

Setup Reference guide for KX-NS Series
(Tested with NS1000 Ver5.0)
“TBC” SIP Trunk service
with External Router



Panasonic

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◆ SUMMARY

This document is a reference for configuring “**TBC**” 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 **TBC**.

◆ 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

FAX

Need to check a FAX communication in advance if need to connect it.

We had a failure between international/Japan.

Refer

Use PBX “Transfer to CO” function. TBC does not work properly for SIP REFER messages.

P-Asserted-Identity (PAI) Header

Not use the PAI header. When PAI header is attached, the number will not be notified correctly.

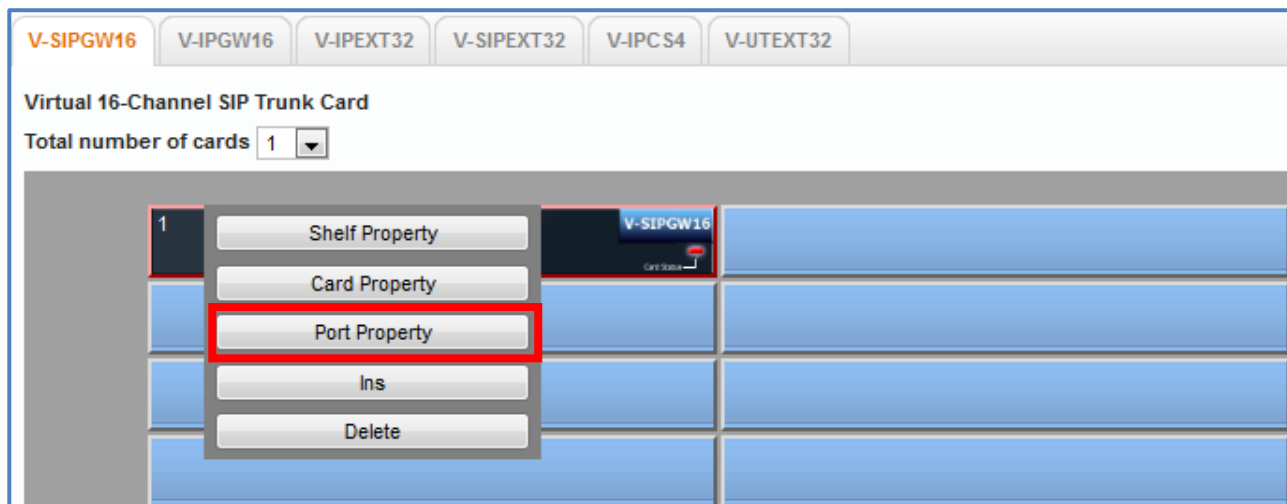
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(1) Provisioning the SIP Trunk

SIP Trunk – Port Property

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



[Main] Tab

- | | |
|--------------------------------------|---|
| 1. Channel Attribute: | <i>Basic Channel</i> |
| 2. Provider Name: | <i>Enter a name – reference only</i> |
| 3. SIP Server Location – Name: | <i>voip.tbc.com.es – (TBC provided)</i> |
| 4. SIP Server Location – IP Address: | <i>Not required</i> |
| 5. SIP Server port Number: | <i>Leave at 5060</i> |
| 6. SIP Service Domain: | <i>Not required</i> |
| 7. Subscriber Number: | <i>Not required</i> |

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	Connectio	Connection 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	Backup	
1	OUS	SIP Provider	Public	Basic channel	TBC	voip.tbc.com.es			5060			Normal	
2	OUS	SIP Provider	Public	Not Used					5060			Normal	

SIP Trunk – Port Property continued

[Account] Tab

1. User name: **Enter the *Username as supplied by TBC.***
(Note this is username without @voip.tbc.com.es)
For example: Username = *tbc_user1*
Enter: *tbc_user1*
2. Authentication ID: **Enter the *Authentication ID as supplied by TBC.***
(Note this is Authentication ID without @voip.tbc.com.es)
For example: Authentication ID = *tbc_user1*
Enter: *tbc_user1*
3. Authentication Password: **Enter the *password as supplied by TBC.***
For example: password = *passABCD*
Enter: *passABCD*

« Main Account Register NAT Option Calling Party Called Party Voice/FAX »									
No.	Shelf	Slot	Port	Connection	User Name (64 characters)	Authentication ID (64 characters)	Authentication Password (32 characters)		
1	Virtual	1	1	OUS	tbc_user1	tbc_user1	passABCD		
2	Virtual	1	2	OUS					

[Register] Tab

1. Register Ability: *Leave enabled*
2. Register Interval: *Leave at 3600*
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	Connection	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	3600	Enable				5060	

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

(2) Outgoing Call CLI

Go to **[Calling Party] Tab**

From Header – User Part:

PBX-CLIP

No.	Shelf	Slot	Port	Connection	Header Type	From Header - User Part	From Header - SIP-URI (100 characters)
1	Virtual	1	1	OUS	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	101	Service	991234001	Extension	Disable	Disable
2	102	Development	991234002	Extension	Disable	Disable
3	103	Sales	991234000	Extension	Disable	Disable
4	104			Extension	Disable	Disable

(3) CLIR Outgoing Call (Withholding Number)

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

Under **CLIR**: select **Enable** and Click **[OK]**.

Extension Settings

Copy to CLIP Generate

« Main Intercept Destination Intercept No Answer Time **CLIP** UM C

No.	Extension Number	Extension Name (20 characters)	CLIP ID	CLIP on Extension/CO	CLIR	COLR
				ALL ▼	ALL ▼	ALL ▼
1	101	Service	991234001	Extension	Enable	Disable
2	102	Development	991234002	Extension	Disable	Disable
3	103	Sales	991234000	Extension	Disable	Disable
4	104			Extension	Disable	Disable

(4) Incoming Call Routing

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

1. **DDI/DID Number:** *Enter the DDI number in the format (as below)*
Example: 99 123 40 00
*Enter: **991234000***
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	991234000	Sales	101	101	101	101
2	991234001	Service	102	102	102	102
3	991234002	Development	103	103	103	103
4						
5						
6						
7						
8						
9						
10						

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

1. **CLIP (Receive):** *From Header*

No.	Shelf	Slot	Port	Connection	Blind Transfer(REFER)	Attended Transfer(REFER)	CLIP (Receive)	CLIR	CNIP (Send)	CNIP (Receive)
1	Virtual	1	1	OUS	No	No	From Header	Yes	Yes	No
2	Virtual	1	2	OUS	No	No	From Header	Yes	Yes	No

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

When using Transfer/Forward, the TBC will not notify the CLIP of the first caller. In order to notify the forwarder CLIP, change the setting of System Options.

Go to **[System]** and select **[9.System Options]**

[Option 4] Tab

1. when call is transferred to CO (CLIP of Held Party): **Disable**

2. when call is forwarded to CO: **Disable**

The screenshot shows the 'System Options' configuration window with the 'Option 4' tab selected. The left sidebar lists various configuration categories, with '9. System Options' highlighted. The main area contains several settings:

- DSS Key**
 - DSS key mode for Incoming Call: ☒ On or Flash, ☐ Off
 - Call Pick-up by DSS key for Direct Incoming Call: ☒ Enable, ☐ Disable
 - Call Pick-up by DSS key for ICD Group Call: ☐ Enable, ☒ Disable
- Automatic Transfer for Extension Call**: ☐ Enable, ☒ Disable
- Caller Information Display before Call Pick-up**: ☐ Enable, ☒ Disable
- Send CLIP of CO Caller**
 - when call is transferred to CO (CLIP of Held Party): ☐ Enable, ☒ Disable
 - when call is forwarded to CO: ☐ Enable, ☒ Disable
- Send CLIP of Extension Caller**
 - when call is forwarded to CO: ☒ Enable, ☐ Disable
- System Wireless**
 - Out of Range Registration: ☐ Enable, ☒ Disable

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

(5) Appendix

UDP hole punching for keeping in Router/Firewall Port Forwarding

If TBC's Keep Alive message (e.g. OPTIONS) is not effect to keep port forwarding for external router, configure the BLANK UDP to enable as keep-alive message 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'. The 'Main' tab is active, displaying the following settings:

Property	Value
SIP Client Port Number	35060
NAT Traversal	Off
NAT - Voice (RTP) UDP Port No.	16000
NAT - Keep Alive Packet Sending Ability	Enable
NAT - Keep Alive Packet Type	Blank UDP
NAT - Keep Alive Packet Sending Interval (s)	20
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 TBC source IP addresses.

END OF DOCUMENT