

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]**

Virtual 16-Channel SIP Trunk Card

Total number of cards 1

1

Shelf Property

Card Property

Port Property

Ins

Delete

[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 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	Connection	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	Backu
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

«	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	48221234000	account12345	passwordABCD	
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	Connec	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 **[Apply]** 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	48221234001	Extension	Disable	Disable
2	102	Development	48221234002	Extension	Disable	Disable
3	103	Sales	48221234000	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]**.

No.	Extension Number	Extension Name (20 characters)	CLIP ID	CLIP on Extension/CO	CLIR	COLR
1	101	Service	48221234001	Extension	Enable	Disable
2	102	Development	48221234002	Extension	Disable	Disable
3	103	Sales	48221234000	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: 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)*

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	48221234000	Sales	103	103	103	103
2	48221234001	Service	101	101	101	101
3	48221234002	Development	102	102	102	102
4						
5						
6						
7						
8						
9						
10						

(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.

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 Peoplefone source IP addresses.

END OF DOCUMENT