

FUSION SIP TRUNKING CONFIGURATION GUIDE
PBX Platform: NS-1000/KX-TDE/NCP
11/03/2015



OVERVIEW

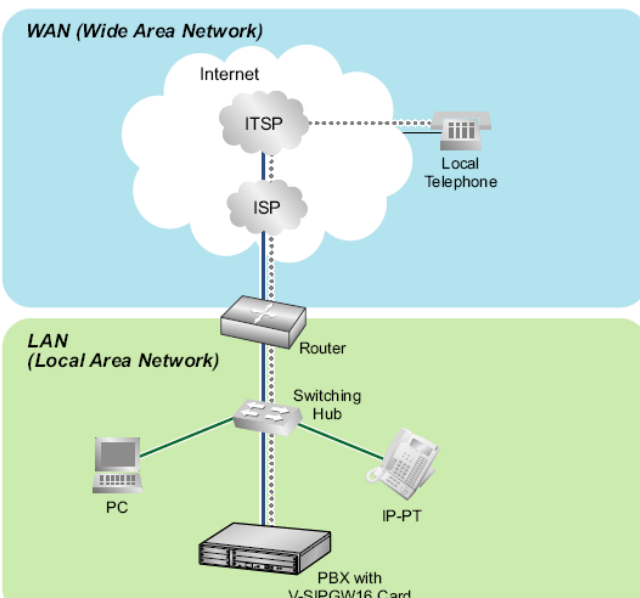
This document describes the configuration procedures required for the Panasonic NS1000, KX-TDE/100/200/600 and NCP1000 to make full use of the capabilities of the Cloud Voice Sip trunk Services.

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.

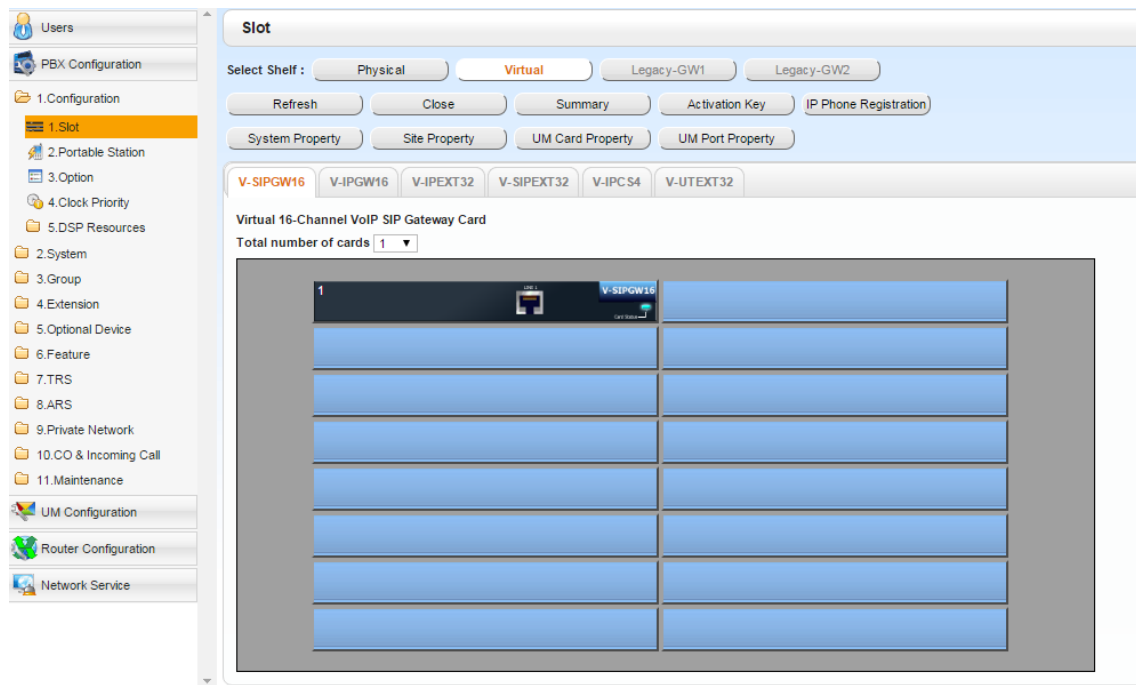
ARCHITECTURE OVERVIEW

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

Shelf Property - Virtual SIP Gateway

Main Timer

SIP Client Port Number : 35060

NAT Traversal : Fixed IP Addr.

NAT - Voice (RTP) UDP Port No. : 16000

NAT - Keep Alive Packet Sending Ability : Enable

NAT - Keep Alive Packet Type : Blank UDP

NAT - Keep Alive Packet Sending Interval (s) : 30

NAT - Fixed Global IP Address : 88.164.124.154

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

SIP QoS Ability : ToS

SIP QoS-ToS Priority : 0

OK Cancel Apply

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

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 DSP

No.	Shelf	Slot	Port	Connection	Connection Attribute	Trunk Property	Channel Attribute	Provider Name (20 characters)
1	Virtual	1	1	INS	SIP Provider	Public	Basic channel	
2	Virtual	1	2	INS	SIP Provider	Public	Additional channel for Slot 1 Ch 1	
3	Virtual	1	3	INS	SIP Provider	Public	Additional channel for Slot 1 Ch 1	
4	Virtual	1	4	INS	SIP Provider	Public	Additional channel for Slot 1 Ch 1	
5	Virtual	1	5	INS	SIP Provider	Public	Additional channel for Slot 1 Ch 1	
6	Virtual	1	6	Fault	SIP Provider	Public	Not Used	
7	Virtual	1	7	Fault	SIP Provider	Public	Not Used	
8	Virtual	1	8	Fault	SIP Provider	Public	Not Used	
9	Virtual	1	9	Fault	SIP Provider	Public	Not Used	
10	Virtual	1	10	Fault	SIP Provider	Public	Not Used	
11	Virtual	1	11	Fault	SIP Provider	Public	Not Used	
12	Virtual	1	12	Fault	SIP Provider	Public	Not Used	
13	Virtual	1	13	Fault	SIP Provider	Public	Not Used	
14	Virtual	1	14	Fault	SIP Provider	Public	Not Used	
15	Virtual	1	15	Fault	SIP Provider	Public	Not Used	

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OK Cancel Apply

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 ability== 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.