



**Setup Reference guide for KX-NS Series
(Tested with NS700 Ver8.0)
“sipcall” SIP trunk service
with External Router**



Panasonic

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◆ SUMMARY

This document is a reference for configuring “[sipcall](#)” 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 [sipcall](#).

◆ 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 and 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

REFER

[Supported](#)

◆ Table of Contents

(1) Provisioning the SIP trunk:	Page 3
(2) Outgoing Call CLI:	Page 5
(3) Incoming Call Routing:	Page 7
(4) CLIR Outgoing Call (Anonymous calls):	Page 8
(5) Router/Firewall Port Forwarding	Page 9

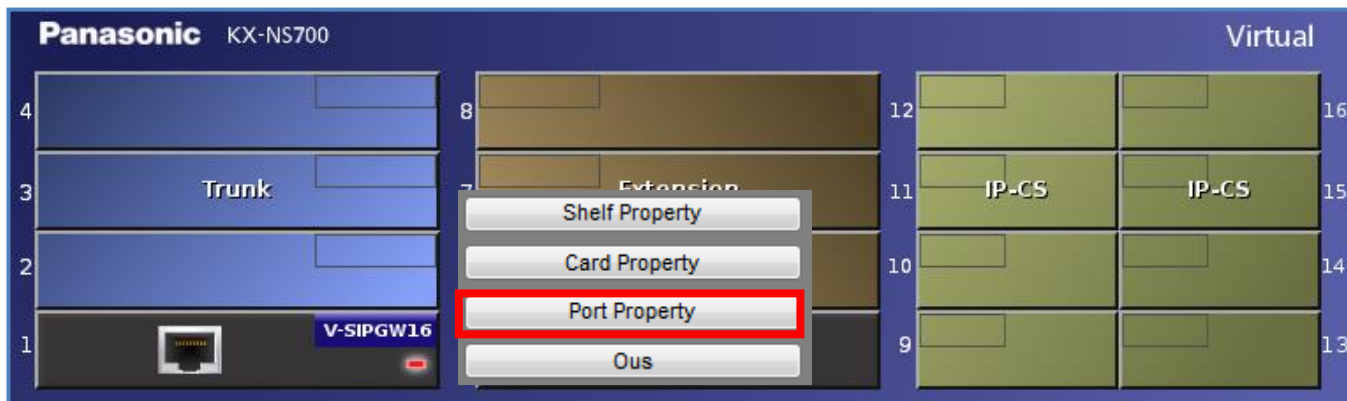
(1) Provisioning the SIP Trunk

V-SIPGW – Port Property

[1.Configuration – 1.Slot > V-SIPGW tab]

First, place the **V-SIPGW** Card into **[OUS]** condition.

Hover over **[V-SIPGW16]** and 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>Not required</i> |
| 4. SIP Server IP Address: | <i>212.117.203.60</i>
<i>Enter the domain as supplied by sipcall</i> |
| 5. SIP Server IP Address for Failover: | <i>Not required</i> |
| 6. SIP Server port Number: | <i>Leave at 5060</i> |
| 7. SIP Service Domain: | <i>Not required</i> |
| 8. Subscriber Number: | <i>Not required</i> |

Port Property - Virtual SIP Gateway

Select Provider Add Provider

« **Main** Account Register NAT Option Calling Party Called Party Voice/FAX RTP/RTCP T.38 T.38 Option »

No.	She	Slot	Port	Connec	Trunk Proper	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
	▼			ALL ▼	ALL ▼	ALL ▼					
1	Virtua1	1		OUS	Public	Basic channel	SIPCALL		212.117.203.60		5060
2	Virtua1	2		OUS	Public	Additional channel for Slot 1 Ch 1					5060

SIP Trunk – Port Property continued

[Account] Tab

- 1. User name: *Enter the **User Name as supplied by sipcall.**
Note this is User name without @ 212.117.203.60)
For example: User name = 41412345000
Enter: 41412345000*
- 2. Authentication ID: *Enter the **Authentication ID as supplied by sipcall.**
(Note this is User name without @ 212.117.203.60)
For example: Authentication ID = 41412345000
Enter: 41412345000*
- 3. Authentication Password: *Enter the **password as supplied by sipcall.**
For example: password = pass1234
Enter: pass1234*

« Main Account Register NAT Option Calling Party Called Party Voice/FAX RT									
No.	She	Slot	Port	Connec	User Name (64 characters)	Authentication ID (64 characters)	Authentication Password (32 characters)		
	↓			ALL ↓					
1	Virtua	1		OUS	41412345000	41412345000	pass1234		
2	Virtua	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 IP Address for Failover: *Not required*
- 7. Registrar Server port number: *Leave at 5060*
- 8. Registrar Resending Interval(s): *Leave at 300*

« Main Account Register NAT Option Calling Party Called Party Voice/FAX RTP/RTCP T.38 T.38 Option DSP »												
No.	She	Slot	Port	Conne	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	Register Resending Interval (s)
	↓			ALL ↓	ALL ↓		ALL ↓					
1	Virtua	1		OUS	Enable	3600	Enable				5060	300
2	Virtua	2		OUS	Enable	3600	Enable				5060	300

(2) Outgoing Call CLI

[Calling Party] Tab

- 1. Header Type: *Leave From Header*
- 2. From Header - User Part: *PBX-CLIP*
- 3. Number Format: *Leave National*

No.	Shelf	Slot	Port	Connect	Header Type	From Header - User Part	From Header - SIP-URI (100 characters)	P-Preferred-Identity Header - User Part	P-Preferred-Identity Header - SIP-URI (100 characters)	Number Format	Remove Digit
	ALL			ALL	ALL	ALL		ALL		ALL	ALL
1	Virtual	1	1	OUS	From Header PBX-CLIP			PBX-CLIP		National	0
2	Virtual	1	2	OUS	From Header User Name			User Name		National	0

[Called Party] Tab

- 1. Number Format: *Leave National*
- 2. Type: *Leave To header*

No.	Shelf	Slot	Port	Connect	Number Format	Type	MEX - Prefix for Incoming /E.164 - Prefix for own system (16 characters)	MEX - Prefix for Outgoing /E.164 - Prefix for other system (16 characters)	MEX /E.164 - Additional Dial (7 digits)
	ALL			ALL	ALL	ALL			
1	Virtual	1	1	OUS	National	To header			0
2	Virtual	1	2	OUS	National	To header			0

Click **[OK]**.

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



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.

1. Extension Name: *Enter: Sales*

2. CLIP ID: *Enter: 0412345000*

No.	Extension Number	Extension Name (20 characters)	CLIP ID	CLIP on Extension/CO	CLIR	COLR
				ALL	ALL	ALL
1	201	Sales	0412345000	Extension	Disable	Disable
2	202	Service	0412345001	Extension	Disable	Disable
3	203	Development	0412345002	Extension	Disable	Disable

Click **[OK]**

(3) Incoming Call Routing

Go to **[10.CO & Incoming call]** and select **[3.DDI /DID Table]**

- 1. **DDI/DID Number:** *Enter the DDI number in the format (as below)*
Example: 41412345000
*Enter: **41412345000***
- 2. **DDI/DID Name:** *Determined by the installer (optional setting)*
- 3. **DDI/DID Destination:** *Determined by the installer (extension number, group etc)*

The screenshot shows the 'DDI / DID Table' configuration window. The left sidebar has a tree view with '10.CO & Incoming Call' expanded to '3.DDI / DID Table'. The main area contains a table with the following data:

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	Tenant / Time Group
1	41412345000	Sales	201	201	201	201	1
2	41412345001	Service	202	202	202	202	1
3	41412345002	Development	203	203	203	203	1
5							1
6							1
7							1
8							1
9							1

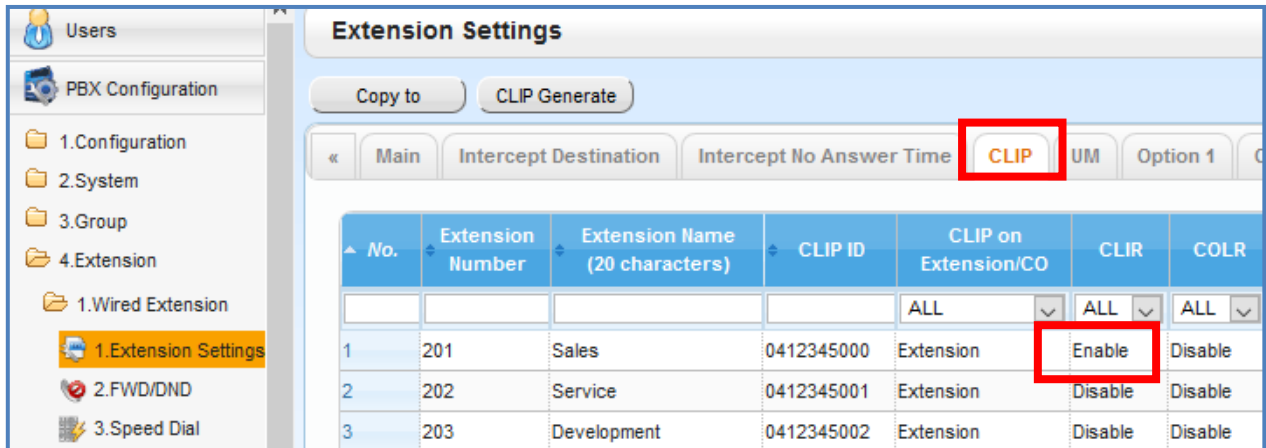
Click **[OK]**

(4) CLIR Outgoing Call (Anonymous Call)

-1 by CLIR setting of PBX

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

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



-2 by Type Anonymous prefix code before phone number

Make dialing *67 + Called Party phone number, it will be Anonymous at called party.

(5) Router/Firewall Port Forwarding

sipcall server does not send any keep alive packets to PBX. But Server respond a register expire value is 40 – 60 seconds to PBX when PBX using a NAT-Off (default) as SIP setting, and PBX sends re-register in half of that value. It might be almost external routers can hold a port forwarding.

Therefore the port forwarding may be not necessary on the router.

If necessary,

For External router setup, configure Port Forwarding on the router as follows:

udp port 35060 (SIP) – to NS LAN IP address (e.g. 192.168.0.101)

IMPORTANT!

To secure the PBX from illegal attacks, please restrict the above port forwarding ports to only be accessible from the [sipcall](#) source IP addresses.

END OF DOCUMENT