



Avaya Solution & Interoperability Test Lab

Application Notes for Bell Canada SIP Trunking Service with Avaya Communication Server 1000 Release 7.5, Avaya Aura ® Session Manager Release 6.1 and Acme Packet Session Border Controller Release 6.2 - Issue 1.0

Abstract

These Application Notes describe steps to configure Session Initiation Protocol (SIP) Trunking between Bell Canada SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Communication Server 1000 Release 7.5, Avaya Aura ® Session Manager Release 6.1, Acme Packet Session Border Controller Release 6.2 and various Avaya endpoints. This documented solution does not extend to configurations without Avaya Aura ® Session Manager or Acme Packet Session Border Controller.

Bell Canada SIP Trunking Service provides PSTN access via a SIP Trunk between enterprise and Bell Canada networks as an alternative to traditional PSTN trunks such as analog or ISDN-PRI. This approach generally results in lower cost for the enterprise.

Bell Canada is a member of the Avaya DevConnect Service Provider Program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing is conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe steps to configure Session Initiation Protocol (SIP) Trunking between Bell Canada SIP Trunking Service (Bell Canada) and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Communication Server 1000 (CS1000) Release 7.5, Avaya Aura® Session Manager (Session Manager) Release 6.1, Acme Packet Session Border Controller (Acme SBC) Release 6.2 and various Avaya endpoints.

Bell Canada SIP Trunking Service referenced within these Application Notes is designed for enterprise business customers. Customers using Bell Canada SIP Trunking Service with the Avaya SIP-enabled enterprise solution are able to place and receive PSTN calls via a broadband WAN connection using SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog or ISDN-PRI.

Bell Canada applies Digest Authentication to outgoing calls from the enterprise. It uses challenge-response authentication with a “401 Unauthorized” responding to each initial outgoing INVITE to Bell Canada. The subsequent INVITE from the enterprise provides the “Authorization” header with a configured user name and password. This credential is provided by Bell Canada and configured on Acme SBC. This call authentication scheme as specified in RFC 3261 provides authentication for SIP signaling.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Bell Canada is a member of the Avaya DevConnect Service Provider Program. The general test approach is to connect a simulated enterprise to Bell Canada via public internet and exercise features and functionalities listed in **Section 2.1**.

2.1. Interoperability Compliance Testing

To verify Bell Canada SIP Trunking interoperability, following features and functionalities were covered during the compliance testing:

- Incoming PSTN calls to various phone types including UNISTim, SIP, digital and analog telephones at the enterprise. All incoming calls from PSTN are routed to the enterprise across the SIP Trunk from service provider.
- Outgoing PSTN calls from various phone types including UNISTim, SIP, digital and analog telephones at the enterprise. All outgoing calls to PSTN are routed from the enterprise across the SIP Trunk to service provider.
- Incoming and outgoing PSTN calls to/from 2050PC softphone.
- Incoming and outgoing PSTN calls to/from Avaya one-X® Communicator (SIP) softphone for CS1000.
- Dialing plans including local, long distance, international, outgoing toll-free, operator assisted calls, local directory assistance (411) calls, etc.
- Calling Party Name presentation and Calling Party Name restriction.
- Proper codec negotiation using G.711MU codec.
- Proper early media transmission using G.711MU codec.
- Proper media transmission using G.711MU codec.
- Incoming and outgoing fax over IP using G.711MU codec.
- DTMF tone transmission as out-of-band RTP events per RFC 2833.
- Voicemail navigation for incoming and outgoing calls.
- Call Pilot voicemail system hosted on CS1000.
- Telephony features such as Hold and Resume, Call Waiting, Call Park, Call Transfer, Call Forward, and Conferencing.
- Music on Hold.
- Off-net call transfer using subsequent INVITE method.
- Off-net call forward using Diversion method.
- Mobility Extension (MobX) twining incoming call to cellular using Diversion method.
- Response to incomplete call attempts and trunk errors.
- Response to OPTIONS heartbeat.
- SIP Digest Authentication.

- Session Timers implementation.

Items that were not supported by Bell Canada on the test environment or not tested as part of the compliance testing, are listed as following:

- Inbound toll-free and outgoing emergency calls (E911) are supported but were not tested as part of the compliance testing because Bell Canada has not provided the necessary configuration.
- G.729 codec is not supported.
- Fax over IP with T.38 codec is not supported.
- Off-net calls transfer using REFER method is not supported.
- Off-net call forward using History-Info method is not supported.

2.2. Test Results

Interoperability testing of Bell Canada SIP Trunking Service with the Avaya SIP-enabled enterprise solution is completed with successful results for all test cases with the exception of the observations/limitations described below.

1. **Incoming call scenario to retrieve voicemail on Call Pilot is now corrected and working properly.** An incoming call to Call Pilot was unexpectedly disconnected by Bell Canada with BYE message responding to 200OK without Session Description Protocol (SDP) from CS1000. According to RFC 3262, this 200OK is valid because the signaling from Bell Canada contains the “Supported: 100rel” header to indicate that Reliable of Provisional Response is supported. A workaround has been implemented to disable Reliable of Provisional Response on the SIP Trunk by provisioning a Header Manipulation Rule (HMR) on Acme SBC to delete the “Supported: 100rel” signaling from both inbound and outbound traffics. By removing the “Supported: 100rel” signaling, it forces CS1000 to send a SDP along with 200OK. This allows the call to establish successfully. For detailed configuration, refer to **Section 7.7**.
2. **For off-net blind transfer, the calling PSTN party does not hear ringback when the called PSTN party is ringing.** This limitation was encountered due to the workaround from the observation #1 above. Without the “Supported: 100rel” signaling which is for Reliable of Provisional Response (RFC3262) support, UPDATE/SDP from CS1000 to complete an off-net call transfer was rejected by Bell Canada with “500 Internal Server Error”. The call failed to transfer. To correct this scenario, patch MPLR30224 needs to be installed on CS1000 SIP Signaling Gateway (SSG). With the patch in-service, CS1000 uses subsequent INVITE method to complete the transferred call instead of using UPDATE/SDP. The call successfully transferred, however, it was then observed that the calling PSTN did not hear ringback when the called PSTN was ringing. This is a known limitation of the patch with no resolution available at this time.
3. **For off-net call transfer, Calling Party Name and Number are not updated to PSTN parties.** When CS1000 transferred off-net an incoming call back to PSTN, it did not update true connected Calling Party Name and Number to PTSN parties. This results both PSTN parties still display Calling Party Name and Number of CS1000 extension. This is

a known issue of CS1000 when it interoperates with Bell Canada where proprietary signaling of CS1000 is not supported. This issue has low user impact and therefore listed here simply as an observation.

4. **CS1000 SIP phone fails to transfer off-net back to PSTN if “Music On Hold” is enabled.** When “Music On Hold” is enabled, CS1000 SIP phone fails to transfer off-net an incoming or outgoing call between CS1000 SIP phone and PSTN_1 back to PSTN_2. PSTN_1 continues to hear ringback after the call has already been answered by PSTN_2. The same call scenario was successful when SIP phone is replaced by other endpoints e.g. UNISTim or digital phones. This issue has been reported to relevant Avaya team for investigation. It, however, is listed here as a limitation.
5. **When CS1000 UNISTim phone places an external PSTN call on hold then retrieves the held call, it causes Calling Party Number to change.** After retrieving the held external PSTN call, Calling Party Number previously displayed on CS1000 phone was replaced by “Route ACOD” – “Trunk Channel ID”. This is a known behavior of CS1000 with no resolution available at this time. This issue has low user impact and therefore listed here simply as an observation.
6. **When CS1000 SIP phone calls local UNISTim phone then blind transfers to PSTN, it causes Calling Party Number to change.** The call was successfully transferred; however, UNISTim phone displayed “Route ACOD” – “Trunk Channel ID” instead of displaying Calling Party Name and Number of PSTN party. This is a known behavior of CS1000 with no resolution available at this time. This issue has low user impact and therefore listed here simply as an observation.
7. **Cellular Voice Mail Avoidance function of Mobility Extension (MobX) feature is now corrected and working properly.** When an incoming call being answered by cellular voicemail, **Mobile extension timer (MBXT)** setting on the SIP Trunk as described in **Section 5.5.6**, cannot ignore the answering as expected. The call then was unexpectedly connected to cellular voicemail instead of connecting to enterprise voicemail (hosted by Call Pilot). This issue has been corrected by patch MPLR32246. With the patch in-service, the answering by cellular voicemail before MBXT is ignored, it allows the incoming call to route to Call Pilot.
8. **Off-net call forward unconditional or busy scenarios fail if the forwarded PSTN party takes longer than 8 seconds to respond.** The issue has been seen on cellular PSTN phone. CS1000 relies on the respond time of cellular PSTN phone on the 2nd call leg to transmit ringback back to the original calling PSTN party on the 1st call leg. If the respond time exceeds 8 seconds, Bell Canada terminates the call with CANCEL request on the 1st call leg. This failed the scenarios. The issue does not happen if cellular PSTN phone in good wireless coverage that can significantly reduce the respond time under 8 second limit. The same call scenarios are successful on regular PSTN phone (instead of cellular PSTN phone). Bell Canada is recommended to increase Session Provisioning Timer to correct the issue. This is acknowledged as a known behavior of Bell Canada SIP Trunking Service with no resolution available at this time. It is listed here as a limitation.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on Bell Canada SIP Trunking Service, contact Bell Canada at http://www.bell.ca/enterprise/EntPrd_SIP_Trunking.page.

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution connected to Bell Canada SIP Trunking Service (Vendor Validation Circuit) through the internet. For confidentiality and privacy purposes, actual public IP addresses and PSTN routable phone numbers used in the certification testing have been masked or replaced with fictitious parameters throughout the Application Notes.

Acme SBC is located at the edge of the enterprise network. It has two connection points, a public side connecting to Bell Canada via the internet and a private side connecting to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise network flow through Acme SBC which can protect the enterprise against any outside SIP-based attacks. Acme SBC provides Network Address Translation (NAT) at both IP and SIP layers. The transport protocol between Acme SBC and Bell Canada across the public network is UDP, the transport protocol between Acme SBC and Session Manager across the enterprise network is also UDP.

In the compliance testing, Bell Canada provided service provider public SIP domain as **sipxxxxxxxx.bell.ca** and enterprise public SIP domain as **cust6xxxx.xxxx.bell.ca**. These public SIP domains will be used for public SIP traffic between Acme SBC and Bell Canada. The Avaya lab was configured with private SIP domain **avaya.com** for the enterprise. Acme SBC was used to adapt this private SIP domain to public SIP domains known to Bell Canada. **Figure 1** below illustrates network diagram for the enterprise.

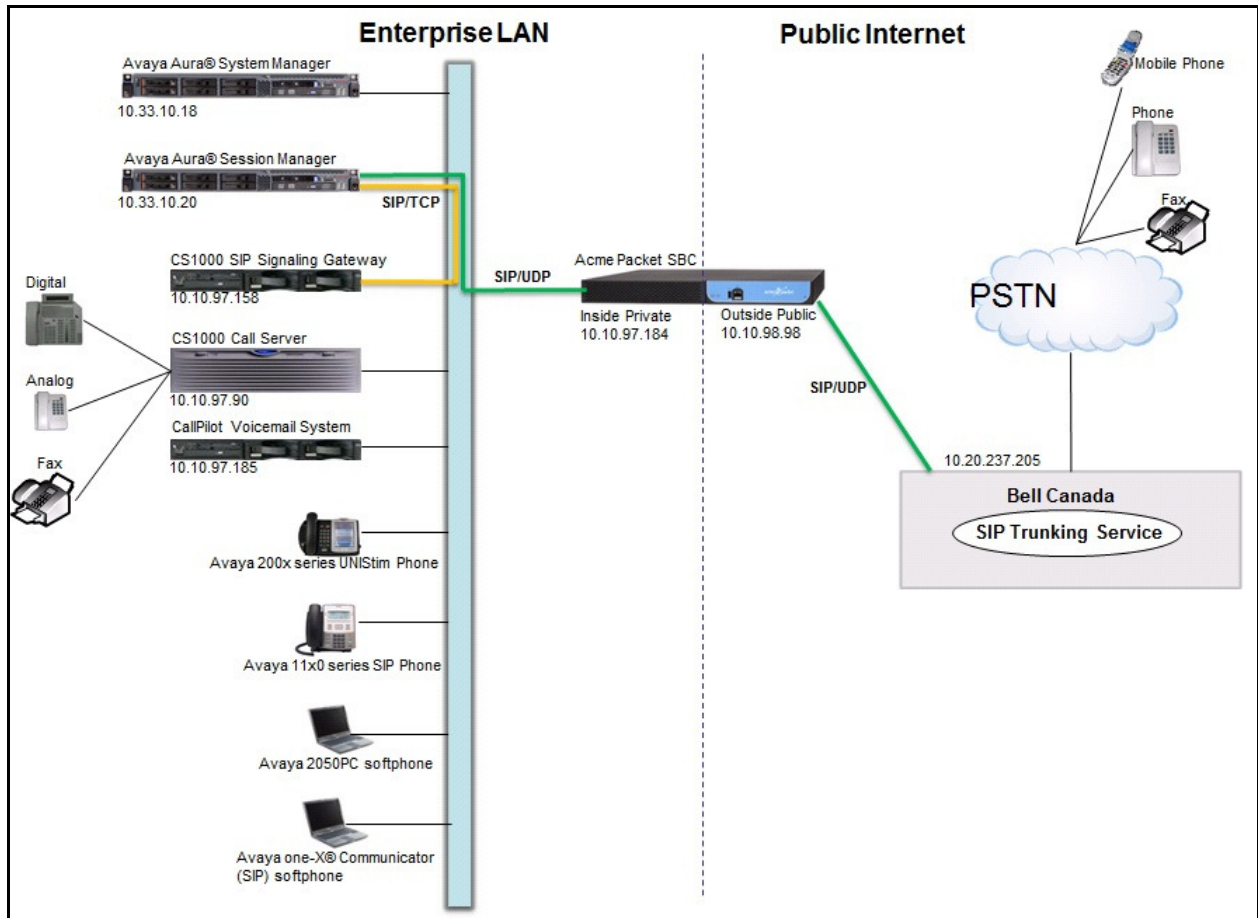


Figure 1: Avaya IP Telephony Network connecting to Bell Canada SIP Trunking Service

4. Equipment and Software Validated

The following equipment and software are used for the sample configuration provided:

Avaya IP Telephony Solution Components	
Equipment/Software	Release/Version
Avaya CS1000 7.5 (CPPM)	<ul style="list-style-type: none"> • Call Server: 7.50 Q GA plus latest DEPLIST Issue: 01 Release: 2012-12-12 18:39:40 (est) and patch MPLR32246 • SSG and SLG Server: 7.50.17 GA plus latest Service_Pack_Linux_7.50_17_20121212.ntl and patch MPLR30224
Avaya Aura® Session Manager running on Avaya S8800 Server	6.1.7.0.617012
Avaya Aura® System Manager running on Avaya S8800 Server	6.1.5.0 (6.1.12.1.1960)
Avaya Call Pilot	05.00.41.141
Avaya IP telephone	<ul style="list-style-type: none"> • 2002 p2: 0604DCJ (UNISim) • 2004 p2: 0604DCJ (UNISim) • 1140: 0625C6O (UNISim) • 1120: 0624C6O (UNISim) • 2007: 0621C6M (UNISim) • SIP 1120, 1140: SIP11x0e04.03.12.00
Avaya 2050PC softphone	3.4
Avaya one-X® Communicator (SIP) for CS1000	CS6.1.0.25-GA-33661
Avaya digital telephone	n/a
Avaya analog telephone	n/a
Fax machine	n/a
Acme Packet Net-Net 3800 Session Border Controller	ACME Net-Net 3800 Firmware SCX6.2.0 MR-12 Patch 3 (Build 1181) Build Date=03/14/13
Bell Canada SIP Trunking Service Components	
Equipment/Software	Release/Version
Bell Canada SIP Trunking Service	Version 1.3

Table 1: Equipment and Software Tested

5. Avaya Communication Server 1000 Configuration

This section describes procedure for configuring CS1000 for inter-operating with Bell Canada.

A two-way SIP Trunk was created between CS1000 and Session Manager to carry traffic to and from service provider respectively. Incoming calls flow from Bell Canada networks to Acme SBC to CS1000 via Session Manager. Incoming calls into CS1000 may undergo call treatments such as incoming digit translations and class of service restrictions. Outgoing calls to PSTN are first processed by CS1000 for call treatments such as route selection and class of service. Once CS1000 selects the proper SIP Trunk, the calls are routed to Acme SBC via Session Manager for egress to Bell Canada networks.

For the compliance testing, Bell Canada applied Digest Authentication for outgoing calls from the enterprise, using challenge-response authentication based on a configured user name and password (provided by Bell Canada and configured on Acme SBC). This call authentication scheme as specified in SIP RFC 3261 provides authentication for SIP signaling.

These Application Notes assume basic configuration for CS1000 has already been administered and therefore not discussed here. For further information on CS1000, see **References** in **Section 11**.

5.1. Log into Avaya Communication Server 1000

5.1.1. Log into Unified Communications Management (UCM) and Element Manager (EM)

Open web browser and connect to UCM <https://<UCM IP address>> as shown in the screenshot below then log in using an appropriate username and password.




This computer system and network is PRIVATE and PROPRIETARY of [company name] and may only be accessed by authorized users. Unauthorized use of this computer system or network is strictly prohibited and may be subject to criminal prosecution, employee discipline up to and including discharge, or the termination of the vendor/service contracts. The owner, or its agents, may monitor any activity or communication on the computer system or network.

User ID:

Password:

On **Avaya Unified Communications Management** page that is shown in the following screenshot, click **Element Name** of CS1000 Element as highlighted in the red box.



Avaya Unified Communications Management
[Help](#) | [Logout](#)

- Network
 - Elements
 - CS 1000 Services
 - IPSec
 - Patches
 - SNMP Profiles
 - Secure FTP Token
 - Software Deployment
- User Services
 - Administrative Users
 - External Authentication
 - Password
- Security
 - Roles
 - Policies

Host Name: car2-sipl-ucm.bwwdev.com
Software Version: 02.20.0029.00(5448)
User Name admin

Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.

	Element Name	Element Type	Release	Address	Description
1	EM on car2-cores	CS1000	7.5	10.10.97.90	New element

The following screenshot shows CS1000 Element Manager **System Overview** page.



5.1.2. Log into Call Server Command Line Interface (CLI)

Using Putty, SSH to the IP address of CS1000 SSG with *admin* account then run command *cslogin* to login with appropriate *admin* account and password.

```
login as: admin

                Avaya Inc. Linux Base 7.50
The software and data stored on this system are the property of,
or licensed to, Avaya Inc. and are lawfully available only
to authorized users for approved purposes. Unauthorized access
to any software or data on this system is strictly prohibited and
punishable under appropriate laws. If you are not an authorized
user then do not try to login. This system may be monitored for
operational purposes at any time.

admin@10.10.97.172's password:
Last login: Wed Nov  2 11:32:26 2011 from 10.10.98.105

[admin@car2-ssg2 ~]$
[admin@car2-ssg2 ~]$ cslogin

SEC054 A device has connected to, or disconnected from, a pseudo tty without
authenticating

TTY 09 SCH MTC TRF BUG OSN   14:05
OVL111 BKGD  44
OVL111 TTY 10      0      ADMIN
```

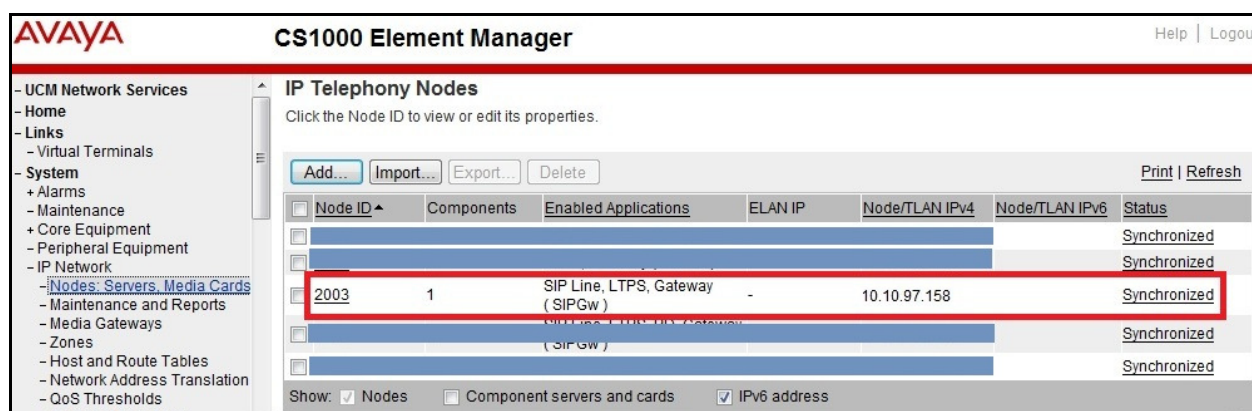
5.2. Administer Node IP Telephony

This section describes how to configure Node IP Telephony on CS1000.

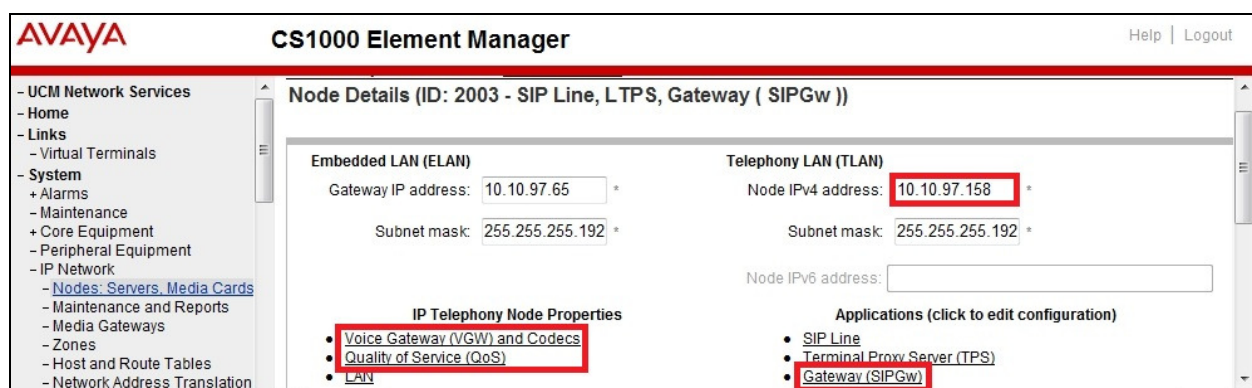
5.2.1. Obtain Node IP Address

These Application Notes assume basic configuration has already been administered and a Node has already been created. This section describes configuration steps for Node ID 2003.

To create Node IP Telephony, select **System → IP Network → Nodes: Servers, Media Cards** from the **CS1000 Element Manager** page. In the **IP Telephony Nodes** page as shown in the screenshot below, click Node ID 2003 of CS1000.

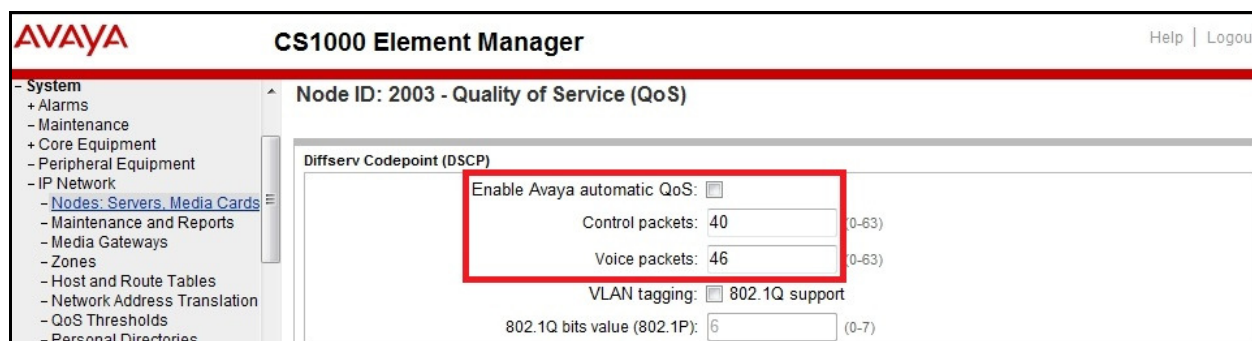


Node Details page is shown in the screenshot below with IP address of Node ID 2003. CS1000 SSG uses **Node IP Address** to connect to Acme SBC for the SIP Trunk to Bell Canada.



5.2.2. Administer Quality of Service (QoS)

On **Node Details** page as shown in **Section 5.2.1**, select **Quality of Service (QoS)** link to open **Node ID: 2003 – Quality of Service (QoS)** page. Retain default Diffserv values as shown in the screenshot below then click **Save** button (not shown).



5.2.3. Synchronize the new configuration

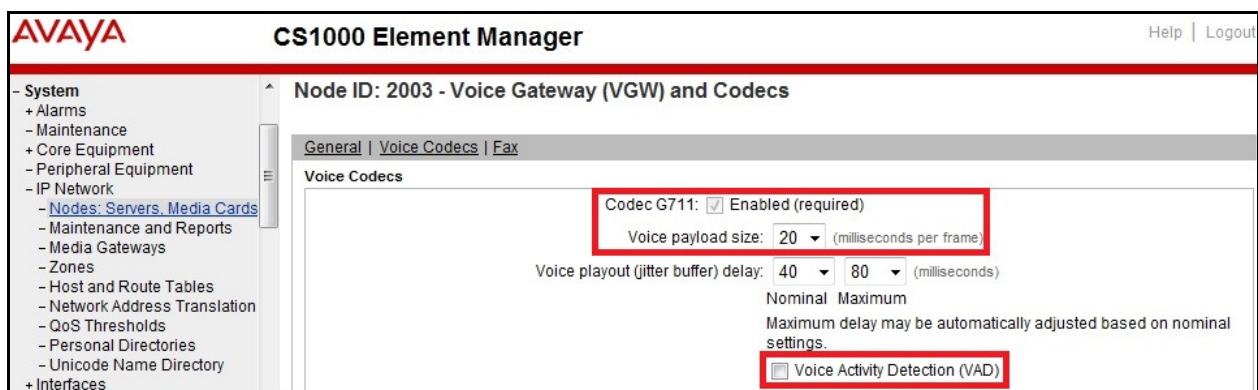
On **Node Details** page as shown in **Section 5.2.1**, perform following procedure to synchronize the new configuration:

- Click **Save** button (not shown).
- On **Node Saved** screen that displays, click **Transfer Now** button (not shown).
- On **Synchronize Configuration Files** screen that displays, check **Signaling Server** checkbox and click **Start Sync** button (not shown).
- When the synchronization completes, check **Signaling Server** checkbox and click **Restart Applications** button (not shown).

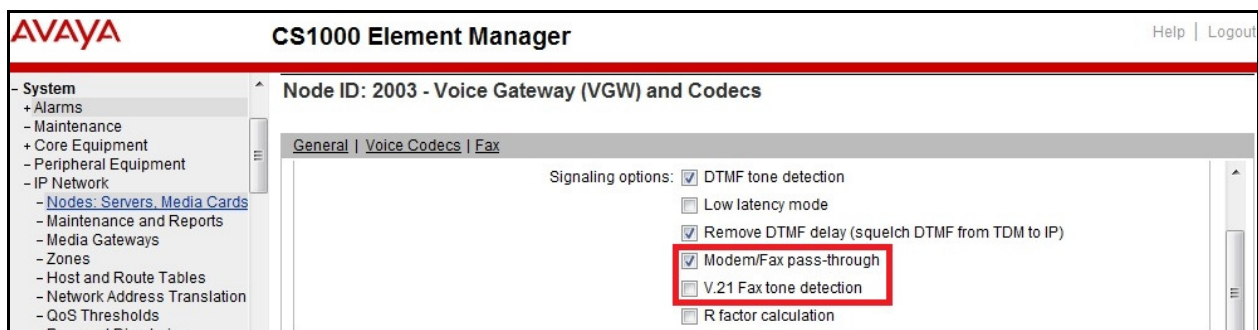
5.3. Administer Voice Codec

5.3.1. Enable Voice Codec, Node IP Telephony

On **Node Details** page as shown in **Section 5.2.1**, click **Voice Gateway (VGW) and Codec**. In this compliance testing, Bell Canada supports voice codec G.711, payload size 20 ms, with VAD disabled. The following screenshot shows appropriate voice codec profile configured on CS1000.



For Fax over IP, Bell Canada supports G.711MU codec as default. The following screenshot shows **Modem Pass Through** was selected for Node ID 2003. This enables G.711MU codec to be used for fax call between CS1000 and Bell Canada. **Note: V.21 Fax tone detection** should be unchecked to disable T.38 fax capability on the SIP Trunk.



Click **Save** (not shown) then synchronize the new configuration (see **Section 5.2.3**).

5.3.2. Administer Voice Codec on Media Gateways

CS1000 uses Media Gateways to support traditional analog and digital phones for voice calls over SIP Trunk. Media Gateways are also needed to support analog terminals to send fax over IP.

To configure Voice Codec on Media Gateways, from the left menu of Element Manager (not shown) select the **IP Network** → **Media Gateways**. On Media Gateways page that appears (not shown), click **MGC** link which is located on the right of the page (not shown).

In this compliance testing, Bell Canada supports voice codec G.711, payload size 20 ms, with VAD disabled. The screenshot below shows appropriate codec profile configured for Media Gateways.

The screenshot shows the AVAYA CS1000 Element Manager interface. On the left is a navigation tree with 'Media Gateways' selected under 'IP Network'. The main panel shows configuration for a G.711 codec. A dropdown menu is set to 'G711'. Below it, 'Voice payload size' is set to '20' ms/frame. 'Voice playout (jitter buffer) nominal delay' is set to '40' and 'Voice playout (jitter buffer) maximum delay' is set to '80'. Red text indicates that modifications may cause changes to dependent settings. At the bottom, the 'VAD' checkbox is unchecked.

For Fax over IP, Bell Canada supports G.711 codec as default. The following screenshot shows **Modem Pass Through** is selected for Media Gateway; this enables G.711MU codec to be used for fax calls between CS1000 and Bell Canada. **Note: V.21 Fax tone detection** should be unchecked to disable T.38 fax capability on the Media Gateway.

The screenshot shows the AVAYA CS1000 Element Manager interface for Fax over IP configuration. The 'Remove DTMF delay (squelch DTMF from TDM to IP)' checkbox is checked. The 'Enable modem/fax pass through mode' checkbox is checked and highlighted with a red box. The 'Enable V.21 FAX tone detection' checkbox is unchecked. Other settings include 'Fax TCF method' set to '2', 'FAX maximum rate' set to '14400' bps, 'FAX playout nominal delay' set to '100' ms (0-300 ms range), and 'FAX no activity timeout' set to '20' ms (10-32000 ms range).

5.4. Administer Zones and Bandwidth

CS1000 uses zone configuration for bandwidth management purposes. This section describes steps to create 2 zones: MO zone for VGW and IP phone, VTRK zone for the SIP Trunk.

Bell Canada supports only G.711 codec on the test environment. In the sample configuration, MO and VTRK zones were configured with **Strategy Best Quality (BQ)** to allow CS1000 to prioritize G.711 codec for both voice and fax calls.

In general, bandwidth zone is configured with parameters described as following:

- **INTRA_STGY**: Bandwidth configuration for local calls.
- **INTER_STGY**: Bandwidth configuration for the calls over trunk.
- **BQ**: G.711 is first choice and G.729 is second choice.
- **BB**: G.729 is first choice and G.711 is second choice.
- **MO**: The zone type which is used for IP phones and Voice Gateway (VGW).
- **VTRK**: The zone type which is used for the SIP Trunk.

5.4.1. Create Zone for VGW and IP phones

To create MO zone **10** for VGW and IP phone, select **IP Network → Zones** from the left pane of the **CS1000 Element Manager** page, then configure as following:

- Click **Bandwidth Zones** link (not shown).
- On **Bandwidth Zones** screen, click **Add** button (not shown).
- On **Add Bandwidth Zone** screen, click **Zone Basic Property and Bandwidth Management**, define the values as shown in the screenshot below then click **Submit** button (not shown).

Input Description	Input Value
Zone Number (ZONE):	10 (1 - 8000)
Intrazone Bandwidth (INTRA_BW):	100000 (0 - 10000000)
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	100000 (0 - 10000000)
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

```
ERROR: undefined
OFFENDING COMMAND: '~
STACK:
```