

Setup Reference guide for KX-NS700/1000
Version 4.2 Firmware
“British Telecom” SIP Trunk service
(with External Router)



Panasonic

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◆ SUMMARY

This document is a reference for configuring “**British Telecom**” SIP trunks with Panasonic 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 BT.

The BT Sip services covered by this Inter-Operability Test include:

- **BT Wholesale SIP Trunks (WSIPT)**
- **BT Global Services One Voice SIP Trunk UK**

◆ 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

SIP Registration

As per British Telecom procedures, the example configurations use a Global IP Address for authentication.

Transfer Function

British Telecom does not support REFER transfer method – use CO to CO Transfer.

FAX

Recommended G.711 Inband codec.

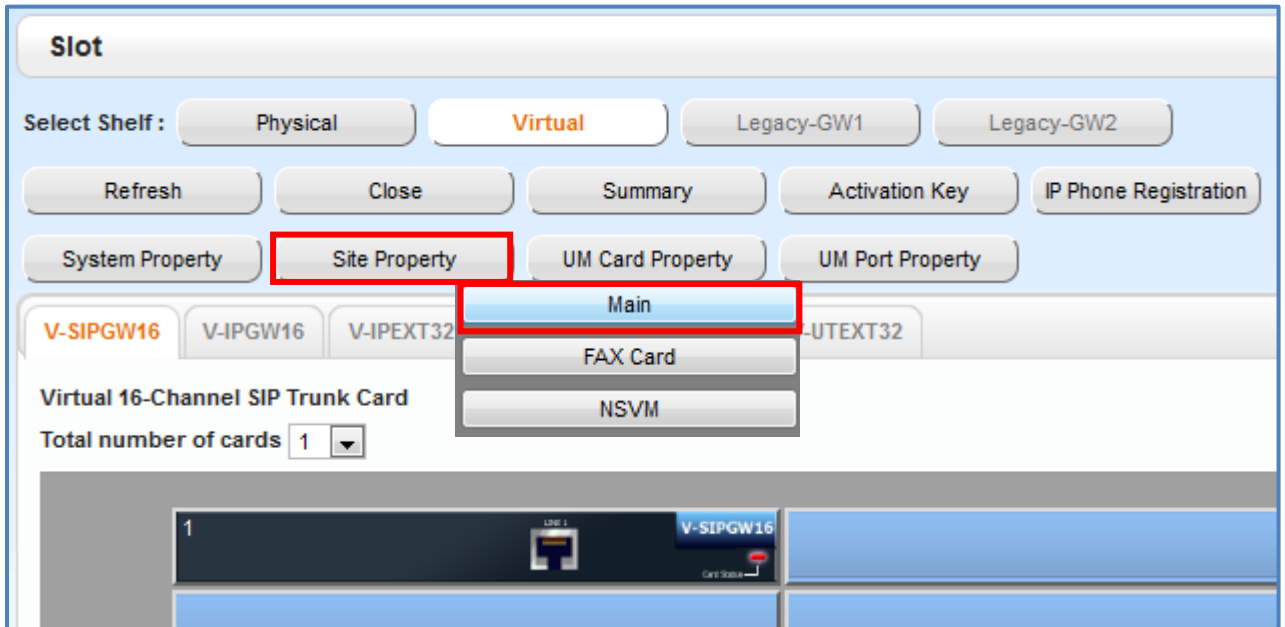
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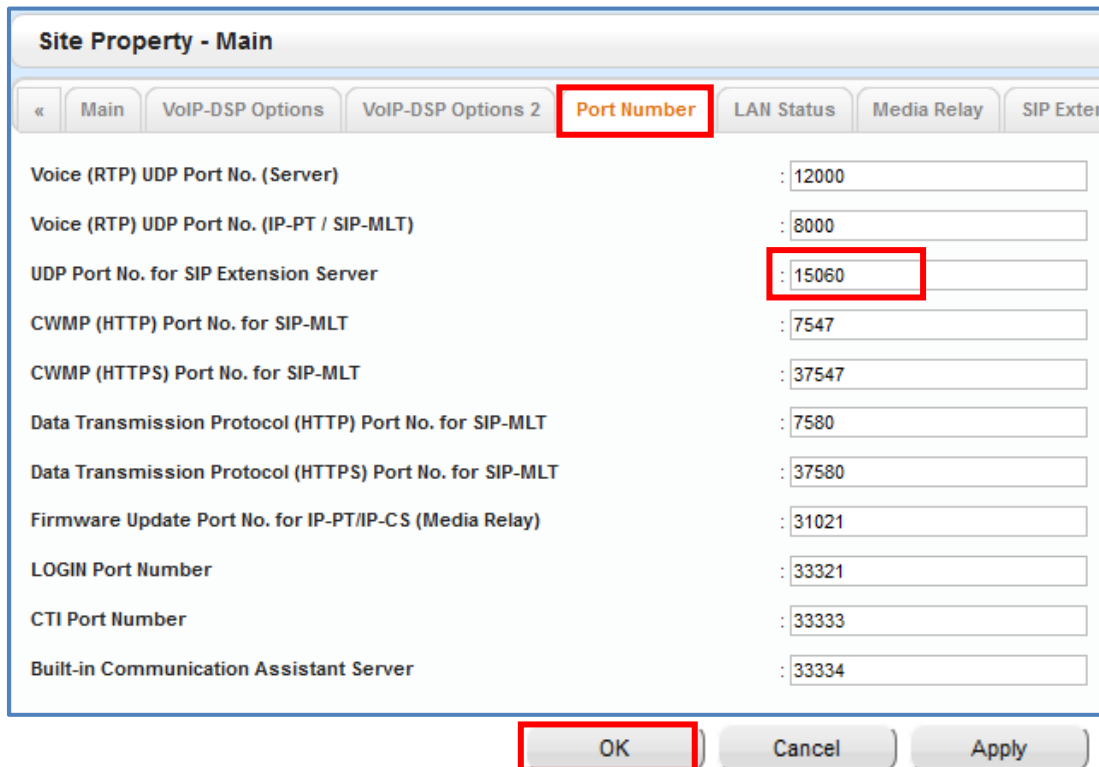
(1) SIP Trunk and Extension Port Number Configuration

- Note:
SIP Trunk Port number for British Telecom must be **5060** therefore SIP Extension ports are reconfigured, in this example to port **15060**.

Click **[1. Slot]** Move mouse over **[Site Property]** and click **[Main]**.



Click **[Port Number]** tab and change **[UDP Port No. for SIP Extension Server]** (Default:5060) to **other number**. (i.e.15060). and click **[OK]**.



Go to **[1.Configuration – 1.Slot]**

Click **[Virtual]** and move mouse over **[V-SIPGW16]**, and click **[Ous]**. Click **[OK]** pop-up window.

Move mouse over **[V-SIPGW16]** again, and click **[Shelf Property]**.

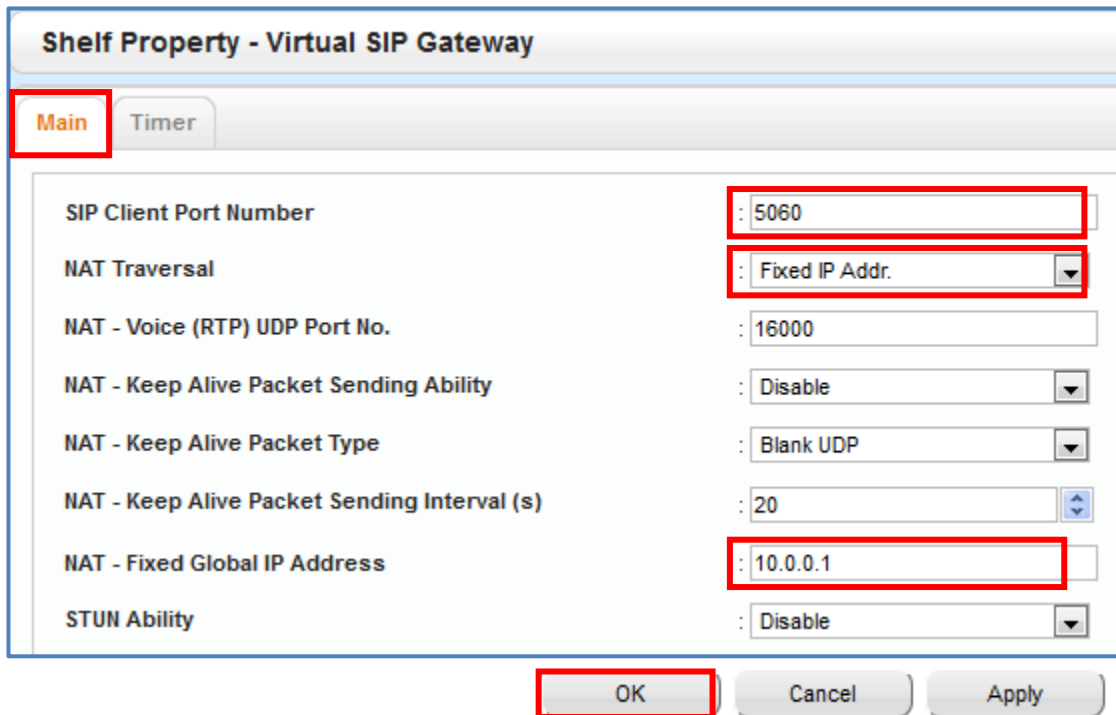


Select **[Main]** tab and change the two items.

[SIP Client Port Number] (Default:35060) to **5060**

[NAT – Traversal] **Fixed IP Addr.**

[NAT – Fixed Global IP] **10.0.0.1 (Enter your actual Global IP address)**



and click **[OK]**.

Move mouse over **[V-SIPGW16]** again, and click **[Ins]**.

***Note: Save the System data and Restart the PBX after making these port changes.**

Port Forwarding

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

udp port 5060 – to NS LAN IP address (e.g. 192.168.0.101)

udp port range 16000-16511 (RTP) – to NS DSP1 LAN IP address (192.168.0.102)

For larger installations where additional DSP resources are installed:

udp port range 16512-17023 (RTP) – to NS DSP2 LAN IP address (192.168.0.103)

udp port range 17024-17535 (RTP) – to NS DSP3 LAN IP address (192.168.0.104)

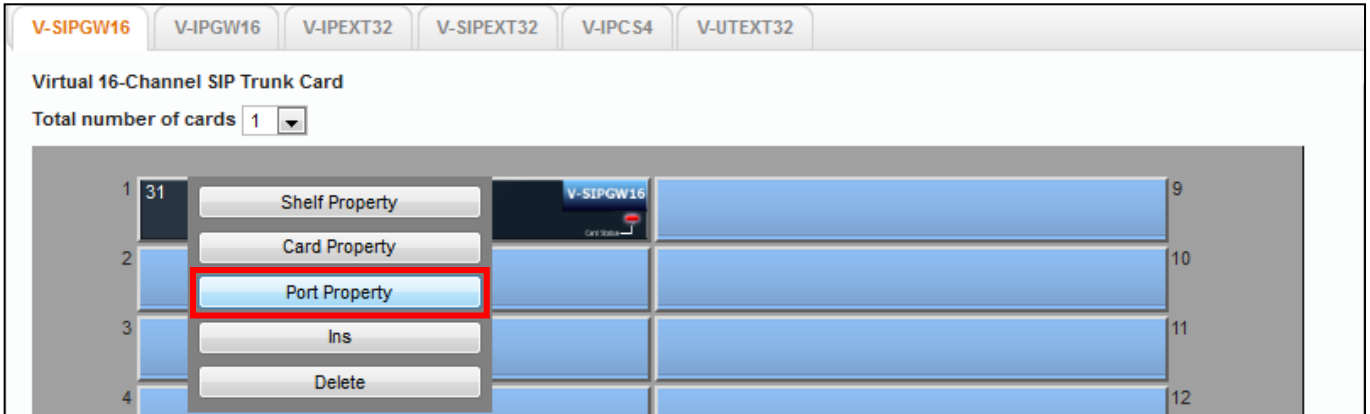
udp port range 17536-18047 (RTP) – to NS DSP4 LAN IP address (192.168.0.105)

IMPORTANT!

To secure the PBX from illegal attacks, please restrict the above port forwarding ports to only be accessible from the British Telecom source IP addresses.

(2) Provisioning the SIP Trunk
SIP Trunk – Port Property

Move mouse over **[V-SIPGW16]** and click **[Ous]** and Select **[Port Property]**



SIP Trunk – Port Property continued

[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: *192.65.221.26 – (British Telecom provided)*
- 5. SIP Server IP for Failover: *192.65.221.23 – (British Telecom provided)*
- 6. SIP Server port Number: *Leave at 5060*
- 7. SIP Service Domain: *Not required*
- 8. 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 »

No.	Shelf	Slot	Port	Connecti	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
1	Virtual	1	1	ALL	SIP Provider	Public	Basic chann	BT Trunk-1		192.65.221.26	192.65.221.23	5060
2	Virtual	1	2		Fault	SIP Provider	Not Used					5060

SIP Trunk – Port Property continued

[Account] Tab

- 1. User name: **Enter the SIP Account (User name) as supplied by British Telecom.**
(Note this is user name without @192.65.221.26)
For example: SIP Account (User name) = +445512340100
Enter: +445512340100

- 2. Authentication ID: **Enter the Authentication ID as supplied by British Telecom.**
(Note this is authentication ID without @192.65.221.26)
For example: Authentication ID = +445512340100
Enter: +445512340100

- 3. Authentication Password: **Not required.**

No.	Shelf	Slot	Port	Connecti	User Name (64 characters)	Authentication ID (64 characters)	Authentication Password (32 characters)
1	Virtual	1	1	OUS	+445512340100	+445512340100	

SIP Trunk – Port Property continued

[Register] Tab

- 1. Register Ability: **Disable**
- 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**

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	Registrar Server Port Number for Failover
1	Virtual	1	1	OUS	Disable	3600	Enable				5060	300

[Voice/FAX] Tab

- 1. IP Codec Priority 1st: **G.711A**
- 2. IP Codec Priority 2nd: **G.711Mu**
- 3. IP Codec Priority 3rd: **G.729A**

No.	Shelf	Slot	Port	Connect	IP Codec Priority 1st	IP Codec Priority 2nd	IP Codec Priority 3rd	Packet Sampling Time (G.711A)
1	Virtual	1	1	OUS	G.711A	G.711Mu	G.729A	20ms

[Option] Tab

- Session Timer Ability: **Enable (Active)**
- Session Expire Timer: **600**

No.	Shelf	Slot	Port	Connect	Session Timer Ability	Session Expire Timer (s)	Session Refresh Method	Session Incoming Refresher Request
1	Virtual	1	1	OUS	Enable(Active)	600	re-INVITE	UAC

(3) Incoming Call Routing

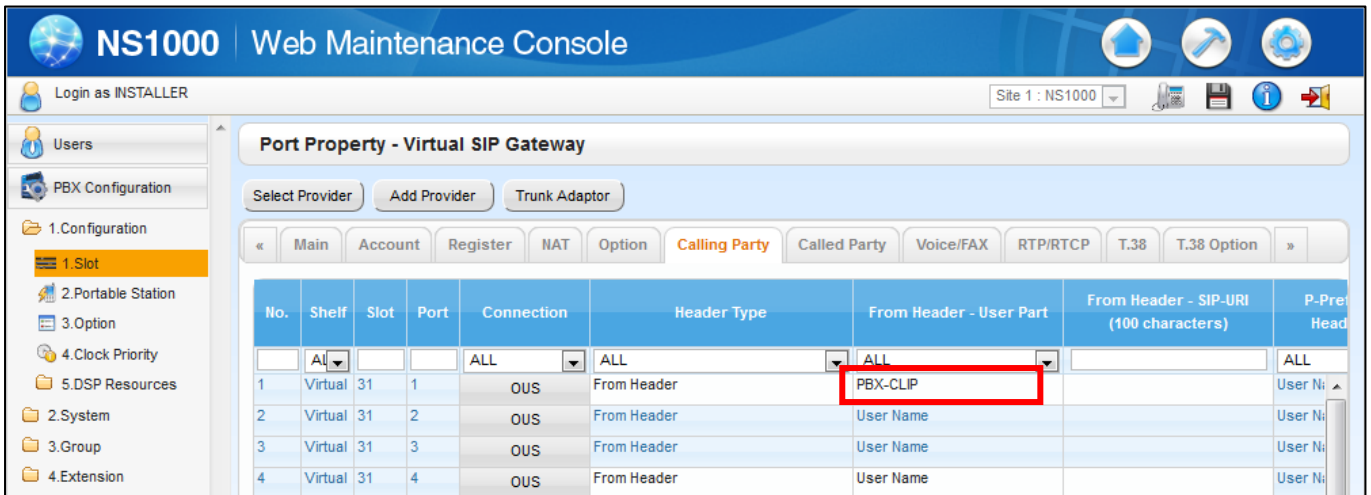
Go to **[CO & Incoming call]** and select **[3.DDI /DID Table]**

- 1. DDI/DID Number:** *Enter the DDI number in the appropriate format (see below)*
Example: 44-55-1234-0100
Enter: 445512340100
- 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
1	445512340100	Sales	201	201	202
2	445512340101	Service	202	202	202
3	445512340102	Development	203	203	202
4					
5					
6					
7					
8					
9					
10					
11					

(4) Outgoing Call CLI

Move mouse over **[V-SIPGW16]** click **[Ous]** Select **[Port Property]** and **[Calling Party] Tab**
From Header – User Part: PBX-CLIP



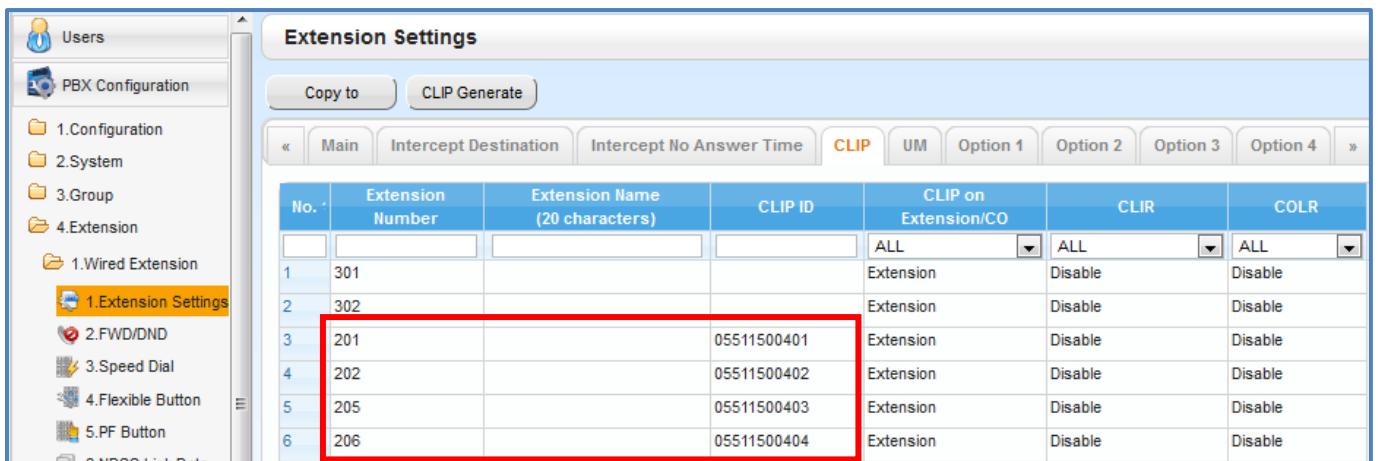
Click **[OK]**.

Go to **[4.Extension, 1.Wired Extension, 1.Extension Settings]** & select **[CLIP]**

[CLIP] tab

Enter a valid CLI number for each required extension in the **CLIP ID** field

Click **[OK]**.



(5) CLIR Outgoing Call (Withholding Number)

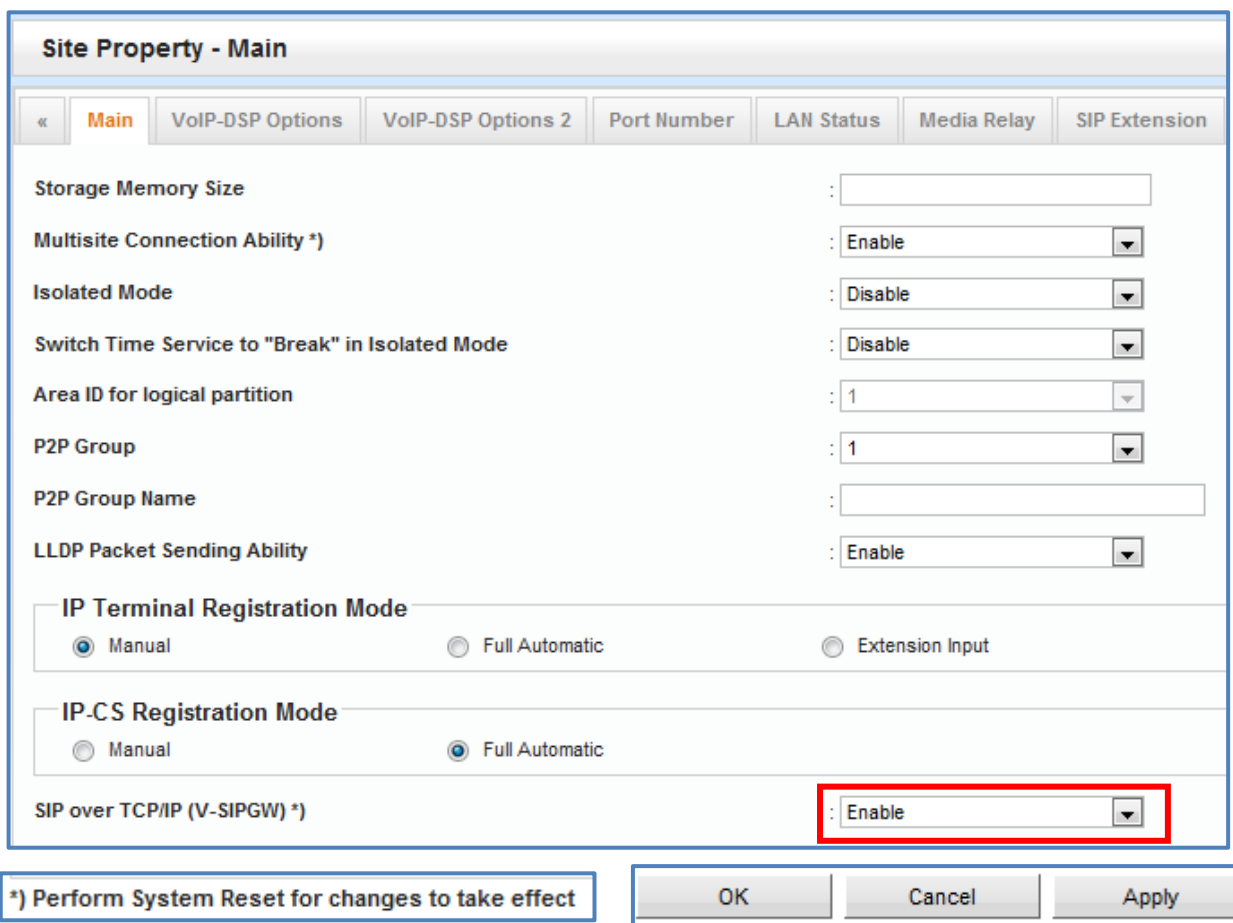
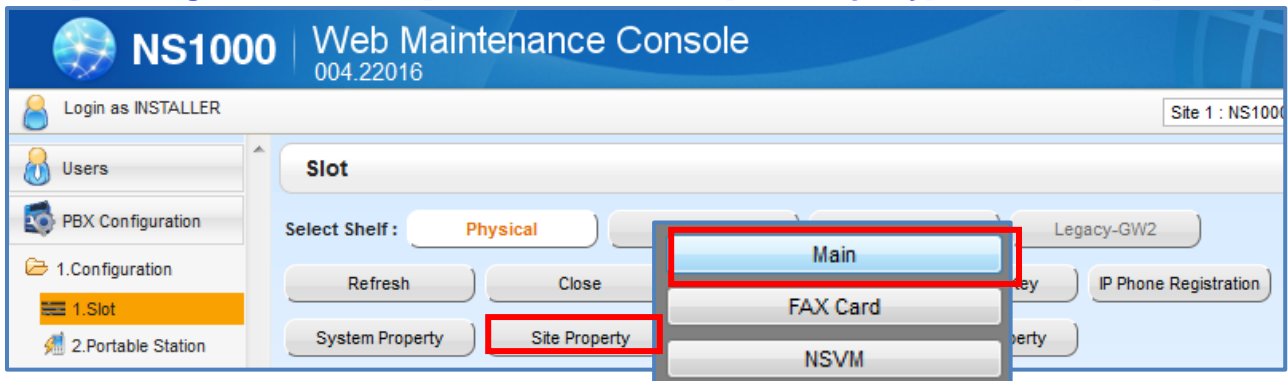
Place call to called party with a 141 prefix, call proceed to endpoint with withholding CLI information.
 (e.g.) 141 055 1234 0020

(6) Enabling SIP over TCP/IP

IMPORTANT NOTES

Enabling SIP over TCP/IP will delete all existing SIPGW and IPGW (H323) cards in the system. Also IPGW (H323) Trunks cannot be configured (see Feature Guide: [Direct SIP Connection](#)). To use SIP over TCP with H323 IPGW (or SIP over UDP) in the same system requires a One-Look network with a second NS controller to provide a second IPGW (or SIPGW) shelf.

Go to [1.Configuration, 1. Slot] & Move mouse over [Site Property] and click [Main]



Click [OK] then Save the System data and Restart the PBX to apply the change to TCP/IP. (Existing SIPGW and IPGW cards in the system will be deleted at this point).

END OF DOCUMENT