VIDEOTRON
Business

SIP Trunking Service

Configuration Guide

Avaya IP Office PBX Ver. 9.0

This document was written using gender-neutral language.

The information contained herein is subject to change without prior notice.

Modification history

Edit	Date	Author	Description
1.0	2019-06-28	Pascal Beauregard	Original draft
1.1	2019-09-12	Martin Lefrançois	Review of the consistency with other manuals

Table of Contents

1 Audience

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

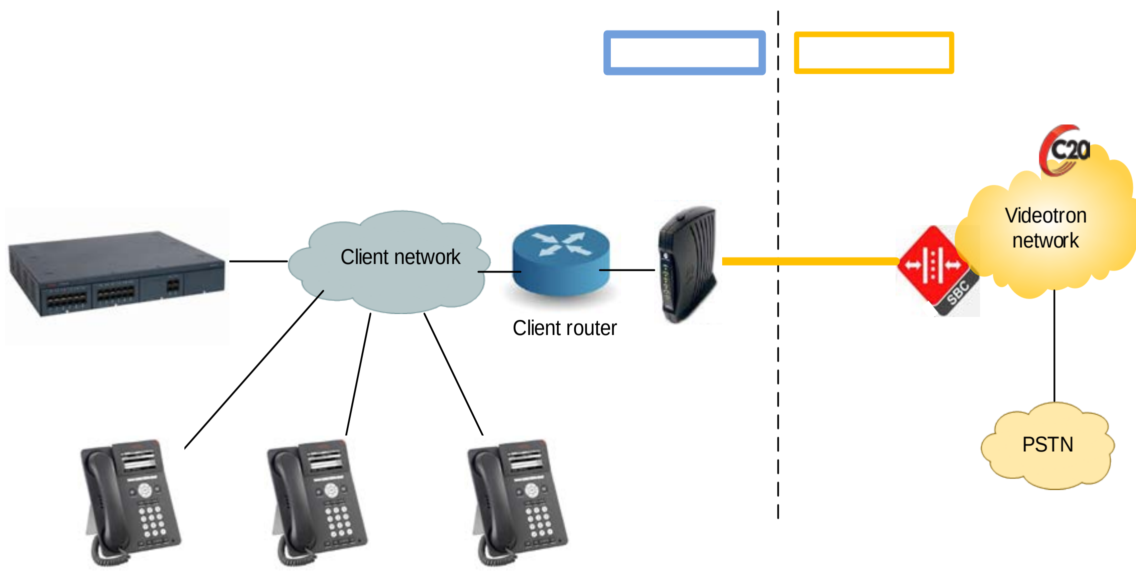
2 Introduction

The *SIP Trunking Service Configuration Guide* details the basic steps for setting up a single SIP trunk between Videotron's SBC and the Avaya IP Office PBX — you can configure several SIP 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 an Avaya IP Office 500 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

4.1 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.	
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.
Failover 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 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.

4.2 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 predefined 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 Registering an 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 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 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.

5.3 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 missing, any calls will be rejected.

5.4 Configuration settings overview

The table below 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
Videotron SBC address	**24.200.242.87**
SIP communication port	**UDP 5060**
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*
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.**

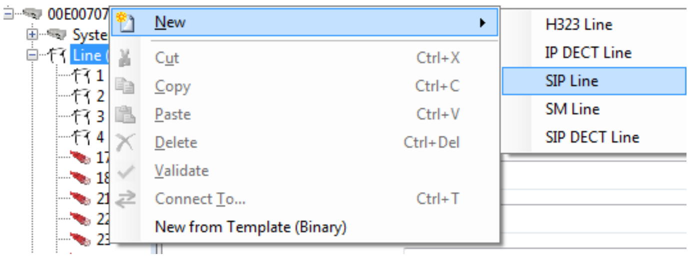
Table 1: Configuration settings overview

6 Configuration

Step 1: Configuring the SIP trunking service

SIP Trunk section

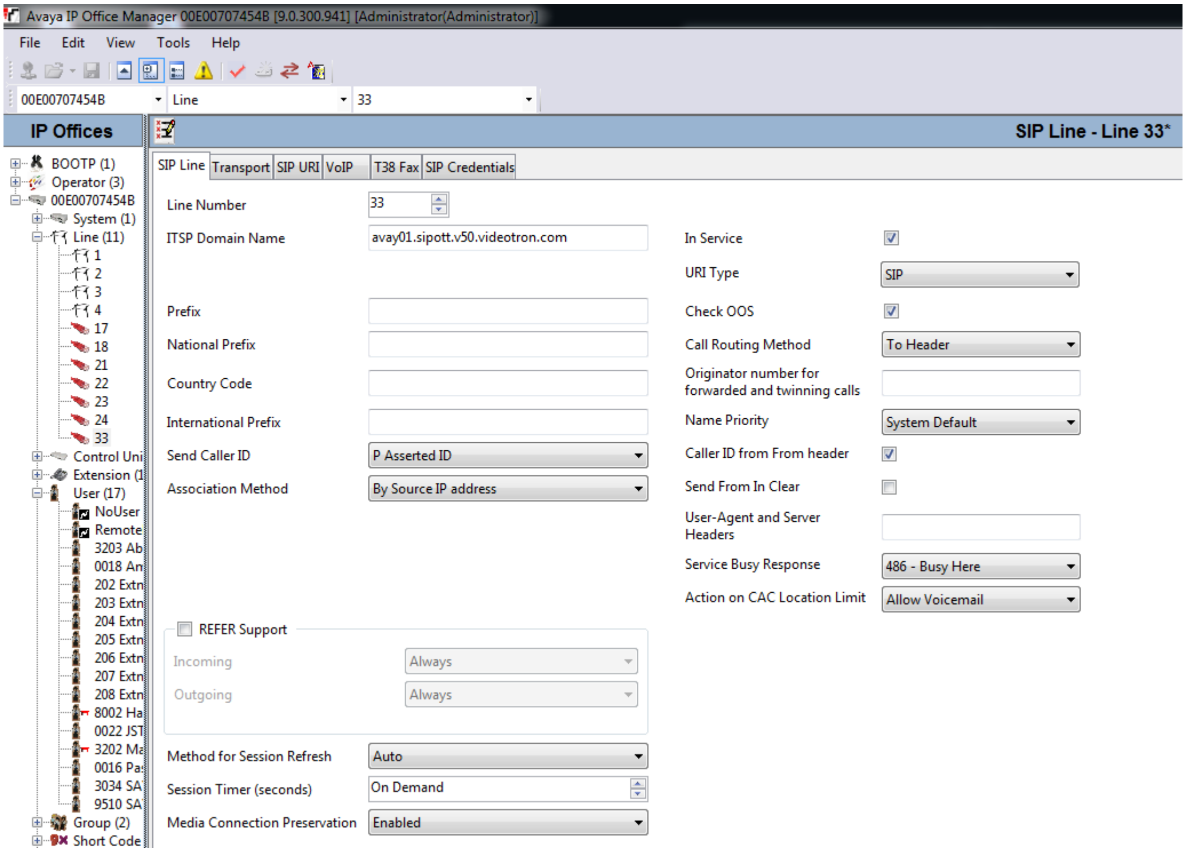
You must add an SIP trunk in the **Line** section. Right click on **Line**, click on **New** and **SIP Line.**



Line - SIP Line tab

Enter the following information in the SIP line's configuration page.

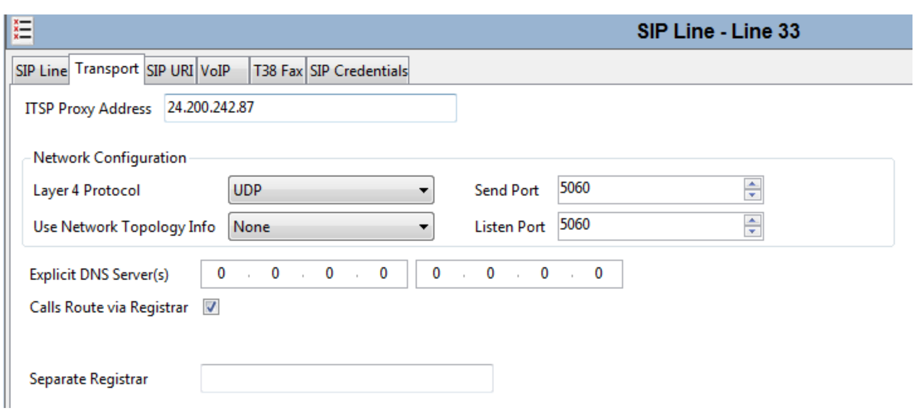
- Line Number: **Enter a free line number.**
- ITSP Domain Name: **Domain name provided by Videotron (e.g., cust01.sipott.v50.videotron.com)**
- Send Caller ID: **P Asserted ID**
- Association Method: **By Source IP address**
- REFER Support: **You must activate the SIP-Refer here if required. However, please contact Videotron to understand the potential issues that could arise when activating this feature.**
- Method for Session Refresh**: Auto**
- Session Timer (seconds): **On Demand**
- Media Connection Preservation**: Enabled**
- In Service: **Selected**
- URI Type**: SIP**
- Check OOS: **Selected**
- Call Routing Method**: To Header**
- Caller ID from From header: **Selected**
- Service Busy Response: **486 – Busy Here**



Line - Transport tab

Enter the following information:

- ITSP Proxy Address: **24.200.242.87**
- Network Configuration – Layer 4 Protocol: **UDP**
- Network Configuration – Send Port: **5060**
- Network Configuration – Use network Topology: **None**
- Network Configuration – Listen Port: **5060**
- Calls Route via Registrar: **Selected**



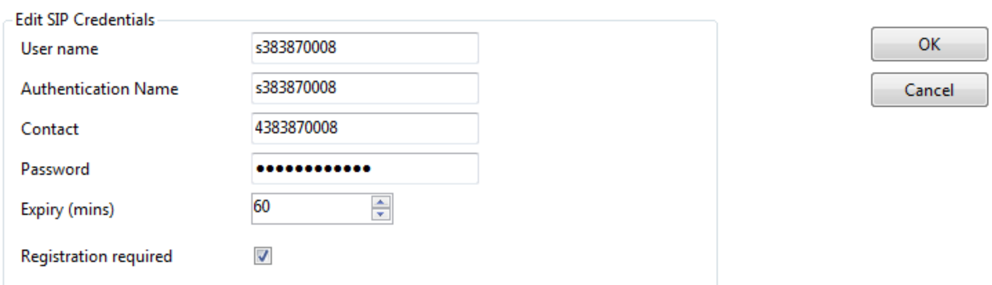
Line - SIP Credentials tab

Two entries must be created: the first entry with the service registration information and the second with fake information required for inbound calls and used in the SIP URI tab.

To add the first entry, click on **Add...** in the **SIP Credentials** tab.

Enter the following information:

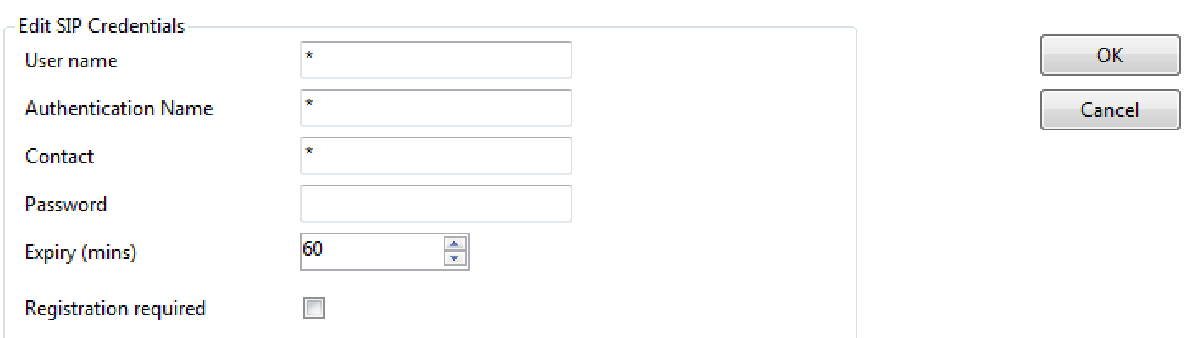
- User name: **SIP User ID provided by Videotron**
- Authentication Name: **SIP User ID provided by Videotron**
- Contact: **The SIP trunk's primary number**
- Password: **Authentication Password provided by Videotron**
- Expiry (mins): **60**
- Registration required: **Selected**



To add the second entry, click on **Add...** in the **SIP Credentials** tab.

Enter the following information:

- User name: *
- Authentication Name: *
- Contact: *
- Password: **Empty field**
- Expiry (mins): **60**
- Registration required: **Not selected**



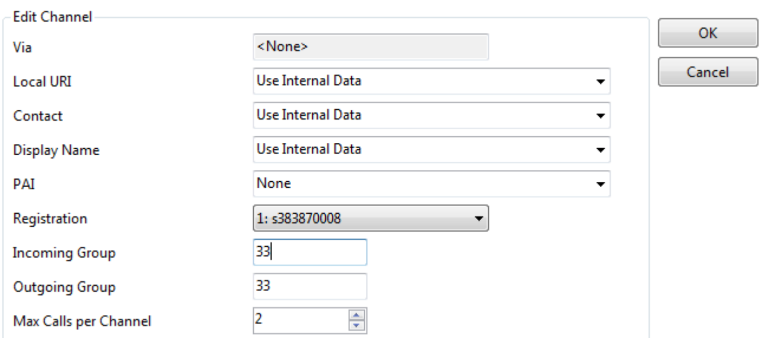
Line - SIP URI tab

Two entries must be created, the first entry with the service registration information and the second with fake information created in the SIP Credentials tab.

To add the first entry, click on **Add...** in the **SIP URI** tab.

Enter the following information:

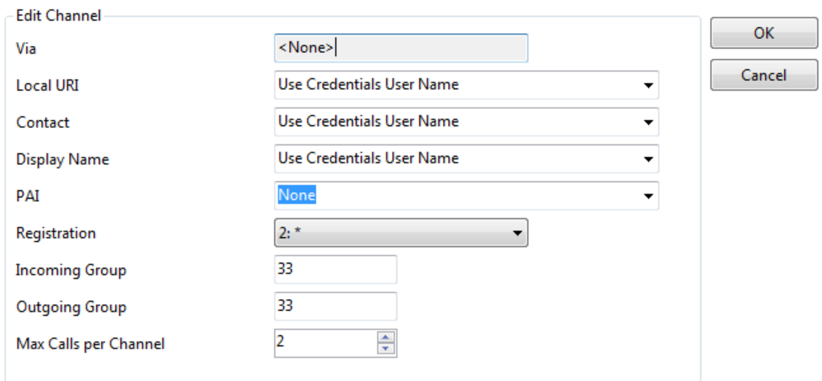
- Via: **Empty field**
- Local URI: **Use Internal Data**
- Contact: **Use Internal Data**
- Display Name: **Use Internal Data**
- PAI: **None**
- Registration: **Select the first entry created in the SIP Credentials tab from the drop-down list.**
- Incoming Group: **Enter the number of the line that you are creating**.
- Outgoing Group: **Enter the number of the line that you are creating.**
- Max Calls per Channel: **Number of simultaneous calls on the SIP trunk agreed upon with Videotron**



To add the second entry, click on **Add...** in the **SIP Credentials** tab.

Enter the following information:

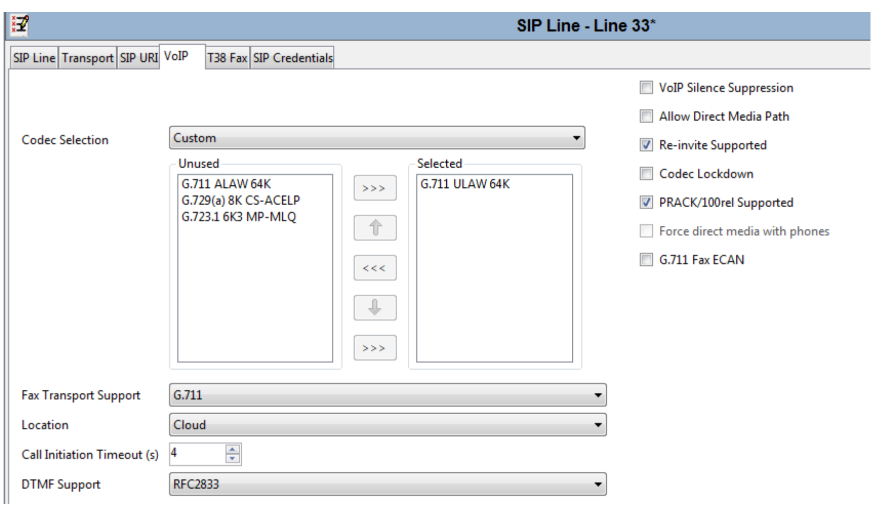
- Via: **Empty field**
- Local URI: **Use Credentials User Name**
- Contact: **Use Credentials User Name**
- Display Name: **Use Credentials User Name**
- PAI: **None**
- Registration: **Select the second entry created in the SIP Credentials tab from the drop-down list**.
- Incoming Group: **Enter the number of the line that you are creating.**
- Outgoing Group: **Enter the number of the line that you are creating**.
- Max Calls per Channel: **Number of simultaneous calls on the SIP trunk agreed upon with Videotron**



Line - VoIP tab

Enter the following information:

- Codec Selection: **Custom**
- Codec Selection - Selected: **G.711 ULAW 64K**
- Fax Transport Support: **G.711**
- Location: **Cloud**
- Call Initiation Timeout (s): **4**
- DTMF Support: **RFC2833**
- VoIP Silence Suppression: **Not selected**
- Allow Direct Media Path: **At the customer's discretion, we have tested without this option selected**.
- Re-invite Supported: **Selected**
- Codec Lockdown: **Not selected**
- PRACK/100rel Supported: **Selected**
- G.711 Fax ECAN: **Not selected**



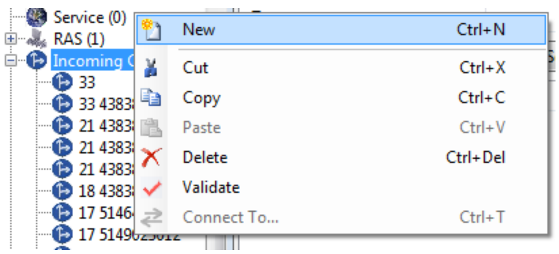
Line - T38 Fax

Nothing needs to be configured in this tab as Videotron does not support the T.38 protocol on the SIP trunks.

Step 2: Setting inbound rules

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

You must add at least one inbound route in the **Incoming Call Route** section. Right click on **Incoming Call Route,** click on **New.**

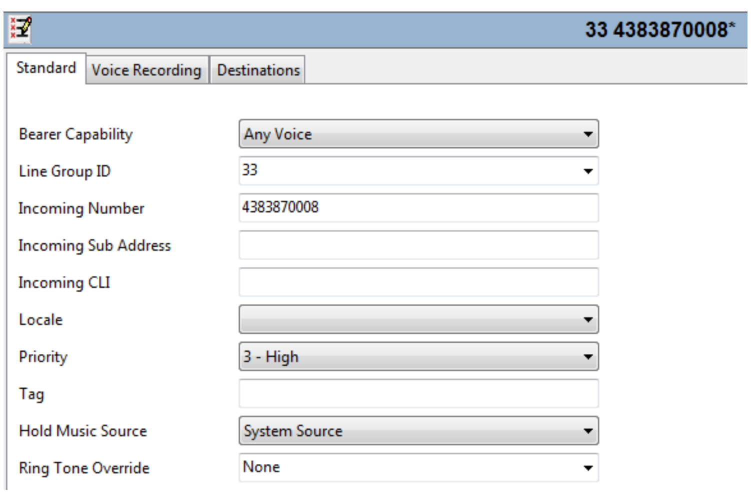


Incoming Call Route – Standard tab

Enter the following information:

- Bearer Capability: **Any Voice**
- Line Group ID: **Enter the number of the line that you created earlier.**
- Incoming Number: **One of the phone numbers (DID) assigned to your SIP trunk**

The other settings in this tab are specific to each customer.

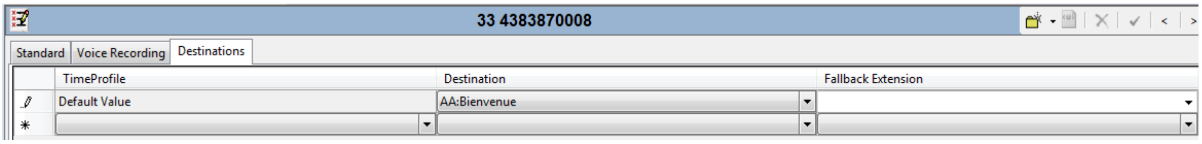


Incoming Call Route – Destinations tab

Enter the following information:

- Time Profile: **A schedule that has previously been configured may be selected and used.**

- Destination: **Choose one of the PBX destinations (voice menu, phone extension, voice messaging, call group) from the drop-down list for this route's destination**.
- Fallback Extension: **Secondary destination (optional), if the primary destination does not work. Same drop-down list.**

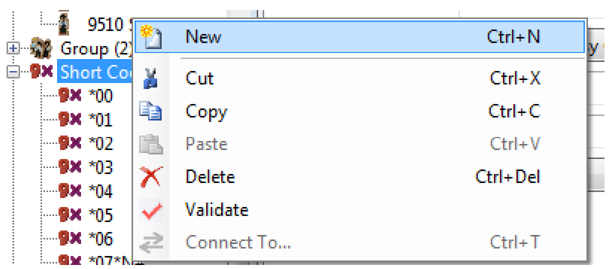


When the first inbound route has been entered, you must enter another route per phone number assigned to your SIP trunk.

Step 3: Setting outbound rules

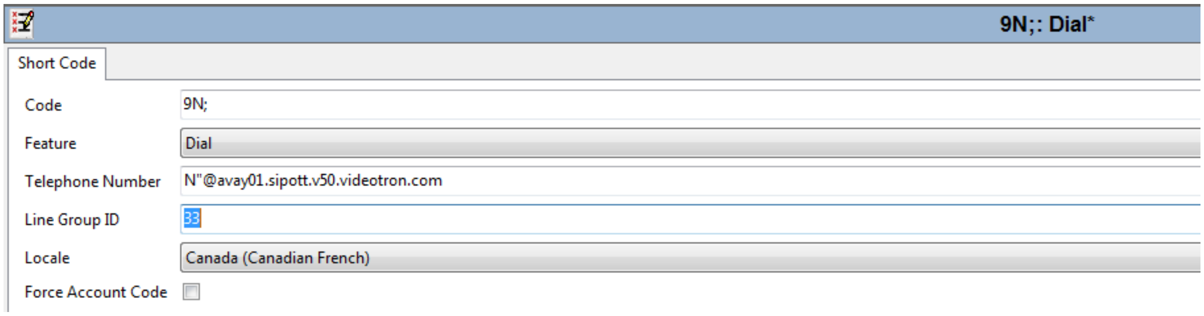
This section explains how to configure outbound routes. These routes are activated when PBX lines dial numbers with a specific sequence. When the conditions that suit this route are met, the call will be made from the PBX to the destination specified by this route. These routes direct outbound calls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

You must add at least one outbound route in the **Short Code** section. Right click on **Short Code,** click on **New.**



Enter the following information:

- Code: **This is your outbound code for calls to the PSTN. Very often, 9 will be the chosen outbound code, but it can be any sequence of numbers. The Avaya syntax must be used here to specify the expected sequence. (E.g., 9N)**
- Feature: **Dial**
- Telephone Number: **The Avaya syntax must again be used here. The dialed number and domain name must be specified (e.g.: N"@cust01.sipott.V50.videotron.com)**
- Line Group ID: **Enter the number of the line that you created earlier.**
- Locale: **At the customer's discretion**



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

7 Glossary

503	Service unavailable Server error code.
bursting	Feature that enables you to temporarily exceed the simultaneous calling limit stipulated in your contract. Simultaneous calls are billed on a pay-per-use basis. Feature currently in development
called number	Number called or requested
called party	Person to whom a call is sent.
calling party	Person sending a call to establish communication.
C20	Videotron telephone switch
CO line	Central office line Communication line that connects a PBX to a telephone service provider's switchboard.
DID	direct inward dialling Telephone feature enabling 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
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 The most commonly used telephone system when few additional extensions are required. Enables users to call each other directly and to communicate with public network subscribers through outbound and inbound calls.
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. Enables 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 enabling you to configure SIP trunk settings.
SIP	session initiation protocol Login 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.
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.