# GETTING STARTED PANASONIC KX-NS SERIES

### Overview

This document is a reference for configuring the Kwebbl SIP trunk into KX-NS Series systems and includes the settings required for incoming Call DDI routing and Outgoing Call CLI presentation.

#### Codecs

- G.711 A

FAX

- T.38 not supported.

CLI

- No need for P-Asserted-Identity header and P-Preferred-Identity header.

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# 1. Configuring a SIP Trunk

### SIP Trunk – Port Property

#### Set the [V-SIPGW16] card to [OUS] and then select [Port Property]

4		8		12	
3	Trunk	7 V-SIPEXT32		11 IP-CS	IP-CS
2		6 V-UTEXT32		10	
1	Shelf Property			9	
	Card Property				
	Port Property	and the second s	-	and in case of the local division of the loc	
	Ins		n a	mm e a	
	Delete	]			
Vietu	al Slot				

#### [Main] Tab

- 1. Channel Attribute:
- 2. Provider Name:
- 3. SIP Server Location Name:
- 4. SIP Server Location IP Address:
- 5. SIP Server port Number:
- 6. SIP Service Domain:
- 7. Subscriber Number:

Basic Channel Enter a name – reference only 1.trunk.sip.kwebbl.net Not required Leave at 5060 Not required Not required

	Mair	Account	Regist	ter NA	T Option	Calling Party Calle	d Party Voice/FAX	RTP/RTCP T.38	T.38 Option DSP
	No.	Shelf	Slot	Port	Connection	Trunk Property	Channel Attribute	Provider Name (20 characters)	SIP Server Name (100 characters)
E		ALL 🗸		1	ALL 🗸	ALL 🗸	ALL	No.	
1		Virtual	1	1	OUS	Public	Basic channel	kwebbl-1	1.trunk.sip.kwebbl.net
2		Virtual	1	2	OUS	Public	Not Used		
3		Virtual	1	3	OUS	Public	Not Used		
4		Virtual	1	4	OUS	Public	Not Used		

## [Account] Tab

1. User name:	Enter the Username (can be found in the panel).
	(Note this is username without @sip.example.com) For example:
	Username = abcdefg Enter: abcdefg
2. Authentication ID:	Enter the Authentication ID (can be found in the panel)
	(Note this is Authentication ID without @sip.example.com) For
	example: Authentication ID = abcdefg Enter: abcdefg
3. Authentication Password	Enter the password (can be found in the panel)
	For example: password = opqrstuvwxyz Enter: opqrstuvwxyz

Ma	in Accoun	t Regi	ster N	AT Option	C	Called F	Party Voice/FAX	RTP/RTCP	T.38	T.38 Option	DSP
No.	Shelf	Slot	Port	Connection		User Name (64 characters)	Authentica (64 charae			ntication Passw 32 characters)	ord
	ALL		1	ALL					Í		_
1	Virtual	1	1	OUS	a	bcdefg	abcdefg		opqrstuvv	wxyz	
2	Virtual	1	2	ous							_
3	Virtual	1	3	OUS							

## [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:

Leave enabled Leave at 3600 Leave enabled Not required Not required Leave at 5060

40	Main	Account	Reg	gister	NAT O	otion Callin	ng Party	Called P	arty Voice/FA)	C RTP/RTC	CP T.38 T.38	Option »
No.	Shelf	Slot	Port	Conner	Register Ability	Register Sending Interval (s)		gister hen port IS	Registrar Server Name (100 characters)	Server	Registrar Server IP Address for Failover	
	AL		-	ALL	A11					1		
1	Virtual	1	1	ous	Enable	3600	Enable					5060

Click [OK] to apply the changes.

## 2. Outgoing Call CLI

#### Go to [Calling Party] Tab

From Header – User Part:

PBX-CLIP

8 Users	Port	Prope	erty - Vir	tual S	IP Gate	way							
PBX Configuration	Select	Provider	) Add P	rovider	) Trun	k Adaptor							
1.Configuration	6.6	Aain	Account	Regi	ster	IAT Option	C	alling Party Call	ed Party Vo	ce/FAX	RTP/RTCP	T.38	T.38 Option a
EE 1.Slot				1 0.0560									
2.Portable Station				1812		100			-	1	2 2 2		From Header - SIP-
3.Option	No.				Port				гТуре			er Part	(100 characters)
3 4.Clock Priority		ALL				ALL		ALL			_		
5.DSP Resources	1	Virtual	-	1	1	OUS		From Header		PBX-C	LIP	-	
2.System	2	Virtual		1	2	OUS		From Header		User 1	rame		
C 3.Group	3	Virtual		1	3	OUS		From Header		Usert	lame		
4.Extension	4	Virtual		1	4	OUS		From Header		User	lame		

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]

Mai	n Intercept Des	stination Intercept No A	Answer Time (	CLIP UM Option	1 Option	n 2 Optio
▲ No.	Extension Number	Extension Name (20 characters)	* CLIP ID	CLIP on Extension/CO	CLIR	COLR
	1	1		ALL [	V ALL V	ALL 🗸
1	101	Ext_101	4912300000001	Extension	Disable	Disable
2	102	Ext_102	4912300000002	Extension	Disable	Disable
3	103			Extension	Disable	Disable
4	104			Extension	Disable	Disable

## 3. CLIR Outgoing call (Anonymous)

Go to [4.Extension] - [1.Wired Extension] - [1.Extension Settings] and select [CLIP] tab Under CLIR: select Enable and Click [OK].

<b>▲</b> No.	Extension Number	Extension Name (20 characters)	CLIP ID	CLIP on Extension/CO	CLIR	COLR
				ALL	ALL 🗸	ALL
1	101	Ext_101	4912300000001	Extension	Enable	[ isable
2	102	Ext_102	491230000002	Extension	Disable	Disable
3	103			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: 4912300000001 (DDI Number) Enter: 4912300000001 (DDI Number)
2. DDI/DID Name:	Determined by the installer (optional setting)
3. DDI/DID Destination:	Determined by the installer (extension number, group etc)

- ID o	(32 digits)	DDI / DID Name (20 characters)	Destination - Day	Destination - Lunch	Destination - Break	Destination - Night
45	912300000001	Sales	101	101	101	101
49	912300000002	Service	102	102	102	102

## 5. Appendix

#### UDP hole punching for keeping Port Forwarding

If Kwebbl's Keep Alive message (e.g. OPTIONS/BLANK UDP packets) 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.

👸 Users	Shelf Property - Virtual SIP Gateway		
PBX Configuration	Main Timer		
😂 1.Configuration			
1.Siot	SIP Client Port Number	: 35060	
A 2.Portable Station	NAT Traversal	: Off	
3.Option	NAT - Voice (RTP) UDP Port No.	: 16000	
3 4.Clock Priority		. 10000	
5.DSP Resources	NAT - Keep Alive Packet Sending Ability	Enable	
2.System	NAT - Keep Alive Packet Type	: Blank UDP	
C 3.Group	NAT - Keep Alive Packet Sending Interval (s)	20	\$
4.Extension			(A)
5.Optional Device	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 Kwebbl source IP addresses.