



Application Notes for Configuring Avaya Communication Server 1000E R7.5, Avaya Aura® Session Manager R6.1, Avaya Session Border Controller for Enterprise R4.0.5 to support Phonect SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describes the steps to configure Session Initiation Protocol (SIP) Trunking between Phonect SIP Trunk Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Session Manager, Avaya Session Border Controller for Enterprise and Avaya Communication Server 1000E.

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

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Phonect SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura[®] Session Manager, Avaya Communication Server 1000E (CS1000E) connected to Phonect SIP Trunk Service via an Avaya Session Border Controller for Enterprise (Avaya SBCE). Customers using this Avaya SIP-enabled enterprise solution with Phonect SIP Trunk Service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach normally results in lower cost for the enterprise.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Session Manager, Avaya SBCE and CS1000E. The enterprise site was configured to use the SIP Trunk to connect to Phonect SIP Trunk Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

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.

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- Incoming PSTN calls were made to Unistim, SIP, Digital and Analog telephones at the enterprise
- Incoming calls to the enterprise site from the PSTN routed to the DDI numbers assigned by Phonect
- Outgoing calls from the enterprise to the PSTN were made from Unistim, SIP, Digital and Analog telephones
- Outgoing calls from the enterprise site were completed via Phonect to PSTN destinations
- Calls using the G.711A, G.711U and G.729 codec supported by Phonect
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using T.38
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls
- User features such as hold and resume, transfer, conference, call forwarding, etc
- Caller ID Presentation and Caller ID Restriction
- Call coverage and call forwarding for endpoints at the enterprise site
- Mobile-X call features
- Off-net call forwarding and mobility (extension to mobile)

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Phonect SIP Trunk with the following observations:

- Issue observed when testing Blind/Consultative Transfers to PSTN. Service Provider wrongly responded 200OK without SDP to (slowstart) reINVITE. As Service Provider does not support slowstart, Session Manager would not forward this 200OK to CS1000E so the Service Provider would then send a BYE to disconnect after the session provisioning had timed out. To resolve the issue, the SDP needs to be embedded in every INVITE request/reply to the Service Provider. This is achieved by enabling “Delayed SDP Handling” on the Avaya SBCE. The guidelines on how to enable this feature is documented in **Section 7.2.1** and **Section 7.2.2** of this document
- As Phonect do not support SIP 302 Moved Temporarily, the CS1000E will not allow a blind transfer to be executed if the parties involved do not support the SIP 302 Moved Temporarily method. With the installation of plugin 501 on the CS1000E, the blind transfer will be allowed and the call will be completed. The limitation of this plugin is that no ringback is provided to the originator of the call for the duration that the destination set is ringing. In addition to plugin 501, it is required that **VTRK SU version “cs1000-vtrk-7.50.17.16-15.i386.000.ntl”** or higher be used on all SSG signaling servers to ensure proper operation of the blind transfer feature. The use of plugin 501 does not restrict the use of the SIP 302 method of blind transfer to other parties that do happen to support the SIP 302 method, but rather extend support to those parties that do not
- All unwanted MIME was stripped on outbound calls using the Adaptation Module in Session Manager
- No inbound toll free numbers were tested as none were available from the Service Provider
- No Emergency Services numbers were tested as test calls to these numbers need to be pre-arranged with the Operator

2.3. Support

For initial setup for your Avaya solution please contact leveranse@phonect.no. For technical support send email to bedriftsupport@phonect.no.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Phonect SIP Trunks Service. Located at the enterprise site are Session Manager, Avaya SBCE and CS1000E. Endpoints are Avaya 1140 series IP telephones, Avaya 1200 series (not shown in **Figure 1**) IP telephones (with Unistim and SIP firmware), Avaya IP Softphones (SMC3456, 2050 and Avaya one-X® Communicator), Avaya Digital telephone, Analog telephone and fax machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are masked in these Application Notes.

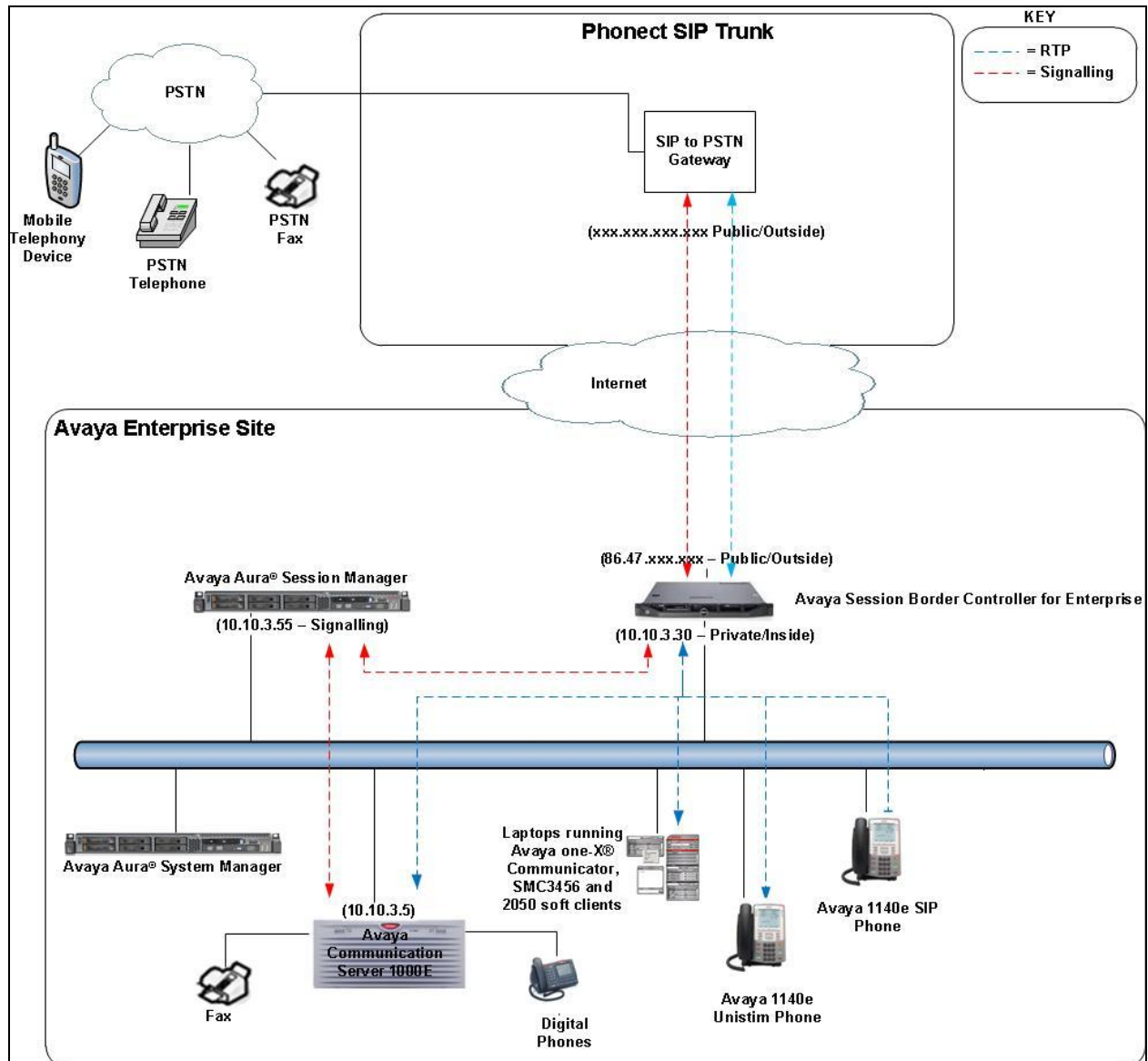


Figure 1: Test Setup Phonect SIP Trunk Service to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Session Manager running on Avaya S8800 server	R6.1 Build: 6.1.0.0.610023
Avaya Aura® System Manager running on Avaya S8800 server	R6.1 Load: 6.1.0.0.7345 Service Pack 6
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	R7.5, Version 7.50.17 Service Update: 7.50_14Mar12 Deplist: X21 07.50Q
Avaya Session Border Controller for Enterprise on Dell R210 V2 server	Build: 4.0.5.Q19
Avaya Communication Server 1000E Media Gateway	CSP Version: MGCC CD02 MSP Version: MGCM AB01 APP Version: MGCA BA07 FPGA Version: MGCF AA18 BOOT Version: MGCB BA15 DSP1 Version: DSP1 AB04
Avaya 1140e and 1230 Unistim Telephones	FW: 0625C8A
Avaya 1140e and 1230 SIP Telephones	FW: 04.01.13.00.bin
Avaya SMC 3456	Version 2.6 build 53715
Avaya 2050 IP Softphone	Release 4.03.0081
Avaya one-X® Communicator	Version cs6.1.0.10
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
Phonect SIP Trunk Service:	
Network Equipment: Dual Cisco ASR 1002 - Cisco IOS Software, IOS-XE Software Version 15.0(1)S setup with HSRP internally and BGP on WAN towards several ISP's for redundancy.	Soft switch: PortaSwitch MR-22

5. Configure Avaya Communication Server 1000E

This section describes the steps required to configure CS1000E for SIP Trunking and also the necessary configuration for terminals (analog, SIP and IP phones). SIP trunks are established between CS1000E and Session Manager. These SIP trunks carry SIP signaling associated with Phonect SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from the Avaya SBCE; through which Phonect's SIP Service directs incoming SIP messages to CS1000E (see **Figure 1**). Once a SIP message arrives at CS1000E, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within CS1000E and may be first subject to outbound features treatment such as route selection, digit manipulation and class of service

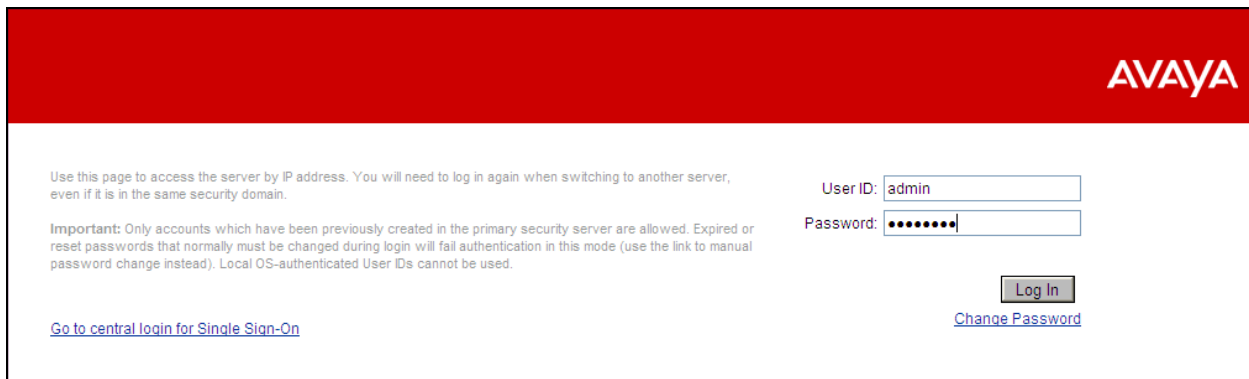
restrictions. Once CS1000E selects a SIP trunk, the SIP signaling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the Avaya SBCE and on to Phonect's network. Specific CS1000E configuration was performed using Element Manager and the system terminal interface. The general installation of the CS1000E, System Manager and Session Manager is presumed to have been previously completed and is not discussed here.

5.1. Log in to the Avaya Communication Server 1000E

Configuration on the CS1000E will be performed by using both SSH Putty session and Avaya Unified Communications Management GUI.

Log in using SSH to the ELAN IP address of the Call Server using a user with correct privileges. Once logged in type **csconsole**, this will take the user into the vxworks shell of the call server. Next type **login**, the user will then be asked to login with correct credentials. Once logged in the user can then progress to load any overlay.

Log in using the web based Avaya Unified Communications Management GUI. The Avaya Unified Communications Management GUI may be launched directly via <http://<ipaddress>> where the relevant <ipaddress> is the TLAN ip address of the CS1000E.



The screenshot shows the Avaya Unified Communications Management GUI login page. At the top right is the Avaya logo. Below it, a message states: "Use this page to access the server by IP address. You will need to log in again when switching to another server, even if it is in the same security domain." To the right of this message are two input fields: "User ID:" with the text "admin" and "Password:" with a masked password of seven dots. Below the password field is a "Log In" button. To the left of the "Log In" button is a link: "Go to central login for Single Sign-On". Below the "Log In" button is a link: "Change Password".

The Avaya Unified Communications Management Elements page will be used for configuration. Click on the Element Name corresponding to CS1000E in the Element Type column. In the abridged screen below, the user would click on the Element Name **EM on cs1kv13**.

Host Name: Software Version: 02.20_SMGR-SNAPSHOT(4554) 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.

<input type="checkbox"/>	Element Name	Element Type ▲	Release	Address	Description ▲
<input type="checkbox"/>	1 smgrv3.avaya.com (primary)	Base OS	7.5	10.10.3.52	Base OS element.
<input type="checkbox"/>	2 EM on cs1kv3	CS1000	7.5	192.168.1.5	New element.
<input type="checkbox"/>	3 cs1kv3.avaya.com (member)	Linux Base	7.5	10.10.3.5	Base OS element.
<input type="checkbox"/>	4 192.168.1.3	Media Gateway Controller	7.5	192.168.1.3	New element.
<input type="checkbox"/>	5 NRSIM on cs1kv3	Network Routing Service	7.5	192.168.1.5	New element.

5.2. Confirm System Features

The keycode installed on the Call Server controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the CS1000E system terminal and manually load overlay 22 to print the System Limits (the required command is **SLT**), and verify that the number of **SIP Access Ports** reported by the system is sufficient for the combination of trunks to Phonect's network, and any other SIP trunks needed. See the following screenshot for a typical System Limits printout. The value of **SIP ACCESS PORTS** defines the maximum number of SIP trunks for the CS1000E.

```
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

IPMGs Registered:          1
IPMGs Unregistered:       0
IPMGs Configured/unregistered: 0

TRADITIONAL TELEPHONES 32767 LEFT 32766 USED 1
DECT USERS              32767 LEFT 32767 USED 0
IP USERS                32767 LEFT 32744 USED 23
BASIC IP USERS          32767 LEFT 32766 USED 1
TEMPORARY IP USERS      32767 LEFT 32767 USED 0
DECT VISITOR USER       10000 LEFT 10000 USED 0
ACD AGENTS              32767 LEFT 32752 USED 15
MOBILE EXTENSIONS       32767 LEFT 32767 USED 0
TELEPHONY SERVICES      32767 LEFT 32767 USED 0
CONVERGED MOBILE USERS  32767 LEFT 32767 USED 0
NORTEL SIP LINES        32767 LEFT 32765 USED 2
THIRD PARTY SIP LINES   32767 LEFT 32761 USED 6
SIP CONVERGED DESKTOPS  32767 LEFT 32767 USED 0
SIP CTI TR87            32767 LEFT 32767 USED 0
SIP ACCESS PORTS      2000 LEFT 1970 USED 30
```

Load **overlay 21**, and confirm the customer is setup to use **ISDN** trunks (see below).

```
REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTD
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
ISDN YES
```

5.3. Configure Codec's for Voice and FAX Operation

Phonect SIP Trunk service supports G.711A, G.711U and G.729 voice codec's and T.38 FAX transmissions. Using the CS1000E element manager sidebar, navigate to the **IP Network → IP Telephony Nodes → Node Details → VGW Gateway (VGW) and Codecs** property page and configure the CS1000E General codec settings as in the next screenshot.

Node ID: 100 - Voice Gateway (VGW) and Codecs

[General](#) | [Voice Codecs](#) | [Fax](#)

General

Echo cancellation: ☒ Use canceller, with tail delay:
 ☒ Dynamic attenuation

Voice activity detection threshold: (-20 - +10 DBM)

Idle noise level: (-327 - +327 DBM)

Signaling options: ☒ DTMF tone detection
 ☐ Low latency mode
 ☒ Remove DTMF delay (squelch DTMF from TDM to IP)
 ☒ Modem/Fax pass-through
 ☒ V.21 Fax tone detection
 ☐ R factor calculation

Next, scroll down and configure the **Codec G.711**. The relevant settings are highlighted in the following screenshot.

Managing: 192.168.1.5 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 100 - Voice Gateway (VGW) and Codecs

General | **Voice Codecs** | Fax

Voice Codecs

Codec G711: ☒ Enabled (required)

Voice payload size: 20 (milliseconds per frame)

Voice playout (jitter buffer) delay: 40 80 (milliseconds)

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

☐ Voice Activity Detection (VAD)

Next, scroll down and configure the **Codec G.729**. The relevant settings are highlighted in the following screenshot.

Managing: 192.168.1.5 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 100 - Voice Gateway (VGW) and Codecs

General | **Voice Codecs** | Fax

Voice Codecs

Codec G729: ☒ Enabled

Voice payload size: 20 (milliseconds per frame)

Voice playout (jitter buffer) delay: 40 80 (milliseconds)

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

☐ Voice Activity Detection (VAD)

Finally, configure the **Fax** settings as in the highlighted section of the next screenshot.

Managing: 192.168.1.5 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 100 - Voice Gateway (VGW) and Codecs

General | Voice Codecs | Fax

Codec G723.1: ☐ Enabled
Voice payload size: 30 (milliseconds per frame)
Voice playout (jitter buffer) delay: 60 120 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Coding rate: 5.3 (kbps)

Fax

Codec name: T.38 FAX
Maximum rate: 14400 (bps)
Fax TCF method: 2
Fax playout nominal delay: 100 (0 - 300 milliseconds)
FAX no activity timeout: 20 (10 - 32000 milliseconds)
Packet size: 30 (bps)

5.4. Virtual Trunk Gateway Configuration

Use CS1000E Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and verify the highlighted section is completed with the correct IP addresses and subnet masks of the Node. At this stage the call server has an IP address and so too does the signalling server. The Node IP is the IP address that the IP phones use to register. This is also where the SIP trunk connection is made to the Session Manager. When an entity link is added in Session Manager for the CS1000E, it is the Node IP that is used (please see **Section 6.5 – Define SIP Entities** for more details).

Managing: 192.168.1.5 Username: admin
System » IP Network » IP Telephony Nodes » Node Details

Node Details (ID: 100 - SIP Line, LTPS, Gateway (SIPGw))

Node ID:	<input type="text" value="100"/>	* (0-9999)	
Call server IP address:	<input type="text" value="192.168.1.5"/>	TLAN address type:	<input checked="" type="radio"/> IPv4 only <input type="radio"/> IPv4 and IPv6
Embedded LAN (ELAN)		Telephony LAN (TLAN)	
Gateway IP address:	<input type="text" value="192.168.1.1"/>	Node IPv4 address:	<input type="text" value="10.10.3.6"/>
Subnet mask:	<input type="text" value="255.255.255.0"/>	Subnet mask:	<input type="text" value="255.255.255.0"/>
		Node IPv6 address:	<input type="text"/>

* Required Value.

Associated Signaling Servers & Cards

[Print](#) | [Refresh](#)

<input type="checkbox"/> Hostname ▲	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/> cs1kv3	Signaling_Server	SIP Line, LTPS, Gateway, PD, Presence Publisher, IP Media Services	192.168.1.5	10.10.3.5	Leader

The next screenshots show the SIP Virtual Trunk Gateway configuration, navigate to **System → IP Networks → IP Telephony Nodes → Node Details → Virtual Trunk Gateway Configuration** and fill in the highlighted areas with the relevant settings.

- **Vtrk gateway application:** Provides option to select Gateway applications. The three supported modes are **SIP Gateway (SIPGw)**, **H.323Gw**, and **SIPGw and H.323Gw**
- **SIP domain name:** The SIP Domain Name is the SIP Service Domain. The SIP Domain Name configured in the Signalling Server properties must match the Service Domain name configured in the Session Manager, in this case **avaya.com**
- **Local SIP port:** The Local SIP Port is the port to which the gateway listens. The default value is **5060**
- **Gateway endpoint name:** This field cannot be left blank so a value is needed here. This field is used when a Network Routing Server is used for registration of the endpoint. In this network a Session Manager is used so any value can be put in here and will not be used
- **Application node ID:** This is a unique value that can be alphanumeric and is for the new Node that is being created, in this case **100**
- **Proxy or Redirect Server:** Primary TLAN IP address is the Security Module IP address of the Session Manager. The **Transport protocol** used for **SIP**, in this case **TCP**
- **SIP URI Map:** **Public National** and **Private Unknown** are left blank. All other fields in the SIP URI Map are left with default values

Node ID: 100 - Virtual Trunk Gateway Configuration Details

[General](#) | [SIP Gateway Settings](#) | [SIP Gateway Services](#)Vtrk gateway application: ☒ Enable gateway service on this node

General

Vtrk gateway application: SIP Gateway (SIPGw) ▼

SIP domain name: avaya.com *

Local SIP port: 5060 * (1 - 65535)

Gateway endpoint name: cs1kv3 *

Gateway password: *

Application node ID: 100 * (0-9999)

Enable failsafe NRS: ☐SIP ANAT: ☒ IPv4☐ IPv6

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)

Information will be captured for the IP addresses listed below.

Monitor IP:

Monitor addresses:

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 10.10.3.55

The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: TCP ▼

Options: ☐ Support registration☐ Primary CDS proxy

SIP URI Map:

Public E.164 domain names

National:

Subscriber: subscriber

Special number: PublicSpecial

Unknown: PublicUnknown

Private domain names

UDP: udp

CDP: cdp.udp

Special number: PrivateSpecial

Vacant number: PrivateUnknown

Unknown:

5.5. Configure Bandwidth Zones

Bandwidth Zones are used for alternate call routing between IP stations and for Bandwidth Management. SIP trunks require a unique zone, not shared with other resources and best practice dictates that IP telephones and Media Gateways are all placed in separate zones. In the sample configuration SIP trunks use zone 01 and IP Telephones use zone 02, system defaults were used for each zone other than the parameter configured for **Zone Intent**. For SIP Trunks (zone 01), **VTRK** is configured for **Zone Intent**. For IP, SIP Telephones (zone 02), **MO** is configured for **Zone Intent**.

Use Element Manager to define bandwidth zones as in the following highlighted example. Use Element Manager and navigate to **System → IP Network → Zones → Bandwidth Zones** and add new zones as required.

Managing: 192.168.1.5 Username: admin
System » IP Network » Zones » Bandwidth Zones

Bandwidth Zones

Add... Edit... Import... Export Maintenance... Delete

Zone	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description
1 1	1000000	BQ	1000000	BQ	SHARED	VTRK	
2 2	1000000	BQ	1000000	BQ	SHARED	MO	

5.6. Configure Incoming Digit Conversion Table

A limited number of Direct Dial Inwards (DDI) numbers were available. The IDC table was configured to translate incoming PSTN numbers to four digit local telephone extension numbers. The digits of the actual PSTN DDI number are obscured for security reasons. The following screenshot shows the incoming PSTN numbers converted to local extension numbers. These were altered during testing to map to various SIP, Analog, Digital or Unistim telephones depending on the particular test case being executed.

Managing: 192.168.1.5 Username: admin
Dialing and Numbering Plans » Incoming Digit Translation » Customer 00 » Digit Conversion Tree 0 Configuration

Digit Conversion Tree 0 Configuration

Regular IDC tree
Send calling party DID disabled

Add... Delete IDC Delete IDC tree

	Incoming Digits	Converted Digits	CPND Name
1 472		5000	
2 472		5000	
3 472		5001	
4 472		5003	
5 472		5004	
6 472		5005	
7 472		5006	
8 472		5009	

5.7. Configure SIP Trunks

CS1000E virtual trunks will be used for all inbound and outbound PSTN calls to Phonect's SIP Trunk Service. Six separate steps are required to configure CS1000E virtual trunks.

- Configure a D-Channel Handler (**DCH**); configure using the CS1000E system terminal and overlay 17
- Configure a SIP trunk Route Data Block (**RDB**); configure using the CS1000E system terminal and overlay 16
- Configure SIP trunk members; configure using the CS1000E system terminal and overlay 14
- Configure a Digit Manipulation Data Block (**DGT**), configure using the CS1000E system terminal and overlay 86
- Configure a Route List Block (**RLB**); configure using the CS1000E system terminal and overlay 86
- Configure Co-ordinated Dialling Plan(s) (**CDP**); configure using the CS1000E system terminal and overlay 87

The following is an example DCH configuration for SIP trunks. Load **Overlay 17** at the CS1000E system terminal and enter the following values. Verify **DCIP** has been selected for **CTYP** (D-Channel Card Type) and **IFC** (Interface Type) has been set to **SL1**. The highlighted entries are required for correct SIP trunk operation. Exit overlay 17 when completed.

```
Overlay 17
ADAN      DCH 1
CTYP DCIP
DES  VIR TRK
USR  ISLD
ISLM 4000
SSRC 3700
OTBF 32
NASA YES
IFC SL1
CNEG 1
RLS  ID 4
RCAP ND2
MBGA NO
H323
OVLN NO
OVLS NO
```

Next, configure the SIP trunk Route Data Block (RDB) using the CS1000E system terminal and overlay 16. Load **Overlay 16**, enter **RDB** at the prompt, press return and commence configuration. The value for **DCH** is the same as previously entered in overlay 17. The value for **NODE** should match the node value in **Section 5.4**. The value for **ZONE** should match that used in **Section 5.5** for **SIP_VTRK**. The remaining highlighted values are important for correct SIP trunk operation.

Overlay 16 TYPE: RDB CUST 00 ROUT 1 TYPE RDB CUST 00 ROUT 1 DES VIR_TRK TKTP TIE NPID_TBL_NUM 0 ESN NO RPA NO CNVT NO SAT NO RCLS EXT VTRK YES ZONE 00001 PCID SIP CRID NO NODE 100 DTRK NO ISDN YES MODE ISLD DCH 1 IFC SL1 PNI 00000 NCNA YES NCRD YES TRO NO FALT NO CTYP UKWN INAC NO ISAR NO DAPC NO MBXR NO MBXOT NPA MBXT 0 PTYP ATT CNDP UKWN AUTO NO DNIS NO DCDR NO ICOG IAO SRCH LIN TRMB YES STEP	ACOD 1111 TCPP NO PII NO AUXP NO TARG CLEN 1 BILN NO OABS INST IDC YES DCNO 0 NDNO 0 * DEXT NO DNAM NO SIGO STD STYP SDAT MFC NO ICIS YES OGIS YES TIMR ICF 1920 OGF 1920 EOD 13952 LCT 256 DSI 34944 NRD 10112 DDL 70 ODT 4096 RGV 640 GTO 896 GTI 896 SFB 3 PRPS 800 NBS 2048 NBL 4096 IENB 5 TFD 0 VSS 0 VGD 6 EESD 1024 SST 5 0 DTD NO SCDT NO 2 DT NO NEDC ORG FEDC ORG	CPDC NO DLTN NO HOLD 02 02 40 SEIZ 02 02 SVFL 02 02 DRNG NO CDR NO NATL YES SSL CFWR NO IDOP NO VRAT NO MUS YES MRT 21 PANS YES RACD NO MANO NO FRL 0 0 FRL 1 0 FRL 2 0 FRL 3 0 FRL 4 0 FRL 5 0 FRL 6 0 FRL 7 0 OHQ NO OHQT 00 CBQ NO AUTH NO TTBL 0 ATAN NO OHTD NO PLEV 2 OPR NO ALRM NO ART 0 PECL NO DCTI 0 TIDY 1600 100 ATRR NO TRRL NO SGRP 0 ARDN NO CTBL 0 AACR NO
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Next, configure virtual trunk members using the CS1000E system terminal and **Overlay 14**. Configure sufficient trunk members to carry both incoming and outgoing PSTN calls. The following example shows a single SIP trunk member configuration. Load **Overlay 14** at the system terminal and type **new X**, where X is the required number of trunks. Continue entering data until the overlay exits. The **RTMB** value is a combination of the **ROUT** value entered in the previous step and the first trunk member (usually 1). The remaining highlighted values are important for correct SIP trunk operation.

```
Overlay 14
TN 100 0 0 0
DATE
PAGE
DES VIR_TRK
TN 100 0 00 00 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 00001
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK ANLG
NCOS 0
RTMB 1 1
CHID 1
TGAR 1
STRI/STRO IMM IMM
SUPN YES
AST NO
IAPG 0
CLS UNR DIP CND ECD WTA LPR APN THFD XREP SPCD MSBT
P10 NTC
TKID
AACR NO
```

Next, configure a Digit Manipulation data block (DGT) in overlay 86. Load **Overlay 86** at the system terminal and type **new**. The following example shows the values used. The value for **DMI** is the same as when inputting the **DMI** value during configuration of the Route List Block.

```

Overlay 86
CUST 0
FEAT dgt
DMI 10
DEL 0
ISPN NO
CTYP NPA

```

Configure a Route List Block (RLB) in overlay 86. Load **Overlay 86** at the system terminal and type **new**. The following example shows the values used. The value for **ROUT** is the same as previously entered in overlay 16. The **RLI** value is unique to each RLB.

```

Overlay 86
CUST 0
FEAT rlb
RLI 10
ELC NO
ENTR 0
LTER NO
ROUT 1
TOD 0 ON 1 ON 2 ON 3 ON
    4 ON 5 ON 6 ON 7 ON
VNS NO
SCNV NO
CNV NO
EXP NO
FRL 0
DMI 10
CTBL 0
ISDM 0

```

```

FCI 0
FSNI 0
BNE NO
DORG NO
SBOC NRR
PROU 1
IDBB DBD
IOHQ NO
OHQ NO
CBQ NO

ISET 0
NALT 5
MFRL 0
OVLL 0

```

Next, configure Co-ordinated Dialling Plan(s) (CDP) which users will dial to reach PSTN numbers. Use the CS1000E system terminal and **Overlay 87**. The following are some example CDP entries used. The highlighted **RLI** value previously configured in overlay 86 is used as the Route List Index (**RLI**), this is the default PSTN route to the SIP Trunk service.

TSC 00353	TSC 18	TSC 800	TSC 08
FLEN 0	FLEN 0	FLEN 0	FLEN 0
RRPA NO	RRPA NO	RRPA NO	RRPA NO
RLI 10	RLI 10	RLI 10	RLI 10
CCBA NO	CCBA NO	CCBA NO	CCBA NO

5.8. Configure Analog, Digital and IP Telephones

A variety of telephone types were used during the testing, the following is the configuration for the Avaya 1140e Unistim IP telephone. Load **Overlay 20** at the system terminal and enter the following values. The value for **CFG_ZONE** is the same value used in **Section 5.5** for **VIRTUALSETS**. A unique four digit number is entered for the **KEY 00** and **KEY 01** value.

Overlay 20 IP Telephone configuration

```
DES 1140
TN 100 0 01 0 VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00002
CUR_ZONE 00002
ERL 0
ECL 0
FDN 0
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 1
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCB
ICDA CDMD LLCN MCTD CLBD AUTR
GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTA AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCB
RTDD RBDD RBHD PGND OCB
FLXD FTTC DNDY DNO3 MCBN
FDSD NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA PKCH MUTA MWTD
```

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```
DVLD CROD CROD
CPND_LANG ENG
RCO 0
HUNT 0
LHK 0
PLEV 02
PUID
DANI NO
AST 00
IAPG 1
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 MCR 5000 0      MARP
      CPND
        CPND_LANG ROMAN
        NAME IP1140
        XPLN 10
        DISPLAY_FMT FIRST, LAST
01 MCR 5000 0
      CPND
        CPND_LANG ROMAN
        NAME IP1140
        XPLN 10
        DISPLAY_FMT FIRST, LAST
02
03 BSY
04 DSP
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
```

Digital telephones are configured using the **Overlay 20**, the following is a sample **3904** digital set configuration. Again, a unique number is entered for the **KEY 00** and **KEY 01** value.

Overlay 20 - Digital Set configuration

TYPE: 3904

```
DES 3904
TN 04 0 02 00 VIRTUAL
TYPE 3904
CDEN 8D
CTYP XDLC
CUST 0
MRT
ERL 0
FDN 0
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 1
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBD WTA LPR PUA MTD FND HTD TDD HFA GRLD CRPA STSD
    MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD SLKD CCSD SWD LNA CNDA
    CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
    ICDA CDMA LLCN MCTD CLBD AUTU
    GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD
    CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
    UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
    DRDD EXR0
    USMD USRD ULAD CCBF RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
    FDSD NOVD CDMR PRED RECA MCDD T87D SBMD PKCH CROD CROD
CPND LANG ENG
RCO 0
HUNT
PLEV 02
PUID
DANI NO
SPID NONE
AST
IAPG 1
AACS
ACQ
ASID
SFNB
SFRB
USFB
CALB
FCTB
ITNA NO
DGRP
PRI 01
MLWU_LANG 0
```

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MLNG ENG

DNDR 0

KEY 00 MCR 5008 0 MARP

CPND

CPND_LANG ROMAN

NAME Digital Set

XPLN 10

DISPLAY_FMT FIRST, LAST

01 MCR 5008 0

CPND

CPND_LANG ROMAN

NAME Digital Set

XPLN 10

DISPLAY_FMT FIRST, LAST

02

03

04

05

06

07

08

09

10

11

12

13

14

15

16

17 TRN

18 AO6

19 CFW 16

20 RGA

21 PRK

22 RNP

23

24 PRS

25 CHG

26 CPN

27 CLT

28 RLT

29

30

31

Analog telephones are also configured using **Overlay 20**, the following example shows an analog port configured for Plain Ordinary Telephone Service (POTS) and also configured to allow T.38 Fax transmission. A unique value is entered for **DN**, this is the extension number. **DTN** is required if the telephone uses DTMF dialing. Values **FAXA** and **MPTD** configure the port for T.38 Fax transmissions.

Overlay 20 – Analog Telephone Configuration

```
DES 500
TN 04 0 03 00
TYPE 500
CDEN 4D
CUST 0
MRT

ERL 00000
WRLS NO
DN 5015
AST NO
IAPG 0
HUNT
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
XLST
SCI 0
SCPW
SFLT NO
CAC_MFC 0
CLS UNR DTN FBD XFD WTA THFD FND HTD ONS
      LPR XRD AGRD CWD SWD MWD RMMD SMWD LPD XHD SLKD CCSD LND TVD
      CFTD SFD MRD C6D CNID CLBD AUTU
      ICDD CDMD LLCN EHTD MCTD
      GPUD DPUD CFXD ARHD OVDD AGTD CLTD LDTD ASCD SDND
      MBXD CPFA CPTA UDI RCC HBTD IRGD DDGA NAMA MIND
      NRWD NRCD NROD SPKD CRD PRSD MCRD
      EXR0 SHL SMSD ABDD CFHD DNDY DNO3
      CWND USMD USRD CCBF BNRD OCBF RTDD RBDD RBHD FAXA CNUD CNAD PGND FTTC
      FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTD
PLEV 02
PUID
AACS NO
MLWU_LANG 0
FTR DCFW 4
```

5.9. Configure the SIP Line Gateway Service

SIP terminal operation requires the Communication Server node to be configured as a SIP Line Gateway (SLG) before SIP telephones can be configured. Prior to configuring the SIP Line node properties, the SIP Line service must be enabled in the customer data block. Use the CS1000E system terminal and **Overlay 15** to activate SIP Line services, as in the following example where **SIPL_ON** is set to **YES**.

```
SLS_DATA
SIPL_ON YES
UAPR 11
NMME NO
```

If a numerical value is entered against the **UAPR** setting, this number will be pre appended to all SIP Line configurations, and is used internally in the SIP Line server to track SIP terminals. Use Element Manager and navigate to the **IP Network → IP Telephony Nodes → Node Details → SIP Line Gateway Configuration** page. See the following screenshot for highlighted critical parameters.

- **SIP Line Gateway Application:** Enable the SIP line service on the node, check the box to enable
- **SIP Domain Name:** The value must match that configured in **Section 6.2**
- **SLG endpoint name:** The endpoint name is the same endpoint name as the SIP Line Gateway and will be used for SIP gateway registration
- **SLG Local Sip port:** Default value is **5070**
- **SLG Local TLS port:** Default value is **5071**
- Click **Save**

CS1000 Element Manager Help | Logout

Managing: 192.168.1.5 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration

Node ID: 100 - SIP Line Configuration Details

General | SIP Line Gateway Settings | SIP Line Gateway Service

SIP Line Gateway Application: ☒ Enable gateway service on this node

General

SIP domain name: *

SLG endpoint name:

SLG Group ID:

SLG Local Sip port: (1 - 65535)

SLG Local Tls port: (1 - 65535)

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)

Information will be captured for the IP addresses listed below.

Monitor IP:

Monitor addresses:

5.10. Configure SIP Line Telephones

When SIP Line service configuration is completed, use the CS1000E system terminal and **Overlay 20** to add a Universal Extension (UEXT). See the following example of a SIP Line extension. The value for **UXTY** must be **SIPL**. This example is for an Avaya SIP telephone, so the value for **SIPN** is 1. The **SIPU** value is the username, **SCPW** is the logon password and these values are required to register the SIP telephone to the SLG. The value for **CFG_ZONE** is the value set for **SIPLINEZONE** in **Section 5.5**. A unique telephone number is entered for value **KEY 00**. The value for **KEY 01** is comprised of the **UAPR** (set in **Section 5.8**) value and the telephone number used in **KEY 00**.

Overlay 20 – SIP Telephone Configuration

```
DES  SIPD
TN    100 0 01 10  VIRTUAL
TYPE  UEXT
CDEN  8D
CTYP  XDLC
CUST  0
UXTY  SIPL
MCCL  YES
SIPN  1
SIP3  0
FMCL  0
TLSV  0
SIPU  5003
NDID  100
SUPR  NO
SUBR  DFLT MWI RGA CWI MSB
UXID
NUID  100
NHTN  100 0 01 10
CFG_ZONE 00002
CUR_ZONE 00002
ERL   0
ECL   0
VSIT  NO
FDN
TGAR  0
LDN   NO
NCOS  0
SGRP  0
RNPG  0
SCI   0
SSU
XLST
SCPW  1234
SFLT  NO
CAC_MFC 0
CLS   UNR FBD WTA LPR MTD FNA HTA TDD HFD CRPD
      MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
      POD SLKD CCSD SWD LND CNDA
      CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
      ICDD CDMD LLCN MCTD CLBD AUTU
      GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
      CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
```

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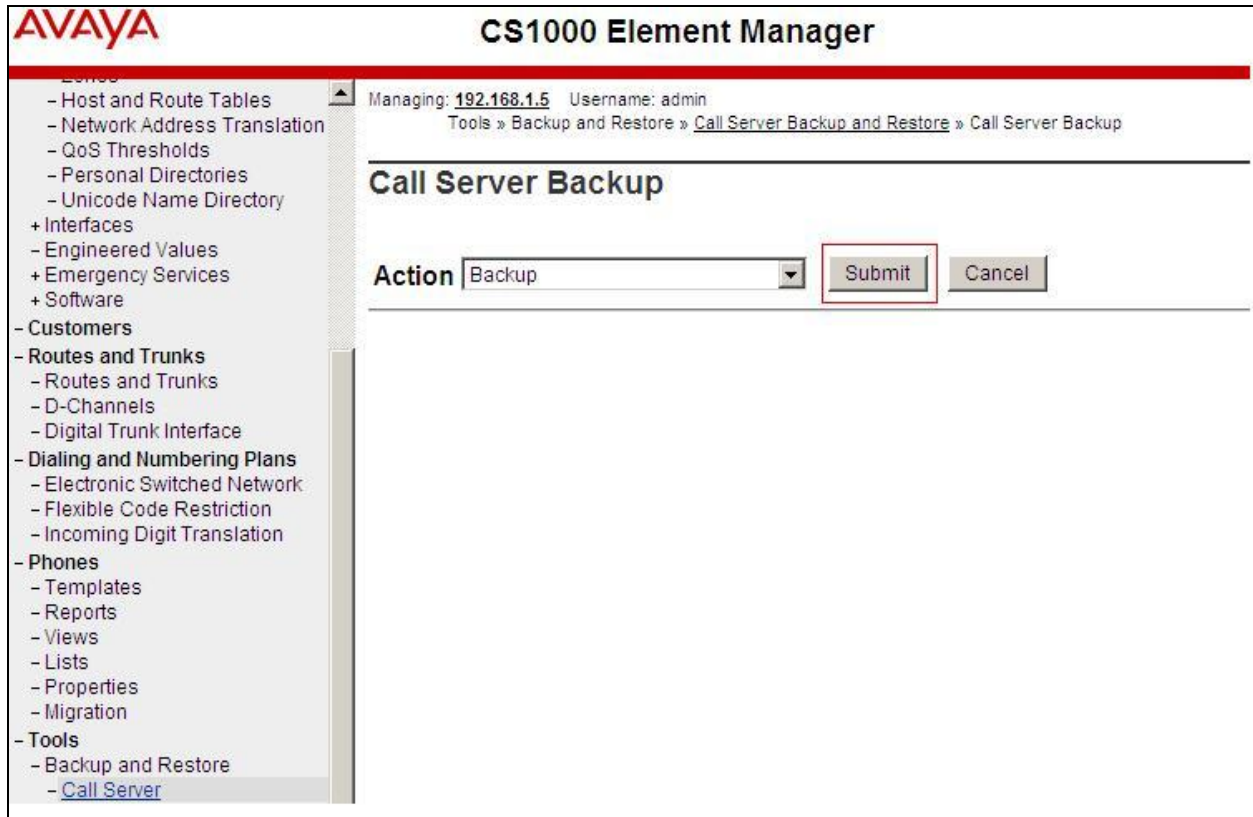
```

      UDI RCC HBTB AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
      DRDD EXRO
      USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
      FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD MSNV FRA  PKCH MWTD DVLD
CROD CROD
CPND_LANG ENG
RCO 0
HUNT
LHK 0
PLEV 02
PUID
DANI NO
AST
IAPG 0 *

AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 5003 0      MARP
      CPND
      CPND_LANG ROMAN
      NAME Sigma 1140
      XPLN 11
      DISPLAY_FMT FIRST, LAST*
01 HOT U 115003 MARP 0
02
03
04
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23 *
24 PRS
25 CHG
26 CPN
27
28
29
30
31
```

5.11. Save Configuration

Expand **Tools** → **Backup and Restore** on the left navigation panel and select **Call Server**. Select **Backup** and click **Submit** to save configuration changes as shown below.



The screenshot shows the AVAYA CS1000 Element Manager web interface. On the left is a navigation tree with categories like Services, Interfaces, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, and Tools. The 'Tools' category is expanded, showing 'Backup and Restore' and 'Call Server'. The main content area is titled 'Call Server Backup'. At the top of this area, it says 'Managing: 192.168.1.5 Username: admin' and 'Tools » Backup and Restore » Call Server Backup and Restore » Call Server Backup'. Below this, there is a section labeled 'Action' with a dropdown menu set to 'Backup'. To the right of the dropdown are two buttons: 'Submit' and 'Cancel'. The 'Submit' button is highlighted with a red rectangle.

Backup process will take several minutes to complete. Scroll to the bottom of the page to verify the backup process completed successfully as shown below.

```
Backing up reten.bkp to "/var/opt/nortel/cs/fs/cf2/backup/single"
Database backup Complete!
TEMU207
Backup process to local Removable Media Device ended successfully.
```

Configuration of CS1000E is complete.

6. Configure Avaya Aura® Session Manager

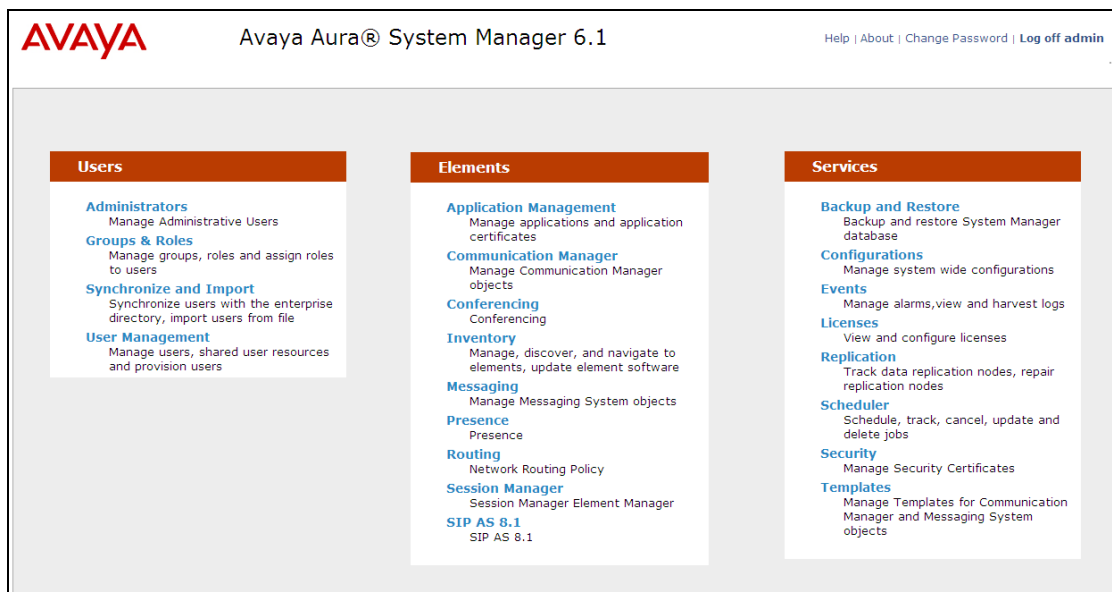
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- SIP Entities corresponding to CS1000E, Avaya SBCE and Session Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager Instance, corresponding to the Session Manager server to be administered in System Manager

It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, Locations, SIP Entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

6.1. Avaya Aura® System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL <https://<ip-address>/SMGR>, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials and click on **Log On** (not shown). The screen shown below is then displayed.



Most of the configuration items are performed in the Routing Element. Click on **Routing** in the Elements column shown above to bring up the Introduction to Network Routing Policy screen.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home /Elements / Routing- Introduction to Network Routing Policy

Introduction to Network Routing Policy

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

- Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Adaptations"
- Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)

6.2. Define SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. Expand **Elements** → **Routing** and select **Domains** from the left navigation menu, click **New** (not shown). Enter the following values and use default values for remaining fields.

- **Name** Enter the Domain name specified for the SIP Gateway in **Section 5.4**. In the sample configuration, in this case, **avaya.com** was used
- **Type** Verify **SIP** is selected
- **Notes** Add a brief description (Optional)

Click **Commit** to save. The screen below shows the SIP Domain defined for the sample configuration.

Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home /Elements / Routing / Domains- Domain Management

Domain Management

Commit Cancel

1 Item Refresh Filter: Enable

Name	Type	Default	Notes
* avaya.com	sip	<input type="checkbox"/>	

6.3. Define Location for Avaya Communication Server 1000E

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a Location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name for the location
- **Notes:** Add a brief description (optional)

The Location Pattern is used to identify call routing based on IP address. Session Manager matches the IP address against the patterns defined in this section. If a call is from a SIP Entity that does not match the IP address pattern then Session Manager uses the Location administered for the SIP Entity.

In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern** Enter the logical pattern used to identify the location. For the sample configuration, **10.10.3.*** was used
- **Notes** Add a brief description [Optional]

Click **Commit** to save. The screenshot below shows the Location defined for CS1000E in the sample configuration.

The screenshot displays the 'Location Details' configuration page for the Avaya Communication Server 1000E. The page is divided into two main sections: 'General' and 'Location Pattern'.

General Section:

- Name:** SMGRVL3
- Notes:** (Empty text box)
- Overall Managed Bandwidth:**
 - Managed Bandwidth Units:** Kbit/sec
 - Total Bandwidth:** (Empty text box)
 - Multimedia Bandwidth:** (Empty text box)
 - Audio Calls Can Take Multimedia Bandwidth:** ☒
- Per-Call Bandwidth Parameters:**
 - Maximum Multimedia Bandwidth (Intra-Location):** 1000 Kbit/Sec
 - Maximum Multimedia Bandwidth (Inter-Location):** 1000 Kbit/Sec
 - Minimum Multimedia Bandwidth:** 64 Kbit/Sec
 - Default Audio Bandwidth:** 80 Kbit/sec

Location Pattern Section:

- Add** and **Remove** buttons are present.
- A table lists the defined location patterns:

IP Address Pattern	Notes
* 10.10.3.*	

Below the table, there is a 'Select' dropdown menu with options 'All' and 'None'.

At the bottom of the page, there is a *** Input Required** message and **Commit** and **Cancel** buttons.

6.4. Configure Adaptation Module

To enable calls to be routed to stations on CS1000E, the Session Manager should be configured to modify the called party number to meet network requirements. Expand **Elements** → **Routing** and select **Adaptations** from the left navigational menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Adaptation Name** Enter an identifier for the Adaptation Module
- **Module Name** Select **DigitConversionAdapter** from drop-down menu
- **Module parameter** **fromto=true** → Modifies from and to headers of a message
MIME=no → Strips MIME message bodies on egress from Session Manager

Home / Elements / Routing / Adaptations - Adaptation Details

Adaptation Details

Help ?

Commit Cancel

General

* Adaptation name: Phonect

Module name: DigitConversionAdapter

Module parameter: fromto=true MIME=no

Egress URI Parameters:

Notes:

In the **Digit Conversion for Incoming Calls to SM** section, click **Add** and enter the following values.

- **Matching Pattern** Enter dialed prefix for calls to SIP endpoints registered to Session Manager. In the sample configuration, **47** was used
- **Min** Enter minimum number of digits that must be dialed. In the sample configuration, **2** was used
- **Max** Enter maximum number of digits that may be dialed. In the sample configuration, the default **36** was used
- **Delete Digits** Enter number of digits that may be deleted
- **Address to modify** Select **both**

Digit Conversion for Incoming Calls to SM

Add Remove

1 Item Refresh Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
<input type="checkbox"/>	*47	*2	*36		*0		both	

Select : All, None

In the **Digit Conversion for Outgoing Calls to SM** section, click **Add** and enter the following values.

- **Matching Pattern** Enter dialed prefix for calls to SIP endpoints registered to Session Manager. In the sample configuration, **5000** was used
- **Min** Enter minimum number of digits that must be dialed. In the sample configuration, **4** was used
- **Max** Enter maximum number of digits that may be dialed. In the sample configuration, **4** was used
- **Delete Digits** Enter number of digits that may be deleted. In the sample configuration, **4** was used
- **Insert Digits** Enter number of digits to be added before the dialed number. In the sample configuration, **4721xxxxxx** was used as this number required to be presented as CLID on outgoing calls
- **Address to Modify** Select **both**

Digit Conversion for Outgoing Calls from SM

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
<input type="checkbox"/>	*5000	*4	*4		*4	4721xxxxxx	both	

Select : All, None

6.5. Define SIP Entities

A SIP Entity must be added for Session Manager and for each SIP server connected to it, which includes CS1000E and Avaya SBCE. Navigate to **Routing → SIP Entities** in the left-hand navigation pane and click on the **New** button in the right pane (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signalling
- **Type:** Enter **Session Manager** for Session Manager, **Other** for CS1000E and **Gateway** for Avaya SBCE
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the **Adaptation Name** that will be applied to this entity
- **Location:** Select one of the locations defined previously
- **Time Zone:** Select the time zone for the location above

In the **SIP Link Monitoring** section:

- **SIP Link Monitoring** Select **Use Session Manager**

The following screen shows the addition of Session Manager. The IP address of the Session Manager signaling interface is entered for **FQDN or IP Address**.

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details Commit

General

* Name: Session Manager

* FQDN or IP Address: 10.10.3.55

Type: Session Manager

Notes:

Location: SMGRVL3

Outbound Proxy:

Time Zone: Europe/Dublin

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities. This section defines a default set of ports that Session Manager will use to listen for SIP requests, typically from registered SIP endpoints. Session Manager can also listen on additional ports defined elsewhere such as the ports specified in the SIP Entity Link definition in **Section 6.6**. In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **Port:** Port number on which Session Manager can listen for requests
- **Protocol:** Transport protocol to be used to send SIP requests
- **Default Domain:** The domain used for the enterprise

Defaults can be used for the remaining fields. Click **Commit** to save.

For the compliance test, three **Port** entries were added. The TCP and UDP ports were used by Session Manager in the reference configuration.

Port

Add

Remove

3 Items

Refresh

Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	
<input type="checkbox"/>	5060	UDP	avaya.com	
<input type="checkbox"/>	5061	TLS	avaya.com	

Select : All, None

* Input Required

Commit

Cancel

In order for Session Manager to send SIP service provider traffic on a separate Entity Link to CS1000E and Avaya SBCE, a new SIP entity is created separate from the one created at Session Manager installation for use with all other SIP traffic.

The following screen shows the addition of CS1000E SIP Entity. The **FQDN or IP Address** field is set to the TLAN Node IP address defined in **Section 5.4**.

The screenshot shows the 'SIP Entity Details' configuration page for a new entity named 'Avaya SBCE'. The page is titled 'Home / Elements / Routing / SIP Entities'. On the left, there are tabs for 'General' (selected) and 'SIP Link Monitoring'. The main form area contains the following fields and options:

- Name:** Avaya SBCE
- * FQDN or IP Address:** 10.10.3.30
- Type:** Gateway (dropdown menu)
- Notes:** (empty text field)
- Adaptation:** Phonect (dropdown menu)
- Location:** SMGRVL3 (dropdown menu)
- Time Zone:** Europe/Dublin (dropdown menu)
- Override Port & Transport with DNS SRV:** ☐
- * SIP Timer B/F (in seconds):** 4
- Credential name:** (empty text field)
- Call Detail Recording:** none (dropdown menu)
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown menu)

At the top right of the form area, there are 'Commit' and 'Cancel' buttons, and a 'Help ?' link.

The following screen shows the addition of Avaya SBCE SIP Entity. The **FQDN or IP Address** field is set to the IP address of its private network interface.

6.6. Define Entity Links

A SIP trunk between Session Manager and a telephony system is described as an Entity Link. Two Entity Links were created; one to CS1000E for use only by service provider traffic and one to Avaya SBCE. To add an Entity Link, navigate to **Routing → Entity Links** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name
- **SIP Entity 1:** Select the SIP Entity for Session Manager
- **Protocol:** Select the transport protocol used for this link
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. Default listen port is **5060**
- **SIP Entity 2:** Select the name of the other system. Select the CS1000E or Avaya SBCE defined in **Section 6.5**
- **Port:** Port number on which the other system receives SIP requests from the Session Manager. Default listen port is **5060**
- **Trusted:** Check this box. **Note:** If this box is not checked, calls from the associated SIP Entity specified in **Section 6.5** will be denied

Click **Commit** to save. The following screens illustrate the Entity Links to CS1000E and Avaya SBCE.

Entity Link to CS1000E.

Home / Elements / Routing / Entity Links - Entity Links

Entity Links Help ? Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* CS1K	* Session Manager	TCP	* 5060	* CS1K	* 5060	Trusted	toCS1K

* Input Required Commit Cancel

Entity Link to Avaya SBCE.

Home / Elements / Routing / Entity Links - Entity Links

Entity Links Help ? Commit Cancel

1 Item Refresh

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* Sipera	* Session Manager	TCP	* 5060	* Sipera	* 5060	Trusted	toSipera

* Input Required Commit Cancel

6.7. Define Routing Policies

Routing Policies describe the conditions under which calls will be routed to CS1000E from either SIP endpoint registered to Session Manager or from other telephony system. It also describes the routing policies for which calls will be routed to the Avaya SBCE and therefore to Phonects SIP network. To add a Routing Policy, Expand **Elements** → **Routing** and select **Routing Policies**. Click **New** (not shown).

In the **General** section, enter the following values.

- **Name** Enter an identifier to define the routing policy
- **Disabled** Leave unchecked
- **Notes** Enter a brief description (Optional)

In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). For Routing Policy to the CS1000E, select the SIP Entity associated with CS1000E defined in **Section 6.5** and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition.

Note: The Routing Policy defined in this section is an example and was used in the sample configuration. Other Routing Policies may be appropriate for different customer networks.

The following screenshot shows the Routing Policy for CS1000E.

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details Help ? Commit Cancel

General

* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CS1K	10.10.3.6	Other	

For Routing Policy to the Avaya SBCE – Phonects SIP Trunk, select the SIP Entity associated with Avaya SBCE defined in **Section 6.5** and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition.

The following screenshot shows the Routing Policy for Avaya SBCE – Phonect SIP Trunk.

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details Help ? Commit Cancel

General

* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Sipera	10.10.3.30	Gateway	

6.8. Define Dial Patterns

Dial Patterns are needed to route calls through Session Manager. For the compliance test, dial patterns were needed to route calls from CS1000E to Phonect and vice versa. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call
- **Min:** Enter a minimum length used in the match criteria
- **Max:** Enter a maximum length used in the match criteria
- **SIP Domain:** Enter the destination domain used in the match criteria
- **Notes:** Add a brief description (optional)

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria.

- **Originating Locations** Select **ALL**
- **Routing Policies** Select the required Routing Policy defined in **Section 6.7**

Two examples of the dial patterns used for the compliance test are shown below. This Session Manager is shared between two test environments. The first example shows that minimum **2** digit dialed numbers that begin with **47** originating from **SMGRVL3** uses route policy **toCS1K**. This will allow DID numbers assigned to the enterprise from Phonect SIP Trunk Service to route to CS1000E.

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details

Commit Cancel Help ?

General

* Pattern: 47

* Min: 2

* Max: 36

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	SMGRVL3		toCS1K	0	<input type="checkbox"/>	CS1K	

Select : All, None

The second example shows that a minimum **5** digit dialed numbers that begin with **00353** originating from **SMGRVL3** uses route policy **toSipera**. This will allow outbound calls to route from the CS1000E to PSTN test numbers in the Avaya enterprise lab.

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details Help ? Commit Cancel

General

* Pattern: 00353

* Min: 5

* Max: 36

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	SMGRVL3		toSipera	0	<input type="checkbox"/>	Sipera	

Select : All, None

6.9. Verify Avaya Aura® Session Manager Instance

The creation of a Session Manager Instance provides the linkage between System Manager and Session Manager. This was most likely done as part of the initial Session Manager installation. To add a Session Manager, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane and click on the **new** button in the right pane (not shown). If the Session Manager instance already exists, click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the following screen:

In the **General** section, enter the following values:

- **SIP Entity Name:** Select the SIP Entity created for Session Manager
- **Description:** Add a brief description (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface

The following screen shows the Session Manager values used for the compliance test.

Home / Elements / Session Manager / Session Manager Administration - Session Manager Administration

Help ?

View Session Manager

Return

General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server |
Expand All | Collapse All

General ▾

SIP Entity Name Session Manager

Description Session Manager

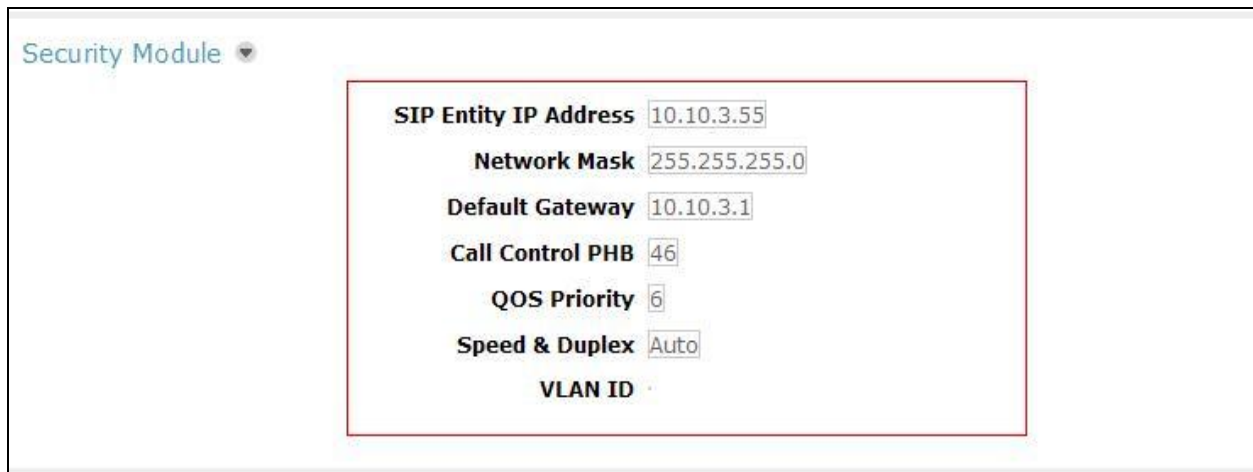
Management Access Point Host Name/IP 10.10.3.54

Direct Routing to Endpoints Enable

In the **Security Module** section, enter the following values:

- **SIP Entity IP Address:** Should be filled in automatically based on the SIP Entity Name. Otherwise, enter IP address of Session Manager signaling interface
- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager

Use default values for the remaining fields. Click **Save** (not shown) to add this Session Manager. The following screen shows the remaining Session Manager values used for the compliance test.



The screenshot displays the 'Security Module' configuration page. A red rectangular box highlights the following fields and their values:

Field	Value
SIP Entity IP Address	10.10.3.55
Network Mask	255.255.255.0
Default Gateway	10.10.3.1
Call Control PHB	46
QOS Priority	6
Speed & Duplex	Auto
VLAN ID	

7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Session Border Controller. The Avaya SBCE is administered using the UC-Sec Control Center.

7.1. Access Avaya Session Border Controller for Enterprise

Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. Select the **UC-Sec Control Center**.



Log in with the appropriate credentials. Click **Sign In**.



The main page of the UC-Sec Control Center will appear.

To view system information that was configured during installation, navigate to **UC-Sec Control Center → System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named Sipera is shown. To view the configuration of this device, click the monitor icon (the third icon from the right).

The **System Information** screen shows the **General Settings**, **Network Settings**, **DNS Configuration** and **Management IP** information provided during installation. The **Box Type** was set to **SIP** and the **Deployment Mode** was set to **Proxy**. Default values were used for all other fields.

System Information: GSSCP_03 ✕

Network Configuration

General Settings

Appliance Name	GSSCP_03
Box Type	SIP
Deployment Mode	Proxy

Device Settings

HA Mode	NO
Secure Channel Mode	NONE
Two Bypass Mode	NO

Network Settings

IP	Public IP	Netmask	Gateway	Interface
10.10.3.30	10.10.3.30	255.255.255.0	10.10.3.1	A1
86.47.xxx.xxx	86.47.xxx.xxx	255.255.255.128	86.47.xxx.xxx	B1

DNS Configuration

Primary DNS	8.8.8.8
Secondary DNS	
DNS Location	DMZ
DNS Client IP	86.47.xxx.xxx

Management IP(s)

IP	10.10.2.55
----	------------

7.2. Global Profiles

When selected, Global Profiles allows for configuration of parameters across all UC-Sec appliances.

7.2.1. Server Interworking - Avaya Side

Server Interworking configures and manages various SIP call server specific capabilities such as call hold and T.38. Navigate to **Global Profiles** → **Server Interworking** and click on **Add Profile** (Not Shown).

- Enter a profile name such as **SM3_CS** and click **Next** (Not Shown)
- **Check Hold Support= RFC2543**
- **Check Delayed SDP Handling**
- **Check T.38 Support**
- All other options on the General Tab can be left at default

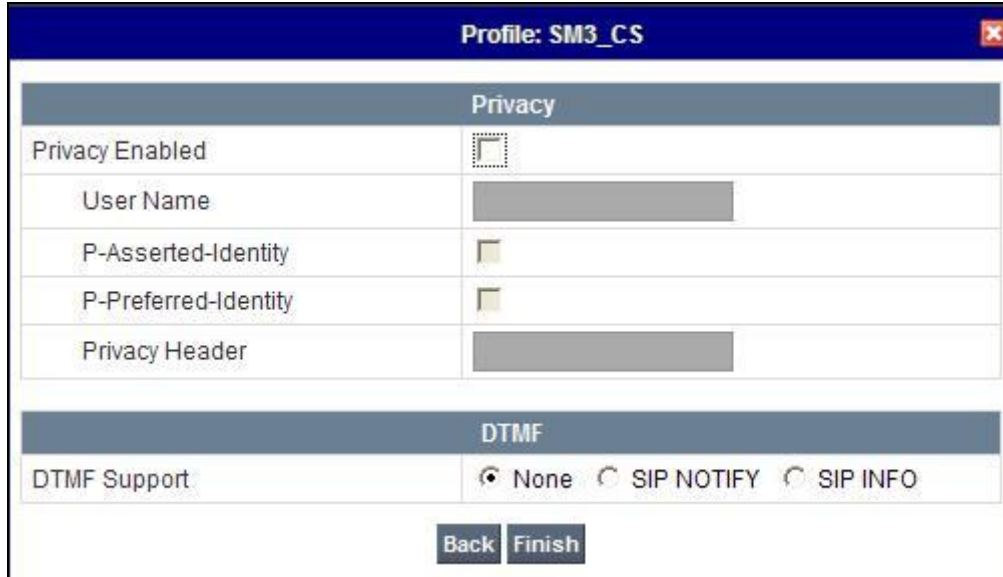
Click **Next** to continue.

Profile: SM3_CS

General	
Hold Support	<input type="radio"/> None <input checked="" type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input checked="" type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Next

Default values can be used for the next window that appears. Click **Finish**.



The screenshot shows a configuration window titled "Profile: SM3_CS". It contains two sections: "Privacy" and "DTMF".

Privacy	
Privacy Enabled	<input type="checkbox"/>
User Name	<input type="text"/>
P-Asserted-Identity	<input type="text"/>
P-Preferred-Identity	<input type="text"/>
Privacy Header	<input type="text"/>

DTMF	
DTMF Support	<input checked="" type="radio"/> None <input type="radio"/> SIP NOTIFY <input type="radio"/> SIP INFO

At the bottom, there are two buttons: "Back" and "Finish".

Default values can be used for the **Advanced Settings** window. Click **Finish**.



The screenshot shows a configuration window titled "Profile: SM3_CS". It contains an "Advanced Settings" section with various options.

Advanced Settings	
Record Routes	<input checked="" type="radio"/> None <input type="radio"/> Single Side <input type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
SLIC Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>

At the bottom, there is a "Finish" button.

7.2.2. Server Interworking – Phonect Side

Server Interworking configures and manages various SIP call server specific capabilities such as call hold and T.38. Navigate to **Global Profiles** → **Server Interworking** and click on **Add Profile** (Not Shown).

- Enter profile name: **SP_Trunk** and click **Next** (Not Shown)
- **Check Hold Support= RFC2543**
- **Check Delayed SDP Handling**
- **Check T.38 Support**
- All other options on the **General** tab can be left at default

Click **Next** to continue.

Profile: SP_Trunk	
General	
Hold Support	<input type="radio"/> None <input checked="" type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input checked="" type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Next

Default values can be used for the next window that appears. Click **Finish**.

The screenshot shows a configuration window titled "Profile: SP_Trunk". It contains two sections: "Privacy" and "DTMF".

Privacy	
Privacy Enabled	<input type="checkbox"/>
User Name	<input type="text"/>
P-Asserted-Identity	<input type="checkbox"/>
P-Preferred-Identity	<input type="checkbox"/>
Privacy Header	<input type="text"/>

DTMF	
DTMF Support	<input checked="" type="radio"/> None <input type="radio"/> SIP NOTIFY <input type="radio"/> SIP INFO

At the bottom, there are two buttons: "Back" and "Finish".

Default values can be used for the **Advanced Settings** window. Click **Finish**.

The screenshot shows a configuration window titled "Profile: SP_Trunk". It contains an "Advanced Settings" section with various options.

Advanced Settings	
Record Routes	<input checked="" type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
SLIC Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>

At the bottom, there is a "Finish" button.

7.2.3. Routing

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, server addresses and resolution methods, next hop routing information, and packet transport types.

Create a Routing Profile for Session Manager and a Routing Profile for Phonet SIP Trunk. To add a routing profile, navigate to **UC-Sec Control Center → Global Profiles → Routing** and select **Add Profile**. Enter a **Profile Name** and click **Next (Not Shown)** to continue.

In the new window that appears, enter the following values. Use default values for all remaining fields:

- **URI Group:** Select “*” from the drop down box
- **Next Hop Server 1:** Enter the Domain Name or IP address of the Primary Next Hop server
- **Next Hop Server 2:** (Optional) Enter the Domain Name or IP address of the secondary Next Hop server
- **Routing Priority Based on Next Hop Server:** Checked
- **Use Next Hop for In-Dialog Messages:** Select only if there is no secondary Next Hopserver
- **Outgoing Transport:** Choose the protocol used for transporting outgoing signaling packets

Click **Finish**.

The following screen shows the Routing Profile to Session Manager. The **Next Hop Server 1** IP address must match the IP address of the Session Manager Security Module in **Section 6.9**. The Outgoing Transport and port number must match the Avaya SBCE Entity Link created on Session Manager in **Section 6.6**.

Global Profiles > Routing: Call Server

Add Profile **Rename Profile** **Clone Profile** **Delete Profile**

Click here to add a description.

Routing Profile

Add Routing Rule

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	10.10.3.55	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	TCP

The following screen shows the Routing Profile to Phonect.

Global Profiles > Routing: Trunk Server

[Add Profile](#) [Rename Profile](#) [Clone Profile](#) [Delete Profile](#)

Routing Profiles

default


Call Server

Trunk Server

Click here to add a description.

Routing Profile

[Add Routing Rule](#)

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport	
1	*	sip-101.xxxxxx.net	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	UDP	

7.2.4. Server - Configuration

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs configure and manage various SIP call server specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling parameters and some advanced options.

7.2.4.1 Server - Configuration – Avaya Side

To add a Server Configuration Profile for Session Manager navigate to **UC-Sec Control Center** → **Global Profiles** → **Server Configuration** and click on **Add Profile** (not shown). In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Server Type:** Select **Call Server** from the drop-down box
- **IP Addresses / Supported FQDNs:** Enter the IP address of the Session Manager signaling interface. This should match the IP address of the Session Manager Security Module in **Section 6.9**
- **Supported Transports:** Select the transport protocol used to create the Avaya SBCE Entity Link on Session Manager in **Section 6.6**
- **TCP and UDP Port:** Port number on which to send SIP requests to Session Manager. This should match the port number used in the Avaya SBCE Entity Link on Session Manager in **Section 6.6**

Click **Finish** to continue.

The screenshot shows a window titled "Server Configuration Profile - General". It contains several fields for configuration:

- Server Type:** A dropdown menu with "Call Server" selected.
- IP Addresses / Supported FQDNs:** A text area with "10.10.3.55" entered. Below the text area is the label "Comma seperated list".
- Supported Transports:** Three checkboxes: "TCP" (checked), "UDP" (checked), and "TLS" (unchecked).
- TCP Port:** A text box with "5060" entered.
- UDP Port:** A text box with "5060" entered.
- TLS Port:** A greyed-out text box.
- Finish:** A button at the bottom of the window.

In the new window that appears, verify **Enable Authentication** is unchecked as Session Manager does not require authentication. Click **Finish**.



Server Configuration Profile - Authentication	
Enable Authentication	<input type="checkbox"/>
User Name	<input type="text"/>
Realm	<input type="text"/>
Password	<input type="password"/>
Confirm Password	<input type="password"/>
Finish	

In the new window that appears, select the **Interworking Profile** created for the enterprise in **Section 7.2.1**. Use default values for all remaining fields. Click **Finish** to save the configuration.



Configuration Profile - Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SM3_CS
Signaling Manipulation Script	None
TCP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
UDP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
Finish	

7.2.4.2 Server - Configuration - Phonect

To add a Server Configuration Profile for Session Manager navigate to **UC-Sec Control Center** → **Global Profiles** → **Server Configuration** and click on **Add Profile** (not shown). In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Server Type:** Select **Trunk Server** from the drop-down box
- **IP Addresses / Supported FQDNs:** Enter the IP address(es) of the SIP proxy(ies) of the service provider. This will associate the inbound SIP messages from Phonect to this Server Configuration
- **Supported Transports:** Select the transport protocol to be used for SIP traffic between Avaya SBCE and Phonect
- **UDP Port:** Enter the port number that Phonect uses to send SIP traffic

Click **Finish** to continue.

The screenshot shows a window titled "Server Configuration Profile - General" with a close button in the top right corner. The window contains several configuration fields:

- Server Type:** A dropdown menu with "Trunk Server" selected.
- IP Addresses / Supported FQDNs:** A text area with "sip-101.xxxxxxx.net" entered. Below the text area is a label "Comma seperated list".
- Supported Transports:** Three checkboxes: "TCP" (unchecked), "UDP" (checked), and "TLS" (unchecked).
- TCP Port:** A text box that is currently empty.
- UDP Port:** A text box with "5060" entered.
- TLS Port:** A text box that is currently empty.

At the bottom center of the window is a "Finish" button.

In the new window that appears, enter the following values as Phonect require authentication to connect to their network:

- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider
- **Realm:** Enter realm details provided by the Service Provider
- **Password** Enter password provided by the Service Provider
- **Confirm Password** Re-enter password provided by the Service Provider

Click **Finish** to continue.

Server Configuration Profile - Authentication	
Enable Authentication	<input checked="" type="checkbox"/>
User Name	xxxxxxxxxx
Realm (Leave blank to detect from server challenge)	sip-101.xxxxxx.net
Password	
Confirm Password	
Finish	

In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Enabled Heartbeat:** Checked
- **Method:** Select **REGISTER** from the drop-down box
- **Frequency:** Choose the desired frequency in seconds the Avaya SBCE will send SIP REGISTERS. **Note:** Phonect recommends a frequency range between 600-3600 for this parameter
- **From URI:** Enter an URI to be sent in the FROM header for SIP REGISTERS
- **TO URI:** Enter an URI to be sent in the TO header for SIP REGISTERS

Click **Next** to continue.

Server Configuration Profile - Heartbeat	
Enable Heartbeat	<input checked="" type="checkbox"/>
Method	REGISTER
Frequency	800 seconds
From URI	xxxxxx@sip-101.xxxxx.n
To URI	xxxxxx@sip-101.xxxxx.n
TCP Probe	<input type="checkbox"/>
TCP Probe Frequency	seconds
Finish	

In the new window that appears, select the **Interworking Profile** created for Phonect in **Section 7.2.2**. Use default values for all remaining fields. Click **Finish** to save the configuration.

Configuration Profile - Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SP_Trunk
Signaling Manipulation Script	None
UDP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
Finish	

7.2.5. Topology Hiding – Avaya Side

The **Topology Hiding** screen allows you to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks. From the left-hand menu select **Global Profiles → Topology Hiding** (not shown).

- Click **default** profile and select **Clone Profile** (not shown)
- Enter Profile Name : **SM3_CS**
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For **Override Value** type **avaya.com**
- Click **Finish** (not shown)

The screen below is a result of the details configured above.

Global Profiles > Topology Hiding: SM3_CS

Buttons: Add Profile, Rename Profile, Clone Profile, Delete Profile

Click here to add a description.

Topology Hiding Profiles

- default
- cisco_th_profile
- SM3_CS**
- SP_Trunk

Topology Hiding

Header	Criteria	Replace Action	Override Value
Record-Route	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	avaya.com
To	IP/Domain	Overwrite	avaya.com
Via	IP/Domain	Auto	---
From	IP/Domain	Overwrite	avaya.com
SDP	IP/Domain	Auto	---

Edit

7.2.6. Topology Hiding – Phonect Side

The **Topology Hiding** screen allows you to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks. From the left-hand menu select **Global Profiles → Topology Hiding** (not shown).

- Click **default** profile and select **Clone Profile** (not shown)
- Enter Profile Name : **SP_Trunk**
- For the Headers Request Line, **To**, and **From** select **IP/Domain** under **Criteria** and **Next Hop** under **Replace Action**
- Click **Finish** (not shown)

The screen below is a result of the details configured above.

Global Profiles > Topology Hiding: SP_Trunk

Buttons: Add Profile, Rename Profile, Clone Profile, Delete Profile

Topology Hiding Profiles:

- default
- cisco_th_profile
- SM3_CS
- SP_Trunk**

Click here to add a description.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
Record-Route	IP/Domain	Auto	---
Request-Line	IP/Domain	Next Hop	---
To	IP/Domain	Next Hop	---
Via	IP/Domain	Auto	---
From	IP/Domain	Next Hop	---
SDP	IP/Domain	Auto	---

Edit

7.3. Device Specific Settings

The Device Specific Settings feature allows aggregate system information to be viewed, and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network.

7.3.1. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of the Avaya SBCE, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc. to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency.

Navigate to **UC-Sec Control Center → Device Specific Settings → Network Management** and verify the IP addresses assigned to the interfaces and that the interfaces are enabled. The following screen shows the private interface is assigned to **A1** and the external interface is assigned to **B1**.

Device Specific Settings > Network Management: GSSCP_03

UC-Sec Devices
GSSCP_03

Network Configuration | Interface Configuration

Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from [System Management](#).

A1 Netmask: 255.255.255.0
A2 Netmask: [Disabled]
B1 Netmask: 255.255.255.128
B2 Netmask: [Disabled]

Add IP | Save Changes | Clear Changes

IP Address	Public IP	Gateway	Interface	
10.10.3.30		10.10.3.1	A1	X
86.47.xxx.xxx		86.47.xxx.xxx	B1	X

Select the **Interface Configuration** Tab and use the **Toggle State** button to enable the interfaces.

ent: GSSCP_03

Network Configuration | Interface Configuration

Name	Administrative Status	
A1	Enabled	Toggle State
A2	Disabled	Toggle State
B1	Enabled	Toggle State
B2	Disabled	Toggle State

7.3.2. Media Interface

The Media Interface screen allows the IP Address and ports to be set for transporting Media over the SIP trunk. The Avaya SBCE listens for SIP media on the defined ports. To create a new Media Interface, navigate to **UC-Sec Control Center → Device Specific Settings → Media Interface** and click **Add Media Interface**.

- Select **Add Media Interface**
- **Name: Int_Media**
- **Media IP: 10.10.3.30** (Internal Address for calls toward CS1000E)
- **Port Range: 35000-40000**
- Click **Finish** (not shown)
- Select **Add Media Interface**
- **Name: Ext_Media**
- **Media IP: 86.47.xxx.xxx** (External Address for calls toward Phonect)
- **Port Range: 35000-40000**
- Click **Finish** (not shown)

The following screen shows the media interfaces created in the sample configuration for the inside and outside IP interfaces. After the media interfaces are created, an application restart is necessary before the changes will take effect.

Note: An application restart can be performed within System Management by clicking the application restart icon (**not shown**).

Device Specific Settings > Media Interface: GSSCP_03

UC-Sec Devices
GSSCP_03

Media Interface

Modifying or deleting an existing media interface will require an application restart before taking effect.
Application restarts can be issued from [System Management](#).

Add Media Interface

Name	Media IP	Port Range		
Int_Media	10.10.3.30	35000 - 40000		
Ext_Media	192.168.102.2	35000 - 40000		

7.3.3. Signalling Interface

The Signalling Interface screen allows the IP Address and ports to be set for transporting Media over the SIP trunk. The Avaya SBCE listens for SIP requests on the defined ports. Create a Signaling Interface for both the inside and outside IP interfaces. To create a new Signaling Interface, navigate to **UC-Sec Control Center → Device Specific Settings → Signaling Interface** and click **Add Signaling Interface**.

- **Name: Int_Sig**
- **Signaling IP: 10.10.3.30** (Internal Address for calls toward CS1000E)
- **TCP Port: 5060**
- **UDP Port: 5060**
- Click **Finish**
- Select **Add Signaling Interface**
- **Name: Ext_Sig**
- **Signaling IP: 86.47.xxx.xxx** (External Address for calls toward Phonetx)
- **TCP Port: 5060**
- **UDP Port: 5060**
- Click **Finish**

The following screen shows the signaling interfaces created in the sample configuration for the inside and outside IP interfaces.

Device Specific Settings > Signaling Interface: GSSCP_03

UC-Sec Devices	
GSSCP_03	

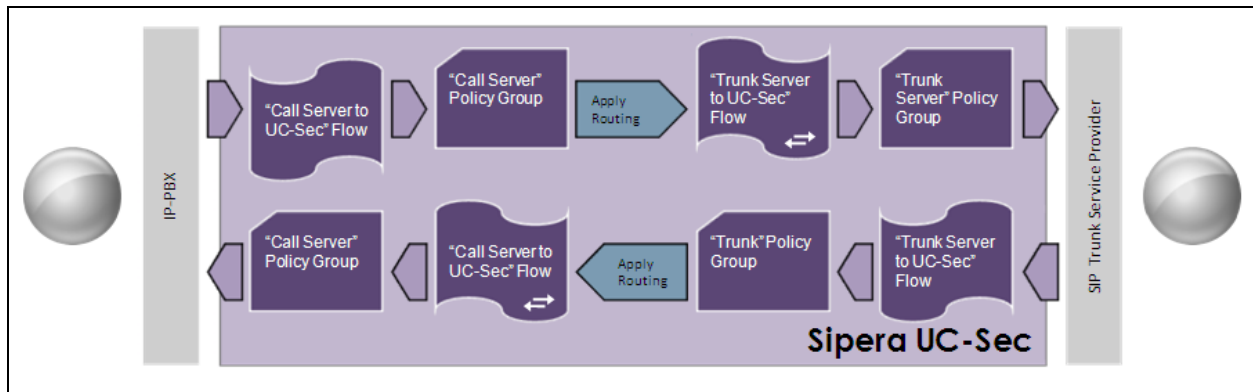
Signaling Interface

Add Signaling Interface

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile		
Int_Sig	10.10.3.30	5060	5060	---	None		
Ext_Sig	86.47.xxx.xxx	5060	5060	---	None		

7.3.4. End Point Flows

When a packet is received by UC-Sec, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.



To create a Server Flow, navigate to **UC-Sec Control Center → Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add Flow**.

- **Flow Name:** Enter a descriptive name
- **Server Configuration:** Select a Server Configuration created in **Section 7.1.5** to assign to the Flow
- **Received Interface:** Select the Signaling Interface the Server Configuration is allowed to receive SIP messages from
- **Signaling Interface:** Select the Signaling Interface used to communicate with the Server Configuration
- **Media Interface:** Select the Media Interface used to communicate with the Server Configuration.
- **End Point Policy Group:** Select the policy assigned to the Server Configuration
- **Routing Profile:** Select the profile the Server Configuration will use to route SIP messages to
- **Topology Hiding Profile:** Select the profile to apply toward the Server Configuration

Click **Finish** to save and exit.

The following screen shows the Sever Flow for Session Manager.

Flow: SM3_Call_Server	
Criteria	
Flow Name	SM3_Call_Server
Server Configuration	SM3_Call_Server
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Ext_Sig
Signaling Interface	Int_Sig
Media Interface	Int_Media
End Point Policy Group	Belgacom
Routing Profile	Trunk Server
Topology Hiding Profile	SM3_CS
File Transfer Profile	None
Finish	

The following screen shows the Sever Flow for Phonect.

Flow: SP_Trunk_Server	
Criteria	
Flow Name	SP_Trunk_Server
Server Configuration	SP_Trunk_Server
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Int_Sig
Signaling Interface	Ext_Sig
Media Interface	Ext_Media
End Point Policy Group	Belgacom
Routing Profile	Call Server
Topology Hiding Profile	SP_Trunk
File Transfer Profile	None
Finish	

8. Phonect SIP Service Provider Configuration

The setup for the use of Phonect is by using the SIP trunk with an authenticated service. The configuration of Phonect's authentication service to support the SIP trunk service is outside of the scope for these Application Notes and will not be covered. To obtain further information on Phonect's equipment and system configuration please contact an authorised Phonect representative (see Section 2.3).

9. Verification

9.1. Verify Avaya Communication Server 1000E Operational Status

Expand **System** on the left navigation panel and select **Maintenance**. Select **LD 96 - D-Channel** from the **Select by Overlay** table and the **D-Channel Diagnostics** function from the **Select Group** table as shown below.

The screenshot displays the Avaya CS1000 Element Manager web interface. The top header shows the Avaya logo and the title "CS1000 Element Manager". Below the header, the navigation pane on the left lists various system components, with "Maintenance" selected under the "System" category. The main content area is titled "Maintenance" and contains two radio buttons: "Select by Overlay" (which is selected) and "Select by Functionality". A table titled "<Select by Overlay>" lists various LD (Line Description) entries, with "LD 96 - D-Channel" highlighted. To the right of this table is a smaller table titled "<Select Group>" which lists "D-Channel Diagnostics", "MSDL Diagnostics", and "TMDI Diagnostics".

<Select by Overlay>
LD 30 - Network and Signaling
LD 32 - Network and Peripheral Equipment
LD 34 - Tone and Digit Switch
LD 36 - Trunk
LD 37 - Input/Output
LD 38 - Conference Circuit
LD 39 - Intergroup Switch and System Clock
LD 45 - Background Signaling and Switching
LD 48 - Multifrequency Sender
LD 48 - Link
LD 54 - Multifrequency Signaling
LD 60 - Digital Trunk Interface and Primary Rate Interface
LD 75 - Digital Trunk
LD 80 - Call Trace
LD 96 - D-Channel
LD 117 - Ethernet and Alarm Management
LD 135 - Core Common Equipment
LD 137 - Core Input/Output
LD 143 - Centralized Software Upgrade

<Select Group>
D-Channel Diagnostics
MSDL Diagnostics
TMDI Diagnostics

Select **Status for D-Channel (STAT DCH)** command and click **Submit** to verify status of virtual D-Channel as shown below. Verify the status of the following fields.

- **APPL_STATUS** Verify status is **OPER**
- **LINK_STATUS** Verify status is **EST ACTV**

Managing: 192.168.1.5 Username: admin
System > Maintenance > D-Channel Diagnostics

D-Channel Diagnostics

Diagnostic Commands	Command Parameters	Action
Status for D-Channel (STAT DCH)		Submit
Disable Automatic Recovery (DIS AUTO)	<input type="checkbox"/> ALL	Submit
Enable Automatic Recovery (ENL AUTO)	<input type="checkbox"/> FDL	Submit
Test Interrupt Generation (TEST 100)		Submit
Establish D-Channel (EST DCH)		Submit

DCH|DES|APPL_STATUS|LINK_STATUS|AUTO_RECV|PDCH|BDCH

DCH	DES	APPL_STATUS	LINK_STATUS	AUTO_RECV	PDCH	BDCH
001	SIP_DCH	OPER	EST ACTV	AUTO		

STAT DCH

Command executed successfully.

9.2. Verify Avaya Aura® Session Manager Operational Status

9.2.1. Verify Avaya Aura® Session Manager is Operational

Navigate to **Elements → Session Manager → Dashboard** (not shown) to verify the overall system status for Session Manager. Specifically, verify the status of the following fields as shown below.

• **Tests Pass** ✓

• **Security Module** Up

• **Service State** Accept New Service

Home / Elements / Session Manager- Session Manager

Session Manager Dashboard

This page provides the overall status and health summary of each administered Session Manager.

Session Manager Instances

Service State: [Dropdown] Shutdown System: [Dropdown] As of 11:56 AM

Session Manager	Type	Alarms	Tests Pass	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Version
Session Manager	Core	0/0/2	Up	Up	Accept New Service	0/3	1	0	6.1.0.0.610023

Select : All, None

Navigate to **Elements → Session Manager → System Status → Security Module Status** (not shown) to view more detailed status information on the status of Security Module for the specific Session Manager. Verify the **Status** column displays **Up** as shown below.

Reset

Synchronize

Certificate Management ▾

Connection Status

1 Item

Refresh

Show

ALL ▾

Filter: Enable

	Details	Session Manager	Type	Status	Connections	IP Address	VLAN	Default Gateway	NIC Bonding	Entity Links (expected / actual)	Certificate Used
<div>C</div>	<div>► Show</div>	Session Manager	SM	Up	6	10.10.3.55/24	---	10.10.3.1	Disabled	3/3	SIP CA

Select : None

9.2.2. Verify SIP Entity Link Status

Navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring** (not shown) to view more detailed status information for one of the SIP Entity Links. Select the SIP Entity for CS1000E from the **All Monitored SIP Entities** table (not shown) to open the **SIP Entity, Entity Link Connection Status** page. In the **All Entity Links to SIP Entity: CS1K** table, verify the **Conn. Status** for the link is **Up** as shown below.

SIP Entity, Entity Link Connection Status							
This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.							
All Entity Links to SIP Entity: CS1K							
Summary View							
1 Item Refresh Filter: Enable							
Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
► Show	Session Manager	10.10.3.6	5060	TCP	Up	200 OK	Up

Verify the status of the SIP link is up between the Session Manager and the Avaya SBCE by going through the same process as outlined above but selecting the SIP Entity for the Avaya SBCE in the **All Monitored SIP Entity** table.

SIP Entity, Entity Link Connection Status							
This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.							
All Entity Links to SIP Entity: Sipera							
Summary View							
1 Item Refresh Filter: Enable							
Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
► Show	Session Manager	10.10.3.30	5060	TCP	Up	200 OK	Up

10. Conclusion

These Application Notes describe the configuration necessary to connect the Avaya CS1000E, Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise to Phonect SIP Service. Interoperability testing of the sample configuration was completed with successful results for the Phonect SIP Trunk with observations which are detailed in **Section 2.2**.

11. Additional References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] Avaya Aura® Session Manager Overview, Doc ID 03-603323, available at <http://support.avaya.com>
- [2] Installing and Configuring Avaya Aura® Session Manager, available at <http://support.avaya.com>
- [3] Avaya Aura® Session Manager Case Studies, available at <http://support.avaya.com>
- [4] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, available at <http://support.avaya.com>
- [5] Administering Avaya Aura® Session Manager, Doc ID 03-603324, available at <http://support.avaya.com>
- [6] IP Peer Networking Installation and Commissioning, Release 7.5, Document Number NN43001-313, available at <http://support.avaya.com>
- [7] Network Routing Service Fundamentals, Release 7.5, Document Number NN43001-130, Issue 03.02, available at <http://support.avaya.com>
- [8] Co-resident Call Server and Signaling Server Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-509, available at <http://support.avaya.com>
- [9] Signaling Server and IP Line Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-125, available at <http://support.avaya.com>
- [10] E-SBC (Avaya Session Border Controller for Enterprise) Administration Guide, November 2011
- [11] RFC 3261 SIP: Session Initiation Protocol, <http://www.ietf.org/>

Appendix A – Avaya Communication Server 1000E Software

Avaya Communication Server 1000E call server patches and plug ins

TID: 46379

VERSION 4121

System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

IPMGs Registered: 1
IPMGs Unregistered: 0
IPMGs Configured/unregistered: 0

RELEASE 7

ISSUE 50 Q +

IDLE_SET_DISPLAY NORTEL

DepList 1: core Issue: 01 ALTERED(created: 2012-03-14 13:55:18 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2012-03-28 11:15:04(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2012-03-27 06:55:16(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE

LOADWARE VERSION: PSWV 100

INSTALLED LOADWARE PEPS : 0

ENABLED PLUGINS : 2

PLUGIN	STATUS	PRS/CR_NUM	MPLR_NUM	DESCRIPTION
501	ENABLED	Q02138637	MPLR30070	Enables blind transfer to a SIP endpoint even if SIP UPDATE is not supported by the far end

Avaya Communication Server 1000E call server deplists

VERSION 4121

RELEASE 7

ISSUE 50 Q +

DepList 1: core Issue: 01 (created: 2012-03-14 13:55:18 (est)) ALTERED

IN-SERVICE PEPS

PAT#	CR #	PATCH REF #	NAME	DATE	FILENAME	SPECINS
000	wi00891626	ISS1:10F1	p31051_1	01/02/2012	p31051_1.cpl	YES
001	wi00951837	ISS1:10F1	p31485_1	01/02/2012	p31485_1.cpl	NO
002	wi00946477	ISS1:10F1	p31426_1	01/02/2012	p31426_1.cpl	NO
003	wi00906163	ISS1:10F1	p31205_1	01/02/2012	p31205_1.cpl	NO
004	wi00962211	ISS1:10F1	p31580_1	01/02/2012	p31580_1.cpl	NO
005	wi00877592	ISS1:10F1	p30880_1	01/02/2012	p30880_1.cpl	NO
006	wi00839134	ISS1:10F1	p30698_1	01/02/2012	p30698_1.cpl	YES
007	wi00958682	ISS1:10F1	p31540_1	01/02/2012	p31540_1.cpl	NO
008	wi00868729	ISS1:10F1	p31163_1	01/02/2012	p31163_1.cpl	NO
009	wi00886321	ISS1:10F1	p31009_1	01/02/2012	p31009_1.cpl	NO
010	wi00946282	ISS1:10F1	p31204_1	01/02/2012	p31204_1.cpl	NO
011	wi00841980	ISS1:10F1	p30618_1	01/02/2012	p30618_1.cpl	NO
012	wi00946681	ISS1:10F1	p31428_1	01/02/2012	p31428_1.cpl	NO
013	wi00945533	ISS1:10F1	p31421_1	01/02/2012	p31421_1.cpl	YES
014	wi00843623	ISS1:10F1	p30731_1	01/02/2012	p30731_1.cpl	YES
015	wi00958776	ISS1:10F1	p31542_1	01/02/2012	p31542_1.cpl	YES
016	wi00857362	ISS1:10F1	p30782_1	01/02/2012	p30782_1.cpl	NO
017	wi00865477	ISS1:10F1	p30893_1	01/02/2012	p30893_1.cpl	YES
018	wi00879526	ISS1:10F1	p31007_1	01/02/2012	p31007_1.cpl	NO
019	wi00894243	ISS1:10F1	p31087_1	01/02/2012	p31087_1.cpl	NO
020	wi00890475	p30952	p31048_1	01/02/2012	p31048_1.cpl	NO

021	WI00927300	ISS1:10F1	p30999_1	01/02/2012	p30999_1.cpl	NO
022	wi00856991	ISS1:10F1	p17588_1	01/02/2012	p17588_1.cpl	NO
023	wi00688381	ISS1:10F1	p30104_1	01/02/2012	p30104_1.cpl	NO
024	wi00881777	ISS1:10F1	p25747_1	01/02/2012	p25747_1.cpl	NO
025	WI00853473	ISS1:10F1	p30625_1	01/02/2012	p30625_1.cpl	NO
026	wi00855423	ISS1:10F1	p31328_1	01/02/2012	p31328_1.cpl	YES
027	wi00943172	ISS1:10F1	p31402_1	01/02/2012	p31402_1.cpl	NO
028	wi00865477	ISS1:10F1	p30898_1	01/02/2012	p30898_1.cpl	YES
029	wi00850521	ISS1:10F1	p30709_1	01/02/2012	p30709_1.cpl	YES
030	wi00898327	ISS1:10F1	p31136_1	01/02/2012	p31136_1.cpl	NO
031	wi00871739	ISS1:10F1	p30856_1	01/02/2012	p30856_1.cpl	NO
032	wi00853031	ISS1:10F1	p30531_1	01/02/2012	p30531_1.cpl	NO
033	wi00839821	ISS1:10F1	p30619_1	01/02/2012	p30619_1.cpl	NO
034	wi00854130	ISS1:10F1	p30443_1	01/02/2012	p30443_1.cpl	NO
035	wi00871969	ISS1:10F1	p30768_1	01/02/2012	p30768_1.cpl	NO
036	wi00952381	ISS1:10F1	p31410_1	01/02/2012	p31410_1.cpl	NO
037	wi00946876	ISS1:10F1	p31430_1	01/02/2012	p31430_1.cpl	NO
038	wi00962557	ISS1:10F1	p31581_1	01/02/2012	p31581_1.cpl	NO
039	wi00833910	ISS2:10F1	p30492_2	01/02/2012	p30492_2.cpl	NO
040	wi00903085	ISS1:10F1	p31164_1	01/02/2012	p31164_1.cpl	NO
041	wi00875425	ISS1:10F1	p30943_1	01/02/2012	p30943_1.cpl	NO
042	wi00862574	iss1:10f1	p30870_1	01/02/2012	p30870_1.cpl	NO
043	wi00859499	ISS1:10F1	p30694_1	01/02/2012	p30694_1.cpl	NO
044	wi00925208	ISS1:10F1	p30986_1	01/02/2012	p30986_1.cpl	NO
045	wi00877442	ISS1:10F1	p30844_1	01/02/2012	p30844_1.cpl	NO
046	wi00900668	ISS1:10F1	p30456_1	01/02/2012	p30456_1.cpl	NO
047	wi00867905	ISS1:10F1	p30640_1	01/02/2012	p30640_1.cpl	NO
048	wi00879322	ISS1:10F1	p30954_1	01/02/2012	p30954_1.cpl	NO
049	wi00865477	ISS1:10F1	p30895_1	01/02/2012	p30895_1.cpl	YES
050	wi00951925	ISS1:10F1	p31486_1	01/02/2012	p31486_1.cpl	NO
051	wi00865477	ISS1:10F1	p30894_1	01/02/2012	p30894_1.cpl	YES
052	wi00865477	ISS1:10F1	p30897_1	01/02/2012	p30897_1.cpl	YES
053	wi00865477	ISS1:10F1	p30892_1	01/02/2012	p30892_1.cpl	YES
054	wi00908933	ISS1:10F1	p31239_1	01/02/2012	p31239_1.cpl	NO
055	wi00931028	ISS1:10F1	p31354_1	01/02/2012	p31354_1.cpl	YES
056	wi00932948	ISS1:10F1	p31077_1	01/02/2012	p31077_1.cpl	NO
057	wi00869695	ISS1:10F1	p30654_1	01/02/2012	p30654_1.cpl	NO
058	wi00838073	ISS1:10F1	p30588_1	01/02/2012	p30588_1.cpl	NO
059	wi00852365	ISS1:10F1	p30707_1	01/02/2012	p30707_1.cpl	NO
060	wi00927321	ISS1:10F1	p31286_1	01/02/2012	p31286_1.cpl	YES
061	wi00937114	ISS1:10F1	p31310_1	01/02/2012	p31310_1.cpl	NO
062	wi00877367	ISS1:10F1	p30534_1	01/02/2012	p30534_1.cpl	NO
063	wi00900096	ISS1:10F1	p31006_1	01/02/2012	p31006_1.cpl	NO
064	wi00905660	ISS1:10F1	p27968_1	01/02/2012	p27968_1.cpl	NO
065	wi00925141	ISS1:10F1	p30802_1	01/02/2012	p30802_1.cpl	NO
066	wi00943748	ISS1:10F1	p31516_1	01/02/2012	p31516_1.cpl	NO
067	wi00827950	ISS2:10F1	p30471_2	01/02/2012	p30471_2.cpl	NO
068	wi00937119	ISS1:10F1	p28005_1	01/02/2012	p28005_1.cpl	NO
069	wi00836981	ISS1:10F1	p30613_1	01/02/2012	p30613_1.cpl	NO
070	wi00961267	ISS1:10F1	p30288_1	01/02/2012	p30288_1.cpl	NO
071	wi00936714	ISS1:10F1	p31379_1	01/02/2012	p31379_1.cpl	NO
072	wi00906022	ISS1:10F1	p31202_1	01/02/2012	p31202_1.cpl	NO
073	wi00852389	ISS1:10F1	p30641_1	01/02/2012	p30641_1.cpl	NO
074	wi00857566	ISS1:10F1	p30766_1	01/02/2012	p30766_1.cpl	NO
075	wi00932204	ISS2:10F1	p31305_2	01/02/2012	p31305_2.cpl	NO
077	wi00865477	ISS1:10F1	p30890_1	01/02/2012	p30890_1.cpl	YES
078	wi00873382	ISS1:10F1	p30832_1	01/02/2012	p30832_1.cpl	NO
079	wi00948274	ISS1:10F1	p31365_1	01/02/2012	p31365_1.cpl	NO
080	wi00923899	ISS1:10F1	p31270_1	01/02/2012	p31270_1.cpl	NO
081	wi00856410	ISS1:10F1	p30749_1	01/02/2012	p30749_1.cpl	NO
082	wi00854415	ISS1:10F1	p30593_1	01/02/2012	p30593_1.cpl	NO
083	wi00896394	ISS1:10F1	p30807_1	01/02/2012	p30807_1.cpl	NO
084	wi00826075	ISS1:10F1	p30452_1	01/02/2012	p30452_1.cpl	NO
085	wi00863876	ISS1:10F1	p30787_1	01/02/2012	p30787_1.cpl	NO
086	wi00880386	ISS1:10F1	p30977_1	01/02/2012	p30977_1.cpl	NO
087	wi00840590	ISS1:10F1	p30767_1	01/02/2012	p30767_1.cpl	NO
088	wi00949627	ISS1:10F1	p31462_1	01/02/2012	p31462_1.cpl	NO
089	wi00842409	ISS1:10F1	p30621_1	01/02/2012	p30621_1.cpl	NO
090	wi00865477	ISS1:10F1	p30896_1	01/02/2012	p30896_1.cpl	YES
091	wi00897096	ISS1:10F1	p30676_1	01/02/2012	p30676_1.cpl	NO

092	wi00899584	ISS1:10F1	p30809_1	01/02/2012	p30809_1.cpl	NO
093	wi00907707	ISS1:10F1	p31228_1	01/02/2012	p31228_1.cpl	NO
094	wi00949273	ISS1:10F1	p31411_1	01/02/2012	p31411_1.cpl	NO
095	wi00839255	ISS1:10F1	p30591_1	01/02/2012	p30591_1.cpl	NO
096	wi00921340	ISS1:10F1	p31266_1	01/02/2012	p31266_1.cpl	NO
097	wi00903369	ISS1:10F1	p31165_1	01/02/2012	p31165_1.cpl	NO
098	wi00875701	ISS1:10F1	p30942_1	01/02/2012	p30942_1.cpl	NO
099	wi00884699	ISS1:10F1	p31000_1	01/02/2012	p31000_1.cpl	YES
100	wi00834382	ISS1:10F1	p30548_1	01/02/2012	p30548_1.cpl	NO
101	wi00960133	ISS2:10F1	p31557_2	01/02/2012	p31557_2.cpl	NO
102	wi00929140	ISS1:10F1	p31284_1	01/02/2012	p31284_1.cpl	NO
103	wi00948931	ISS1:10F1	p31407_1	01/02/2012	p31407_1.cpl	NO
104	wi00887744	ISS2:10F1	p31026_2	01/02/2012	p31026_2.cpl	NO
105	wi00905600	ISS1:10F1	p31201_1	01/02/2012	p31201_1.cpl	NO
106	wi00869243	ISS1:10F1	p30848_1	01/02/2012	p30848_1.cpl	NO
107	WI00854150	ISS1:10F1	p30468_1	01/02/2012	p30468_1.cpl	NO
108	wi00897176	ISS1:10F1	p30418_1	01/02/2012	p30418_1.cpl	NO
109	wi00903381	ISS1:10F1	p30421_1	01/02/2012	p30421_1.cpl	NO
110	wi00959854	ISS1:10F1	p31556_1	01/02/2012	p31556_1.cpl	NO
111	wi00908598	ISS1:10F1	p31235_1	01/02/2012	p31235_1.cpl	NO
112	wi00903437	ISS1:10F1	p31167_1	01/02/2012	p31167_1.cpl	NO
113	wi00900766	ISS1:10F1	p31159_1	01/02/2012	p31159_1.cpl	NO
114	wi00946558	ISS1:10F1	p31358_1	01/02/2012	p31358_1.cpl	NO
115	wi00932958	ISS1:10F1	p31115_1	01/02/2012	p31115_1.cpl	NO
116	wi00895090	ISS1:10F1	p31105_1	01/02/2012	p31105_1.cpl	NO
117	wi00824257	ISS1:10F1	p30447_1	01/02/2012	p30447_1.cpl	NO
118	wi00895181	ISS1:10F1	p31106_1	01/02/2012	p31106_1.cpl	NO
119	WI00928455	ISS1:10F1	p31297_1	01/02/2012	p31297_1.cpl	NO
120	wi00832106	ISS1:10F1	p30550_1	01/02/2012	p30550_1.cpl	NO
121	wi00953900	ISS1:10F1	p31494_1	01/02/2012	p31494_1.cpl	NO
122	wi00942734	ISS1:10F1	p31409_1	01/02/2012	p31409_1.cpl	NO
123	wi00898200	ISS1:10F1	p31274_1	01/02/2012	p31274_1.cpl	NO
124	wi00882293	ISS1:10F1	p31010_1	01/02/2012	p31010_1.cpl	NO
125	WI00843571	ISS1:10F1	p30627_1	01/02/2012	p30627_1.cpl	NO
126	wi00835294	ISS1:10F1	p30565_1	01/02/2012	p30565_1.cpl	NO
127	WI00836292	ISS1:10F1	p30554_1	01/02/2012	p30554_1.cpl	NO
128	WI00900213	ISS1:10F1	p30656_1	01/02/2012	p30656_1.cpl	NO
129	wi00921295	ISS1:10F1	p31265_1	01/02/2012	p31265_1.cpl	NO
130	wi00957141	ISS1:10F1	p31579_1	01/02/2012	p31579_1.cpl	NO
131	WI00836334	ISS1:10F1	p30481_1	01/02/2012	p30481_1.cpl	NO
132	wi00858335	ISS1:10F1	p30819_1	01/02/2012	p30819_1.cpl	NO
133	wi00859123	ISS1:10F1	p30648_1	01/02/2012	p30648_1.cpl	NO
134	wi00959820	ISS1:10F1	p31562_1	01/02/2012	p31562_1.cpl	NO
135	wi00905297	ISS1:10F1	p31195_1	01/02/2012	p31195_1.cpl	NO
136	wi00907697	ISS1:10F1	p31227_1	01/02/2012	p31227_1.cpl	NO
137	wi00951427	ISS1:10F1	p31478_1	01/02/2012	p31478_1.cpl	NO
138	wi00883604	ISS1:10F1	p30973_1	01/02/2012	p30973_1.cpl	NO
139	wi00962955	ISS1:10F1	p31585_1	01/02/2012	p31585_1.cpl	NO
140	wi00860279	ISS1:10F1	p30789_1	01/02/2012	p30789_1.cpl	NO
141	wi00909476	ISS1:10F1	p31340_1	01/02/2012	p31340_1.cpl	NO
142	wi00925218	ISS1:10F1	p30675_1	01/02/2012	p30675_1.cpl	NO
143	wi00836182	ISS1:10F1	p30450_1	01/02/2012	p30450_1.cpl	NO
144	wi00841273	ISS1:10F1	p30713_1	01/02/2012	p30713_1.cpl	NO
145	WI00889786	ISS1:10F1	p30750_1	01/02/2012	p30750_1.cpl	NO
146	wi00894443	ISS1:10F1	p31093_1	01/02/2012	p31093_1.cpl	NO
147	wi00896420	ISS1:10F1	p30867_1	01/02/2012	p30867_1.cpl	NO
148	wi00941500	ISS1:10F1	p31394_1	01/02/2012	p31394_1.cpl	NO
149	wi00950592	ISS1:10F1	p31499_1	01/02/2012	p31499_1.cpl	NO
150	wi00927678	ISS1:10F1	p31399_1	01/02/2012	p31399_1.cpl	NO
151	wi00930864	ISS1:10F1	p31325_1	01/02/2012	p31325_1.cpl	NO
152	wi00957252	ISS1:10F1	p31530_1	01/02/2012	p31530_1.cpl	NO
153	wi00880836	ISS1:10F1	p30976_1	01/02/2012	p30976_1.cpl	NO
154	wi00865477	ISS1:10F1	p30891_1	01/02/2012	p30891_1.cpl	YES
155	wi00896680	ISS1:10F1	p30357_1	01/02/2012	p30357_1.cpl	NO
156	wi00856702	ISS1:10F1	p30573_1	01/02/2012	p30573_1.cpl	NO
157	wi00897082	ISS1:10F1	p31124_1	01/02/2012	p31124_1.cpl	NO
158	wi00853178	ISS1:10F1	p30719_1	01/02/2012	p30719_1.cpl	NO
159	wi00938555	ISS1:10F1	p30881_1	01/02/2012	p30881_1.cpl	YES
160	WI00839794	ISS1:10F1	p28647_1	01/02/2012	p28647_1.cpl	NO
MDP>LAST SUCCESSFUL MDP REFRESH :2012-01-24 11:17:37(Local Time)						

MDP>USING DEPLIST ZIP FILE DOWNLOADED :2012-01-11 11:07:13(est)

Avaya Communication Server 1000E signaling server service updates

Product Release: 7.50.17.00

In system patches: 2

PATCH#	NAME	IN_SERVICE	DATE	SPECINS	TYPE	RPM
2	p27159_1	Yes	13/04/12	NO	FRU	cs1000-pi-control-1.00.00.00-00.noarch
4	p30260_1	Yes	25/07/12	NO	FRU	cs1000-pi-control-1.00.00.00-00.noarch

In System service updates: 24

PATCH#	IN SERVICE	DATE	SPECINS	REMOVABLE	NAME
0	Yes	27/03/12	NO	YES	cs1000-csmWeb-7.50.17.16-3.i386.000
1	Yes	20/01/12	NO	YES	cs1000-baseWeb-7.50.17.16-1.i386.001
3	Yes	20/01/12	NO	YES	cs1000-dbcom-7.50.17-02.i386.000
5	Yes	20/01/12	NO	YES	cs1000-shared-pbx-7.50.17.16-1.i386.000
6	Yes	20/01/12	NO	YES	cs1000-kcv-7.50.17.16-1.i386.000
7	Yes	20/01/12	NO	YES	cs1000-nrsmWebService-7.50.17.16-1.i386.000
8	Yes	01/08/12	NO	YES	cs1000-vtrk-7.50.17.16-105TMP.i386.000
10	Yes	20/01/12	NO	YES	cs1000-ipsec-7.50.17.16-1.i386.000
14	Yes	20/01/12	NO	YES	ipsec-tools-0.6.5-14.el5.3_avaya_1.i386.000
15	Yes	20/01/12	NO	YES	spiritAgent-6.1-1.0.0.108.208.i386.000
16	Yes	20/01/12	NO	YES	cs1000-EmCentralLogic-7.50.17.16-1.i386.000
22	Yes	27/03/12	NO	YES	cs1000-linuxbase-7.50.17.16-07.i386.000
23	Yes	27/03/12	NO	YES	cs1000-mscAnnc-7.50.17.16-1.i386.000
24	Yes	27/03/12	NO	YES	cs1000-mscTone-7.50.17.16-1.i386.000
25	Yes	27/03/12	NO	YES	cs1000-mscMusc-7.50.17.16-2.i386.000
26	Yes	27/03/12	NO	YES	cs1000-dmWeb-7.50.17.16-2.i386.000
27	Yes	27/03/12	NO	YES	cs1000-patchWeb-7.50.17.16-4.i386.000
28	Yes	27/03/12	NO	YES	cs1000-nrsm-7.50.17.16-3.i386.000
29	Yes	27/03/12	NO	YES	cs1000-ftrpkg-7.50.17.16-9.i386.000
30	Yes	27/03/12	NO	YES	cs1000-sps-7.50.17.16-4.i386.000
31	Yes	27/03/12	NO	YES	cs1000-tps-7.50.17.16-13.i386.000
32	Yes	27/03/12	NO	YES	cs1000-emWeb_6-0-7.50.17.16-19.i386.000
33	Yes	27/03/12	NO	YES	cs1000-bcc-7.50.17.16-51.i386.000
34	Yes	27/03/12	NO	YES	cs1000-Jboss-Quantum-7.50.17.16-16.i386.000

Avaya Communication Server 1000E system software

Product Release: 7.50.17.00

Base Applications

base	7.50.17	[patched]
NTAFS	7.50.17	
sm	7.50.17	
cs1000-Auth	7.50.17	
Jboss-Quantum	7.50.17	[patched]
lhmonitor	7.50.17	
baseAppUtils	7.50.17	[patched]
dfoTools	7.50.17	
nnnm	7.50.17	
cppmUtil	7.50.17	
oam-logging	7.50.17	[patched]
dmWeb	n/a	[patched]
baseWeb	n/a	[patched]
ipsec	n/a	[patched]
Snmp-Daemon-TrapLib	7.50.17	
ISECSH	7.50.17	
patchWeb	n/a	[patched]
EmCentralLogic	n/a	[patched]

Application configuration: CS+SS+NRS+EM

Packages:

CS+SS+NRS+EM

Configuration version:	7.50.17-00
cs	7.50.17
dbcom	7.50.17 [patched]

cslogin	7.50.17	
sigServerShare	7.50.17	[patched]
csv	7.50.17	
tps	7.50.17.16	[patched]
vtrk	7.50.17.16	[patched]
pd	7.50.17	
sps	7.50.17.16	[patched]
ncs	7.50.17	
gk	7.50.17	
nrsm	7.50.17	[patched]
nrsmWebService	7.50.17	[patched]
managedElementWebService	7.50.17	
EmConfig	7.50.17	
emWeb 6-0	7.50.17	[patched]
emWebLocal_6-0	7.50.17	
csmWeb	7.50.17	[patched]
bcc	7.50.17	[patched]
ftrpkg	7.50.17	[patched]
cs1000WebService 6-0	7.50.17	
mscAnnc	7.50.17	
mscAttn	7.50.17	
mscConf	7.50.17	
mscMusc	7.50.17	
mscTone	7.50.17	

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