Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.5, Avaya Aura® Session Manager R6.1 and Avaya Session Border Controller for Enterprise to support BT Wholesale/HIPCOM SIP Trunk Service – Issue 1.0

Abstract

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

BT is a member of the DevConnect 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 lab.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between BT Wholesale/HIPCOM SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager and Avaya Communication Server 1000E (CS1K) connected to BT Wholesale/HIPCOM SIP Trunk Service via an Avaya Session Border Controller for Enterprise (Avaya SBCE). Customers using this Avaya SIP-enabled enterprise solution with BT Wholesale/HIPCOM's 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 CS1K, Session Manager and Avaya SBCE. The enterprise site was configured to use the SIP Trunk Service provided by BTW/HIPCOM.

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

The interoperability test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DDI numbers assigned by BTW/HIPCOM. Incoming PSTN calls were made to Unistim, SIP, Digital and Analogue telephones at the enterprise
- Outgoing calls from the enterprise to the PSTN were made from Unistim, SIP, Digital and Analogue telephones
- G.729 annex b (silence suppression) is not supported by BTW/HIPCOM's SIP Trunk Service and thus was not tested
- Calls using G.729, G.711A and G.711MU codec's were tested
- Fax calls to/from a Group 3 fax machine to a PSTN connected fax machine using the T.38 mode
- User features such as hold and resume, transfer, conference, call forwarding, etc
- Caller ID Presentation and Caller ID Restriction
- DTMF transmission using RFC 2833 with successful Voice Mail navigation for inbound and outbound calls
- Call coverage and call forwarding for endpoints at the enterprise site

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for BTW/HIPCOM SIP Trunk Service with the following observations.

- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested
- No Emergency Services numbers tested as test calls to these numbers should be pre-arranged with the Operator
- Mobile-X call to service DN, once secondary dial tone is heard a call is then placed off-net to the PSTN. This call scenario only works with patch p30260_1.ntl loaded on the CS1000E
- Mobile-X mid call features were not tested
- Blind Transfer back out to PSTN only work with plug-in 501 enabled on the CS1K. This enables the reINVITE method. No ring back tone heard when the call is transferred but this is by design intent if the UPDATE method isn't used

2.3. Support

For technical support on BTW/HIPCOM products please contact the following website: http://www.hipcom.co.uk/support or http://ipvoicesupport.btwholesale.com.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to BTW/HIPCOM using SIP Trunks. Located at the enterprise site are Session Manager, Avaya SBCE and a Communication Server 1000E. 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 one-X Communicator), Avaya Digital telephone, Analogue telephone and fax machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

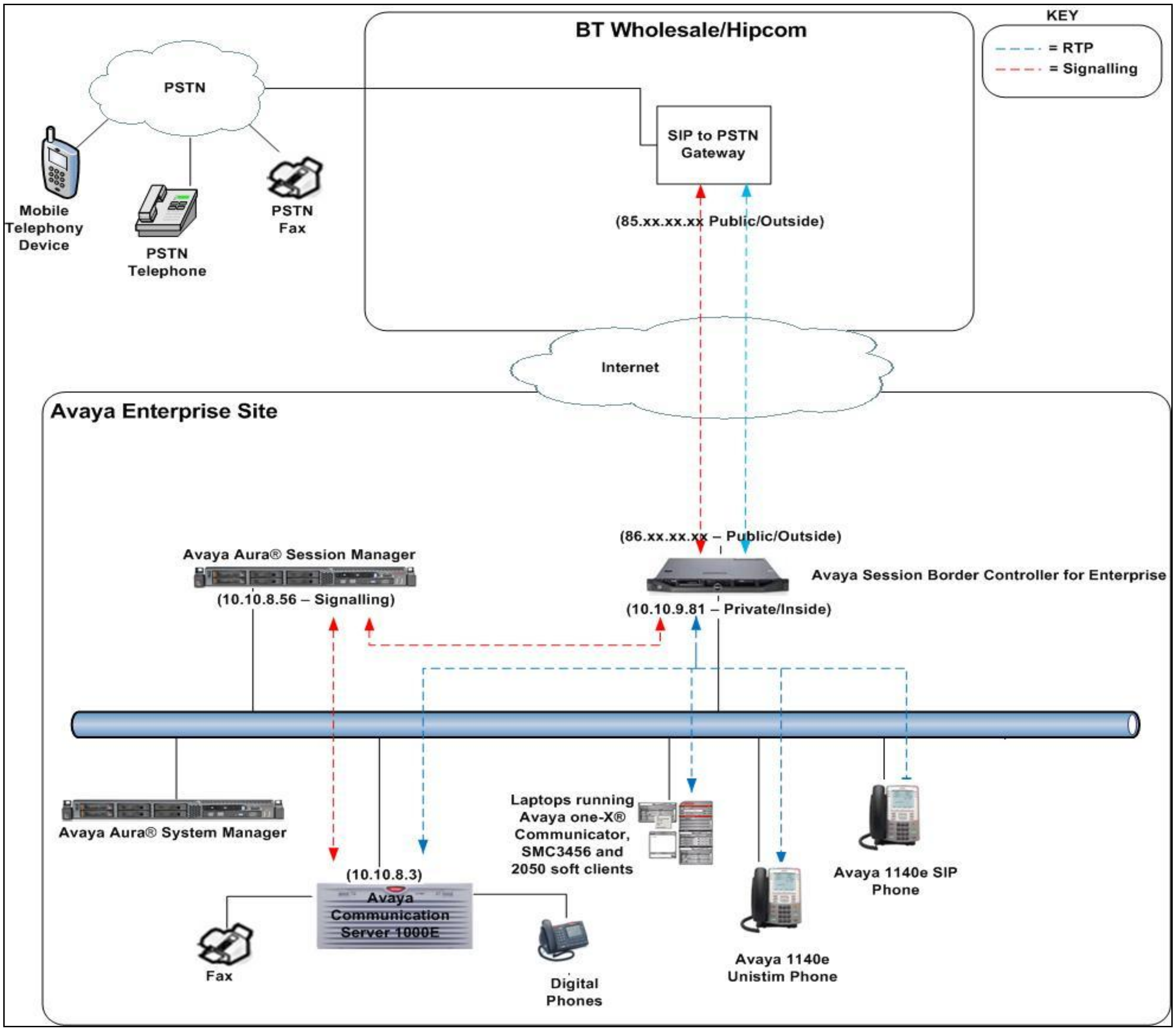


Figure 1: BT Wholesale/HIPCOM SIP Trunk topology with Avaya Communication Server 1000E

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 server running Avaya Aura® Session Manager	Avaya Aura® Session Manager R6.1 Service Pack 6 (6.1.6.0.616008)
Avaya S8800 server running Avaya Aura® System Manager	Avaya Aura® System Manager R6.1 Service Pack 6 (6.1.10.1.1806)
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	Avaya Communication Server 1000E R7.5 Version 7.50.17 Deplist: CPL_X21_07_50Q All CS1000E patches listed in **Appendix A**
Dell R210 Server	Avaya Session Border Controller for Enterprise version R4.0.5.Q09
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 1140e: 0625C8G FW 1230e: 062AC8G
Avaya 1140e and 1230 SIP Telephones	FW: 04.03.09.00.bin
Avaya SMC 3456	Version 2.6 build 57666
Avaya one-X® Communicator	Version cs6.1.0.25
Avaya 2050 IP Soft phone	Version 4.02.0062
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
BTW/HIPCOM SIP Trunk Service	Acme Packet 4500 Net-Net SBC ver SCX6.1.0 Broadsoft - ver 14 Sevice Pack 9 Configuration version -

5. Configure Avaya Communication Server 1000E

This section describes the steps required to configure Communication Server 1000E for SIP Trunking and also the necessary configuration for terminals (Analogue, Digital, SIP and IP phones). SIP trunks are established between Communication Server 1000E and Session Manager. These SIP trunks carry SIP Signaling associated with BTW/HIPCOM's SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from the Avaya SBCE, through which the BTW/HIPCOM SIP Service directs incoming SIP messages to Communication Server 1000E (see **Figure 1**). Once a SIP message arrives at Communication Server 1000E, 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 Communication Server 1000E and may be first subject to outbound features such as route selection, digit manipulation and class of service restrictions. Once Communication Server

1000E 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 BTW/HIPCOM's network. Specific Communication Server 1000E configuration was performed using Element Manager and the system terminal interface. The general installation of the Communication Server 1000E, System Manager and Session Manager is presumed to have been previously completed and is not discussed here.

5.1. Logging into the Avaya Communication Server 1000E

Login to the CS1K using SSH to the ELAN IP address of the Call Server. Once logged in type **csconsole** (not shown)**,** this will take the user into the vxworks shell of the call server. Next type **logi** (not shown), the user will then be asked to login with correct credentials. Once logged in the user can then progress to load any overlay.

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 Communication Server 1000E 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 BTW/HIPCOM'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 Communication Server 1000E.

```
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       32767    LEFT 32752    USED    15
```

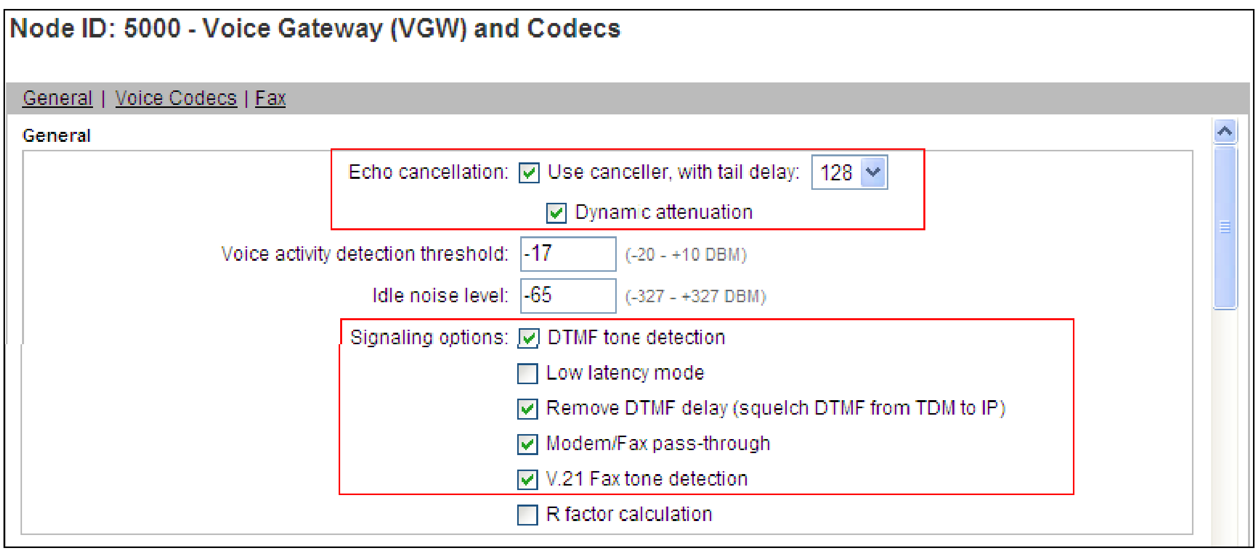
Load overlay 21 and confirm the customer is setup to use **ISDN** trunks (see below).

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

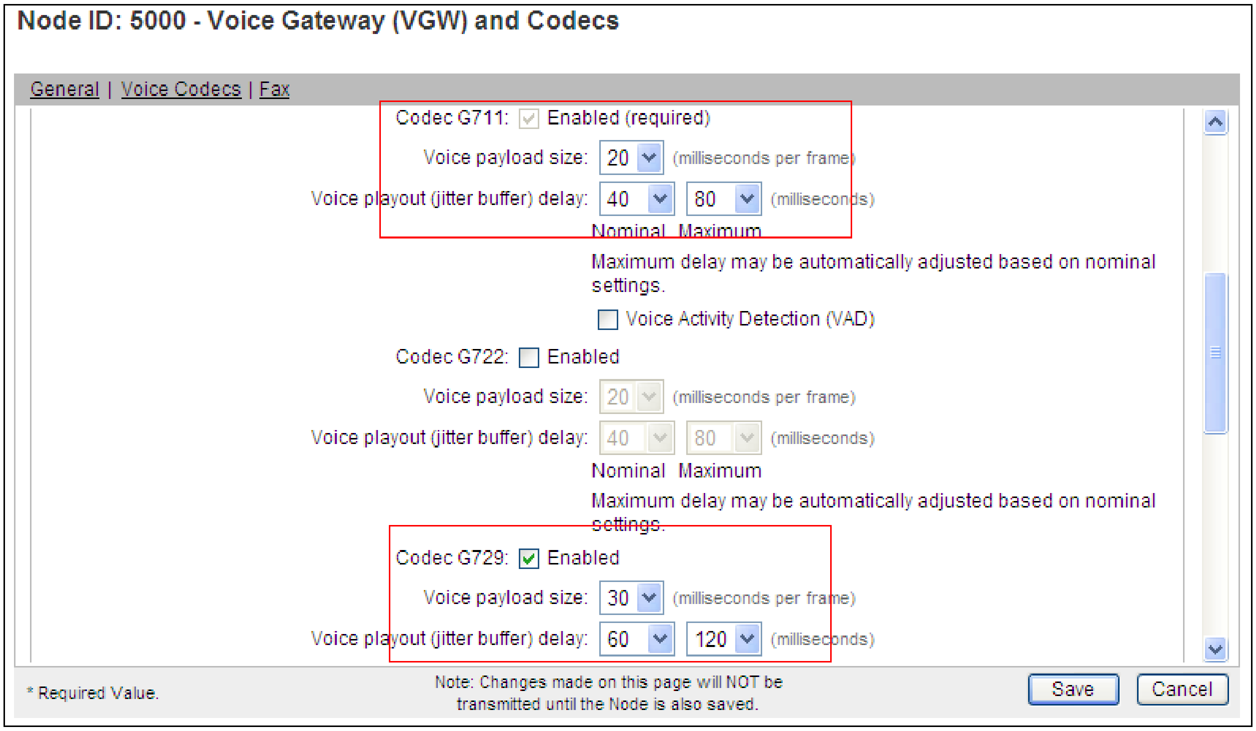
TYPE NET_DATA
CUST 00
OPT RTD
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
ISDN YES
```

5.3. Configure Codec's for Voice and FAX operation

BTW/HIPCOM SIP Trunk service supports G.711A, G.711MU and G.729 voice codec and T.38 FAX transmissions. Using the Communication Server 1000E element manager sidebar, navigate to the **IP Network → IP Telephony Nodes → Node Details → Voice Gateway (VGW) and Codecs** property page and configure the Communication Server 1000E General codec settings as shown in the screenshot below. The values highlighted are required for correct operation; most of the options are turned on by default but it is good practice to ensure that they are set as shown below



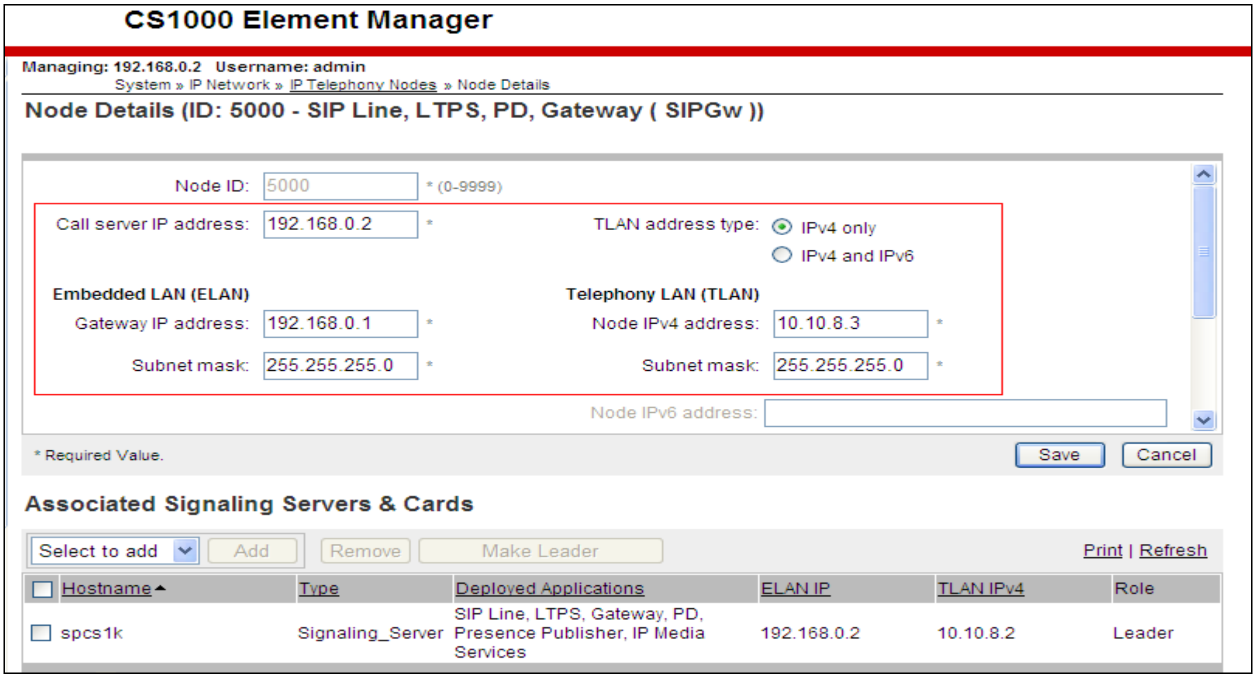
Scroll down the page and configure the CS1000E to use **Codec G.711** and **G.729**. G.711 is enabled as default and cannot be disabled or enabled on the CS1000E. Default values were configured for G.729. This aligns with what BTW/HIPCOM support on their SIP network.



Finally configure **Fax** settings as highlighted in the screenshot below. System defaults were used. Please note T.38 cannot be disabled or enabled at the Node level and by default is enabled. Turning T.38 on or off is done at the endpoint level, by using different class of service as shown in **Section 5.8 Configure Analogue, Digital and IP Telephones**.

5.4. Virtual Trunk Gateway Configuration

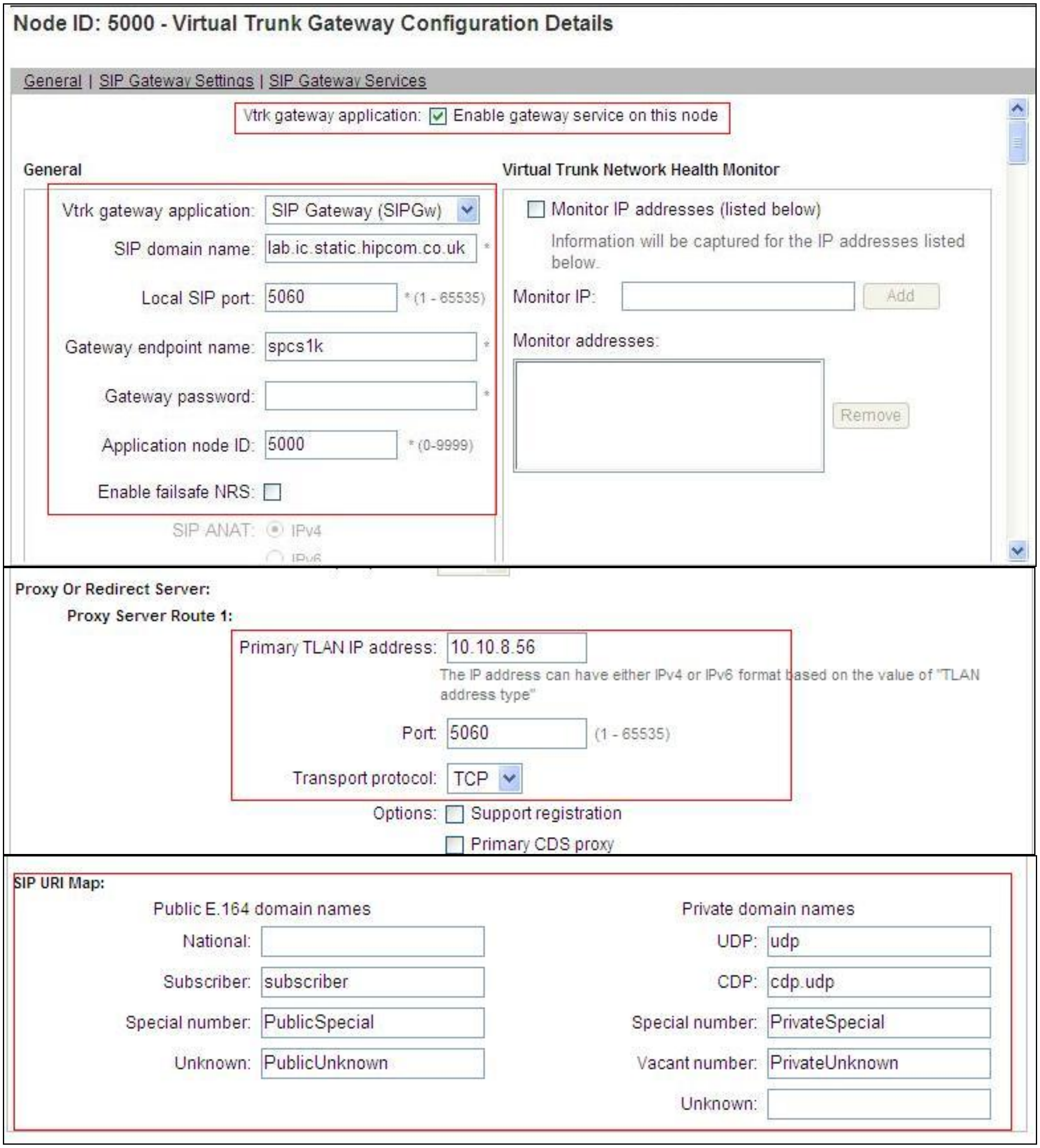
Use Communication Server 1000E Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and verify the highlighted section is completed with the correct IP addresses and subnet masks of the Node. At this stage the call server 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 CS1K it is the Node ip that is used (see **Section 6.4** for more details).



The next two screenshots show the SIP Virtual Trunk Gateway configuration, navigate to **System → IP Networks → IP Telephony Nodes → Node Details → Gateway (SIPGW) Virtual Trunk Configuration Details** 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 used in the enterprise, in this case **lab.ic.static.hipcom.co.uk**
- **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 (NRS) is used for registration of the endpoint. In this network a NRS is not used so anything can be data filled
- **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 **5000**
- **Proxy or Redirect Server:** Primary TLAN IP address is the signalling interface of the Session Manager. The **Transport protocol** used for **SIP**, in this case is **TCP**

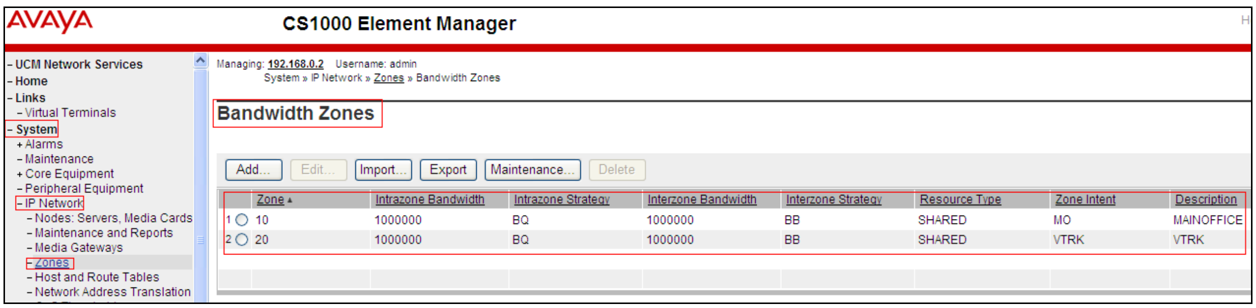
- **Options:** Support registration is checked
- **SIP URI Map: Public E.164 - National** and **Private - Unknown** are left blank. All other fields in the SIP URI Map are left with default values.



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 that are not shared with other resources and best practice dictates that IP telephones and Media Gateways are all placed in a separate zone than SIP trunks. In the sample configuration SIP trunks use zone 20 and IP Telephones use zone 10, system defaults were used for each zone other than the parameter configured for **Zone Intent**. For SIP Trunks (zone 20), **VTRK** is configured for **Zone Intent**. For IP Telephones (zone 10), **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.



5.6. Configure Incoming Digit Conversion Table

A limited number of Direct Dial Inwards (DDI) numbers were available; an IDC table was configured to translate incoming PSTN numbers to five digit local telephone extension numbers. The last four 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, Analogue, Digital or Unistim telephones depending on the particular test case being executed.

5.7. Configure SIP Trunks

Communication Server 1000E virtual trunks will be used for all inbound and outbound PSTN calls to BTW/HIPCOM's SIP Trunk Service. Five separate steps are required to configure Communication Server 1000E virtual trunks:-

- Configure a D-Channel Handler (DCH); configure using the Communication Server 1000E system terminal and overlay 17
- Configure a SIP trunk Route Data Block (RDB); configure using the Communication Server 1000E system terminal and overlay 16
- Configure SIP trunk members; configure using the Communication Server 1000E system terminal and overlay 14
- Configure a Route List Block (RLB); configure using the Communication Server 1000E system terminal and overlay 86
- Configure Special Prefix Numbers (SPN's); configure using the Communication Server 1000E system terminal and overlay 90

The following is an example DCH configuration for SIP trunks. Load **Overlay 17** at the Communication Server 1000E system terminal and enter the following values. The highlighted entries are required for correct SIP trunk operation. Exit overlay 17 when completed.

```
Overlay 17
ADAN     DCH 10
  CTYP DCIP
  DES  VIR_TRK
  USR  ISLD
  ISLM 4000
  SSRC 1800
  OTBF 32
  NASA YES
  IFC  SL1
  CNEG 1
  RLS  ID  5
  RCAP ND2
  MBGA NO
  H323
    OVLR NO
    OVLS NO
```

Next, configure the SIP trunk Route Data Block (RDB) using the Communication Server 1000E 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 100
TYPE RDB
CUST 00
ROUT 100
DES  VIR_TRK
TKTP TIE
NPID_TBL_NUM   0
ESN  NO
RPA  NO
CNVT NO
SAT  NO
RCLS EXT
VTRK YES
ZONE 00020
PCID SIP
CRID NO
NODE 5000
DTRK NO
ISDN YES
    MODE ISLD
    DCH  10
    IFC  SL1
    PNI  00001
    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 1600
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 Communication Server 1000E 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
new 30
TN   160 0 0 0
DATE
PAGE
DES  VIR_TRK
TN   160 0 00 00  VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 00020
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK  ANLG
NCOS 0
RTMB 100 1
CHID 1
TGAR 1
STRI/STRO WNK WNK
SUPN YES
AST  NO
IAPG 0
CLS  TLD DTN CND ECD WTA LPR APN THFD XREP SPCD MSBT
     P10 NTC
TKID
AACR NO
```

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                        FCI  0
new                               FSNI 0
CUST 0                            BNE  NO
FEAT rlb                          DORG NO
RLI  24                           SBOC NRR
ELC  NO                           PROU 1
ENTR 0                            IDBB DBD
LTER NO                           IOHQ NO
ROUT 100                          OHQ  NO
TOD  0 ON  1 ON  2 ON  3 ON       CBQ  NO
     4 ON  5 ON  6 ON  7 ON
VNS  NO                           ISET 0
SCNV NO                           NALT 5
CNV  NO                           MFRL 0
EXP  NO                           OVLL 0
FRL  0
DMI  0
CTBL 0
ISDM 0
```

Next, configure Special Prefix Number(s) (SPN) which users will dial to reach PSTN numbers. Use the Communication Server 1000E system terminal and overlay 90. The following are some example SPN 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.

SPN 999	SPN 90	SPN 2	SPN 15
FLEN 3	FLEN 7	FLEN 7	FLEN 3
ITOH NO	ITOH NO	ITOH NO	ITOH NO
CLTP NONE	CLTP NONE	CLTP NONE	CLTP NONE
RLI 24	**RLI 24**	**RLI 24**	**RLI 24**
SDRR NONE	SDRR NONE	SDRR NONE	SDRR NONE
ITEI NONE	ITEI NONE	ITEI NONE	ITEI NONE

5.8. Configure Analogue, 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 shown in the screen shot. A unique five digit number is entered for the **KEY 00** and **KEY 01** value. The value for **CFG_ZONE** is the same value used in **Section 5.4** for **VIRTUALSETS**.

```
Overlay 20 IP Telephone configuration
DES  1140
TN   096 0 01 16  VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
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 8000 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 8000 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   000 0 09 08  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 8010 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 8010 0
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     02 DSP
     03 MSB
     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
```

Analogue telephones are also configured using **Overlay 20**, the following example shows an Analogue 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. In the class of service (**CLS**) field **DTN** is required if the telephone uses DTMF dialing. Values **FAXA** and **MPTD** configure the port for T.38 Fax transmissions.

```
Overlay 20 – Analogue Telephone Configuration
DES  500
TN   100 0 00 03
TYPE 500
CDEN 4D
CUST 0
MRT

ERL 00000
WRLS NO
DN   8021
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
```

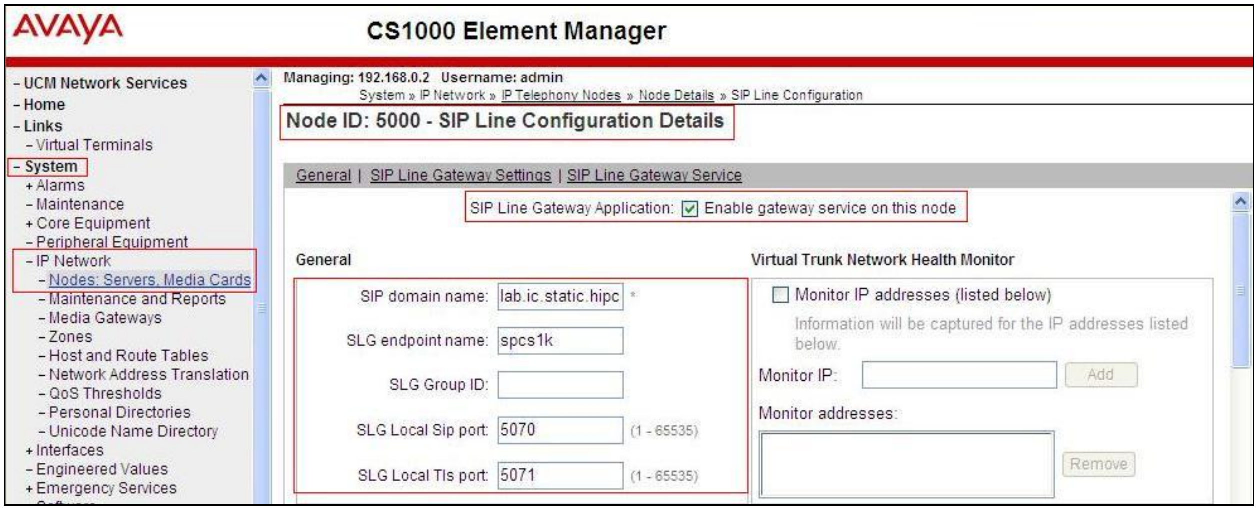
5.9. Configure the SIP Line Gateway Service

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 Communication Server 1000E 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 78
  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. The value for **SIP Domain Name** must match that configured in **Section 6.1**.

- **SIP Line Gateway Application: Enable gateway service on this node**, check the box to enable
- **SIP domain name:** Enter the SIP domain, in this case **lab.ic.static.hipcom.co.uk**
- **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**



5.10. Configure SIP Line Telephones

When SIP Line service configuration is completed, use the Communication Server 1000E 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 5.4**. A unique telephone number is entered for value **KEY 00**. The value for **KEY 01** is comprised of the **UAPR** value (set to 78 previously in this section) and the telephone number used in **KEY 00**.

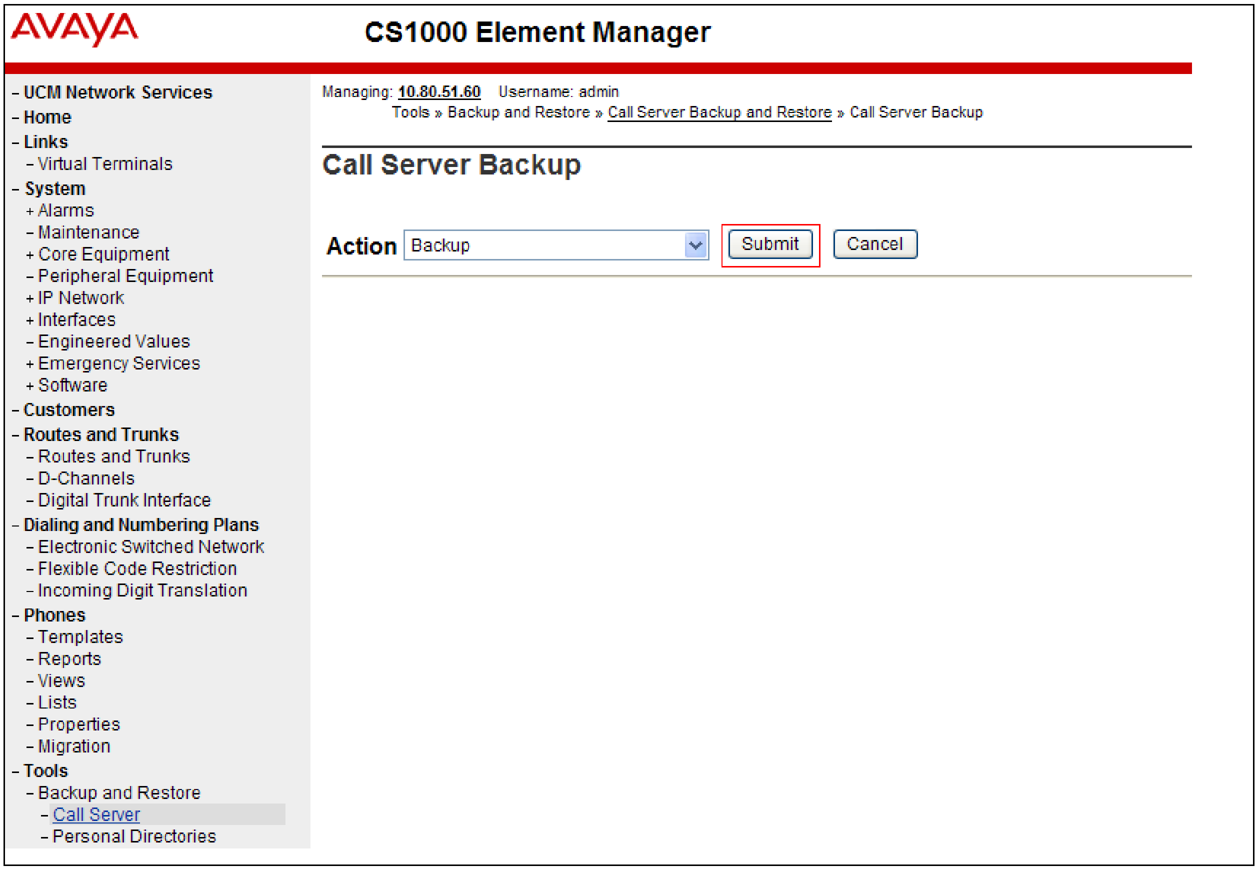
```
Overlay 20 – SIP Telephone Configuration
DES  SIPD
TN   096 0 01 15  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TLSV 0
SIPU 8889
NDID 5
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
ERL  0
ECL  0
VSIT NO
FDN
TGAR 0
LDN  NO
NCOS 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
---continued on next page---
---continued from previous page---
```

```
     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 *

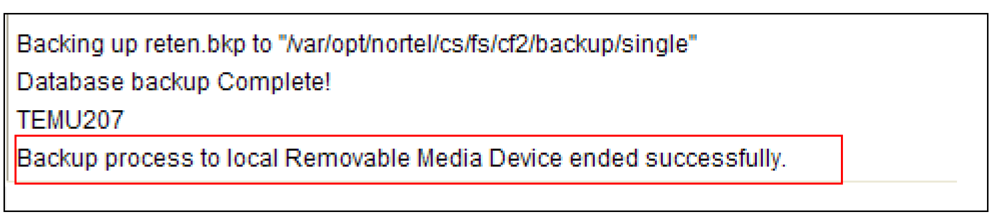
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 MCR 8889 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Sigma 1140
            XPLN 11
            DISPLAY_FMT FIRST,LAST*
     01 HOT U 788889 MARP 0
     02
     03
     04
     05
     06
     07
     08
     09
     10
     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** (not shown) 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 Communication Server 1000E is complete.

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager to receive and route calls over the SIP trunk between Communication Server 1000E and Session Manager. These instructions assume other administration activities have already been completed such as defining the SIP entity for Session Manager, defining the network connection between System Manager and Session Manager, and adding SIP endpoints. The following administration activities will be described.

- Define SIP Domain
- Define Location
- Configure Adaptation Module.
- Define SIP Entities
- Define Entity Links
- Define Routing Policies
- Define Dial Patterns

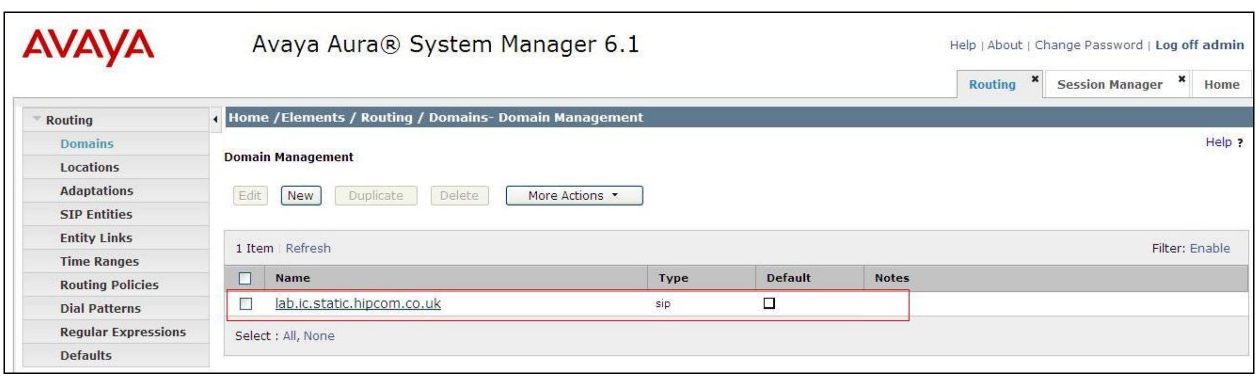
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Login with the appropriate credentials. Some administration screens have been abbreviated for clarity.

6.1. Define SIP Domains

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, **lab.ic.static.hipcom.co.uk** was used
- **Type** Verify **sip** is selected
- **Notes** Add a brief description [Optional]

Click **Commit** (not shown) to save. The screen below shows the SIP Domain defined for the sample configuration.



6.2. Define Location

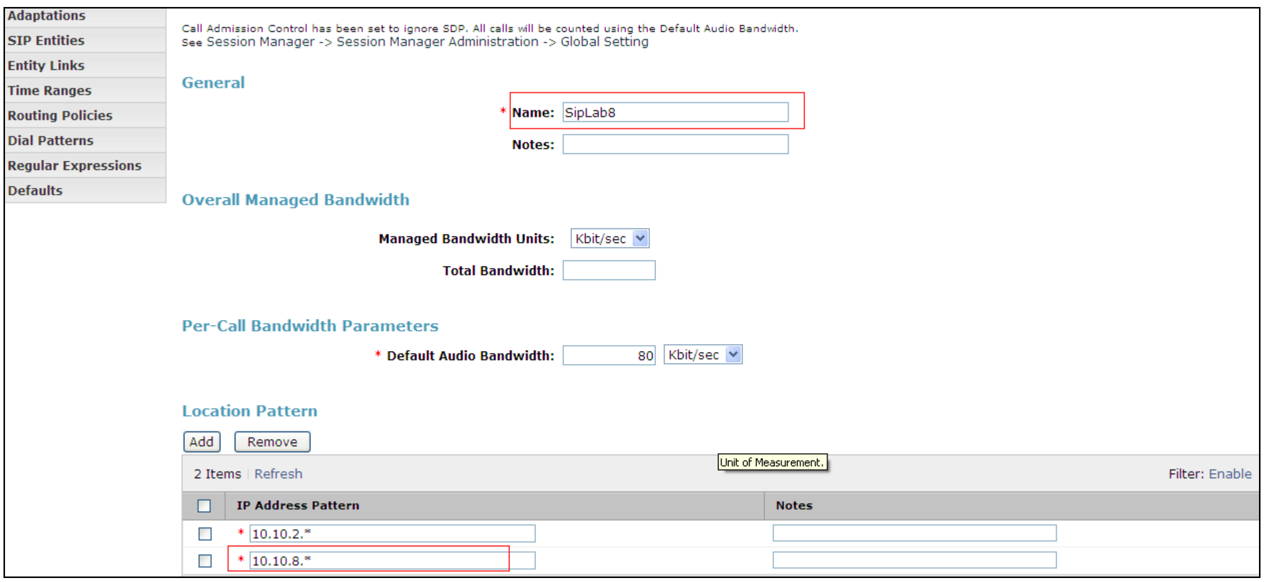
Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing. Expand **Elements → Routing** and select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section**,** enter the following values and use default values for remaining fields**.**

- **Name** Enter a descriptive name for the location
- **Notes** Add a brief description [Optional]

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, **10.10.8.*** was used
- **Notes** Add a brief description [Optional]

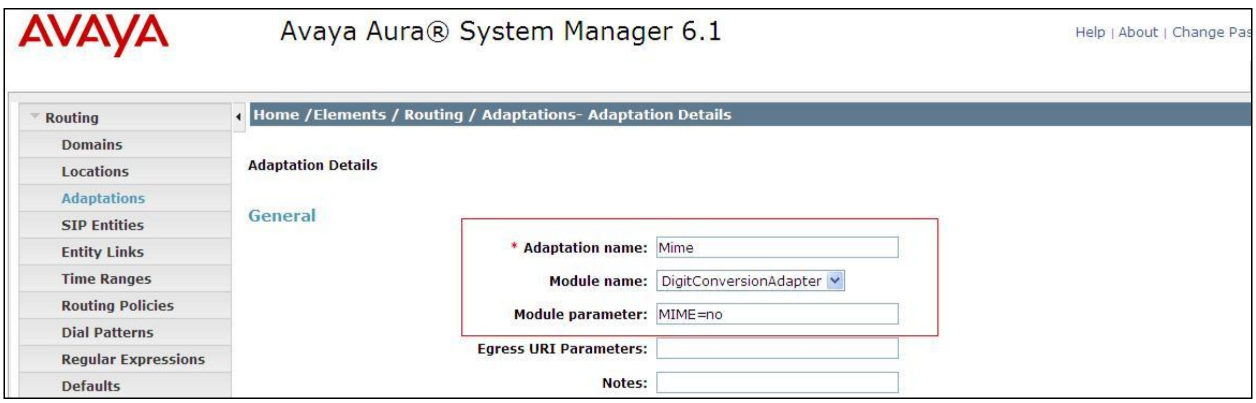
Click **Commit** to save. The screenshot below shows the Location defined for the enterprise network in the sample configuration.



6.3. Configure Adaptation Module

Session Manager is installed with a module called DigitConversionAdapter, which can convert digit strings in various message headers as well as host names in the Request-URI (Uniform Resource Identifier). In this configuration the adaptation is used by the Session Manager to strip MIME messages before being sent on to BTW/HIPCOM. To add an adaptation, select **Adaptations** on the left panel menu and then click on the **New** button (not shown). Under **General:**

- **Adaptation Name:** Enter an informative name, in the sample configuration **plus** was used
- **Module Name:** **<click to add module>** from the drop down list and enter "DigitConversionAdapter" in the resulting **New Module Name** field
- **Module Parameter:** Enter **MIME=no** to have Session Manager strip MIME message bodies on egress to BTW/HIPCOM, such that only SDP is present in the message body sent to BTW/HIPCOM's SBC



6.4. Define SIP Entities

A SIP Entity must be added for each SIP-based telephony system, supported by a SIP connection to the Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity. Under **General:**

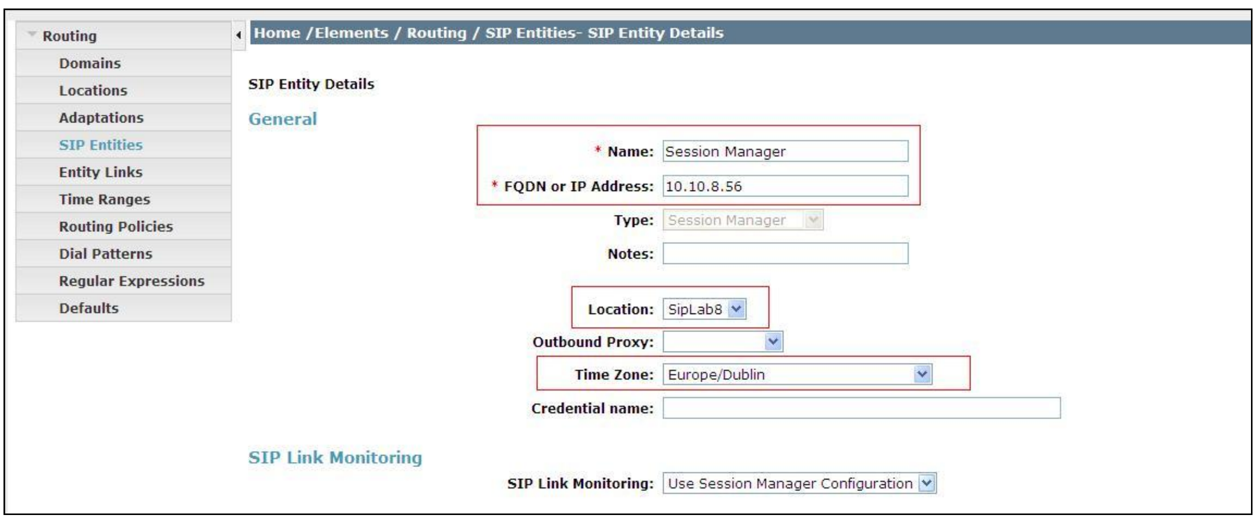
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **Other** for a Communication Server 1000E SIP entity and **Gateway** for the Avaya SBCE SIP entity
- In the **Location** field select the appropriate location from the drop down menu
- In the **Time Zone** field enter the time zone for the SIP Entity

In this configuration there are three SIP Entities:

- Avaya Aura® Session Manager SIP Entity
- Avaya Communication Server 1000E SIP Entity
- Avaya Session Border Controller for Enterprise SIP Entity

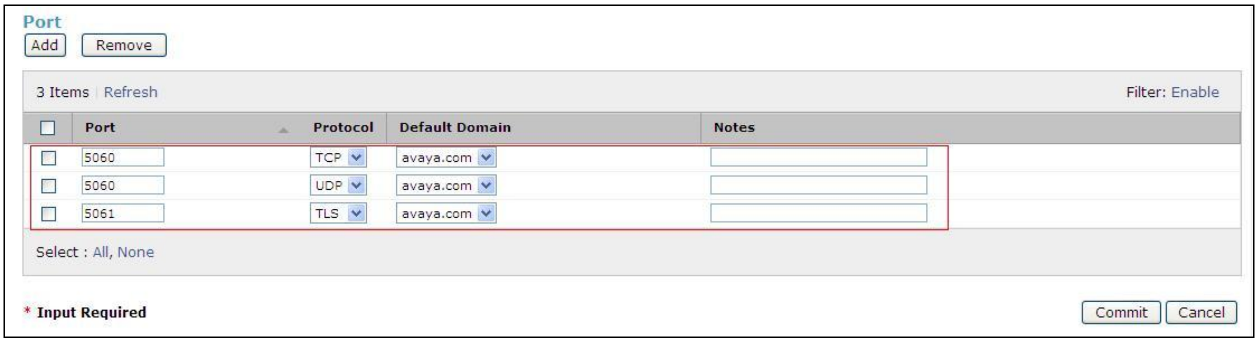
6.4.1. Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signalling interface.



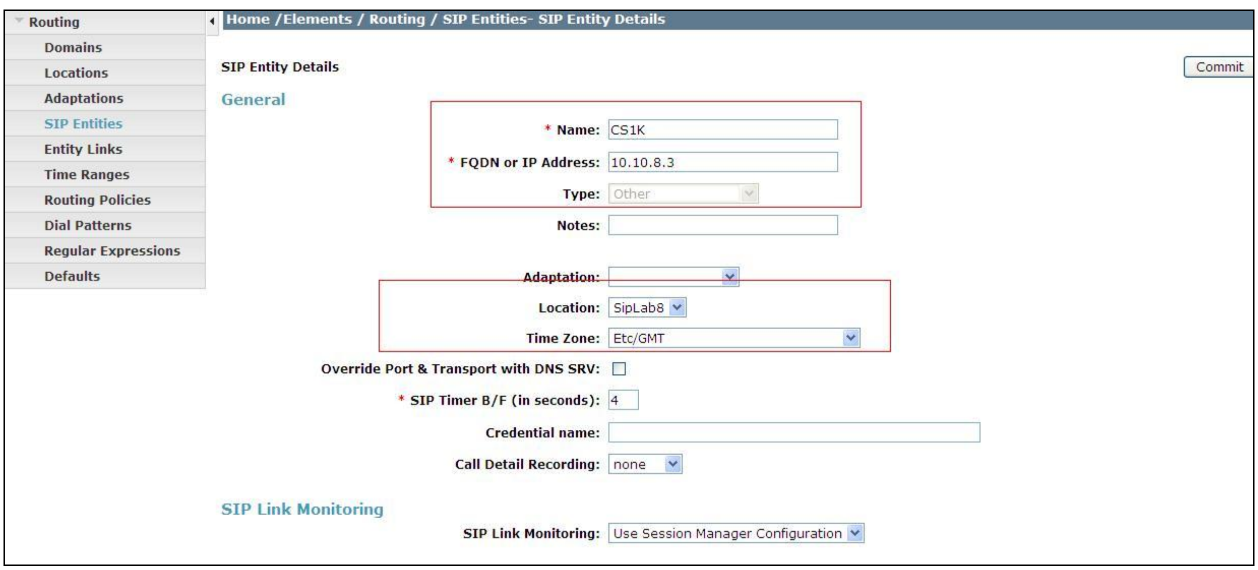
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add,** then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests
- In the **Protocol** field enter the transport protocol to be used for SIP requests
- In the **Default Domain** field, from the drop down menu select **avaya.com** as the default domain



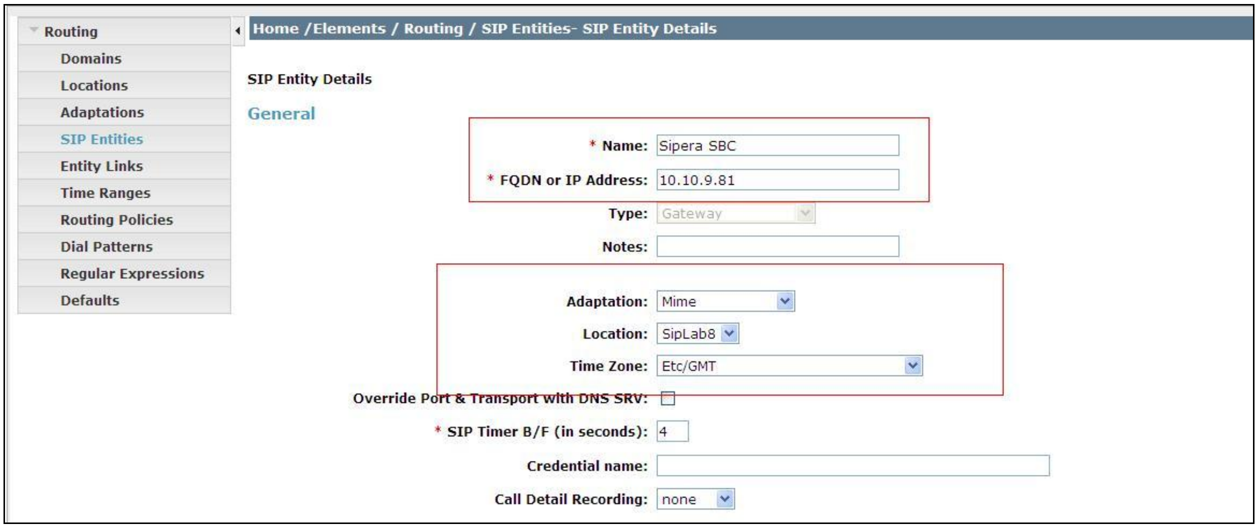
6.4.2. Avaya Communication Server 1000E SIP Entity

The following screen shows the SIP entity for Communication Server 1000E. The **FQDN or IP Address** field is set to the Node IP address of the interface on CS1000E that will be providing SIP signalling, as shown in **Section 5.4**.



6.4.3. Avaya Session Border Controller Advanced for Enterprise SIP Entity

The following screen shows the SIP Entity for the Session Border Controller. The **FQDN or IP Address** field is set to the IP address of the Session Border Controller private network interface (see **Figure 1**). Note the adaption module configured in **Section 6.3** is applied to this entity.

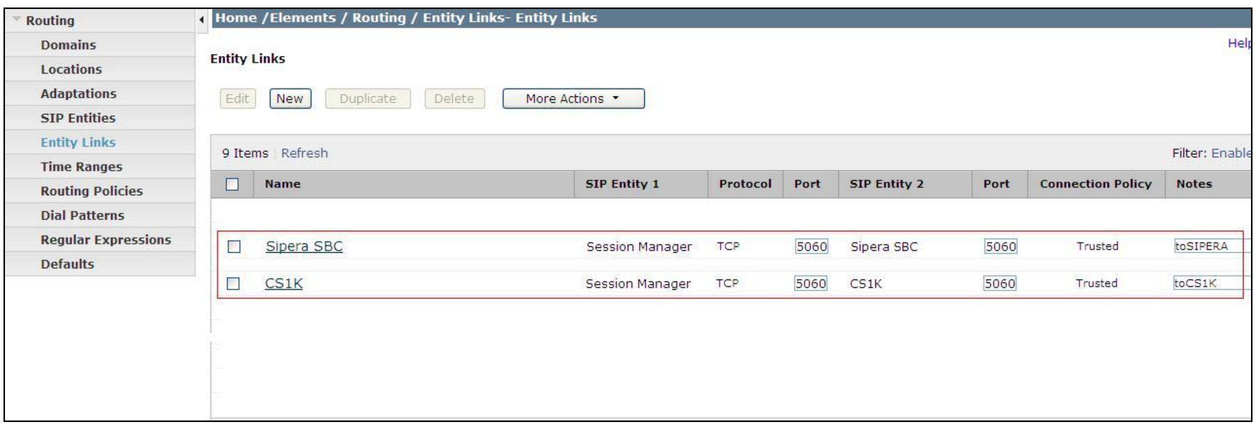


6.5. Define Entity links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button. Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name
- In the **SIP Entity 1** field select **Session Manager 1**
- In the **Port** field enter the port number to which the other system sends its SIP requests
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.4**
- In the **Port** field enter the port number to which the other system expects to receive SIP requests
- Select the **Trusted** tick box to make the other system trusted
- In the **Protocol** field enter the transport protocol to be used to send SIP requests

Next click **Commit** (not shown) to save changes. The following screen shows the Entity Links used in this configuration.



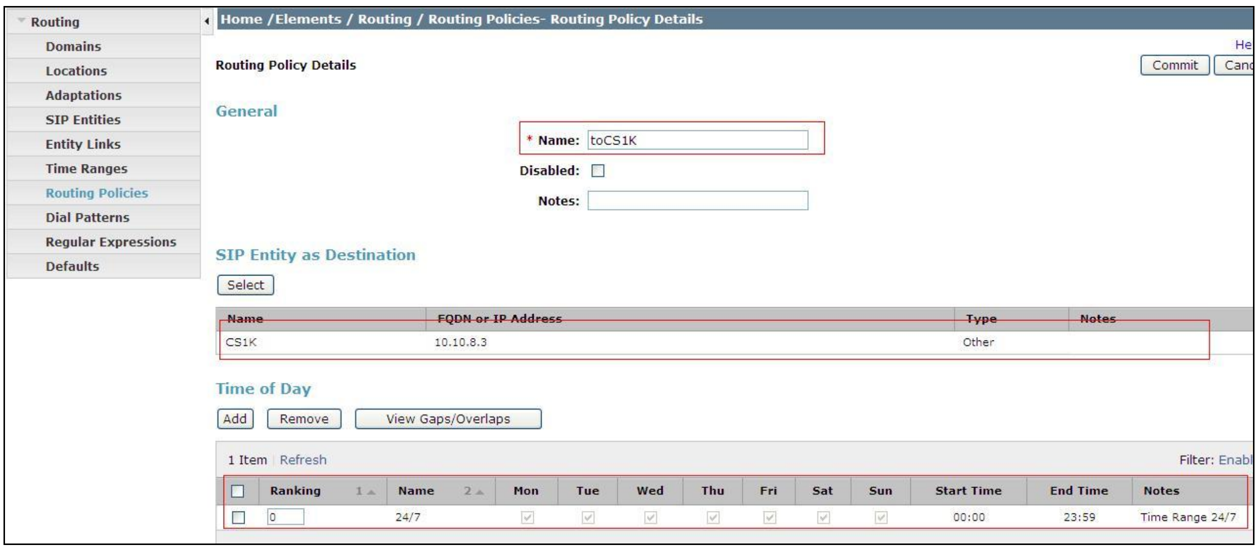
6.6. Define Routing Policy

Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

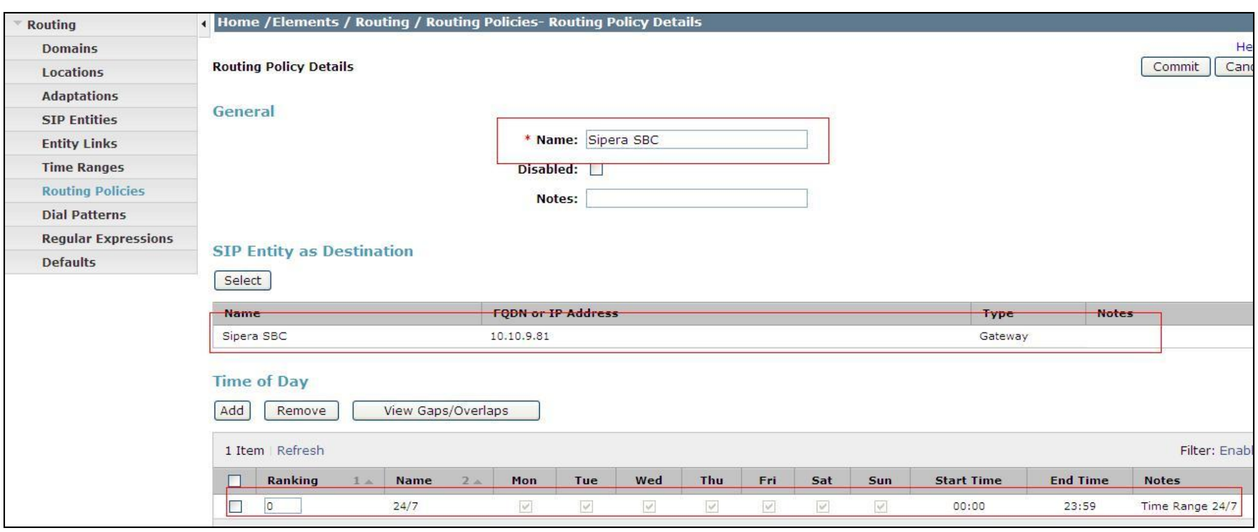
Under **General:**

- Enter an informative name in the **Name** field
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies
- Under **Time of Day**, click **Add**, and then select the time range

The following screen shows the routing policy for Communication Server 1000E



The following screen shows the routing policy for the Avaya SBCE.

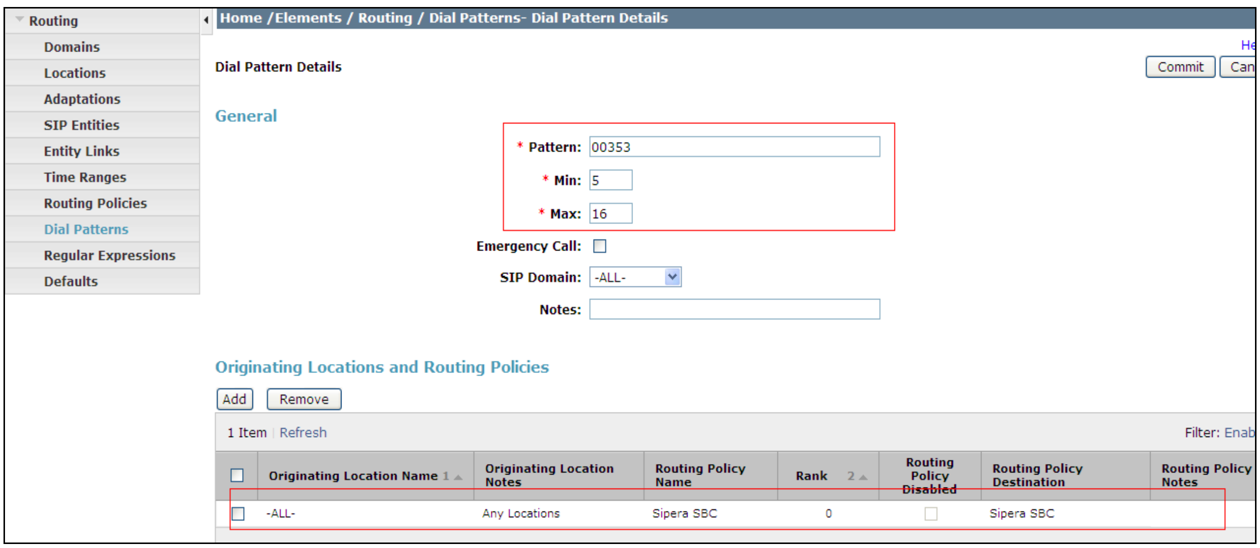


6.7. Define Dial Pattern

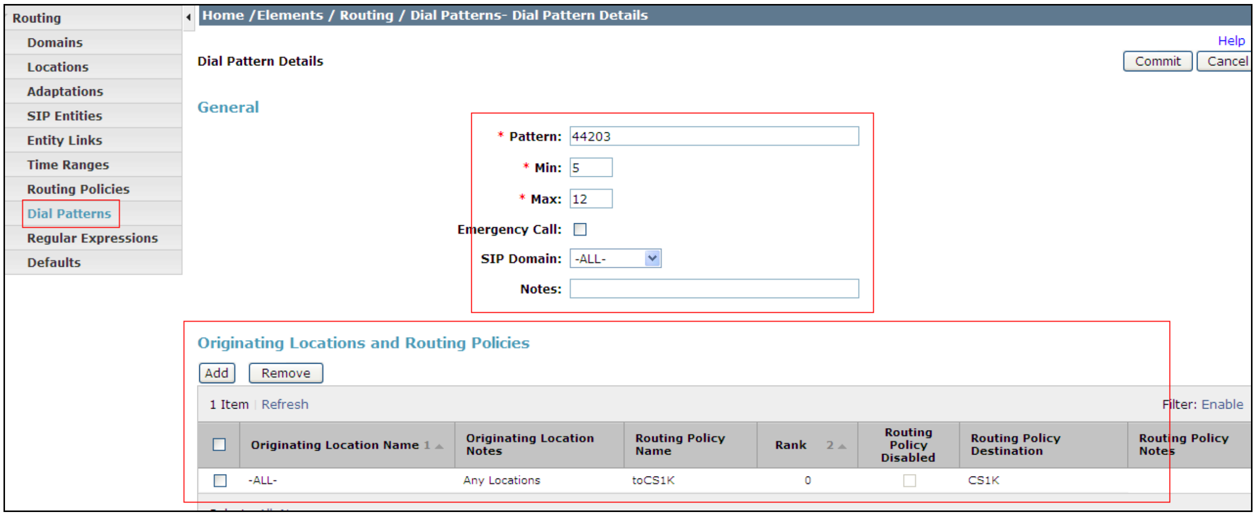
A dial pattern must be defined to direct calls to the appropriate entity. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).
Under **General:**

- In the **Pattern** field enter a dialed number or prefix to be matched
- In the **Min** field enter the minimum length of the dialled number
- In the **Max** field enter the maximum length of the dialled number
- In the **SIP Domain** field select **ALL** or alternatively one of those configured in **Section 6.2**

Under **Originating Locations and Routing Policies.** Click **Add**, in the resulting screen (not shown), under **Originating Location** select **ALL** and under **Routing Policies** select one of the routing policies defined in **Section 6.6**. Click **Select** button to save. The following screen shows an example dial pattern configured for the Avaya SBCE which will route the calls out to the BTW/HIPCOM's network.



The following screen shows an example dial pattern configured for the CS1000E. This dial pattern will route the calls to the CS1000E endpoints e.g. Unistim and SIP sets.

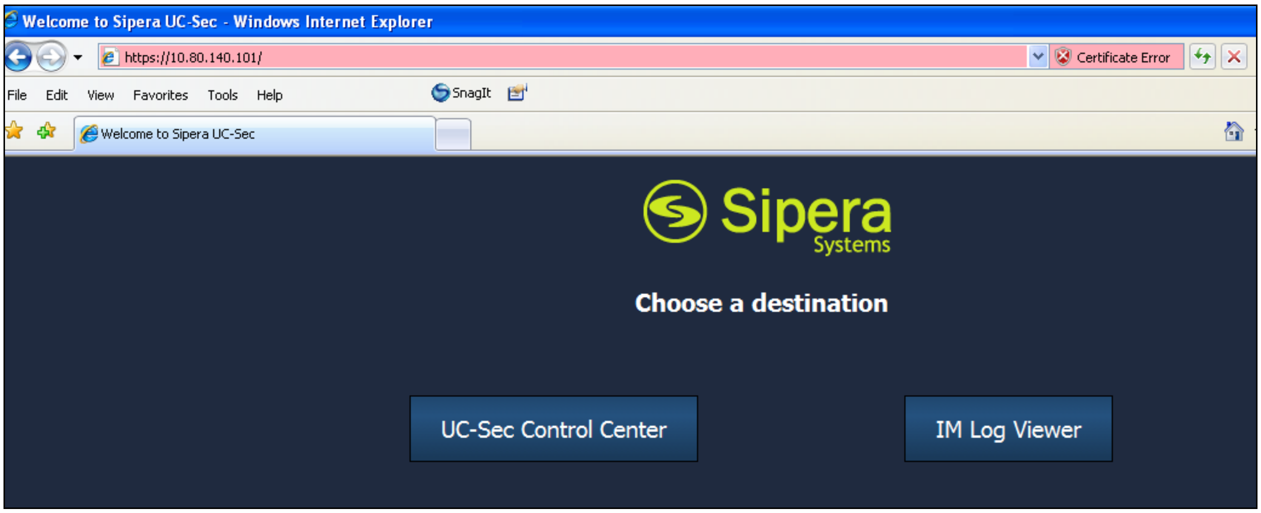


7. Configure Avaya Session Border Controller for Enterprise

This section provides the procedures for configuring Session Border Controller Advanced or Enterprise.

7.1. Accessing UC-Sec Control Centre

Access the web interface by typing **https://x.x.x.x** (where x.x.x.x is the management IP of the Avaya SBCE).



Select **UC-Sec Control Center** and enter the **Login ID** and **Password**.

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 allows you to configure and manage various SIP call server specific capabilities such as call hold and T.38. From the left hand menu select **Global Profiles → Server Interworking** and click on **Add Profile.**

- Enter profile name**: SM9_Call_Server** and click **Next**
- **Check Hold Support= RFC2543**
- Enable **T.38 support**
- All other options on the General Tab can be left at default

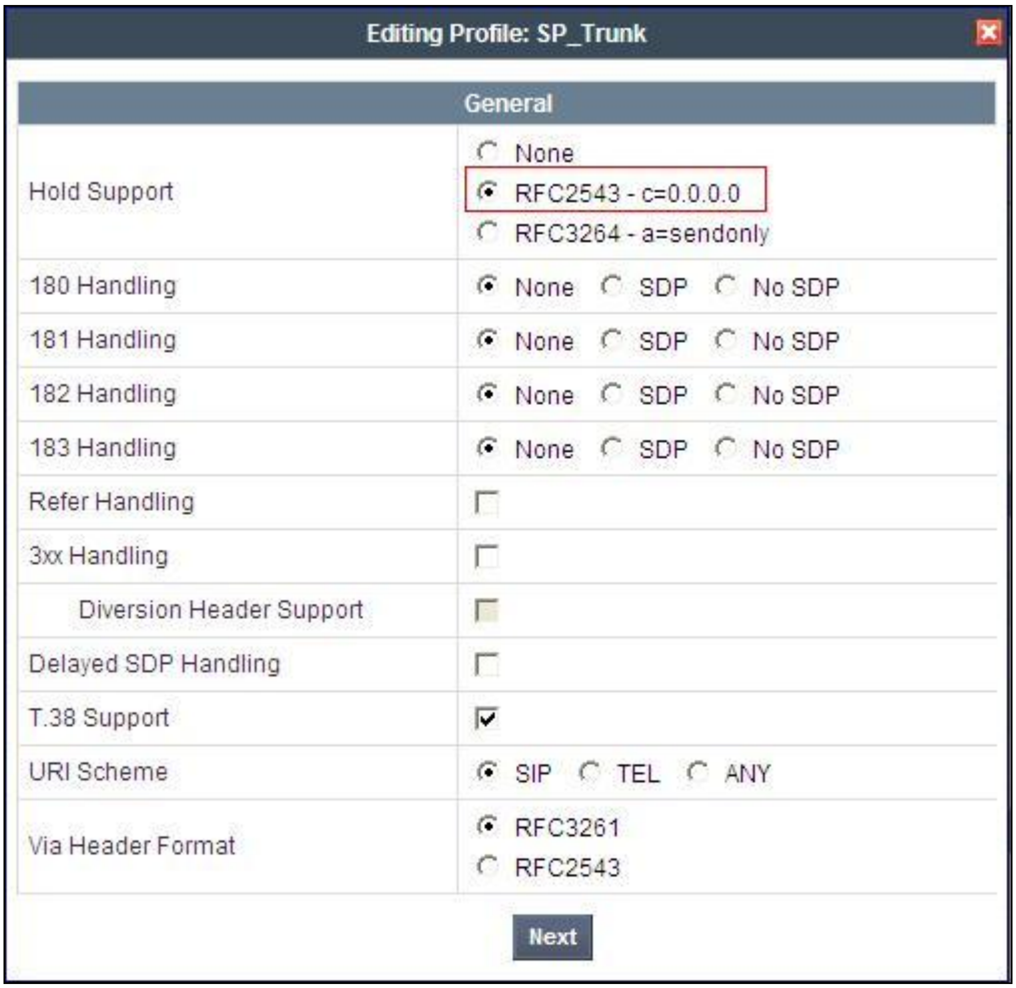
Click on **Next** on the following screens and then **Finish**.

7.2.2. Server Interworking – BTW/HIPCOM side

Server Internetworking allows you to configure and manage various SIP call server specific capabilities such as call hold and T.38. From the left hand menu select **Global Profiles → Server Interworking** and click on **Add Profile.**

- Enter profile name**: SP_Trunk** and click on **Next**
- **Check Hold Support= RFC2543**
- All other options on the General Tab can be left at default.

Click on **Next** on the following screens and then **Finish**.

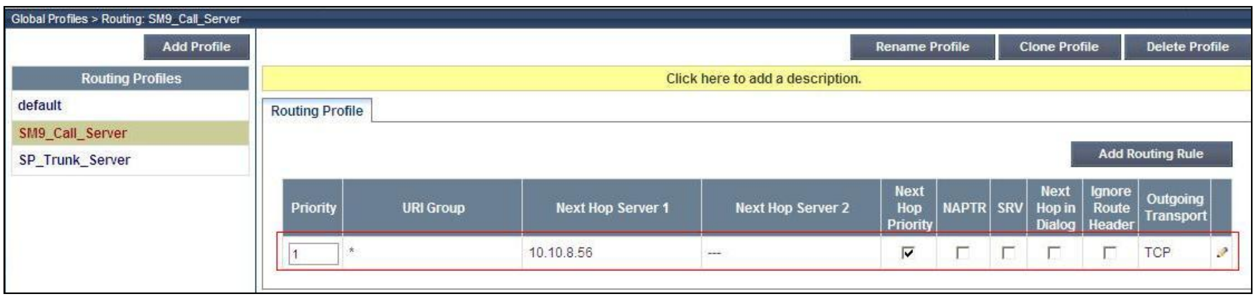


7.2.3. Routing – Avaya side

The Routing Profile allows you to manage parameters related to routing SIP signaling messages. From the left hand menu select **Global Profiles → Routing** and click on **Add Profile**.

- Enter Profile Name: **SM9_Call_Server**
- Hit **Next** (not shown)
- **Next Hop Server 1: 10.10.8.56** (Session Manager signalling interface ip address)
- Select **Routing Priority Based on Next Hop Server**
- **Outgoing Transport: TCP**

Click **Finish** (not shown).

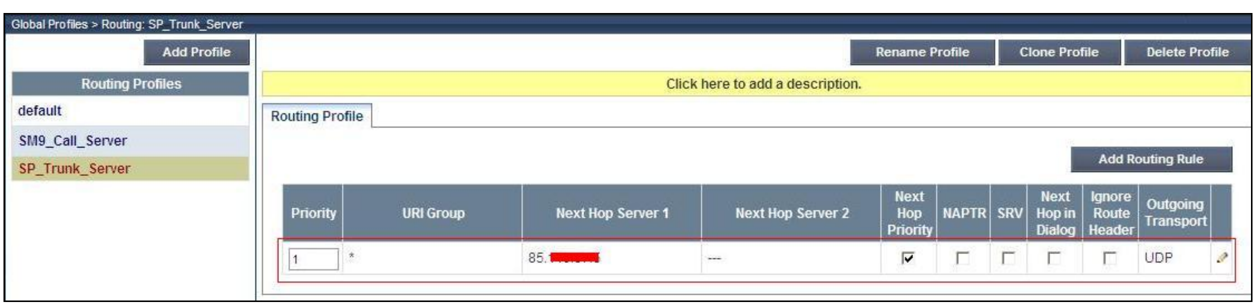


7.2.4. Routing – BTW/HIPCOM side

The Routing Profile allows you to manage parameters related to routing SIP signaling messages. From the left hand menu select **Global Profiles → Routing** and click on **Add Profile**.

- Enter Profile Name: **SP_Trunk_Server**
- Hit **Next**
- **Next Hop Server 1: 85.xx.xx.xx (IP Address provided by BTW/HIPCOM, partially hidden for security purposes)**
- Select **Routing Priority Based on Next Hop Server**
- **Outgoing Transport: UDP**

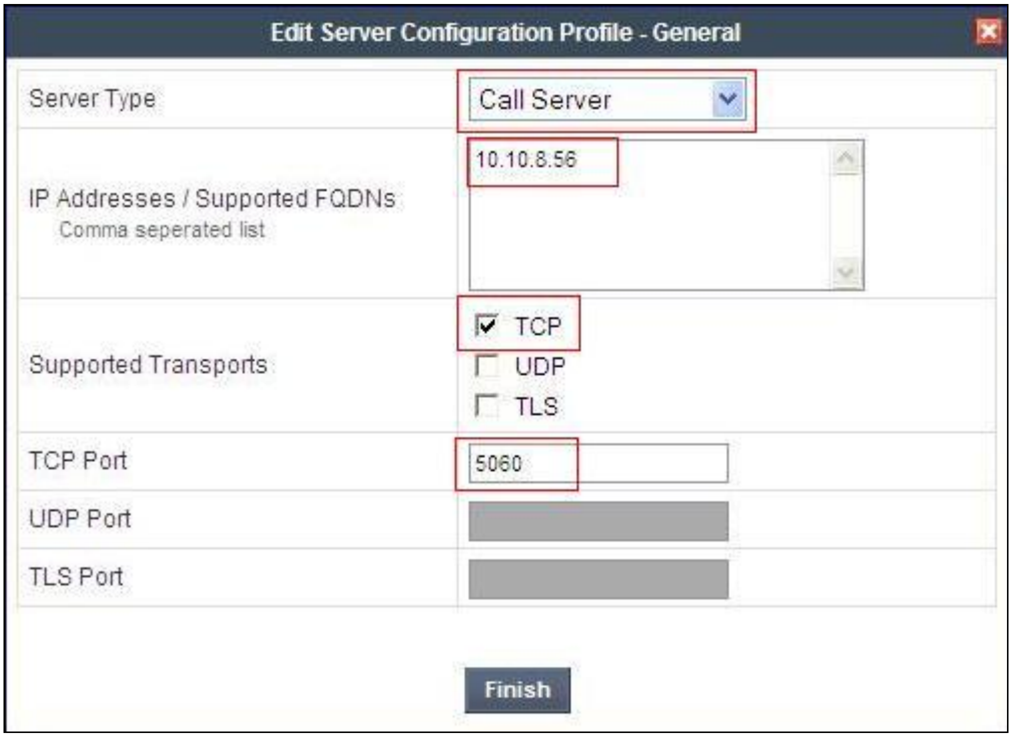
Click **Finish** (not shown)

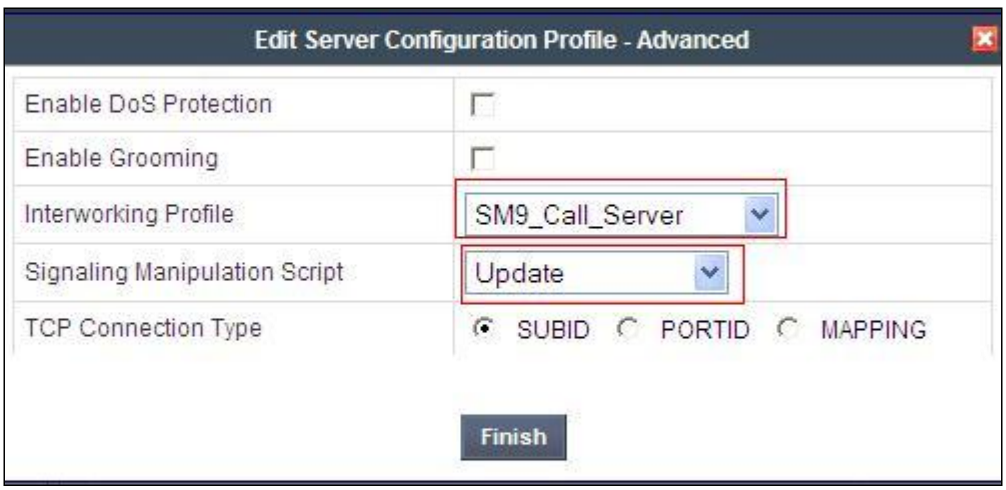


7.2.5. Server Configuration – Avaya side

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to 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.

- Select **Global Profiles → Server Configuration**
- Click on **Add Profile**.
- **Enter profile name: SM9_Call_Server**
- On the **Add Server Configuration Profile** tab
    - Select Server Type**: Call Server**
    - **IP Address: 10.10.8.56 (Session Manager signalling interface ip address, configured in Section 5.4)**
    - **Supported Transports**: Check **TCP**
    - **TCP Port: 5060**
    - Click **Next**
- At the **Authentication** tab
    - Click **Next**
- At the **Heartbeat** tab
    - Click **Next**.
- On the **Advanced** Tab
    - Select **SM9_Call_Sever** for Interworking Profile
    - Select **Update** for Signaling Manipulation Script, this will be discussed in **Section 7.3**
    - Hit **Next**
- Click **Finish**


Edit Server Configuration Profile - General
Server Type
Call Server
IP Addresses / Supported FQDNs
Comma seperated list
10.10.8.56
Supported Transports
TCP
UDP
TLS
TCP Port
5060
UDP Port
TLS Port
Finish


Edit Server Configuration Profile - Advanced
Enable DoS Protection
Enable Grooming
Interworking Profile
SM9_Call_Server
Signaling Manipulation Script
Update
TCP Connection Type
SUBID
PORTID
MAPPING
Finish

7.2.6. Server Configuration – BTW/HIPCOM side

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

- Select **Global Profiles → Server Configuration**
- Select **Add Profile.**
- **Enter profile name: SP_Trunk_Server**
- On the **Add Server Configuration Profile** tab
    - Select Server Type **Trunk Server**
    - **IP Address: 85.xx.xx.xx (BTW/HIPCOM's SBC ip address, partially hidden for security purposes)**
    - **Supported Transports**: Check **UDP**
    - **UDP Port: 5060**
    - Click **Next**
- At the **Authentication** tab
    - Select **Enable Authentication**
    - Enter the **User Name, Realm** and **Password** provided by HIPCOM
- At the **Heartbeat** tab
    - Click **Next**.
- On the **Advanced** Tab
    - Select **SP_Trunk** for Interworking Profile
    - Hit **Next**
- Click **Finish.**


Edit Server Configuration Profile - General
Server Type
Trunk Server
IP Addresses / Supported FQDNs
Comma seperated list
[redacted]
Supported Transports
TCP
UDP
TLS
TCP Port
UDP Port
5060
TLS Port
Finish


Edit Server Configuration Profile - Authentication
Enable Authentication
User Name
[redacted]
Realm
[redacted]
Password
(Leave blank to keep existing password)
Confirm Password
Finish


Edit Server Configuration Profile - Advanced
Enable DoS Protection
Enable Grooming
Interworking Profile
SP_Trunk
Signaling Manipulation Script
None
UDP Connection Type
SUBID
PORTID
MAPPING
Finish

7.2.7. 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.**

- Click **default** profile and select **Clone Profile**
- Enter Profile Name**: SM9_CS**
- For the **Header To** and **Request Line** select **IP/Domain** under **Criteria** and **Next Hop** under **Replace Action.**
- Click **Finish**

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

7.2.8. Topology Hiding – BTW/HIPCOM side

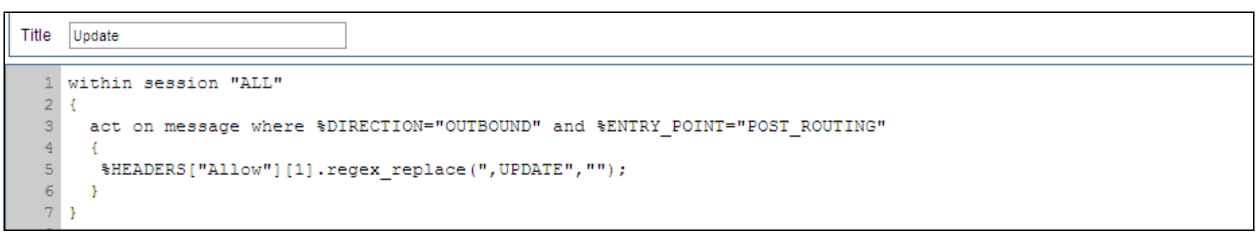
The **Topology Hiding** screen allows you to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks. From the left-hand menu select **Global Profiles → Topology Hiding.**

- Click **default** profile and select **Clone Profile**
- **Enter Profile Name: SP_Trunk**
- For the Headers **To**, **From** and **Request Line** select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For **Override Value** type **lab.ic.static.hipcom.co.uk**
- Click **Finish**

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

7.3. Signalling Manipulation Scripts

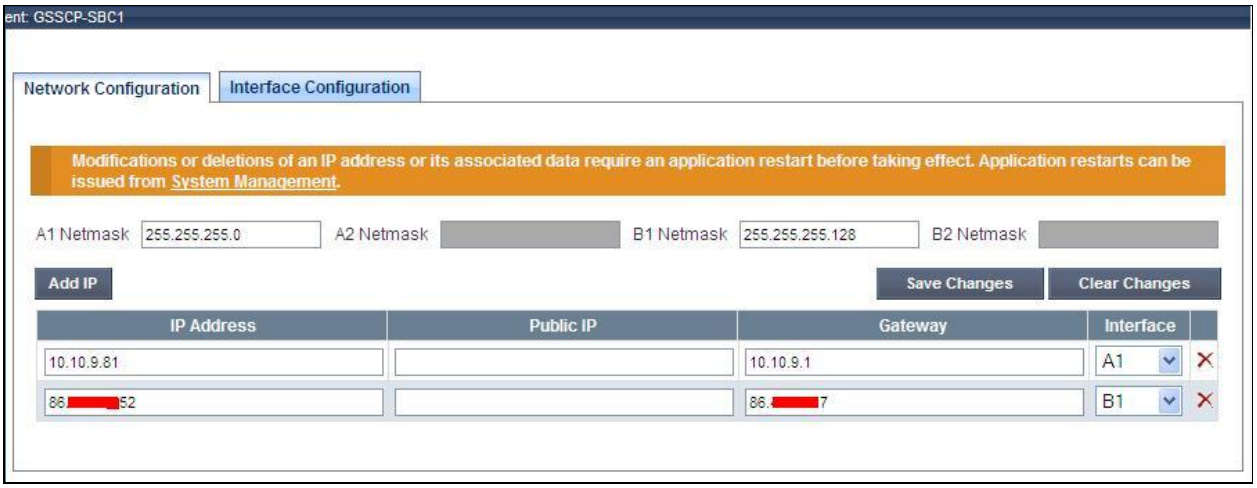
This feature adds the ability to add, change and delete any of the headers and other information in a SIP message. During test, a script was created so that the UPDATE header is removed so that Blind Transfer back out to the PSTN would work using the re-INVITE method on the CS1K. From the left hand menu select **Global Profiles → Signaling Manipulation** and click on **Add Script**, in this case the title of the script is **Update**. The script shown below was used for this test:



7.4. Device Specific Settings

The **Network Management** feature allows the public and private interface addresses and state to be set. From the left-hand menu select **Device Specific Settings → Network Management.**

- Enter in the **IP Address** and **Gateway Address** for both the Inside and the Outside interfaces
- Select the physical interface used in the **Interface** column



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

The **Media Interfaces** feature allows the IP Address and ports to be set for transporting Media over the SIP trunk. From the left hand menu select **Device Specific Settings → Media Interface.**

- Select **Add Media Interface**
- **Name**: **Int_Media**
- **Media IP**: **10.10.9.81** (Internal Address for calls toward CS1000E)
- **Port Range**: **35000-40000**
- Click **Finish**
- Select **Add Media Interface**

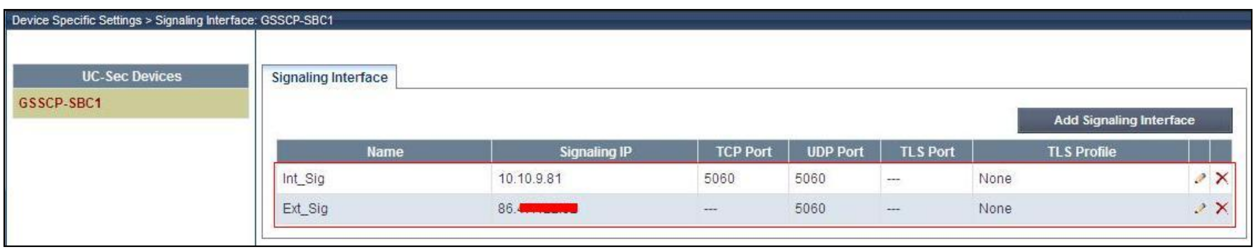
- **Name**: **Ext_Media**
- **Media IP**: **86.xx.xx.xx** (External Address for calls toward BTW/HIPCOM's SIP trunk, hidden for security purposes)
- **Port Range**: **35000-40000**
- Click **Finish**

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

The **Signalling Interfaces** feature allows the IP Address and ports to be set for transporting Signalling over the SIP trunk. From the left hand menu select **Device Specific Settings → Signalling Interface.**

- Select **Add Signaling Interface**
- **Name**: **Int_Sig**
- **Signaling IP**: **10.10.9.81** (Internal Address for calls toward CS1000E)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**
- Select **Add Signaling Interface**
- **Name**: **Ext_Sig**
- **Signaling IP: 86.xx.xx.xx** (External Address for calls toward BTW/HIPCOM's SIP trunk, hidden for security purposes)
- **UDP Port**: **5060**
- Click **Finish**

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



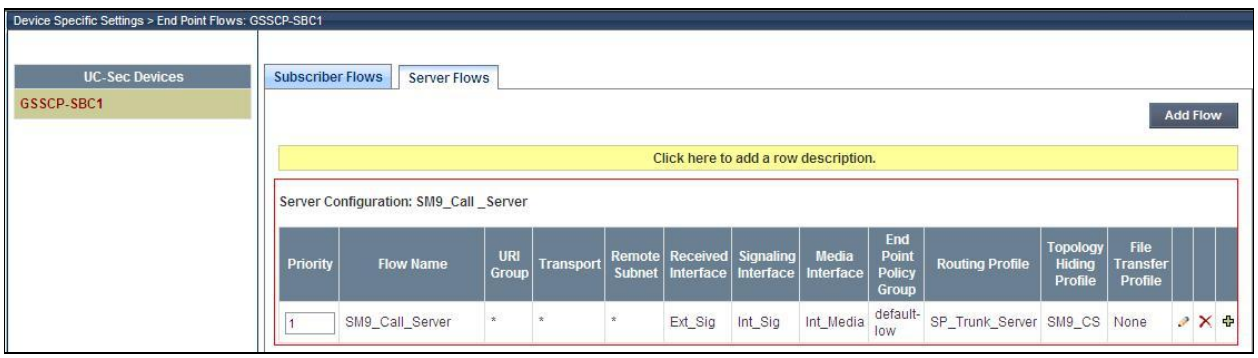
Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile
Int_Sig	10.10.9.81	5060	5060	---	None
Ext_Sig	86.[redacted]	---	5060	---	None

The **End Point Flows** allow the Interfaces**,** Policies and Profiles administered to be used to transport the SIP traffic. From the left-hand menu select **Device Specific Settings → Endpoint Flows** (not shown).

- Select the **Server Flows** tab

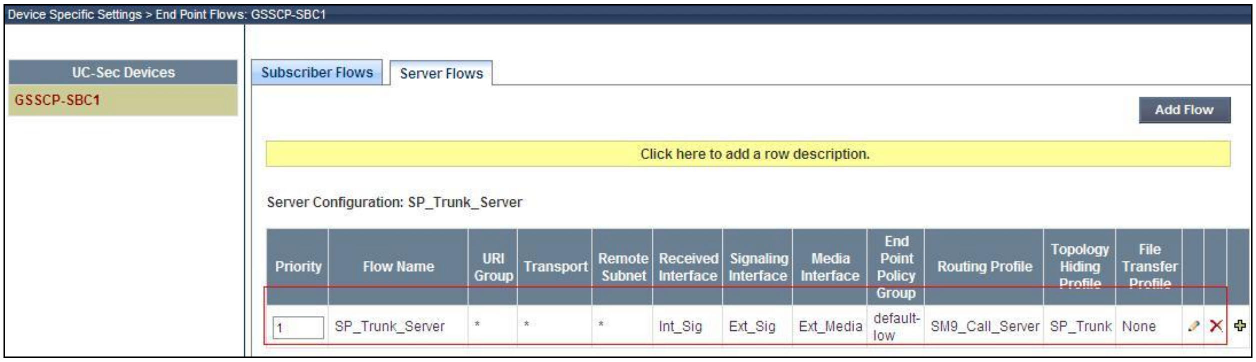
To add the settings for the call flow to the CS1000E, click on select **Add Flow.**

- **Name**: **SM9_Call_Server**
- **Server Configuration**: **SM9_Call_Server**
- **URI Group: ***
- **Transport**: *****
- **Remote Subnet**: *****
- **Received Interface**: **Ext_Sig**
- **Signaling Interface**: **Int_Sig**
- **Media Interface**: **Int_Media**
- **End Point Policy Group**: **default-low**
- **Routing Profile**: **SP_Trunk_Server**
- **Topology Hiding Profile**: **SM9_CS**
- **File Transfer Profile**: **None**
- Click **Finish** (not shown)



To add the settings for the call flow to BTW/HIPCOM, click on select **Add Flow.**

- **Name**: **SP_Trunk_Server**
- **Server Configuration**: **SP_Trunk_Server**
- **URI Group**: *****
- **Transport**: *****
- **Remote Subnet**: *****
- **Received Interface**: **Int_Sig**
- **Signaling Interface**: **Ext_Sig**
- **Media Interface**: **Ext_Media**
- **End Point Policy Group**: **default-low**
- **Routing Profile**: **SM9_Call_Server**
- **Topology Hiding Profile**: **SP_Trunk**
- **File Transfer Profile**: **None**
- Click **Finish**



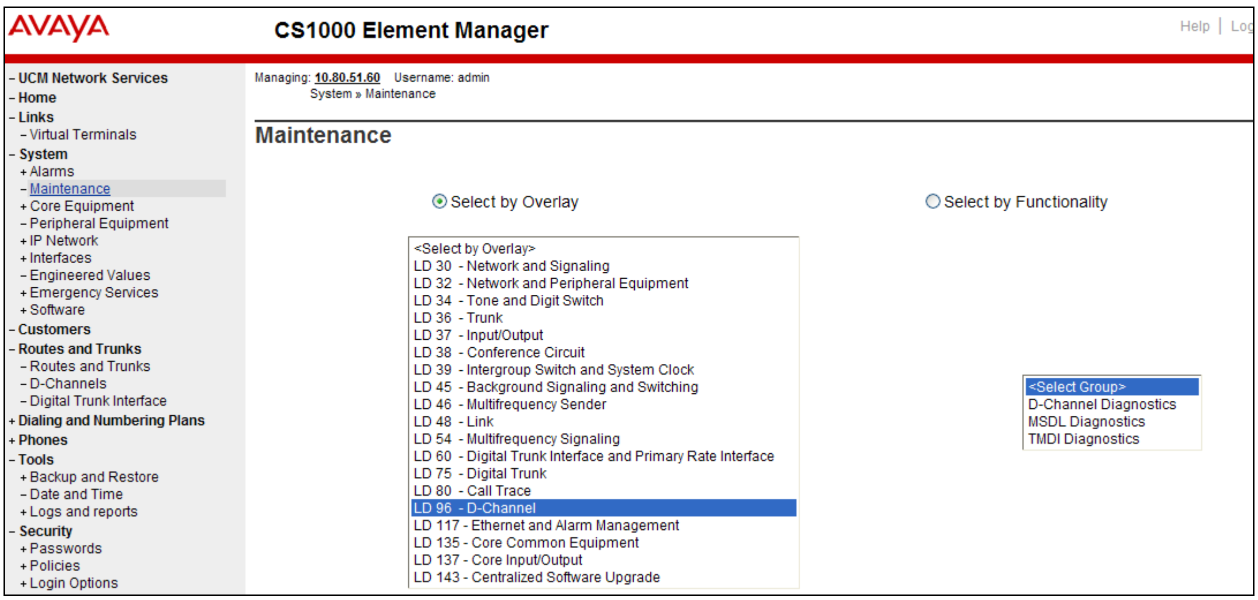
8. BT Wholesale/HIPCOM Service Provider Configuration

The configuration of BTW/HIPCOM's equipment used 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 BTW/HIPCOM's equipment and system configuration please contact an authorised BTW/HIPCOM representative.

9. Verification

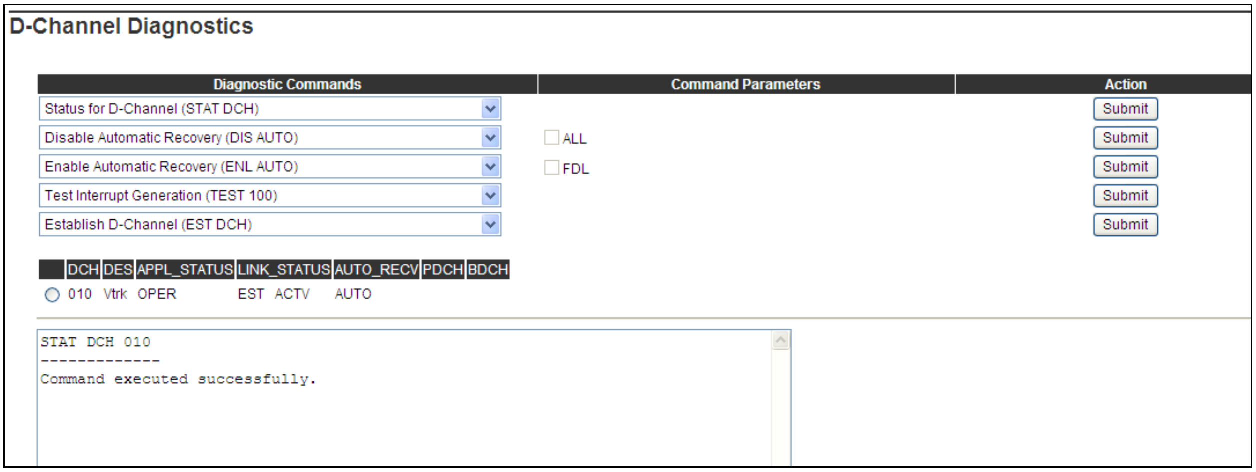
9.1. Verify Avaya Communication Server 1000E Operational Status

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**

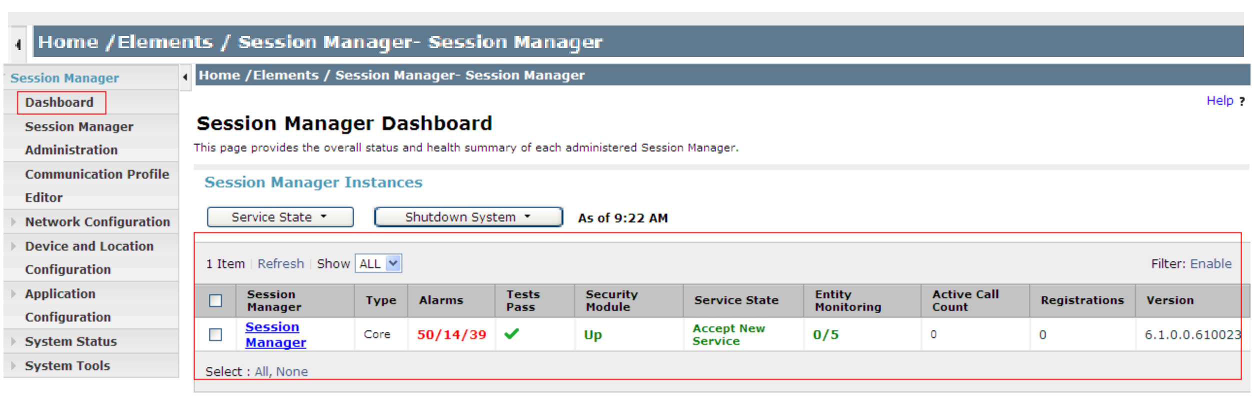


9.2. Verify Avaya Aura® Session Manager Operational Status

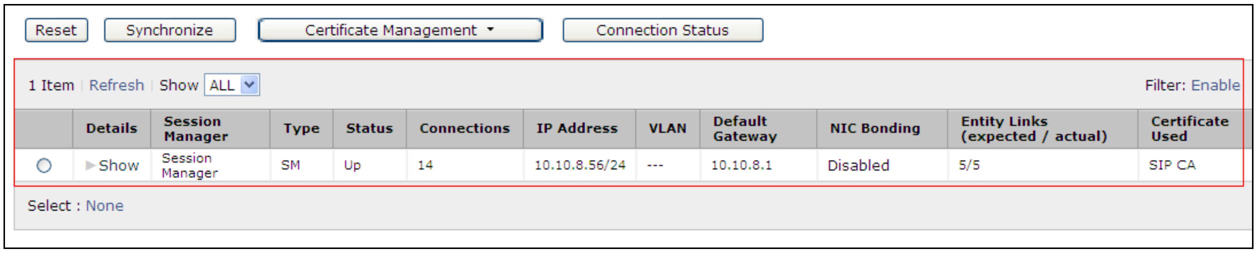
9.2.1. Verify Avaya Aura® Session Manager is Operational

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.

- **Tests Pass**
- **Security Module**
- **Service State**

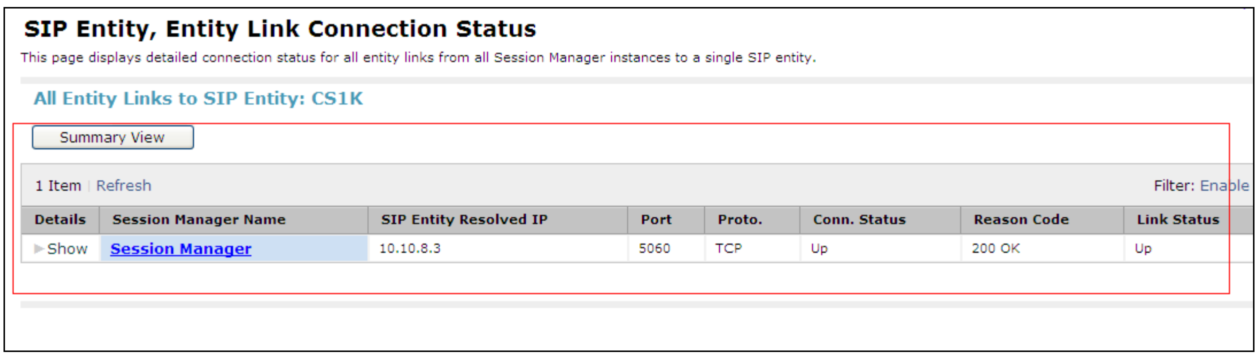


Navigate to **Elements → Session Manager → System Status → Security Module Status** (not shown) to view more detailed status information on the status of Security Module for the specific Session Manager. Verify the **Status** column displays **Up** as shown below.



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 Communication Server 1000Efrom 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: CS1000 Rel7.5** table, verify the **Conn. Status** for the link is **Up** as shown below.



Verify 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 Entities** table (not shown).

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Communication Server 1000E, Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise to BTW/HIPCOM SIP Trunk Service. The service was successfully tested with a number of observations listed 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] Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-116, 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 Advanced for Enterprise) Administration Guide, November 2011

[11] RFC 3261 SIP: Session Initiation Protocol, http://www.ietf.org/

Appendix A
Avaya Communication Server 1000E Software

Communication Server 1000E call server patches and plug ins

```
01/05/12 14:07:37
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(created: 2012-03-14 13:55:18 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2012-03-27 14:39:08(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 : 3
PAT#  CR #            PATCH REF #     NAME            DATE        FILENAME
00    wi00890367      ISS1:1OF1       MGCCCD02        24/04/2012  MGCCCD02.LW
01    wi00832543      ISS1:1OF1       DSP1AB04        24/04/2012  DSP1AB04.LW
02    wi00946113      ISS1:1OF1       MGCBBA15        24/04/2012  MGCBBA15.LW

ENABLED PLUGINS : 1

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
```

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))

IN-SERVICE PEPS
PAT# CR #            PATCH REF #     NAME      DATE        FILENAME       SPECINS
000  wi00832106      ISS1:1OF1       p30550_1  30/04/2012  p30550_1.cpl   NO
001  wi00835294      ISS1:1OF1       p30565_1  30/04/2012  p30565_1.cpl   NO
002  wi00897176      ISS1:1OF1       p30418_1  30/04/2012  p30418_1.cpl   NO
003  wi00925218      ISS1:1OF1       p30675_1  30/04/2012  p30675_1.cpl   NO
004  wi00839821      ISS1:1OF1       p30619_1  30/04/2012  p30619_1.cpl   NO
005  wi00937672      ISS1:1OF1       p31276_1  30/04/2012  p31276_1.cpl   NO
006  wi00842409      ISS1:1OF1       p30621_1  30/04/2012  p30621_1.cpl   NO
007  wi00838073      ISS1:1OF1       p30588_1  30/04/2012  p30588_1.cpl   NO
008  wi00937114      ISS1:1OF1       p31310_1  30/04/2012  p31310_1.cpl   NO
009  wi00841980      ISS1:1OF1       p30618_1  30/04/2012  p30618_1.cpl   NO
010  wi00955753      ISS1:1OF1       p31733_1  30/04/2012  p31733_1.cpl   NO
011  wi00839255      ISS1:1OF1       p30591_1  30/04/2012  p30591_1.cpl   NO
012  wi00843623      ISS1:1OF1       p30731_1  30/04/2012  p30731_1.cpl   YES
013  WI00843571      ISS1:1OF1       p30627_1  30/04/2012  p30627_1.cpl   NO
```

```
014  wi00871739      ISS1:1OF1       p30856_1  30/04/2012  p30856_1.cpl   NO
015  wi00852365      ISS1:1OF1       p30707_1  30/04/2012  p30707_1.cpl   NO
016  wi00852389      ISS1:1OF1       p30641_1  30/04/2012  p30641_1.cpl   NO
017  wi00839134      ISS1:1OF1       p30698_1  30/04/2012  p30698_1.cpl   YES
018  wi00856702      ISS1:1OF1       p30573_1  30/04/2012  p30573_1.cpl   NO
019  wi00857566      ISS1:1OF1       p30766_1  30/04/2012  p30766_1.cpl   NO
020  wi00850521      ISS1:1OF1       p30709_1  30/04/2012  p30709_1.cpl   YES
021  wi00903381      ISS1:1OF1       p30421_1  30/04/2012  p30421_1.cpl   NO
022  wi00863876      ISS1:1OF1       p30787_1  30/04/2012  p30787_1.cpl   NO
023  WI00853473      ISS1:1OF1       p30625_1  30/04/2012  p30625_1.cpl   NO
024  wi00854130      ISS1:1OF1       p30443_1  30/04/2012  p30443_1.cpl   NO
025  wi00875425      ISS1:1OF1       p30943_1  30/04/2012  p30943_1.cpl   NO
026  wi00978883      ISS1:1OF1       p31770_1  30/04/2012  p31770_1.cpl   NO
027  wi00875701      ISS1:1OF1       p30942_1  30/04/2012  p30942_1.cpl   NO
028  wi00936935      ISS1:1OF1       p31362_1  30/04/2012  p31362_1.cpl   NO
029  wi00877367      ISS1:1OF1       p30534_1  30/04/2012  p30534_1.cpl   NO
030  wi00871969      ISS1:1OF1       p30768_1  30/04/2012  p30768_1.cpl   NO
031  wi00886321      ISS1:1OF1       p31009_1  30/04/2012  p31009_1.cpl   NO
032  WI00836334      ISS1:1OF1       p30481_1  30/04/2012  p30481_1.cpl   NO
033  wi00836182      ISS1:1OF1       p30450_1  30/04/2012  p30450_1.cpl   NO
034  wi00858335      ISS1:1OF1       p30819_1  30/04/2012  p30819_1.cpl   NO
035  wi00860279      ISS1:1OF1       p30789_1  30/04/2012  p30789_1.cpl   NO
036  wi00953900      ISS1:1OF1       p31494_1  30/04/2012  p31494_1.cpl   NO
037  wi00854415      ISS1:1OF1       p30593_1  30/04/2012  p30593_1.cpl   NO
038  WI00836292      ISS1:1OF1       p30554_1  30/04/2012  p30554_1.cpl   NO
039  WI00839794      ISS1:1OF1       p28647_1  30/04/2012  p28647_1.cpl   NO
040  wi00824257      ISS1:1OF1       p30447_1  30/04/2012  p30447_1.cpl   NO
041  wi00827950      ISS2:1OF1       p30471_2  30/04/2012  p30471_2.cpl   NO
042  wi00949273      ISS1:1OF1       p31411_1  30/04/2012  p31411_1.cpl   NO
043  WI00854150      ISS1:1OF1       p30468_1  30/04/2012  p30468_1.cpl   NO
044  wi00873382      ISS1:1OF1       p30832_1  30/04/2012  p30832_1.cpl   NO
045  wi00853178      ISS1:1OF1       p30719_1  30/04/2012  p30719_1.cpl   NO
046  wi00869695      ISS1:1OF1       p30654_1  30/04/2012  p30654_1.cpl   NO
047  wi00834382      ISS1:1OF1       p30548_1  30/04/2012  p30548_1.cpl   NO
048  wi00951427      ISS1:1OF1       p31478_1  30/04/2012  p31478_1.cpl   NO
049  wi00946558      ISS1:1OF1       p31358_1  30/04/2012  p31358_1.cpl   NO
050  wi00903369      ISS1:1OF1       p31165_1  30/04/2012  p31165_1.cpl   NO
051  wi00927321      ISS1:1OF1       p31286_1  30/04/2012  p31286_1.cpl   YES
052  wi00923899      ISS1:1OF1       p31270_1  30/04/2012  p31270_1.cpl   NO
053  wi00949627      ISS1:1OF1       p31462_1  30/04/2012  p31462_1.cpl   NO
054  wi00962557      ISS1:1OF1       p31581_1  30/04/2012  p31581_1.cpl   NO
055  wi00865477      ISS1:1OF1       p30894_1  30/04/2012  p30894_1.cpl   YES
056  wi00962211      ISS1:1OF1       p31580_1  30/04/2012  p31580_1.cpl   NO
057  wi00883604      ISS1:1OF1       p30973_1  30/04/2012  p30973_1.cpl   NO
058  wi00898327      ISS1:1OF1       p31136_1  30/04/2012  p31136_1.cpl   NO
059  wi00856410      ISS1:1OF1       p30749_1  30/04/2012  p30749_1.cpl   NO
060  wi00932948      ISS1:1OF1       p31077_1  30/04/2012  p31077_1.cpl   NO
061  wi00905600      ISS1:1OF1       p31201_1  30/04/2012  p31201_1.cpl   NO
062  wi00865477      ISS1:1OF1       p30897_1  30/04/2012  p30897_1.cpl   YES
063  wi00879526      ISS1:1OF1       p31007_1  30/04/2012  p31007_1.cpl   NO
064  wi00962955      ISS1:1OF1       p31585_1  30/04/2012  p31585_1.cpl   NO
065  wi00984178      ISS1:1OF1       p31786_1  30/04/2012  p31786_1.cpl   NO
066  wi00907707      ISS1:1OF1       p31228_1  30/04/2012  p31228_1.cpl   NO
067  wi00857362      ISS1:1OF1       p30782_1  30/04/2012  p30782_1.cpl   NO
068  wi00974635      ISS1:1OF1       p31695_1  30/04/2012  p31695_1.cpl   YES
069  wi00894443      ISS1:1OF1       p31093_1  30/04/2012  p31093_1.cpl   NO
070  wi00942734      ISS1:1OF1       p31409_1  30/04/2012  p31409_1.cpl   NO
071  wi00841273      ISS1:1OF1       p30713_1  30/04/2012  p30713_1.cpl   NO
072  wi00974272      ISS1:1OF1       p31690_1  30/04/2012  p31690_1.cpl   YES
073  wi00948931      ISS1:1OF1       p31407_1  30/04/2012  p31407_1.cpl   NO
074  wi00891626      ISS1:1OF1       p31051_1  30/04/2012  p31051_1.cpl   YES
075  wi00929140      ISS1:1OF1       p31284_1  30/04/2012  p31284_1.cpl   NO
076  wi00925208      ISS1:1OF1       p30986_1  30/04/2012  p30986_1.cpl   NO
077  wi00958776      ISS1:1OF1       p31542_1  30/04/2012  p31542_1.cpl   YES
078  wi00880836      ISS1:1OF1       p30976_1  30/04/2012  p30976_1.cpl   NO
079  WI00927300      ISS1:1OF1       p30999_1  30/04/2012  p30999_1.cpl   NO
080  wi00943172      ISS1:1OF1       p31402_1  30/04/2012  p31402_1.cpl   NO
081  wi00826075      ISS1:1OF1       p30452_1  30/04/2012  p30452_1.cpl   NO
082  wi00881777      ISS1:1OF1       p25747_1  30/04/2012  p25747_1.cpl   NO
083  wi00948274      ISS1:1OF1       p31365_1  30/04/2012  p31365_1.cpl   NO
```

084	wi00908933	ISS1:1OF1	p31239_1	30/04/2012	p31239_1.cpl	NO
085	wi00865477	ISS1:1OF1	p30892_1	30/04/2012	p30892_1.cpl	YES
086	wi00968531	ISS1:1OF1	p31645_1	30/04/2012	p31645_1.cpl	NO
087	wi00961267	ISS1:1OF1	p30288_1	30/04/2012	p30288_1.cpl	NO
088	wi00930864	ISS1:1OF1	p31325_1	30/04/2012	p31325_1.cpl	NO
089	wi00898200	ISS1:1of1	p31274_1	30/04/2012	p31274_1.cpl	NO
090	wi00946876	ISS1:1OF1	p31430_1	30/04/2012	p31430_1.cpl	NO
091	wi00936714	ISS1:1OF1	p31379_1	30/04/2012	p31379_1.cpl	NO
092	wi00959457	ISS1:1OF1	p31551_1	30/04/2012	p31551_1.cpl	NO
093	wi00969581	ISS1:1OF1	p31661_1	30/04/2012	p31661_1.cpl	YES
094	wi00956885	ISS1:1OF1	p31489_1	30/04/2012	p31489_1.cpl	NO
095	wi00973241	ISS1:1OF1	p31715_1	30/04/2012	p31715_1.cpl	NO
096	wi00946282	ISS1:1OF1	p31204_1	30/04/2012	p31204_1.cpl	NO
097	wi00840590	ISS1:1OF1	p30767_1	30/04/2012	p30767_1.cpl	NO
098	wi00897082	ISS1:1OF1	p31124_1	30/04/2012	p31124_1.cpl	NO
099	wi00896394	ISS1:1OF1	p30807_1	30/04/2012	p30807_1.cpl	NO
100	wi00909476	ISS1:1OF1	p31340_1	30/04/2012	p31340_1.cpl	NO
101	wi00887744	ISS2:1OF1	p31026_2	30/04/2012	p31026_2.cpl	NO
102	wi00865477	ISS1:1OF1	p30896_1	30/04/2012	p30896_1.cpl	YES
103	wi00957252	ISS1:1OF1	p31530_1	30/04/2012	p31530_1.cpl	NO
104	wi00859123	ISS1:1OF1	p30648_1	30/04/2012	p30648_1.cpl	NO
105	wi00895181	ISS1:1OF1	p31106_1	30/04/2012	p31106_1.cpl	NO
106	wi00938555	ISS1:1OF1	p30881_1	30/04/2012	p30881_1.cpl	YES
107	wi00941500	ISS1:1OF1	p31394_1	30/04/2012	p31394_1.cpl	NO
108	wi00931028	ISS1:1OF1	p31354_1	30/04/2012	p31354_1.cpl	YES
109	wi00907697	ISS1:1OF1	p31227_1	30/04/2012	p31227_1.cpl	NO
110	wi00905660	ISS1:1OF1	p27968_1	30/04/2012	p27968_1.cpl	NO
111	wi00900096	ISS1:1OF1	p31006_1	30/04/2012	p31006_1.cpl	NO
112	wi00900766	ISS1:1OF1	p31159_1	30/04/2012	p31159_1.cpl	NO
113	wi00865477	ISS1:1OF1	p30898_1	30/04/2012	p30898_1.cpl	YES
114	wi00906022	ISS1:1OF1	p31202_1	30/04/2012	p31202_1.cpl	NO
115	wi00856991	ISS1:1OF1	p17588_1	30/04/2012	p17588_1.cpl	NO
116	wi00880386	ISS1:1OF1	p30977_1	30/04/2012	p30977_1.cpl	NO
117	wi00688381	ISS1:1OF1	p30104_1	30/04/2012	p30104_1.cpl	NO
118	wi00908598	ISS1:1OF1	p31235_1	30/04/2012	p31235_1.cpl	NO
119	wi00890475	p30952	p31048_1	30/04/2012	p31048_1.cpl	NO
120	wi00868729	ISS1:1OF1	p31163_1	30/04/2012	p31163_1.cpl	NO
121	wi00956788	ISS1:1OF1	p31638_1	30/04/2012	p31638_1.cpl	NO
122	wi00859499	ISS1:1OF1	p30694_1	30/04/2012	p30694_1.cpl	NO
123	wi00895090	ISS1:1OF1	p31105_1	30/04/2012	p31105_1.cpl	NO
124	wi00869243	ISS1:1OF1	p30848_1	30/04/2012	p30848_1.cpl	NO
125	wi00930649	ISS1:1OF1	p31570_1	30/04/2012	p31570_1.cpl	NO
126	wi00899584	ISS1:1OF1	p30809_1	30/04/2012	p30809_1.cpl	NO
127	wi00932204	ISS2:1OF1	p31305_2	30/04/2012	p31305_2.cpl	NO
128	wi00951837	ISS1:1OF1	p31485_1	30/04/2012	p31485_1.cpl	NO
129	wi00865477	ISS1:1OF1	p30893_1	30/04/2012	p30893_1.cpl	YES
130	wi00946477	ISS1:1OF1	p31426_1	30/04/2012	p31426_1.cpl	NO
131	wi00959284	ISS1:1OF1	p31531_1	30/04/2012	p31531_1.cpl	NO
132	wi00855423	ISS1:1OF1	p31328_1	30/04/2012	p31328_1.cpl	YES
133	wi00900668	ISS1:1OF1	p30456_1	30/04/2012	p30456_1.cpl	NO
134	wi00862574	iss1:1of1	p30870_1	30/04/2012	p30870_1.cpl	NO
135	wi00894243	ISS1:1OF1	p31087_1	30/04/2012	p31087_1.cpl	NO
136	wi00959820	ISS1:1OF1	p31562_1	30/04/2012	p31562_1.cpl	NO
137	WI00889786	ISS1:1OF1	p30750_1	30/04/2012	p30750_1.cpl	NO
138	wi00943748	ISS1:1OF1	p31516_1	30/04/2012	p31516_1.cpl	NO
139	wi00959463	ISS1:1OF1	p31528_1	30/04/2012	p31528_1.cpl	NO
140	WI00928455	ISS1:1OF1	p31297_1	30/04/2012	p31297_1.cpl	NO
141	wi00896680	ISS1:1OF1	p30357_1	30/04/2012	p30357_1.cpl	NO
142	wi00925141	ISS1:1OF1	p30802_1	30/04/2012	p30802_1.cpl	NO
143	wi00968157	ISS1:1OF1	p31637_1	30/04/2012	p31637_1.cpl	NO
144	wi00884699	ISS1:1OF1	p31000_1	30/04/2012	p31000_1.cpl	YES
145	wi00932958	ISS1:1OF1	p31115_1	30/04/2012	p31115_1.cpl	NO
146	wi00921295	ISS1:1OF1	p31265_1	30/04/2012	p31265_1.cpl	NO
147	wi00906163	ISS1:1OF1	p31205_1	30/04/2012	p31205_1.cpl	NO
148	wi00903437	ISS1:1OF1	p31167_1	30/04/2012	p31167_1.cpl	NO
149	wi00960133	ISS2:1OF1	p31557_2	30/04/2012	p31557_2.cpl	NO
150	wi00879322	ISS1:1OF1	p30954_1	30/04/2012	p30954_1.cpl	NO
151	wi00896420	ISS1:1OF1	p30867_1	30/04/2012	p30867_1.cpl	NO
152	wi00924886	ISS1:1OF1	p31062_1	30/04/2012	p31062_1.cpl	YES
153	wi00877592	ISS1:1OF1	p30880_1	30/04/2012	p30880_1.cpl	NO

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154  wi00958682      ISS1:1OF1       p31540_1  30/04/2012  p31540_1.cpl   NO
155  wi00882293      ISS1:1OF1       p31010_1  30/04/2012  p31010_1.cpl   NO
156  wi00905297      ISS1:1OF1       p31195_1  30/04/2012  p31195_1.cpl   NO
157  wi00968353      ISS1:1OF1       p31412_1  30/04/2012  p31412_1.cpl   NO
158  wi00975133      ISS1:1OF1       p31731_1  30/04/2012  p31731_1.cpl   NO
159  wi00897096      ISS1:1OF1       p30676_1  30/04/2012  p30676_1.cpl   NO
160  wi00969890      ISS1:1OF1       p31664_1  30/04/2012  p31664_1.cpl   YES
161  wi00967510      ISS1:1OF1       p31147_1  30/04/2012  p31147_1.cpl   NO
162  wi00891621      ISS1:1OF1       p31037_1  30/04/2012  p31037_1.cpl   NO
163  wi00968448      ISS1:1OF1       p31648_1  30/04/2012  p31648_1.cpl   YES
164  wi00945997      ISS1:1OF1       p31641_1  30/04/2012  p31641_1.cpl   NO
165  wi00967509      ISS1:1OF1       p31294_1  30/04/2012  p31294_1.cpl   NO
166  wi00969208      ISS1:1OF1       p31656_1  30/04/2012  p31656_1.cpl   NO
167  wi00976209      ISS1:1OF1       p31717_1  30/04/2012  p31717_1.cpl   YES
168  wi00969039      ISS1:1OF1       p31643_1  30/04/2012  p31643_1.cpl   NO
169  wi00973858      ISS1:1OF1       p31691_1  30/04/2012  p31691_1.cpl   NO
170  wi00950575      ISS1:1OF1       p31724_1  30/04/2012  p31724_1.cpl   NO
171  wi00975659      ISS1:1OF1       p31707_1  30/04/2012  p31707_1.cpl   NO
172  wi00949410      ISS1:1OF1       p31248_1  30/04/2012  p31248_1.cpl   NO
173  wi00967754      ISS1:1OF1       p31653_1  30/04/2012  p31653_1.cpl   YES
174  wi00965285      ISS1:1OF1       p31476_1  30/04/2012  p31476_1.cpl   NO
175  wi00978007      ISS1:1OF1       p31737_1  30/04/2012  p31737_1.cpl   NO
176  wi00982243      ISS1:1OF1       p31797_1  30/04/2012  p31797_1.cpl   NO
177  wi00960809      ISS1:1OF1       p31564_1  30/04/2012  p31564_1.cpl   NO
178  wi00964006      ISS1:1OF1       p31595_1  30/04/2012  p31595_1.cpl   YES
179  wi00965838      ISS1:1OF1       p31623_1  30/04/2012  p31623_1.cpl   NO
180  wi00977393      ISS1:1OF1       p31744_1  30/04/2012  p31744_1.cpl   YES
MDP>LAST SUCCESSFUL MDP REFRESH :2012-03-27 14:39:08(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2012-03-27 06:55:16(est)
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Communication Server 1000E signaling server service updates

```
Product Release: 7.50.17.00
In system patches: 0

In System service updates: 22
PATCH#  IN_SERVICE    DATE        SPECINS   REMOVABLE   NAME
1       Yes           27/03/12    NO        YES         cs1000-ftrpkg-7.50.17.16-9.i386.000
2       Yes           29/02/12    NO        YES         cs1000-patchWeb-7.50.17.16-4.i386.000
3       Yes           01/03/12    NO        YES         cs1000-csmWeb-7.50.17.16-3.i386.000
4       Yes           18/04/11    NO        YES         cs1000-dbcom-7.50.17-02.i386.000
5       Yes           01/03/12    NO        YES         cs1000-mscAnnc-7.50.17.16-1.i386.000
6       Yes           01/03/12    NO        YES         cs1000-mscTone-7.50.17.16-1.i386.000
7       Yes           01/03/12    NO        YES         cs1000-mscMusc-7.50.17.16-2.i386.000
8       Yes           01/03/12    NO        YES         cs1000-dmWeb-7.50.17.16-2.i386.000
10      Yes           27/03/12    NO        YES         cs1000-linuxbase-7.50.17.16-07.i386.000
11      Yes           17/01/12    NO        YES         cs1000-baseWeb-7.50.17.16-1.i386.001
12      Yes           17/01/12    NO        YES         cs1000-shared-pbx-7.50.17.16-1.i386.000
13      Yes           17/01/12    NO        YES         cs1000-kcv-7.50.17.16-1.i386.000
14      Yes           27/03/12    NO        YES         cs1000-sps-7.50.17.16-4.i386.000
15      Yes           17/01/12    NO        YES         cs1000-ipsec-7.50.17.16-1.i386.000
16      Yes           27/03/12    NO        YES         cs1000-tps-7.50.17.16-13.i386.000
17      Yes           27/03/12    NO        YES         cs1000-emWeb_6-0-7.50.17.16-19.i386.000
18      Yes           27/03/12    NO        YES         cs1000-bcc-7.50.17.16-51.i386.000
19      Yes           17/01/12    NO        YES         ipsec-tools-0.6.5-14.el5.3_avaya_1.i386.0
20      Yes           17/01/12    NO        YES         spiritAgent-6.1-1.0.0.108.208.i386.000
21      Yes           17/01/12    NO        YES         cs1000-EmCentralLogic-7.50.17.16-1.i386.0
22      Yes           27/03/12    NO        YES         cs1000-Jboss-Quantum-7.50.17.16-16.i386.0
23      Yes           27/03/12    NO        YES         cs1000-vtrk-7.50.17.16-46.i386.000
```

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+EM
Packages:
CS+SS+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
   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
   managedElementWebService     7.50.17
   mscAnnc                      7.50.17.16  [patched]
   mscAttn                      7.50.17
   mscConf                      7.50.17
   mscMusc                      7.50.17.16  [patched]
   mscTone                      7.50.17.16  [patched]
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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.