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 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

REFER

Supported

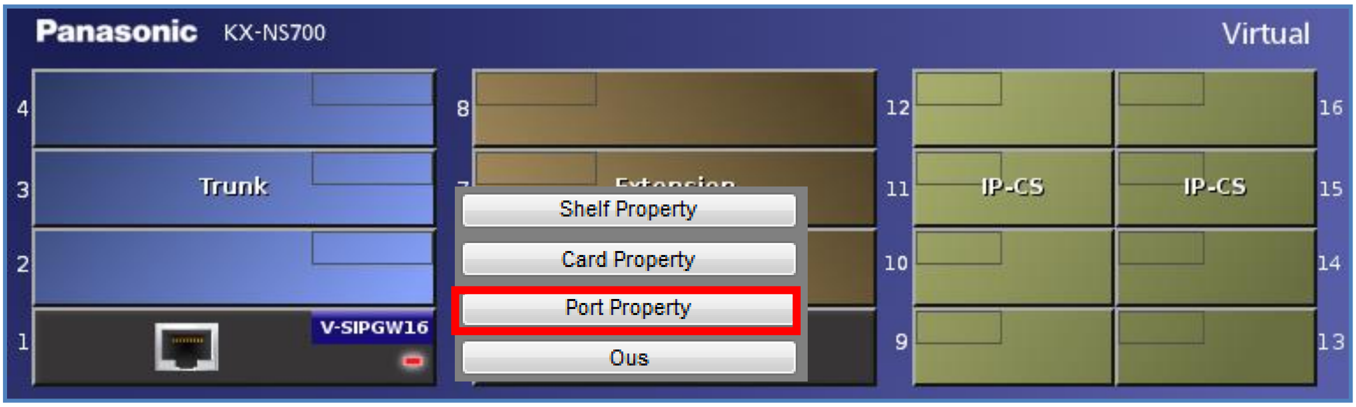
◆ Table of Contents

(1) Provisioning the SIP Trunk
V-SIPGW – Port Property

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

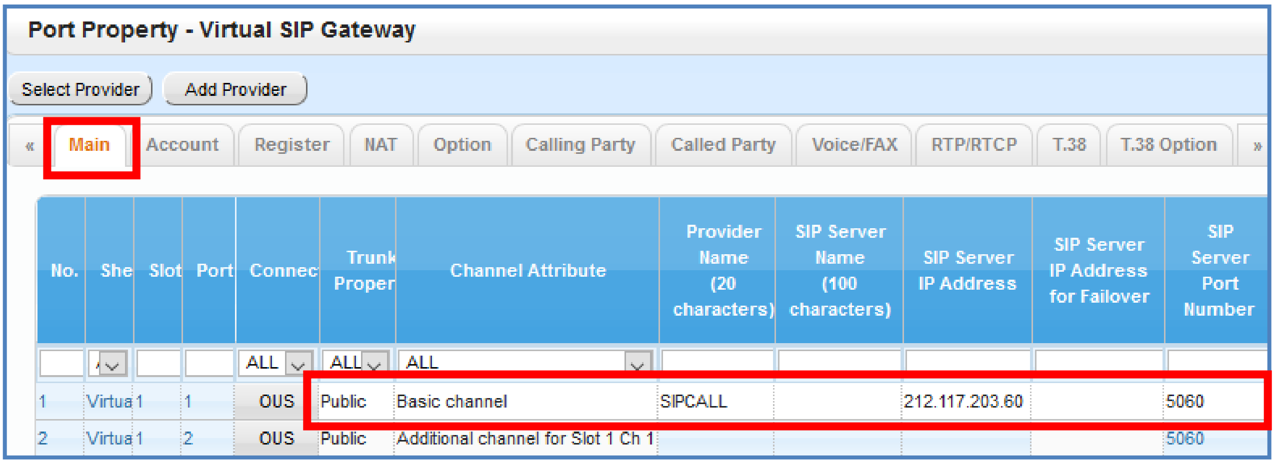
First, place the **V-SIPGW** Card into **[OUS]** condition.

Hover over **[V-SIPGW16]** and select **[Port Property]**



[Main] Tab

1. **Channel Attribute:** ***Basic Channel***
2. **Provider Name:** ***Enter a name – reference only***
3. **SIP Server Location - Name:** *Not required*
4. **SIP Server IP Address:** ***212.117.203.60***
   Enter the domain as supplied by sipcall
5. **SIP Server IP Address for Failover:** *Not required*
6. **SIP Server port Number:** *Leave at 5060*
7. **SIP Service Domain:** *Not required*
8. **Subscriber Number:** *Not required*

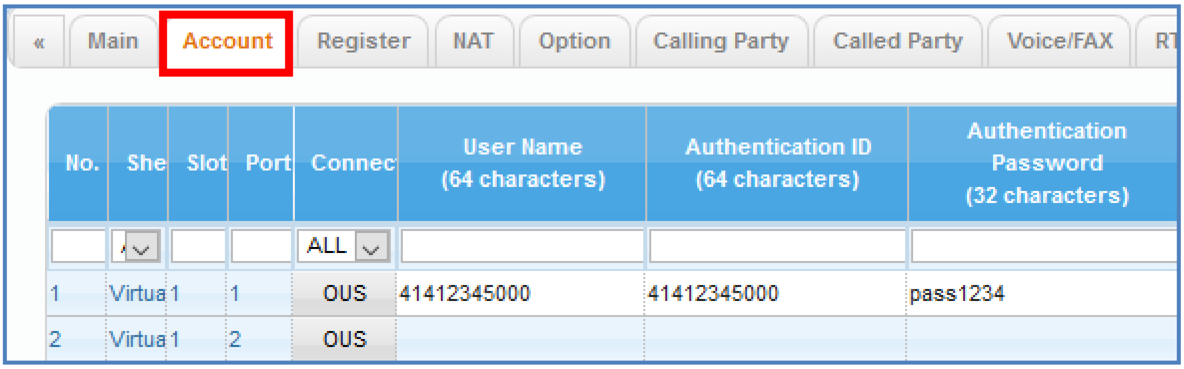


No.	She	Slot	Port	Connec	Trunk Proper	Channel Attribute	Provider Name (20 characters)	SIP Server Name (100 characters)	SIP Server IP Address	SIP Server IP Address for Failover	SIP Server Port Number
				ALL	ALL	ALL					
1	Virtua	1	1	OUS	Public	Basic channel	SIPCALL		212.117.203.60		5060
2	Virtua	1	2	OUS	Public	Additional channel for Slot 1 Ch 1					5060

SIP Trunk – Port Property continued

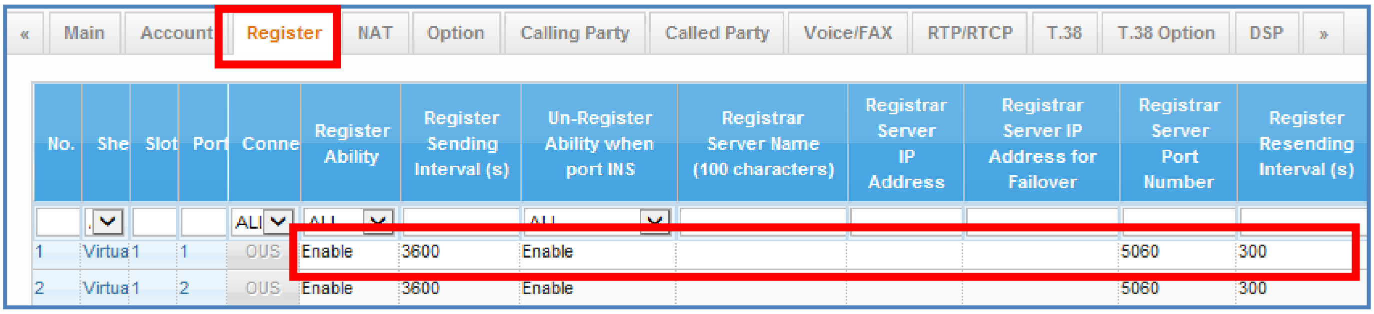
[Account] Tab

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



[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*



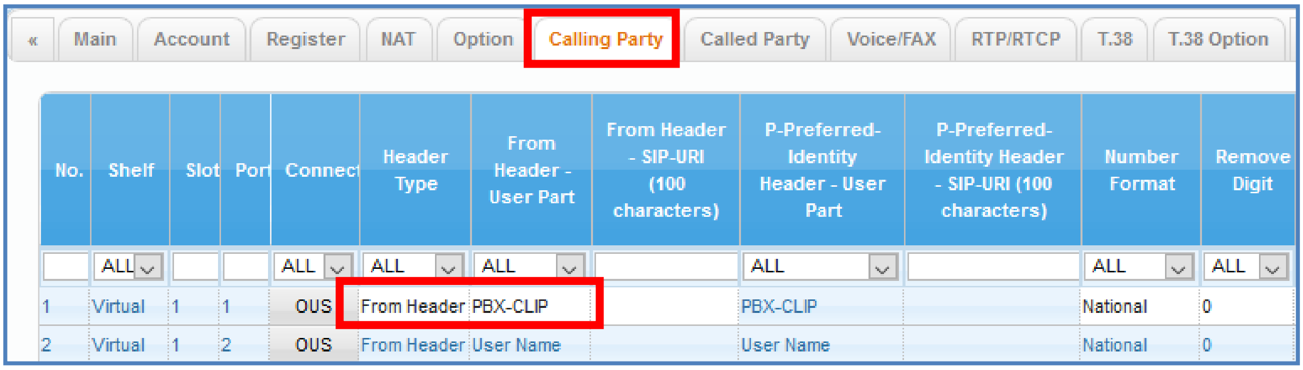
(2) Outgoing Call CLI

[Calling Party] Tab

1. Header Type: *Leave From Header*

2. From Header - User Part: ***PBX-CLIP***

3. Number Format: *Leave National*

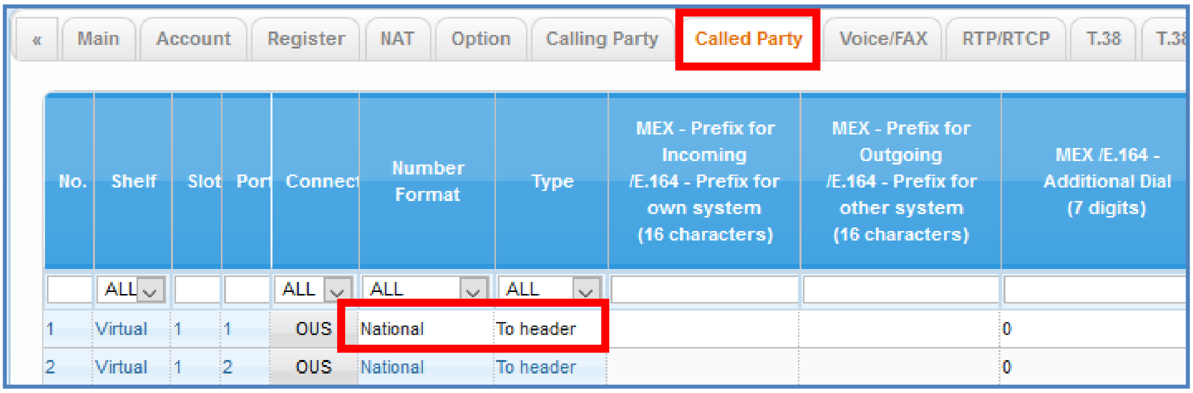


No.	Shelf	Slot	Port	Connect	Header Type	From Header - User Part	From Header - SIP-URI (100 characters)	P-Preferred-Identity Header - User Part	P-Preferred-Identity Header - SIP-URI (100 characters)	Number Format	Remove Digit
	ALL			ALL	ALL	ALL		ALL		ALL	ALL
1	Virtual	1	1	OUS	From Header	PBX-CLIP		PBX-CLIP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

[Called Party] Tab

1. Number Format: *Leave National*

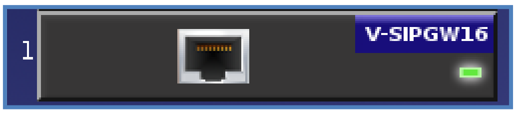
2. Type: *Leave To header*



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]**.

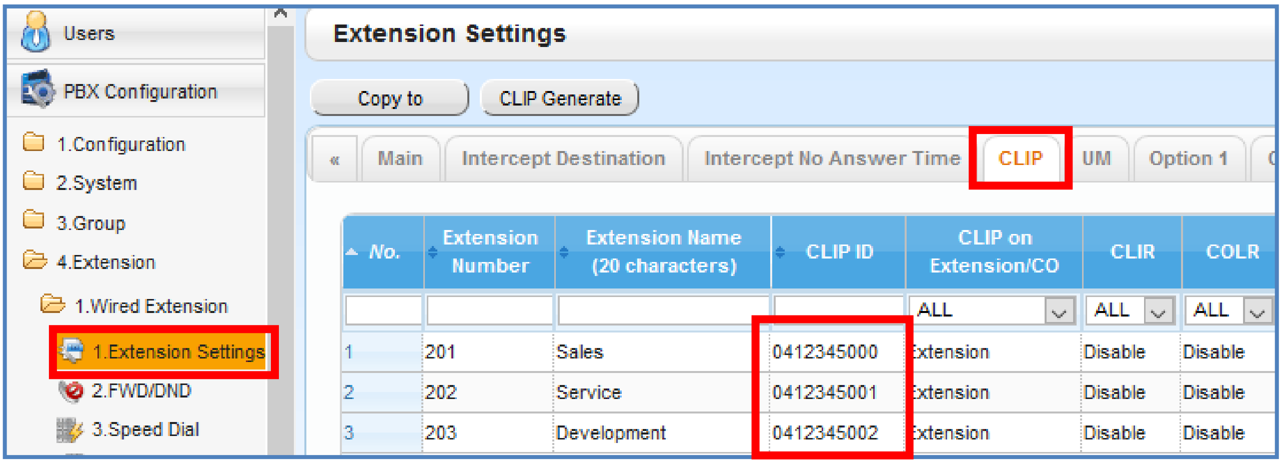
Select **[INS]** on the **[V-SIPGW16]** card to bring the SIP trunk ports into service.



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***

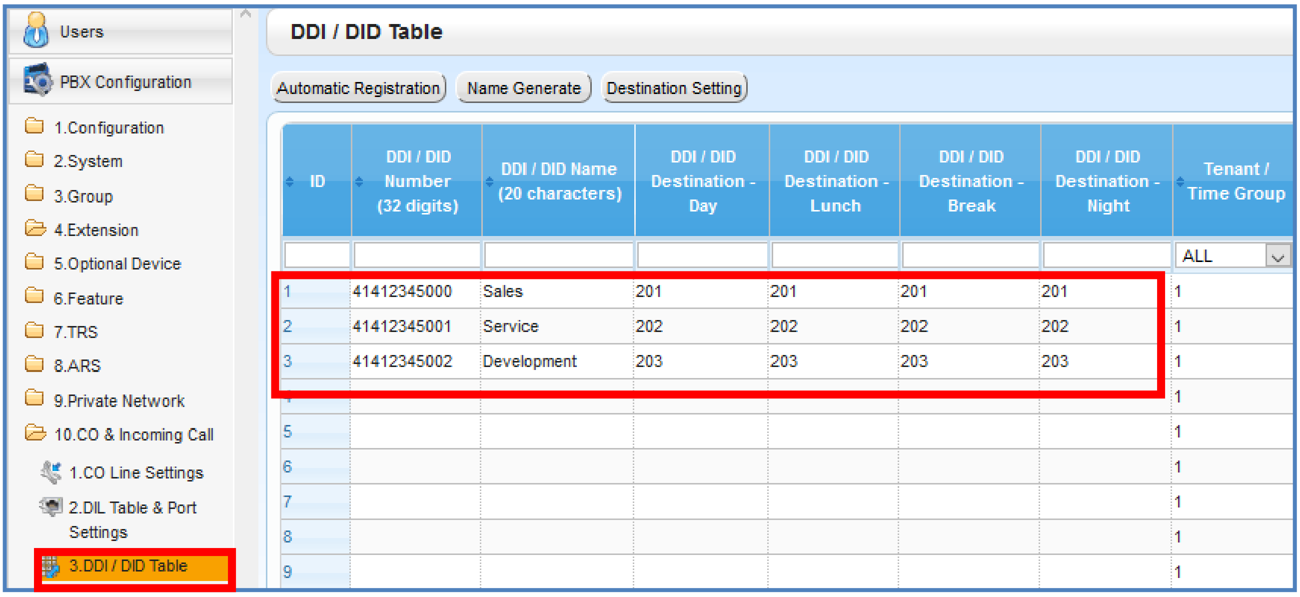


Click **[OK]**

(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)*

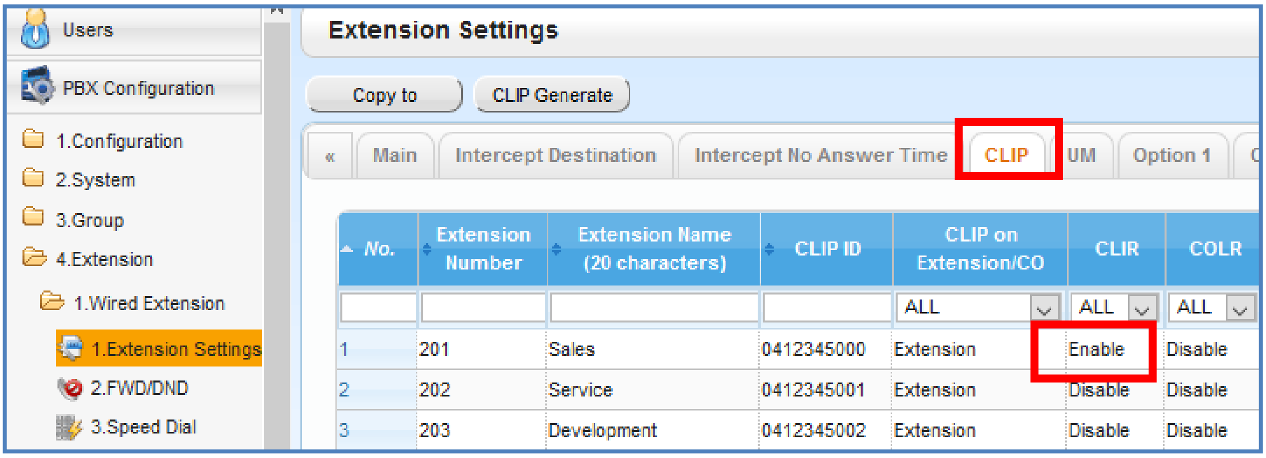


Click **[OK]**

(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