Avaya Solution & Interoperability Test Lab

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 CS100E 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.

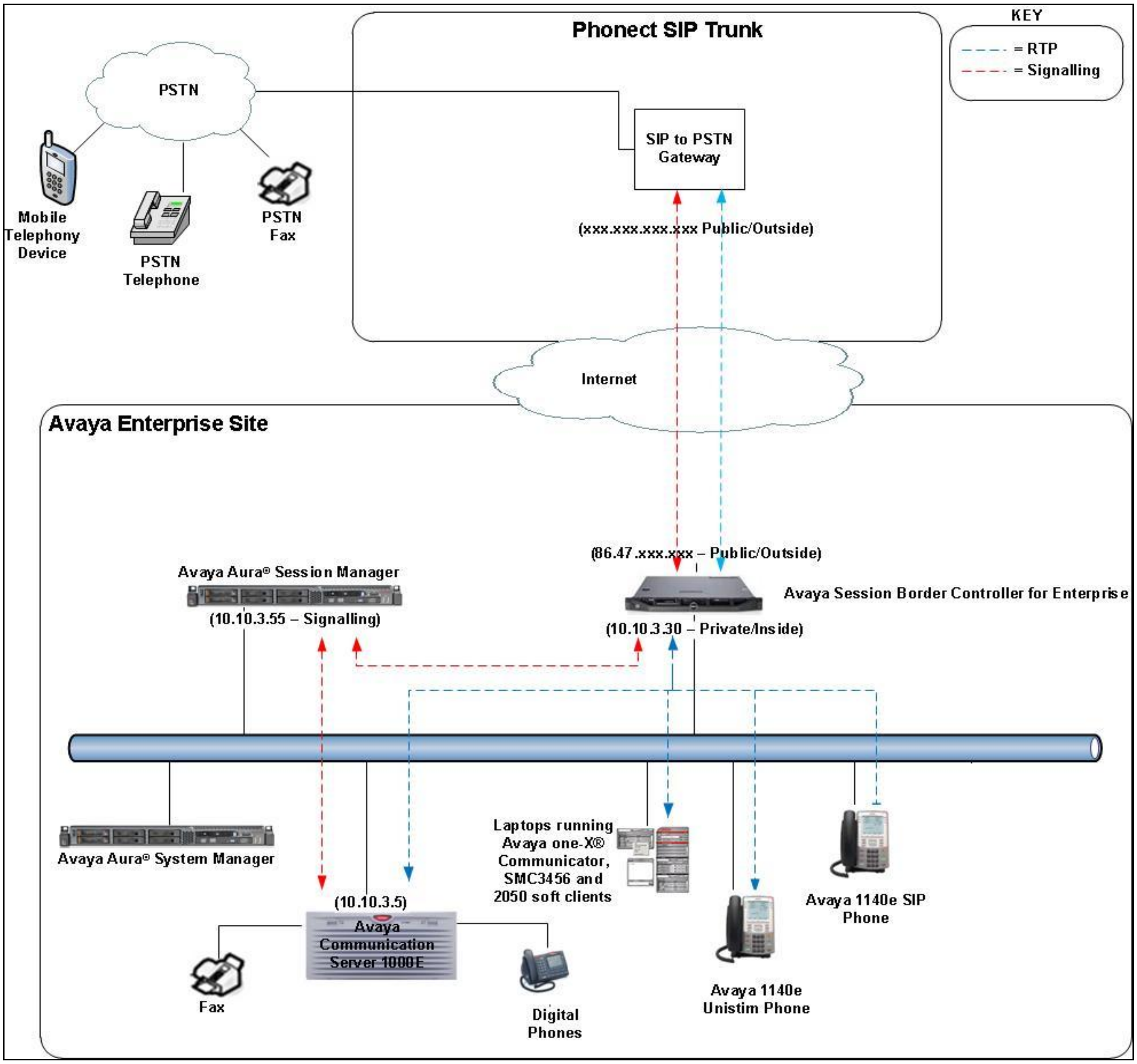


Figure 1: Test Setup Phonect SIP Trunk Service to Avaya Enterprise

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

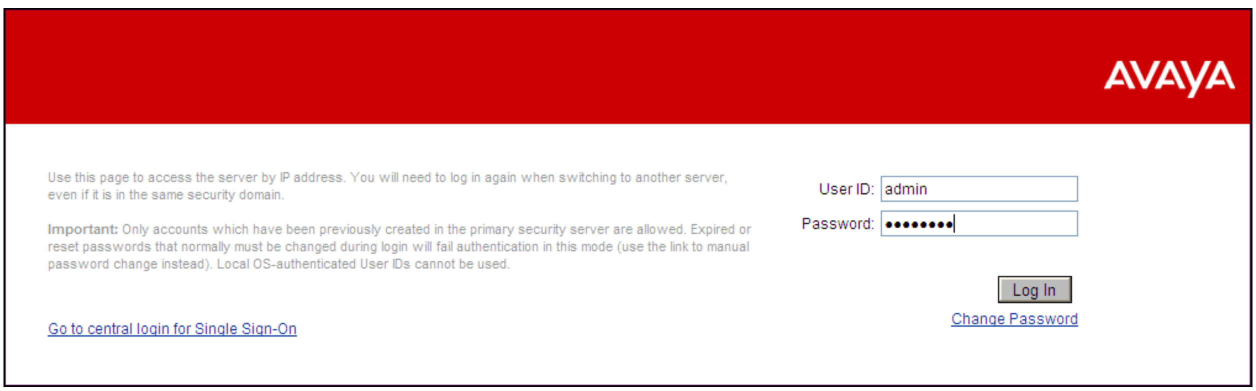
This section describes the steps required to configure CS1000E for SIP Trunking and also the necessary configuration for terminals (analog, SIP and IP phones). SIP trunks are established between CS1000E and Session Manager. These SIP trunks carry SIP signaling associated with Phonect SIP Trunk Service. 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

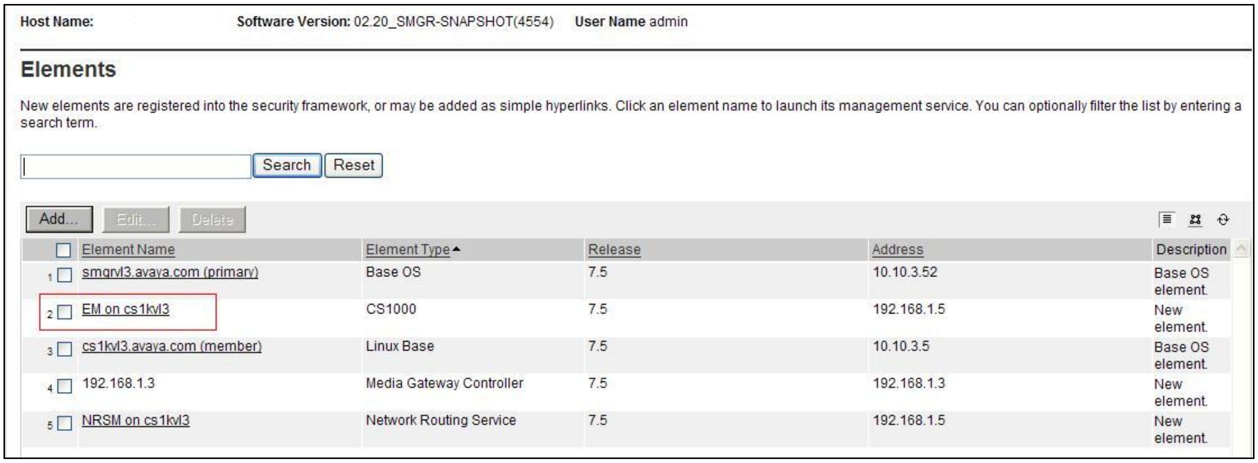
Configuration on the CS1000E will be performed by using both SSH Putty session and Avaya Unified Communications Management GUI.

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 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 cs1kvl3**.



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.

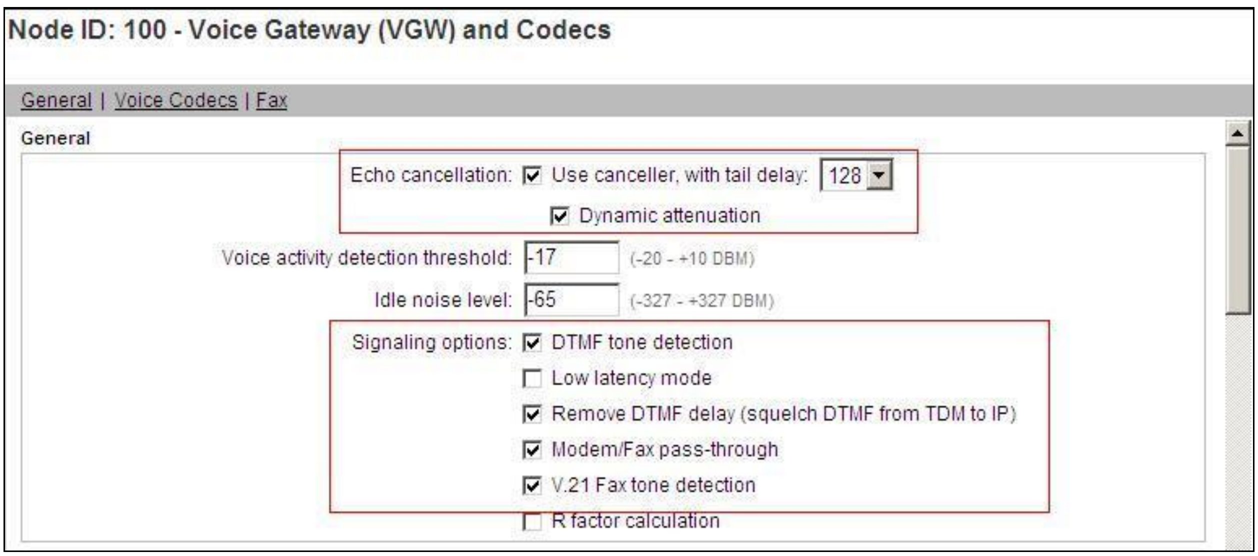
```
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

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).

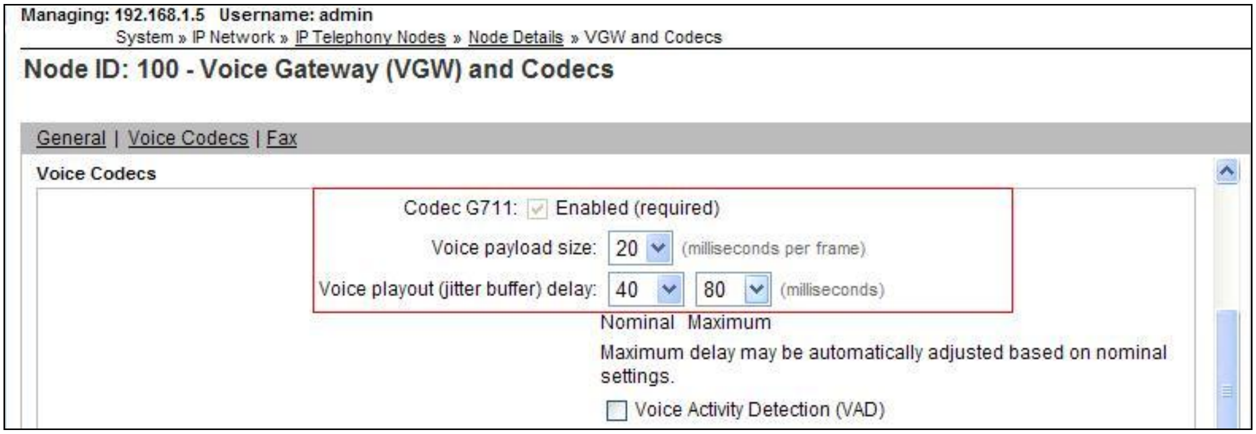
```
REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

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

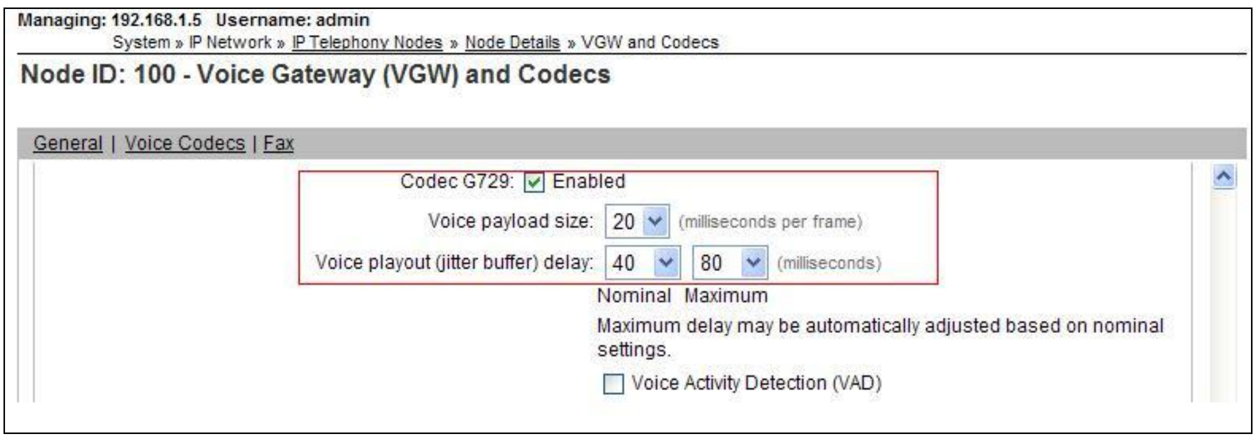
Phonect SIP Trunk service supports G.711A, G.711U and G.729 voice codec's and T.38 FAX transmissions. Using the CS1000E element manager sidebar, navigate to the **IP Network → IP Telephony Nodes → Node Details → VGW Gateway (VGW) and Codecs** property page and configure the CS1000E General codec settings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



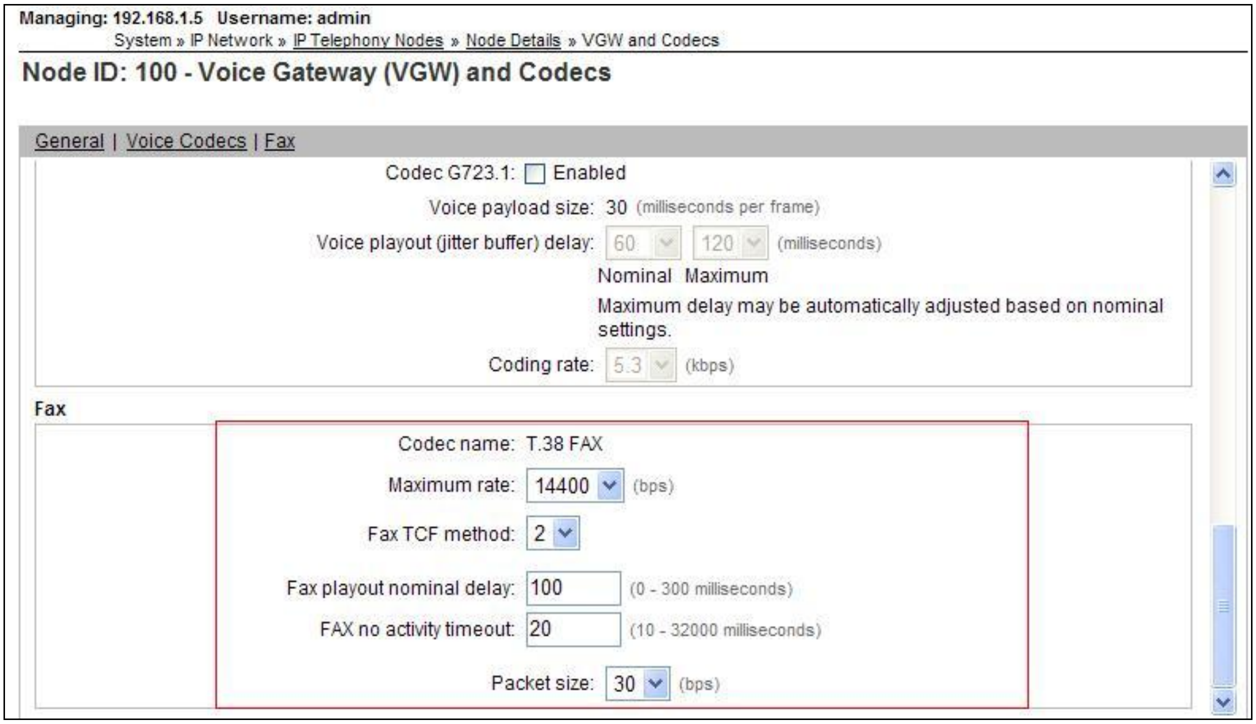
Next, scroll down and configure the **Codec G.711**. The relevant settings are highlighted in the following screenshot.



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

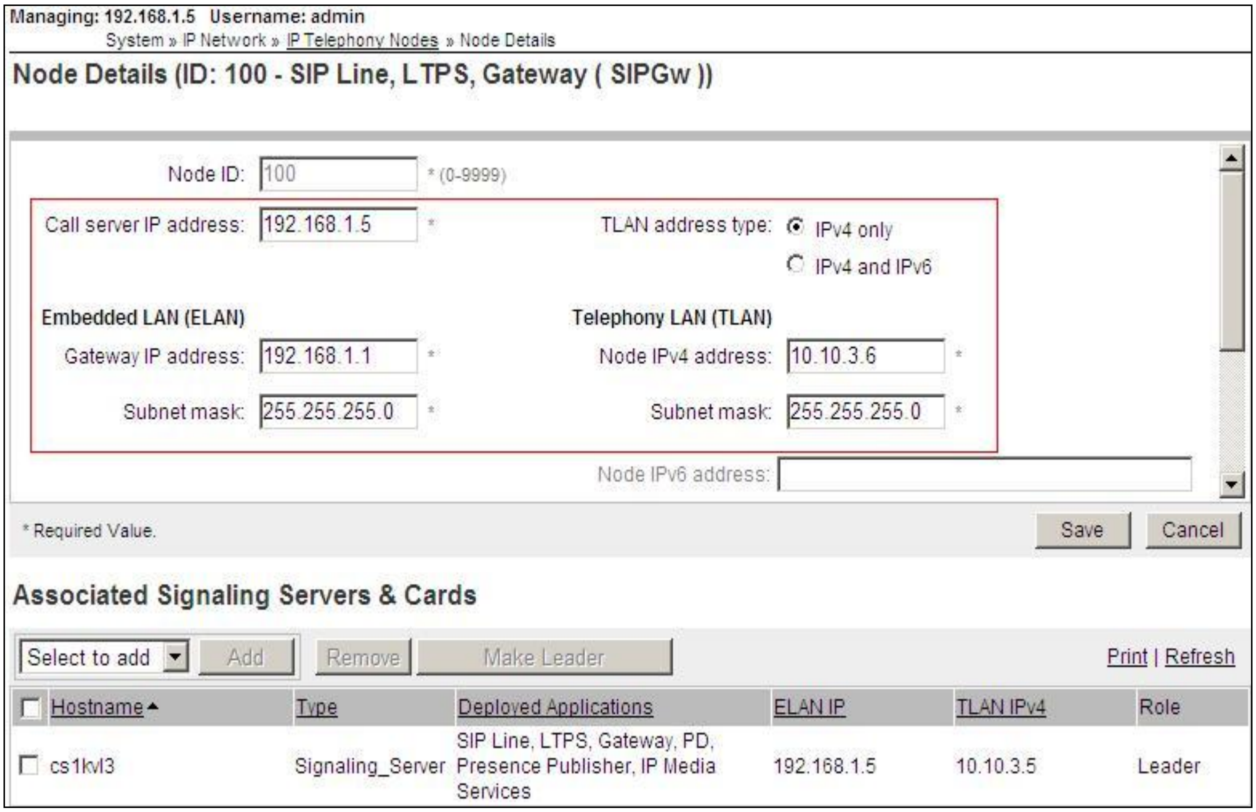


Finally, configure the **Fax** settings as in the highlighted section of the next screenshot.



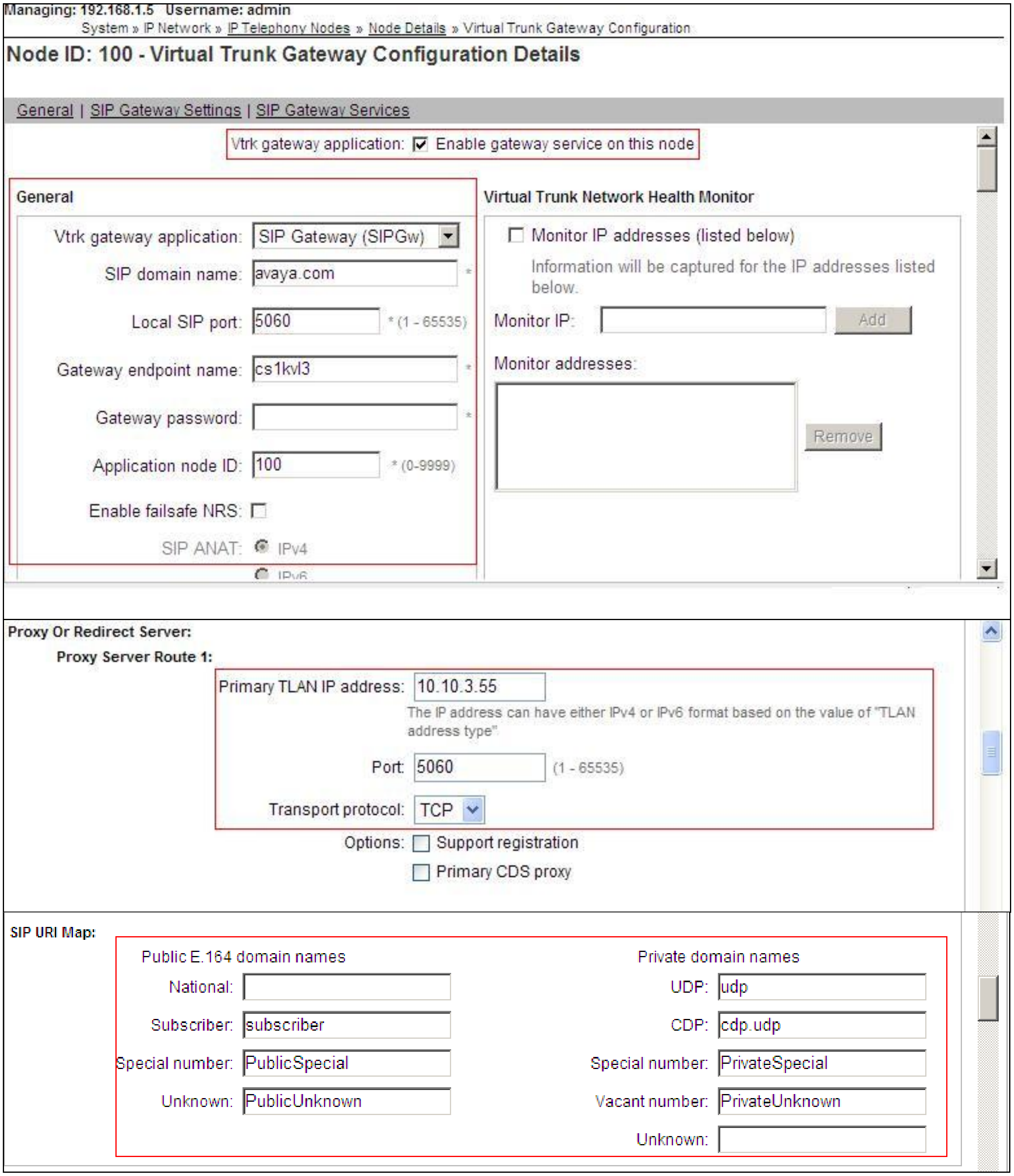
5.4. Virtual Trunk Gateway Configuration

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).



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.

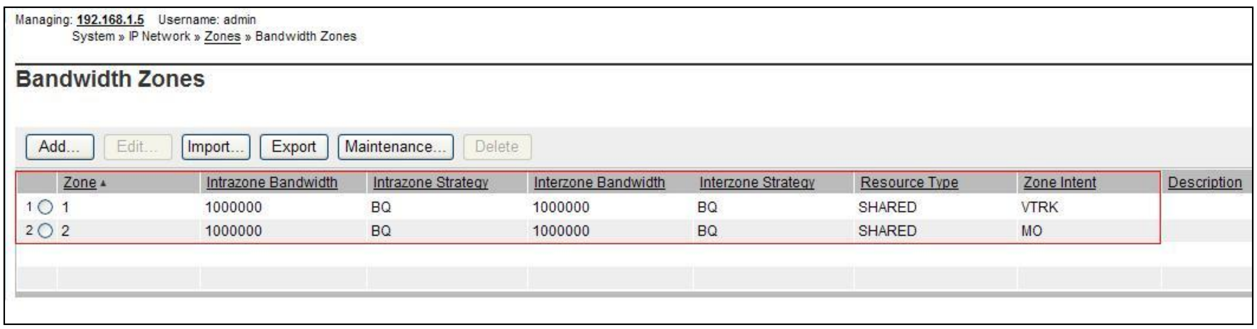
- **Vtrk gateway application:** Provides option to select Gateway applications. The three supported modes are **SIP Gateway (SIPGw)**, **H.323Gw**, and **SIPGw and H.323Gw**
- **SIP domain name:** The SIP Domain Name is the SIP Service Domain. The SIP Domain Name configured in the Signalling Server properties 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


Managing: 192.168.1.5 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration
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: cs1kvl3
Gateway password:
Application node ID: 100 * (0-9999)
Enable failsafe NRS:
SIP ANAT: IPv4
Virtual Trunk Network Health Monitor
Monitor IP addresses (listed below)
Information will be captured for the IP addresses listed below.
Monitor IP:
Add
Monitor addresses:
Remove
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**.

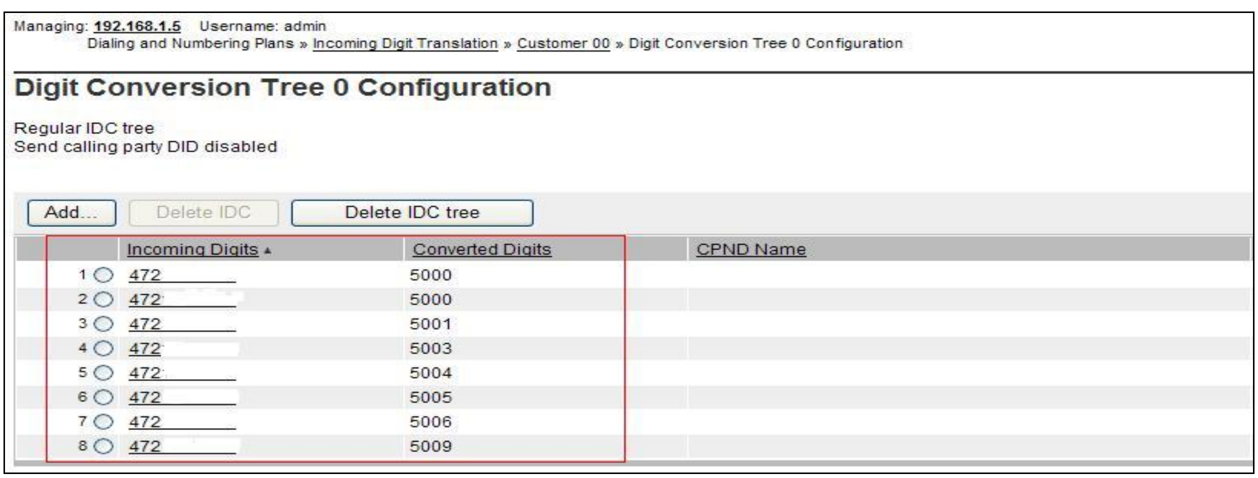
Use Element Manager to define bandwidth zones as in the following highlighted example. Use Element Manager and navigate to **System → IP Network → Zones → Bandwidth Zones** and add new zones as required.



	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.



	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
    OVLR 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
```

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 FLEN 0 RRPA NO **RLI 10** CCBA NO	TSC 18 FLEN 0 RRPA NO **RLI 10** CCBA NO	TSC 800 FLEN 0 RRPA NO **RLI 10** CCBA NO	TSC 08 FLEN 0 RRPA NO **RLI 10** 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 RCBD
     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 CCBD RTDD RBDD RBHD PGND OCBD 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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```

```
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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 RCBD
     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 CCBD RTDD RBDD RBHD PGND OCBD 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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```

```
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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 CCBD BNRD OCBD 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
```

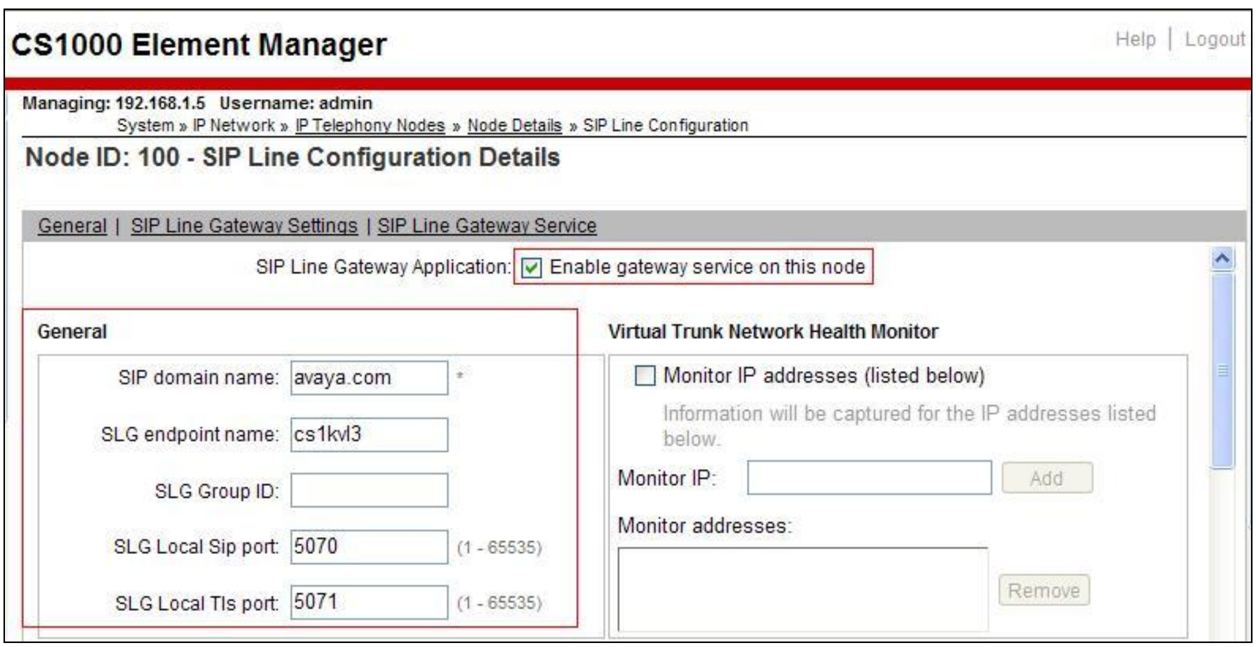
5.9. Configure the SIP Line Gateway Service

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**



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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```

```
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     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD 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
```
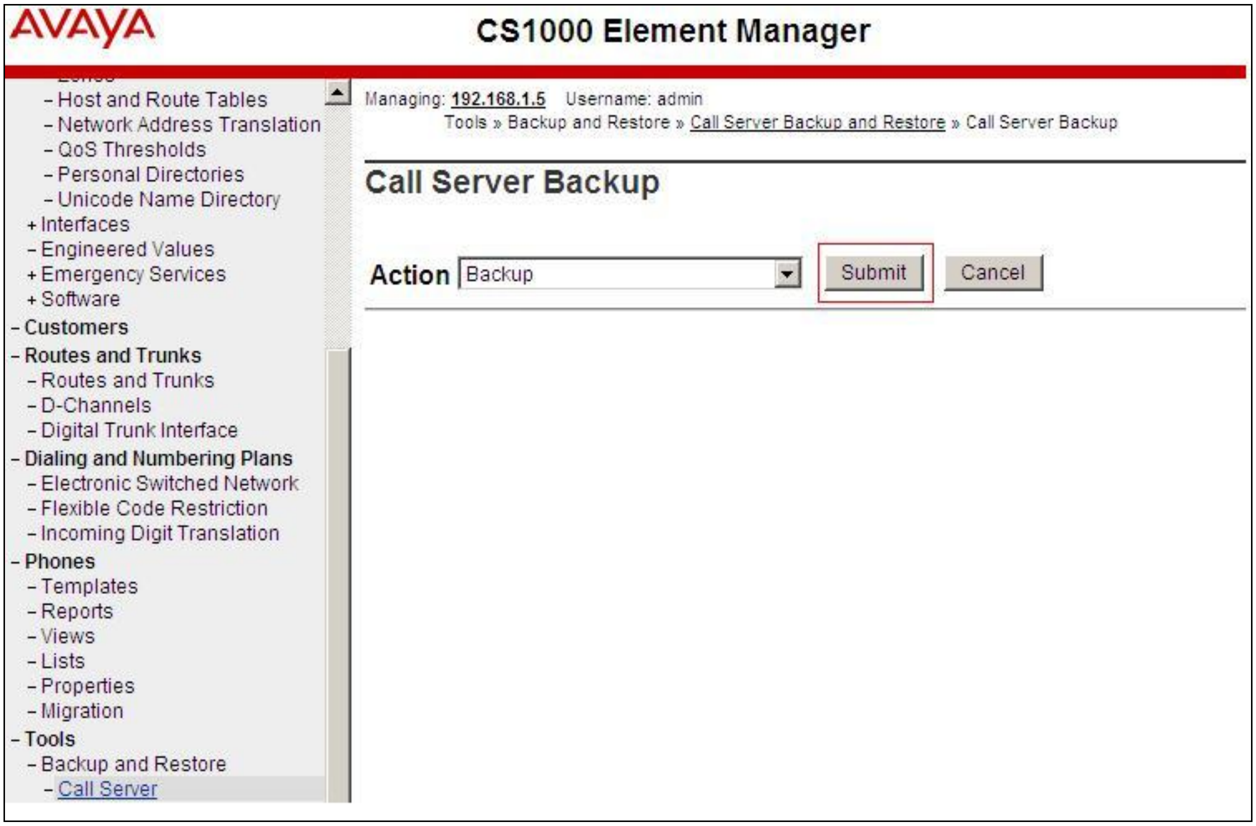

5.11. Save Configuration

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.



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.

Configuration of CS1000E is complete.

6. Configure Avaya Aura® Session Manager

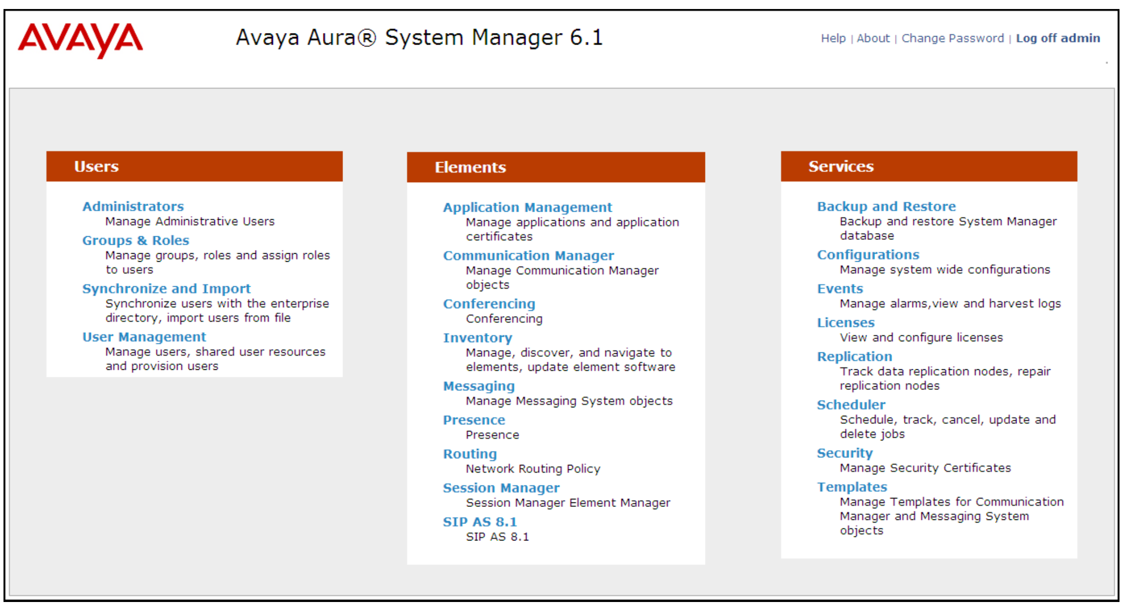
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- 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

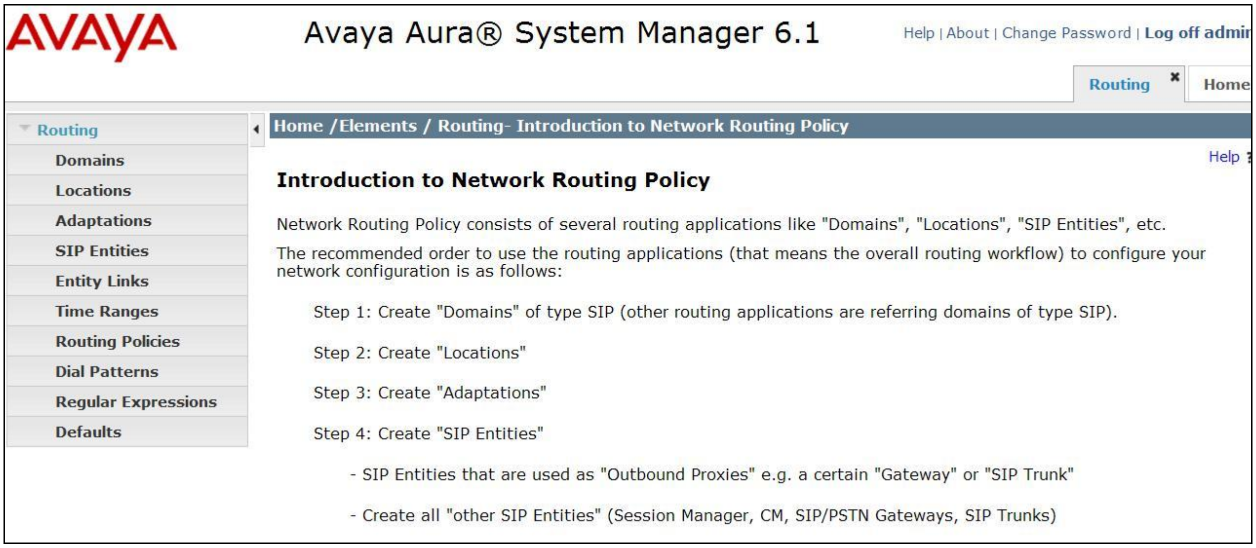
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, Locations, SIP Entities, and Session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

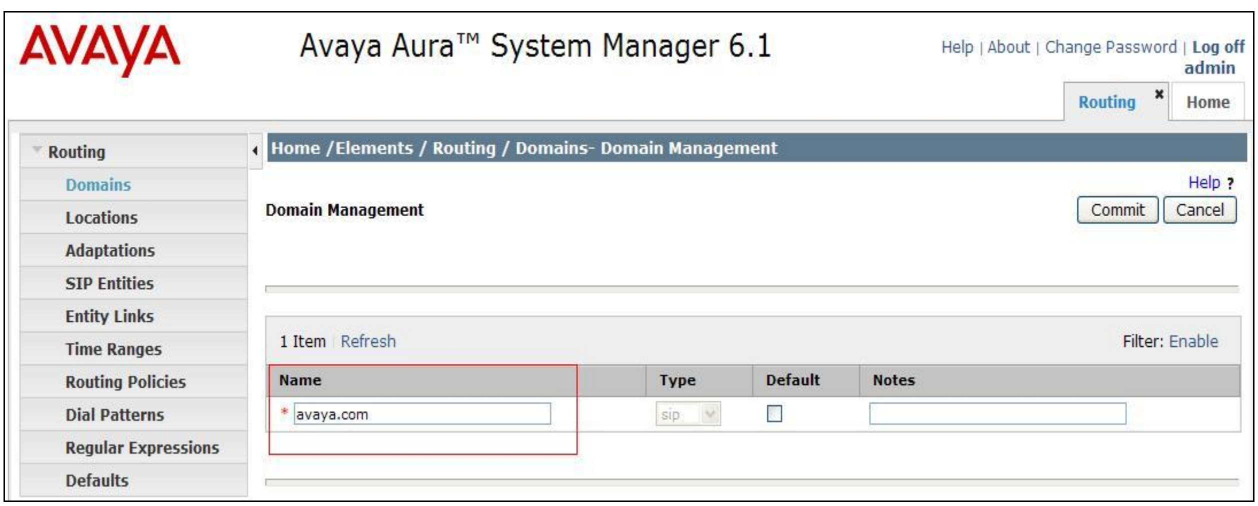


6.2. Define SIP Domain

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.



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:

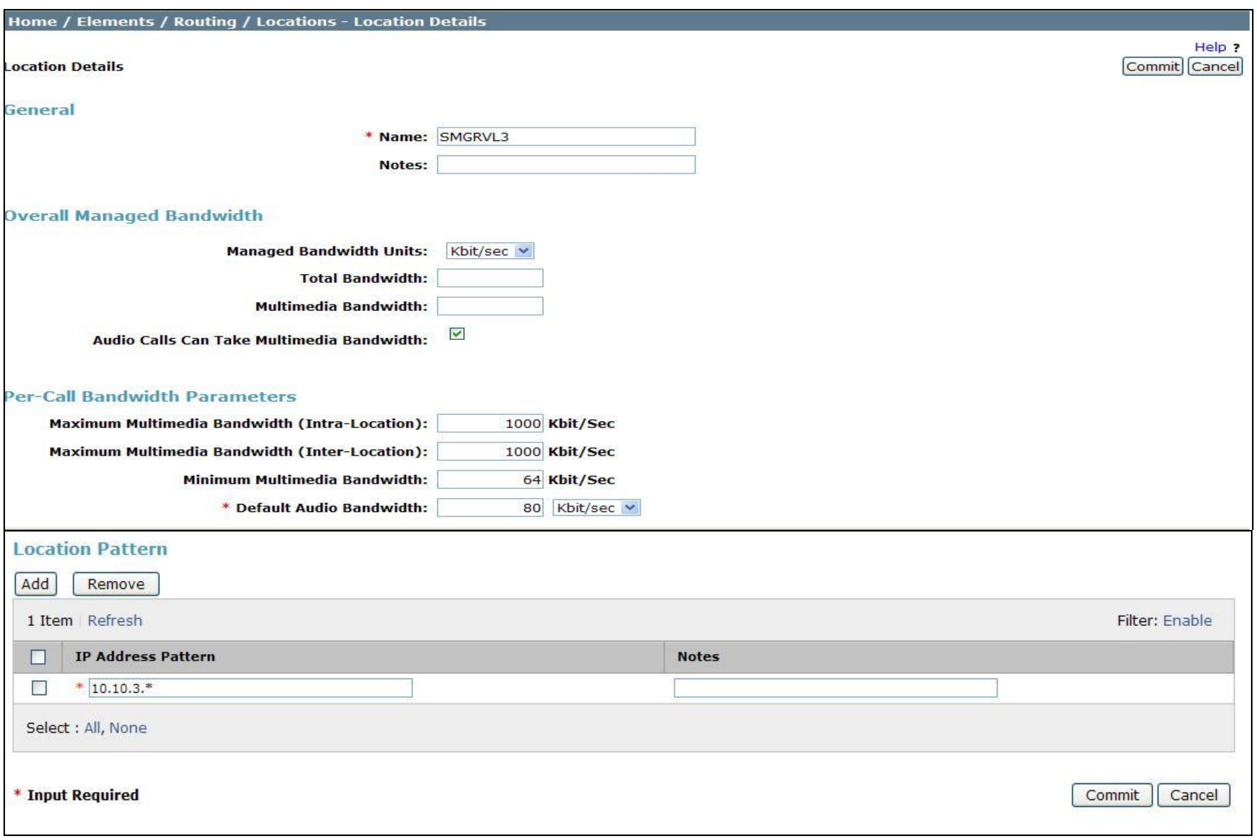
- **Name:** Enter a descriptive name for the location
- **Notes:** Add a brief description (optional)

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]

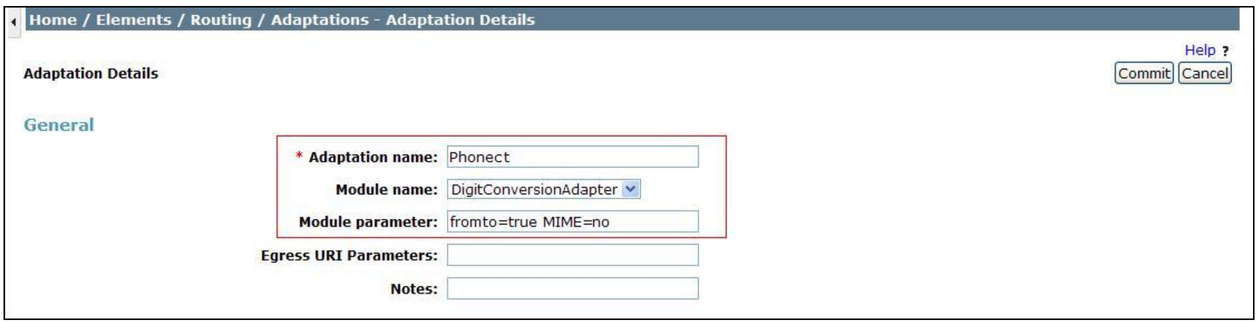
Click **Commit** to save. The screenshot below shows the Location defined for CS1000E in the sample configuration.



6.4. Configure Adaptation Module

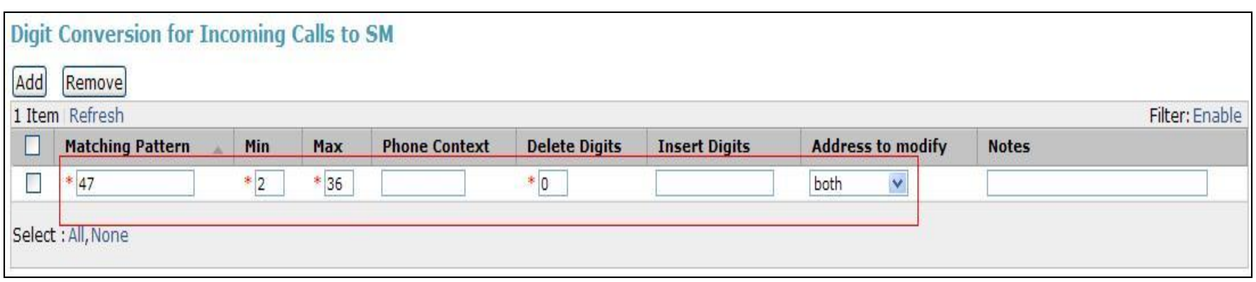
To enable calls to be routed to stations on CS1000E, the Session Manager should be configured to modify the called party number to meet network requirements. Expand **Elements → Routing** and select **Adaptations** from the left navigational menu. Click **New (**not shown**)**. In the **General** section,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

- **Adaptation Name** Enter an identifier for the Adaptation Module
- **Module Name** Select **DigitConversonAdaptor** 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



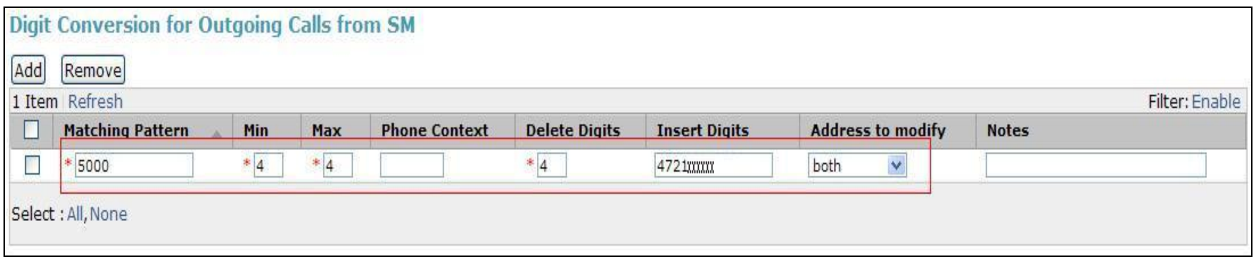
In the **Digit Conversion for Incoming Calls to SM** section, click **Add** and enter the following values.

- **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**



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**



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).

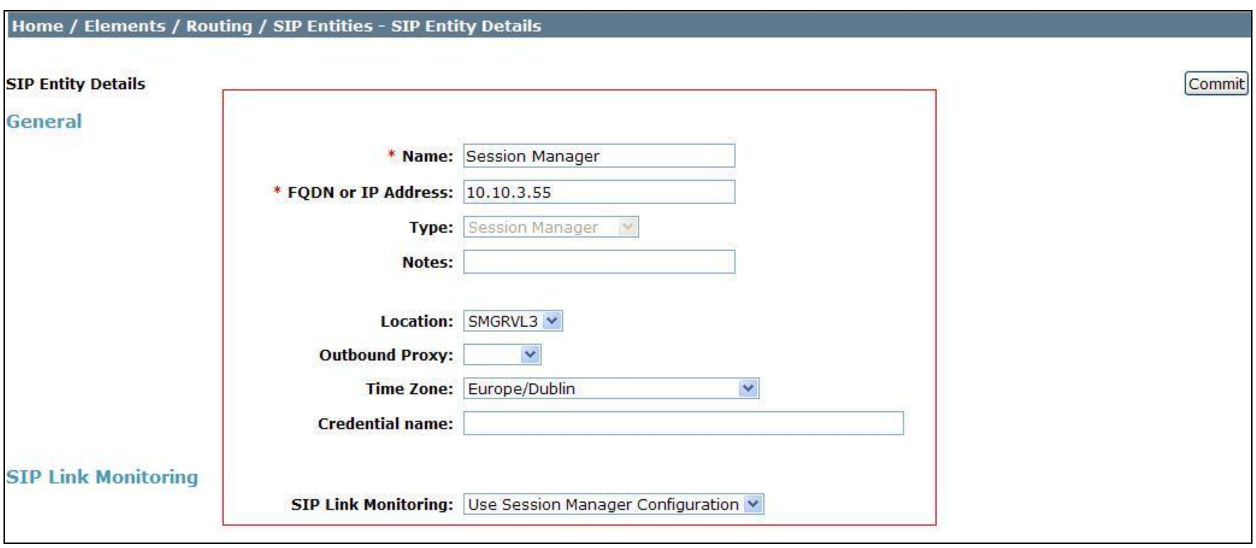
In the **General** section, enter the following values. Use default values for all remaining fields:

- **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 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**.

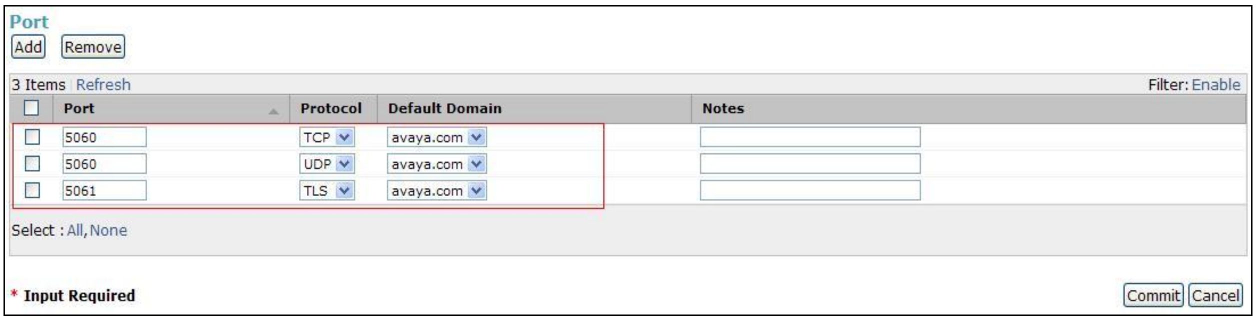


To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities. This section defines a default set of ports that Session Manager will use to listen for SIP requests, typically from 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

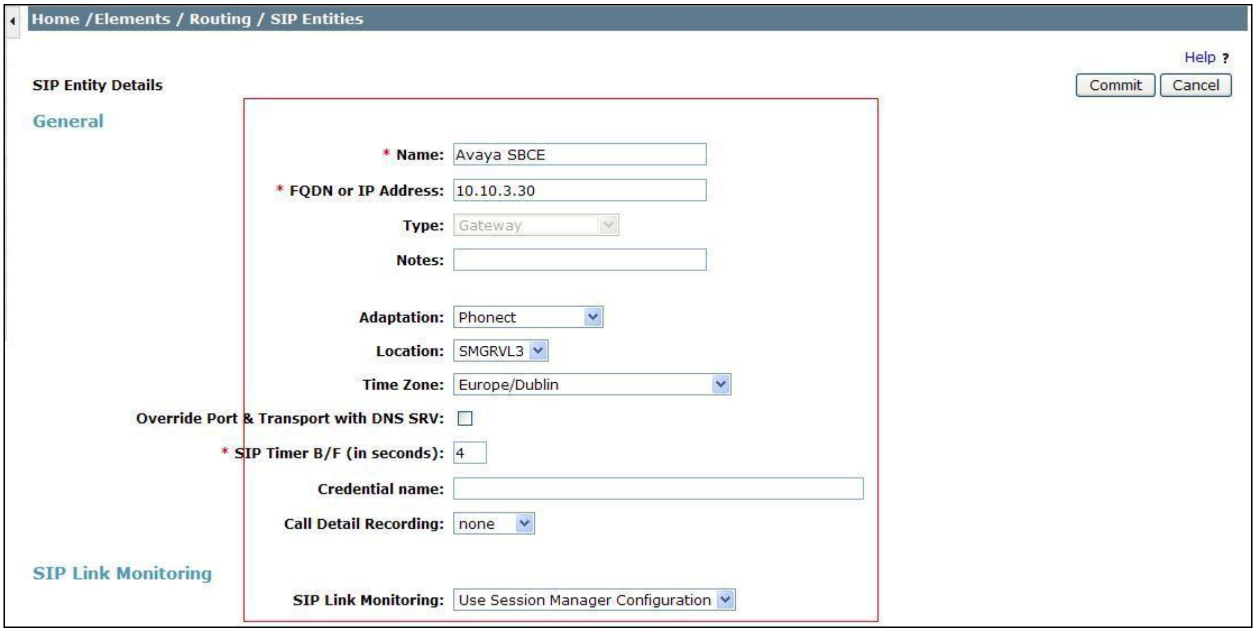
Defaults can be used for the remaining fields. Click **Commit** to save.

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



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 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.

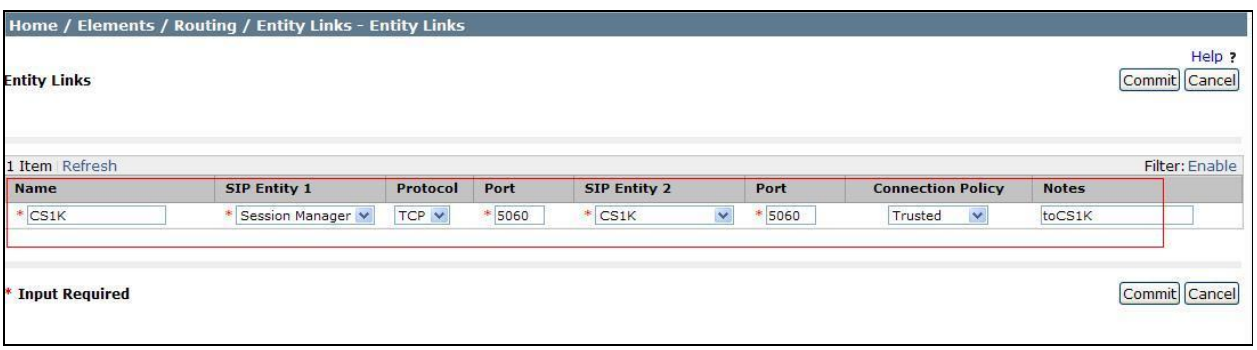
6.6. Define Entity Links

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:

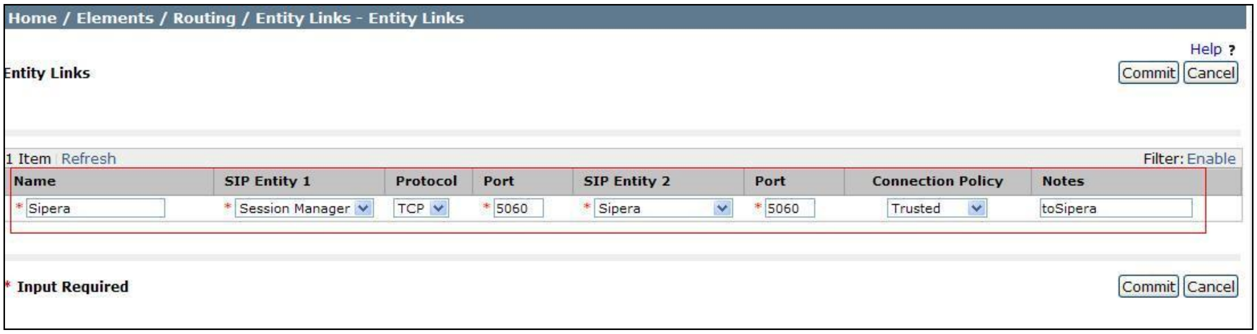
- **Name:** Enter a descriptive name
- **SIP Entity 1:** Select the SIP Entity for Session Manager
- **Protocol:** Select the transport protocol used for this link
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. Default listen port is **5060**
- **SIP Entity 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.



Entity Link to Avaya SBCE.



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 polices 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).

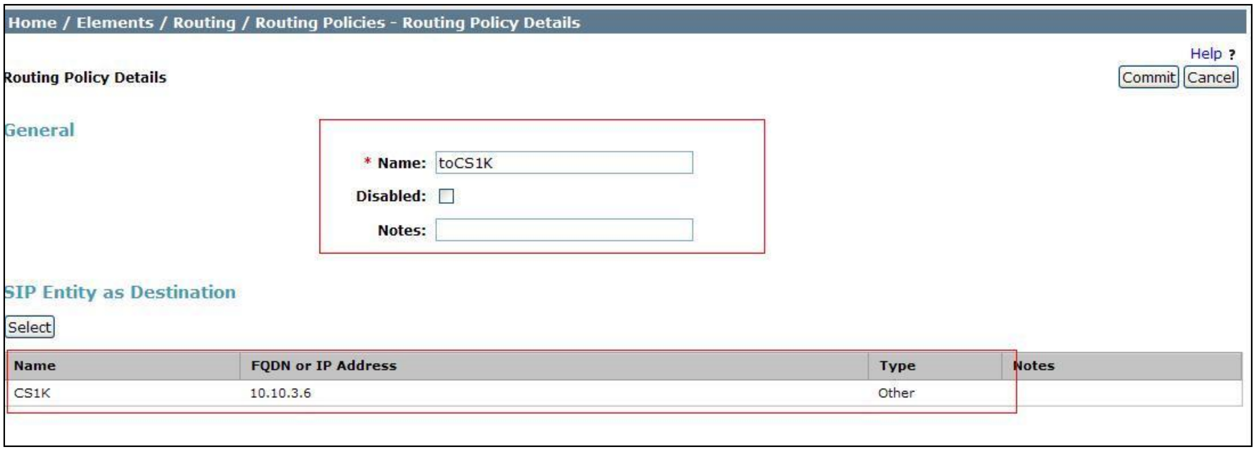
In the **General** section, enter the following values.

- **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.

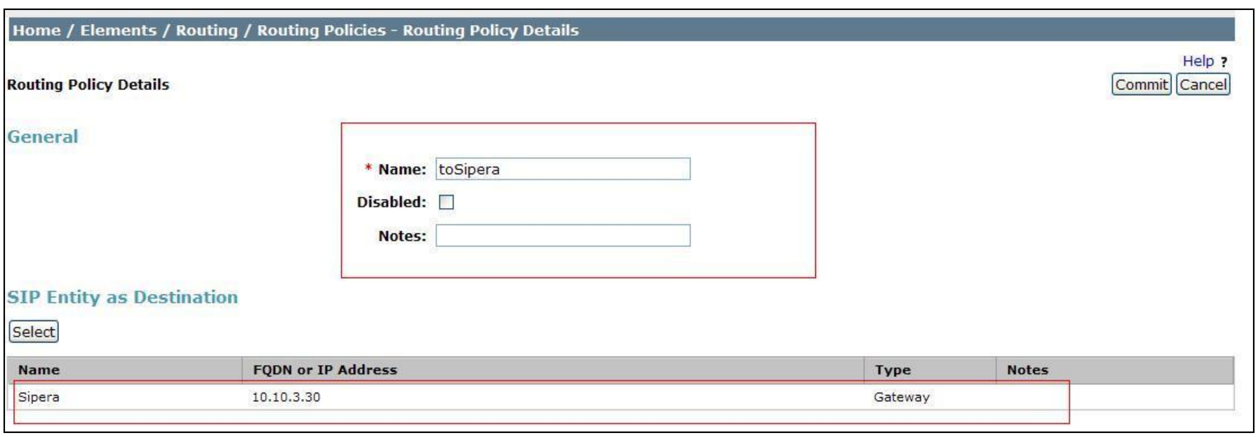
Note: The Routing Policy defined in this section is an example and was used in the sample configuration. Other Routing Policies may be appropriate for different customer networks.

The following screenshot shows the Routing Policy for CS1000E.



For Routing Policy to the Avay 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.



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:

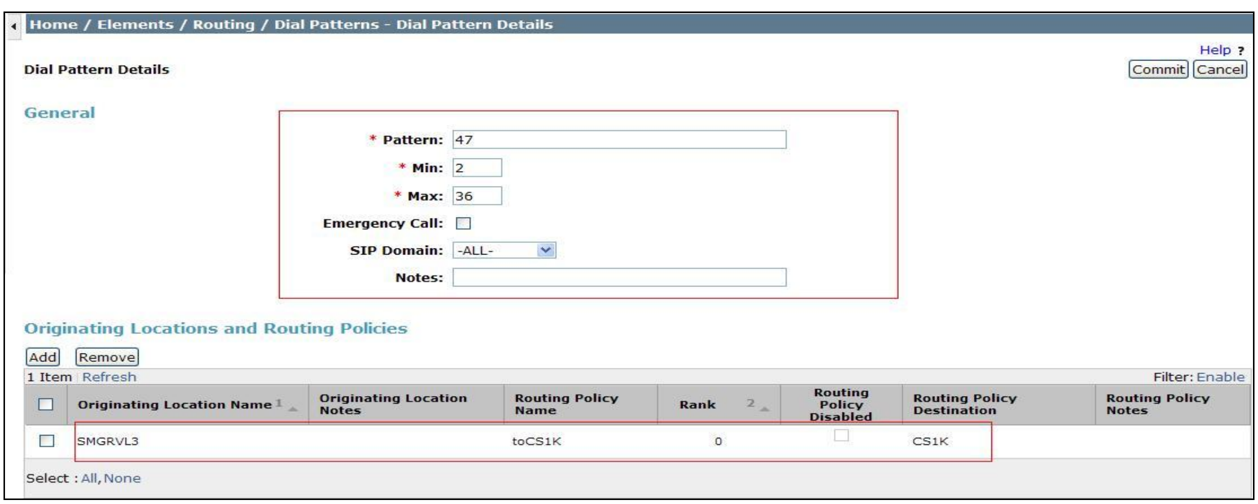
In the **General** section, enter the following values. Use default values for all remaining fields:

- **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)

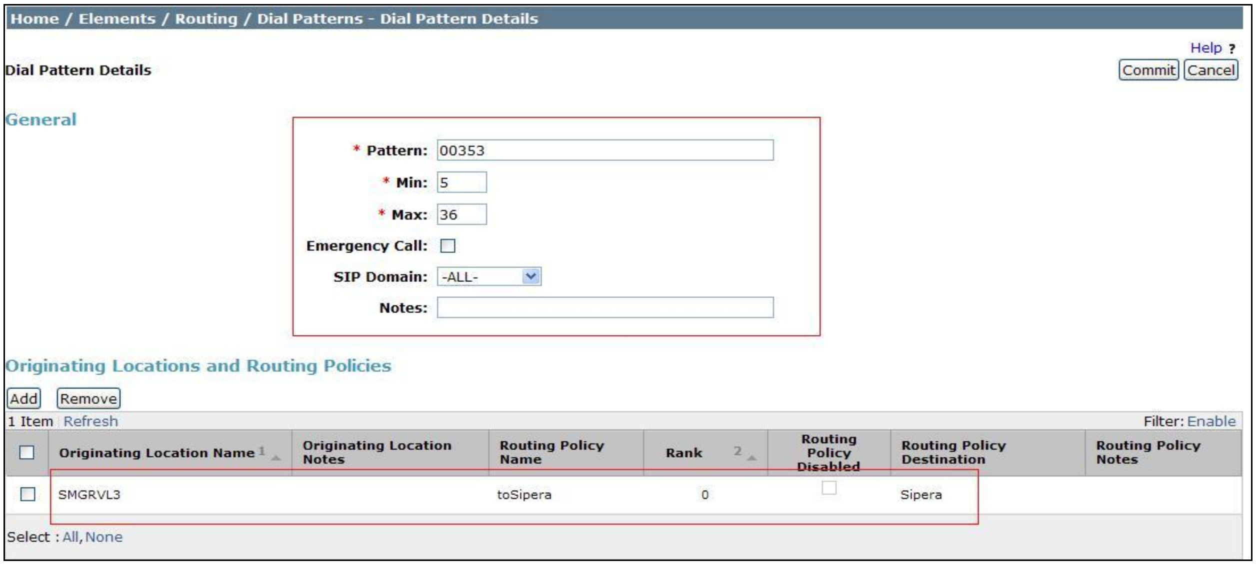
In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria.

- **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.



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.



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:

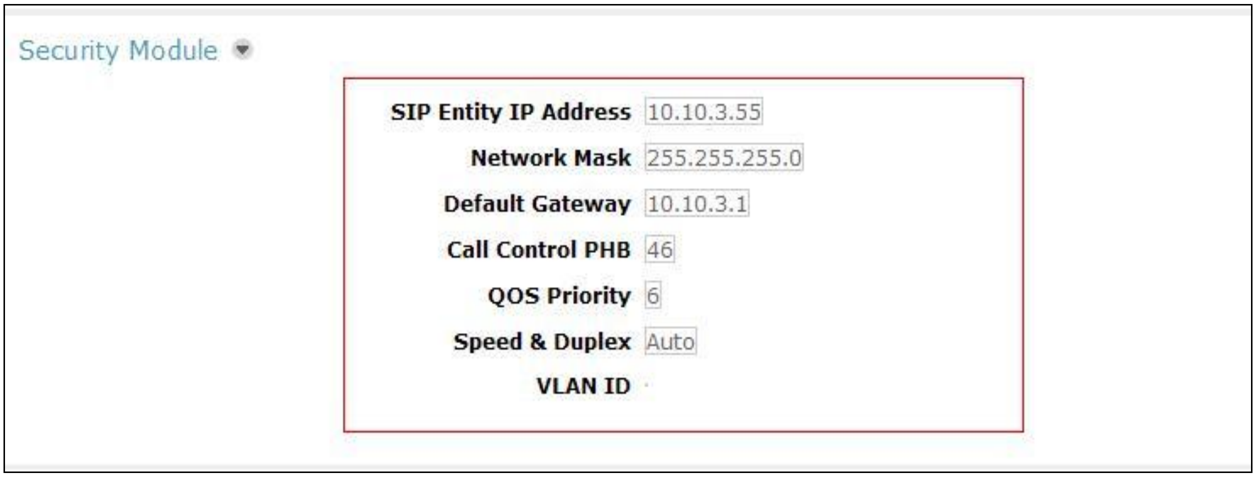
- **SIP Entity Name:** Select the SIP Entity created for Session Manager
- **Description**: Add a brief description (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface

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

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.



7. Configure Avaya Session Border Controller for Enterprise

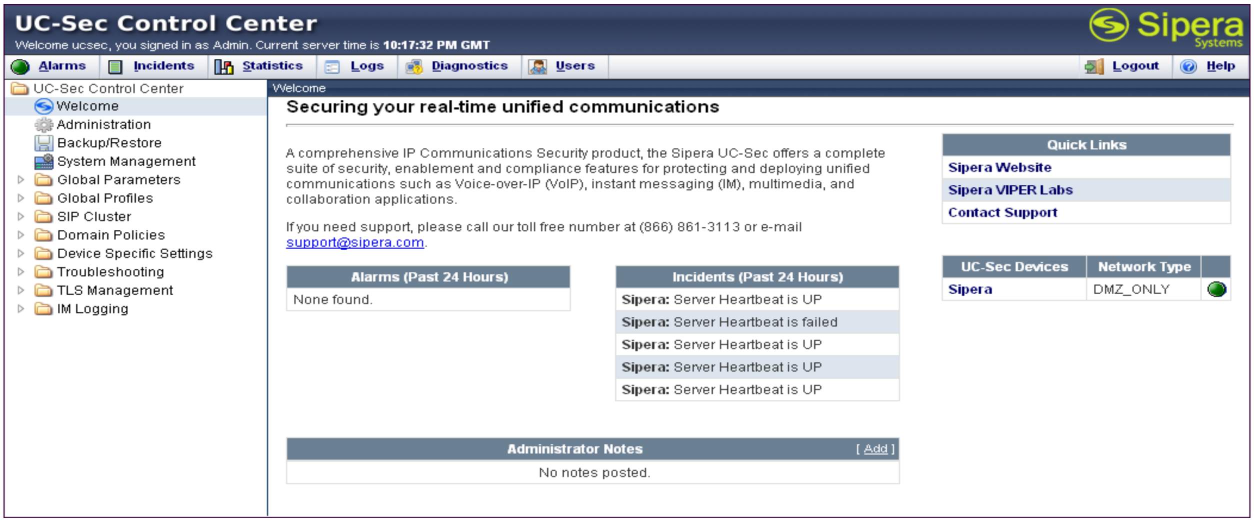
This section describes the configuration of the Session Border Controller. The Avaya SBCE is administered using the UC-Sec Control Center.

7.1. Access Avaya Session Border Controller for Enterprise

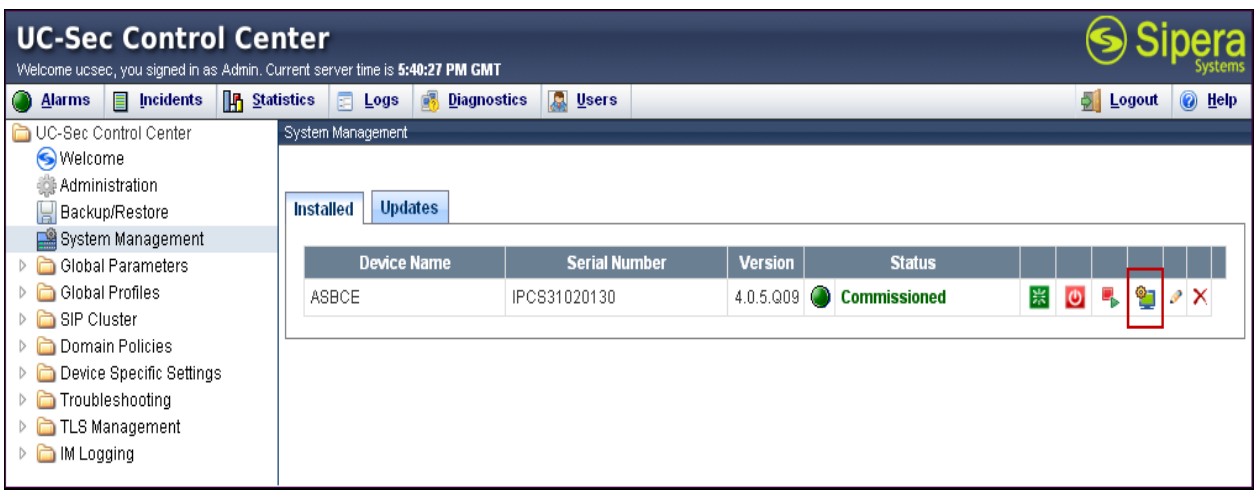
Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. Select the **UC-Sec Control Center**.

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

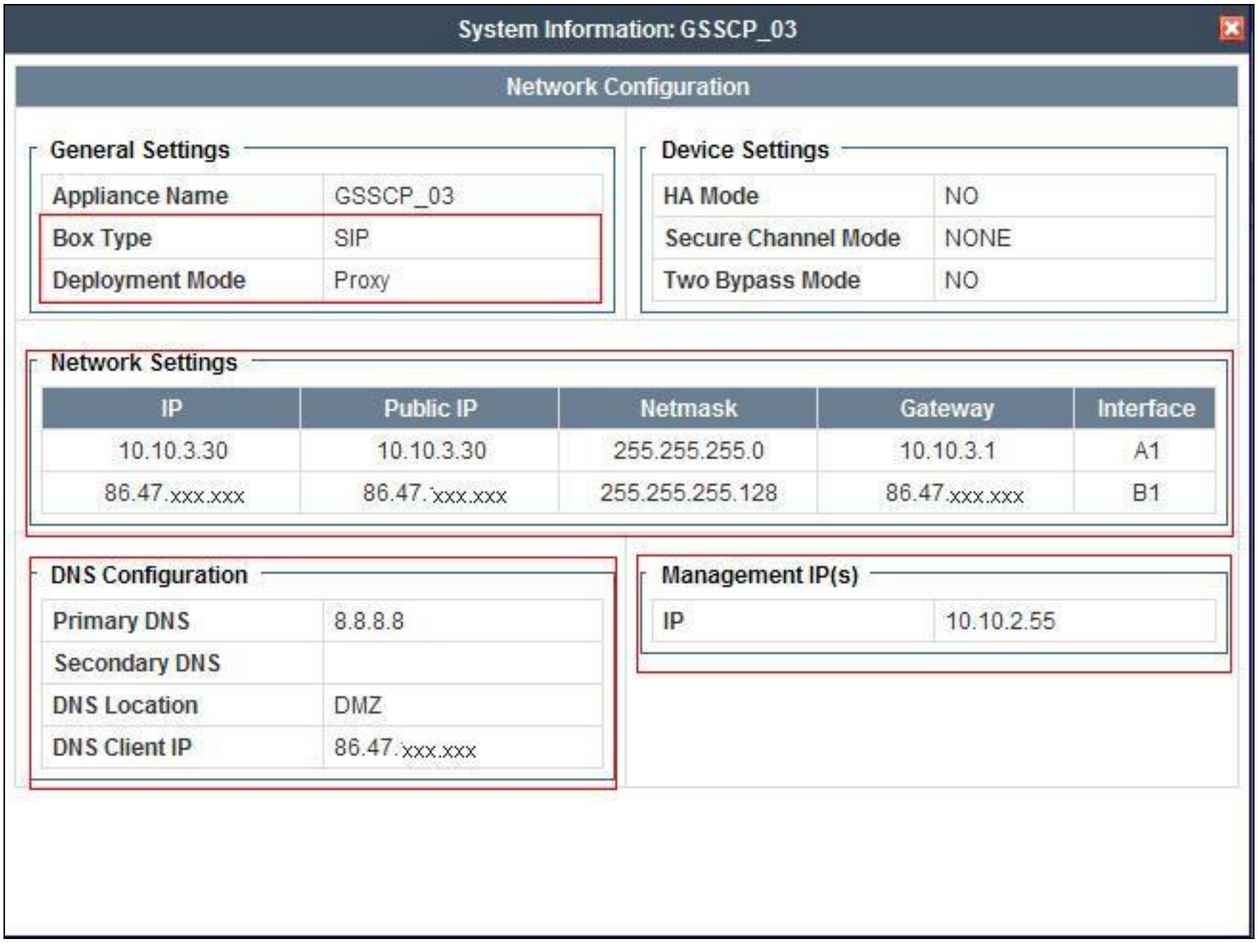
The main page of the UC-Sec Control Center will appear.



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.



7.2. Global Profiles

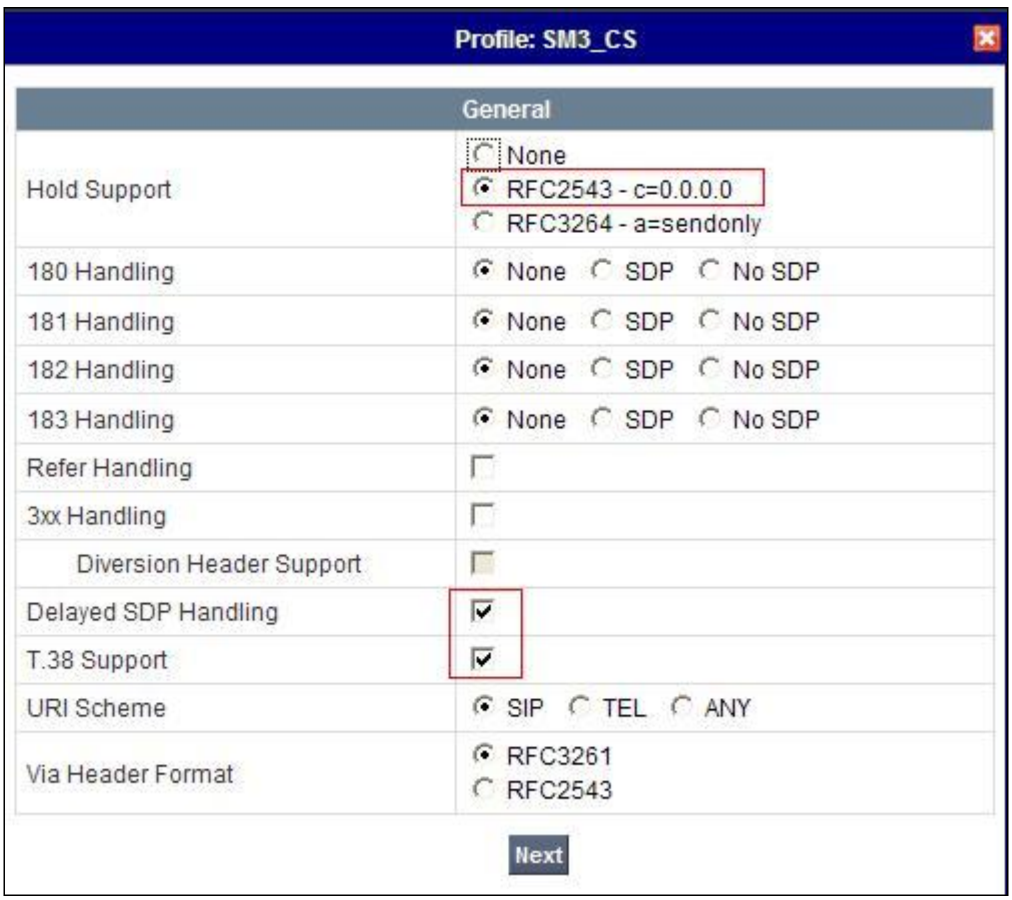
When selected, Global Profiles allows for configuration of parameters across all UC-Sec appliances.

7.2.1. Server Interworking - Avaya Side

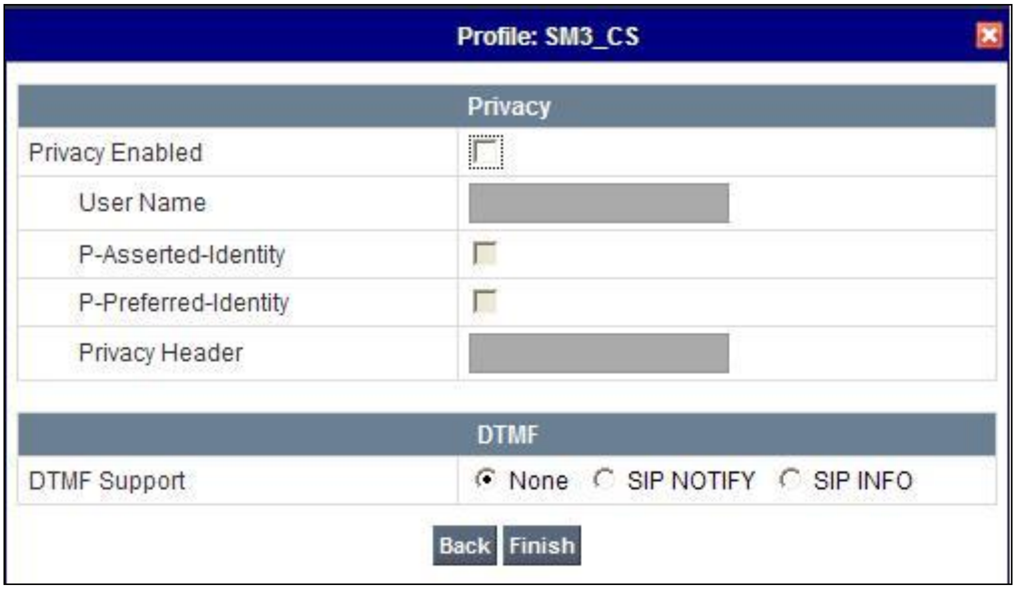
Server Internetworking configures and manages various SIP call server specific capabilities such as call hold and T.38. Navigate to **Global Profiles → Server Interworking** and click on **Add Profile** (Not Shown).

- 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

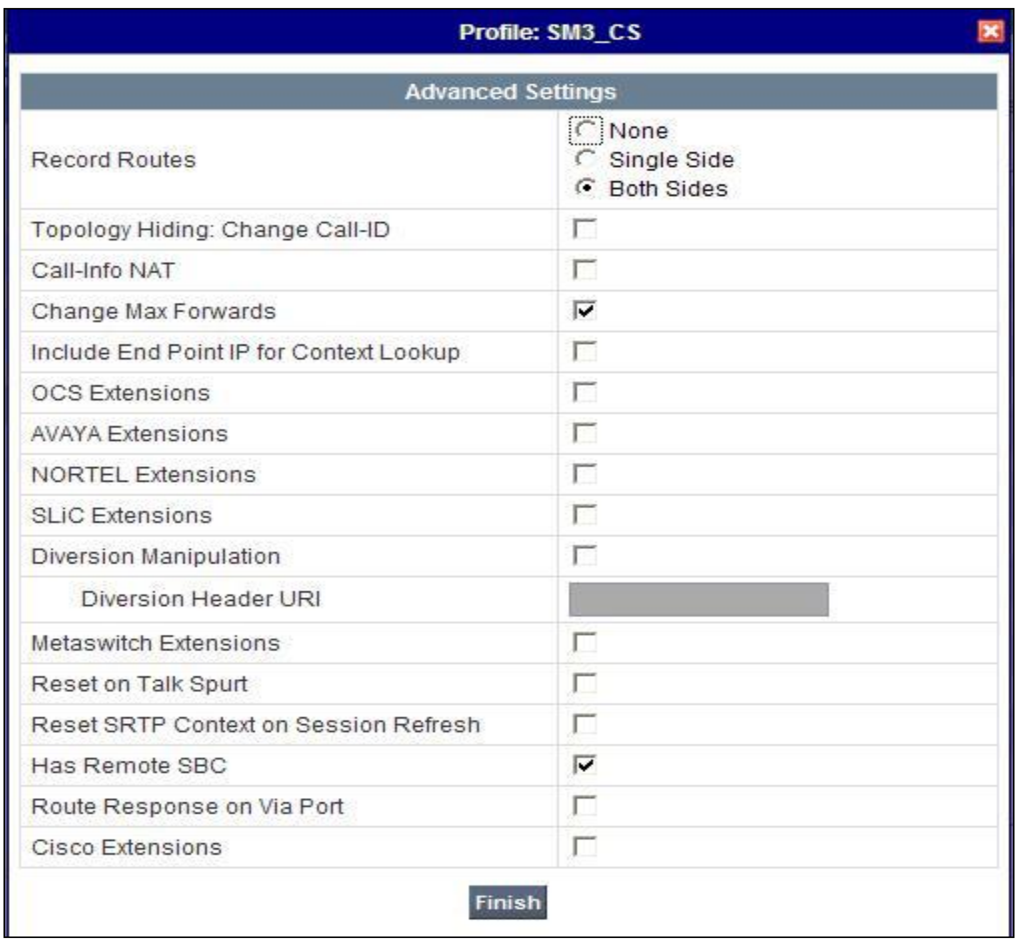
Click **Next** to continue.



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



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

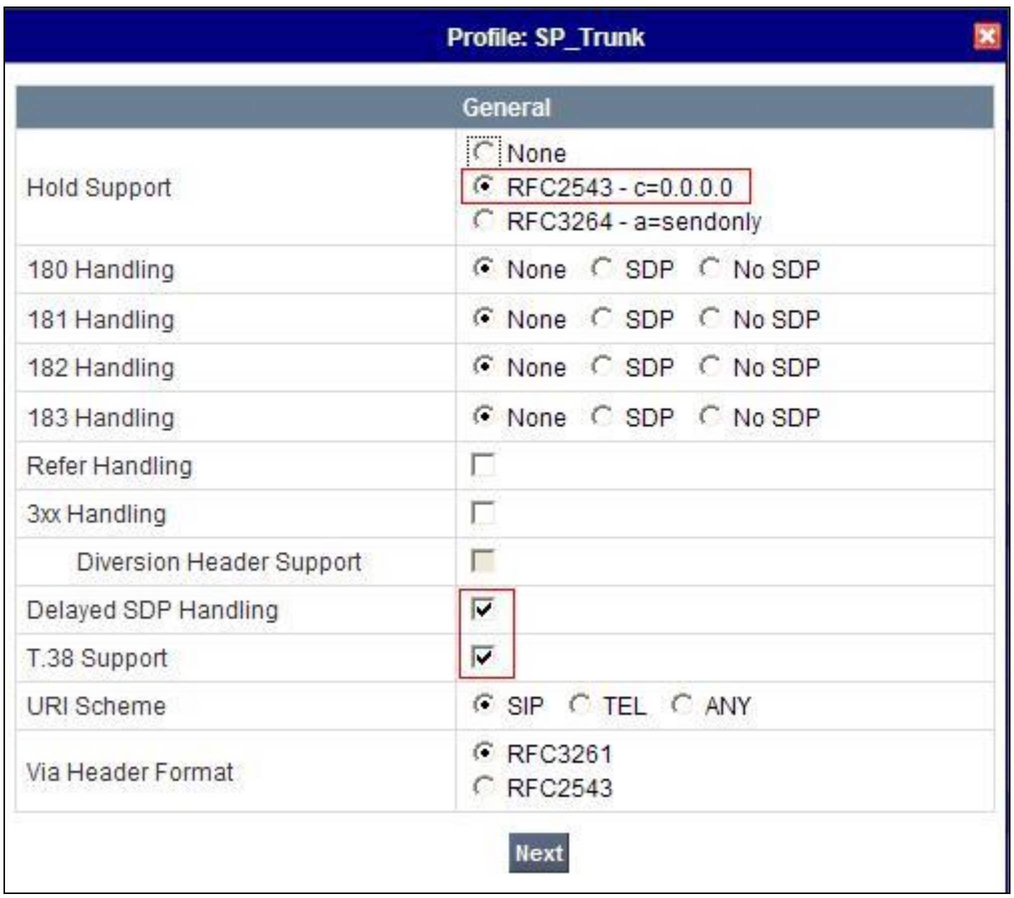


7.2.2. Server Interworking – Phonect Side

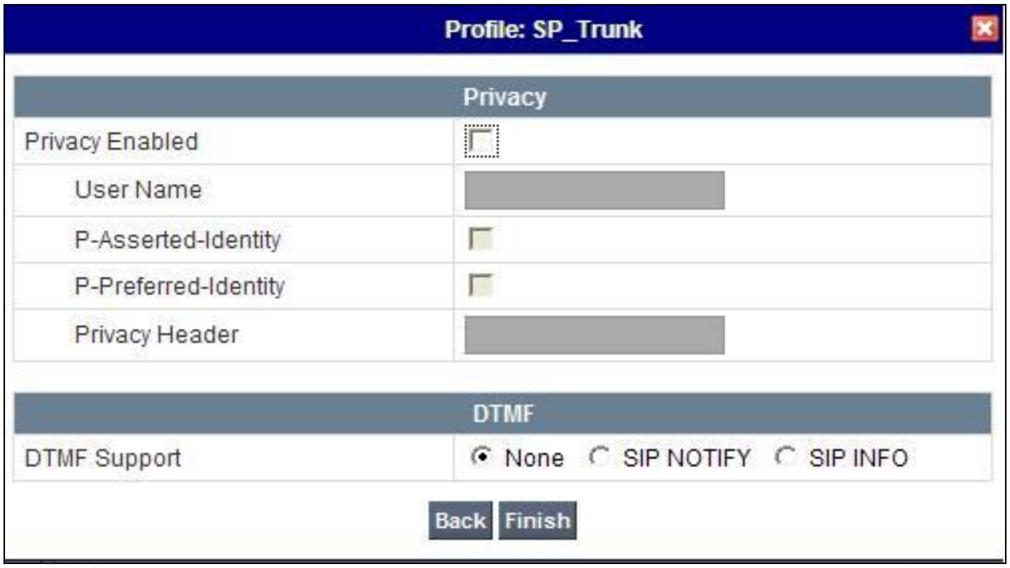
Server Internetworking configures and manages various SIP call server specific capabilities such as call hold and T.38. Navigate to **Global Profiles → Server Interworking** and click on **Add Profile** (Not Shown)**.**

- 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

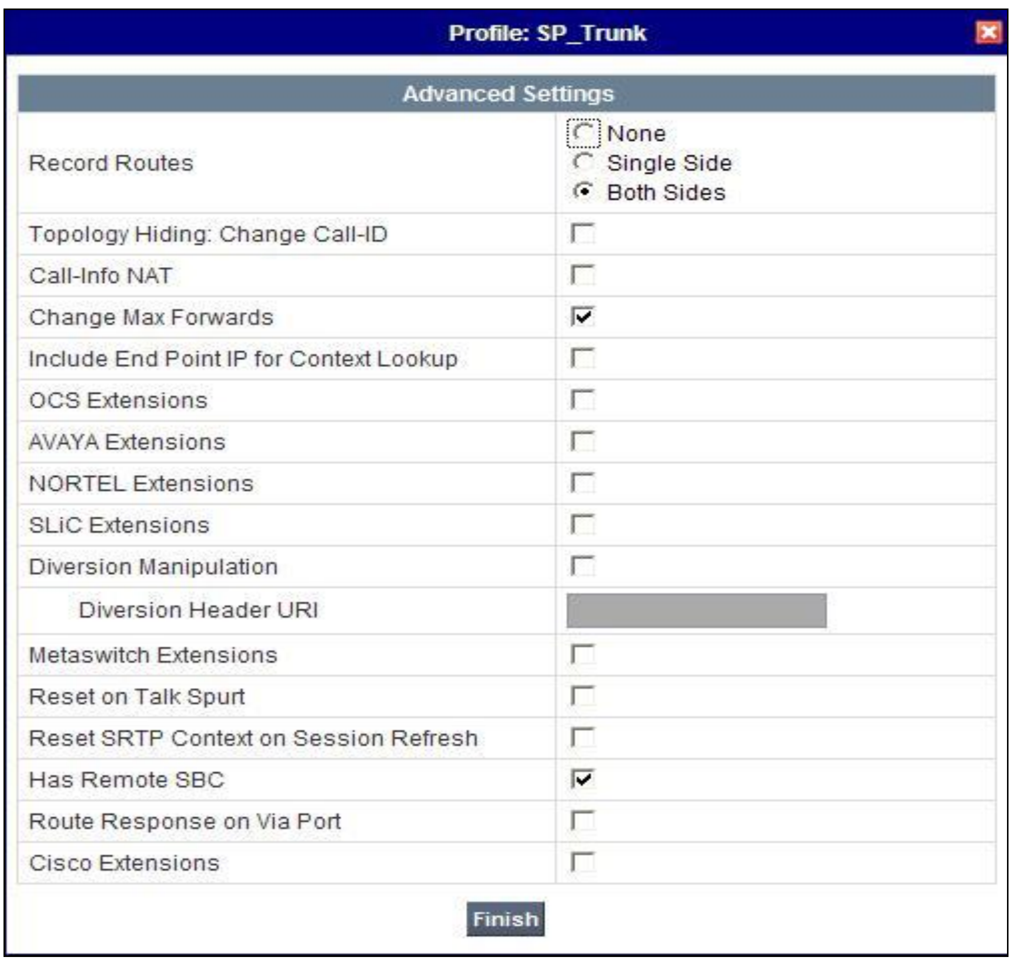
Click **Next** to continue.



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



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



7.2.3. Routing

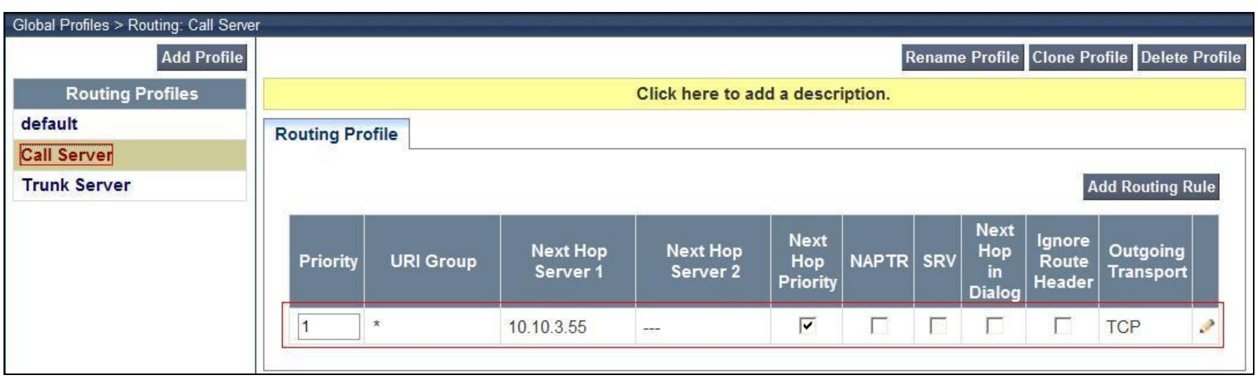
Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport 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 Phonect 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:

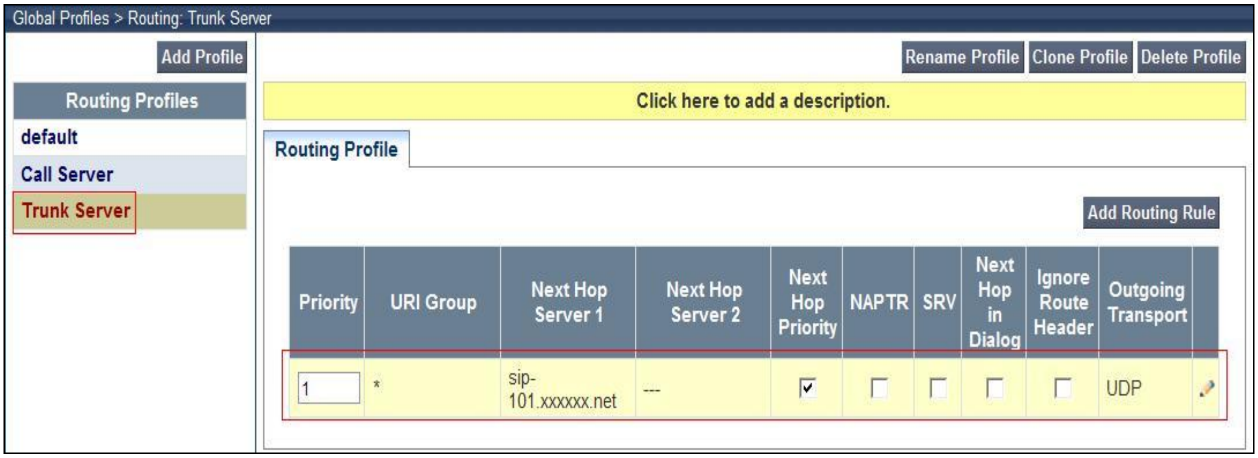
- **URI Group:** Select "*" from the drop down box
- **Next Hop Server 1:** Enter the Domain Name or IP address of the Primary Next Hop server
- **Next Hop Server 2:** (Optional) Enter the Domain Name or IP address of the secondary Next Hop server
- **Routing Priority Based on Next Hop Server**: Checked
- **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**.



The following screen shows the Routing Profile to Phonect.



7.2.4. Server - Configuration

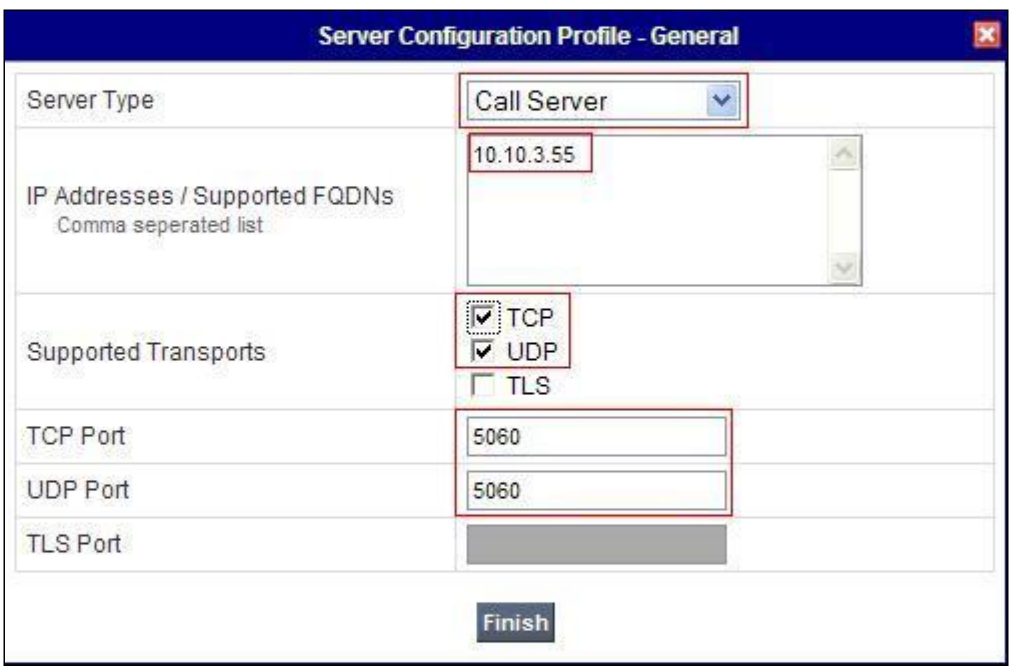
The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs configure and manage various SIP call server specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling parameters and some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

7.2.4.1 Server - Configuration – Avaya Side

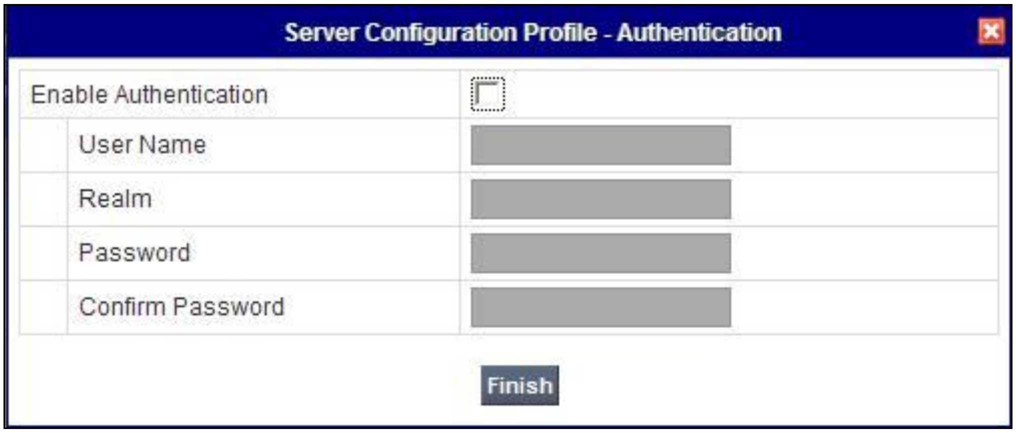
To add a Server Configuration Profile for Session Manger navigate to **UC-Sec Control Center → Global Profiles → Server Configuration** and click on **Add Profile** (not shown). In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Server Type:** Select **Call Server** from the drop-down box
- **IP Addresses / Supported FQDNs:** Enter the IP address of the Session Manager signaling interface 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**

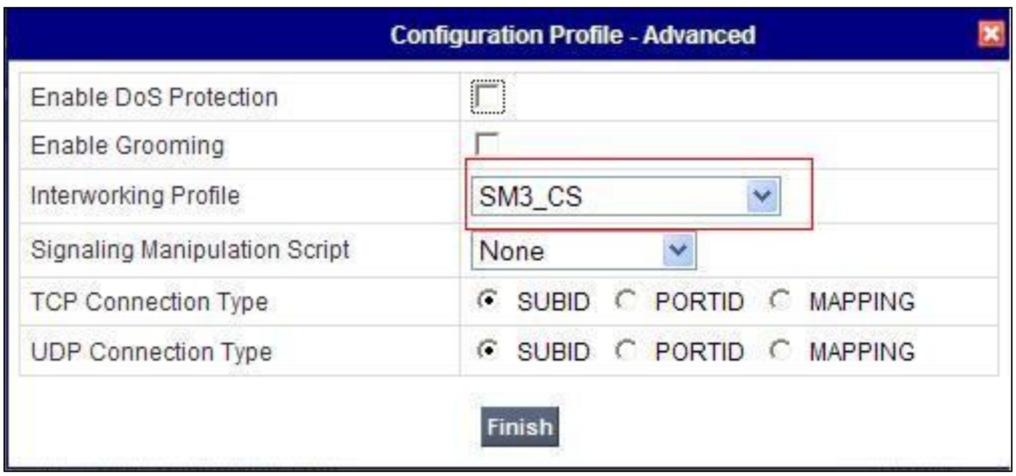
Click **Finish** to continue.



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

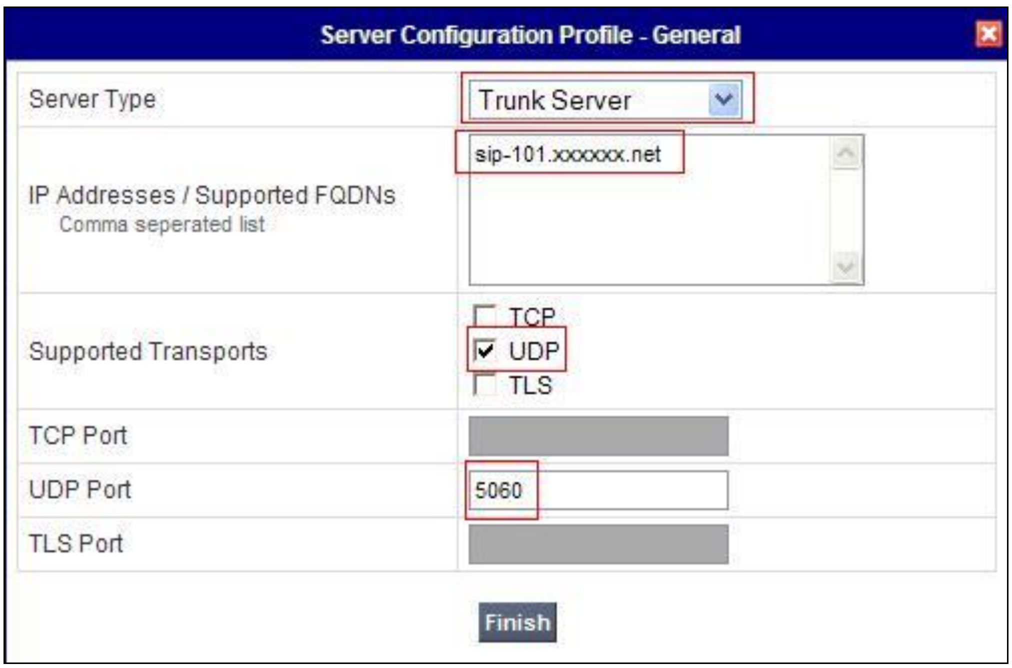


7.2.4.2 Server - Configuration - Phonect

To add a Server Configuration Profile for Session Manger 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 Sever 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.



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

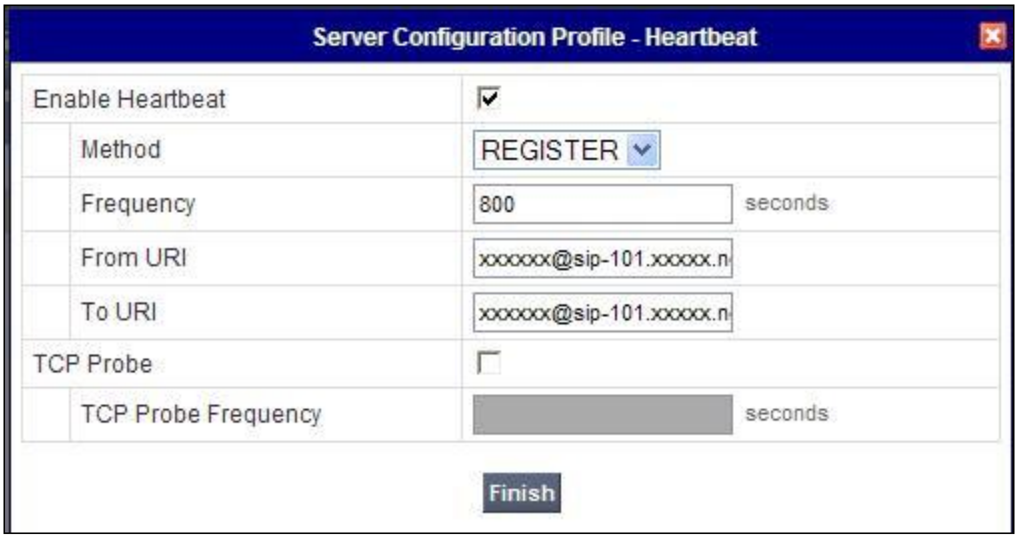
- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider
- **Realm:** Enter realm details provided by the Service Provider
- **Password** Enter password provided by the Service Provider
- **Confirm Password** Re-enter password provided by the Service Provider

Click **Finish** to continue.

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.



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.

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.

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.

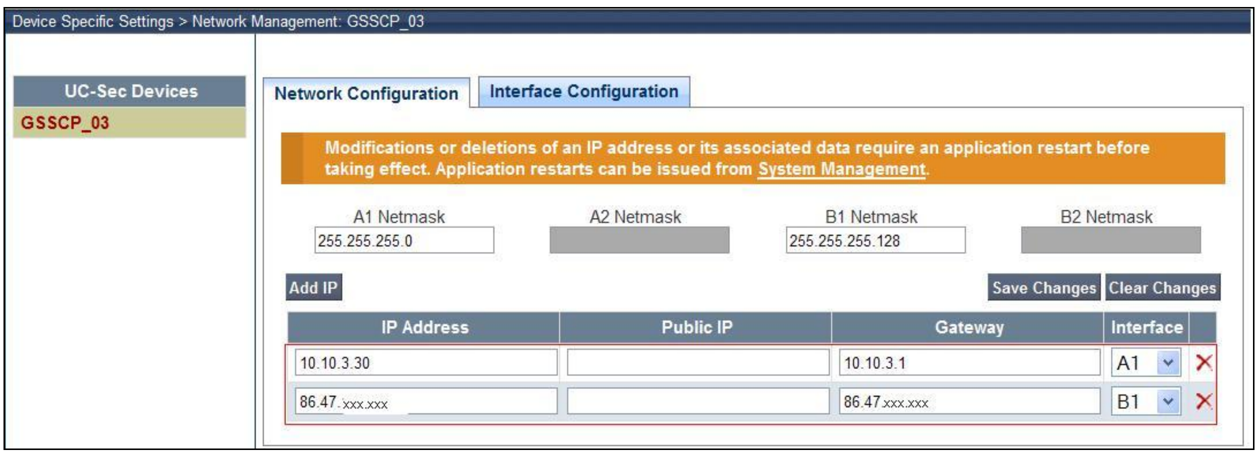
7.3. Device Specific Settings

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.**



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

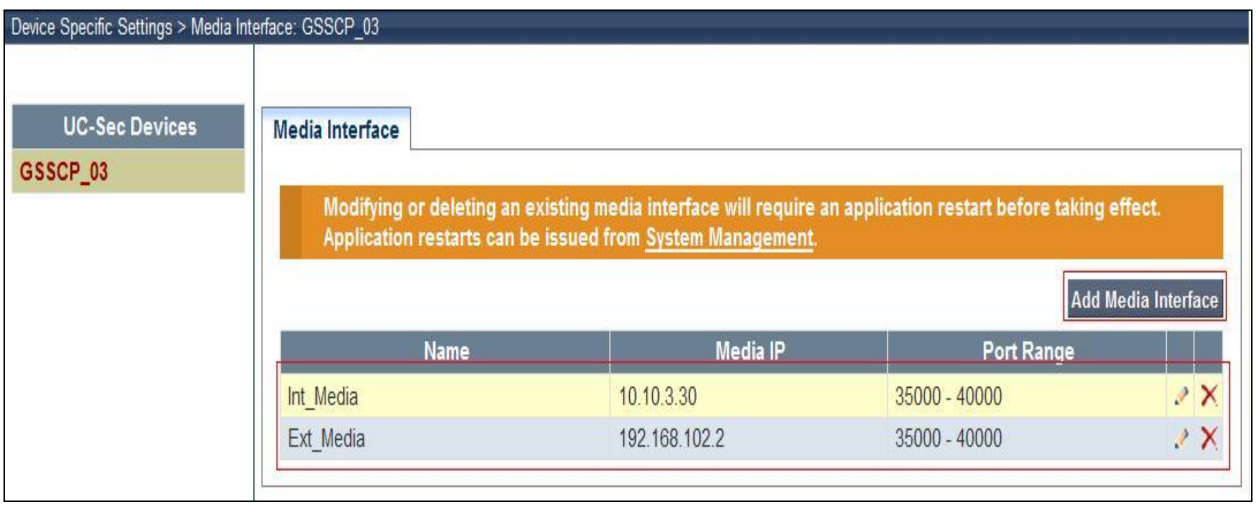
7.3.2. Media Interface

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).**

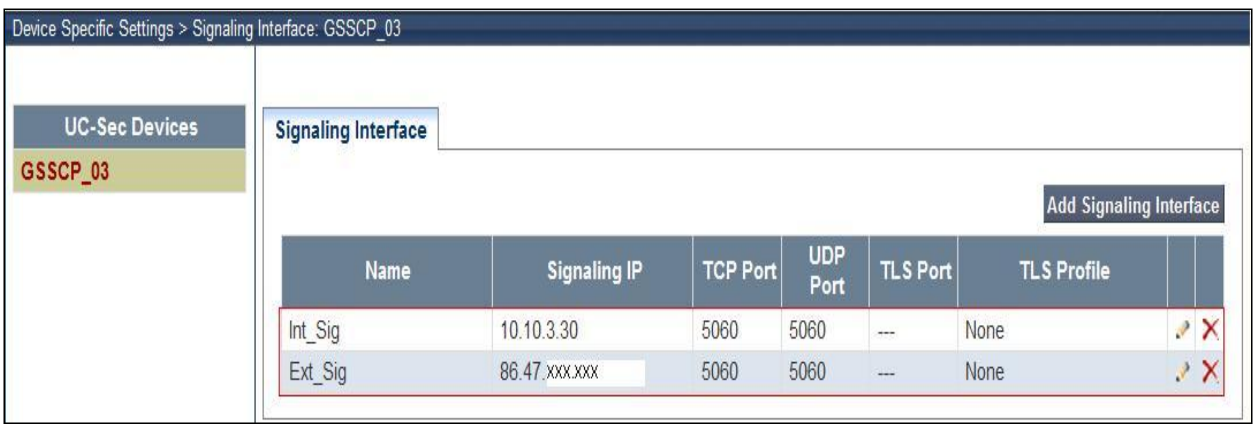


7.3.3. Signalling Interface

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**.

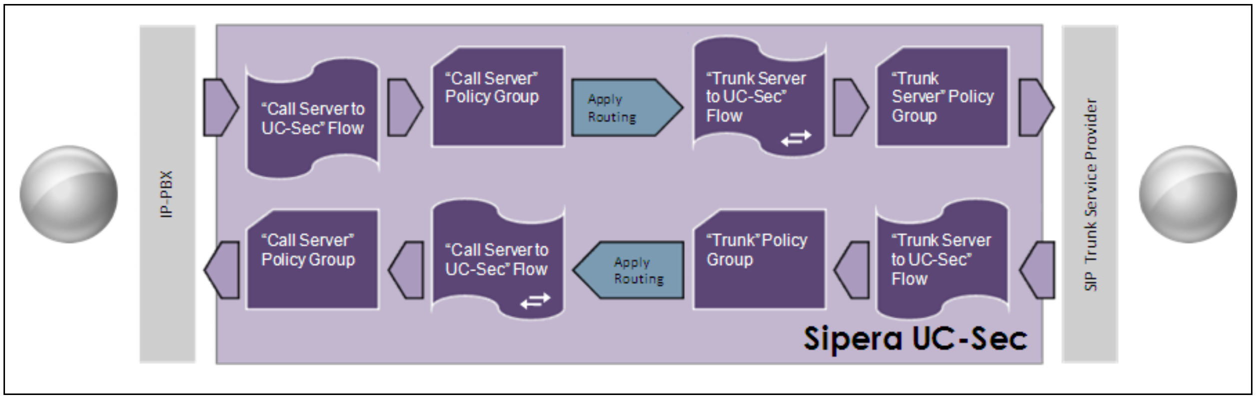
- **Name**: **Int_Sig**
- **Signaling IP**: **10.10.3.30** (Internal Address for calls toward CS1000E)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**
- Select **Add Signaling Interface**
- **Name**: **Ext_Sig**
- **Signaling IP: 86.47.xxx.xxx** (External Address for calls toward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



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.

The following screen shows the Sever Flow for Phonect.

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 authentification 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).

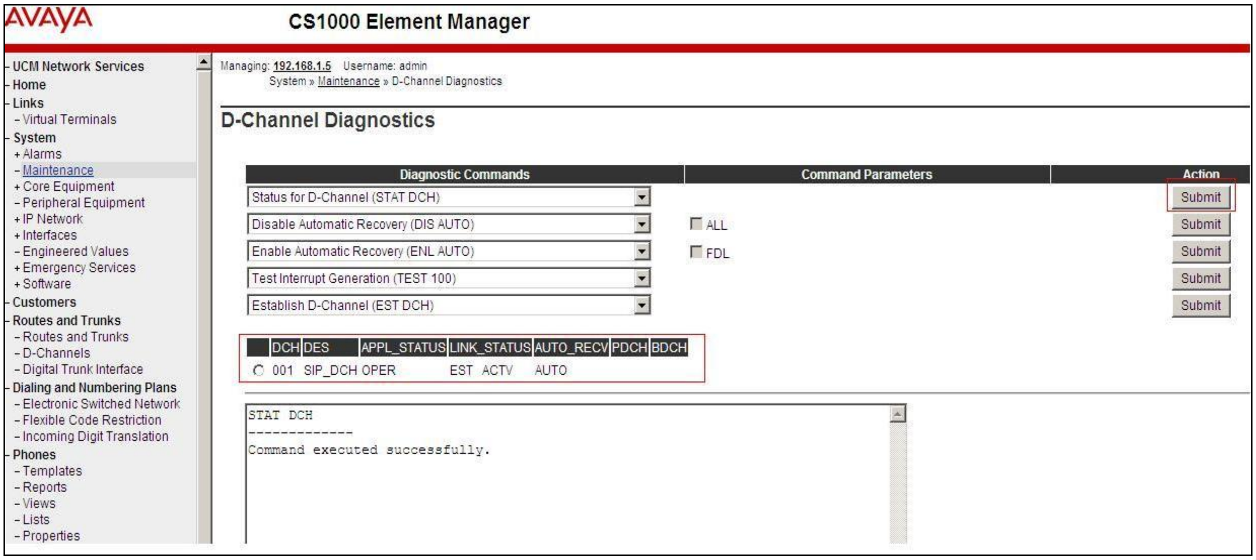
9. Verification

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.

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.

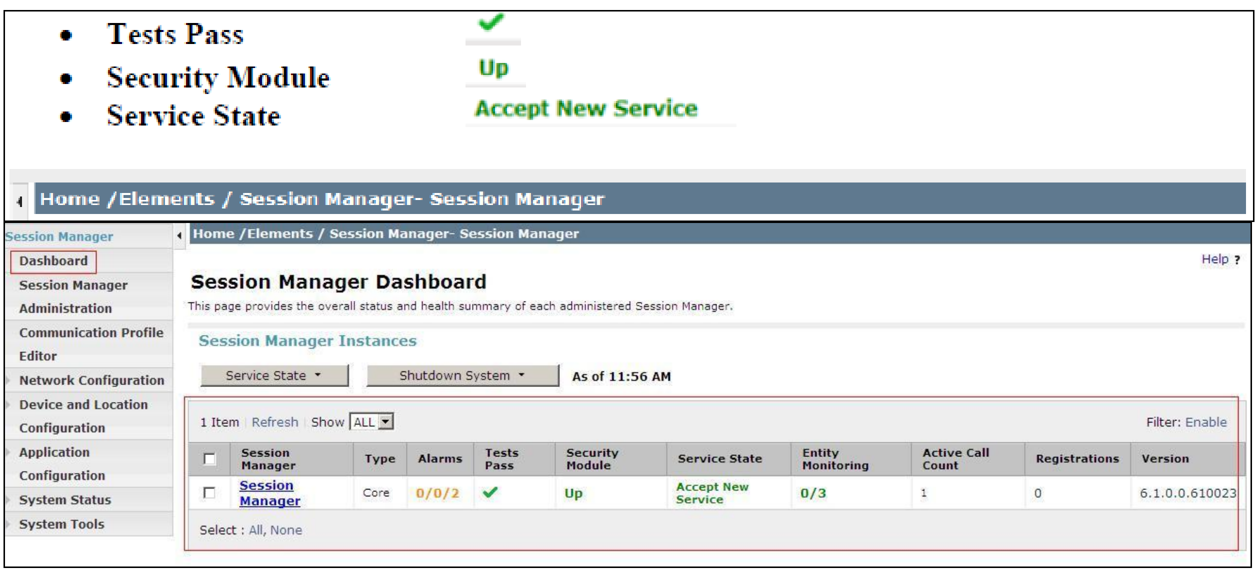
- **APPL_STATUS** Verify status is **OPER**
- **LINK_STATUS** Verify status is **EST ACTV**



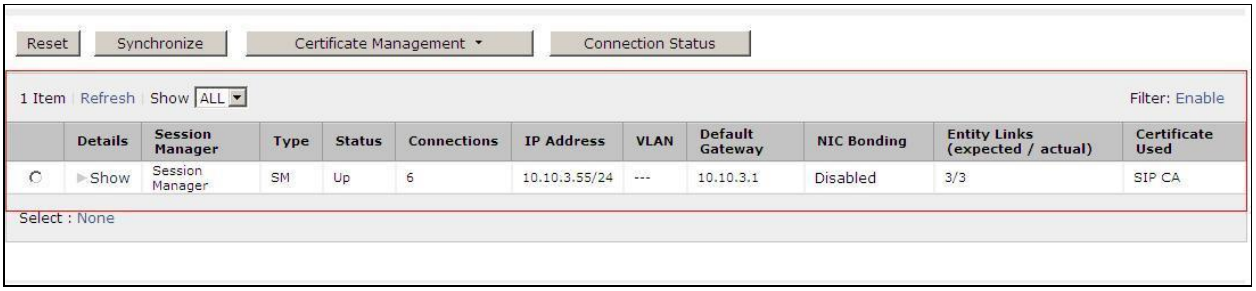
9.2. Verify Avaya Aura® Session Manager Operational Status

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.

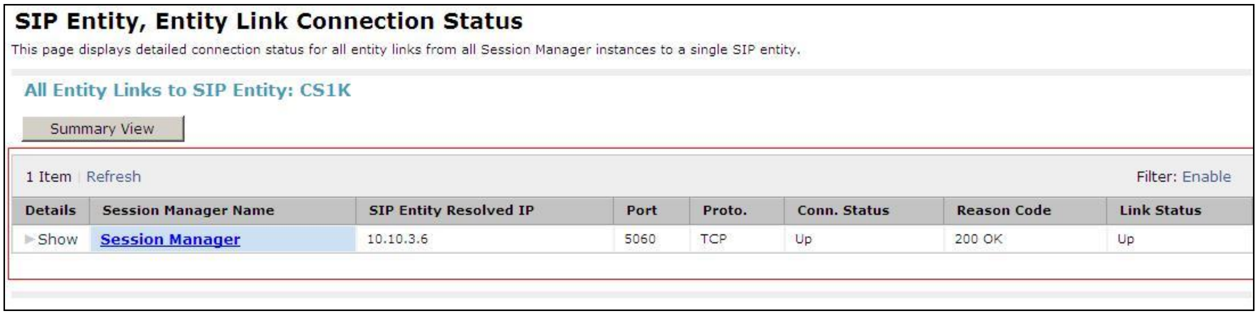


Navigate to **Elements → Session Manager → System Status → Security Module Status** (not shown) to view more detailed status information on the status of Security Module for the specific Session Manager. Verify the **Status** column displays **Up** as shown below.

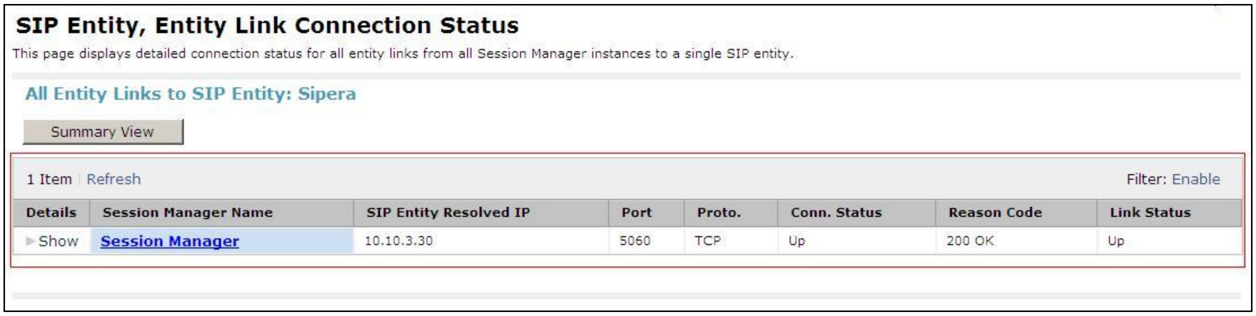


9.2.2. Verify SIP Entity Link Status

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.



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.



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     ISS1:1OF1       p31051_1  01/02/2012  p31051_1.cpl   YES
001  wi00951837     ISS1:1OF1       p31485_1  01/02/2012  p31485_1.cpl   NO
002  wi00946477     ISS1:1OF1       p31426_1  01/02/2012  p31426_1.cpl   NO
003  wi00906163     ISS1:1OF1       p31205_1  01/02/2012  p31205_1.cpl   NO
004  wi00962211     ISS1:1OF1       p31580_1  01/02/2012  p31580_1.cpl   NO
005  wi00877592     ISS1:1OF1       p30880_1  01/02/2012  p30880_1.cpl   NO
006  wi00839134     ISS1:1OF1       p30698_1  01/02/2012  p30698_1.cpl   YES
007  wi00958682     ISS1:1OF1       p31540_1  01/02/2012  p31540_1.cpl   NO
008  wi00868729     ISS1:1OF1       p31163_1  01/02/2012  p31163_1.cpl   NO
009  wi00886321     ISS1:1OF1       p31009_1  01/02/2012  p31009_1.cpl   NO
010  wi00946282     ISS1:1OF1       p31204_1  01/02/2012  p31204_1.cpl   NO
011  wi00841980     ISS1:1OF1       p30618_1  01/02/2012  p30618_1.cpl   NO
012  wi00946681     ISS1:1OF1       p31428_1  01/02/2012  p31428_1.cpl   NO
013  wi00945533     ISS1:1OF1       p31421_1  01/02/2012  p31421_1.cpl   YES
014  wi00843623     ISS1:1OF1       p30731_1  01/02/2012  p30731_1.cpl   YES
015  wi00958776     ISS1:1OF1       p31542_1  01/02/2012  p31542_1.cpl   YES
016  wi00857362     ISS1:1OF1       p30782_1  01/02/2012  p30782_1.cpl   NO
017  wi00865477     ISS1:1OF1       p30893_1  01/02/2012  p30893_1.cpl   YES
018  wi00879526     ISS1:1OF1       p31007_1  01/02/2012  p31007_1.cpl   NO
019  wi00894243     ISS1:1OF1       p31087_1  01/02/2012  p31087_1.cpl   NO
020  wi00890475     p30952          p31048_1  01/02/2012  p31048_1.cpl   NO
```

021	WI00927300	ISS1:1OF1	p30999_1	01/02/2012	p30999_1.cpl	NO
022	wi00856991	ISS1:1OF1	p17588_1	01/02/2012	p17588_1.cpl	NO
023	wi00688381	ISS1:1OF1	p30104_1	01/02/2012	p30104_1.cpl	NO
024	wi00881777	ISS1:1OF1	p25747_1	01/02/2012	p25747_1.cpl	NO
025	WI00853473	ISS1:1OF1	p30625_1	01/02/2012	p30625_1.cpl	NO
026	wi00855423	ISS1:1OF1	p31328_1	01/02/2012	p31328_1.cpl	YES
027	wi00943172	ISS1:1OF1	p31402_1	01/02/2012	p31402_1.cpl	NO
028	wi00865477	ISS1:1OF1	p30898_1	01/02/2012	p30898_1.cpl	YES
029	wi00850521	ISS1:1OF1	p30709_1	01/02/2012	p30709_1.cpl	YES
030	wi00898327	ISS1:1OF1	p31136_1	01/02/2012	p31136_1.cpl	NO
031	wi00871739	ISS1:1OF1	p30856_1	01/02/2012	p30856_1.cpl	NO
032	wi00853031	ISS1:1OF1	p30531_1	01/02/2012	p30531_1.cpl	NO
033	wi00839821	ISS1:1OF1	p30619_1	01/02/2012	p30619_1.cpl	NO
034	wi00854130	ISS1:1OF1	p30443_1	01/02/2012	p30443_1.cpl	NO
035	wi00871969	ISS1:1OF1	p30768_1	01/02/2012	p30768_1.cpl	NO
036	wi00952381	ISS1:1OF1	p31410_1	01/02/2012	p31410_1.cpl	NO
037	wi00946876	ISS1:1OF1	p31430_1	01/02/2012	p31430_1.cpl	NO
038	wi00962557	ISS1:1OF1	p31581_1	01/02/2012	p31581_1.cpl	NO
039	wi00833910	ISS2:1OF1	p30492_2	01/02/2012	p30492_2.cpl	NO
040	wi00903085	ISS1:1OF1	p31164_1	01/02/2012	p31164_1.cpl	NO
041	wi00875425	ISS1:1OF1	p30943_1	01/02/2012	p30943_1.cpl	NO
042	wi00862574	iss1:1of1	p30870_1	01/02/2012	p30870_1.cpl	NO
043	wi00859499	ISS1:1OF1	p30694_1	01/02/2012	p30694_1.cpl	NO
044	wi00925208	ISS1:1OF1	p30986_1	01/02/2012	p30986_1.cpl	NO
045	wi00877442	ISS1:1OF1	p30844_1	01/02/2012	p30844_1.cpl	NO
046	wi00900668	ISS1:1OF1	p30456_1	01/02/2012	p30456_1.cpl	NO
047	wi00867905	ISS1:1OF1	p30640_1	01/02/2012	p30640_1.cpl	NO
048	wi00879322	ISS1:1OF1	p30954_1	01/02/2012	p30954_1.cpl	NO
049	wi00865477	ISS1:1OF1	p30895_1	01/02/2012	p30895_1.cpl	YES
050	wi00951925	ISS1:1OF1	p31486_1	01/02/2012	p31486_1.cpl	NO
051	wi00865477	ISS1:1OF1	p30894_1	01/02/2012	p30894_1.cpl	YES
052	wi00865477	ISS1:1OF1	p30897_1	01/02/2012	p30897_1.cpl	YES
053	wi00865477	ISS1:1OF1	p30892_1	01/02/2012	p30892_1.cpl	YES
054	wi00908933	ISS1:1OF1	p31239_1	01/02/2012	p31239_1.cpl	NO
055	wi00931028	ISS1:1OF1	p31354_1	01/02/2012	p31354_1.cpl	YES
056	wi00932948	ISS1:1OF1	p31077_1	01/02/2012	p31077_1.cpl	NO
057	wi00869695	ISS1:1OF1	p30654_1	01/02/2012	p30654_1.cpl	NO
058	wi00838073	ISS1:1OF1	p30588_1	01/02/2012	p30588_1.cpl	NO
059	wi00852365	ISS1:1OF1	p30707_1	01/02/2012	p30707_1.cpl	NO
060	wi00927321	ISS1:1OF1	p31286_1	01/02/2012	p31286_1.cpl	YES
061	wi00937114	ISS1:1OF1	p31310_1	01/02/2012	p31310_1.cpl	NO
062	wi00877367	ISS1:1OF1	p30534_1	01/02/2012	p30534_1.cpl	NO
063	wi00900096	ISS1:1OF1	p31006_1	01/02/2012	p31006_1.cpl	NO
064	wi00905660	ISS1:1OF1	p27968_1	01/02/2012	p27968_1.cpl	NO
065	wi00925141	ISS1:1OF1	p30802_1	01/02/2012	p30802_1.cpl	NO
066	wi00943748	ISS1:1OF1	p31516_1	01/02/2012	p31516_1.cpl	NO
067	wi00827950	ISS2:1OF1	p30471_2	01/02/2012	p30471_2.cpl	NO
068	wi00937119	ISS1:1OF1	p28005_1	01/02/2012	p28005_1.cpl	NO
069	wi00836981	ISS1:1OF1	p30613_1	01/02/2012	p30613_1.cpl	NO
070	wi00961267	ISS1:1OF1	p30288_1	01/02/2012	p30288_1.cpl	NO
071	wi00936714	ISS1:1OF1	p31379_1	01/02/2012	p31379_1.cpl	NO
072	wi00906022	ISS1:1OF1	p31202_1	01/02/2012	p31202_1.cpl	NO
073	wi00852389	ISS1:1OF1	p30641_1	01/02/2012	p30641_1.cpl	NO
074	wi00857566	ISS1:1OF1	p30766_1	01/02/2012	p30766_1.cpl	NO
075	wi00932204	ISS2:1OF1	p31305_2	01/02/2012	p31305_2.cpl	NO
077	wi00865477	ISS1:1OF1	p30890_1	01/02/2012	p30890_1.cpl	YES
078	wi00873382	ISS1:1OF1	p30832_1	01/02/2012	p30832_1.cpl	NO
079	wi00948274	ISS1:1OF1	p31365_1	01/02/2012	p31365_1.cpl	NO
080	wi00923899	ISS1:1OF1	p31270_1	01/02/2012	p31270_1.cpl	NO
081	wi00856410	ISS1:1OF1	p30749_1	01/02/2012	p30749_1.cpl	NO
082	wi00854415	ISS1:1OF1	p30593_1	01/02/2012	p30593_1.cpl	NO
083	wi00896394	ISS1:1OF1	p30807_1	01/02/2012	p30807_1.cpl	NO
084	wi00826075	ISS1:1OF1	p30452_1	01/02/2012	p30452_1.cpl	NO
085	wi00863876	ISS1:1OF1	p30787_1	01/02/2012	p30787_1.cpl	NO
086	wi00880386	ISS1:1OF1	p30977_1	01/02/2012	p30977_1.cpl	NO
087	wi00840590	ISS1:1OF1	p30767_1	01/02/2012	p30767_1.cpl	NO
088	wi00949627	ISS1:1OF1	p31462_1	01/02/2012	p31462_1.cpl	NO
089	wi00842409	ISS1:1OF1	p30621_1	01/02/2012	p30621_1.cpl	NO
090	wi00865477	ISS1:1OF1	p30896_1	01/02/2012	p30896_1.cpl	YES
091	wi00897096	ISS1:1OF1	p30676_1	01/02/2012	p30676_1.cpl	NO

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092  wi00899584     ISS1:1OF1      p30809_1  01/02/2012  p30809_1.cpl   NO
093  wi00907707     ISS1:1OF1      p31228_1  01/02/2012  p31228_1.cpl   NO
094  wi00949273     ISS1:1OF1      p31411_1  01/02/2012  p31411_1.cpl   NO
095  wi00839255     ISS1:1OF1      p30591_1  01/02/2012  p30591_1.cpl   NO
096  wi00921340     ISS1:1OF1      p31266_1  01/02/2012  p31266_1.cpl   NO
097  wi00903369     ISS1:1OF1      p31165_1  01/02/2012  p31165_1.cpl   NO
098  wi00875701     ISS1:1OF1      p30942_1  01/02/2012  p30942_1.cpl   NO
099  wi00884699     ISS1:1OF1      p31000_1  01/02/2012  p31000_1.cpl   YES
100  wi00834382     ISS1:1OF1      p30548_1  01/02/2012  p30548_1.cpl   NO
101  wi00960133     ISS2:1OF1      p31557_2  01/02/2012  p31557_2.cpl   NO
102  wi00929140     ISS1:1OF1      p31284_1  01/02/2012  p31284_1.cpl   NO
103  wi00948931     ISS1:1OF1      p31407_1  01/02/2012  p31407_1.cpl   NO
104  wi00887744     ISS2:1OF1      p31026_2  01/02/2012  p31026_2.cpl   NO
105  wi00905600     ISS1:1OF1      p31201_1  01/02/2012  p31201_1.cpl   NO
106  wi00869243     ISS1:1OF1      p30848_1  01/02/2012  p30848_1.cpl   NO
107  WI00854150     ISS1:1OF1      p30468_1  01/02/2012  p30468_1.cpl   NO
108  wi00897176     ISS1:1OF1      p30418_1  01/02/2012  p30418_1.cpl   NO
109  wi00903381     ISS1:1OF1      p30421_1  01/02/2012  p30421_1.cpl   NO
110  wi00959854     ISS1:1OF1      p31556_1  01/02/2012  p31556_1.cpl   NO
111  wi00908598     ISS1:1OF1      p31235_1  01/02/2012  p31235_1.cpl   NO
112  wi00903437     ISS1:1OF1      p31167_1  01/02/2012  p31167_1.cpl   NO
113  wi00900766     ISS1:1OF1      p31159_1  01/02/2012  p31159_1.cpl   NO
114  wi00946558     ISS1:1OF1      p31358_1  01/02/2012  p31358_1.cpl   NO
115  wi00932958     ISS1:1OF1      p31115_1  01/02/2012  p31115_1.cpl   NO
116  wi00895090     ISS1:1OF1      p31105_1  01/02/2012  p31105_1.cpl   NO
117  wi00824257     ISS1:1OF1      p30447_1  01/02/2012  p30447_1.cpl   NO
118  wi00895181     ISS1:1OF1      p31106_1  01/02/2012  p31106_1.cpl   NO
119  WI00928455     ISS1:1OF1      p31297_1  01/02/2012  p31297_1.cpl   NO
120  wi00832106     ISS1:1OF1      p30550_1  01/02/2012  p30550_1.cpl   NO
121  wi00953900     ISS1:1OF1      p31494_1  01/02/2012  p31494_1.cpl   NO
122  wi00942734     ISS1:1OF1      p31409_1  01/02/2012  p31409_1.cpl   NO
123  wi00898200     ISS1:1of1      p31274_1  01/02/2012  p31274_1.cpl   NO
124  wi00882293     ISS1:1OF1      p31010_1  01/02/2012  p31010_1.cpl   NO
125  WI00843571     ISS1:1OF1      p30627_1  01/02/2012  p30627_1.cpl   NO
126  wi00835294     ISS1:1OF1      p30565_1  01/02/2012  p30565_1.cpl   NO
127  WI00836292     ISS1:1OF1      p30554_1  01/02/2012  p30554_1.cpl   NO
128  WI00900213     ISS1:1OF1      p30656_1  01/02/2012  p30656_1.cpl   NO
129  wi00921295     ISS1:1OF1      p31265_1  01/02/2012  p31265_1.cpl   NO
130  wi00957141     ISS1:1OF1      p31579_1  01/02/2012  p31579_1.cpl   NO
131  WI00836334     ISS1:1OF1      p30481_1  01/02/2012  p30481_1.cpl   NO
132  wi00858335     ISS1:1OF1      p30819_1  01/02/2012  p30819_1.cpl   NO
133  wi00859123     ISS1:1OF1      p30648_1  01/02/2012  p30648_1.cpl   NO
134  wi00959820     ISS1:1OF1      p31562_1  01/02/2012  p31562_1.cpl   NO
135  wi00905297     ISS1:1OF1      p31195_1  01/02/2012  p31195_1.cpl   NO
136  wi00907697     ISS1:1OF1      p31227_1  01/02/2012  p31227_1.cpl   NO
137  wi00951427     ISS1:1OF1      p31478_1  01/02/2012  p31478_1.cpl   NO
138  wi00883604     ISS1:1OF1      p30973_1  01/02/2012  p30973_1.cpl   NO
139  wi00962955     ISS1:1OF1      p31585_1  01/02/2012  p31585_1.cpl   NO
140  wi00860279     ISS1:1OF1      p30789_1  01/02/2012  p30789_1.cpl   NO
141  wi00909476     ISS1:1OF1      p31340_1  01/02/2012  p31340_1.cpl   NO
142  wi00925218     ISS1:1OF1      p30675_1  01/02/2012  p30675_1.cpl   NO
143  wi00836182     ISS1:1OF1      p30450_1  01/02/2012  p30450_1.cpl   NO
144  wi00841273     ISS1:1OF1      p30713_1  01/02/2012  p30713_1.cpl   NO
145  WI00889786     ISS1:1OF1      p30750_1  01/02/2012  p30750_1.cpl   NO
146  wi00894443     ISS1:1OF1      p31093_1  01/02/2012  p31093_1.cpl   NO
147  wi00896420     ISS1:1OF1      p30867_1  01/02/2012  p30867_1.cpl   NO
148  wi00941500     ISS1:1OF1      p31394_1  01/02/2012  p31394_1.cpl   NO
149  wi00950592     ISS1:1OF1      p31499_1  01/02/2012  p31499_1.cpl   NO
150  wi00927678     ISS1:1OF1      p31399_1  01/02/2012  p31399_1.cpl   NO
151  wi00930864     ISS1:1OF1      p31325_1  01/02/2012  p31325_1.cpl   NO
152  wi00957252     ISS1:1OF1      p31530_1  01/02/2012  p31530_1.cpl   NO
153  wi00880836     ISS1:1OF1      p30976_1  01/02/2012  p30976_1.cpl   NO
154  wi00865477     ISS1:1OF1      p30891_1  01/02/2012  p30891_1.cpl   YES
155  wi00896680     ISS1:1OF1      p30357_1  01/02/2012  p30357_1.cpl   NO
156  wi00856702     ISS1:1OF1      p30573_1  01/02/2012  p30573_1.cpl   NO
157  wi00897082     ISS1:1OF1      p31124_1  01/02/2012  p31124_1.cpl   NO
158  wi00853178     ISS1:1OF1      p30719_1  01/02/2012  p30719_1.cpl   NO
159  wi00938555     ISS1:1OF1      p30881_1  01/02/2012  p30881_1.cpl   YES
160  WI00839794     ISS1:1OF1      p28647_1  01/02/2012  p28647_1.cpl   NO
MDP>LAST SUCCESSFUL MDP REFRESH :2012-01-24 11:17:37(Local Time)
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```
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
```

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