Setup Reference guide for KX-NS Series
(Tested with NS1000 Ver4.5)
"Peoplefone" SIP Trunk service
with External Router

Panasonic

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◆ SUMMARY

This document is a reference for configuring "**Peoplefone**" 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 **Peoplefone**.

◆ 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

FAX between Peoplefone and Japan result was NG. Between 2 Peoplefone accounts, FAX was OK by 2 methods, G.711 and T.38.

P-Asserted-Identity (PAI) Header

Not need PAI header for inform the CLIP. Set the CLIP in the From header, then Peoplefone inform it.

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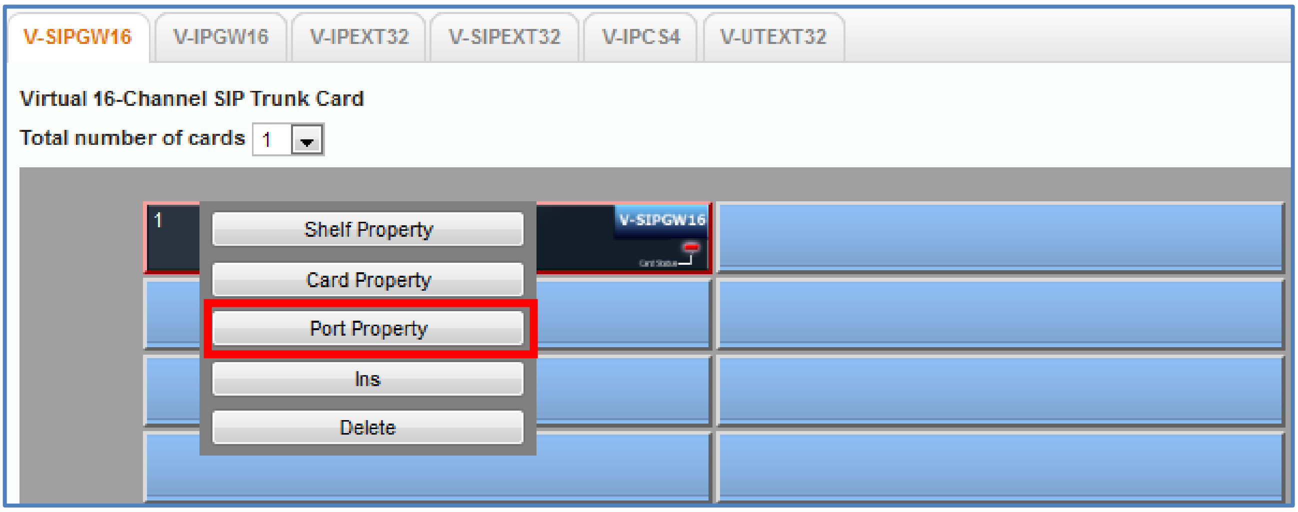
(1) Provisioning the SIP Trunk

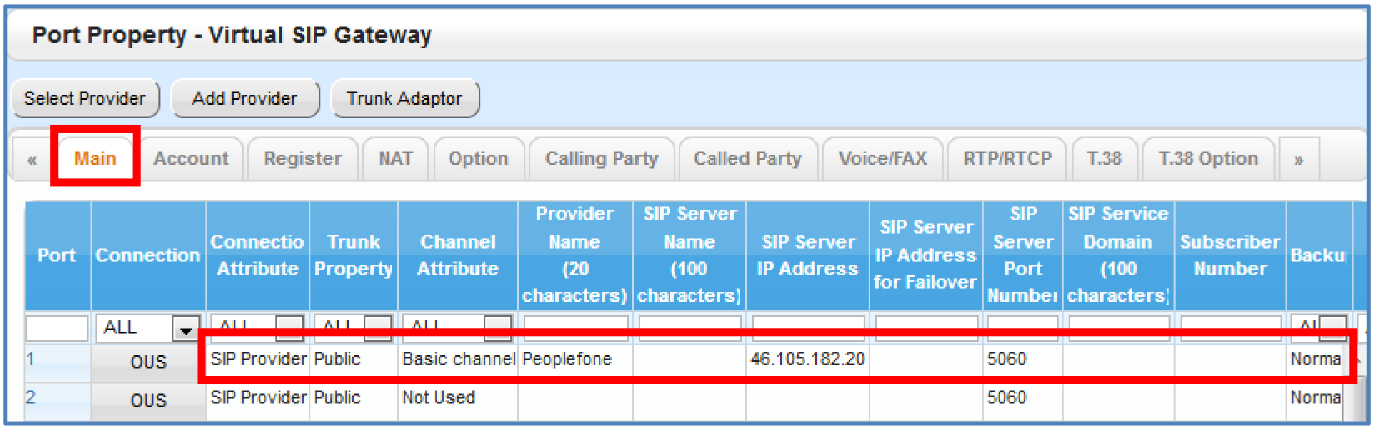
SIP Trunk – Port Property

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:** *Not required*
4. **SIP Server Location – IP Address:** ***46.105.182.20 – (Peoplefone provided)***
5. **SIP Server port Number:** *Leave at 5060*
6. **SIP Service Domain:** *Not required*
7. **Subscriber Number:** *Not required*



Port	Connection	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	Backu
	ALL	ALL	ALL	ALL								AL
1	OUS	SIP Provider	Public	Basic channel	Peoplefone		46.105.182.20		5060			Norma
2	OUS	SIP Provider	Public	Not Used					5060			Norma

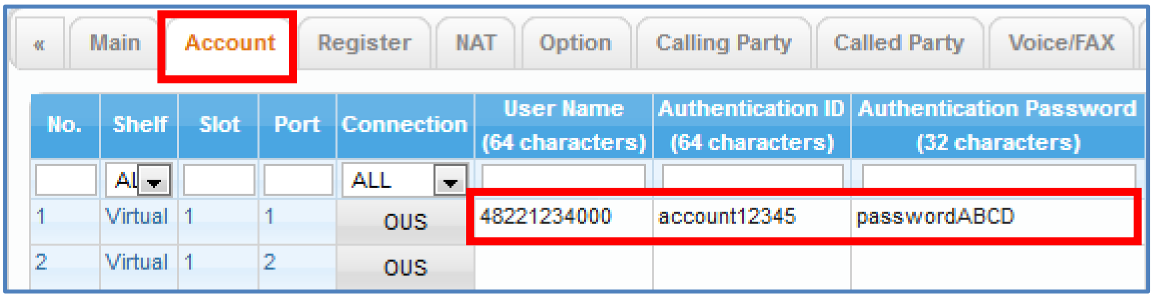
SIP Trunk – Port Property continued

[Account] Tab

1. User name: ***Enter the pilot number as supplied by Peoplefone.***
(Note this is pilot number without @46.105.182.20)
For example: pilot number = 48221234000
Enter: 48221234000

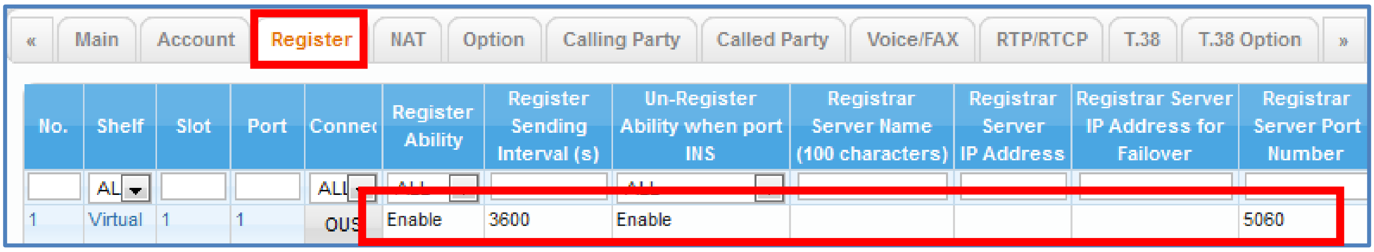
2. Authentication ID: ***Enter the Username as supplied by Peoplefone.***
(Note this is Username without @46.105.182.20)
For example: Username = account12345
Enter: account12345

3. Authentication Password: ***Enter the password as supplied by Peoplefone.***
For example: password = passwordABCD
Enter: passwordABCD



[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*

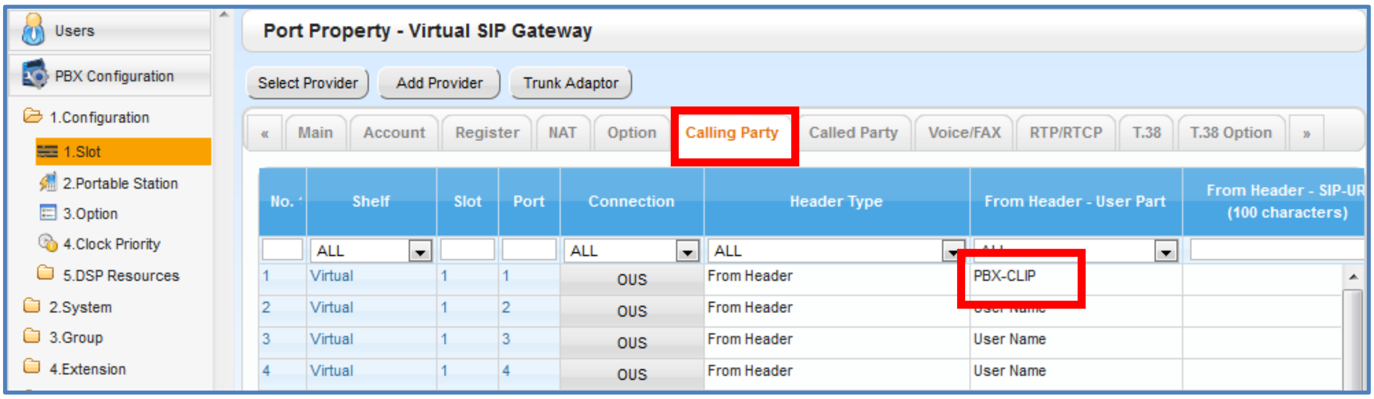


Click **[Apply]** to apply the changes.

(2) Outgoing Call CLI

Go to **[Calling Party] Tab**

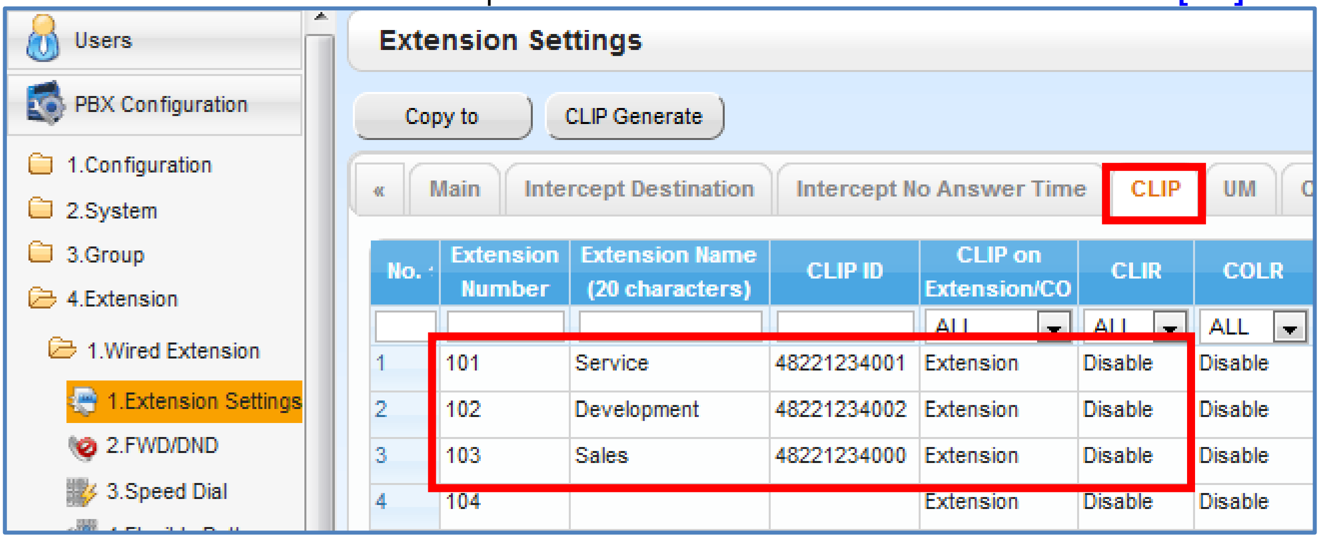
From Header – User Part: ***PBX-CLIP***



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

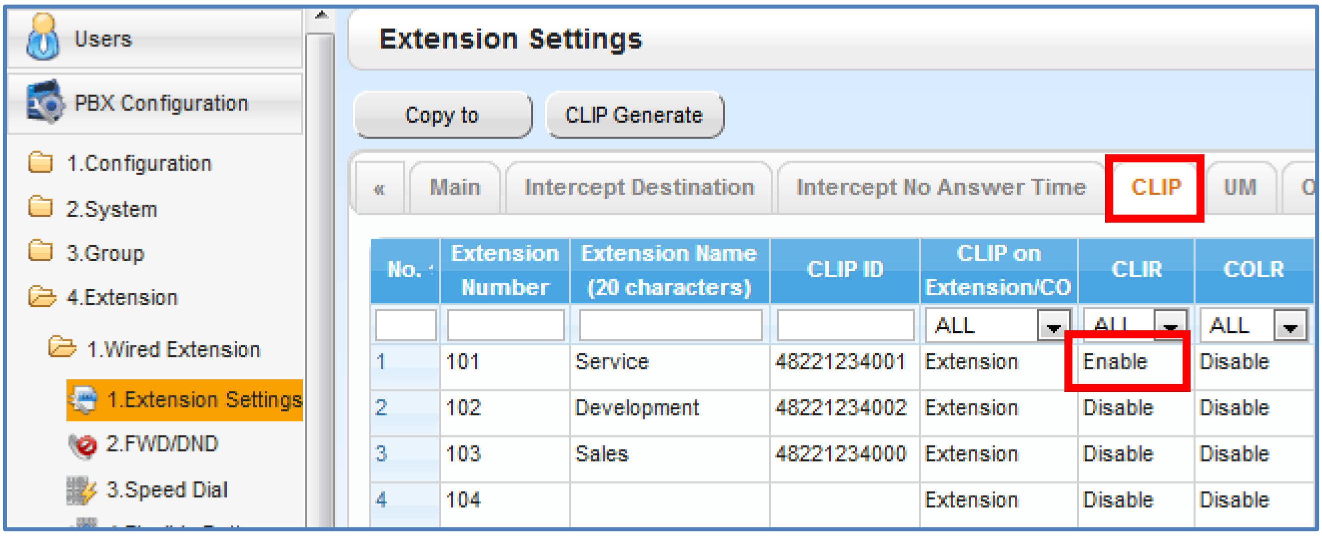
Go to **[4.Extension] – [1.Wired Extension] – [1.Extension Settings]** and select **[CLIP] tab**

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



(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]**.



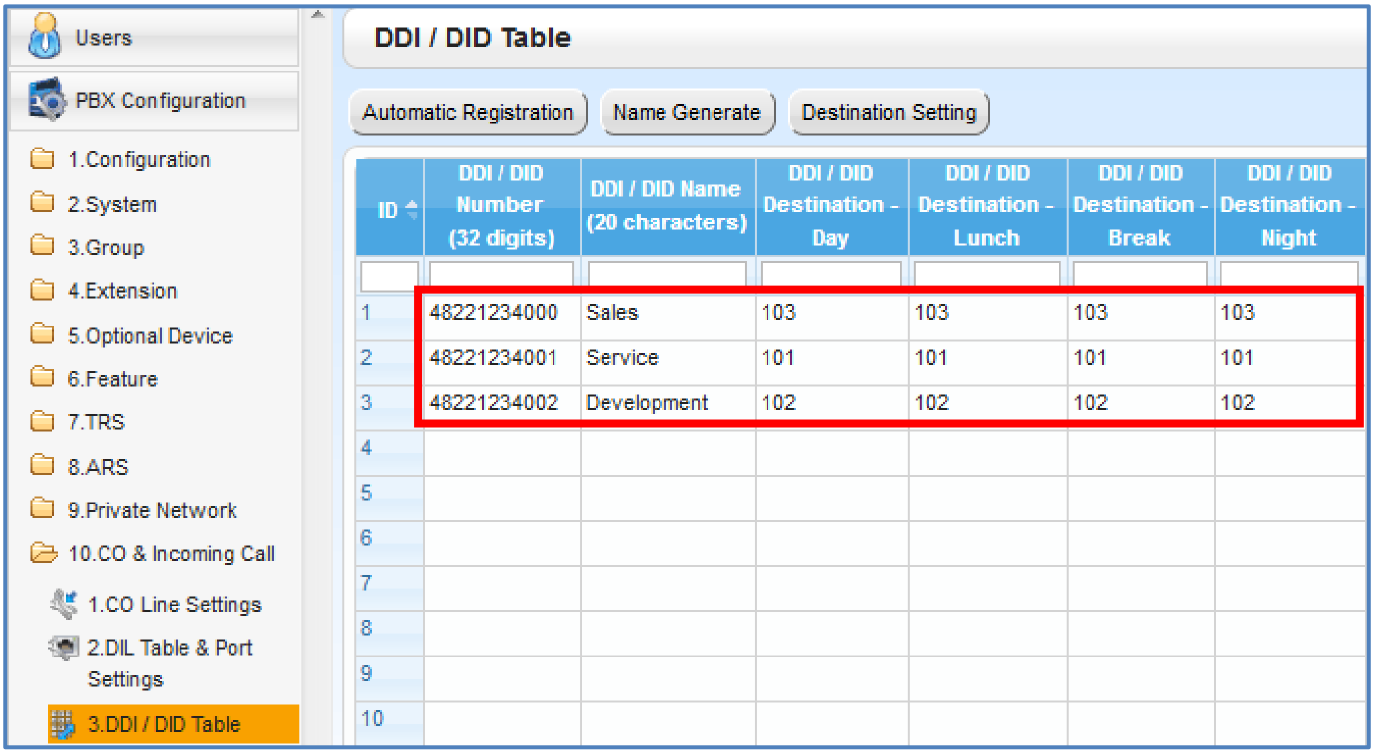
(4) Incoming Call Routing

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: 48221234001
Enter: ***48221234001***

2. DDI/DID Name: *Determined by the installer (optional setting)*

3. DDI/DID Destination: *Determined by the installer (extension number, group etc)*



(5) Appendix

UDP hole punching for keeping in Router/Firewall Port Forwarding

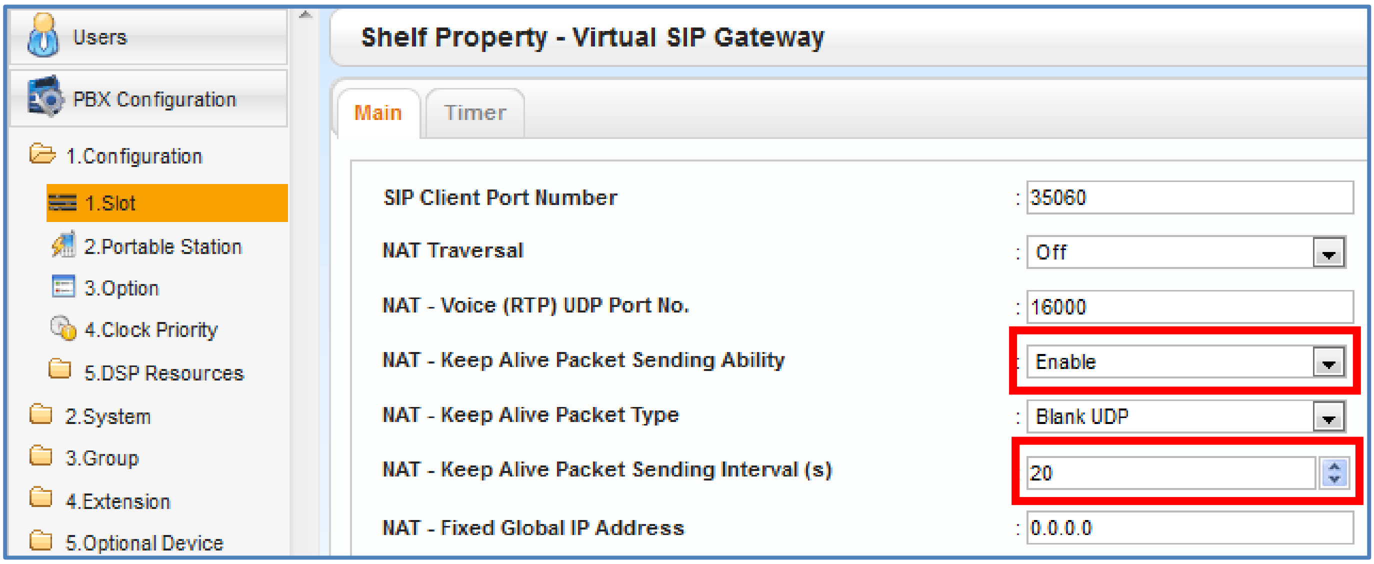
If REGISTER every 30 seconds is not effect to keep port forwarding for external router, configure the BLANK UDP to enable as keep-alive message on the PBX shelf property.

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.



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 *Peoplefone* source IP addresses.

END OF DOCUMENT