VIDEOTRON
Business

SIP Trunking Service

Configuration Guide

3CX PBX Ver. 16.0

Confidentiality and copyright statement

Modification history

Edit	Date	Author	Description
1.0	2019-06-27	Pascal Beauregard	Original draft
1.1	2019-07-02	Martin Lefrançois	Edit
1.2	2019-07-17	Louis Villemur	Linguistic revision
1.3	2019-08-19	Martin Lefrançois	Validation and correction. Integration of edits from another document.

Table of Contents

1 Audience

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

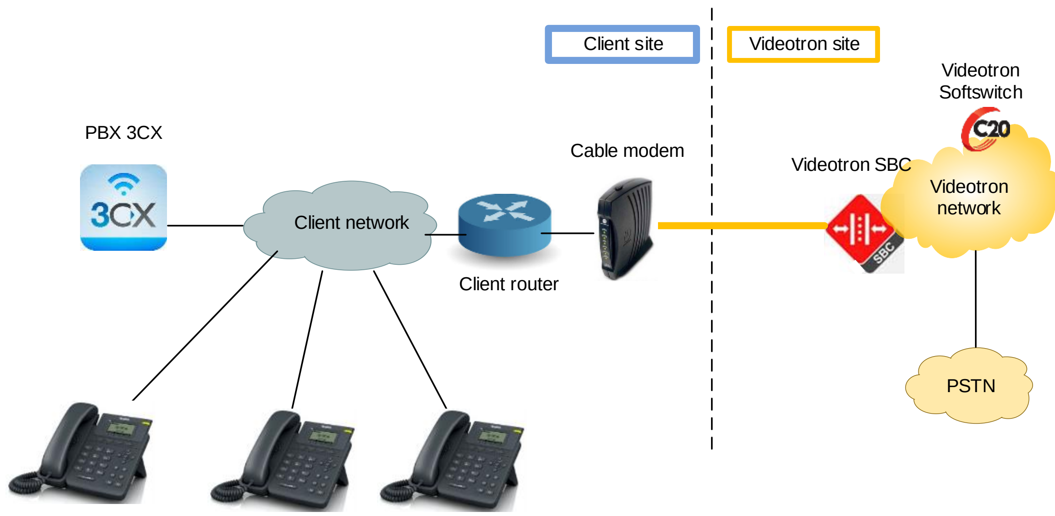
2 Introduction

The *SIP Trunking Service Configuration Guide* details the basic steps for setting up a single SIP trunk between Videotron's SBC and the 3CX PBX; you can configure additional trunks following the steps described herein.

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

3 Network and equipment diagram

The diagram below is an overhead view of SIP trunking with a 3CX PBX.



The solution includes:

Customer site:

- Telephones
- PBX
- Router/Firewall
- Cable modem

Videotron site:

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

4 Features

4.2 Supported features

The SIP trunking service supports the following features:

Feature	Description	Limit(s)
Simultaneous calls	The simultaneous calls limit is established when the SIP trunk order is placed. Any excess calls are extra.	
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	Outbound Caller ID number, as transmitted via PBX to the public network.	
DID display for 911 emergency call centre	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 SIP trunking can handle is exceeded.	The other SIP trunk must be on the same Videotron telephone switch as the primary SIP trunk.
Direct trunk overflow	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 preset 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 customer’s PBX no longer responds to calls sent to it on the SIP trunk. 2. The customer's PBX responds with the message “SIP 503 Service Unavailable.” 3. The SIP trunk is faulty.	If the PBX responds with a SIP message other than “503 Service Unavailable,” there will be no call failover.
Failover to another phone number	Calls are routed to another phone number in the same three cases as above.	If the PBX responds with a SIP message other than “503 Service Unavailable,” there will be no call failover.

		Same limitation as “Direct trunk overflow” with respect to the fields and the need for a Permanent Redirect Line.
“Redirect number” field (Remote Party ID)		The Videotron telephone switch transmits the original called number to the Remote-Party-ID header.
Class of restriction call blocking	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 a customer’s telephone number from their current service to the SIP trunking service.	The customer 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 original 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.

4.3 Unsupported or limited features

Our SIP trunking does not support the following features:

Feature	Description
Numbers outside our coverage area	Only telephone numbers in Videotron service areas will be accepted.
Fixed 911	This feature allows calls to be forwarded directly to the 911 emergency call centre in the municipality where the caller is located. Instead, the SIP trunking service uses an intermediary (“nomad”) 911 emergency call centre to forward calls. See videotron.com/ip-911 for details.
Emergency call forwarding	Allows you to forward calls to different destinations based on a preset phone tree for emergency scenarios. This is an advanced feature 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..
Equity of access	Allows you to use another long-distance provider. This feature is largely irrelevant considering that Videotron offers unlimited calling plans for Canada and the United States. This feature is reserved for the dedicated local fibre optic SIP trunking service.
Occasional calls	Used to dial the 101-XXXX code in order to temporarily change long-distance provider. This feature is largely irrelevant considering that Videotron offers unlimited calling plans for Canada and the United States. This 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). Encrypted MD5 hash password.

5 Service requirements

5.1 Responding to SIP INFO (or SIP OPTIONS) messages

Videotron's telephone switch periodically sends SIP INFO messages to the customer's PBX. If these messages do not reach the PBX (i.e., they are blocked by the customer's firewall), or the PBX does not answer, the switch will consider the PBX out of order.

5.2 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 missing, calls will be rejected.

5.3 Registering a SIP trunk

Once the SIP trunk has been configured at the Videotron site, our technical team will send the following information to the customer:

- domain name
- username
- password

The customer PBX must be registered with Videotron in order to connect calls via SIP trunking. The customer, or more commonly the integrator-interconnector, must configure the PBX such as to be able 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.

6 Configuration

Configuration settings overview

Table 4 provides an overview of the settings required to set up the SIP trunking service.

Domain name	**Provided by Videotron:** <customer acronym>.sipott.v50.videotron.com ***E.g.:*** *cust01.sipott.v50.videotron.com*
Username	**Provided by Videotron: s**<last 9 numbers of primary telephone number> *E.g.: s143801234*
Password	**Provided by Videotron:** 12 alphanumeric characters with at least 1 lowercase letter, 1 uppercase letter and 1 number *E.g.: aQkTZaxvHz7phrLY*
Videotron SBC address	**24.200.242.87**
SIP communication port	**UDP 5060**
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 in relation to the original dialled number, the call may be dropped rather than forwarded.**

Table 1: Configuration settings overview

The 3CX PBX can use SBC features and provides instructions on how to install an SBC on a separate server. After installation, the SBC must be added to the PBX configuration. This document does not cover how to configure a PBX when a 3CX SBC has been installed.

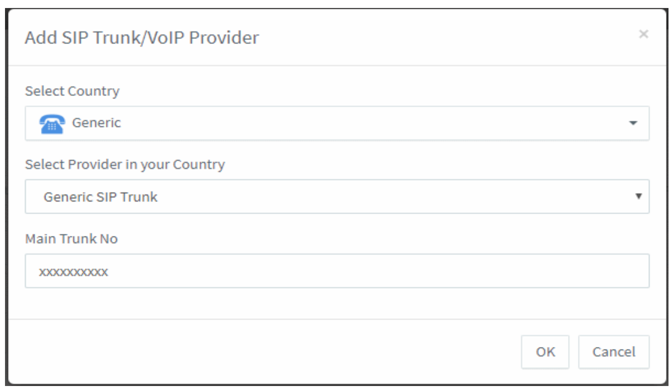
Step 1: Configuring the SIP trunking service

SIP Trunk section

In the **SIP Trunk** section, click **Add SIP Trunk**.

Enter the following information:

- Select Country: **Generic**
- Select Provider in Your Country: **Generic SIP Trunk**
- Main Trunk No.: **Enter the primary phone number provided by Videotron**

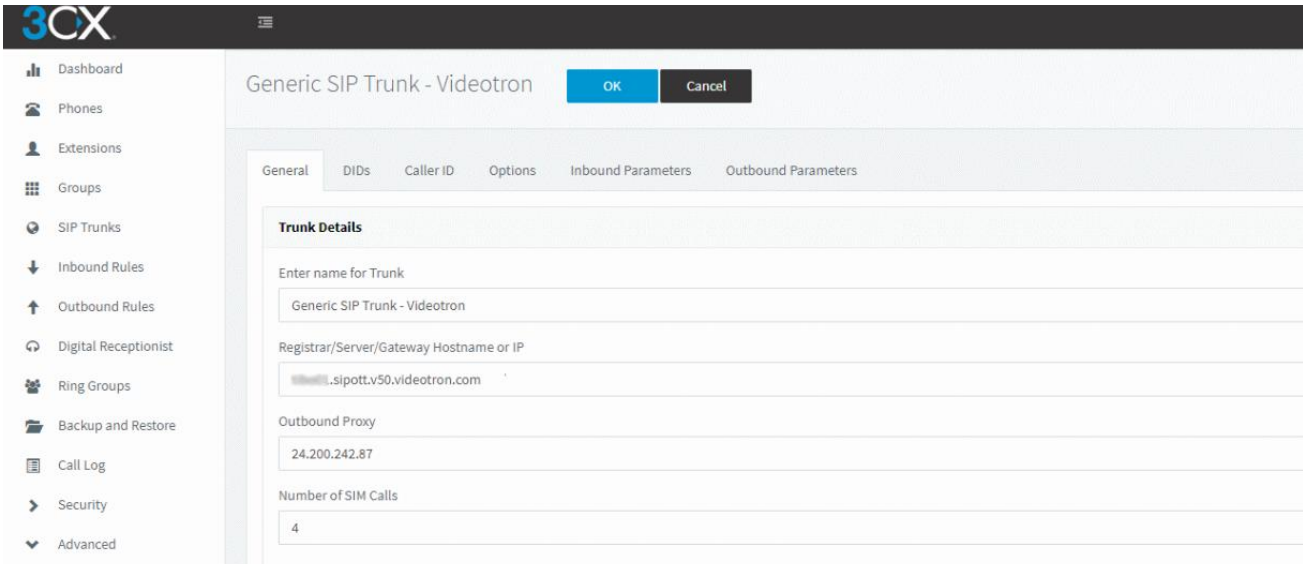


Click OK and the SIP Trunk configuration window will open.

SIP Trunk - General tab - Trunk Details section

Enter the following information:

- Enter name for Trunk: **A relevant name (e.g., Videotron)**
- Registrar/Server/Gateway Hostname or IP: **Domain name provided by Videotron (e.g., hotfan.sipott.v50.videotron.com)**
- Outbound Proxy: **24.200.242.87**
- Number of SIM Calls: **Number of simultaneous calls included in your plan**



SIP Trunk - General tab - Authentication section

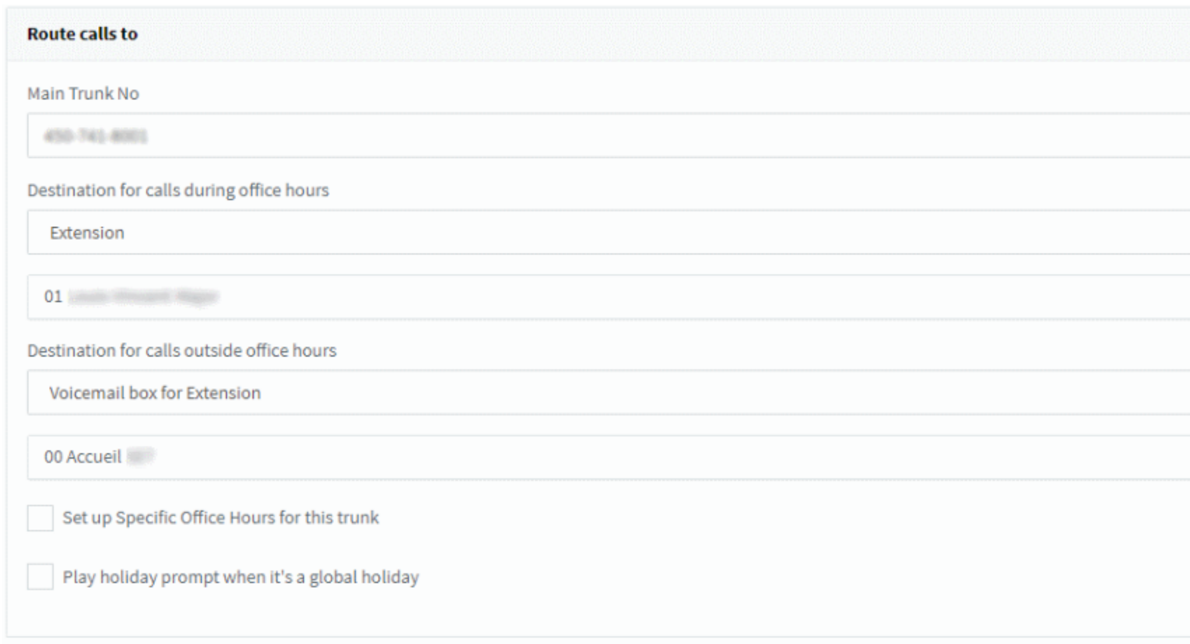
Enter the following information:

- Type of Authentication: **Register/Account based**
- Authentication ID: **SIP User ID provided by Videotron**
- Authentication Password: **Authentication Password provided by Videotron**
- Number of SIM Calls: **Number of simultaneous calls included in your plan**

SIP Trunk - General tab - Route calls to section

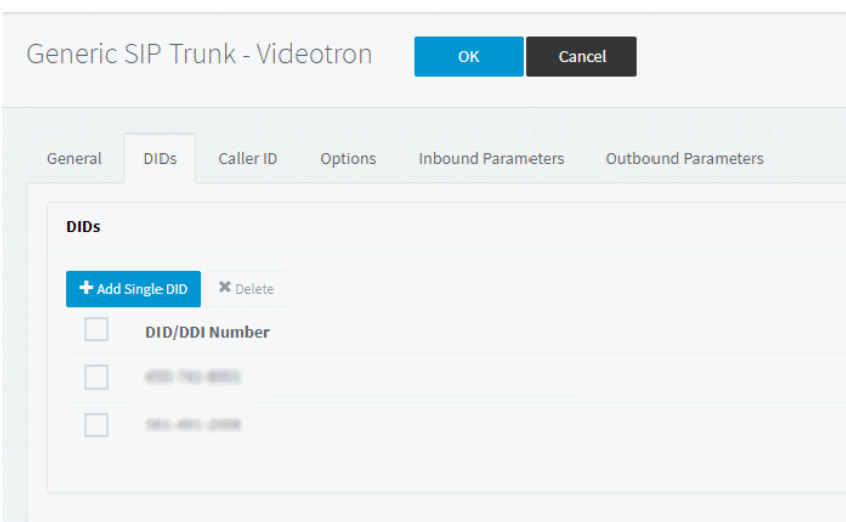
Enter the following information:

- Main Trunk No.: **One of the numbers defined in the DID section (e.g., the primary number)**
- Destination for calls during office hours: **Many destination options are available (e.g., Extension). Second option: One of the preset extensions in the PBX's Extensions section**
- Destination for calls outside office hours: **Same as above (e.g., Voicemail box for Extension).**



SIP Trunk - DID tab

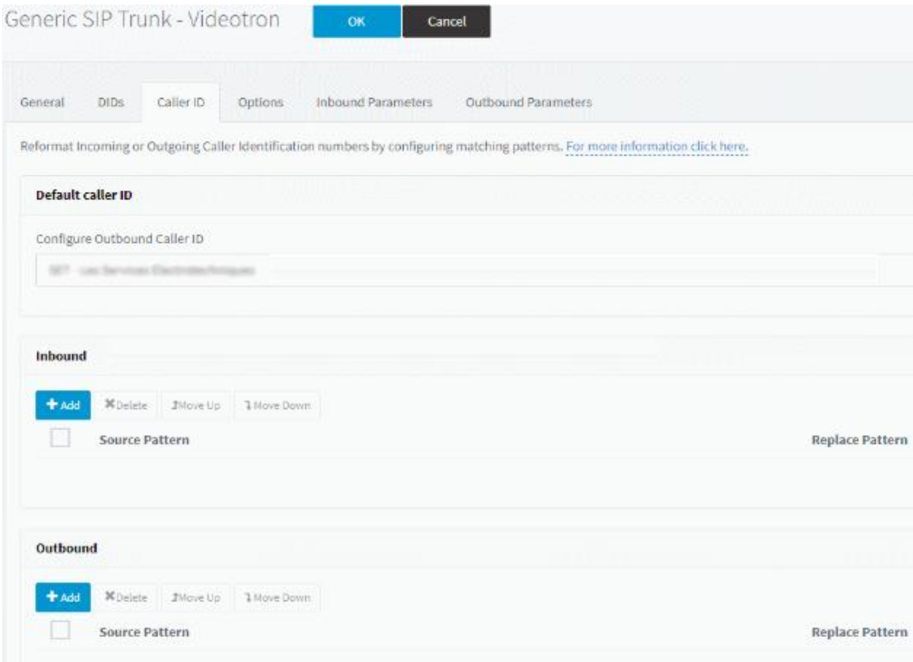
Add all the DID numbers assigned to your SIP trunk:



SIP Trunk – Caller ID tab

In the **Default Caller ID** section, enter the default Caller ID number you want to be displayed for outbound calls. This number can be changed in other display settings in the **Outbound** section of this tab or in the **Outbound Parameters** tab.

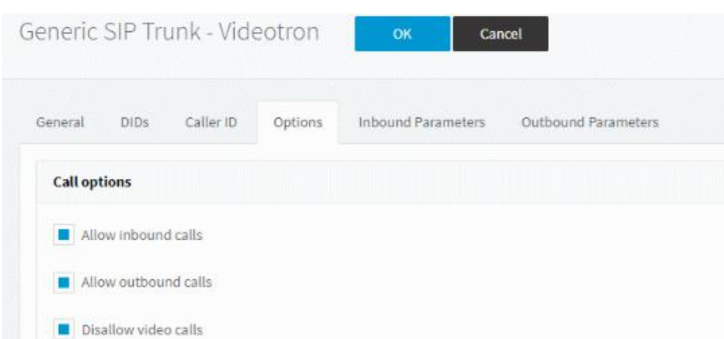
The **Inbound** and **Outbound** sections are used to reformat the Caller ID number using more specific criteria. See link for more details: https://www.3cx.com/docs/cid-reformatting/



SIP Trunk – Options tab – Call Options section

Enter the following information:

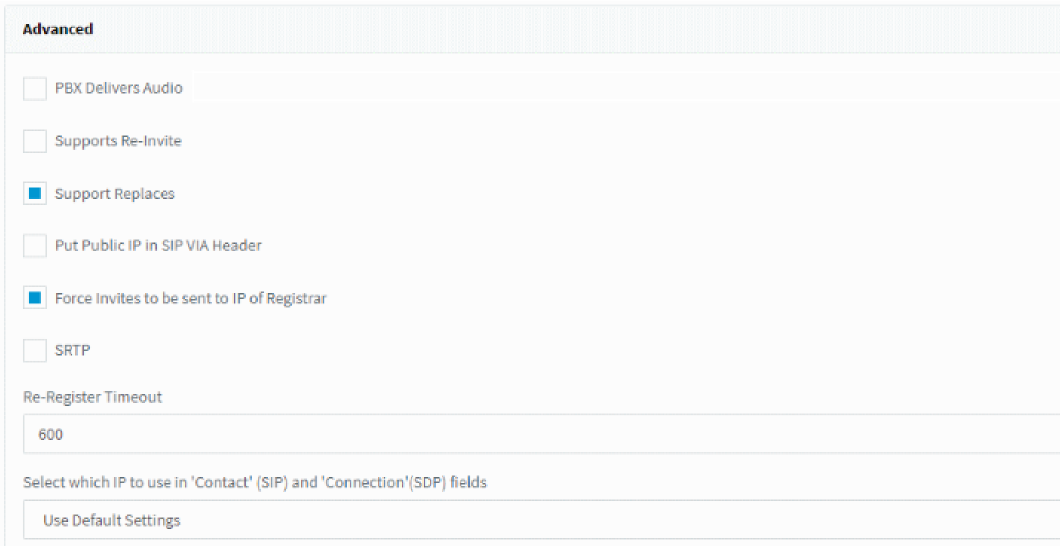
- Allow inbound calls: **Selected**
- Allow outbound calls: **Selected**
- Disallow video calls: **Selected**



SIP Trunk – Options tab – Advanced section

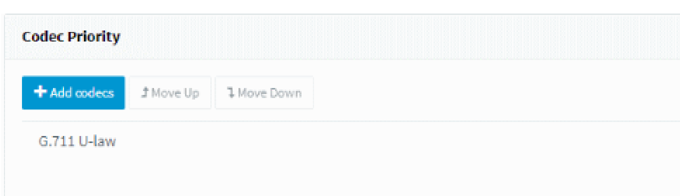
Enter the following information:

- PBX Delivers Audio: **This option determines if the audio streams through the PBX or if it streams directly from the IP telephones to the SIP trunk. Whether or not this option should be selected depends on each customer's network topology.**
- Support Re-Invite: **This option is recommended.**
- Support Replaces: **Option used with SIP-REFER feature.**
- Force Invites to be sent to IP of Registrar: **Selected**
- SRTP: **Not selected**
- Re-Register Timeout: **3600**



SIP Trunk – Options tab – Codec Priority section

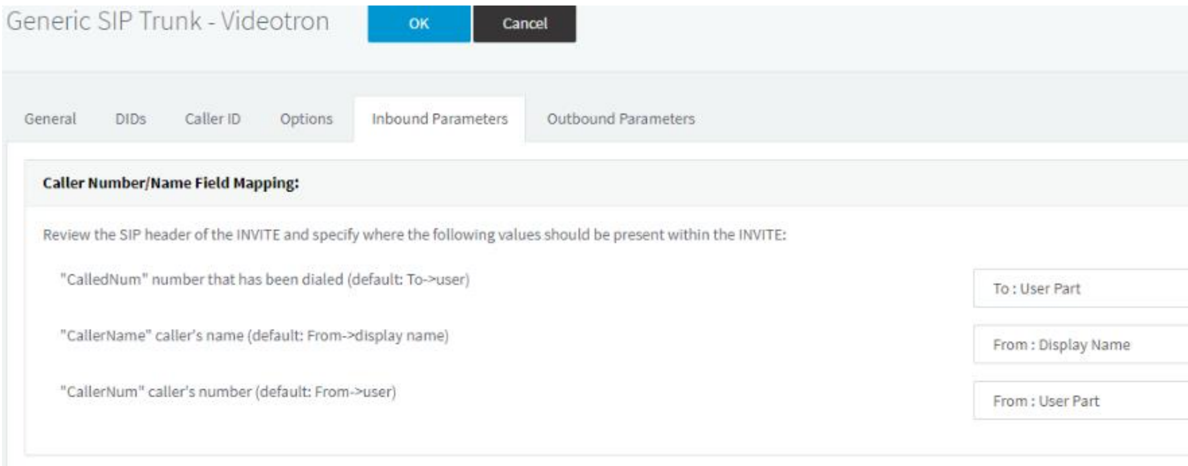
We recommend leaving only G.711 U-Law in the codec list, because it's the only codec Videotron supports.



SIP Trunk – Inbound Parameters tab – Caller Number/Name Field Mapping section

Enter the following information:

- “CalledNum” number that has been dialled: **To: User Part**
- “CallerName” caller’s name: **From: Display Name**
- “CallerNum” caller’s number: **From: User Part**



SIP Trunk – Inbound Parameters tab – Call Source Identification section

Do not do anything in this section.

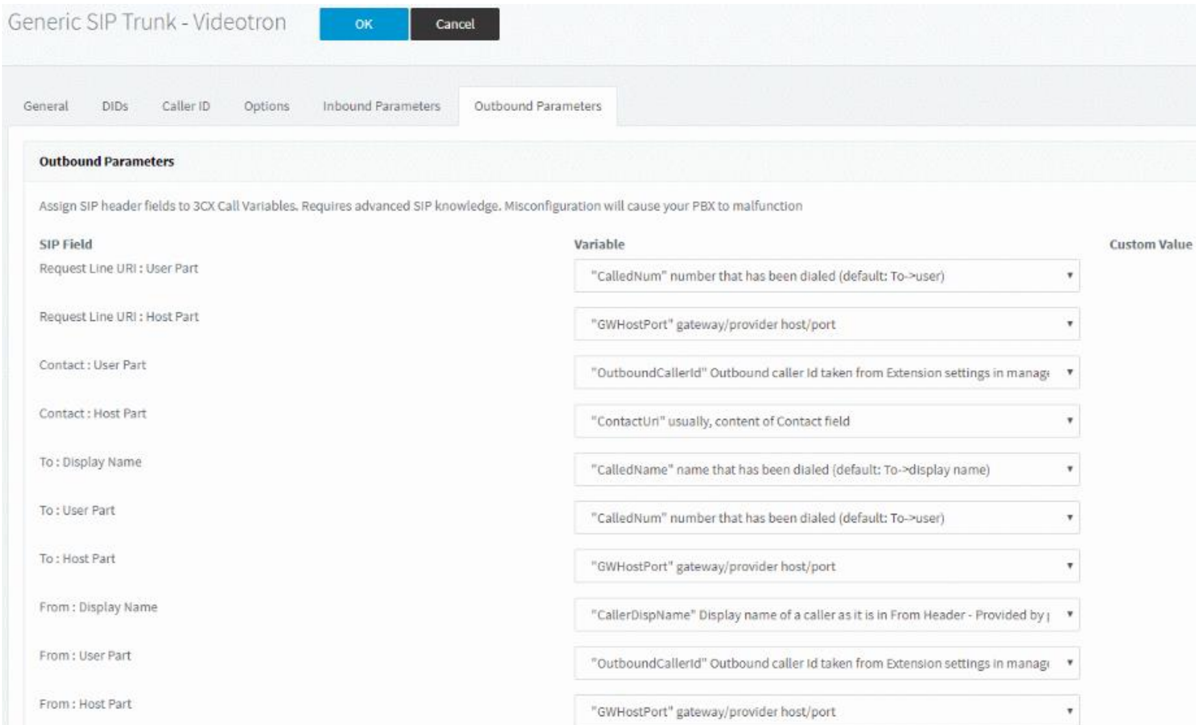
SIP Trunk – Outbound Parameters tab

The default parameters in this section work quite well. Customers may want to make adjustments, particularly with respect to the caller’s name and number (**From: Display Name**, **User Part** and **Host Part**), which must be transmitted for outbound calls from the PBX to the SIP trunk.

Here’s a list of key options that are confirmed to work well with the other recommended settings in this document:

- Request Line URI: User Part: **“CalledNum” number that has been dialled (default: To->user)**
- Request Line URI: Host Part: “**GWHostPort” gateway/provider host/port**
- To: Display Name: **“CalledName” name that has been dialled (default: To->display name)**
- To: User Part: **“CalledNum” number that has been dialled (default: To->user)**
- To: Host Part: **“GWHostPort” gateway/provider host/port**
- From: Display Name: **“CallerDispName” Display name of a caller as it is in From Header**
- From: User Part: **“OutboundCallerId” Outbound caller id taken from Extension settings**
- From: Host Part: **“GWHostPort” gateway/provider host/port**
- Remote Party ID – Calling Party: User Part: **Custom Field (enter the SIP trunk’s primary number)**

Keep the default settings for the rest.



Step 2: Setting inbound rules

This section explains how to configure inbound routes. These routes are used when the PBX gets a new call from the SIP trunk. They direct inbound calls to the appropriate PBX destination (voice menu, voicemail, telephone, etc.) based on the number dialled.

In the **Inbound Rules** section, click **Add DID Rule**.

Enter the following information:

- Name: **A relevant name for this rule (e.g., Videotron – primary number)**
- DID/DDI: **Enter one of the DID numbers assigned to the SIP trunk (e.g., the primary number). You can also use the asterisk symbol (*) to create a rule that applies to a range of dialled numbers instead of just one.**
- Destination for calls during office hours: **Enter a destination for this rule during office hours. Here are the options: End call, Extension,Voicemail box forExtension, Forward to Outside Numberand Send fax to email of extension. For some options, a second destination can be selected. E.g.: for Extension, you must specify the extension.**
- Destination for calls during office hours: **Enter a destination for this rule outside office hours. The options are the same as the previous setting.**

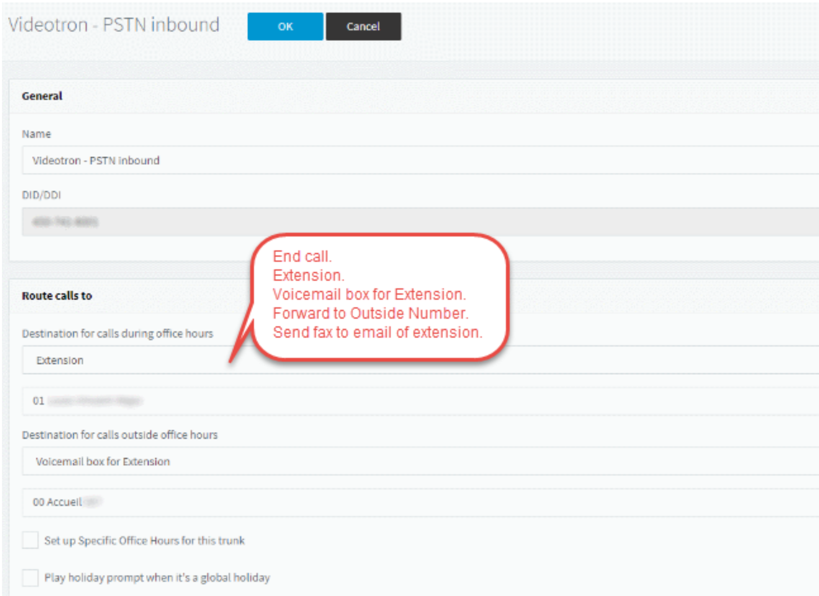
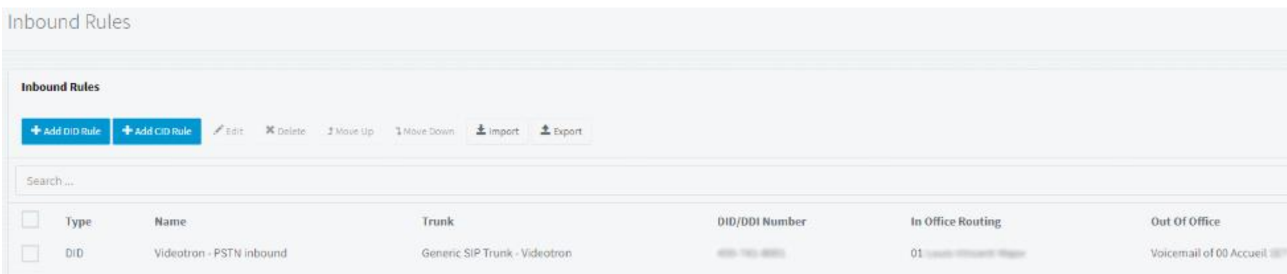


Table after setup is complete.



Step 2: Setting outbound rules

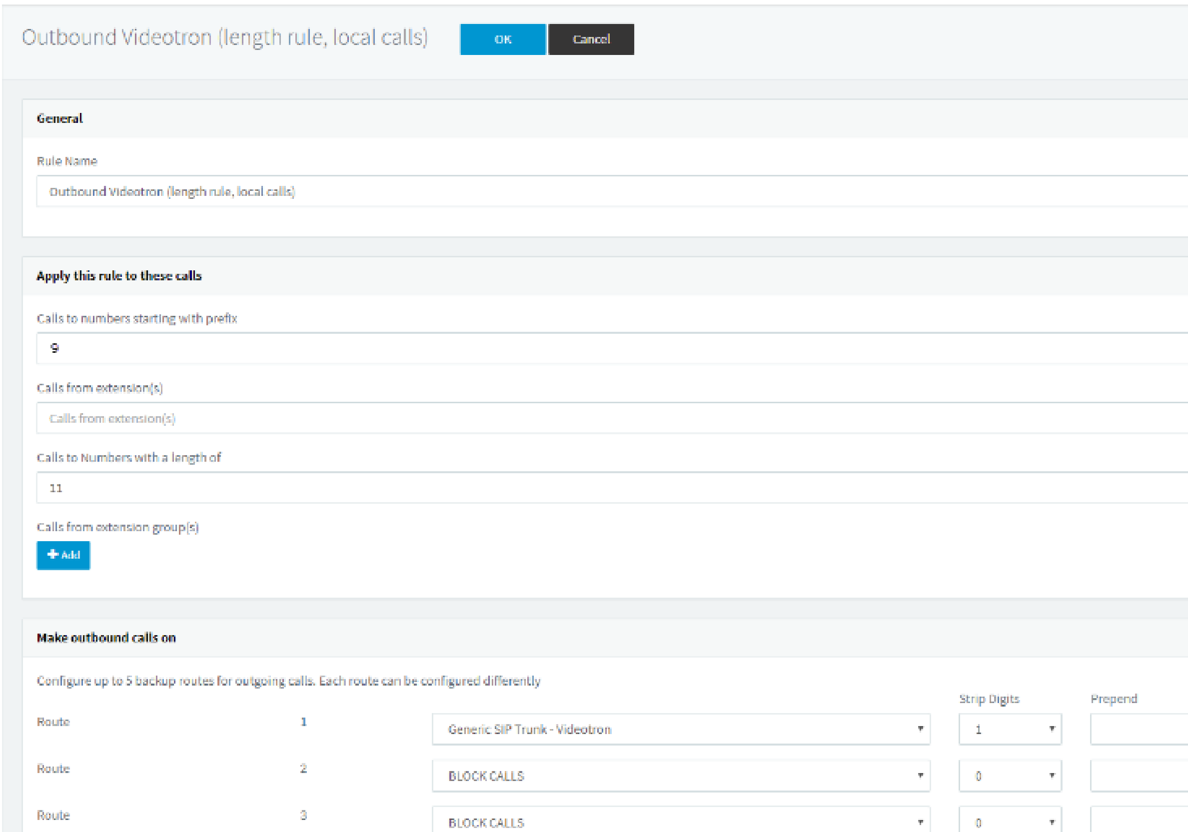
This section explains how to configure outbound routes. These routes are used when PBX lines dial numbers with a specific sequence. When the conditions for this route are met, the call will be made from the PBX to the specified destination. These routes direct outbound calls to Videotron's SIP trunk.

In the **Outbound Rules** section, click **Add**.

Enter the following information:

- Rule Name: **A relevant name for this rule (e.g., Videotron – local calls)**
- Calls to numbers starting with prefix: **This is your outside line code for PSTN calls. The most common prefix is "9." The prefix can be any sequence of digits.**
- Calls from extension(s): **Calls from extension(s):**
- Calls to Numbers with a length of: **The number of digits to be dialled under this rule.**
- Make outbound calls on Route X: **Select the SIP trunk's name (defined in Step 1) from the drop down list.**

- Make outbound calls on Route X – Strip Digits: **Indicate the number of digits to strip before transmitting the call over the SIP trunk. In our example, we would want to strip the 9, which is the outside line code, so we would enter 1.**
- Make outbound calls on Route X – Prepend: **Indicate (if applicable) the digit(s) to add at the beginning of dialled numbers before calls are transmitted over the SIP trunk. In our example, no digits need to be added.**



Telephone systems require multiple outbound routes in order to handle all types of PSTN calls.

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 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. DID number.
G.711	Digital voice encoding standard (audio compression standard)
H.323	Standard for transmitting audio, data and images in real time across packet networks. Used for local networks, like an intranet, or public networks, like the Internet. Less commonly used than 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.
Original Called Number	
PBX	Private branch exchange A company's private telephone switch
PSTN	*Public switched telephone network*
Redirect information	
REFER	SIP method for transferring calls whereby the call is sent to a number indicated in the transfer request. Allows you to free up lines after a call is forwarded from an external number to another external number, such as a cellphone.
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 Logon 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 signalling, 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 to go in the reverse.
T.38	Encoding standard for sending faxes across IP networks in a real-time mode. FoIP

Trunk	A line that connects switches with each other and is used to route information sequentially.
Trunk group (TG)	Circuitry starting from a single switch and terminating at 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.