FUSION SIP TRUNKING CONFIGURATION GUIDE

PBX Platform: NS-1000/KX-TDE/NCP

11/03/2015

OVERVIEW

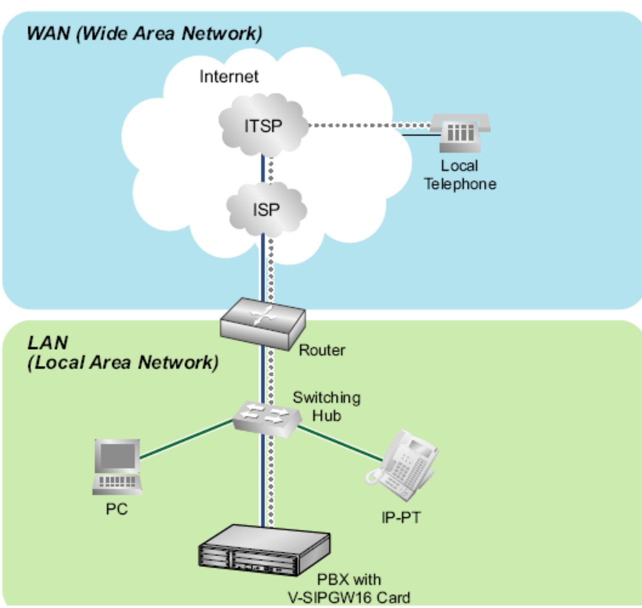
This document describes the configuration procedures required for the Panasonic NS1000, KX-TDE/100/200/600 and NCP1000 to make full use of the capabilities of the Cloud Voice Sip trunk Services.

The SIP trunks services of the NS1000/TDE/NCP are provided through virtual CO line cards (V-SIPGW16) which are designed to be easily integrated into an Internet Telephony Service provided by an ITSP (Internet Telephony Service Provider).

This guide describes the specific configuration items for the virtual SIP Gateway card in addition to the PBX basic configuration related to SIP trunks functionality. It does not describe the purpose and use of all configuration options on the virtual SIP Gateway card. For those details, see the NS1000, KXNCP500/1000 and KX-TDE100/200/600 Programming Manual for Virtual SIP CO Line Card and the KX-TDE100/200/600 and KX-NCP500/1000 Manuals available from Panasonic Communication Solutions.

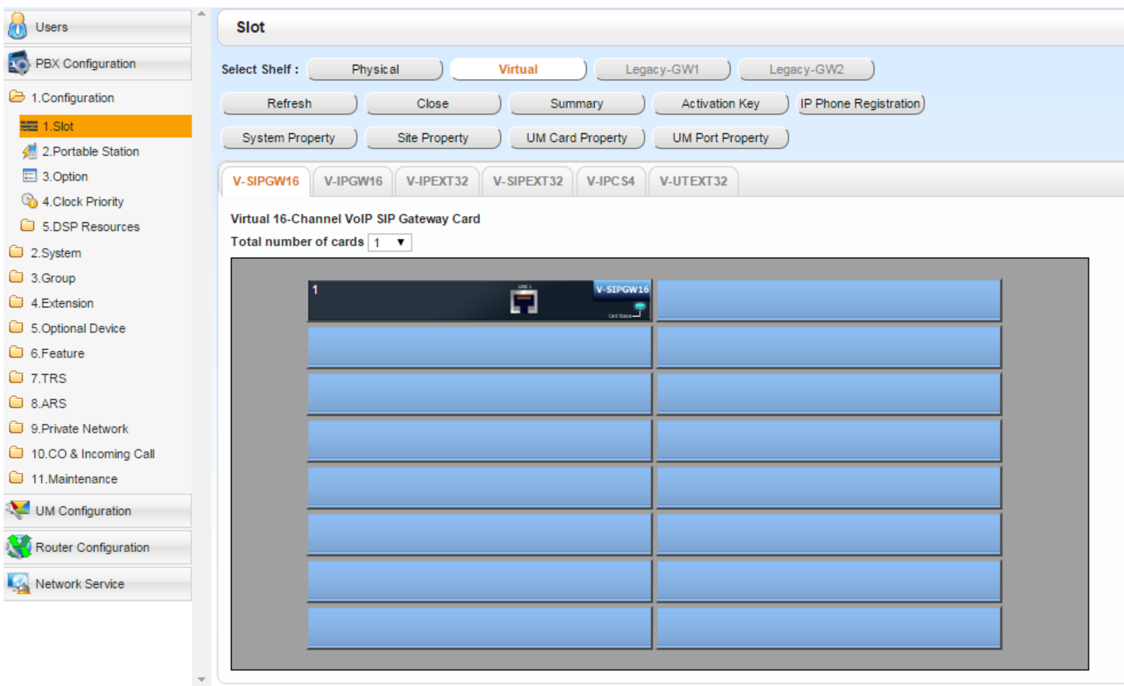
ARCHITECTURE OVERVIEW

The following Diagram illustrates simple VOIP networks connecting the PBX.

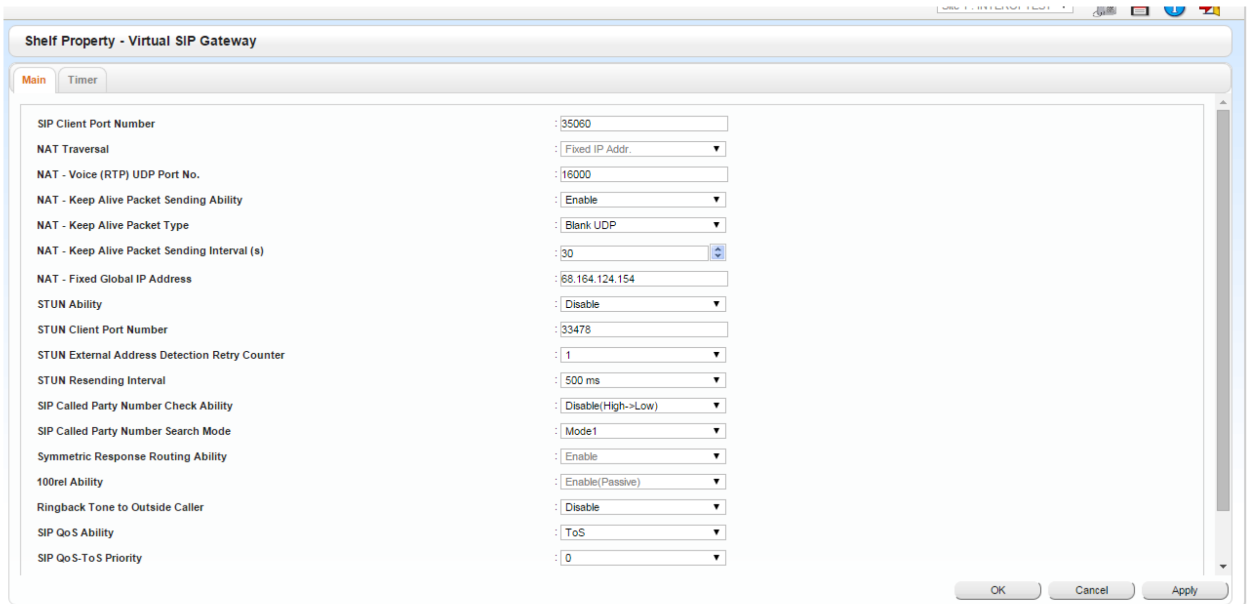


BASIC V-SIPGW SETTINGS FOR FUSION CLOUD VOICE TRUNKS

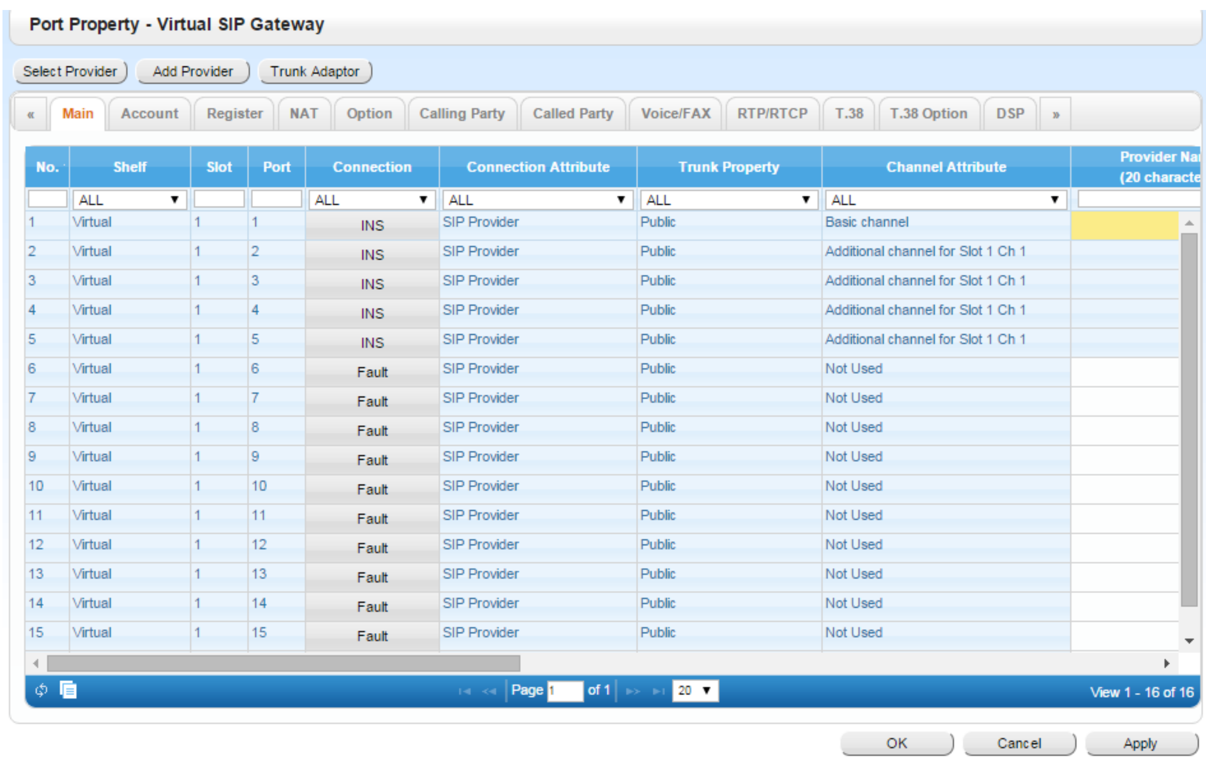
1. Please see Panasonic Communication Solutions Installation Manuals for the corresponding PBX to access configuration tools and initial Network Setup.
2. Installing V-SIPW16 Gateway card
   a. Go to Virtual V-SIPGW16 Tab. Select total amount of cards relating to total amount of SIP trunks configured.



   b. Move the mouse over the VSIPGW16 card and Choose Shelf Properties
   ** (NAT Traversal = Fixed IP Address and NAT-Fixed Global IP Address= The IP address of the WAN side of the Internet router= Public Static IP address obtained from your Internet SIP Provider)
   c. Please Enable NAT Keep Alive and Set Keep Alive Type to Blank UDP



d. Move you mouse over the VSIPGW16 card and choose port property.



Please use the account information/server as per the technical document provider by Fusion to configure the following:

Main tab: According to the account info above, we need to configure the Basic
Channel Port as follows:

1- Port 1 channel attribute = Basic channel
2- Provider name= Fusion Connect
3- SIP Server Name= "nbsvoice.net" (Please see Documentation for Correct Server)
4- SIP server port number =5060
5- SIP service Domain = "Blank"
6- Subscriber number = "Blank"

Account tab:
1- User name ==BTN& Username==1234567890
2- Authentication ID== same as BTN & Username 1234567890
3- Password == Password=ZMLnAEJuERWZ (This is just an example, please enter your account password. Case Sensitive Entry. Please do not use Copy/Paste)

Register Tab:
1- Register abilty== Enable
2- Register sending interval==90
3- Unregister ability when port INS==Enable
4- Register server name/IP address== "Blank"
5- Register server port==5060

NAT Tab:
Same as default

Option Tab:

1- Session Timer Ability=Enable(Passive)
2- Session Expire timer= 1800
3- 3- Refresh Method= re-INVITE
4- 4- Proxy-Require Option= "Blank"

Calling Party tab: Please use the following Settings if you would like to control the CLID on an extension based level. If this is not required please use default settings:

1- Header Type change to P-Preferred-Identity Header
2- P-Preferred-Identity Header User Part will be PBX-CLIP
3- CLID will be controlled on an Extension level In the PBX CLIP. Please see the Panasonic Programming manual for further information.

ADDITIONAL DID'S

Please set the channel attribute for number of ports depending on the number of active SIP Lines as "additional Channel for Channel 1.

	ALL			ALL	ALL	ALL		ALL		ALL
1	Virtual	1	1	INS	P-Preferred-Identity Header	User Name		PBX-CLIP		National
2	Virtual	1	2	INS	From Header	User Name		User Name		National
3	Virtual	1	3	INS	From Header	User Name		User Name		National
4	Virtual	1	4	INS	From Header	User Name		User Name		National
5	Virtual	1	5	INS	From Header	User Name		User Name		National

SIP TRUNKS ACTIVATION KEYS:

Please Contact Panasonic Vendor for More information.