VIDEOTRON Business

SIP Trunking Service

Configuration Guide

Panasonic KX-NS700 PBX Ver.6.0

Confidentiality and copyright statement

Modification history

Edit	Date	Originator	Description
0.1	2019-04-04	Pascal Beauregard	Original draft
1.1	2019-04-11	Martin Lefrançois	Minor edits to descriptions
1.2	2019-04-30	Danielle Arsenault and Louis Villemur	Linguistic revision
1.3	2019-06-25	Martin Lefrançois	Post-revision technical edits
1.4	2019-08-19	Martin Lefrançois	Revision of translation.
1.5	2019-09-12	Martin Lefrançois	Revision of coherency with other manuals

Table of Contents

1 Audience

The *SIP Trunking Service Configuration Guide* is intended for service users, technical managers, and authorized integrators.

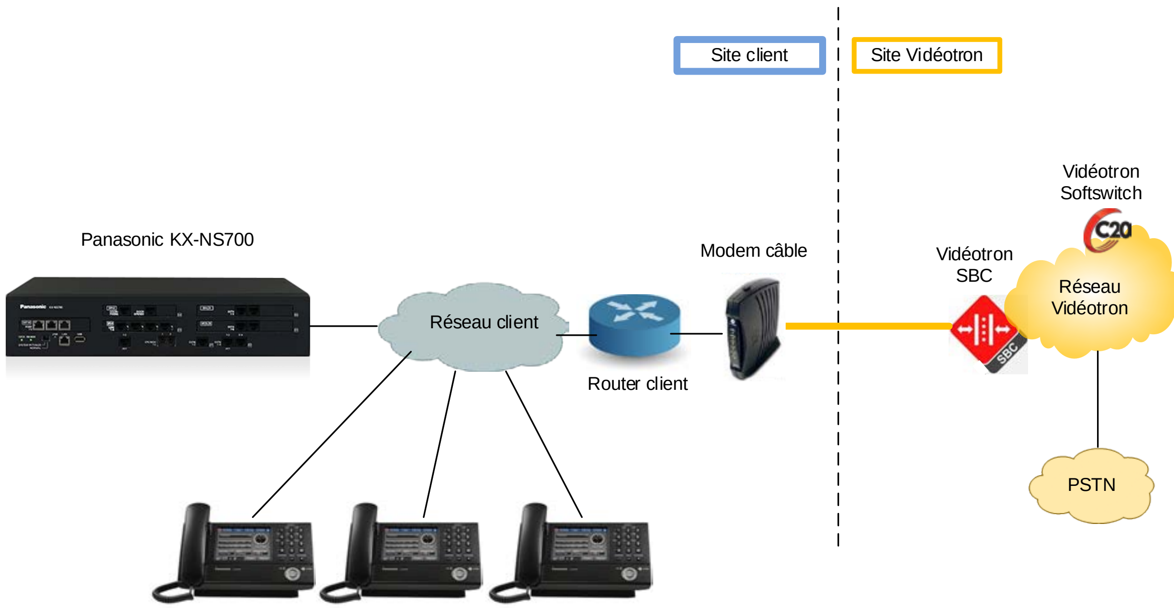
2 Introduction

The *SIP Trunking Service Configuration Guide* details the basic requirements for setting up a single SIP trunk between Videotron's SBC and the Panasonic KX-NS700 PBX—you can configure additional trunks following these very same steps.

That said, this guide is not intended to help you configure various PBX user/application features.

3 Network and equipment diagram

The diagram below is an overhead view of SIP trunking with a Panasonic KX-NS700 PBX.



The solution includes:

Client site:

- Telephones
- PBX
- Router/Firewall
- Cable modem

Videotron site:

- Videotron SBC: Oracle (Acme Packet)
- Videotron softswitch: Genband C20
- PSTN connection

4 Features

4.1 Supported features

The SIP trunking service supports the following features:

Feature	Description	Limit(s)
Simultaneous calls	The maximum limit of simultaneous calls is established when the SIP trunk order is placed.	
Voice	G.711 µ-law standard used exclusively	
Fax	G.711 µ-law standard used	T.38 not supported
Other kinds of data (modem, alarm, etc.)	G.711 µ-law standard used	
Inbound Caller ID name and number	Inbound Caller ID name and number transmitted from the Videotron site to the PBX.	
Outbound Caller ID name	Outbound Caller ID name, as transmitted via PBX, to the public network.	
Outbound Caller ID number	Sends Outbound Caller ID number, as transmitted via PBX, to the public network.	
DID display for 911	DID display for 911 emergency call centre transmitted via PBX if on the predefined list of numbers.	
SIP trunk overflow	Calls are routed to another SIP trunk when the number of simultaneous calls that the SIP trunking can handle is exceeded.	The other SIP trunk must be part of the same Videotron telephone switch as the primary SIP trunk.
Overflow to another phone number	Calls are routed to another phone number when the number of simultaneous calls that the SIP trunk can handle is exceeded.	The “Redirect information” or “Original called number” field is not transmitted. The “Called number” is the actual forwarding number and not the DID. An overflow to another phone number requires an additional service called a “Permanent Redirect Line (PRL).” This service is billed according to the predefined number of simultaneous PRL calls. If the phone number is long distance, charges will apply.
Failover to another SIP trunk	Calls are routed to another SIP trunk in the following three cases of failure: 1. The client’s PBX no longer responds to calls sent to it on the SIP trunk. 2. The client's PBX responds with a "SIP 503 Service unavailable" message. 3. The SIP trunk is faulty.	If the PBX responds with a SIP message other than "503 Service unavailable", there will be no call overflow.
Failover to another phone number	Calls are routed to another phone number in the same three cases above.	If the PBX responds with a SIP message other than

		"503 Service unavailable", there will be no call overflow. Same limitation as "Overflow to another phone number": Due to maximum capacity reached with respect to fields and the need for a Permanent Diverting Line.
"Redirect number" field (Remote Party ID)		The Videotron telephone switch transmits the initial called number in the Remote-Party-ID header.
Call blocking Restriction class	No blocking for local calls, in Quebec, Canada, the United States and abroad, and for 411, 0-, 0+, 00 and 900 numbers.	1-976 calls are blocked.
Number portability	Videotron handles the transfer of the customer's telephone number from their current service to the SIP trunking service.	The client must provide all required documentation.
SIP REFER	Allows you to free up lines after a call is forwarded from an external number to another external number, such as a cellphone.	If the external number is long distance in relation to the originally dialled number, the call may be dropped rather than forwarded. Especially when the call is transferred through another Videotron switch. Routing between Videotron switches is subject to change without prior notice.

4.2 Unsupported or limited features

Our SIP trunking does not support the following features:

Features	Description
Numbers outside coverage area	Only the telephone numbers in Videotron Wired service areas will be accepted.
Fixed 911	This feature allows calls to be routed directly to the 911 emergency call centre in the municipality where the caller is located. Instead, the SIP trunking service uses an intermediary ("nomadic") 911 emergency call centre to route calls. See videotron.com/911-ip for details.
Emergency call rerouting	Allows you to route calls to different destinations based on a predefined phone tree for emergency scenarios. This is an advanced feature that is reserved for the dedicated fibre optic SIP trunking service.
Authorization and billing codes	The authorization code is used to limit access to long distance calls. The billing code is used to count calls per user for internal

	billing and customer billing purposes. These are advanced features reserved for the dedicated fibre optic SIP trunking service.
Equal Access	Allow to use another provider for long-distance calls. This feature is irrelevant considering that Videotron offers unlimited calling plans for Canada and the United States. This is a feature reserved for the dedicated fibre optic local SIP trunking service.
Occasional calls	Allows you to dial 101-XXXX to temporarily change your long-distance service provider. This feature is irrelevant considering that Videotron offers unlimited calling plans for Canada and the United States. This is a feature is reserved for the dedicated local fibre optic SIP trunking service.
Signalling and voice channel encryption	Videotron does not currently support signalling encryption (SIP TLS) and voice channel encryption (SRTP). However, the password is encrypted using MD5 hash.

5 Service requirements

5.1 Registering a SIP trunk

Once SIP trunking has been initialized at the Videotron site, our technical team will send the following information to the client:

- domain name
- username
- password

The client PBX must be registered with Videotron in order to connect calls via SIP trunking. The client— or more commonly the integrator-interconnector—configures the PBX to register the SIP trunk with Videotron's switch. The Videotron team will set up a phone conference with the interconnector to complete the registration and ensure the SIP trunk is functioning properly.

The PBX is registered by sending SIP REGISTER messages to Videotron's SBC IP address. These messages contain a username, password and domain name.

5.2 Responding to SIP INFO (or SIP OPTIONS) messages

Videotron's telephone switch periodically sends SIP INFO messages to the client's PBX. If said messages do not reach the PBX (i.e. are blocked by the client's firewall), or remain unanswered by the PBX, the switch will consider the PBX out of order.

5.3 Sending the domain name in the Req URI header of SIP INVITE messages

The PBX must be capable of sending a domain name in the Req URI of SIP INVITE messages. If the domain name is lacking, any calls will be rejected.

5.4 Configuration settings overview

The table below provides an overview of the settings required to set up SIP trunking service.

Domain name	**Provided by Videotron:** <client acronym>.sipott.v50.videotron.com
Videotron SBC address	**24.200.242.87**
SIP communication port	**UDP 5060**
Username	**Provided by Videotron: s**<last 9 numbers of primary telephone number> *Ex.: s143801234*
Password	**Provided by Videotron:** 12 alphanumeric characters with at least: 1 lowercase letter, 1 uppercase letter, and 1 number *Ex.: aQkTZaxvHz7phrLY*
Number of simultaneous calls on the SIP trunk	**Provided by Videotron**
Codec	**G.711 µ-law only**
Fax protocol	**In-Band (T.38 not supported)**
DTMF	**RFC2833**
SIP REFER	**The SIP REFER function must only be activated after discussion with the Videotron team. If the external number is long distance compared to the number first dialled, the call may be dropped rather than forwarded.**

Table 1: Configuration settings overview

6 Configuration

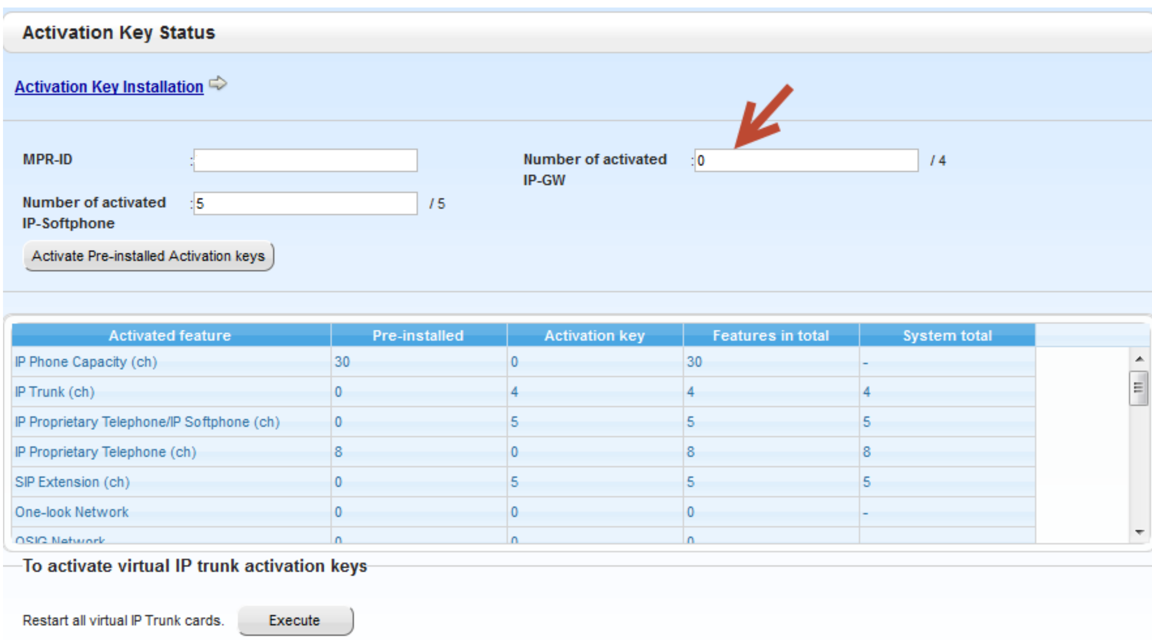
Step 1: Activating licenses

In the **PBX Configuration** section, select **1. Slot**, then click on the **Activation Key** button.

Make sure you reserve enough IP-GW licences for the SIP trunking service.

A 0/4 setting will allow you to allot all four IP-GW licenses for SIP trunking.

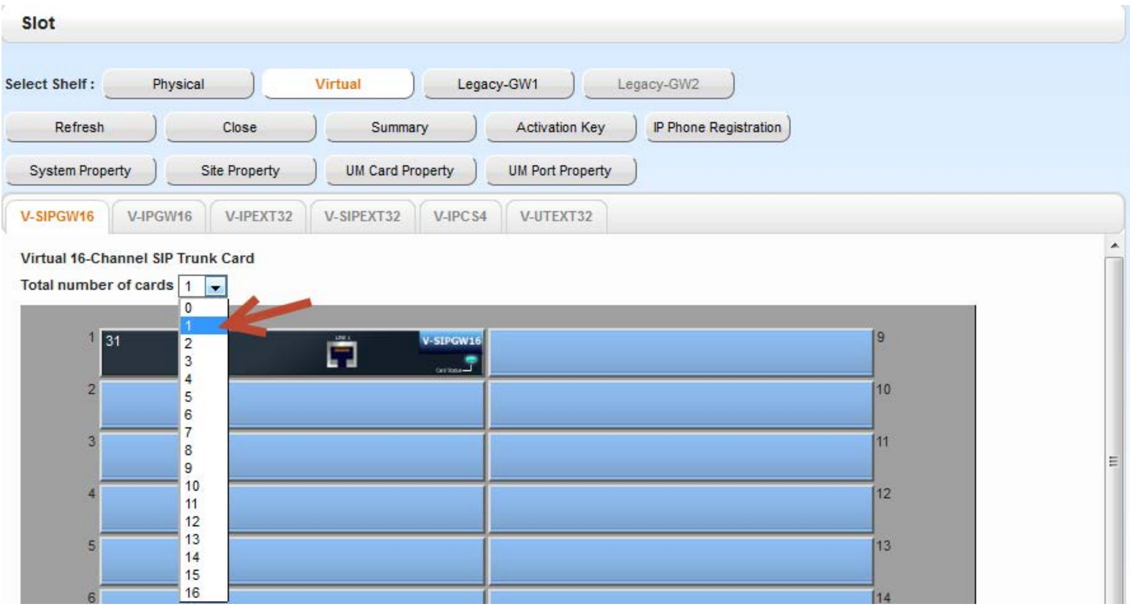
A 4/4 setting will allow you to allot all four IP-GW for H.323.



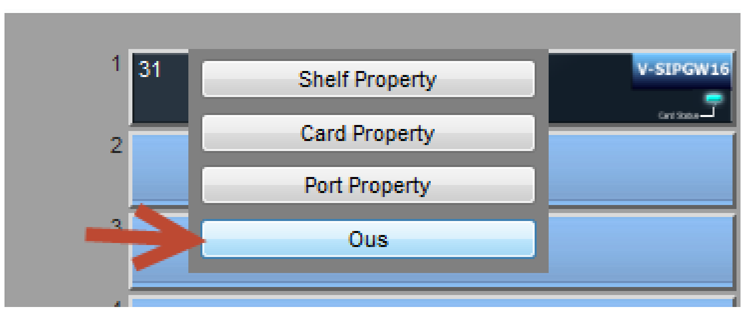
Step 2: Adding a V-SIPGW16 card

In the **PBX Configuration** section, select **1. Slot**, then click on the **Virtual** button.

Add one or several **V-SIPGW16** cards, depending on how many channels you want to activate. One **V-SIPGW16** card supports 16 channels.



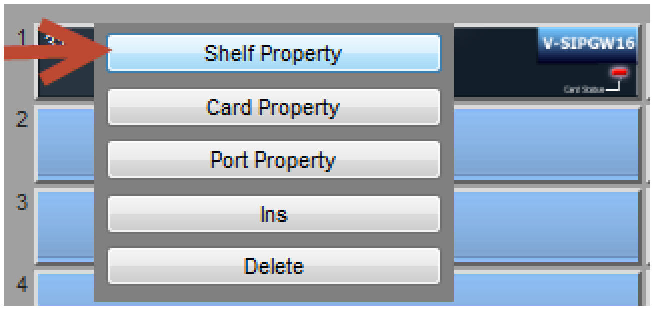
It is important that, firstly, the **V-SIPGW16** card be set to out of service (To set a **V-SIPGW16** card to out of service, hover over it with the cursor, then in pop-up menu, click on the **Ous** (*out of service*) button.



Step 3: Configuring the SIP trunking service

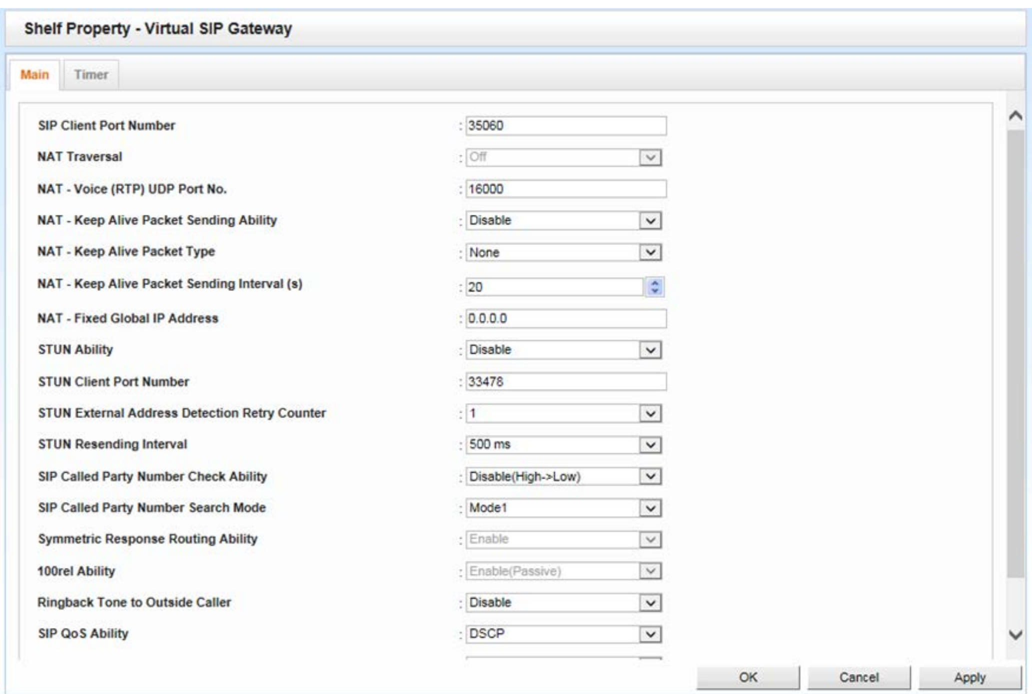
Shelf Property button

Hover over the card with the cursor, then click on the **Shelf Property** button in the pop-up menu.



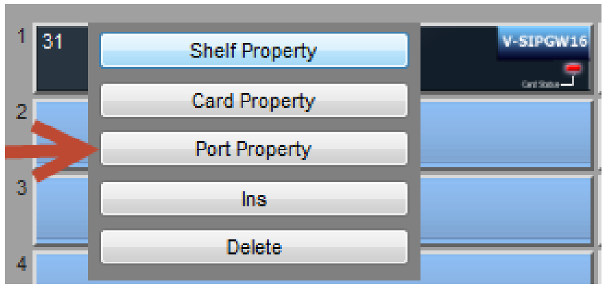
In the **Main** tab, make sure that the following settings have been entered or selected:

- SIP Client Port Number: **35060**
- NAT Traversal: **Off**
- NAT – Keep Alive Packet Sending Ability: **Disable**
- NAT – Keep Alive Packet Type: **None**



Port Property button

Hover above the menu with the cursor, then select **Port Property** in the pop-up menu.

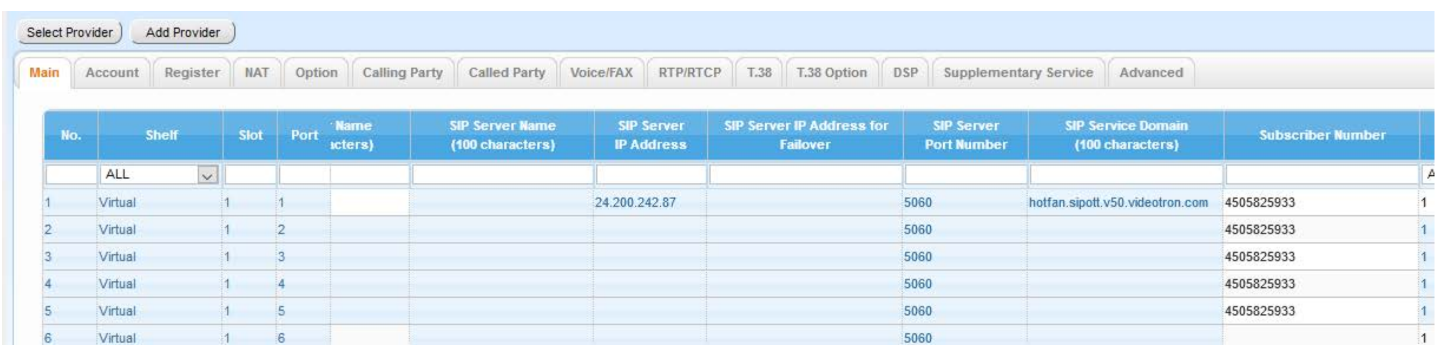


Note: sample screenshots were taken on different systems, so it's possible that card or port numbers vary from one screenshot to the next.

Main tab

In the **Main** tab, for the first channel of the SIP trunk, select or enter the following settings:

- Trunk Property: **Public**
- Channel Attribute: **Basic Channel**
- Provider Name: **Videotron**
- SIP Server IP Address: **24.200.242.87**
- SIP Server Port Number: **5060**
- SIP Service Domain: **domain name provided by Videotron (ex.: hotfan.sipott.v50.videotron.com)**
- Subscriber Number: **phone number displayed for an outbound call**

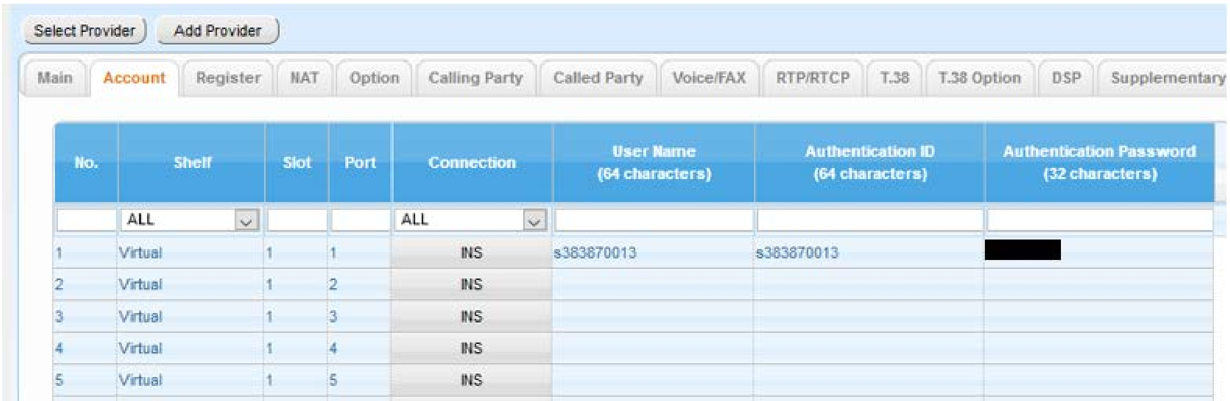


No.	Shelf	Slot	Port	Name cters)	SIP Server Name (100 characters)	SIP Server IP Address	SIP Server IP Address for Failover	SIP Server Port Number	SIP Service Domain (100 characters)	Subscriber Number	
	ALL										A
1	Virtual	1	1			24.200.242.87		5060	hotfan.sipott.v50.videotron.com	4505825933	1
2	Virtual	1	2					5060		4505825933	1
3	Virtual	1	3					5060		4505825933	1
4	Virtual	1	4					5060		4505825933	1
5	Virtual	1	5					5060		4505825933	1
6	Virtual	1	6					5060			1

Account tab

In the **Account** tab, enter or select the following settings for the first SIP trunk channel:

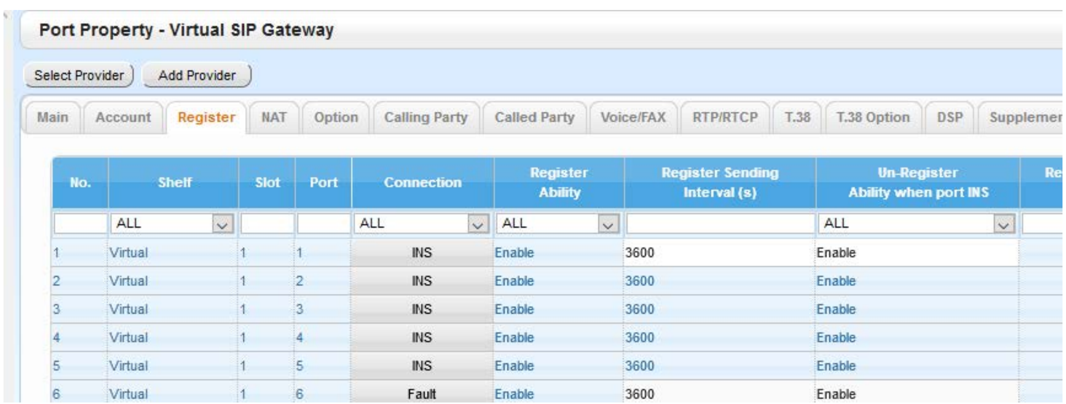
- User Name: **username provided by Videotron**
- Authentication ID: **username provided by Videotron** (same as **User Name**)
- Authentication Password: **password provided by Videotron**



Register tab

In the **Register** tab, enter or select the following settings for the first SIP trunk channel:

- Register Ability: **Enable**
- Register Sending Interval: **3600**
- Un-Register Ability when port INS: **Enable**



NAT tab

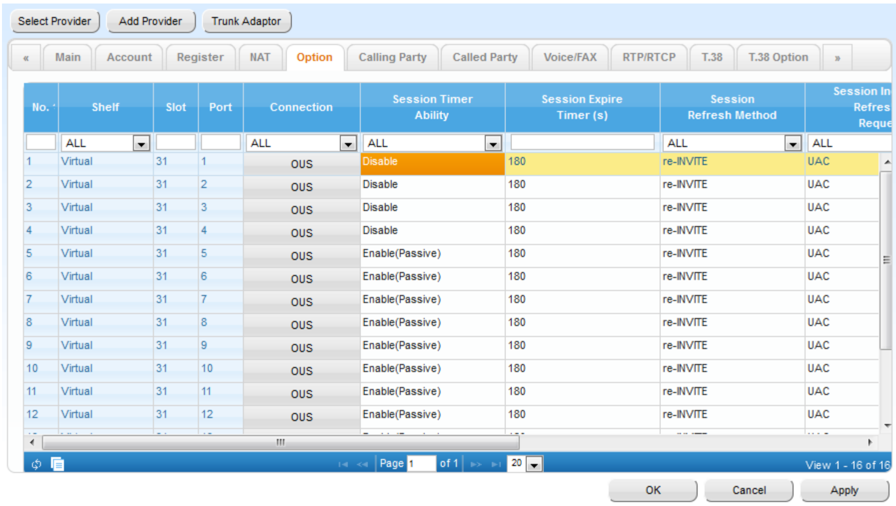
No configuration necessary.

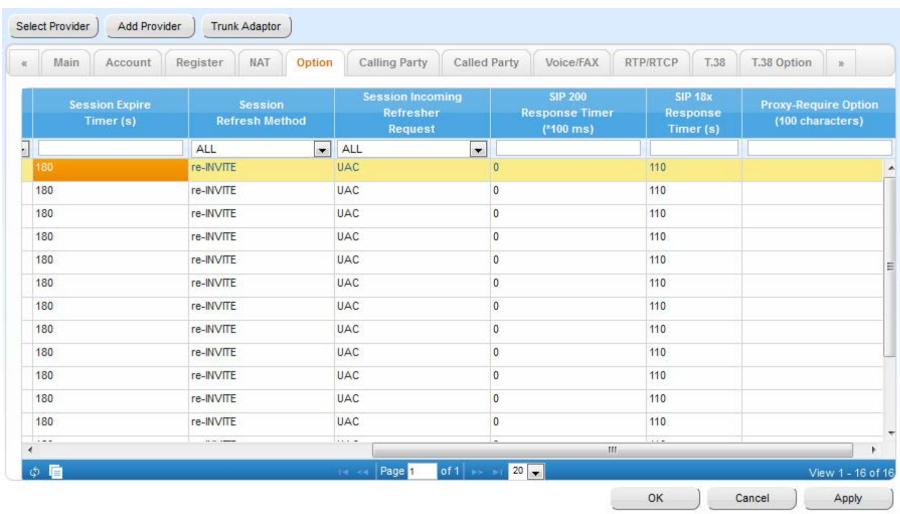
Option tab

In the **Option** tab, for <u>all SIP trunk channels, enter or select the following</u>:

- Session Timer Ability: **Disable**

In all the other fields, leave the default settings be.



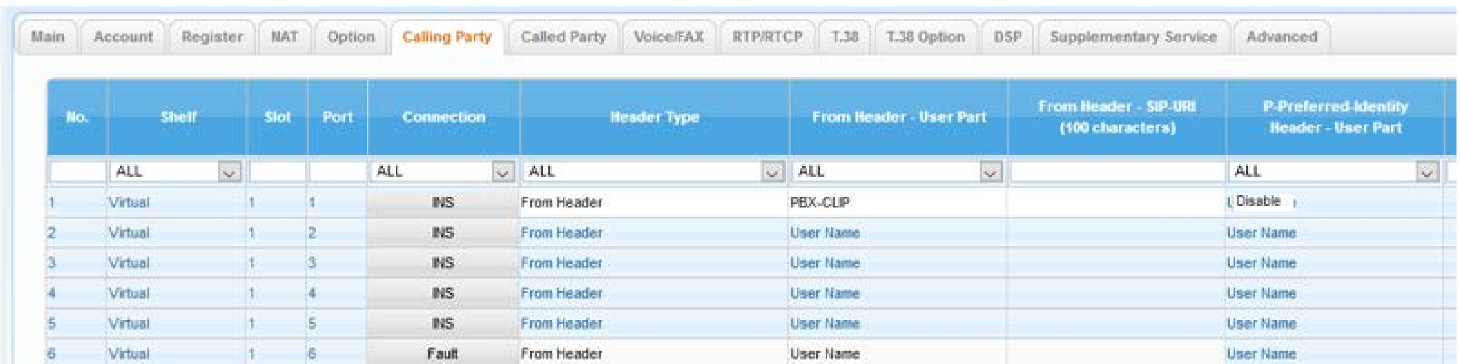


Calling Party tab

In the **Calling Party** tab, for all SIP trunk channels, enter or select the following:

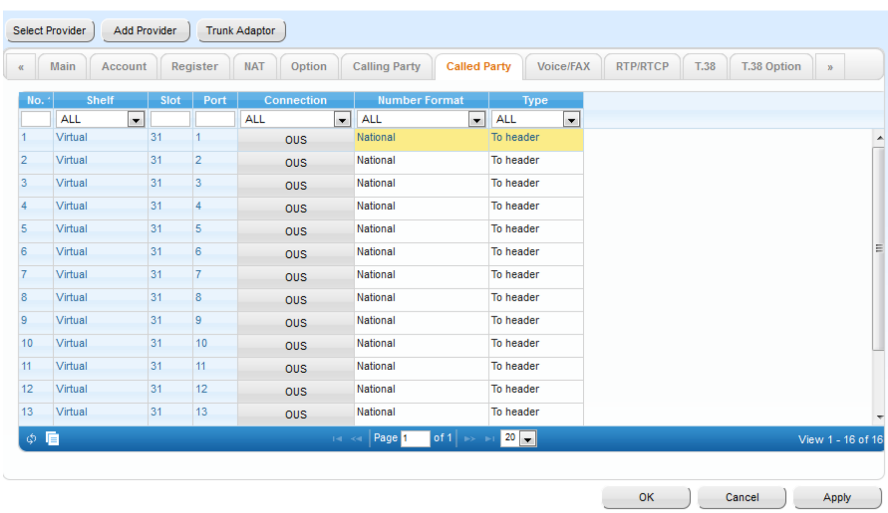
- Header Type: **From Header**
- From Header – User part: **PBX-CLID**
- P-Preferred-Identity Header: **Disable**
- P-Asserted-Identity header: **Enable**

In all the other fields, leave the default settings be.



Called Party tab

In the **Called Party** tab, leave the default settings be in all the other fields.

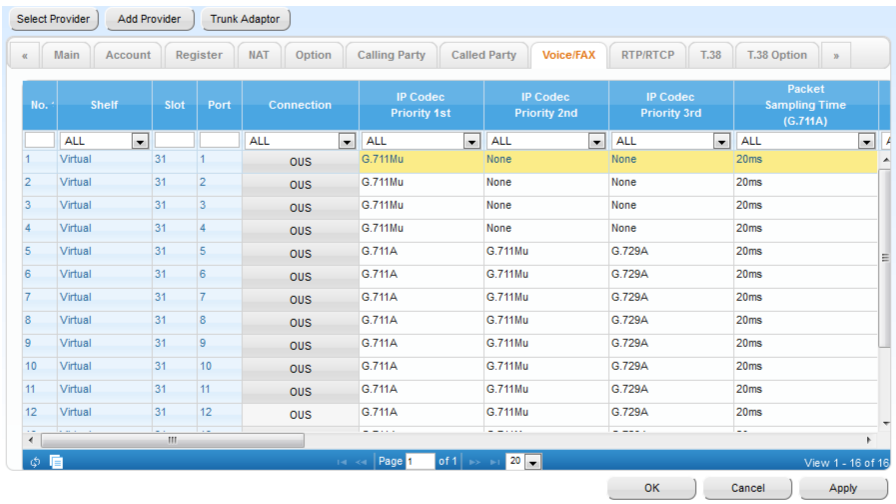


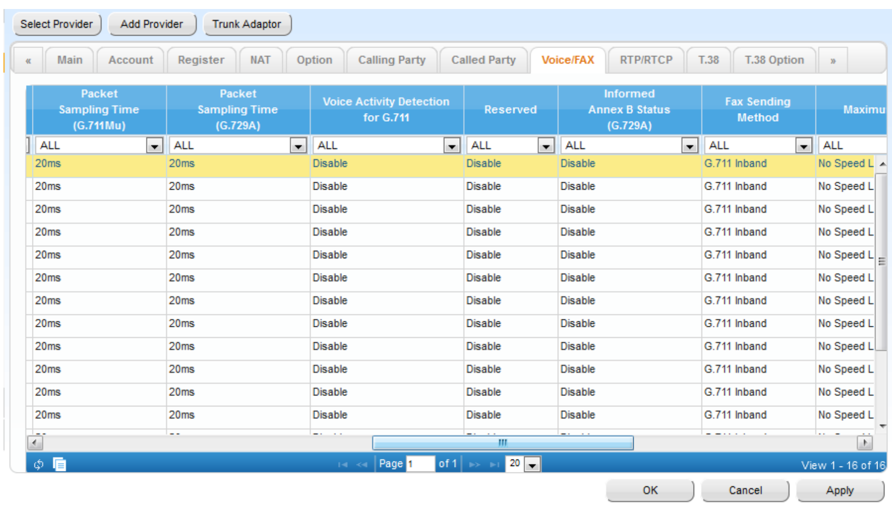
Voice/Fax tab

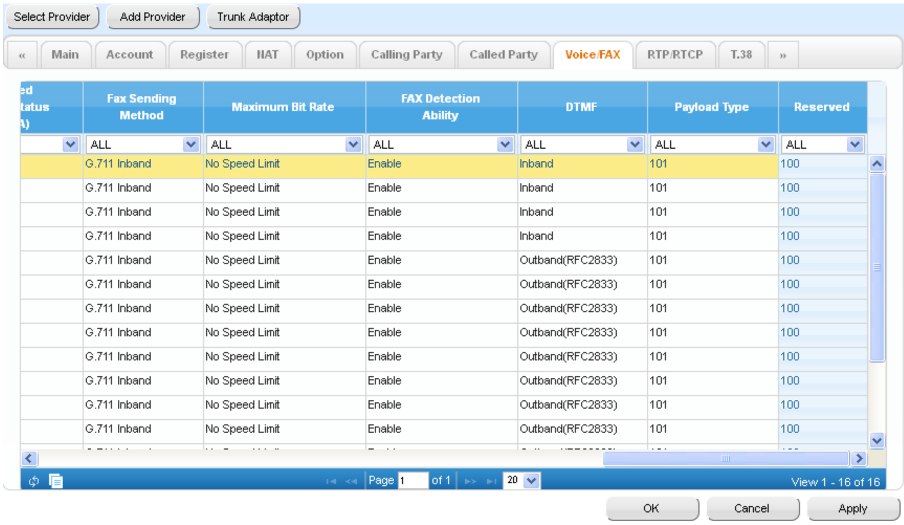
In the **Voice/FAX** tab, for <u>all SIP trunk channels, enter or select the following</u>:

- IP Codec Priority 1st: **G.711Mu**
- IP Codec Priority 2nd: **None**
- IP Codec Priority 3rd: **None**
- Fax sending method: **G711 inband**

In all the other fields, leave the default settings be.





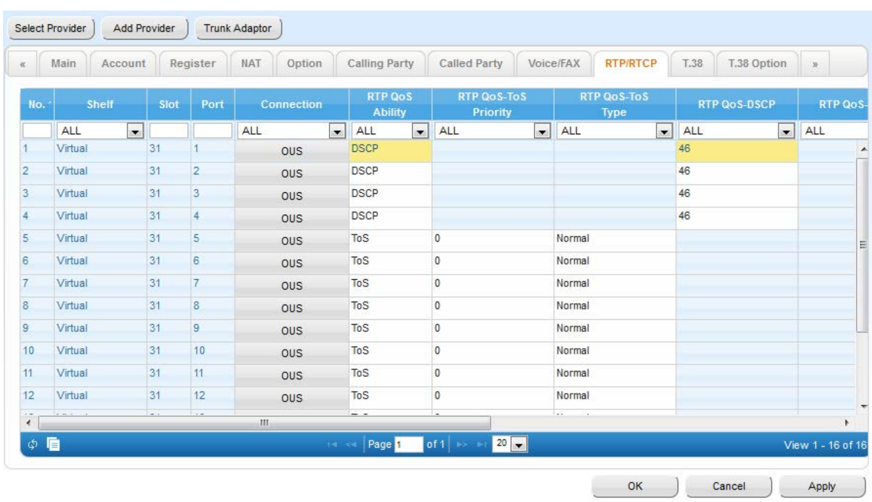


RTP/RTCP tab

In the **RTP/RTCP** tab, for all SIP trunk channels, enter or select the following:

- RTP QoS Ability: **DSCP**
- RTP QoS-DSCP: **46**

In all the other fields, leave the default settings be.



T.38 tab

Do not alter the default settings.

T.38 Option tab

Do not alter the default settings.

DSP tab

Do not alter the default settings.

Supplementary Service tab

In the **Supplementary Service** tab, enter or select the following for all SIP trunk channels:

- CLIP (Receive): **P-Asserted-Identity Header**
- CNIP (Send): **Yes**
- CNIP(Receive): **Yes**
- Blind Transfer (REFER): **No** **
- Attended Transfer (REFER): **No** **

**** The SIP REFER function must only be activated after discussion with the Videotron team. If the external number is long distance compared to the number first dialled, the call may be dropped**

No.	Shelf	Slot	Port	Connection	CLIP (Receive)	CLIR	CNIP (Send)	CNIP (Receive)	Blind Transfer(REFER)	Attended Transfer(REFER)
	ALL			ALL	ALL	ALL	ALL	ALL	ALL	ALL
1	Virtual	1	1	INS	P-Asserted-Identity Header	Yes	Yes	Yes	No	No
2	Virtual	1	2	INS	P-Asserted-Identity Header	Yes	Yes	Yes	Yes	Yes
3	Virtual	1	3	INS	P-Asserted-Identity Header	Yes	Yes	Yes	Yes	Yes
4	Virtual	1	4	INS	P-Asserted-Identity Header	Yes	Yes	Yes	Yes	Yes
5	Virtual	1	5	INS	P-Asserted-Identity Header	Yes	Yes	Yes	Yes	Yes

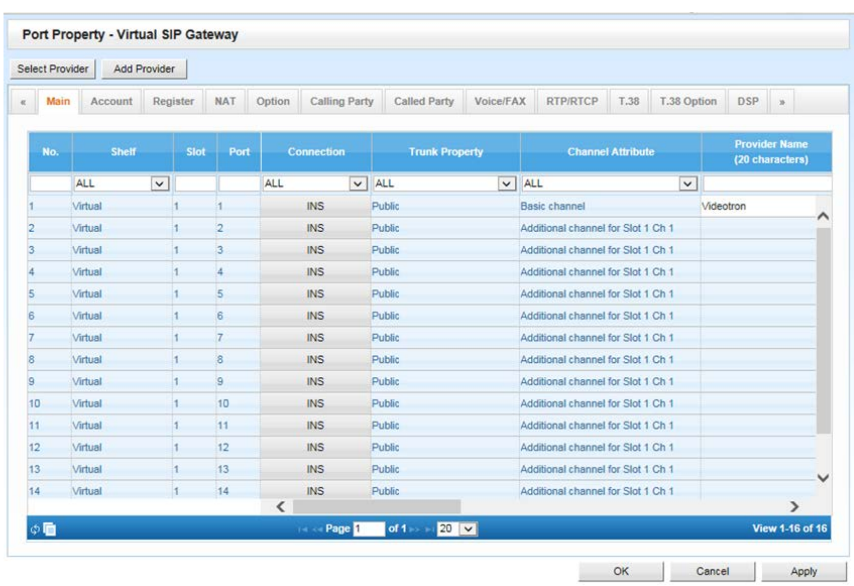
rather than forwarded.

Advanced tab

Do not alter the default settings.

Back to the Main tab

Select **Additional channel for Slot 1 Ch 1** to configure the supplementary SIP trunk channels you want to activate. The fields from the **Main**, **Account** and **Register** tabs will also inherit the settings you enter or select for the first channel.



No.	Shelf	Slot	Port	Connection	Trunk Property	Channel Attribute	Provider Name (20 characters)
	ALL			ALL	ALL	ALL	
1	Virtual	1	1	INS	Public	Basic channel	Videotron
2	Virtual	1	2	INS	Public	Additional channel for Slot 1 Ch 1	
3	Virtual	1	3	INS	Public	Additional channel for Slot 1 Ch 1	
4	Virtual	1	4	INS	Public	Additional channel for Slot 1 Ch 1	
5	Virtual	1	5	INS	Public	Additional channel for Slot 1 Ch 1	
6	Virtual	1	6	INS	Public	Additional channel for Slot 1 Ch 1	
7	Virtual	1	7	INS	Public	Additional channel for Slot 1 Ch 1	
8	Virtual	1	8	INS	Public	Additional channel for Slot 1 Ch 1	
9	Virtual	1	9	INS	Public	Additional channel for Slot 1 Ch 1	
10	Virtual	1	10	INS	Public	Additional channel for Slot 1 Ch 1	
11	Virtual	1	11	INS	Public	Additional channel for Slot 1 Ch 1	
12	Virtual	1	12	INS	Public	Additional channel for Slot 1 Ch 1	
13	Virtual	1	13	INS	Public	Additional channel for Slot 1 Ch 1	
14	Virtual	1	14	INS	Public	Additional channel for Slot 1 Ch 1	

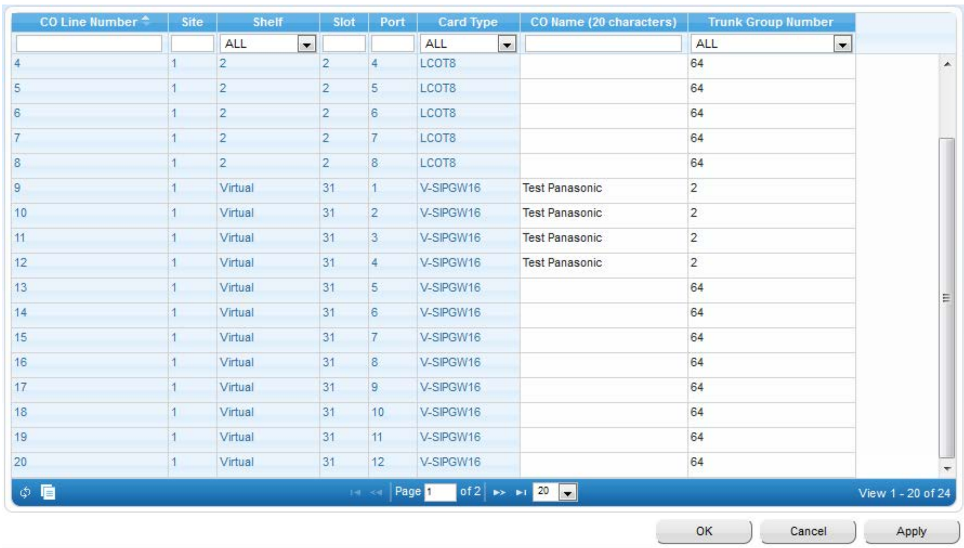
Using the cursor, hover over the **V-SIPGW16** card, then click on **INS** to get the SIP trunk up and running.

Step 4: Configuring the SIP trunk, Trunk Group section

In the **PBX Configuration** section, under the **10. CO & Incoming Call** tab, select point **1. CO Line Settings**.

In the **CO Name** column, enter a name for each activated channel of the **V-SIPGW16** card. This name will be displayed for an outbound call.

In the **Trunk Group Number** column, assign an available SIP trunk number to each activated channel of the **V-SIPGW16** card.



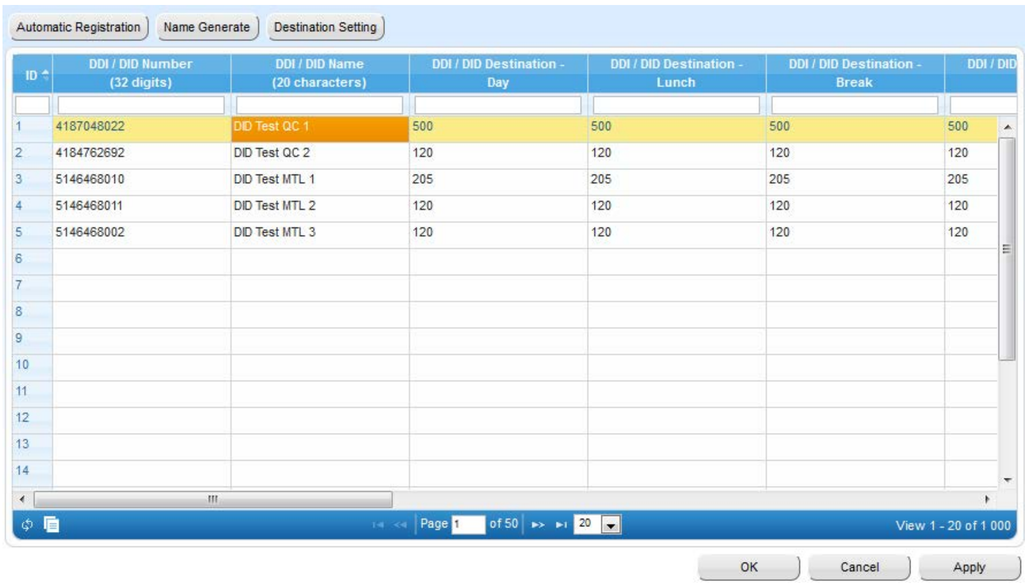
CO Line Number	Site	Shelf	Slot	Port	Card Type	CO Name (20 characters)	Trunk Group Number
		ALL			ALL		ALL
4	1	2	2	4	LCOT8		64
5	1	2	2	5	LCOT8		64
6	1	2	2	6	LCOT8		64
7	1	2	2	7	LCOT8		64
8	1	2	2	8	LCOT8		64
9	1	Virtual	31	1	V-SIPGW16	Test Panasonic	2
10	1	Virtual	31	2	V-SIPGW16	Test Panasonic	2
11	1	Virtual	31	3	V-SIPGW16	Test Panasonic	2
12	1	Virtual	31	4	V-SIPGW16	Test Panasonic	2
13	1	Virtual	31	5	V-SIPGW16		64
14	1	Virtual	31	6	V-SIPGW16		64
15	1	Virtual	31	7	V-SIPGW16		64
16	1	Virtual	31	8	V-SIPGW16		64
17	1	Virtual	31	9	V-SIPGW16		64
18	1	Virtual	31	10	V-SIPGW16		64
19	1	Virtual	31	11	V-SIPGW16		64
20	1	Virtual	31	12	V-SIPGW16		64

Step 5: Configuring DID numbers

In the **PBX Configuration** section, under the **10. CO & Incoming Call** tab, select **3. DDI / DID Table**.

In the **DDI / DID Number** column, enter the active DID numbers for the SIP trunk and, under the **DDI / DID Destination**, **Day**, **Lunch**, **Break** and **Night** columns, enter the destination extension.

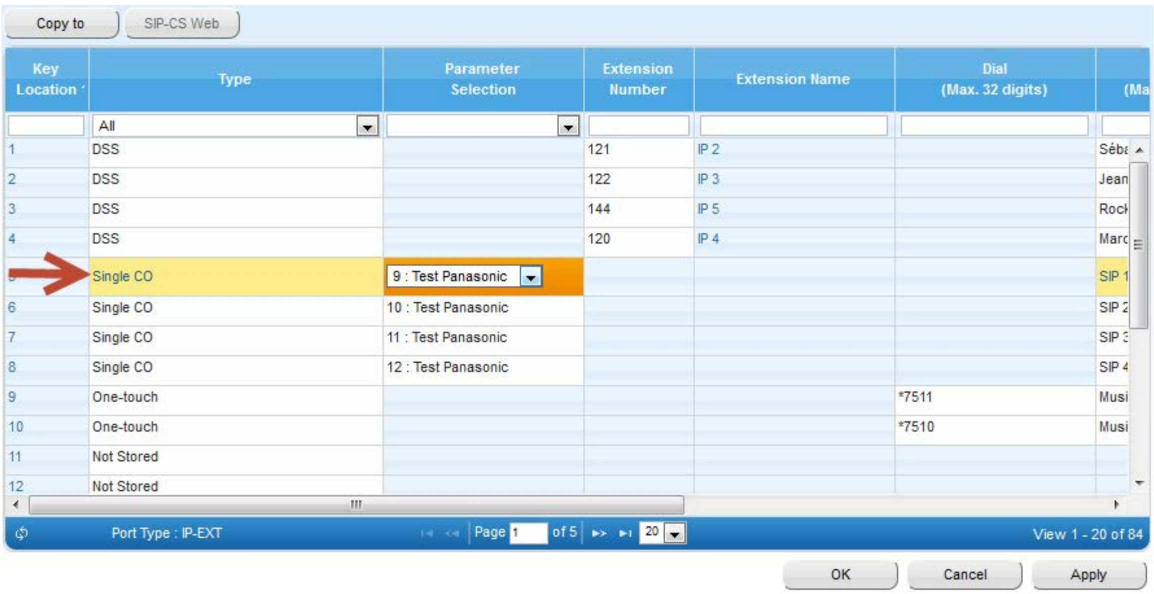
To save changes, click on **OK**.



Step 6: Configuring the SIP trunk for outbound calls

You can configure an SIP trunk directly via touchtone phone, emulating a key system.

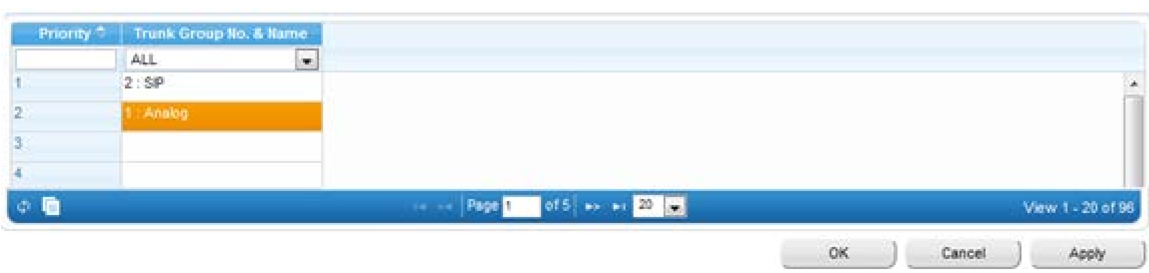
In the **PBX Configuration** section, under the **4. Extension** tab, select **1. Wired Extension**, followed by **4. Flexible Button**.



You must assign a number and name to the SIP trunk in order to use the SIP trunking service, which you can access by dialling 9 before the phone number.

In the **PBX Configuration** section, under the **3. Group** tab, select **1. Trunk Group**, followed by **2. Local Access Priority**.

Enter the necessary information in the fields under the **Trunk Group No. & Name** column.



Step 7: Managing outbound caller ID

Outbound caller ID is extension-specific and two display options are available to you:

- Extension name and custom phone number (**Extension**). This option allows you to display a name and extension-specific number for outbound caller ID.
- Name and number assigned to the SIP trunk (**CO**). This option allows you to display the number and name of the company rather than the extension number. This number was already configured in Step 3, under the Main tab, in the Subscriber Number field. The name was configured in Step 4 under the CO Name column.

<u>Selecting the call display mode</u>

In the **PBX Configuration** section, under the **4. Extension** tab, select **1. Wired Extension**, followed by **1. Extension Settings**, then click on the **CLIP** tab.

Displaying the extension name and custom phone number

In the **PBX Configuration** section, under the **4. Extension** tab, select **1. Wired Extension**, followed by **1. Extension Settings**, then click on the **CLIP** (calling line identification presentation) tab.

In the **CLIP ID** column, enter the phone number, then select **Extension** in the **CLIP on Extension/CO** column.

Displaying the name and number assigned to the SIP trunk

In the **PBX Configuration** section, under the **4. Extension** tab, select **1. Wired Extension**, followed by **1. Extension Settings**, then click on the **CLIP** tab.

Under the **CLIP on Extension/CO** column, select **CO**.

The displayed name will be the one you entered when configuring the SIP trunk.

In the **PBX Configuration** section, under the **10. CO & Incoming Call** tab, select **1. CO Line Settings**, followed by **1. Extension Settings**, then click on the **CLIP** tab.

The displayed phone number will be the one you entered in the **Main** tab, under the **Subscriber Number**, when configuring the SIP trunk.

7 Glossary

503	Service unavailable Server error code.
Bursting	Feature that allows you to temporarily exceed your calling limit. Simultaneous calls are billed on a pay-per-use basis. (Feature currently in development)
C20	Videotron's telephone switch
Called number	Number called or requested
Called Party	Person to whom a call is sent.
Calling Party	Person sending a call to establish communication.
CO line	Central Office Line Communication line that connects a PBX to a telephone service provider's switchboard.
DID	Direct inward dialling Telephone feature allowing an outbound caller to reach a subscriber directly without going through an operator or dialling an extension. Commonly refers to an external phone number.
G.711	Digital voice encoding standard (audio compression standard). Audio codec
H.323	Standard for transmitting audio, data and images in real time across packet-based networks—which applies to local networks, like an intranet, or public ones, like the Internet. It is a lesser used standard when compared to SIP.
IP	Internet protocol
IP-GW	IP gateway
key system	Intercom system, key telephone system Most commonly used telephone system when few additional extensions are required. Allows users to call each other directly and to communicate with public network subscribers via outbound and inbound calls.
Local SIP	Dedicated fibre-optic SIP telephony service offered by Videotron. It's the standard local service.
PBX	private branch exchange A company's private telephone switch
PSTN	public switched telephone network
REFER	SIP method for forwarding calls wherein a call is forwarded to a number indicated in the forwarding request. Allows you to free up lines after a call is forwarded from an external number towards another external number, such as a cellphone.
Remote Party ID	The Remote-Party-ID header indicates the identity of the calling or called party.
SBC	Session Border Controller A network element to monitor and protect SIP-based communications from fraud, and allowing you to configure SIP trunk settings.
SIP	Session Initiation Protocol Signaling protocol used in IP telephony. Refers to an IP telephony service allowing a telephone switch to access the PSTN, thereby supporting the management of call signaling, over IP links using SIP trunking.

Softswitch	Software switch, media gateway controller, call controller, call server Interconnection equipment that manages the operation of a media gateway that allows signals carrying voice, data, or images to move from a circuit-switched public telephone network to a private packet-switched network, such as a private IP network—or go in the reverse.
T.38	Encoding standard for sending faxes across IP networks in a real-time mode.
Trunk	A line that connects switches with each other and is used to route information sequentially.
trunk group; TG	Group of circuits whose start end belongs to the same switch and whose terminating end belongs to one or more switches giving access to the same subscribers. In the specific case of the Videotron SIP trunking service, TG refers to a SIP trunk. In certain exceptional situations, there may be more than one TG, or multiple SIP trunks, between a PBX and Videotron.