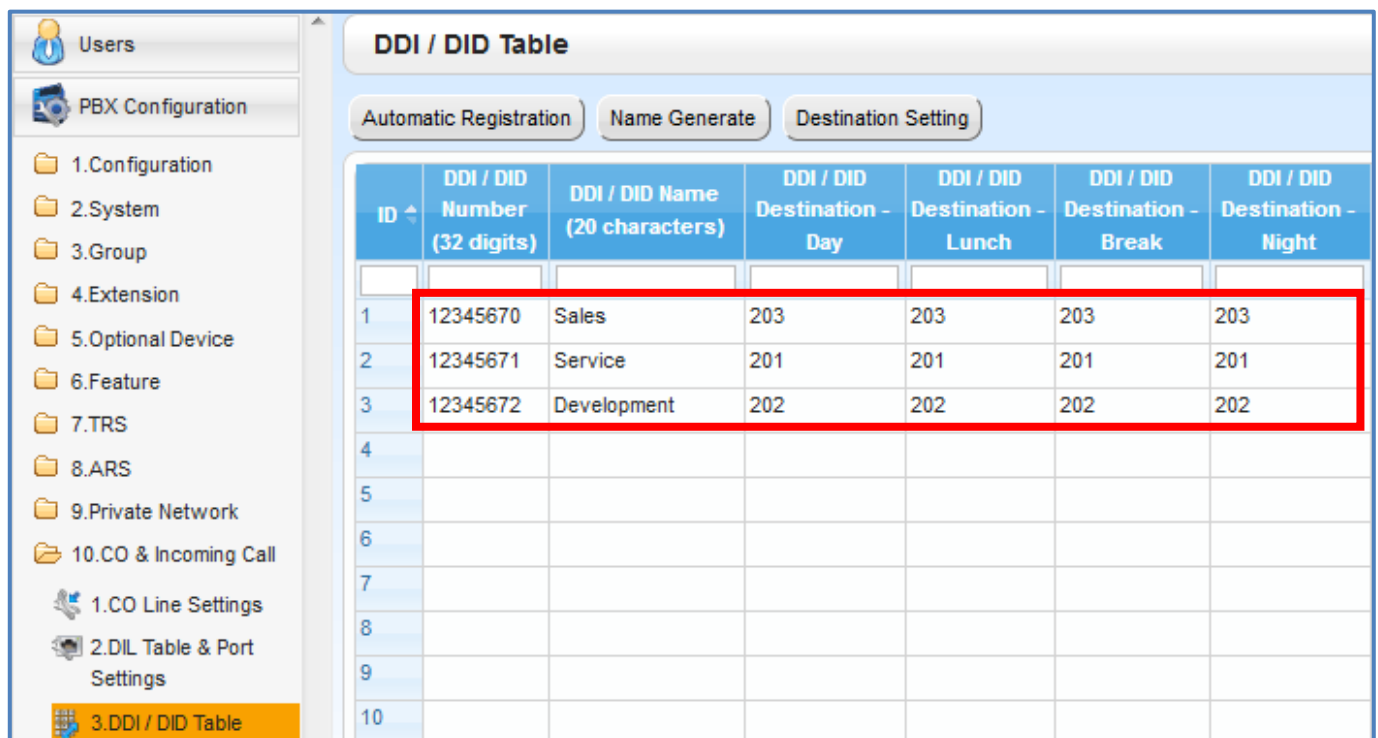
Click **[OK]** to apply the changes.

Select **[INS]** on the **[V-SIPGW16]** card to bring the SIP trunk ports into service.

(2) Incoming Call Routing

Go to **[CO & Incoming call]** and select **[3.DDI /DID Table]**

1. **DDI/DID Number:** *Enter the Username in the format (as below)*
Example: 12345671
*Enter: **12345671***
2. **DDI/DID Name:** *Determined by the installer (optional setting)*
3. **DDI/DID Destination:** *Determined by the installer (extension number, group etc)*



ID	DDI / DID Number (32 digits)	DDI / DID Name (20 characters)	DDI / DID Destination - Day	DDI / DID Destination - Lunch	DDI / DID Destination - Break	DDI / DID Destination - Night
1	12345670	Sales	203	203	203	203
2	12345671	Service	201	201	201	201
3	12345672	Development	202	202	202	202
4						
5						
6						
7						
8						
9						
10						

(3) Outgoing Call CLI

Set the **[V-SIPGW16]** card to **[OUS]** and then select **[Port Property]** and **[Calling Party]** Tab

From Header – User Part:

PBX-CLIP

No.	Shelf	Slot	Port	Connection	Header Type	From Header - User Part	From Header - SIP-UR (100 characters)
1	Virtual	1	1	ALL	From Header	PBX-CLIP	
2	Virtual	1	2	OUS	From Header	User Name	
3	Virtual	1	3	OUS	From Header	User Name	
4	Virtual	1	4	OUS	From Header	User Name	

Click **[OK]** and then set the **[V-SIPGW16]** card back to **[INS]**.

Go to **[4.Extension] – [1.Wired Extension] – [1.Extension Settings]** and select **[CLIP]** tab

Enter a valid CLI number for each required extension in the **CLIP ID** field and then Click **[OK]**

No.	Extension Number	Extension Name (20 characters)	CLIP ID	CLIP on Extension/CO	CLIR	COLR
1	201	Service	02012345001	Extension	Disable	Disable
2	202	Development	02012345002	Extension	Disable	Disable
3	203	Sales	02012345000	Extension	Disable	Disable
4	204			Extension	Disable	Disable

(4) CLIR Outgoing Call (Withholding Number)

Dial using the 141 prefix in order to withhold CLI presentation number.

(e.g.) 141 055 1234 0020

(5) Appendix

UDP hole punching for keeping in Router/Firewall Port Forwarding

If required, for keeping port forwarding table in installed router, configure the BLANK UDP to enable as keep-alive with UDP hole punching on the PBX shelf property.

Set the [V-SIPGW16] card to [OUS] and then select [Shelf Property]

NAT - Keep Alive Packet Sending Ability: *Enable*

NAT - Keep Alive Packet Sending Interval(s): *leave at 20*

Note: It is desirable shorter than expire time of router port forwarding.

The screenshot shows the 'Shelf Property - Virtual SIP Gateway' configuration window. The left sidebar contains a tree view with 'PBX Configuration' expanded, showing '1. Configuration' and '1. Slot' selected. The main area has two tabs: 'Main' and 'Timer'. Under the 'Main' tab, several settings are listed on the left and their values on the right. The 'NAT - Keep Alive Packet Sending Ability' is set to 'Enable' and the 'NAT - Keep Alive Packet Sending Interval (s)' is set to '20'. Both of these settings are enclosed in red rectangular boxes. Other settings include 'SIP Client Port Number' (35060), 'NAT Traversal' (Off), 'NAT - Voice (RTP) UDP Port No.' (16000), 'NAT - Keep Alive Packet Type' (Blank UDP), and 'NAT - Fixed Global IP Address' (0.0.0.0).

Click [OK] and Select [INS] on the [V-SIPGW16] card to bring the SIP trunk ports into service.

IMPORTANT!

To secure the PBX from illegal attacks, please restrict the above port forwarding ports to only be accessible from the I-Net Connect source IP addresses.

END OF DOCUMENT