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





Version 1.0 (PSCEU) 5 October 2015

# 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**.

			cy-GW1 Legacy-GW2
Refresh	Close	Summary	Activation Key IP Phone Registration
System Property S	te Property	UM Card Property	UM Port Property
		Main	UTENT22
V-SIPGW10 V-IPGW10	V-IPEAT 32	FAX Card	-OTEXT32
Virtual 16-Channel SIP Trunk	Card	NSVM	
Total number of cards 1			•
-			
1		V-SIPGW16	

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].

Site Property - Main												
« Main VolP-DSP Options Vo	IP-DSP Options 2	Port Number	LAN Status	Media Relay	SIP Exte							
Voice (RTP) UDP Port No. (Server)			: 12000									
Voice (RTP) UDP Port No. (IP-PT / SIP-MLT) : 8000												
UDP Port No. for SIP Extension Server			: 15060									
CWMP (HTTP) Port No. for SIP-MLT			: 7547									
CWMP (HTTPS) Port No. for SIP-MLT			: 37547									
Data Transmission Protocol (HTTP) Po	rt No. for SIP-MLT		: 7580									
Data Transmission Protocol (HTTPS) P	ort No. for SIP-MLT		: 37580									
Firmware Update Port No. for IP-PT/IP-	CS (Media Relay)		: 31021									
LOGIN Port Number			: 33321									
CTI Port Number			: 33333									
Built-in Communication Assistant Ser	ver		: 33334									
		ок	Cancel	) App	oly )							

3

## 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].



5060

Select [Main] tab and change the two items.

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

- [NAT Traversal]
- [NAT Fixed Global IP]

Fixed IP Addr.

10.0.0.1 (Enter your <u>actual</u> Global IP address)

Shelf Property - Virtual SIP Gateway	
Main Timer	
SIP Client Port Number	: 5060
NAT Traversal	: Fixed IP Addr.
NAT - Voice (RTP) UDP Port No.	: 16000
NAT - Keep Alive Packet Sending Ability	: Disable
NAT - Keep Alive Packet Type	: Blank UDP 💌
NAT - Keep Alive Packet Sending Interval (s)	: 20
NAT - Fixed Global IP Address	: 10.0.0.1
STUN Ability	: Disable
ОК	Cancel Apply

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 <u>only</u> be accessible from the British Telecom source IP addresses.

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

#### V-SIPGW16 V-IPGW16 V-IPEXT32 V-SIPEXT32 V-IPC S4 V-UTEXT32 Virtual 16-Channel SIP Trunk Card Total number of cards 1 9 1 31 Shelf Property Card Property 10 2 Port Property 11 3 Ins Delete 12

#### 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. 1	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	
	Al 🗸			ALL 🚽	ALL 👻	ALL 👻	ALL 👻						
1	Virtual	1	1	OUS	SIP Provider	Public	Basic chanr	BT Trunk-1		192.65.221.26	192.65.221.23	5060	
2	Virtual	1	2	Fault	SIP Provider	Public	Not Used					5060	

#### SIP Trunk – Port Property continued

#### [Account] Tab

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

« Main	Accoun	t Re	gister	NAT Option Calling	Party	Called Party	Voice/	FAX	RTP/RTCP	T.38	
No. 1 Shelf	Slot	Port	Connecti	User Name (64 characters)		Authentication ID (64 characters)			Authentication Password (32 characters)		
	1	1	ALL 👻	+445512240100		5512240100					

# SIP Trunk – Port Property continued

# [Register] Tab

- 1. Register Ability:
- 2. Register Interval:
- 3. Un-Register Ability:
- 4. Registrar Server Name:
- 5. Registrar Server IP Address:
- 6. Registrar Server port number:

#### Disable

Leave at 3600 Leave enabled Not required Not required

Leave at 5060

Users	Port	Prope	rty - V	/irtual	SIP Ga	teway								
PBX Configuration	Select	ect Provider ) Add Provider ) Trunk Adaptor )												
1.Configuration	« I	Main A	Accoun	t Re	gister	NAT	Option Cal	ing Party Ca	lled Party Voic	e/FAX RTP/	RTCP T.38	T.38 Option	ı »	1
2.Portable Station							Deviates		Desisters	Destinters	Registrar	Deviatore		
3.Option	No.	Shelf	Slot	Port	Conne	Register Ability	Sending	Ability when	Server Name	Server	Server IP Address for	Server Port	Re	gist: Ir
4.Clock Priority							Interval (s)	port INS	(100 characters)	IP Address	Failover	Number		
5.DSP Resources	1	Virtual	1	1	ous	Disable	3600	Enable				5060	300	*
<ul><li>5.DSP Resources</li><li>2.System</li></ul>	1	Virtual	1	1	OUS	Disable	3600	ALL . Enable				5060	300	

# [Voice/FAX] Tab

- 1. IP Codec Priority 1st:
- 2. IP Codec Priority 2nd:
- 3. IP Codec Priority 3rd:

•	« Main Account Register NAT Option Calling Party Called Party Voice/FAX RTP/RTCP T.38 T.38 Optio										
	No. <sup>-</sup> Shelf Slot Port Connecti IP Codec IP Codec IP Codec Sampling Time Priority 1st Priority 2nd Priority 3rd (G.711A)										
		Al 🗸			ALL 👻	ALL 🔻	ALL 🔻	ALL 💌	ALL 🔻		
	1	Virtual	1	1	OUS	G.711A	G.711Mu	G.729A	20ms		

# [Option] Tab

Session Timer Ability: Session Expire Timer:

#### Called Party Voice/FAX RTP/RTCP Main Account Register NAT Option Calling Party Session Incoming Session Expire Shelf Slot No. Port Refresher Ability Timer (s) **Refresh Method** Request Al 👻 ALL 👻 ALL ALL ALL -• Ŧ 600 re-INVITE UAC Virtual 1 Enable(Active) 1 OUS

# Enable (Active)

600

G.711A

G.729A

G.711Mu

# (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 Determined by the installer (optional setting)

- 2. DDI/DID Name:
- 3. DDI/DID Destination:

Determined by the installer (optional setting) Determined by the installer (extension number, group etc)

Users		DDI	/ DID Table								
PBX Configuration	A	Automatic Registration Name Generate Destination Setting									
1.Configuration	ſ	DDI / DID Number DDI / DID Name DDI / DID Destination - DDI / DID Destination - DDI / DID Destination									
2.System 3.Group			(32 digits)	(20 characters)	Day	Lunch	Break				
4.Extension	1		445512340100	Sales	201	201	202				
5.Optional Device	2		445512340101	Service	202	202	202				
6.Feature	3		445512340102	Development	203	203	202				
7.TRS	4										
8.ARS	5										
9.Private Network	6	;									
🗁 10.CO & Incoming Call	7										
💐 1.CO Line Settings	8	5									
🐲 2.DIL Table & Port	9	)									
Settings	1	0									
3.DDI / DID Table	1	1									

# (4) Outgoing Call CLI

# Move mouse over [V-SIPGW16] click [Ous] Select [Port Property] and [Calling Party] Tab

From Header – User Part:

**PBX-CLIP** 

🛞 NS1000	NS1000 Web Maintenance Console									
Login as INSTALLER							Site 1 : NS	1000 🖵 归 🤅	) 🇾	
👸 Users	Port	t Prop	erty -	Virtua	I SIP Gateway					
PBX Configuration	Select	Provider		ld Provid	ler Trunk Adap	otor				
1.Configuration 1.Slot	«	« Main Account Register NAT Option Calling Party Called Party Voice/FAX RTP/RTCP T.38 T.38 Option »								
<ul><li>2.Portable Station</li><li>3.Option</li></ul>	No.	Shelf	Slot	Port	Connection	Header Type	From Header - User Part	From Header - SIP-URI (100 characters)	P-Pref Head	
a.Clock Priority		Al 🗸			ALL 👻	ALL	ALL		ALL	
5.DSP Resources	1	Virtual	31	1	ous	From Header	PBX-CLIP		User Ni 🔺	
2.System	2	Virtual	31	2	OUS	From Header	User Name		User Ni	
3.Group	3	Virtual	31	3	OUS	From Header	User Name		User Ni	
4.Extension	4	Virtual	31	4	OUS	From Header	User Name		User Ni	

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].

Users	1	E	xte	ension Settings								
PBX Configuration		Copy to CLIP Generate										
1.Configuration 2.System		« Main         Intercept Destination         Intercept No Answer Time         CLIP         UM         Option 1         Option 2         Option 3         Option 4         »										
3.Group			۱o. ۱	Extension Number	Extension Name (20 characters)	CLIP ID	CLIP on Extension/CO	CLIR	COLR			
							ALL 🔻	ALL 💌	ALL 🔻			
1.Wired Extension		1		301			Extension	Disable	Disable			
😁 1.Extension Settings	L	2		302			Extension	Disable	Disable			
🧐 2.FWD/DND	L	3		201		05511500401	Extension	Disable	Disable			
3.Speed Dial	L	4		202		05511500402	Extension	Disable	Disable			
4.Flexible Button	=	5		205		05511500403	Extension	Disable	Disable			
5.PF Button		6		206		05511500404	Extension	Disable	Disable			

# (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 <u>delete</u> all existing SIPGW and IPGW (H323) cards in the system. Also IPGW (H323) Trunks cannot be configured (see Feature Guide: <u>Direct SIP Connection</u>). 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]

💮 NS1000	Web Maintenance Cc 004.22016	onsole	
Login as INSTALLER			Site 1 : NS1000
Users	Slot		
PBX Configuration	Select Shelf : Physical		Legacy-GW2
1.Configuration	Refresh Close	Main	tey IP Phone Registration
E 1.Slot	Sustam Broparty Site Broparty	FAX Card	uerty .
2.Portable Station	System Property Sile Property	NSVM	Jerty

Site Property - Main						
« Main VoIP-DSP Options	VoIP-DSP Options 2	Port Number	hber LAN Status Media Relay		SIP Extension	
Storage Memory Size						
Multisite Connection Ability *)			: Enat	: Enable		
Isolated Mode			: Disa	: Disable		
Switch Time Service to "Break" in Isolated Mode			: Disa	: Disable		
Area ID for logical partition			: 1	: 1		
P2P Group			: 1	: 1		
P2P Group Name			:			
LLDP Packet Sending Ability			: Enat	: Enable		
IP Terminal Registration Mode						
Manual	Full Automatic			Extension Input		
IP-CS Registration Mode						
Manual	Full Automat	ic				
SIP over TCP/IP (V-SIPGW) *)			: Enat	ole	•	
) Perform System Reset for cha	ок		Cancel	Apply		

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