Avaya Solution & Interoperability Test Lab

Application notes for Bell Canada Communication System with Avaya™ Communication Server 1000 release 5.5

Abstract

These Application Notes describe a solution comprised of Avaya™ Communication Server 1000E Release 5.5 and Bell Canada SIP Trunking Service. The Primary focus of testing is the system verification of SIP trunk interoperability which includes the call scenario such as basic call, call forward (all calls, busy, no answer), call transfer (blind and consult) and conference. Calls should be placed in both directions and should involve various set types

Information in these Application Notes has been obtained through testing by Bell Canada at Bell Canada facilities.

Contents

1. Introduction

This document provides a typical network deployment of Communication Server 1000 (CS1000) utilizing the Bell Canada SIP Trunking Service. This document should serve as general guideline only, since it is not possible to document every possible variation of configuration. Further information may be obtained from your Bell or Avaya support representative.
The CS1000E system is configured as a SIP gateway endpoint on the Bell Canada Communication network. The enterprise customer will require an additional signaling server for each SIP gateway that will be deployed as SIP trunking to the carrier. In the Deployment Options section, the signaling server is shown as the onboard CP-PM option, but it can also be the outboard, rack-mounted 1U server.
The CS1000, in this configuration, does not use SIP Redirect or Proxy for Carrier SIP trunking, the SIP Virtual Gateway is simply provisioned with the SBC as the static SIP endpoint of the SIP Trunk..

1.1. Interoperability Compliance Testing

System verification testing of SIP Trunking between CS1000 Rel. 5.5 and the Bell SIP Trunking SBC:

- General call processing between systems including:
  - Codec/ptime negotiation and transcoding (G.711 and G.729 verification / 20ms)
  - Hold/Retrieve on both ends
  - CLID displayed
  - Ringback tone
  - Speech path
  - Dialing plan support
  - Advanced features (Call on Mute, Call Park, Call Waiting, use Feature Access Code)
  - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- DTMF on both direction
- SIP Transport UDP
- Thru dialing via PBX Call Pilot
- Voice Mail Server (hosted on Nortel system)
- Early Media Transmission
- Inter-office tandem Call
- FAX using Modem Pass Through (T38 not supported)

1.2. Support

For technical support on the Bell Canada system, please contact Bell technical support through the www.bell.ca/enterprise web site.

2. Equipment and Software Validated

The following equipment was used by Bell Canada during certification testing:

Call Server: 5.50J
Signaling Server: SSE 5.50.12
CS1000 Deplist: Please follow the Avaya Procedures when applying Customer deplist.

Patch ID	Issue	Title
MPLR25946	1	PI: SIP Line on CS1000 5.5: Remove MCDN from outgoing INVITE
MPLR29393	1	Provides provisioning of ingress and egress domain names on the SIP gateway Server
MPLR23632	1	Null values should be allowed for Public E.164/National or Subscriber fields on Element Manager
MPLR25529	3	PI: SIP: Partial support of DIVERSION
MPLR27408	1	PI: SIP: Disable SIP Session Timer on CS1K
MPLR28415	1	Ringback tone and speech path support in slow start CFNA scenarios

Notes:

1. MPLR25529 issue 3 may causes conflicts with MPLR25757 issue 1. Result: MPLR25757 is limited (LTD) patch and should not be in the system/dedicated GW.
2. MPLR27408 issue 1 may causes conflicts with MPLR25993 issue 1. Result: MPLR25993 is not required on the dedicated SIP GW as subscribe is not used. It can be safely removed.
3. This list of patches is for the Bell Sip Trunking Dedicated Sig Server ONLY. Any other non-carrier trunking gateways are to be patched with the DEPLIST.

Additional software and patch lineup for the configuration is as follows:

Tested equipment and software:

	System	**Software/Loadware Version**
CPE	Nortel CS1000E 5.5 (CPPM)	Call Server: 5.50J Signaling Server: 5.50.12
CPE	Nortel IP phone	2002 p2: 0604DCG 2004 p2: 0604DCG 1140: 0625C6J 1120: 06246J 2007: 0621C6H 1220: 062AC6J

3. Deployment Options

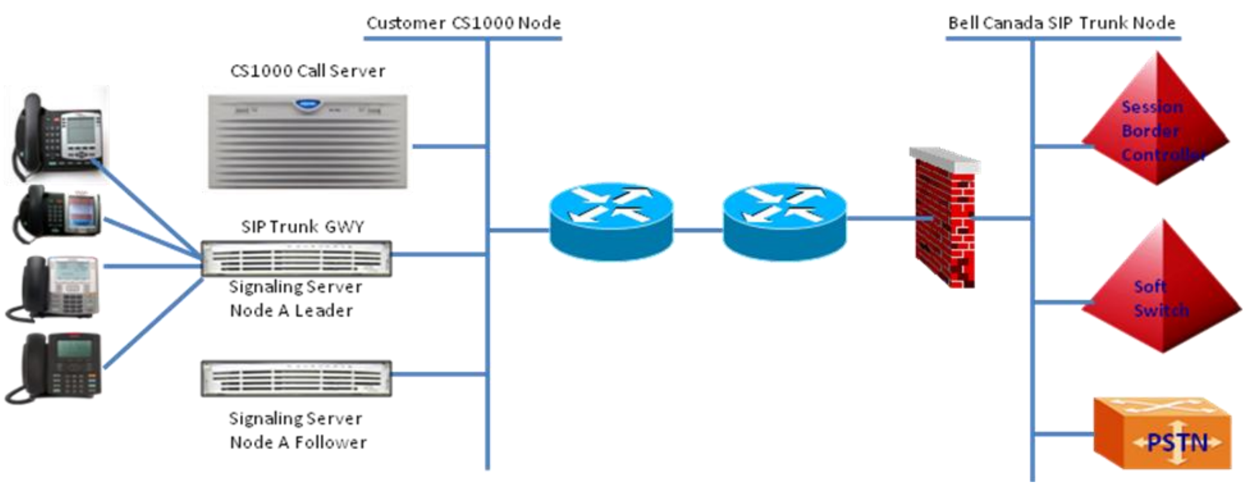
3.1. Deployment Option 1

There are a number of possible engineering options to consider when deploying SIP based trunks. The two main considerations that affect hardware and patching are redundancy and Private Networking requirements.

Avaya's CS1000 array of Signaling servers can be used for the SIP Trunk Gateway examples; (ISP1100 , COTS and CP-PM).

Figure 1 below depicts the simplest deployment model, utilizing one or more Signaling Servers to provide Telephone Proxy and SIP GW capability. The single Node configuration would have TPS and SIP GW applications enabled for all servers in the Node. The engineering rules for co-resident TPS and SIP GW would apply. Patching on the Signaling Servers would consist of the current DEPLIST plus the required patches to support Bell Canada SIP Trunk Service.

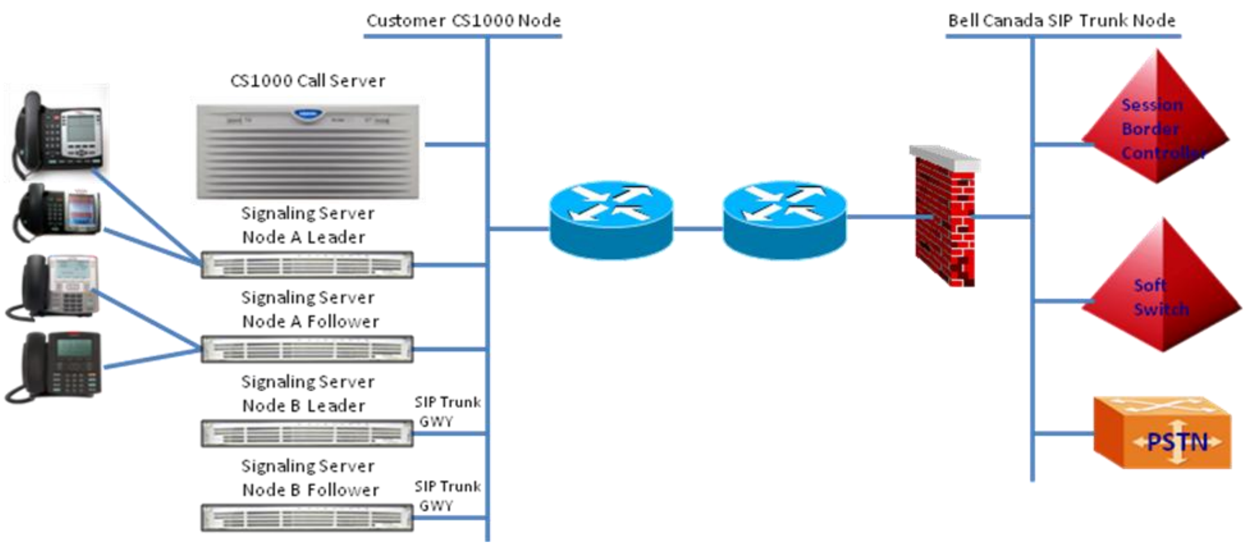
FIGURE 1: DEPLOYMENT OPTIONS 1



3.2. Deployment Option 2

In environments that will experience call rates that would exceed the co-resident TPS/SIP GW engineering recommendations, the TPS and SIP GW functionality must be separated into different Nodes. See Figure 2 as an example where Node A is configured for TPS and is patched with the current DEPLIST. Node B is configured with SIP GW only and is patched with the current DEPLIST plus the patches required to support Bell Canada SIP Trunk service.

FIGURE 2: DEPLOYMENT OPTION 2



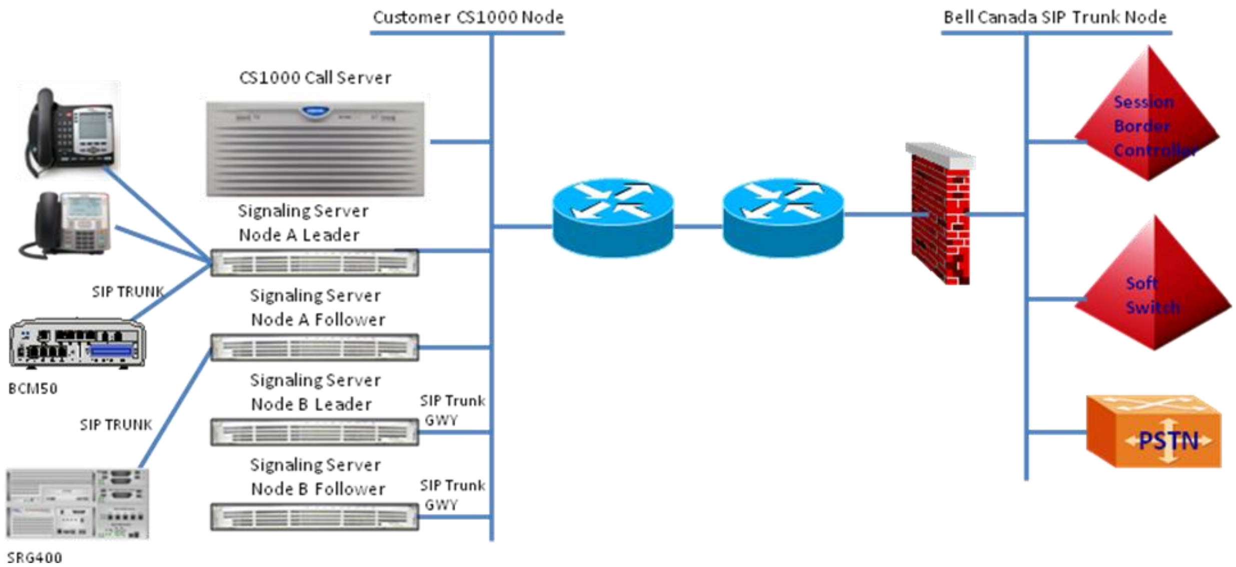
3.3. Deployment Option 3

The deployment options become more complex when additional Call Servers or SIP Applications are introduced behind the CS1K. such as SIP, H323, NRS, SRG etc.

See Figure 3 for an example utilizing SIP Private Networking to a BCM and a SRG.

Node A is configured for NRS, TPS and SIP GW, and patched with the current DEPLIST. Node B is configured with SIP GW only and is patched with the current DEPLIST plus the patches required to support Bell Canada SIP Trunks Service.

FIGURE 3: DEPLOYMENT OPTIONS 3



4. System Test Information

4.1. Tested Equipment

Bell service is compatible with CS1000E on Rel. 5.50 J and above.
The following list summarizes the version and patch levels of the hardware that has been validated as compatible in the Bell lab:

FIGURE 4: AVAYA CS1000E CS (CALL SERVER CP-PM)

CS VERSION CP PM	4021
System type is	Communication Server 1000E/CP P
CP PM - Pentium M 1.4 GHz	
IPMGs Registered:	1
IPMGs Unregistered:	0
IPMGs Configured/unregistered:	0
RELEASE	5
ISSUE	50 J +

FIGURE 5: AVAYA CS1000E SS ISP1100 (NRS\TPS\EM)

SS - ELAN IP address: NRS\TPS	**XXX.XXX.XXX.XXX**
Hostname:	MyHostname1
Type:	ISP1100
H323 ID:	Toronto
Software version:	sse-5.50.12
Role:	Leader
Element manager:	Equipped
Line TPS (UNIStim):	Equipped
IP peer gateway (virtual trunk TPS):	Equipped
SIP proxy/redirect server:	Enabled
SIP gateway:	Enabled
Gatekeeper configuration:	Primary
Gatekeeper configuration:	Off

FIGURE 6: AVAYA CS1000E SS ISP1100 (SIP TRUNK GATEWAY)

SS - ELAN IP address: SIP-GWY	**XXX.XXX.XXX.XXX**
Hostname:	MyHostname2
Type:	ISP1100
H323 ID:	pbx.customer.ca
Software version:	sse-5.50.12
Role:	Leader
Element manager:	Equipped
Line TPS (UNIStim):	Unequipped
IP peer gateway (virtual trunk TPS):	Equipped
SIP proxy/redirect server:	Enabled
SIP gateway:	Enabled
Gatekeeper configuration:	Off

4.2. SIP Trunking Dial Plan Overview

The SIP Trunking service will accept call requests with destinations according to the North American Number Plan (NANP: http://en.wikipedia.org/wiki/North_American_Numbering_Plan) If the service receives a request for a destination number that is not in service or has a malformed number, a proper announcement treatment will be applied.

** Please note that the service will accept 10 and 11 digit call requests for local and long distance calls. Depending on the purchased rate centers and method of trunk selection, long distance charges may apply. The following list shows a non-exhaustive dial plan example we used in our testing

- **10 Digits and 11 Digits for Local and Long Distance Call (long distance charge may apply)**
- **0+10D Call (operator assistant NA LD Call)**
- **01+International NDC (from 8 to 35 digits, Operator Assistant International Call)**
- **Toll Free Call 1-800,1-888,1-877**
- **011+ International Call**
- **101+xxxx+NDC call(from 13 to 40 digits, Casual Dial Call)**
- **911**
- **411, 611, 711, 211, 311, 511, 811**
- **1-xxx-555-1212**
- **310-xxxx Call**
- **1-900 and 1-976 Call**

5. Configure the Avaya Communication Server 1000E

5.1. Element Manager Configuration

Configure SIP in CS1000 network

The CS1000 IP Peer Networking Installation and Commissioning NTP NN43001-313 contains all configuration steps and screen shots basic configuration of SIP trunking. The details provided in the sections below build on this information, adding specific detail required for Bell SIP Trunking.

This section describes the steps for creating Node ID (1001) in CS 1000 network. Enter Element Manager through the IE browser (in IE address bar, type IP address of the Node IP or TLAN of Signaling Server).

Input Node ID and press add...

Enter TLAN, ELAN IP address of Signalling Server.

Node Configuration

+Node: 1 Node IP: 192.168.7.101

admin1	13de4348000000	context/voipsvr/r	TrfSingleNode	1	192.168.8.93	192.168.8.95

Telephony LAN (TLAN) IP address		TN
Signaling Server		
	192.168.7.47	
Voice Gateway Media Card		
	192.168.7.48	0 0 2 0

+Node: 1001 Node IP: 192.168.7.106

admin1	13de4348000000	context/voipsvr/r	TrfSingleNode	1001	192.168.8.96

Telephony LAN (TLAN) IP address		TN
Signaling Server		
	192.168.7.49	

The IP addresses here are customer assigned and provided and are related to their network only.

Node Configuration Main Menu

This menu is standard for all ISP1000 Servers, and each title has multiple drill down table entries.

+IP Telephony Node
+VGW and IP phone codec profile
+QoS
+LAN configuration
+SNTP
+Virtual Trunk Network Health Monitor configuration
+H323 GW Settings
+Firmware
+SIP GW Settings
+SIP URI Map
+SIP CD Services
+SIP CTI Services
+Microsoft Unified Messaging
+Cards
+Signaling Servers

IP Telephony Node

The IP address information displayed is entered during Installation and is not covered in this document. This is the IP address of the SIP Trunk Gateway and is translated to PBX domain name.

−IP Telephony Node	
Node ID	**1001**
Telephony LAN (TLAN) Node IP address	192.168.7.106 *
Embedded LAN (ELAN) gateway IP address	192.168.8.96
Embedded LAN (ELAN) subnet mask	255.255.255.0
Voice LAN (TLAN) subnet mask	255.255.255.0

Node IP address is used by the SBC, and must be supplied to Bell sip trunk service.

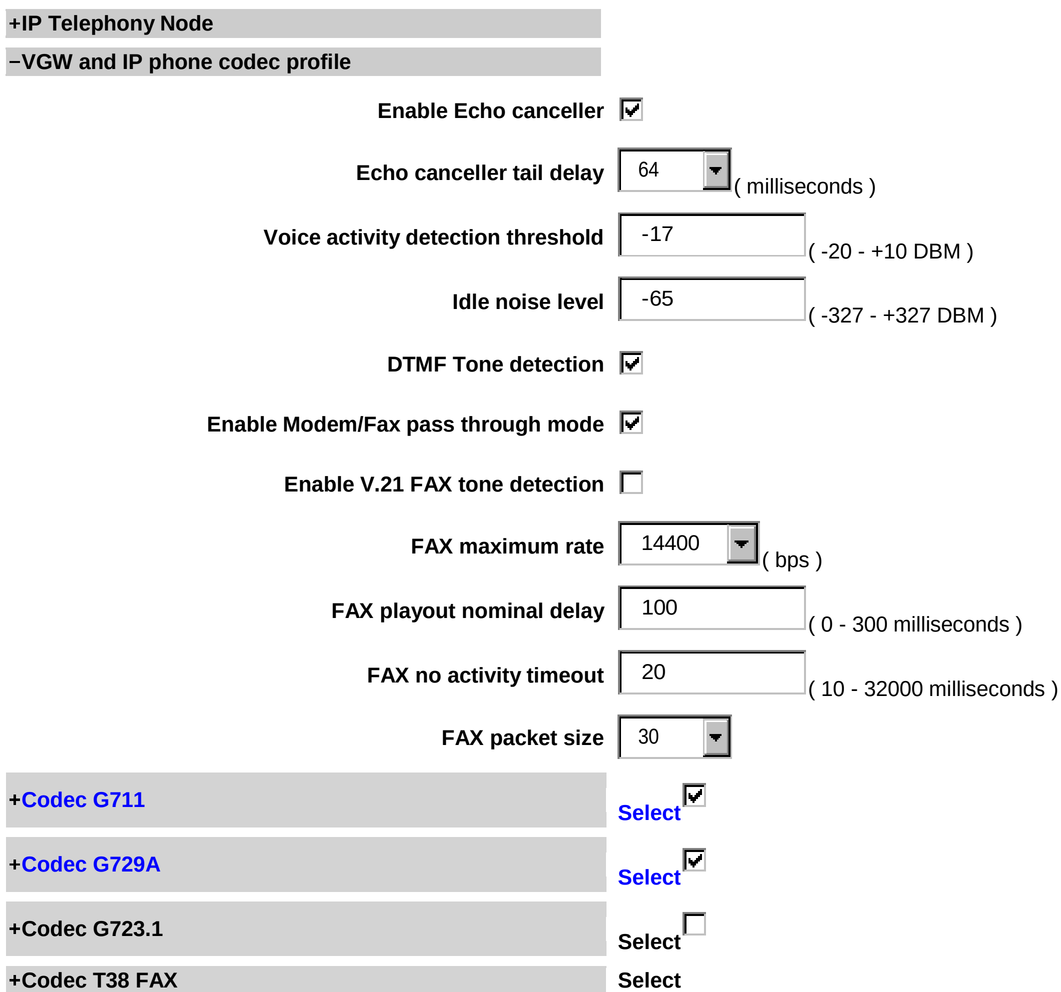
Voice Codec Settings

Configure Voice Codec for Nortel IP Phone

This section describes the steps for administering a set of codecs in CS1000. This set of codecs is used in IP network for communication between Nortel IP Phones.

Access EM by IE browser.
Choose "IP Network", then choose "Nodes: Servers, Media Cards", select proper Node and press "Edit".
To change Codec profile for IP Phone, select "VGW and IP phone codec profile".

The Codes of Choice used for voice during testing were G711 and G729.



Voice Codec Settings Cont.

+IP Telephony Node	
−VGW and IP phone codec profile	
Enable Echo canceller	[x]
Echo canceller tail delay	64 (milliseconds)
Voice activity detection threshold	-17 (-20 - +10 DBM)
Idle noise level	-65 (-327 - +327 DBM)
DTMF Tone detection	[x]
Enable Modem/Fax pass through mode	[x]
Enable V.21 FAX tone detection	[ ]
FAX maximum rate	14400 (bps)
FAX playout nominal delay	100 (0 - 300 milliseconds)
FAX no activity timeout	20 (10 - 32000 milliseconds)
FAX packet size	30
−Codec G711	Select [x]
Codec Name	G711
Voice payload size	20 (ms/frame)
Voice playout (jitter buffer) nominal delay	40
	Modifications may cause changes to maximal delay settings
Voice playout (jitter buffer) maximum delay	80
−Codec G729A	Select [x]
Codec Name	G729A
Voice payload size	20 (ms/frame)
Voice playout (jitter buffer) nominal delay	40
	Modifications may cause changes to maximal delay settings
Voice playout (jitter buffer) maximum delay	80
VAD	[ ]

−Codec G723.1	**Select** ☐
Codec Name	**G723.1**
Voice payload size	**30**
Voice playout (jitter buffer) nominal delay	60
	Modifications may cause changes to maximal delay settings
Voice playout (jitter buffer) maximum delay	120
−Codec T38 FAX	
Codec Name	

−QoS	
Diffserv Codepoint(DSCP) Control packets	40 (0 - 63)
Diffserv Codepoint(DSCP) Voice packets	46 (0 - 63)
Enable 802.1Q support	☐
802.1Q Bits value (802.1p)	6 (0 - 7)

Lan Configuration

The IP address information displayed is entered during Installation and is not covered in this document. The DNS server IP listed below were customer related for test purposes only.

−LAN configuration		
Embedded LAN (ELAN) configuration		
Call server IP address	192.168.8.91	
Unistem Signaling port	**15000**	
Broadcast port	15001	(1024 - 65535)
Telephony LAN (TLAN) configuration		
Unistem Signaling port	**5000**	
RTP/RTCP Starting port	5200	(1024 - 65535)
Embedded LAN (ELAN) Routes		
Host Table		
DNS Servers		
Primary DNS Server IP address		
Alternate DNS Server1 IP address		
Alternate DNS Server2 IP address	0.0.0.0	

SNTP Configuration

−SNTP	
SNTP Server	
Mode	active
Interval	256 (1 - 2147483647)
Port	21001
SNTP Client	
Mode	passive
Interval	256 (1 - 2147483647)
Port	21001
SNTP server IP address	0.0.0.0

VTRK Network Health Monitor

FIG 15: VTRK NETWORK MONITOR

−Virtual Trunk Network Health Monitor configuration	
Monitor	☑

Lan Configuration

The H323 settings were not related to our testing and therefore not assigned.

SIP Gateway Configuration

The Bell SIP Trunk service primary proxy IP and its protocol type are highlighted below.

–SIP GW Settings		
TLS Security		
Security Policy	Security Disabled	
TLS Security Port	5061	(1 - 65535)
Client Authentication	☐	
Re-negotiation	☐	
X.509 Certificate Authentication	☐	
Primary Proxy or Re-direct Server		
Primary Proxy or Redirect (TLAN) IP address	10.1.109.150	**(Bell Canada SIP Trunk Service)**
Port	5060	
Supports Registration	☐	
Primary CDS Proxy or Re-direct server flag	☐	
Transport Protocol	UDP	
Secondary Proxy or Re-direct Server		
Secondary Proxy or Redirect (TLAN) IP address	**10.1.109160 (Secondary SBC)**	
Port	5060	
Supports Registration	☐	
Secondary CDS Proxy or Re-direct server flag	☐	
Transport Protocol	UDP	
CLID Parameters		
Country Code (CCC)		
Area Code (AreaCode)	7611	*Note:The NPA in North America*

	# Digits to Strip	**Prefix to Insert**	**Format of CLID**
Subscriber Number (SN)	0		**+<CCC><AreaCode><SN>**
National Number (NN)	0		**+<CCC><NN>**
International number			**+<International number>**

SIP URI MAP Configuration

Configure SIP URI

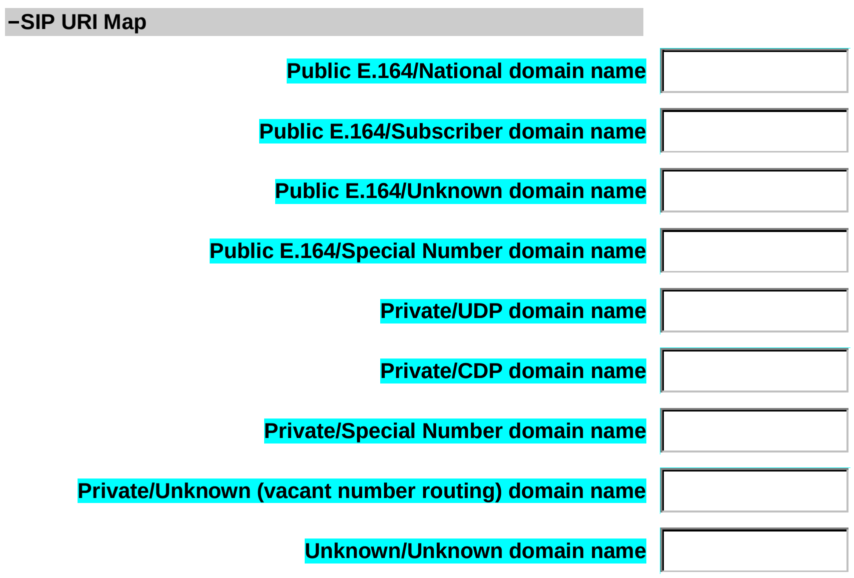
This section describes the steps for administering SIP URI configuration in CS1000.

Access EM by IE browser

Choose "IP Network", then choose "Nodes: Servers, Media Cards", select proper Node and press "Edit".

To change SIP URI, select "SIP URI Map".

The SIP URI fields must remain blank for E.164.



SIP Gateway Signaling Server Configuration

−Signaling Servers	
−Signaling Server 192.168.8.96 Properties	
Role	**Leader**
Type	**ISP1100**
Embedded LAN (ELAN) IP address	192.168.8.96 *
Embedded LAN (ELAN) MAC address	00:02:b3:f7:bb:d4 *
Telephony LAN (TLAN) IP address	192.168.7.49 *
Telephony LAN (TLAN) gateway IP address	192.168.7.1
Hostname	cs1ktrunk *
H323 ID	cs1ktrunk *
Enable Line TPS	☐
Enable IP Peer Gateway (Virtual Trunk TPS)	SIP only If Telephony LAN(TLAN) IP address and Telephony LAN(TLAN) gateway IP address are not in the same subnet as Telephony LAN(TLAN) Node IP address when Line TPS or IP Peer Gateway is enabled, then the TPS and/or VTRK applications will not run.
Enable SIP Proxy / Redirect Server	☑
Local SIP TCP/UDP Port to Listen to	5060
SIP Domain name	siptrunking.bell.ca (Bell Canada SIP Trunk Service)
SIP Gateway Endpoint Name	Pbx1.customer.com (CPE)
SIP Gateway Authentication Password	
Enable Gatekeeper	☐
Network Routing Service Role	Failsafe

In the signalling server properties, the Line TPS will not be enabled if this signalling server is dedicated for SIP Trunk Gateway. As the role of this server is SIP gateway only then this can be left unchecked.

The NRS is not enabled as all PSTN bound calls are routed by the SBC.

Zone Detail Configuration

Configure Zones and Bandwidth Management

This section describes the steps for administering Zone configuration in CS1000.
Access EM by IE browser: Choose "IP Network", then choose "Zones", select proper "Zone Basic Property and Bandwidth Management"
Please follow all documented Bandwidth Engineering rules for the CS1000, When applying the Sip trunk service. The existing bandwidth strategy can be adjusted to preference.
If it does not already exist, create a zone for IP sets.
Create a new zone for the SIP trunks.

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	0
Intrazone Bandwidth (INTRA_BW):	10000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	10000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	1
Intrazone Bandwidth (INTRA_BW):	1000000
Intrazone Strategy (INTRA_STGY):	Best Bandwidth (BB)
Interzone Bandwidth (INTER_BW):	1000000
Interzone Strategy (INTER_STGY):	Best Bandwidth (BB)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	ZONE1

Zone Detail Configuration Cont.

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	20
Intrazone Bandwidth (INTRA_BW):	1000000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	MGCIPMG

SIP Routes and Trunks

Configure SIP trunk

This section describes the steps for establishing a SIP connection between CS 1000 switch and Carrier system.

Routes and Trunks

– Customer: 0	Total routes: 4	Total trunks: 23
+Route: 20	Type: MUS	Description: DESMUS
+Route: 60	Type: TIE	Description: SIPROUTE
+Route: 61	Type: TIE	Description: H323TRUNK
+Route: 62	Type: TIE	Description: NRS H323 TRUNK

SIP Trunk Configuration

Trunks

–	**Customer: 0**	Total routes: 4	Total trunks: 23
	–Route: 60	Type: TIE	Description: SIPROUTE
	–Trunk: 1 - 12	Total trunks: 12	
	–Trunk: 1	TN: 108 0 02 01	Description: SIPTRUNK
	–Trunk: 2	TN: 108 0 02 02	Description: SIPTRUNK
	–Trunk: 3	TN: 108 0 02 03	Description: SIPTRUNK
	–Trunk: 4	TN: 108 0 02 04	Description: SIPTRUNK
	–Trunk: 5	TN: 108 0 02 05	Description: SIPTRUNK
	–Trunk: 6	TN: 108 0 02 06	Description: SIPTRUNK
	–Trunk: 7	TN: 108 0 02 07	Description: SIPTRUNK
	–Trunk: 8	TN: 108 0 02 08	Description: SIPTRUNK
	–Trunk: 9	TN: 108 0 02 09	Description: SIPTRUNK
	–Trunk: 10	TN: 108 0 02 10	Description: SIPTRUNK
	–Trunk: 11	TN: 108 0 02 11	Description: SIPTRUNK
	–Trunk: 12	TN: 108 0 02 12	Description: SIPTRUNK

SIP Route Configuration

Customer 0, Route 60 Property Configuration

−Basic Configuration

Input Description	Input Value
Route Data Block (RDB) (TYPE)	RDB
Customer number (CUST)	00
Route Number (ROUT)	60
Designator field for trunk (DES)	SIPROUTE
Trunk Type (TKTP)	TIE
Incoming and Outgoing trunk (ICOG)	Incoming and Outgoing (IAO)
Access Code for the trunk route (ACOD)	4000 *
Trunk type M911P (M911P)	☐
The route is for a virtual trunk route (VTRK)	☑
- Zone for codec selection and bandwidth management (ZONE)	001 Range: 0 - 255
- Node ID of signaling server of this route (NODE)	1001 Range: 0 - 9999
- Protocol ID for the route (PCID)	SIP (SIP)
- Print Correlation ID in CDR for the route (CRID)	☐
Integrated Services Digital Network option (ISDN)	☑
- Mode of operation (MODE)	Route uses ISDN Signaling Link (ISLD)
- D channel number (DCH)	12 Range: 0 - 254
- Interface type for route (IFC)	Meridian M1 (SL1)
- Private Network Identifier (PNI)	00001 Range: 0 - 32700
- Network Calling Name Allowed (NCNA)	☑
- Network Call Redirection (NCRD)	☑
- - Trunk Route Optimization (TRO)	☐
- Recognition of DTI2 ABCD FALT signal for ISL (FALT)	☐
- Channel Type (CHTY)	B-channel (BCH)
- Call Type for outgoing direct dialed TIE route (CTYP)	Unknow n Call type (UKWN)

- Insert ESN Access Code (INAC) ☐

- Integrated Service Access Route (ISAR) ☐

- Display of Access Prefix on CLID (DAPC) ☐

- Mobile Extension Route (MBXR) ☐

+Basic Route Options

+Network Options

+General Options

+Advanced Configurations

SIP Route Configuration- Basic Route Options

−Basic Route Options

Input Description	Input Value
Billing Number Required (BILN)	☐
Call Detail Recording (CDR)	☐
Controls or timers (CNTL)	☐
Conventional (Tie trunk only) (CNVT)	☐
Incoming DID Digit Conversion on this route (IDC)	☑
- Day IDC tree number (DCNO)	0 Range: 0 - 254
- Night IDC tree number (NDNO)	0 Range: 0 - 254
- Display External dialed digits (DEXT)	☐
MFC feature options (MFC_FEAT)	☐

SIP Route Configuration- Network Options

−Network Options

Input Description	Input Value
Electronic Switched Network pad control (ESN)	☐
Signaling arrangement (SIGO)	Standard (STD)
Route Class (RCLS)	Route Class marked as external (EXT)
Off-Hook Queuing (OHQ)	☐
Off-Hook Queue Threshold (OHQT)	0
Call Back Queuing (CBQ)	☐
Number of Digits (NDIG)	2
Authcode (AUTH)	☐

SIP Route Configuration- General Options

−General Options

Input Description	Input Value
M1 is the only Controlling Party on incoming calls (CPDC)	☐
Dial Tone on originating calls (DLTN)	☐
Hold failure threshold (HOLD)	02 02 40
Trunk Access Restriction Group (TARG)	01
Alternate trunk route for outgoing trunks (STEP)	Range: 0 - 511
Actual outgoing toll digits to be ignored for Code Restriction (OABS)	
Display IDC Name (DNAM)	☐
Enable Equal Access Restrictions (EQAR)	☐
ACD DNIS route (DNIS)	☐
Include DNIS number in CDR records (DCDR)	☐

SIP Route Configuration- Advanced Configurations

−Advanced Configurations

Input Description	Input Value
Allow last Re-directing Number (ARDN)	ARDN (NO)
ANI identifier number (ANTK)	
Auto terminate (AUTO)	☐
Maximum number of CNI digits (CLEN)	1
North American Distinctive Ringing for incoming calls (DRNG)	☐
Home Local Number (HLCL)	
Home National Number (HNTN)	
In-Band Automatic Number Identification route (IANI)	☐
Internal/external definition (IDEF)	Use network info (NET)
Insert (INST)	

Manual Outgoing trunk route (MANO)	☐
Manual Route (MNL)	☐
Music On-Hold (MUS)	☐
North American Toll scheme (NATL)	☑
Off-Hook Timer Delay (OHTD)	☐
Protocol Selection (PSEL)	DM-DM Protocol Selection (DMDM)
Port Type at far end (PTYP)	Analog TIE trunks (ATT)
Route traffic information in ACD Reports (RACD)	☐
Route Number (RTN)	Range: 0 - 511
Satellite used for trunk route (SAT)	☐
Scheduled Access Restriction Group (SGRP)	0 Range: 0 - 999
Special Service List number (SSL)	
Standard Signaling Type (STYP)	Standard Data (SDAT)
CPP/CPPO flag for incoming non-ISDN trunk call tandemed to this trunk route (TCPP)	☐
Tone Detector required (TDET)	☐
Tromboning (TRMB)	☐
Tone Table number (TTBL)	0
Answer an Attendant Extended Call over VNS immediately on the incoming bearer trunk (VRAT)	☐

Sip Trunk Channel Member Configuration

Customer 0, Route 60, Trunk 1 Property Configuration

-Basic Configuration

Input Description	Input Value
Trunk data block (TYPE)	IPTI
Terminal Number (TN)	108 0 02 01
Designator field for trunk (DES)	SIPTRUNK
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	60 1 *
Level 3 Signaling (SIGL)	
Card Density (CDEN)	8D
Start arrangement Incoming (STRI)	Immediate (IMM)
Start arrangement Outgoing (STRO)	Immediate (IMM)
Trunk Group Access Restriction (TGAR)	1
Channel ID for this trunk. (CHID)	1
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	

+Advanced Trunk Configurations

Sip Trunk D-Channel Configuration

D-Channels 12 Property Configuration

Configuration

Choose a D-Channel Number: 0 and type: DCH

–	**Channel: 12**	Type: DCH	Card Type: DCIP	Description: PSTNSIPTRUNK
–	**Channel: 13**	Type: DCH	Card Type: DCIP	Description: NRSH323TRUNK

Sip Trunk D-Channel Detail Configuration

–Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	PSTNSIPTRUNK
Recovery to Primary (RCVP)	☐
PRI loop number for Backup D-channel (BCHL)	
User (USR)	Integrated Services Signaling Link Dedicated (ISLD) *
Interface type for D-channel (IFC)	Meridian Meridian1 (SL1)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	
Secondary PRI2 loops (PRI2)	
Meridian 1 node type (SIDE)	Slave to the controller (USR)
Release ID of the switch at the far end (RLS)	5
Central Office switch type (CO_TYPE)	100% compatible w ith Bellcore standard (STD)
Integrated Services Signaling Link Maximum (ISLM)	4000 Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1800 Range: 0 - 4000

+Basic options (BSCOPT)

+Advanced options (ADVOPT)

+Feature Packages

Sip Trunk D-Channel Detail Configuration-Basic options (BSCOPT)

Basic options (BSCOPT)

Primary D-channel for a backup DCH (PDCH)	Range: 0 - 254
- PINX customer number (PINX_CUST)	
- Progress signal (PROG)	
- Calling Line Identification (CLID)	
- Output request Buffers (OTBF)	32
- D-channel transmission Rate (DRAT)	56 kb/s w hen LCMT is AMI (56K)
- Channel Negotiation option (CNEG)	No alternative acceptable, exclusive. (1)
- Remote Capabilities (RCAP)	

Sip Trunk D-Channel Detail Configuration- Remote Capabilities Configuration

- Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	☐
Call completion on busy using integer value (CCBI)	☐
Call completion on busy using object identifier (CCBO)	☐
Call completion on busy for QSIG and EuroISDN BRI (CCBS)	☐
Call completion on no response using integer value (CCNI)	☐
Call completion on no response using object identifier (CCNO)	☐
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)	☐
Network call park (CPK)	☐
Connected line identification presentation (COLP)	☐
Call transfer integer (CTI)	☐
Call transfer object (CTO)	☐
Diversion info. is sent using integer value (DV1I)	☐
Diversion info. is sent using object identifier (DV1O)	☐
Rerouting requests processed using integer value (DV2I)	☐
Rerouting requests processed using object identifier (DV2O)	☐
Diversion info. sent. rerouting requests processed (DV3I)	☐
EuroISDN - div. info sent. rerouting req. processed (DV3O)	☐
Call transfer notification and invocation to EuroISDN (ECTO)	☐
Malicious call identification (MCID)	☐
MCDN QSIG conversion (MQC)	☐
Remote D-channel is on a MSDL card (MSL)	☐
Message waiting interworking with DMS-100 (MWI)	☐
Network access data (NAC)	☐
Network call trace supported (NCT)	☐
Network name display method 1 (ND1)	☐
Network name display method 2 (ND2)	☑
Network name display method 3 (ND3)	☐

Name display - integer ID coding (NDI) ☐

Name display - object ID coding (NDO) ☐

Path replacement uses integer values (PRI) ☐

Path replacement uses object identifier (PRO) ☐

Release Link Trunks over IP (RLTI) ☑

Remote virtual queuing (RVQ) ☐

Trunk anti-tromboning operation (TAT) ☑

User to user service 1 (UUS1) ☐

NI-2 name display option. (NDS) ☐

Message waiting indication using integer values (QMWI) ☐

Message waiting indication using object identifier (QMWO) ☐

User to user signalling (UUI) ☐

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.