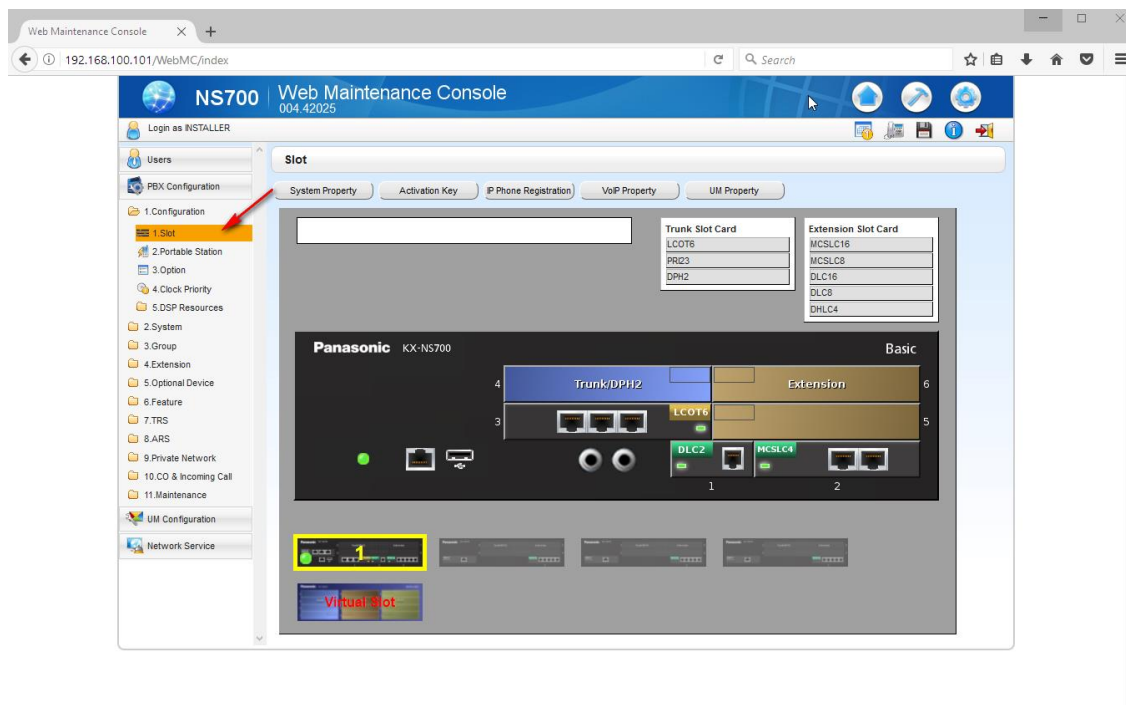


Valcom Session Initiation Protocol (SIP) VIP devices are compatible with the Panasonic Unified Communications Platform. The Valcom device can be added to the Panasonic as a SIP trunk.

The configuration example in this document is based on a Panasonic KX-NS700 software version 4.42025 and a Valcom VIP-821A. The SIP trunk will be configured for inbound and outbound. This example should be similar for any other Panasonic IP PBX in the Unified Communications Platform.

The following steps outline the typical configuration process:

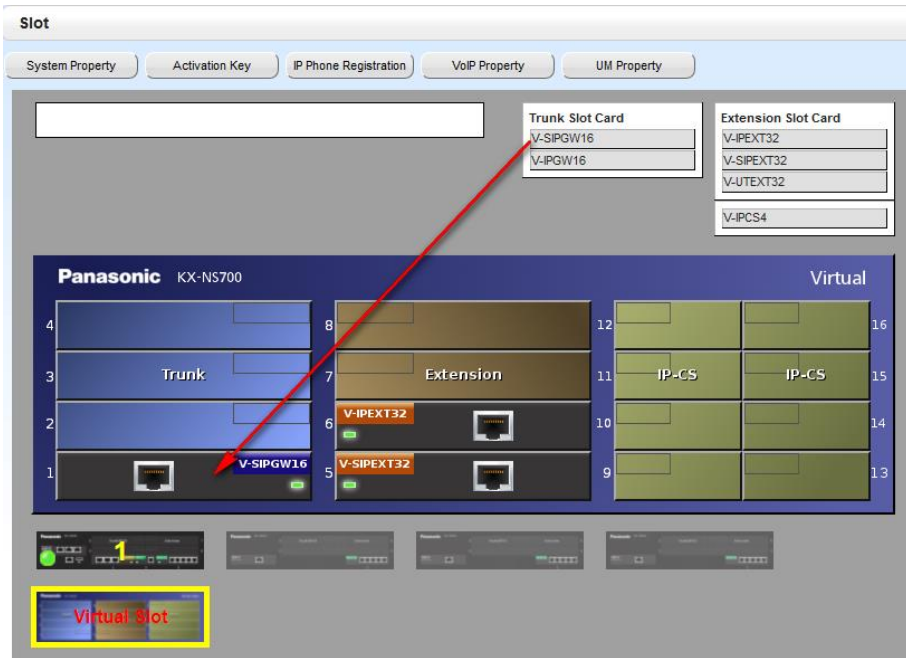
1. Open the Web Management console for the Panasonic PBX via a web browser using the IP address of the PBX and login as INSTALLER.
2. The PBX Configuration/Slot screen should display by default.



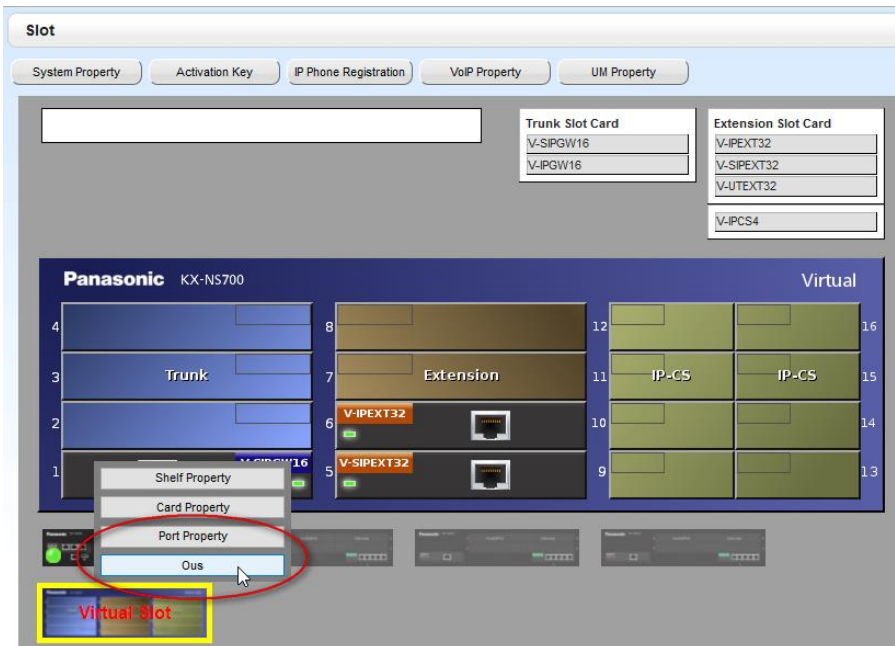
3. Hover the mouse over the Virtual Slot and click on Select Shelf.



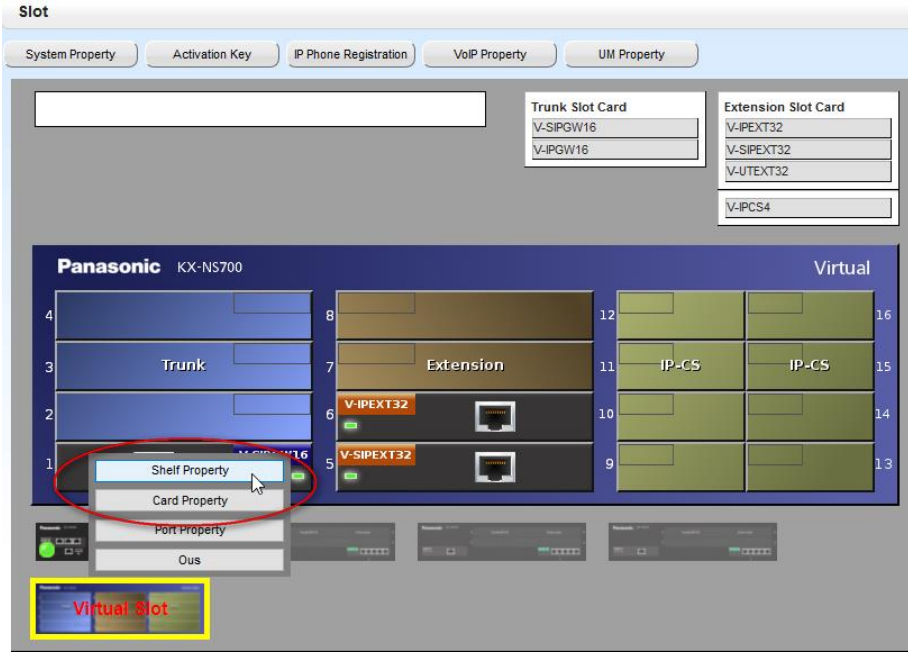
4. Virtual Slot screen should display and have at least 1 V-SIPGW16 Trunk Slot Card.  
If not, drag and drop one from the Trunk Slot card window into the next available trunk Slot.



5. The card must be taken out of service before making configuration changes. Hover the mouse over the V-SIPGW16 card and click on OUS to take the card out of service. Note that any other SIP trunk that is installed on the same card will also become OUS.



6. Hover the mouse over the V-SIPGW16 card and click on Shelf Property

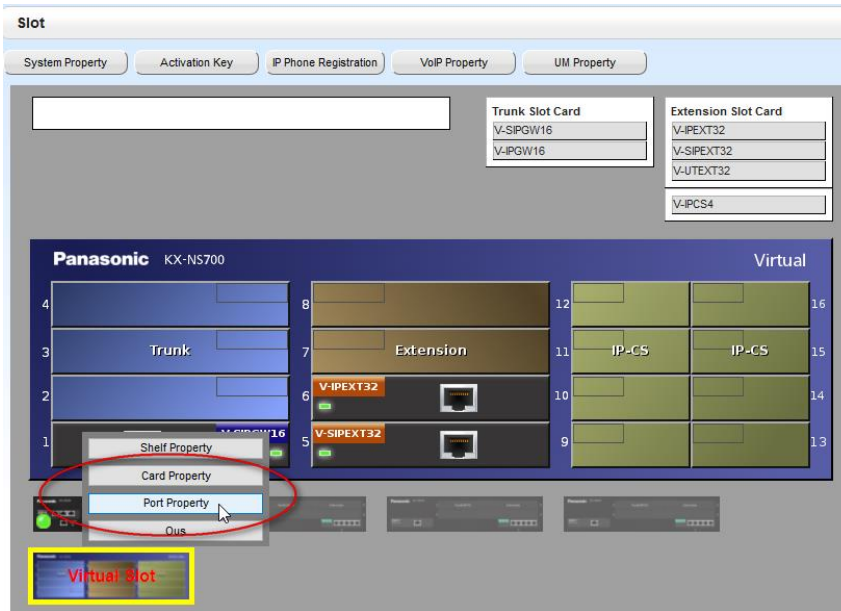


- From the Shelf property screen set the “SIP Client Port Number” (you can use the default 35060 or set your own. For this example we will use 35060) and “SIP Called Party Check Ability” to Disable (High->Low).

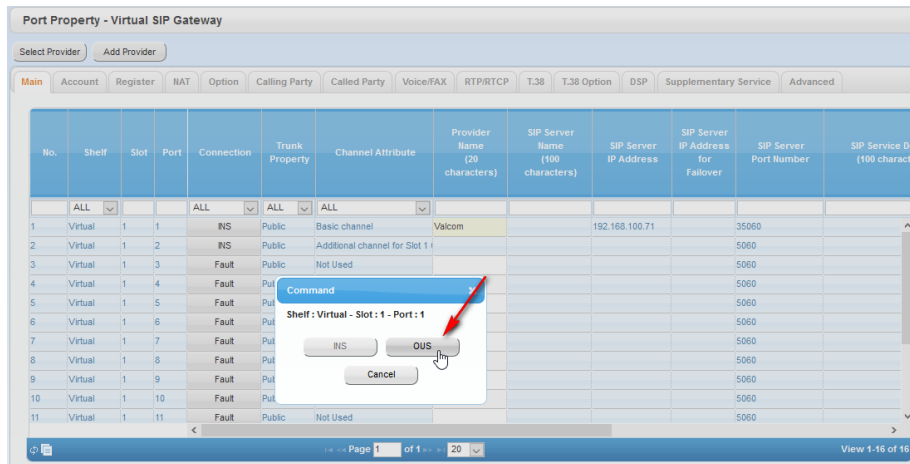
Shelf Property - Virtual SIP Gateway	
Main Timer	
SIP Client Port Number	: 35060
NAT Traversal	: Off
NAT - Voice (RTP) UDP Port No.	: 16000
NAT - Keep Alive Packet Sending Ability	: Disable
NAT - Keep Alive Packet Type	: Blank UDP
NAT - Keep Alive Packet Sending Interval (s)	: 20
NAT - Fixed Global IP Address	: 0.0.0.0
STUN Ability	: Disable
STUN Client Port Number	: 33478
STUN External Address Detection Retry Counter	: 1
STUN Resending Interval	: 500 ms
SIP Called Party Number Check Ability	: Disable(High->Low)
SIP Called Party Number Search Mode	: Mode1
Symmetric Response Routing Ability	: Enable
100rel Ability	: Enable(Passive)
Ringback Tone to Outside Caller	: Disable

- Click Apply button on the screen at the lower right and then the OK button to return to the Slot screen.

9. Hover the mouse over the V-SIPGW16 card and click on "Port Property".



10. At this point, the cards should still be out of service. However, if not, click on the Connection field for the port being configured. A Command window will come up. Click OUS.



- On the Main tab of the Port Property screen, choose Basic Channel from the Channel Attribute column, then add a Provider Name (eg. Valcom) SIP Server IP address (eg 192.168.100.71), and change the SIP server port number to the setting from step 7 (SIP Client Port Number). (Do not click on OK at this point)

Port Property - Virtual SIP Gateway

Select Provider Add Provider

Main Account Register NAT Option Calling Party Called Party Voice/FAX RTP/RTCP T.38 T.38 Option DSP Supplementary Service Advanced

No.	Shelf	Slot	Port	Connection	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	INS	Public	Basic channel	Valcom		192.168.100.71		35060
2	Virtual	1	2	INS	Public	Additional channel for Slot 1					5060
3	Virtual	1	3	Fault	Public	Not Used					5060
4	Virtual	1	4	Fault	Public	Not Used					5060
5	Virtual	1	5	Fault	Public	Not Used					5060
6	Virtual	1	6	Fault	Public	Not Used					5060
7	Virtual	1	7	Fault	Public	Not Used					5060
8	Virtual	1	8	Fault	Public	Not Used					5060
9	Virtual	1	9	Fault	Public	Not Used					5060
10	Virtual	1	10	Fault	Public	Not Used					5060
11	Virtual	1	11	Fault	Public	Not Used					5060

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- On the Account tab of the Port Property screen set the “User Name” to any value (Valcom devices do not accept registration. This info is required by Panasonic PBX but not required by Valcom devices). Likewise the Authentication ID and password can be any value. (Do not click on OK at this point)
- On the Register tab of the Port Property screen set “Register Ability” to “Disable”. (Do not click on OK at this point)
- On the Calling Party tab of the Port Property screen set the “From Header – User Part” to “PBX-CLIP”. (Do not click on OK at this point)
- On the Called Party tab of the Port Property screen set the “Type” to “To Header”. Click Apply.
- Click on the Connection field for the port being configured. A Command window will come up. Click on INS to place the SIP Trunk port back in service. If you had taken the card OUS, go to the next step.
- Click OK to save all changes and return to the Slot Screen. If you had the card still OUS, hover over the SIPGW card and choose INS.

18. From the PBX Configuration menu tree click on **3. Group** to configure Trunk information.
  - a. Click on **1. Trunk Group**-
    - i. Click on **1. TRG Settings** – Select an available Trunk Group and provide a “Group Name” and an unused Dialing Plan Table number. (e.g. Trunk Group 3 for the Group Name, 3 for the Dialing Plan Table).
    - ii. Click OK to continue
    - iii. Click on **4. Dialing Plan** – Select the Dialing plan table number chosen previously
    - iv. and set Digits dialed pattern. (e.g. table #3 “Leading Number” 767XXX where anything dialed outgoing that starts with 767 will be directed to this SIP trunk).

19. Click OK to continue

20. From the PBX Configuration Menu tree select **10. CO & Incoming Call**

- a. Click on **1. CO Line Settings** – Select an available CO line number for a V-SIPGW16 card
  - i. Provide CO name (eg. Valcom SIP Trunk)
  - ii. Enter the Trunk Group number from step 18 (eg. Group Number 3)
  - iii. Click OK to continue

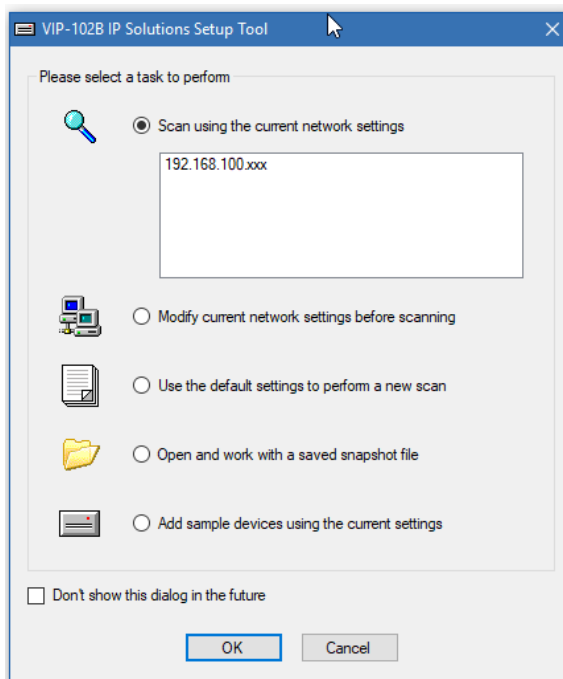
CO Line Number	Shelf	Slot	Port	Card Type	CO Name (20 characters)	Trunk Group Number
	ALL			ALL		ALL
1	1	3	1	LCOT6	Valcom-Trunk1	2
2	1	3	2	LCOT6		1
3	1	3	3	LCOT6		1
4	1	3	4	LCOT6		1
5	1	3	5	LCOT6		1
6	1	3	6	LCOT6		1
7	Virtual	1	1	V-SIPGW16	Valcom-SP-Trunk	3
8	Virtual	1	2	V-SIPGW16		1
9	Virtual	1	3	V-SIPGW16		1
10	Virtual	1	4	V-SIPGW16		1
11	Virtual	1	5	V-SIPGW16		1
12	Virtual	1	6	V-SIPGW16		1
13	Virtual	1	7	V-SIPGW16		1
14	Virtual	1	8	V-SIPGW16		1
15	Virtual	1	9	V-SIPGW16		1
16	Virtual	1	10	V-SIPGW16		1
17	Virtual	1	11	V-SIPGW16		1
18	Virtual	1	12	V-SIPGW16		1

- b. *If inbound calls to the PBX from a Valcom device is not required, the following step may be omitted.*  
 Click on **3. DDI/DID Table** – Select an available slot and enter DDI/DID Number (eg. 23456) and then the destination this incoming number should go to (eg. extension 1041) add this or a different extension to all applicable times – Day, Lunch, Break and Night.


21. Click OK to continue.



22. At this point we can now configure the VIP Device in the VIP-102B tool. The latest version of the VIP-102B IP Solutions Setup Tool may be downloaded from our website at <http://www.valcom.com/vipsetuptool> .
23. After installing the VIP-102B tool, launch it and select “Scan using the current network settings” if you have already predefined the subnet the VIP device is on. Otherwise if you are on the same subnet you can select “Use the default settings to perform a new scan”. Then click OK to start the scan.



24. If successful, the device should appear in the discovery window. Click Continue.
25. If you need to assign an IP address to this device or set it to DHCP refer to the VIP-102B Reference manual on our website <http://www.valcom.com/vipsetuptool> .
26. After assigning the IP address and rescan there will be additional tabs to program. Specifically for this example with the VIP device we will focus on the SIP tab to configure the device for the SIP Trunk that was created on the Panasonic IP PBX.

27. After keying in the necessary fields, click the Apply button at the bottom then click on the  icon to update, then when prompted for Reset, click Yes.

Summary Properties Network Time Channels Relays Group Membership SIP

1

Phone Number: 7677070 number called from PBX

Description:

Authentication Name: 12345 Auth ID and Auth Password from PBX

Secret: 12345

Realm:

SIP Servers:

	Server	Port
Primary	192.168.100.101	35060
Backup 1		5060
Backup 2	IP of PBX	5060
Backup 3		5060

Register:

DNS SRV:

Outbound Proxy:

Outbound Port: 35060 Change to the port number from step #7

SIP Port: 35060 Change to the port number from step #7

Idle Timeout (secs): 0

RTP Port: 20000

Max Call Timer (secs): 0

CID Number: 7677070

CID Name:

Auto Destination: 23456 Number to call on PBX (DDI/DID) for inbound call (not needed for outbound only)

28. For outbound call test for this example you would dial, from a phone, the access code to get to the SIP trunk line you created. In the CO Line settings we used line 7, which belongs to Trunk Group 3. This can be accessed by dialing 803 (8 to access trunk group and 03 for the trunk group). When secondary dial tone is heard, dial the number assigned to the VIP device, in this example it is 7677070.

29. The VIP device in this example is a VIP-821A.

- a. For outbound calls: When the connection is made from the PBX via the SIP trunk the VIP-821A will answer the call and open its analog connection. This analog connection may be connected to an analog page control or amplifier that requires a loop start trunk port. The analog connected device should either go direct to speakers or provide dial tone for a zone selection. If no zones, simply speak from the phone used to call the VIP-821A via the SIP trunk. Paging should come through the speakers. If there are zones, you select the zone first then page.



## *Panasonic Unified Communications Platform SIP Trunk Configuration Guide*

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- b. For Inbound calls: A call can be placed from an analog talkback paging controller. In this case a “button” is pressed that causes the controller to go off hook on its connection to the VIP-821A. The Auto destination field that was set up in the example (23456) is used to place a SIP call back to the PBX via the SIP trunk. The PBX will see the 23456 and should find the phone number in the DDI/DID table and route the call to extension 1041 per our example.