

Panasonic KX-NS700 **Trusted IP SIP Trunk Service** **aka: Non Registering SIP Trunk** **Telquest Tech Support**

1. Mouse Over

2. Click Here

3. Click Here

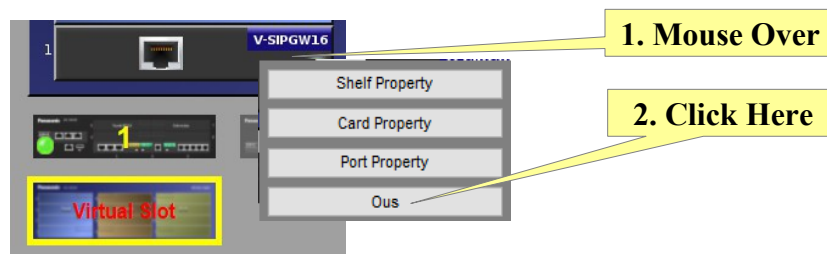
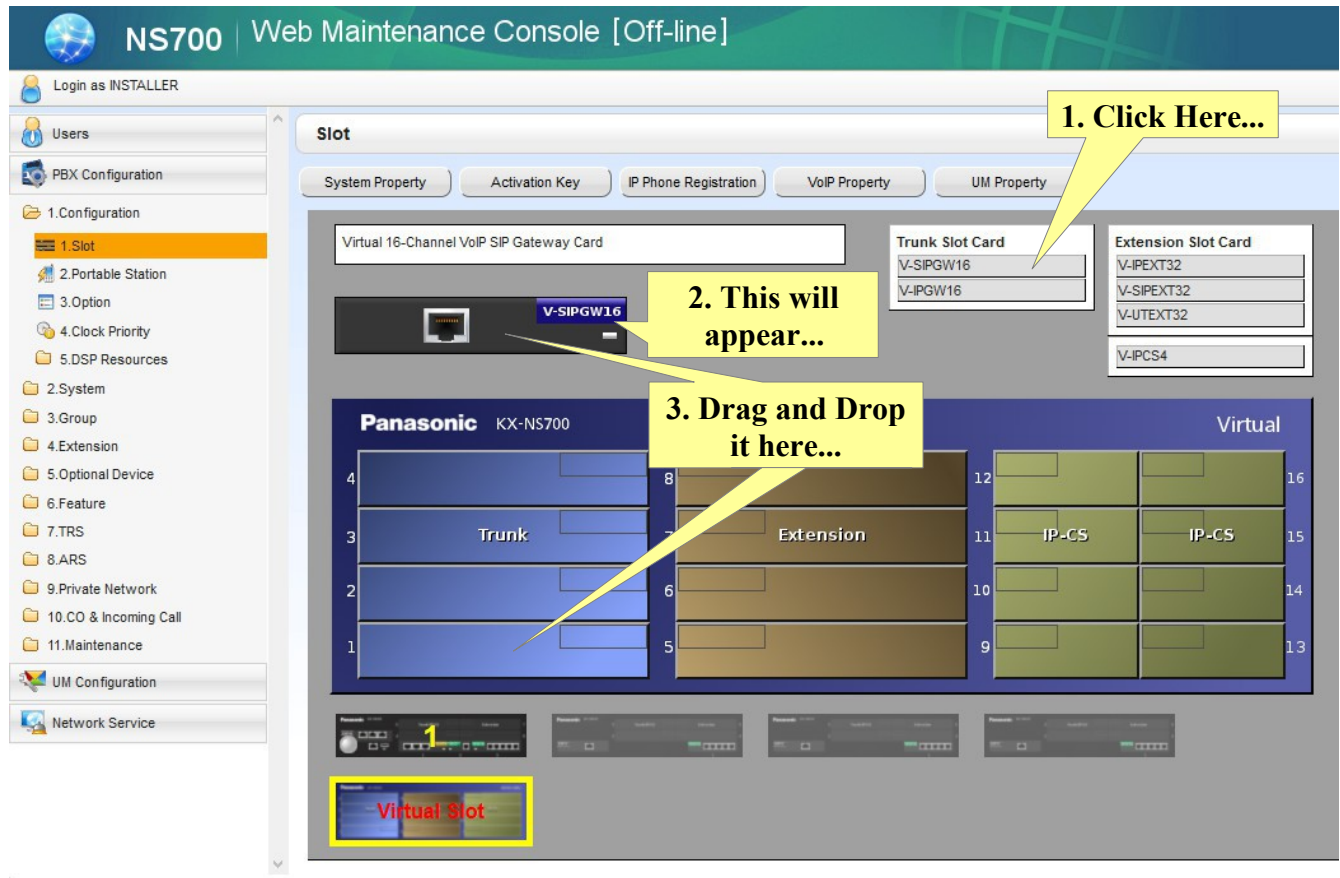
4. Change to 5065

2. Click Here...

1. Click Here...

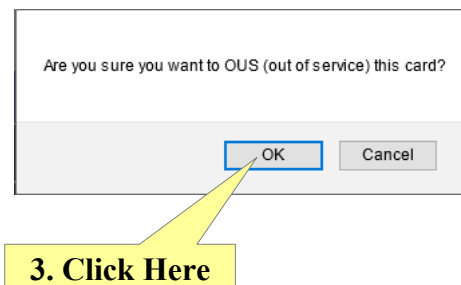
3. Mouse Over and Click Here...

This screen will appear: V-SIPGW16 = Virtual SIP Trunk Card for 16 Trunks and or Channels

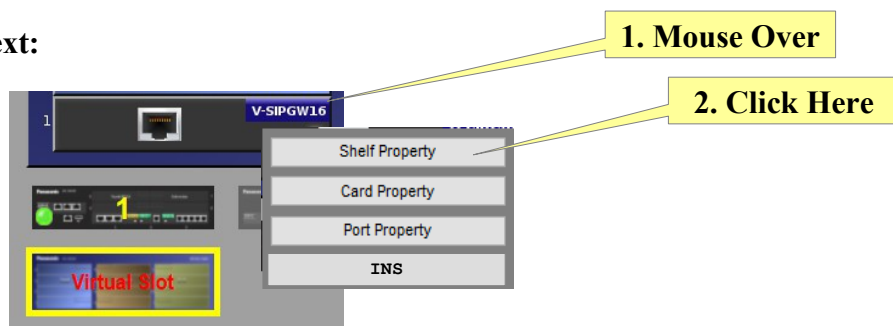


You need to take the card out of service before you make the changes on Page 3.

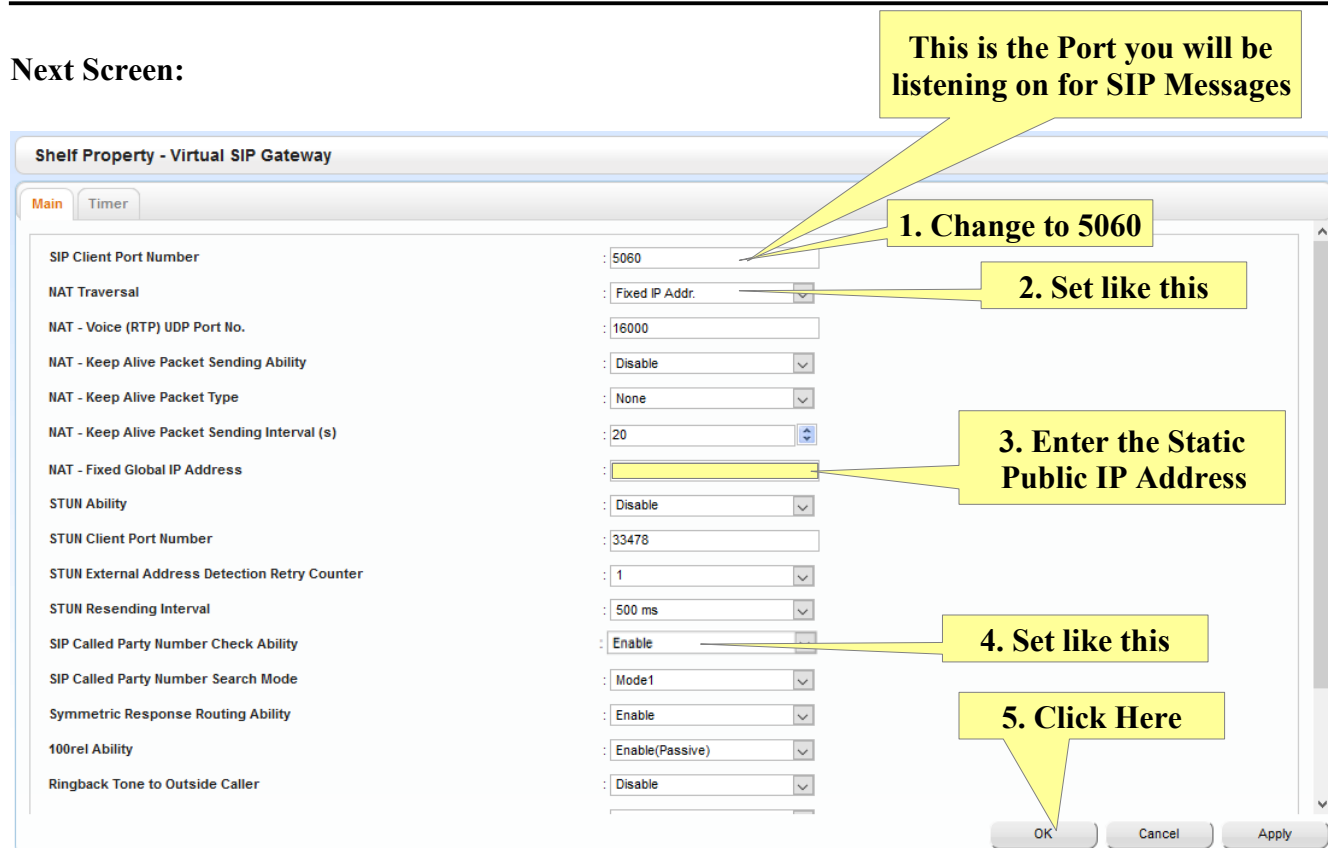
When you are done, you need to Reset the KSU.



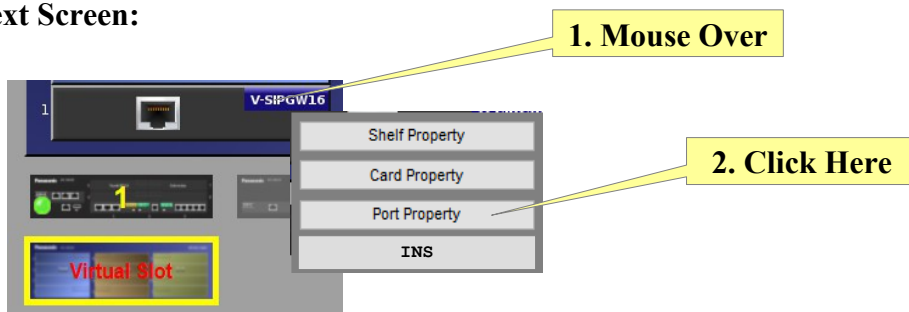
Next:



Next Screen:



Next Screen:



Next Screen: I have cut down the screen to show only the areas of concern....

This example is set up for a SIP Trunk with a total of 5 Channels

1. Don't Touch

2. First Channel like this...

3. Additional Channels like this...

4. Info ONLY Has nothing to do with connection...

No.	Shelf	Slot	Port	Connection	Trunk Property	Channel Attribute	Provider Name (20 characters)
1	Virtual	1	1	OUS	Public	Basic channel	VOIP Innovations
2	Virtual	1	2	OUS	Public	Additional channel for Slot 1 Ch 1	
3	Virtual	1	3	OUS	Public	Additional channel for Slot 1 Ch 1	
4	Virtual	1	4	OUS	Public	Additional channel for Slot 1 Ch 1	
5	Virtual	1	5	OUS	Public	Additional channel for Slot 1 Ch 1	

This is the same screen, I have just scrolled over to the right....

This is the Port you will be Sending out SIP Messages

See below

No.	Shelf	Slot	Port	SIP Server Name (100 characters)	SIP Server IP Address	SIP Server IP Address for Failover	SIP Server Port Number	SIP Service Domain (100 characters)	Subscriber Number
1	Virtual	1	1	SIP.VOIPINNOV.COM	64.125.14.21		5060		
2	Virtual	1	2				5060		
3	Virtual	1	3				5060		
4	Virtual	1	4				5060		
5	Virtual	1	5				5060		

You only need to enter either the SIP Server Name **OR** the SIP Server IP Address.

You **DO NOT** need to enter both.

Enter which ever the SIP Provider has given to you as well as the SIP Server Port Number.

This is the same screen, I have just scrolled over more to the right.....

No.	Shelf	Slot	Port	Server IP Address for Failover	SIP Server Port Number	SIP Service Domain (100 characters)	Subscriber Number	P2P Group	P2P Group Name
	ALL							ALL	
1	Virtual	1	1		5060		2125551212	1	
2	Virtual	1	2		5060			1	
3	Virtual	1	3		5060			1	
4	Virtual	1	4		5060			1	
5	Virtual	1	5		5060			1	

Next Screen:

No.	Shelf	Slot	Port	Connection	User Name (64 characters)	Authentication ID (64 characters)	Authentication Password (32 characters)
	ALL			ALL			
1	Virtual	1	1	OUS	Info from SIP Provider	Info from SIP Provider	Info from SIP Provider
2	Virtual	1	2	OUS			
3	Virtual	1	3	OUS			
4	Virtual	1	4	OUS			
5	Virtual	1	5	OUS			

Enter the information provided by the SIP Provider.

Note: 1

They do not always call their info by the same Names shown above.

Note: 2

It has been noted that you **MUST** enter information in these 3 areas for calls to go out.
At the very least, enter the Main Telephone Number in all 3 areas.

You only need to add the info to the Basic Channel (first line) as shown above.

Enter the “User Name” / Main Telephone Number.

If you do not have an “Authentication ID”, or “Authentication Password” just enter the “User Name” / Main Telephone Number in these 2 areas.

The NS700 will not allow you to enter just a “User Name”.

Next Screen:

1. Click Here

2. Set each to Disable

No.	Shelf	Slot	Port	Connection	Register Ability	Register Sending Interval (s)	Un-Register Ability when port INS	Registrar Server Name (100 characters)	Registrar S IP Address
1	Virtual	1	1	OUS	Disable	3600	Enable		
2	Virtual	1	2	OUS	Disable	3600	Enable		
3	Virtual	1	3	OUS	Disable	3600	Enable		
4	Virtual	1	4	OUS	Disable	3600	Enable		
5	Virtual	1	5	OUS	Disable	3600	Enable		

Next Screen:

1. Click Here

2. Set like this

No.	Shelf	Slot	Port	Connection	IP Codec Priority 1st	IP Codec Priority 2nd	IP Codec Priority 3rd	Packet Sampling Time (G.711A)	Packet Sampling Time (G.711Mu)
1	Virtual	1	1	OUS	G.711Mu	G.711A	G.729A	20ms	20ms
2	Virtual	1	2	OUS	G.711Mu	G.711A	G.729A	20ms	20ms
3	Virtual	1	3	OUS	G.711Mu	G.711A	G.729A	20ms	20ms
4	Virtual	1	4	OUS	G.711Mu	G.711A	G.729A	20ms	20ms
5	Virtual	1	5	OUS	G.711Mu	G.711A	G.729A	20ms	20ms

Note:

You can leave this at the default settings if you want to.

Most SIP Providers in the USA accept the G.711Mu Codec.

G711.A is used mainly in Europe.

3. Click Here

Now you need to go back to the Slot and put the V-SIPGW16 Card In Service (INS)

1

NS700 | Web Maintenance Console [Off-line]

Login as INSTALLER

Users

PBX Configuration

1. Configuration

1. Slot

2. Portable Station

3. Option

4. Clock Priority

5. DSP Resources

2. System

3. Group

4. Extension

5. Optional Device

6. Feature

7. TRS

8. ARS

9. Private Network

10. CO & Incoming Call

11. Maintenance

UM Configuration

Network Service

Slot

System Property

Activation Key

IP Phone Registration

VoIP Property

UM Property

Trunk Slot Card

LCOT6

PRIZ3

DPH2

Extension Slot Card

MCSLC16

MCSLC8

DLC16

DLC8

DHLC4

Panasonic KX-NS700 Basic

4

3

Trunk/DPH2

Extension

6

5

DLC2

MCSLC4

1

2

Select Shelf

2. Mouse Over and Click Here...

3. Mouse Over

1

V-SIPGW16

Shelf Property

Card Property

Port Property

INS

4. Click Here

LED turns Green

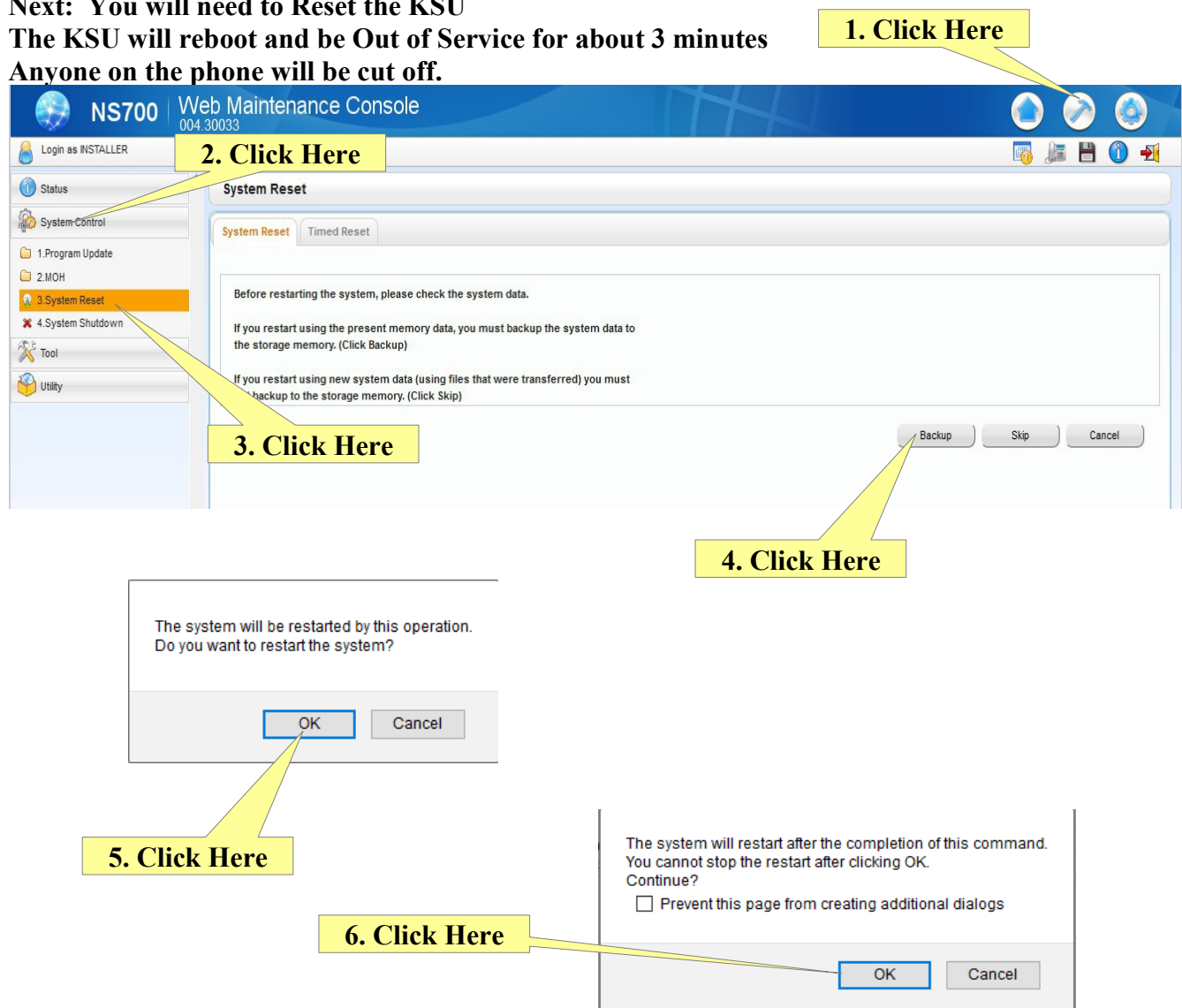
Virtual Slot

1

V-SIPGW16

Virtual Slot

Next: You will need to Reset the KSU
The KSU will reboot and be Out of Service for about 3 minutes
Anyone on the phone will be cut off.



When the system comes back up, log back into it:

Go back to the Port Properties, as shown on Page 4, where it says “Don't Touch”.

If the “Connection” shows INS (In Service) then you can start testing your SIP Service.

If the “Connection” shows OUS (Out of Service”) just click on each one and change it to INS.

Note:

When it says INS that does not mean that the SIP Service is working.

You could have a single or multiple errors in your SIP configuration and it will still indicate INS.

What it does mean is that the KSU is ready to use the SIP Service.

Assigning SIP Service to Dial 9

The NS700 uses Trunk Group Number 1 as the Dial 9 Access Trunks.

1. Click Here

2. Click Here

3. Click Here

4. Name Them

5. Set to 1

6. Set to 2

CO Line Number	Shelf	Slot	Port	Card Type	CO Name (20 characters)	Trunk Group Number
1	Virtual	1	1	V-SIPGW16	Basic SIP Channel	1
2	Virtual	1	2	V-SIPGW16	Additional Channel	1
3	Virtual	1	3	V-SIPGW16	Additional Channel	1
4	Virtual	1	4	V-SIPGW16	Additional Channel	1
5	Virtual	1	5	V-SIPGW16	Additional Channel	1
6	Virtual	1	6	V-SIPGW16		2
7	Virtual	1	7	V-SIPGW16		2
8	Virtual	1	8	V-SIPGW16		2
9	Virtual	1	9	V-SIPGW16		2
10	Virtual	1	10	V-SIPGW16		2
11	Virtual	1	11	V-SIPGW16		2
12	Virtual	1	12	V-SIPGW16		2
13	Virtual	1	13	V-SIPGW16		2
14	Virtual	1	14	V-SIPGW16		2
15	Virtual	1	15	V-SIPGW16		2
16	Virtual	1	16	V-SIPGW16		2

Unused Trunks are assigned to Trunk Group Number 2 so they cannot be accessed by Dial 9.

Assign SIP Service to ring phones:

This is a simple, basic set up to send incoming calls to an Incoming Call Distribution Group.

Your application may be different.

1. Click Here

2. Set all to DIL

3. Set all to 601

No.	Shelf	Slot	Port	Card Type	Trunk Property	Distribution Method	DIL Destination - Day	DIL Destination - Lunch	DIL Destination - Break
1	Virtual	1	1	V-SIPGW16	Public	DIL	601	601	601
2	Virtual	1	2	V-SIPGW16	Public	DIL	601	601	601
3	Virtual	1	3	V-SIPGW16	Public	DIL	601	601	601
4	Virtual	1	4	V-SIPGW16	Public	DIL	601	601	601
5	Virtual	1	5	V-SIPGW16	Public	DIL	601	601	601

Port Opening/Forwarding Information

You will need to “Open” ports 16000 through 16130 UDP to the LAN IP Address of the DSP Card –1 (See “Get the IP Address of the DSP Card -1” below)

This is to allow RTP (Voice Packets) to reach the KSU.

You will also need to “Open” port 5060 UDP to the KSU's LAN IP Address. (See “A” below)
This allows SIP Signaling to reach the KSU.

Testing

To test outgoing calls, press the Intercom Button on any telephone.
Dial 9 and a telephone number and the call should go through.

Note:

Some SIP Service Providers require you to dial, 1 then the area code and then a 7 digit number.
Example: 18885551212

Some MAY allow 7 digit dialing for local calls.
Ask them how to dial using their service.

To test inbound calls, just call the main number from your cell phone.]
The call should ring all the extensions (Members) of Incoming Call Distribution Group 601.

If calls fail to go through you will need to speak with the SIP Provider to see if you are reaching their server.

Get the IP Address of the DSP Card -1

NS700 Web Maintenance Console
004.30033

Login as INSTALLER

Users
PBX Configuration
UM Configuration
Network Service
1. IP Address/Ports
2. Server Feature
3. Client Feature
4. Other

IP Address/Ports

Basic Settings | Advanced Settings | Reference

IP Address : 192.168.111.201
MAC Address : 08:00:23:A6:2E:94
Subnet Mask : 255.255.255.0
Default Gateway : 192.168.111.1

DNS Setting

Port Number : 53

☐ Obtain DNS server address automatically
☒ Use the following DNS server address

Preferred DNS IP Address : 8.8.8.8
Alternative DNS IP Address : 8.8.4.4

DSP IP Setting

☐ Obtain DSP IP address automatically
☒ Use the following DSP IP address

DSP Card - 1

IP Address : 192.168.111.202
MAC Address : 08:00:23:A6:2E:95

OK Cancel Apply

A. This is the KSU IP address
Yours will be different

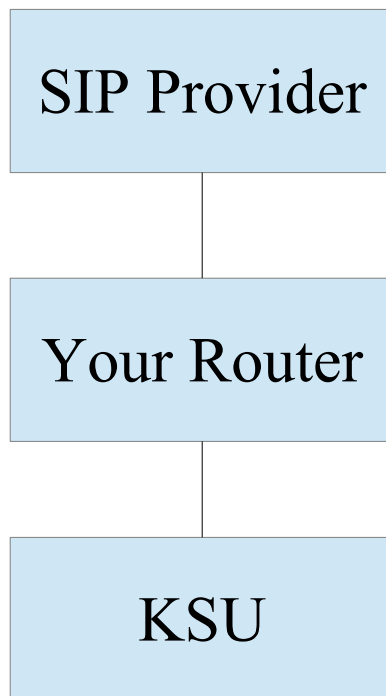
1. Click Here
2. Click Here

3. This is the DSP Card -1 address
Yours will be different

Scroll Down

Special Note

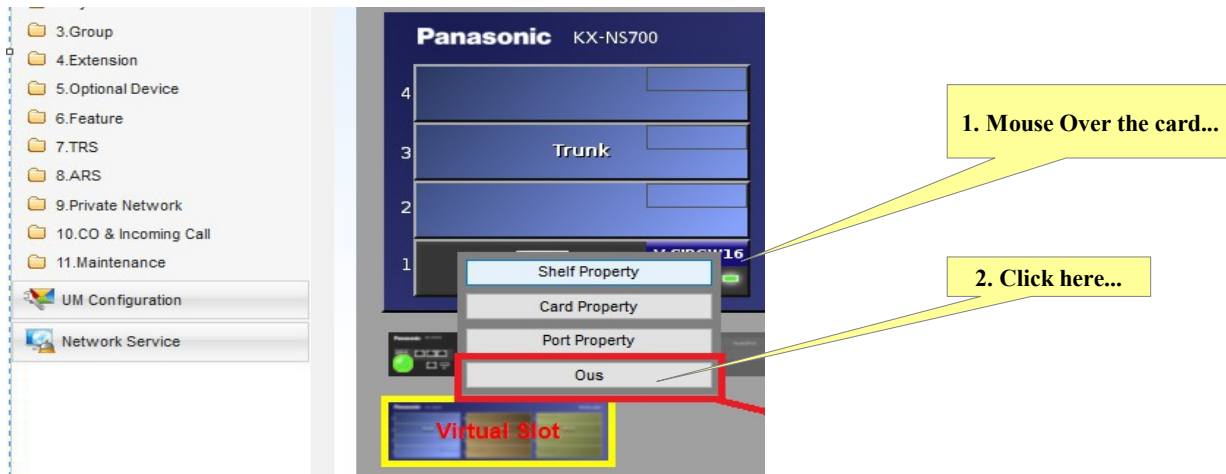
Since the Audio (RTP) Packets come in to the KSU via the DSP Cards IP Address, you need a router to do the Port Forwarding between the KSU and the SIP Providers Interface box.



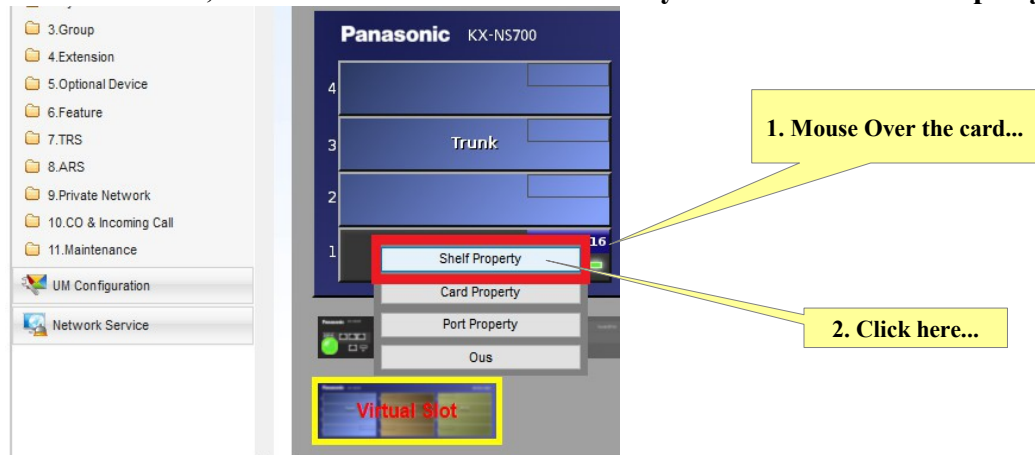
Special DID Numbers Information

If you have DID Numbers on your SIP Service then you need to do the following

Go to the Virtual SIP Gateway Card and put it **Out of Service (OUS)**



Now that the card is Out of Service, on the same Virtual SIP Gateway Card select Shelf Property



Change the SIP Called Party Number Search Mode to **Mode 2**

The screenshot shows the NS700 Web Maintenance Console interface. The left sidebar contains a navigation tree with categories like Users, PBX Configuration, 1. Configuration, 2. System, 3. Group, 4. Extension, 5. Optional Device, 6. Feature, 7. TRS, 8. ARS, 9. Private Network, 10. CO & Incoming Call, 11. Maintenance, UM Configuration, and Network Service. The main content area is titled 'Shelf Property - Virtual SIP Gateway' and has two tabs: 'Main' and 'Timer'. The 'Main' tab is active, displaying a list of configuration parameters and their values. The 'SIP Called Party Number Search Mode' is highlighted with a red box and a yellow callout pointing to it with the text '1. Click here... Set to Mode2'. Another yellow callout points to the 'Apply' button at the bottom right with the text '2. Click here...'. A red arrow points to the 'SIP Called Party Number Check Ability' dropdown with the text 'Set like this....'.

Parameter	Value
SIP Client Port Number	5367
NAT Traversal	Fixed IP Addr.
NAT - Voice (RTP) UDP Port No.	16000
NAT - Keep Alive Packet Sending Ability	Disable
NAT - Keep Alive Packet Type	None
NAT - Keep Alive Packet Sending Interval (s)	20
NAT - Fixed Global IP Address	73.149.137.132
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	Enable
SIP Called Party Number Search Mode	Mode2
Symmetric Response Routing Ability	Enable
100rel Ability	Enable(Passive)
Ringback Tone to Outside Caller	Disable

OK Cancel Apply

Now go back and put the Virtual SIP Gateway Card back In Service (INS)

Add the DID Numbers

This will route INBOUND CALLS to the correct extension

Note:

SIP Providers can use a few different formats for the DID Numbers.

10 digits: 2125551212
11 Digits: 12125551212
+ with 11 digits: +12125551212

You must use the same format when you enter the DID Numbers or they will not work....

It is up to you to determine what format they are using.

Now you can call in and test the DID Numbers.

Add the PBX-CLIP Numbers

This will send out the correct Caller ID for each extension

NS700 Web Maintenance Console
004.30033

Login as INSTALLER

Extension Settings

Copy to CLIP Generate

« Main No Answer Time **CLIP** UM Option 1 Option 2 Option 3 Option 4 Option 5 Option 6 Option 7 Option 8 »

Ext. No.	Extension Number	Extension Name (20 characters)	CLIP ID	CLIP on Extension/CO	CLIR	COLR
1	101	Bob Johnson	2122351234	Extension	Disable	Disable
2	102	Big Boss	5618323801	Extension	Disable	Disable
3	103			Extension	Disable	Disable
4	104			Extension	Disable	Disable
5	105			Extension	Disable	Disable

Note:

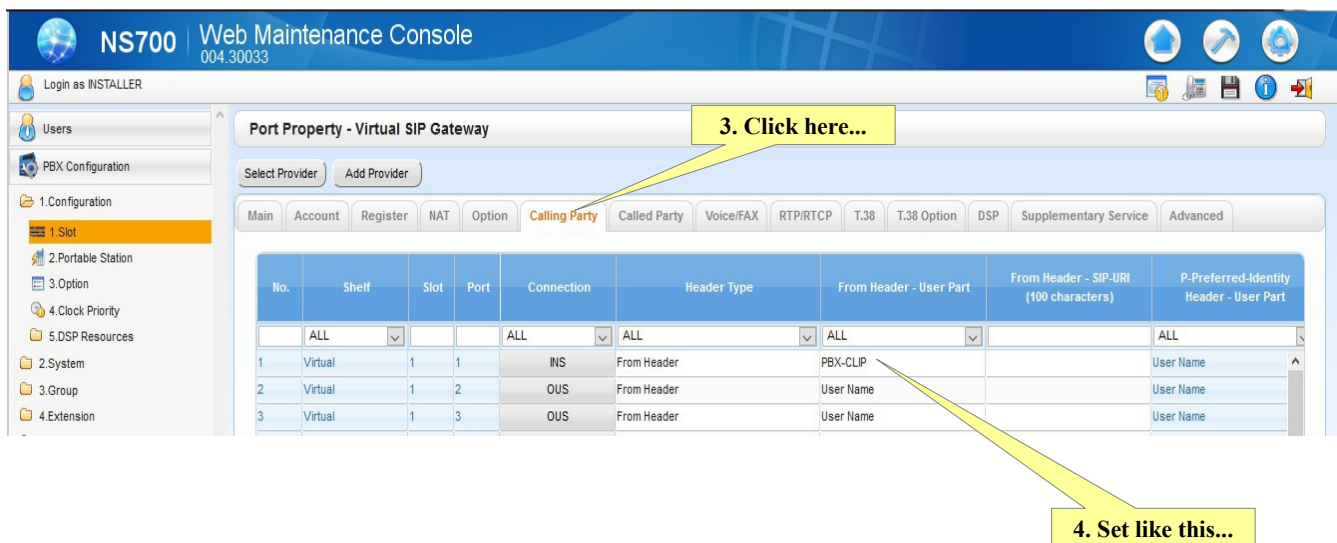
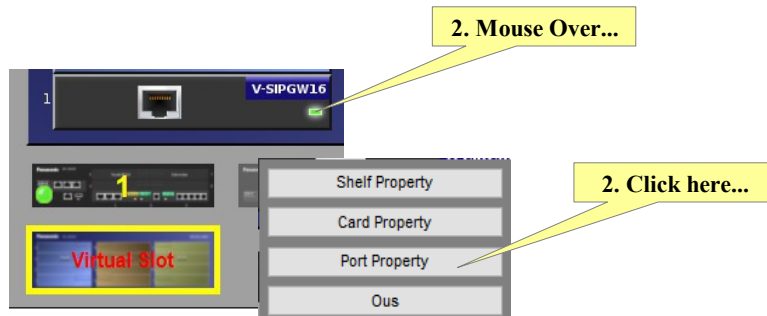
A CLIP Number is the telephone number that will be sent out when the extension makes outgoing calls.

The SIP Provider will usually only allow you to send out telephone numbers that have been assigned by them, that is the DID Numbers.

Activate the PBX-CLIP feature in the SIP Trunk

Step 1

Go to 1. Configuration then 1. Slot and select Virtual Slot



Note:

You must set EACH Port that is INS (In Service) to PBX-CLIP

My example only shows Port 1 as INS.

Testing:

Call your cell phone from an extension that has its CLIP ID set and you should see that extensions CLIP ID appear as the Caller ID in your cell phone.