SIP Trunking Service Configuration Guide

Generic Guide

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Modification history

Edit	Date	Author	Description
1.0	2019-05-30	Pascal Beauregard	Original draft
1.1	2019-06-25	Martin Lefrançois	Edits based on another document
1.2	2019-07-17	Louis Villemur	Linguistic revision
1.3	2019-08-19	Martin Lefrançois	Validation and correction

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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 customer's IP PBX. This document goes over the steps for setting up multiple SIP trunks.

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

It's a generic configuration guide for all PBX models.

3 Network and equipment diagram

The diagram below is an overhead view of SIP trunking with a customer's PBX.

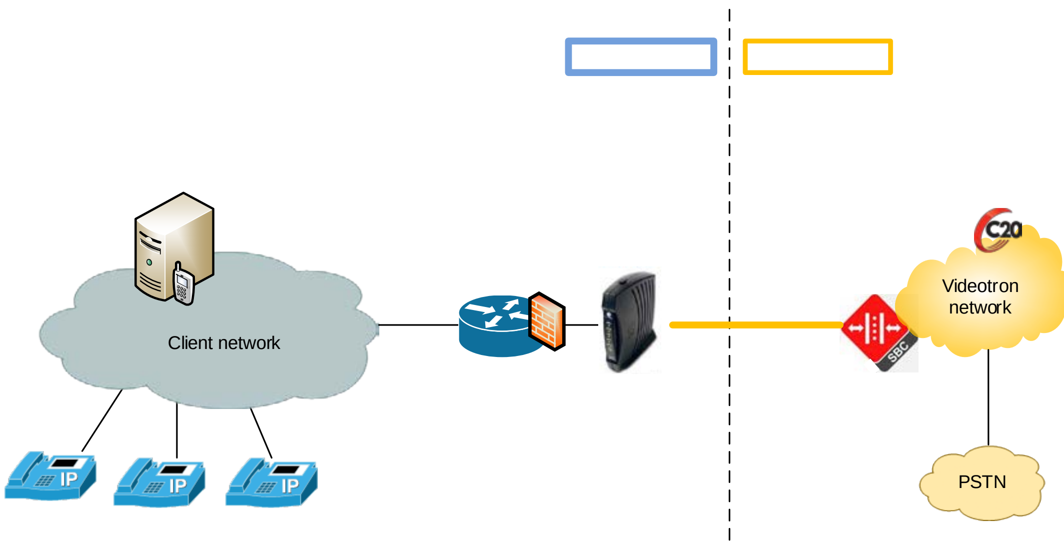

Figure 1: SIP trunking service network components

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. 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 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.
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 if the call is forwarded through another Videotron switch. Routing between Videotron switches is subject to change without prior notice.

Table 2: Supported features

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.

Table 3: Unsupported features

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, any 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:

- the domain name
- the username
- the 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.

6 Configuration

6.1 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>
Password	**Provided by Videotron:** 12 alphanumeric characters with at least 1 lowercase letter, 1 uppercase letter, and 1 number
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 4: Configuration settings overview

The following sections explain the table’s various settings and go over the other settings required to ensure proper communication over the SIP trunk. These sections also provide common names for these settings and screenshots to help integrators set up a PBX.

6.1.1 Authentication data

PBX systems use various names for the registration fields you have to configure. Here’s a table of registration settings and the most common field names associated with them.

Domain name	Provided by Videotron: <customer acronym>.sipott.v50.videotron.com
	E.g.: cust01.sipott.v50.videotron.com
	Common field names: Domain name, SIP Service Domain, ITSP Domain Name, Proxy Domain Name, Registrar Domain name
Username	Provided by Videotron: s<last 8 numbers of primary telephone number>
	E.g.: s143801234
	Common field names: UserName, UserID, Authentication ID, Authentication Name
Password	Provided by Videotron: 12 alphanumeric characters with at least 1 lowercase letter, 1 uppercase letter, and 1 number
	E.g.: aQkTZaxvHz7phrLY
	Common field names: Authentication password, password
Videotron SBC address	**24.200.242.87**
	Common field names: ITSP Proxy address, SIP Server IP Address, Registrar Address Some PBX systems use the same setting for the registration address and the call server address (e.g., Registrar/SIP Server address). This setting is therefore listed below.
Communication port	**UDP 5060**
	Common field names: SIP server port, Proxy port, port

Table 5: Authentication settings overview

6.1.2 Format of required SIP REGISTER messages

The format of the SIP REGISTER messages transmitted during registration is important. Most PBX systems can capture SIP messages sent to and from their network interfaces. The integrator can use these captured SIP messages to check if the SIP messages sent from the customer’s PBX use the format requested by Videotron.

See Appendix 1 for an example of a SIP trunking service registration transaction showing the format required for SIP REGISTER messages sent by the PBX to be accepted by Videotron servers.

6.2 Configuring the settings to make calls on the SIP trunk

The remaining SIP trunk settings are configured very differently from one PBX model to another. This section covers the minimum required settings for a PBX to be able to use a SIP trunk.

Trunking service settings can usually be divided into three broad categories:

1. General SIP trunk settings
2. Outbound settings
3. Inbound settings

6.2.1 General SIP trunk settings

Here is a list of general PBX settings that must be configured in order to use a SIP trunk.

Number of simultaneous calls on the SIP trunk	**Provided by Videotron**
	Common field names: Max Calls, Maximum Channels
	Some PBX systems do not limit the number of simultaneous calls on a SIP trunk.
Videotron SBC address	**24.200.242.87**
	Common field names: Proxy Address, SIP Server IP Address
	Some PBX systems use the same setting for the registration address and the call server address (e.g., Registrar/SIP Server address). Registrar/SIP Server address.
SIP communication port	**UDP 5060**
	Common field names: SIP server port, Proxy port, port
	This is the standard SIP protocol communication port.
Codec	**G.711 µ-law only**
	Common field name: Codec
	Voice encoding method: G.711 is an uncompressed encoding method.
Fax protocol	**In-Band (T.38 not supported)**
	Common field names: Fax protocol, Fax support, T.38 support, T.38 Fax
	Videotron does not support the T.38 transmission mode. If the field is named T.38, it must be deactivated. The system will then default to In-Band signalling.
DTMF	**RFC2833**
	Common field names: DTMF mode, DTMF Support, DTMF
	DTMF (dialled number) transmission method once call is in progress.
SIP REFER	**Deactivated (may be activated, but must take into account what's mentioned in Section 7.1.)**
	Common field names: SIP REFER
	Call management method that frees up lines when a call is transferred to the PSTN. Videotron supports the protocol, but recommends deactivating the feature because calls may not be transferred properly. Please contact Videotron if you want to activate it.

Table 6: General SIP trunk settings

6.2.2 Outbound settings

Outbound routes must be configured on all PBX systems so calls sent to the SIP trunk know which number sequence to use. In addition to outbound routes, there are certain standardized settings to configure for outbound calls.

The names of the settings to configure and the section to find them in vary widely from one PBX to another. Here are some general guidelines to help you set up your PBX.

Outbound routes

Common section names: Trunk access code, Short code, Route pattern, Outbound Route, Outbound Rule.

Businesses usually use "9" as the outside line code. Consequently, most PBX systems have at least one route that starts with 9 and connects to the SIP trunk.

Common routes: 9*, 9XXXXXXXXXX, 9N or another code

For reference, here are screenshots of two known PBX systems: Avaya IP Office 500 and 3CX.

Avaya IP Office 500 – Short Code Section

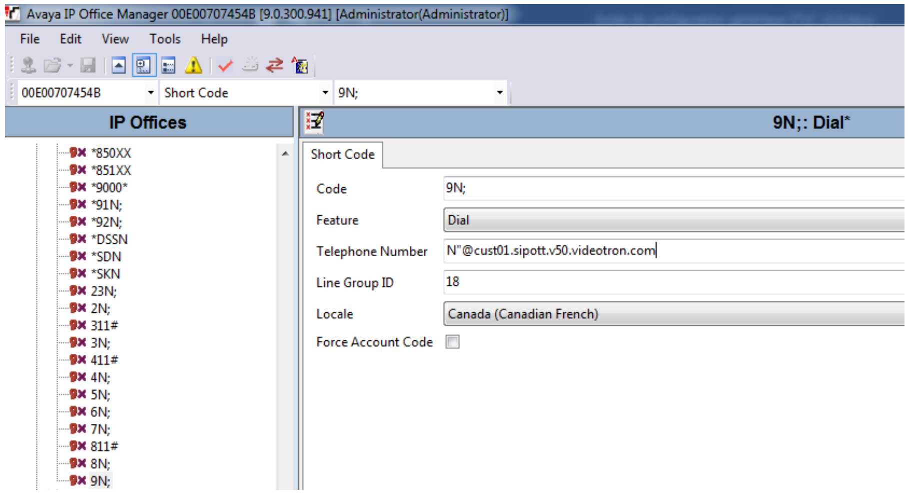

Figure 7: Short Code, Avaya IP Office 500

Using the exit code “9” followed by any number will connect to the SIP trunk (Line Group ID 18) with this PBX.

3CX – Outbound Rules Section

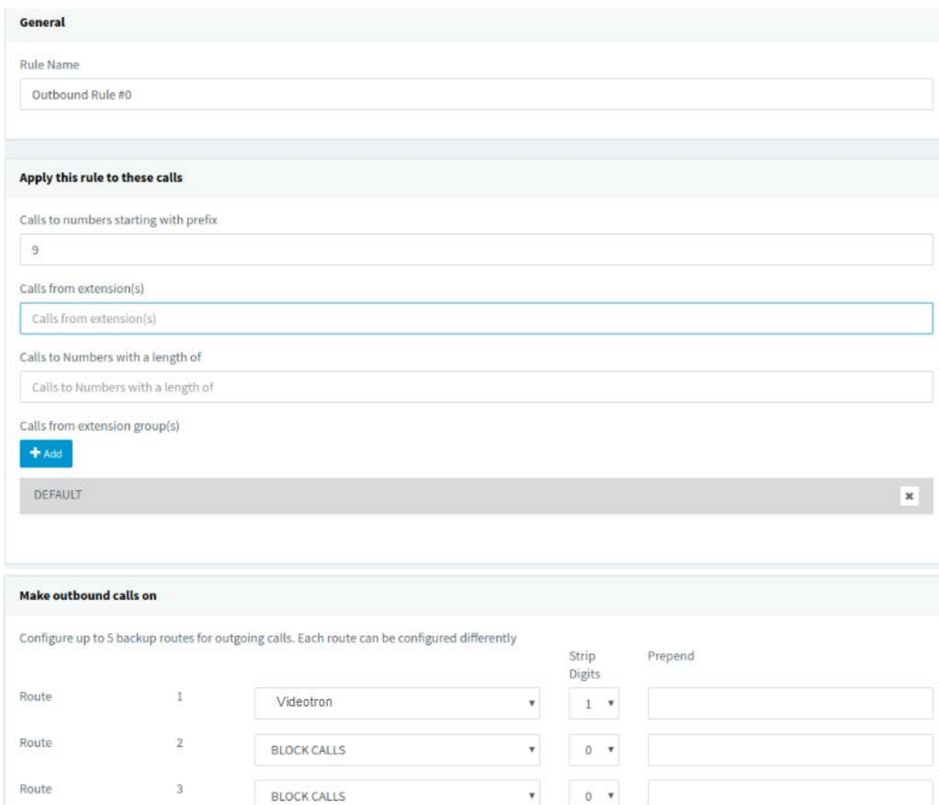

Figure 8: Outbound Rules, 3CX

Using the outside line code “9” will send calls to the preconfigured “Videotron” SIP trunk

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Outbound display for regular calls

For outbound calls (from PBX to PSTN), the caller ID number transmitted by your PBX will be relayed to the PSTN by the Videotron switch at any time, except for 911 calls.

Outbound display for 911 calls

For 911 calls, the caller ID number transmitted by the PBX will be relayed to the 911 call centre only if the number is on the predefined list of numbers the customer gave Videotron. If the caller ID number transmitted by the PBX for a 911 call is not on the list, the caller ID number will be replaced by the customer's primary number for that call.

> **E.g.:** The customer's primary number is 514-380-1234.
> The list of numbers given to Videotron for 911 calls is: 514-380-1234, 514-380-5678, 438-387-2468.
> A 911 call is made from the number 514-380-0010 (not on the list).
> The Videotron switch replaces the number 514-380-0010 with 514-380-1234.

PBX setting used for caller ID number

You can choose what PBX setting will be used for caller ID numbers when making calls to the PSTN.

The options are usually as follows:

1. The calling phone's extension (CLID, PBX-CLIP, User Extension, etc.)
2. A unique company number
3. The SIP line's user name

This information is transmitted to the "From" header of SIP messages. You must select option 1 or 2, but you must not select the user name because it can contain characters that the Videotron switch does not accept.

Private calls

PBX telephones may be used to make private calls to the PSTN. If the PBX's numbering plan allows private outbound calls (no caller ID displayed), an additional PBX setting must be configured.

The "Privacy ID" or "P-Asserted-ID" must be selected for the Videotron switch to properly handle the Caller ID for this type of call.

See Appendix 3 for an example of a SIP INVITE message that shows the format accepted by Videotron servers for "private" calls.

6.2.3 Inbound settings

DIDs must be configured on all PBX systems in order to determine where inbound calls go (voice menu, specific phone, call group, etc.). There are also display settings to configure for inbound calls.

DID settings

Common section names: Inbound Routes, Inbound Rules, Incoming Call Route, Translation Pattern, etc.

All DIDs must be entered individually or as a range of numbers in the appropriate PBX section. Each DID leads to a specific PBX resource.

For reference, here are screenshots of two known PBX systems: Avaya IP Office 500 and Yeastar S-Series.

Avaya IP Office 500 – Incoming Call Route Section

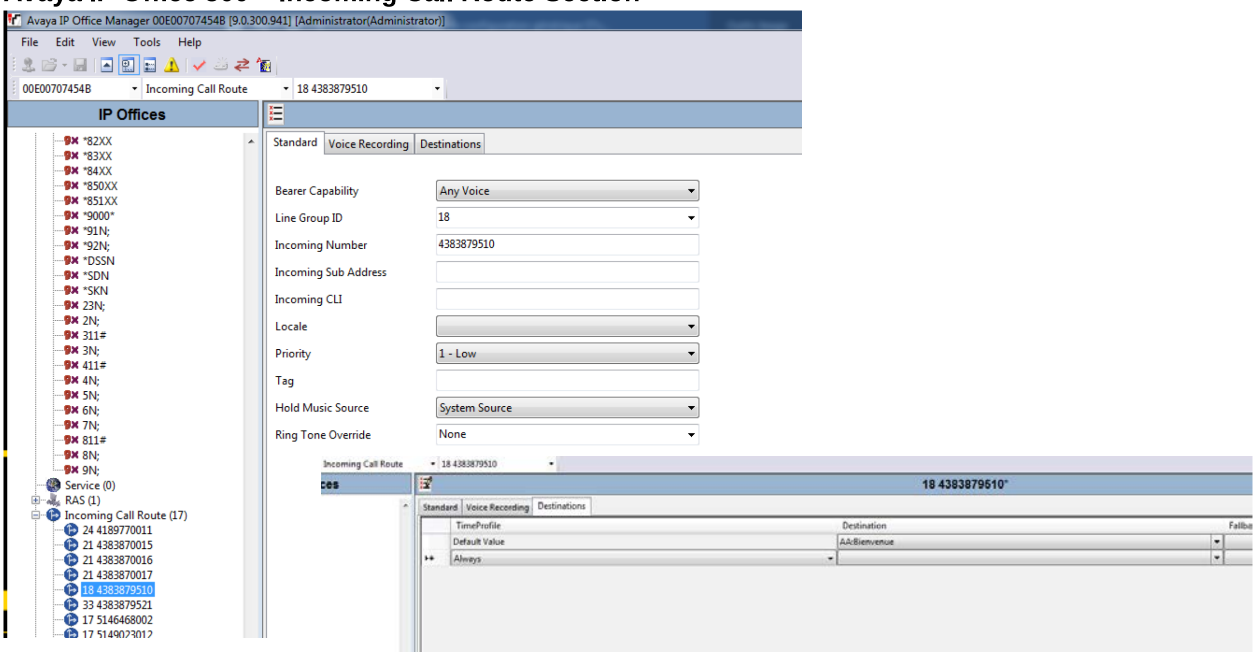

Figure 9: Incoming Call Route, Avaya IP Office 500

For inbound calls on the SIP trunk (Line group ID 18), if the dialled number is 438-387-9510, the call will be routed to the voice menu “AA: Welcome.”

Yeastar – “Inbound Route” section

Figure 10: Inbound Route, Yeastar S-series

For inbound calls on the SIP trunk (Videotron), if the dialled number is 514-380-0018, the call will be routed to the sales call group.

Call routing method

Many PBX systems allow you to choose which headers of inbound SIP INVITE messages are used to rout calls.

The two usual options are:

1. The user part of the Req URI
2. The user part of the “To” header

The Videotron switch sends the same value (caller ID number) to the Req URI and the “To” header, so both options are valid.

6.2.4 Format of SIP INVITE messages for outbound calls

The format of SIP INVITE messages sent from the PBX to the SIP trunk for outbound calls is important. If the requested format is not used, the call will be dropped. Most PBX systems can capture SIP messages sent to and from their network interfaces.

Here are the main requirements:

1. The Req URI must contain the called number and the customer's domain name.
   E.g.: INVITE sip:5143801234@cust01.sipott.v50.videotron.com:5060 SIP/2.0

2. The "From" header must contain the caller's number and, if available, the caller's name.
   E.g.: From: "ABC inc"<sip:4383870016@<PBX IP address>:5060>;

3. The "To" header must contain the called number.
   E.g.: To: <sip:5143801234@24.200.242.87:5060>

Note: The "host" part of the "From" and "To" headers can also be the domain name.

See Appendix 2 for an example of a PBX call setup with properly formatted SIP INVITE messages.

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 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.
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 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.

Appendix 1: Registration example: SIP REGISTER request in the right format

Sent by PBX (original request without username and password)

```
Sent :
REGISTER sip:cust01.sipott.v50.videotron.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.247.44.55:5060;branch=z9hG4bK79A429262C
From: <sip:s143801234@cust01.sipott.v50.videotron.com>;tag=E91DABD0-2575
To: <sip:s143801234@cust01.sipott.v50.videotron.com>
Date: Fri, 24 May 2019 20:47:58 GMT
Call-ID: 8F0D75A9-7D9B11E9-B6C0B188-A1E25E41
User-Agent: Cisco-SIPGateway/IOS-15.5.3.M
Max-Forwards: 70
Timestamp: 1558730878
CSeq: 4 REGISTER
Contact: <sip:s143801234@10.247.44.55:5060>
Expires: 3600
Supported: path
Content-Length: 0
```

Received by PBX (Videotron server sends SIP Trying response)

```
Received :
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK79A429262C;rport=5060
From: <sip:s143801234@cust01.sipott.v50.videotron.com>;tag=E91DABD0-2575
To: <sip:s143801234@cust01.sipott.v50.videotron.com>
Call-ID: 8F0D75A9-7D9B11E9-B6C0B188-A1E25E41
Timestamp: 1558730878
Cseq: 4 REGISTER
Content-Length: 0
```

Received by PBX (Videotron server requests another SIP Register message with the password this time)

```
Received :
SIP/2.0 407 Proxy Authentication Required
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK79A429262C;rport=5060
From: <sip:s143801234@cust01.sipott.v50.videotron.com>;tag=E91DABD0-2575
To: <sip:s143801234@cust01.sipott.v50.videotron.com>;tag=907391964
Call-ID: 8F0D75A9-7D9B11E9-B6C0B188-A1E25E41
Timestamp: 1558730878
Cseq: 4 REGISTER
User-Agent: Nortel SESM 18.0.31.0
Supported: com.nortelnetworks.firewall,p-3rdpartycontrol,nosec,join,x-nortel-sipvc,gin
Proxy-Authenticate: Digest
realm=”Realm”,nonce=”MTU1OdczMDg2MTE3N2FhNDMzN2Y3NzlkNjJjMmM3ZmQ1NjQ5NzQzZjZhMGFl”,stale=false,algorithm=MD5,qop=”
auth”
Content-Length: 0
```

Sent by PBX (PBX sends another SIP Register request with the username, encrypted [hashed] password and domain)

```
Sent:
REGISTER sip:cust01.sipott.v50.videotron.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.247.44.55:5060;branch=z9hG4bK79A42A86D
From: <sip:s143801234@cust01.sipott.v50.videotron.com>;tag=E91DABD0-2575
To: <sip:s143801234@cust01.sipott.v50.videotron.com>
Call-ID: 8F0D75A9-7D9B11E9-B6C0B188-A1E25E41
User-Agent: Cisco-SIPGateway/IOS-15.5.3.M
Max-Forwards: 70
Timestamp: 1558730878
Cseq: 5 REGISTER
Contact: <sip:s143801234@10.247.44.55:5060>
Expires: 3600
Proxy-Authorization: Digest
username=”s143801234”,realm=”Realm”,uri=”sip:cust01.sipott.v50.videotron.com:5060”,response=”d50e282901b8ca6573a34
057c8c198d0”,nonce=”MTU1OdczMDg2MTE3N2FhNDMzN2Y3NzlkNjJjMmM3ZmQ1NjQ5NzQzZjZhMGFl”,cnonce=”003BBF04”,qop=auth,algor
ithm=MD5,nc=00000001
Content-Length: 0
```

Received by PBX (Videotron server sends SIP Trying response)

```
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK79A42A86D;rport=5060
From: <sip:s143801234@cust01.sipott.v50.videotron.com>;tag=E91DABD0-2575
To: <sip:s143801234@cust01.sipott.v50.videotron.com>
Call-ID: 8F0D75A9-7D9B11E9-B6C0B188-A1E25E41
Timestamp: 1558730878
Cseq: 5 REGISTER
Content-Length: 0
```

Received by PBX (Videotron server indicates that registration is successful)

```
Received:
SIP/2.0 200 Registration Successful
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK79A42A86D;rport=5060
From: <sip:s143801234@cust01.sipott.v50.videotron.com>;tag=E91DABD0-2575
To: <sip:s143801234@cust01.sipott.v50.videotron.com>;tag=828935865
Call-ID: 8F0D75A9-7D9B11E9-B6C0B188-A1E25E41
Timestamp: 1558730878
Cseq: 5 REGISTER
Contact: <sip:s143801234@10.247.44.55:5060>;expires=45
User-Agent: Nortel SESM 18.0.31.0
Supported: com.nortelnetworks.firewall,p-3rdpartycontrol,nosec,join,x-nortel-sipvc,gin
Content-Length : 0
```

Appendix 2: Outbound call example: SIP INVITE request in the right format

Sent by PBX (PBX sends a SIP INVITE request including the called number, the domain [Req URI] and the caller's name and number [From])

```
Sent:
INVITE sip:5143725767@cust01.sipott.v50.videotron.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.247.44.55:5060;branch=z9hG4bK7B0A90B58
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>
Date: Wed, 29 May 2019 20:20:29 GMT
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
Supported: rel100,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0844016128-0000065536-0000000006-3114727178
User-Agent: Cisco-SIPGateway/IOS-15.5.3.M
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1559161229
Contact: <sip:4383870018@10.247.44.55:5060>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
P-Asserted-Identity: "Pascal CUCM" <sip:4383870018@10.247.44.55>
Session-Expires: 1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 271

v=0
o=CiscoSystemsSIP-GW-UserAgent 6602 8254 IN IP4 10.247.44.55
s=SIP Call
c=IN IP4 10.247.44.55
t=0 0
m=audio 29524 RTP/AVP 0 101 19
c=IN IP4 10.247.44.55
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:19 CN/8000
a=ptime:20
```

Received by PBX (Videotron server sends SIP Trying response)

```
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK7B0A90B58;rport=5060
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
CSeq: 101 INVITE
Timestamp: 1559161229
```

Received by PBX (Videotron server requests another SIP INVITE message with the username and password)

```
Received:
SIP/2.0 407 Proxy Authentication Required
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK7B0A90B58;rport=5060
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>;tag=1565155830
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
CSeq: 101 INVITE
Timestamp: 1559161229
User-Agent: Nortel SESM 18.0.31.0
Supported: com.nortelnetworks.firewall,p-3rdpartycontrol,nosec,join,x-nortel-sipvc,gin
Proxy-Authenticate: Digest
realm="Realm",nonce="MTU1OTE2MTIyMzk1NDAxMmMyMDY1NTU4MDZhZWIxNWIyMTRiZTA2NTRmMjQ1",stale=false,algorithm=MD5,qop="
auth"
Content-Length: 0
```

Sent by PBX (PBX responds that it understood the last request)

```
Sent:
ACK sip:5143725767@24.200.242.87:5060 SIP/2.0
Via: SIP/2.0/UDP 10.247.44.55:5060;branch=z9hG4bK7B0A90B58
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>;tag=1565155830
Date: Wed, 29 May 2019 20:20:29 GMT
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: telephone-event
Content-Length: 0
```

Sent by PBX (PBX sends another SIP Register request with the username, encrypted [hashed] password and domain)

```
Sent:
INVITE sip:5143725767@cust01.sipott.v50.videotron.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.247.44.55:5060;branch=z9hG4bK7B0A91C2C
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>
Date: Wed, 29 May 2019 20:20:29 GMT
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
Supported: rel100,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0844016128-0000065536-0000000006-3114727178
User-Agent: Cisco-SIPGateway/IOS-15.5.3.M
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 102 INVITE
Timestamp: 1559161229
Contact: <sip:4383870018@10.247.44.55:5060>
Expires: 180
Allow-Events: telephone-event
Proxy-Authorization: Digest
username="s383870018",realm="Realm",uri="sip:5143725767@24.200.242.87:5060",response="560b9f3f8bea30445fcf4f61f
8a62c83",nonce="MTU1OTE2MTIyMzk1NDAxMmMyMDY1NTU4MDZhZWIxNWIyMTRiZTA2NTRmMjQ1",cnonce="8C55BCEC",qop=auth,algorithm
=MD5,nc=00000001
Max-Forwards: 68
P-Asserted-Identity: "Pascal CUCM" <sip:4383870018@10.247.44.55>
Session-Expires: 1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 271

v=0
o=CiscoSystemsSIP-GW-UserAgent 6602 8254 IN IP4 10.247.44.55
s=SIP Call
c=IN IP4 10.247.44.55
t=0 0
m=audio 29524 RTP/AVP 0 101 19
c=IN IP4 10.247.44.55
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:19 CN/8000
a=ptime:20
```

Received by PBX (Videotron server sends SIP Trying response)

```
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK7B0A91C2C;rport=5060
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
CSeq: 102 INVITE
Timestamp: 1559161229
```

Received by PBX (Videotron server sends SIP Ringing response)

```
Received:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK7B0A91C2C;rport=5060
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>;tag=71964
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
CSeq: 102 INVITE
Timestamp: 1559161229
Contact: <sip:5143725767@24.200.242.87:5060;transport=udp>
User-Agent: Nortel SESM 18.0.31.0
Supported: replaces,tdialog,100rel
Allow: INVITE,BYE,CANCEL,ACK,REGISTER,SUBSCRIBE,NOTIFY,UPDATE,MESSAGE,INFO,REFER,OPTIONS,PUBLISH,PRACK
Content-Length: 0
```

Received by PBX (Videotron server sends 2nd SIP Ringing response)

```
Received:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK7B0A91C2C;rport=5060
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>;tag=92129
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
CSeq: 102 INVITE
Timestamp: 1559161229
Contact: <sip:5143725767@24.200.242.87:5060;transport=udp>
User-Agent: Nortel SESM 18.0.31.0
Supported: com.nortelnetworks.firewall,p-3rdpartycontrol,nosec,join,x-nortel-sipvc,gin
Allow: INVITE,BYE,CANCEL,ACK,REGISTER,SUBSCRIBE,NOTIFY,UPDATE,MESSAGE,INFO,REFER,OPTIONS,PUBLISH,PRACK
Content-Length: 0
```

Received by PBX (Videotron server sends a 200 OK response for the INVITE and the call connects)

```
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.247.44.55:5060;received=24.201.245.130;branch=z9hG4bK7B0A91C2C;rport=5060
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>;tag=92129
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
CSeq: 102 INVITE
Timestamp: 1559161229
Content-Type: application/sdp
Contact: <sip:5143725767@24.200.242.87:5060;transport=udp>
User-Agent: Nortel SESM 18.0.31.0
Supported: replaces,tdialog
x-nt-party-id: 95143725767@v50.videotron.com/
Call-Info: <http://pm50.videotron.com:80/pa/direct/pictureServlet?user=95143725767@v50.videotron.com>;Purpose=icon
Allow: INVITE,BYE,CANCEL,ACK,REGISTER,SUBSCRIBE,NOTIFY,UPDATE,MESSAGE,INFO,REFER,OPTIONS,PUBLISH,PRACK
x-nt-location: 1263
Require: timer
x-nt-service: answering-party=95143725767@v50.videotron.com
Session-Expires: 1800;refresher=uac
Content-Length: 212

v=0
o=- 1694963124 3 IN IP4 24.200.242.87
s=-
e=phxV2_95143725767@v50.videotron.com
c=IN IP4 24.200.242.87
t=0 0
m=audio 43884 RTP/AVP 0 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Sent by PBX (PBX responds that it understood the 200 OK response)

```
Sent:
ACK sip:5143725767@24.200.242.87:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 10.247.44.55:5060;branch=z9hG4bK7B0A92120E
From: "Pascal CUCM" <sip:4383870018@10.247.44.55>;tag=2C44734-1FC
To: <sip:5143725767@24.200.242.87>;tag=92129
Date: Wed, 29 May 2019 20:20:29 GMT
Call-ID: 9570E29-818611E9-9324B188-A1E25E41@10.247.44.55
Max-Forwards: 70
CSeq: 102 ACK
Allow-Events: telephone-event
Content-Length: 0
```

Received by PBX (Videotron server sends BYE request to say that the call has been terminated on their end)

```
Received:
BYE sip:00085143725767@10.247.166.178:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.247.166.185:5060;branch=z9hG4bK10a56575e1623
From: "Pascal CUCM" <sip:4383870018@10.247.166.185>;tag=139809~a44183b7-fd80-49c0-8d0f-ee3b6a95cfb9-26634415
To: <sip:00085143725767@10.247.166.178>;tag=2C449DC-228F
Date: Wed, 29 May 2019 20:20:29 GMT
Call-ID: 324eaa00-cee1e98d-10a14-b9a6f70a@10.247.166.185
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
P-Asserted-Identity: "Pascal CUCM" <sip:4383870018@10.247.166.185>
CSeq: 102 BYE
Reason: Q.850;cause=16
Content-Length: 0
```

Sent by PBX (PBX responds that it understood the BYE request)

```
May 29 16:20:35.545 EDT: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.247.166.185:5060;branch=z9hG4bK10a56575e1623
From: "Pascal CUCM" <sip:4383870018@10.247.166.185>;tag=139809~a44183b7-fd80-49c0-8d0f-ee3b6a95cfb9-26634415
To: <sip:00085143725767@10.247.166.178>;tag=2C449DC-228F
Date: Wed, 29 May 2019 20:20:35 GMT
Call-ID: 324eaa00-cee1e98d-10a14-b9a6f70a@10.247.166.185
Server: Cisco-SIPGateway/IOS-15.5.3.M
CSeq: 102 BYE
Reason: Q.850;cause=16
P-RTP-Stat: PS=79,OS=12640,PR=80,OR=12800,PL=0,JI=0,LA=0,DU=1
Content-Length: 0
```

Appendix 3: Private outbound call example: SIP INVITE request in the right format

Sent by PBX (PBX sends the Privacy: ID header and the P-Asserted-Identity header that contains the caller's real number)

```
Sent:
INVITE sip:5143725767@cust01.sipott.v50.videotron.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.247.44.55:5060;branch=z9hG4bK7B0C9B1ADF
From: <sip:anonymous@anonymous.invalid>;tag=2E93D10-888
To: <sip:5143725767@24.200.242.87>
Date: Wed, 29 May 2019 21:00:51 GMT
Call-ID: AD1C407C-818B11E9-95DEB188-A1E25E41@10.247.44.55
Supported: rel100,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 3589179648-0000065536-0000000008-3114727178
User-Agent: Cisco-SIPGateway/IOS-15.5.3.M
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1559163651
Contact: <sip:anonymous@10.247.44.55:5060>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
P-Asserted-Identity: "Pascal CUCM" <sip:4383870018@10.247.44.55>
Privacy: id
Session-Expires: 1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 271

v=0
o=CiscoSystemsSIP-GW-UserAgent 1878 4176 IN IP4 10.247.44.55
s=SIP Call
c=IN IP4 10.247.44.55
t=0 0
m=audio 29532 RTP/AVP 0 101 19
c=IN IP4 10.247.44.55
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:19 CN/8000
a=ptime:20
```

Note:

The “From” header contains the caller’s name and the “anonymous” caller’s number. While this information may seem sufficient for the call to be considered “private,” the Privacy: ID header is necessary for the Videotron switch. In this example, if the Privacy: ID header is missing from the SIP INVITE message, the Videotron switch will replace the “anonymous” caller’s number with the customer’s primary number, and the call will not be received privately on the PSTN side.