

**Setup Reference guide for KX-NS Series
(Tested with NS1000 Ver4.5)
“**Ecotel Sip-Trunk 2.0**” SIP Trunk service
with External Router**



Panasonic

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◆ SUMMARY

This document is a reference for configuring “**Ecotel Sip-Trunk 2.0**” 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 **Ecotel Sip-Trunk 2.0**.

◆ 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

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

Transfer / Forward

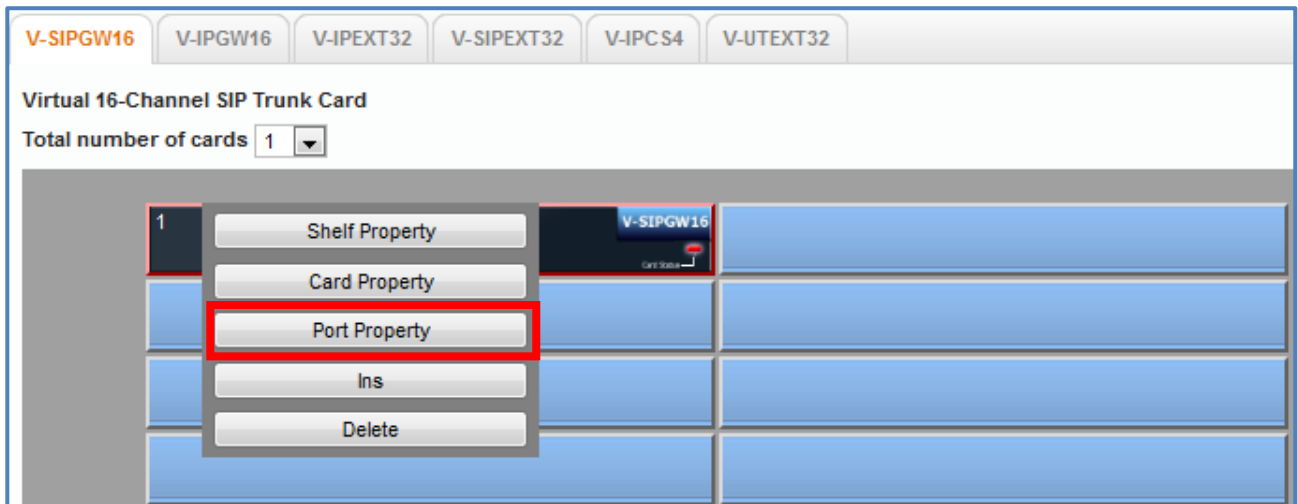
When the Transfer/forwarding destination is PSTN, the CLI is not call originator but PBX's CLI.

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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: *trunk.sip-ecotel.de – (ecotel provided)*
- 4. SIP Server Location – IP Address: *Not required*
- 5. SIP Server port Number: *Leave at 5060*
- 6. SIP Service Domain: *12345678.sip-ecotel.de (12345678=ContractNo.)*
- 7. Subscriber Number: *Not required*

Port Einstellungen - Virtuelles SIP-Gateway

Provider auswählen Provider hinzufügen

Hauptmenü Account Registrieren NAT Option Calling Party Called Party Voice/Fax RTP/RTCP T.38 T.38 Option DSP Zusatzdienste Erweiterte SIP-Option (Bits)

Nr.	Shelf	Slot	Port	Verbindung	Betrieb	Kanal-Attribut	Providername (20 Zeichen)	SIP-Server-Name (100 Zeichen)	SIP-Server-IP-Adresse	SIP-Server IP-Adresse für das Failover	SIP-Server-Portnummer	SIP-Service-Domain (100 Zeichen)
1	Virtuell	1	1	OUS	Public	Basiskanal	ecotel sip-trunk 2.0	trunk.sip-ecotel.de			5060	12345678.sip-ecotel.de
2	Virtuell	1	2	OUS	Public	Nicht verwendet					5060	
3	Virtuell	1	3	OUS	Public	Nicht verwendet					5060	
4	Virtuell	1	4	OUS	Public	Nicht verwendet					5060	
5	Virtuell	1	5	OUS	Public	Nicht verwendet					5060	

SIP Trunk – Port Property continued

[Account] Tab

- 1. User name: **Enter the Username as supplied by ecotel.**
 (Note this is username without @zertifizierung.sip-ecotel.de)
 For example: Username = 99123456
Enter: 99123456

- 2. Authentication ID: **Enter the Authentication ID as supplied by ecotel.**
 (Note this is Authentication ID without @zertifizierung.sip-ecotel.de)
 For example: Authentication ID = 99123456
Enter: 99123456

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

« Main Account Register NAT Option Calling Party Called Party Voice/FAX							
No.	Shelf	Slot	Port	Connect	User Name (64 characters)	Authentication ID (64 characters)	Authentication Password (32 characters)
1	Virtual	1	1	OUS	99123456	99123456	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	Connect	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.

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

(2) 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: 033123 45600 0
*Enter: **004933123456000***
- 2. **DDI/DID Name:** *Determined by the installer (optional setting)*
- 3. **DDI/DID Destination:** *Determined by the installer (extension number, group etc)*

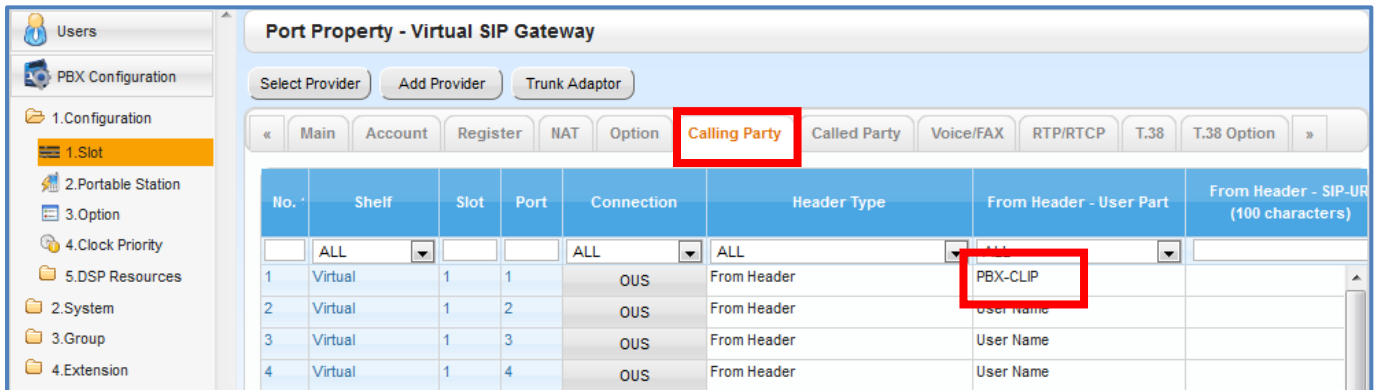
ID	DDI / DID Number (32 digits)	DDI / DID Name (20 characters)	DDI / DID Destination - Day	DDI / DID Destination - Lunch	DDI / DID Destination - Break	DDI / DID Destination - Night
1	004933123456000	Sales	103	103	103	103
2	004933123456001	Service	101	101	101	101
3	004933123456002	Development	102	102	102	102
4						
5						
6						
7						
8						
9						
10						

(3) Outgoing Call CLI

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

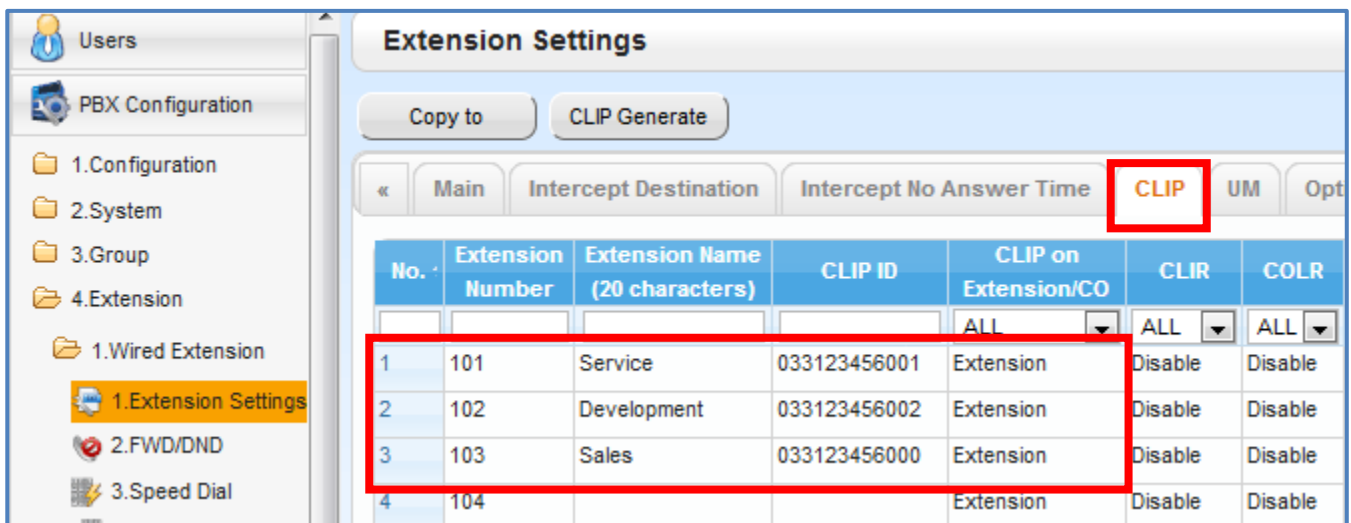
From Header – User Part:

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]**



(4) 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 [\[OK\]](#).

The screenshot displays the 'Extension Settings' window. On the left is a navigation tree with 'PBX Configuration' > '4.Extension' > '1.Wired Extension' > '1.Extension Settings' selected. The main area shows the 'CLIP' tab. At the top are 'Copy to' and 'CLIP Generate' buttons. Below are tabs for 'Main', 'Intercept Destination', 'Intercept No Answer Time', 'CLIP', 'UM', and 'Opt'. A table lists four extensions with their settings. The 'CLIR' column for extension 101 is highlighted with a red box and set to 'Enable'.

No.	Extension Number	Extension Name (20 characters)	CLIP ID	CLIP on Extension/CO	CLIR	COLR
1	101	Service	033123456001	Extension	Enable	Disable
2	102	Development	033123456002	Extension	Disable	Disable
3	103	Sales	033123456000	Extension	Disable	Disable
4	104			Extension	Disable	Disable

(5) Appendix

UDP hole punching for keeping in Router/Firewall Port Forwarding

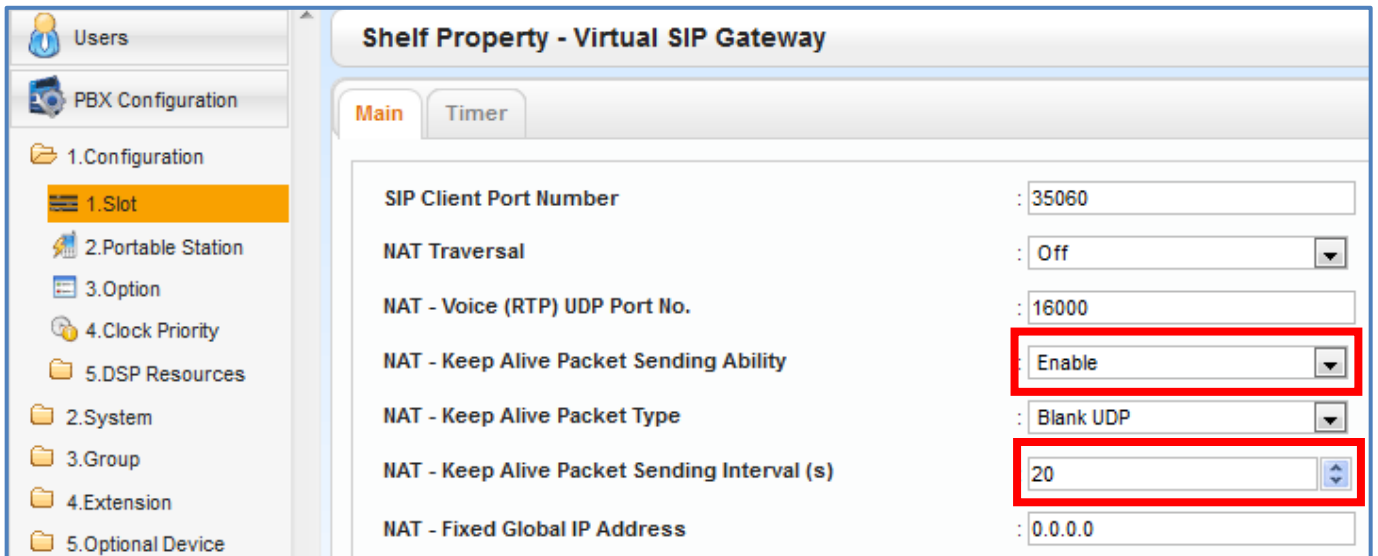
If required, for keeping port forwarding table in installed router, configure the BLANK UDP to enable as keep-alive with UDP hole punching on the PBX shelf property.

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 ecote! source IP addresses.

END OF DOCUMENT