



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.

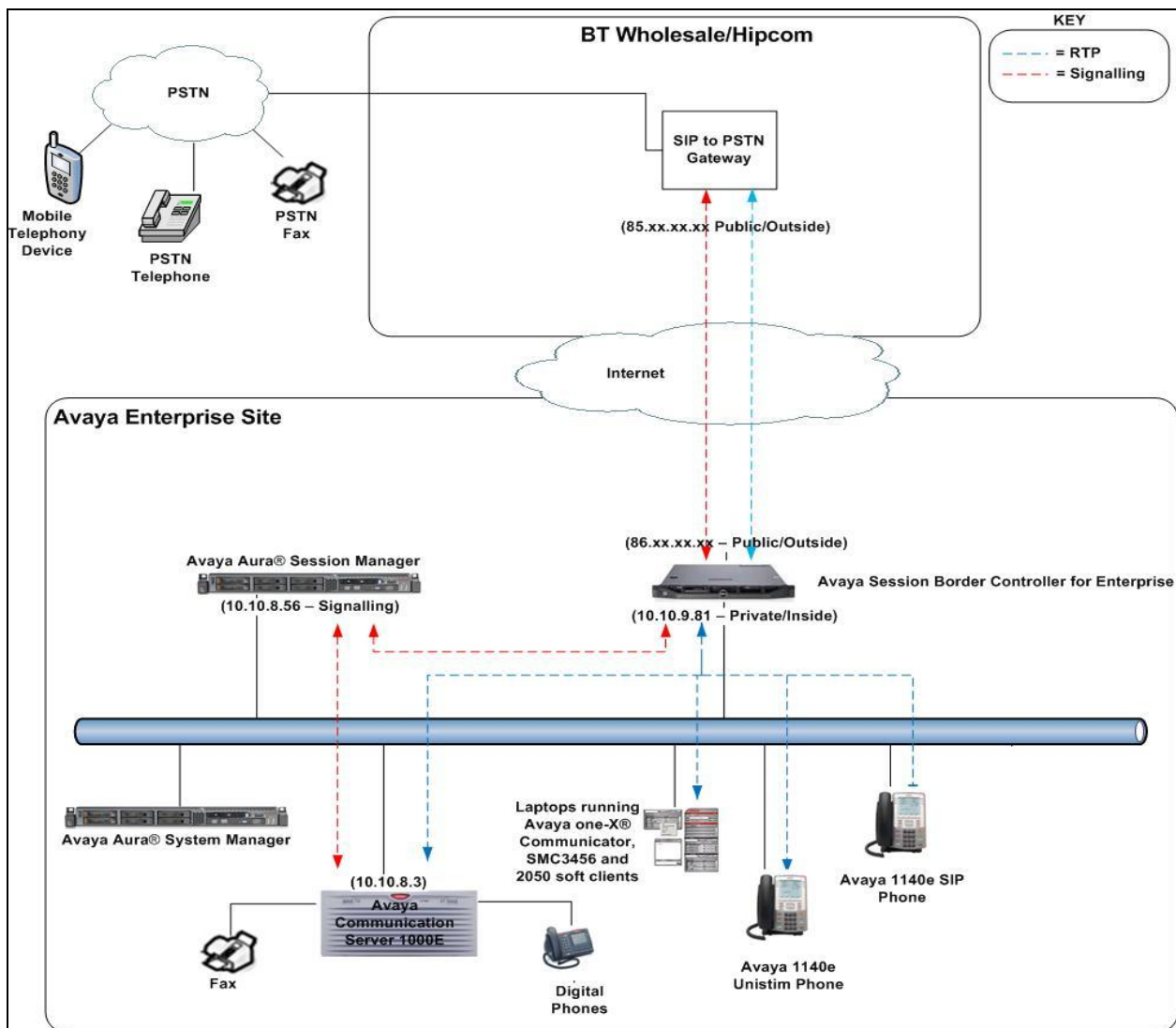


Figure 1: BT Wholesale/HIPCOM SIP Trunk topology with Avaya Communication Server 1000E

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 Deplst: 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

Node ID: 5000 - Voice Gateway (VGW) and Codecs

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

General

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

Voice activity detection threshold: (-20 - +10 DBM)
Idle noise level: (-327 - +327 DBM)

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

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.

Node ID: 5000 - Voice Gateway (VGW) and Codecs

General | Voice Codecs | Fax

Codec G711: ☒ Enabled (required)

Voice payload size: 20 (milliseconds per frame)

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

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

☐ Voice Activity Detection (VAD)

Codec G722: ☐ Enabled

Voice payload size: 20 (milliseconds per frame)

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

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

Codec G729: ☒ Enabled

Voice payload size: 30 (milliseconds per frame)

Voice playout (jitter buffer) delay: 60 120 (milliseconds)

* Required Value.

Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Save Cancel

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**.

Node ID: 5000 - Voice Gateway (VGW) and Codecs

General | Voice Codecs | Fax

Codec G723.1: ☐ Enabled

Voice payload size: 30 (milliseconds per frame)

Voice playout (jitter buffer) delay: 60 120 (milliseconds)

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

Coding rate: 5.3 (kbps)

Fax

Codec name: T.38 FAX

Maximum rate: 14400 (bps)

Fax TCF method: 2

Fax playout nominal delay: 100 (0 - 300 milliseconds)

FAX no activity timeout: 20 (10 - 32000 milliseconds)

Packet size: 20 (bps)

* Required Value.

Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Save Cancel

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).

CS1000 Element Manager

Managing: 192.168.0.2 Username: admin
System » IP Network » IP Telephony Nodes » Node Details

Node Details (ID: 5000 - SIP Line, LTPS, PD, Gateway (SIPGw))

Node ID: * (0-9999)

Call server IP address: * TLAN address type: ☒ IPv4 only
☐ IPv4 and IPv6

Embedded LAN (ELAN) Telephony LAN (TLAN)

Gateway IP address: * Node IPv4 address: *

Subnet mask: * Subnet mask: *

Node IPv6 address:

* Required Value.

Associated Signaling Servers & Cards

Select to add [Print](#) [Refresh](#)

Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/> spcs1k	Signaling_Server	SIP Line, LTPS, Gateway, PD, Presence Publisher, IP Media Services	192.168.0.2	10.10.8.2	Leader

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.

Node ID: 5000 - 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: lab.ic.static.hipcom.co.uk *

Local SIP port: 5060 *(1 - 65535)

Gateway endpoint name: spcs1k *

Gateway password: *

Application node ID: 5000 *(0-9999)

Enable failsafe NRS: ☐

SIP ANAT: ☒ IPv4 ☐ IPv6

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)

Information will be captured for the IP addresses listed below:

Monitor IP: Add

Monitor addresses:

Remove

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 10.10.8.56

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	Private domain names
National: <input type="text"/>	UDP: <input type="text"/>
Subscriber: <input type="text"/>	CDP: <input type="text"/>
Special number: <input type="text"/>	Special number: <input type="text"/>
Unknown: <input type="text"/>	Vacant number: <input type="text"/>
	Unknown: <input type="text"/>

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.

The screenshot shows the CS1000 Element Manager interface. The left sidebar contains a navigation tree with 'Zones' highlighted. The main panel is titled 'Bandwidth Zones' and contains a table with the following data:

Zone	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description
1 10	1000000	BQ	1000000	BB	SHARED	MO	MAINOFFICE
2 20	1000000	BQ	1000000	BB	SHARED	VTRK	VTRK

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.

The screenshot shows the CS1000 Element Manager interface for 'Digit Conversion Tree 0 Configuration'. The main panel contains a table with the following data:

Incoming Digits	Converted Digits	CPND Name	CPND language
1 16159	8000		
2 16160	8889		
3 16161	8001		
4 16162	8050		

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
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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 IPT1
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 new CUST 0 FEAT rlb RLI 24 ELC NO ENTR 0 LTER NO ROUT 100 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 0 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 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 FLEN 3 ITOH NO CLTP NONE RLI 24 SDRR NONE ITEI NONE	SPN 90 FLEN 7 ITOH NO CLTP NONE RLI 24 SDRR NONE ITEI NONE	SPN 2 FLEN 7 ITOH NO CLTP NONE RLI 24 SDRR NONE ITEI NONE	SPN 15 FLEN 3 ITOH NO CLTP NONE RLI 24 SDRR 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 RCBF
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 CCBF RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
FDSF NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA PKCH MUTA MWTD
```

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```
DVLD CROD CROD
CPND_LANG ENG
RCO 0
HUNT 0
LHK 0
PLEV 02
PUID
DANI NO
AST 00
IAPG 1
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 MCR 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 RCBF
    ICDA CDMA LLCN MCTD CLBD AUTU
    GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD
    CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
    UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
    DRDD EXR0
    USMD USRD ULAD CCBF RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
    FDSF 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
---continued on next page---
```

---continued from previous page----

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 CCBF BNRD OCBF RTDD RBDD RBHD FAXA CNUD CNAD PGND FTTC
      FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTD
PLEV 02
PUID
AACS NO
MLWU_LANG 0
FTR DCFW 4
```

5.9. Configure the SIP Line Gateway Service

SIP terminal operation requires the Communication Server node to be configured as a SIP Line Gateway (SLG) before SIP telephones can be configured. Prior to configuring the SIP Line node properties, the SIP Line service must be enabled in the customer data block. Use the 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**

AVAYA CS1000 Element Manager

Managing: 192.168.0.2 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration

Node ID: 5000 - SIP Line Configuration Details

General | SIP Line Gateway Settings | SIP Line Gateway Service

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

General

SIP domain name: lab.ic.static.hipcom.co.uk *

SLG endpoint name: sps1k

SLG Group ID:

SLG Local Sip port: 5070 (1 - 65535)

SLG Local TLS port: 5071 (1 - 65535)

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

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 RCBF
      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 HBTB AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD MSNV FRA PKCH MWTD DVLD
CROD CROD
CPND_LANG ENG
RCO 0
HUNT
LHK 0
PLEV 02
PUID
DANI NO
AST
IAPG 0 *

AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 MCR 8889 0      MARP
      CPND
      CPND_LANG ROMAN
      NAME Sigma 1140
      XPLN 11
      DISPLAY_FMT FIRST, LAST*
01 HOT U 78889 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.

The screenshot shows the AVAYA CS1000 Element Manager web interface. On the left is a navigation tree with categories like UCM Network Services, System, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, and Tools. The 'Tools' category is expanded, showing 'Backup and Restore' and 'Call Server'. The main content area is titled 'Call Server Backup'. It shows the managing IP as 10.80.51.60 and the user as admin. Below the title, there is an 'Action' dropdown menu set to 'Backup', and two buttons: 'Submit' (highlighted with a red box) and 'Cancel'.

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

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

Configuration of 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.

The screenshot displays the Avaya Aura® System Manager 6.1 interface. The top header includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.1', and links for 'Help | About | Change Password | Log off admin'. Below the header is a breadcrumb trail: 'Home / Elements / Routing / Domains - Domain Management'. The left navigation pane shows 'Routing' expanded, with 'Domains' selected. The main content area is titled 'Domain Management' and includes buttons for 'Edit', 'New', 'Duplicate', 'Delete', and 'More Actions'. Below these buttons is a table with one item. The table has columns for 'Name', 'Type', 'Default', and 'Notes'. The row shows the domain 'lab.ic.static.hipcom.co.uk' with type 'sip' and a checkbox for 'Default'.

Name	Type	Default	Notes
lab.ic.static.hipcom.co.uk	sip	<input type="checkbox"/>	

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.

The screenshot displays the configuration interface for a Location. On the left is a sidebar with a menu containing: Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area has a top message: "Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth. See Session Manager -> Session Manager Administration -> Global Setting".

The **General** section includes a "Name" field with the value "SipLab8" and an empty "Notes" field.

The **Overall Managed Bandwidth** section shows "Managed Bandwidth Units" set to "Kbit/sec" and an empty "Total Bandwidth" field.

The **Per-Call Bandwidth Parameters** section shows "Default Audio Bandwidth" set to "80" Kbit/sec.

The **Location Pattern** section has "Add" and "Remove" buttons. Below is a table with 2 items. The table has columns for "IP Address Pattern" and "Notes". The first row shows "10.10.2.*" and the second row shows "10.10.8.*".

IP Address Pattern	Notes
* 10.10.2.*	
* 10.10.8.*	

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

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Pas](#)

Home / Elements / Routing / Adaptations- Adaptation Details

Adaptation Details

General

* Adaptation name:

Module name:

Module parameter:

Egress URI Parameters:

Notes:

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 screenshot shows the 'SIP Entity Details' configuration page for a Session Manager SIP Entity. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities (selected), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and 'General'. The configuration fields are as follows:

- Name:** Session Manager
- FQDN or IP Address:** 10.10.8.56
- Type:** Session Manager (dropdown)
- Notes:** (empty text field)
- Location:** SipLab8 (dropdown)
- Outbound Proxy:** (empty dropdown)
- Time Zone:** Europe/Dublin (dropdown)
- Credential name:** (empty text field)
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown)

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

Port

Add Remove

3 Items Refresh Filter: Enable

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

Select : All, None

* Input Required

Commit Cancel

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**.

Routing

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

SIP Entity Details

General

* Name: CS1K

* FQDN or IP Address: 10.10.8.3

Type: Other

Notes:

Adaptation: SipLab8

Location: SipLab8

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Commit

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.

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

SIP Entity Details

General

* Name: Sipera SBC

* FQDN or IP Address: 10.10.9.81

Type: Gateway

Notes:

Adaptation: Mime

Location: SipLab8

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

Credential name:

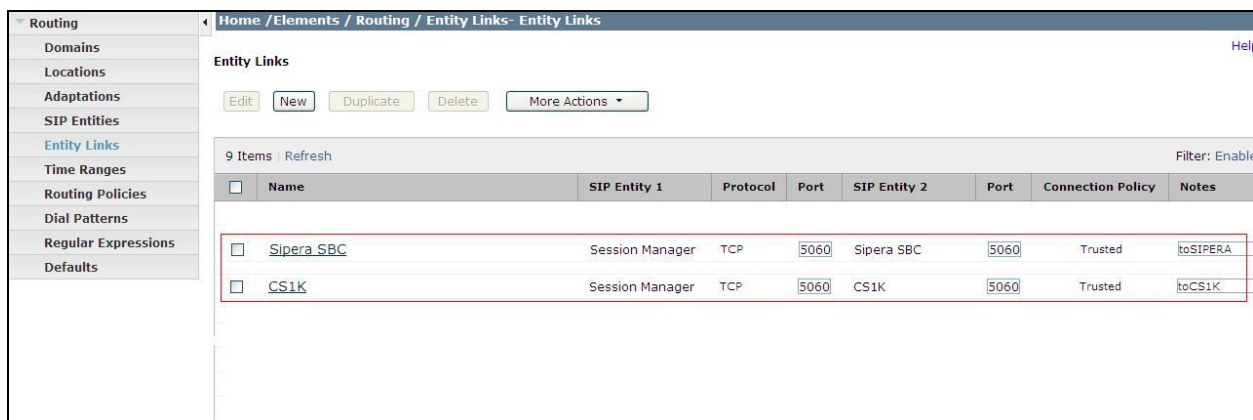
Call Detail Recording: none

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.



	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
<input type="checkbox"/>	Sipera SBC	Session Manager	TCP	5060	Sipera SBC	5060	Trusted	toSIPERA
<input type="checkbox"/>	CS1K	Session Manager	TCP	5060	CS1K	5060	Trusted	toCS1K

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 screenshot shows the 'Routing Policy Details' form for 'toCS1K'. The 'General' section has 'Name' set to 'toCS1K'. The 'SIP Entity as Destination' section shows 'CS1K' selected. The 'Time of Day' section shows a single time range '24/7' from 00:00 to 23:59.

Name	FQDN or IP Address	Type	Notes
CS1K	10.10.8.3	Other	

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	✓	✓	✓	✓	✓	✓	✓	00:00	23:59	Time Range 24/7

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

The screenshot shows the 'Routing Policy Details' form for 'Sipera SBC'. The 'General' section has 'Name' set to 'Sipera SBC'. The 'SIP Entity as Destination' section shows 'Sipera SBC' selected. The 'Time of Day' section shows a single time range '24/7' from 00:00 to 23:59.

Name	FQDN or IP Address	Type	Notes
Sipera SBC	10.10.9.81	Gateway	

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	✓	✓	✓	✓	✓	✓	✓	00:00	23:59	Time Range 24/7

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 screenshot displays the 'Dial Pattern Details' configuration page. The left sidebar contains a menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, **Dial Patterns**, Regular Expressions, and Defaults. The main content area is titled 'Dial Pattern Details' and has a 'Commit' button. The 'General' tab is active, showing fields for Pattern (00353), Min (5), Max (16), SIP Domain (-ALL-), and Notes. Below this is the 'Originating Locations and Routing Policies' section, which includes an 'Add' button and a table with one item. The table has columns: Originating Location Name, Originating Location Notes, Routing Policy Name, Rank, Routing Policy Disabled, Routing Policy Destination, and Routing Policy Notes. The item in the table is '-ALL-' with Rank 0 and Routing Policy Destination 'Sipera SBC'.

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
-ALL-	Any Locations	Sipera SBC	0	<input type="checkbox"/>	Sipera SBC	

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.

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

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

Dial Pattern Details

Commit

Cancel

Help

General

* Pattern:

44203

* Min:

5

* Max:

12

Emergency Call:

☐

SIP Domain:

-ALL-

Notes:

Originating Locations and Routing Policies

Add

Remove

1 Item

Refresh

Filter: Enable

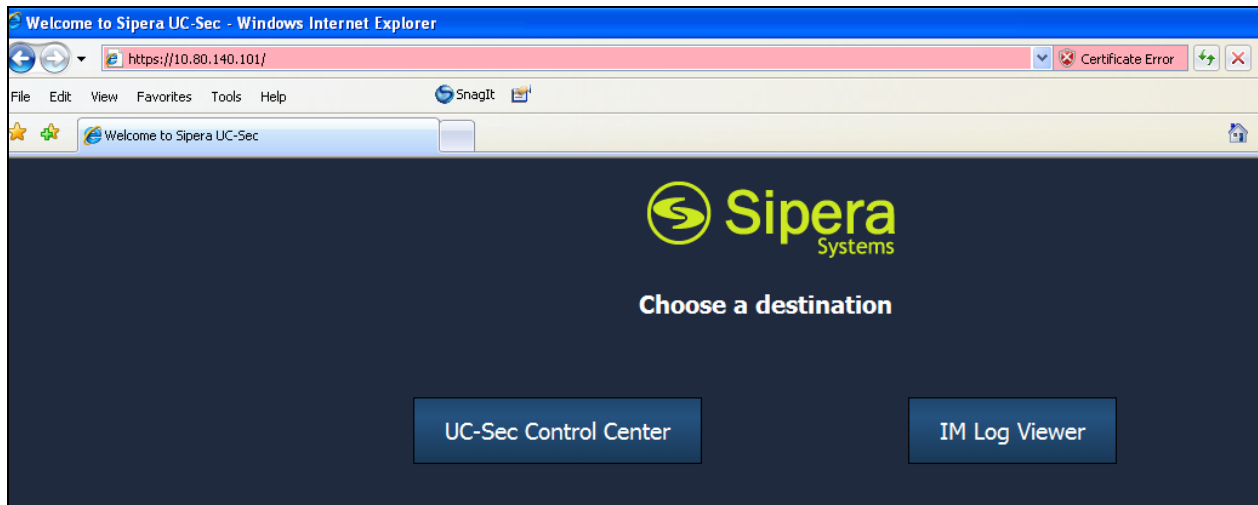
<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	toCS1K	0	<input type="checkbox"/>	CS1K	

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 Interworking 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**.

Editing Profile: SM9_Call_Server	
General	
Hold Support	<input type="radio"/> None <input checked="" type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543
Next	

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**.

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

Next

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).

Global Profiles > Routing: SM9_Call_Server

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

Click here to add a description.

Routing Profile

Add Routing Rule

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	10.10.8.56	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	TCP

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)

Global Profiles > Routing: SP_Trunk_Server

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

Click here to add a description.

Routing Profile

Add Routing Rule

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	85. [REDACTED]	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	UDP

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_Server** 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 <small>Comma separated list</small>	85.
Supported Transports	<input type="checkbox"/> TCP <input checked="" type="checkbox"/> UDP <input type="checkbox"/> TLS
TCP Port	
UDP Port	5060
TLS Port	

Finish

Edit Server Configuration Profile - Authentication

Enable Authentication	<input checked="" type="checkbox"/>
User Name	442
Realm	
Password <small>(Leave blank to keep existing password)</small>	
Confirm Password	

Finish

Edit Server Configuration Profile - Advanced

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

Finish

7.2.7. 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.

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

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.

Header	Criteria	Replace Action	Override Value
To	IP/Domain	Overwrite	lab.ic.static.hipcom.co.uk
Request-Line	IP/Domain	Overwrite	lab.ic.static.hipcom.co.uk
From	IP/Domain	Overwrite	lab.ic.static.hipcom.co.uk
Record-Route	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Via	IP/Domain	Auto	---

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:

```
1 within session "ALL"
2 {
3   act on message where $DIRECTION="OUTBOUND" and $ENTRY_POINT="POST_ROUTING"
4   {
5     $HEADERS["Allow"][1].regex_replace("UPDATE", "");
6   }
7 }
```

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

ent: GSSCP-SBC1

Network Configuration | Interface Configuration

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

A1 Netmask: 255.255.255.0 A2 Netmask: B1 Netmask: 255.255.255.128 B2 Netmask:

Add IP Save Changes Clear Changes

IP Address	Public IP	Gateway	Interface	
10.10.9.81		10.10.9.1	A1	✖
86. [redacted] S2		86. [redacted] 7	B1	✖

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

Network Configuration		Interface Configuration
Name		Administrative Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

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.

Device Specific Settings > Media Interface: GSSCP-SBC1

UC-Sec Devices

GSSCP-SBC1

Media Interface

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

Add Media Interface

Name	Media IP	Port Range		
Int_Media	10.10.9.81	35000 - 40000		
Ext_Media	86.52	35000 - 40000		

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.

Device Specific Settings > Signaling Interface: GSSCP-SBC1																														
<div>UC-Sec Devices</div> <div>GSSCP-SBC1</div>	Signaling Interface																													
	<div>Add Signaling Interface</div> <table> <tr> <th>Name</th><th>Signaling IP</th><th>TCP Port</th><th>UDP Port</th><th>TLS Port</th><th>TLS Profile</th><th></th><th></th></tr> <tr> <td>Int_Sig</td><td>10.10.9.81</td><td>5060</td><td>5060</td><td>---</td><td>None</td><td></td><td></td></tr> <tr> <td>Ext_Sig</td><td>86. [REDACTED]</td><td>---</td><td>5060</td><td>---</td><td>None</td><td></td><td></td></tr> </table>							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	
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)

Device Specific Settings > End Point Flows: GSSCP-SBC1

UC-Sec Devices
GSSCP-SBC1

Subscriber Flows | Server Flows

Add Flow

Click here to add a row description.

Server Configuration: SM9_Call_Server

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile			
1	SM9_Call_Server	*	*	*	Ext_Sig	Int_Sig	Int_Media	default-low	SP_Trunk_Server	SM9_CS	None			

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**

Device Specific Settings > End Point Flows: GSSCP-SBC1

UC-Sec Devices
GSSCP-SBC1

Subscriber Flows Server Flows

Add Flow

Click here to add a row description.

Server Configuration: SP_Trunk_Server

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile		
1	SP_Trunk_Server	*	*	*	Int_Sig	Ext_Sig	Ext_Media	default-low	SM9_Call_Server	SP_Trunk	None		

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.

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

Maintenance	
Managing: 10.80.51.80 Username: admin System » Maintenance	
Maintenance	
<input checked="" type="radio"/> Select by Overlay <input type="radio"/> Select by Functionality	
<Select by Overlay>	
LD 30 - Network and Signaling	
LD 32 - Network and Peripheral Equipment	
LD 34 - Tone and Digit Switch	
LD 36 - Trunk	
LD 37 - Input/Output	
LD 38 - Conference Circuit	
LD 39 - Intergroup Switch and System Clock	
LD 45 - Background Signaling and Switching	
LD 46 - Multifrequency Sender	
LD 48 - Link	
LD 54 - Multifrequency Signaling	
LD 60 - Digital Trunk Interface and Primary Rate Interface	
LD 75 - Digital Trunk	
LD 80 - Call Trace	
LD 96 - D-Channel	
LD 117 - Ethernet and Alarm Management	
LD 135 - Core Common Equipment	
LD 137 - Core Input/Output	
LD 143 - Centralized Software Upgrade	

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

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

- **Appl_Status** Verify status is **OPER**
- **Link_Status** Verify status is **EST ACTV**

D-Channel Diagnostics

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

DCH
DES
APPL_STATUS
LINK_STATUS
AUTO_RECV
PDCH
BDCH

☐ 010
Vtrk
OPER
EST
ACTV
AUTO

STAT DCH 010

Command executed successfully.

9.2. Verify Avaya Aura® Session Manager Operational Status

9.2.1. Verify Avaya Aura® Session Manager is Operational

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

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

Home /Elements / Session Manager- Session Manager

Session Manager
Dashboard
Session Manager
Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
System Tools

Session Manager Dashboard

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

Session Manager Instances

Service State
Shutdown System
As of 9:22 AM

1 Item Refresh Show ALL
Filter: Enable

	Session Manager	Type	Alarms	Tests Pass	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Version
<input type="checkbox"/>	Session Manager	Core	50/14/39	✓	Up	Accept New Service	0/5	0	0	6.1.0.0.610023

Select : All, None

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

Reset

Synchronize

Certificate Management ▾

Connection Status

1 Item

Refresh

Show

ALL ▾

Filter: Enable

	Details	Session Manager	Type	Status	Connections	IP Address	VLAN	Default Gateway	NIC Bonding	Entity Links (expected / actual)	Certificate Used
○	► Show	Session Manager	SM	Up	14	10.10.8.56/24	---	10.10.8.1	Disabled	5/5	SIP CA

Select : None

9.2.2. Verify SIP Entity Link Status

Navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring** (not shown) to view more detailed status information for one of the SIP Entity Links. Select the SIP Entity for Communication Server 1000E 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: CS1000 Rel7.5** table, verify the **Conn. Status** for the link is **Up** as shown below.

SIP Entity, Entity Link Connection Status								
This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.								
All Entity Links to SIP Entity: CS1K								
Summary View								
1 Item Refresh Filter: Enable								
Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status	
► Show	Session Manager	10.10.8.3	5060	TCP	Up	200 OK	Up	

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	ISS1:10F1	MGCCCD02	24/04/2012	MGCCCD02.LW
01	wi00832543	ISS1:10F1	DSP1AB04	24/04/2012	DSP1AB04.LW
02	wi00946113	ISS1:10F1	MGCBB15	24/04/2012	MGCBB15.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	ISS1:10F1	p30550_1	30/04/2012	p30550_1.cpl	NO
001	wi00835294	ISS1:10F1	p30565_1	30/04/2012	p30565_1.cpl	NO
002	wi00897176	ISS1:10F1	p30418_1	30/04/2012	p30418_1.cpl	NO
003	wi00925218	ISS1:10F1	p30675_1	30/04/2012	p30675_1.cpl	NO
004	wi00839821	ISS1:10F1	p30619_1	30/04/2012	p30619_1.cpl	NO
005	wi00937672	ISS1:10F1	p31276_1	30/04/2012	p31276_1.cpl	NO
006	wi00842409	ISS1:10F1	p30621_1	30/04/2012	p30621_1.cpl	NO
007	wi00838073	ISS1:10F1	p30588_1	30/04/2012	p30588_1.cpl	NO
008	wi00937114	ISS1:10F1	p31310_1	30/04/2012	p31310_1.cpl	NO
009	wi00841980	ISS1:10F1	p30618_1	30/04/2012	p30618_1.cpl	NO
010	wi00955753	ISS1:10F1	p31733_1	30/04/2012	p31733_1.cpl	NO
011	wi00839255	ISS1:10F1	p30591_1	30/04/2012	p30591_1.cpl	NO
012	wi00843623	ISS1:10F1	p30731_1	30/04/2012	p30731_1.cpl	YES
013	WI00843571	ISS1:10F1	p30627_1	30/04/2012	p30627_1.cpl	NO

014	wi00871739	ISS1:10F1	p30856_1	30/04/2012	p30856_1.cpl	NO
015	wi00852365	ISS1:10F1	p30707_1	30/04/2012	p30707_1.cpl	NO
016	wi00852389	ISS1:10F1	p30641_1	30/04/2012	p30641_1.cpl	NO
017	wi00839134	ISS1:10F1	p30698_1	30/04/2012	p30698_1.cpl	YES
018	wi00856702	ISS1:10F1	p30573_1	30/04/2012	p30573_1.cpl	NO
019	wi00857566	ISS1:10F1	p30766_1	30/04/2012	p30766_1.cpl	NO
020	wi00850521	ISS1:10F1	p30709_1	30/04/2012	p30709_1.cpl	YES
021	wi00903381	ISS1:10F1	p30421_1	30/04/2012	p30421_1.cpl	NO
022	wi00863876	ISS1:10F1	p30787_1	30/04/2012	p30787_1.cpl	NO
023	WI00853473	ISS1:10F1	p30625_1	30/04/2012	p30625_1.cpl	NO
024	wi00854130	ISS1:10F1	p30443_1	30/04/2012	p30443_1.cpl	NO
025	wi00875425	ISS1:10F1	p30943_1	30/04/2012	p30943_1.cpl	NO
026	wi00978883	ISS1:10F1	p31770_1	30/04/2012	p31770_1.cpl	NO
027	wi00875701	ISS1:10F1	p30942_1	30/04/2012	p30942_1.cpl	NO
028	wi00936935	ISS1:10F1	p31362_1	30/04/2012	p31362_1.cpl	NO
029	wi00877367	ISS1:10F1	p30534_1	30/04/2012	p30534_1.cpl	NO
030	wi00871969	ISS1:10F1	p30768_1	30/04/2012	p30768_1.cpl	NO
031	wi00886321	ISS1:10F1	p31009_1	30/04/2012	p31009_1.cpl	NO
032	WI00836334	ISS1:10F1	p30481_1	30/04/2012	p30481_1.cpl	NO
033	wi00836182	ISS1:10F1	p30450_1	30/04/2012	p30450_1.cpl	NO
034	wi00858335	ISS1:10F1	p30819_1	30/04/2012	p30819_1.cpl	NO
035	wi00860279	ISS1:10F1	p30789_1	30/04/2012	p30789_1.cpl	NO
036	wi00953900	ISS1:10F1	p31494_1	30/04/2012	p31494_1.cpl	NO
037	wi00854415	ISS1:10F1	p30593_1	30/04/2012	p30593_1.cpl	NO
038	WI00836292	ISS1:10F1	p30554_1	30/04/2012	p30554_1.cpl	NO
039	WI00839794	ISS1:10F1	p28647_1	30/04/2012	p28647_1.cpl	NO
040	wi00824257	ISS1:10F1	p30447_1	30/04/2012	p30447_1.cpl	NO
041	wi00827950	ISS2:10F1	p30471_2	30/04/2012	p30471_2.cpl	NO
042	wi00949273	ISS1:10F1	p31411_1	30/04/2012	p31411_1.cpl	NO
043	WI00854150	ISS1:10F1	p30468_1	30/04/2012	p30468_1.cpl	NO
044	wi00873382	ISS1:10F1	p30832_1	30/04/2012	p30832_1.cpl	NO
045	wi00853178	ISS1:10F1	p30719_1	30/04/2012	p30719_1.cpl	NO
046	wi00869695	ISS1:10F1	p30654_1	30/04/2012	p30654_1.cpl	NO
047	wi00834382	ISS1:10F1	p30548_1	30/04/2012	p30548_1.cpl	NO
048	wi00951427	ISS1:10F1	p31478_1	30/04/2012	p31478_1.cpl	NO
049	wi00946558	ISS1:10F1	p31358_1	30/04/2012	p31358_1.cpl	NO
050	wi00903369	ISS1:10F1	p31165_1	30/04/2012	p31165_1.cpl	NO
051	wi00927321	ISS1:10F1	p31286_1	30/04/2012	p31286_1.cpl	YES
052	wi00923899	ISS1:10F1	p31270_1	30/04/2012	p31270_1.cpl	NO
053	wi00949627	ISS1:10F1	p31462_1	30/04/2012	p31462_1.cpl	NO
054	wi00962557	ISS1:10F1	p31581_1	30/04/2012	p31581_1.cpl	NO
055	wi00865477	ISS1:10F1	p30894_1	30/04/2012	p30894_1.cpl	YES
056	wi00962211	ISS1:10F1	p31580_1	30/04/2012	p31580_1.cpl	NO
057	wi00883604	ISS1:10F1	p30973_1	30/04/2012	p30973_1.cpl	NO
058	wi00898327	ISS1:10F1	p31136_1	30/04/2012	p31136_1.cpl	NO
059	wi00856410	ISS1:10F1	p30749_1	30/04/2012	p30749_1.cpl	NO
060	wi00932948	ISS1:10F1	p31077_1	30/04/2012	p31077_1.cpl	NO
061	wi00905600	ISS1:10F1	p31201_1	30/04/2012	p31201_1.cpl	NO
062	wi00865477	ISS1:10F1	p30897_1	30/04/2012	p30897_1.cpl	YES
063	wi00879526	ISS1:10F1	p31007_1	30/04/2012	p31007_1.cpl	NO
064	wi00962955	ISS1:10F1	p31585_1	30/04/2012	p31585_1.cpl	NO
065	wi00984178	ISS1:10F1	p31786_1	30/04/2012	p31786_1.cpl	NO
066	wi00907707	ISS1:10F1	p31228_1	30/04/2012	p31228_1.cpl	NO
067	wi00857362	ISS1:10F1	p30782_1	30/04/2012	p30782_1.cpl	NO
068	wi00974635	ISS1:10F1	p31695_1	30/04/2012	p31695_1.cpl	YES
069	wi00894443	ISS1:10F1	p31093_1	30/04/2012	p31093_1.cpl	NO
070	wi00942734	ISS1:10F1	p31409_1	30/04/2012	p31409_1.cpl	NO
071	wi00841273	ISS1:10F1	p30713_1	30/04/2012	p30713_1.cpl	NO
072	wi00974272	ISS1:10F1	p31690_1	30/04/2012	p31690_1.cpl	YES
073	wi00948931	ISS1:10F1	p31407_1	30/04/2012	p31407_1.cpl	NO
074	wi00891626	ISS1:10F1	p31051_1	30/04/2012	p31051_1.cpl	YES
075	wi00929140	ISS1:10F1	p31284_1	30/04/2012	p31284_1.cpl	NO
076	wi00925208	ISS1:10F1	p30986_1	30/04/2012	p30986_1.cpl	NO
077	wi00958776	ISS1:10F1	p31542_1	30/04/2012	p31542_1.cpl	YES
078	wi00880836	ISS1:10F1	p30976_1	30/04/2012	p30976_1.cpl	NO
079	WI00927300	ISS1:10F1	p30999_1	30/04/2012	p30999_1.cpl	NO
080	wi00943172	ISS1:10F1	p31402_1	30/04/2012	p31402_1.cpl	NO
081	wi00826075	ISS1:10F1	p30452_1	30/04/2012	p30452_1.cpl	NO
082	wi00881777	ISS1:10F1	p25747_1	30/04/2012	p25747_1.cpl	NO
083	wi00948274	ISS1:10F1	p31365_1	30/04/2012	p31365_1.cpl	NO

084	wi00908933	ISS1:10F1	p31239_1	30/04/2012	p31239_1.cpl	NO
085	wi00865477	ISS1:10F1	p30892_1	30/04/2012	p30892_1.cpl	YES
086	wi00968531	ISS1:10F1	p31645_1	30/04/2012	p31645_1.cpl	NO
087	wi00961267	ISS1:10F1	p30288_1	30/04/2012	p30288_1.cpl	NO
088	wi00930864	ISS1:10F1	p31325_1	30/04/2012	p31325_1.cpl	NO
089	wi00898200	ISS1:10F1	p31274_1	30/04/2012	p31274_1.cpl	NO
090	wi00946876	ISS1:10F1	p31430_1	30/04/2012	p31430_1.cpl	NO
091	wi00936714	ISS1:10F1	p31379_1	30/04/2012	p31379_1.cpl	NO
092	wi00959457	ISS1:10F1	p31551_1	30/04/2012	p31551_1.cpl	NO
093	wi00969581	ISS1:10F1	p31661_1	30/04/2012	p31661_1.cpl	YES
094	wi00956885	ISS1:10F1	p31489_1	30/04/2012	p31489_1.cpl	NO
095	wi00973241	ISS1:10F1	p31715_1	30/04/2012	p31715_1.cpl	NO
096	wi00946282	ISS1:10F1	p31204_1	30/04/2012	p31204_1.cpl	NO
097	wi00840590	ISS1:10F1	p30767_1	30/04/2012	p30767_1.cpl	NO
098	wi00897082	ISS1:10F1	p31124_1	30/04/2012	p31124_1.cpl	NO
099	wi00896394	ISS1:10F1	p30807_1	30/04/2012	p30807_1.cpl	NO
100	wi00909476	ISS1:10F1	p31340_1	30/04/2012	p31340_1.cpl	NO
101	wi00887744	ISS2:10F1	p31026_2	30/04/2012	p31026_2.cpl	NO
102	wi00865477	ISS1:10F1	p30896_1	30/04/2012	p30896_1.cpl	YES
103	wi00957252	ISS1:10F1	p31530_1	30/04/2012	p31530_1.cpl	NO
104	wi00859123	ISS1:10F1	p30648_1	30/04/2012	p30648_1.cpl	NO
105	wi00895181	ISS1:10F1	p31106_1	30/04/2012	p31106_1.cpl	NO
106	wi00938555	ISS1:10F1	p30881_1	30/04/2012	p30881_1.cpl	YES
107	wi00941500	ISS1:10F1	p31394_1	30/04/2012	p31394_1.cpl	NO
108	wi00931028	ISS1:10F1	p31354_1	30/04/2012	p31354_1.cpl	YES
109	wi00907697	ISS1:10F1	p31227_1	30/04/2012	p31227_1.cpl	NO
110	wi00905660	ISS1:10F1	p27968_1	30/04/2012	p27968_1.cpl	NO
111	wi00900096	ISS1:10F1	p31006_1	30/04/2012	p31006_1.cpl	NO
112	wi00900766	ISS1:10F1	p31159_1	30/04/2012	p31159_1.cpl	NO
113	wi00865477	ISS1:10F1	p30898_1	30/04/2012	p30898_1.cpl	YES
114	wi00906022	ISS1:10F1	p31202_1	30/04/2012	p31202_1.cpl	NO
115	wi00856991	ISS1:10F1	p17588_1	30/04/2012	p17588_1.cpl	NO
116	wi00880386	ISS1:10F1	p30977_1	30/04/2012	p30977_1.cpl	NO
117	wi00688381	ISS1:10F1	p30104_1	30/04/2012	p30104_1.cpl	NO
118	wi00908598	ISS1:10F1	p31235_1	30/04/2012	p31235_1.cpl	NO
119	wi00890475	p30952	p31048_1	30/04/2012	p31048_1.cpl	NO
120	wi00868729	ISS1:10F1	p31163_1	30/04/2012	p31163_1.cpl	NO
121	wi00956788	ISS1:10F1	p31638_1	30/04/2012	p31638_1.cpl	NO
122	wi00859499	ISS1:10F1	p30694_1	30/04/2012	p30694_1.cpl	NO
123	wi00895090	ISS1:10F1	p31105_1	30/04/2012	p31105_1.cpl	NO
124	wi00869243	ISS1:10F1	p30848_1	30/04/2012	p30848_1.cpl	NO
125	wi00930649	ISS1:10F1	p31570_1	30/04/2012	p31570_1.cpl	NO
126	wi00899584	ISS1:10F1	p30809_1	30/04/2012	p30809_1.cpl	NO
127	wi00932204	ISS2:10F1	p31305_2	30/04/2012	p31305_2.cpl	NO
128	wi00951837	ISS1:10F1	p31485_1	30/04/2012	p31485_1.cpl	NO
129	wi00865477	ISS1:10F1	p30893_1	30/04/2012	p30893_1.cpl	YES
130	wi00946477	ISS1:10F1	p31426_1	30/04/2012	p31426_1.cpl	NO
131	wi00959284	ISS1:10F1	p31531_1	30/04/2012	p31531_1.cpl	NO
132	wi00855423	ISS1:10F1	p31328_1	30/04/2012	p31328_1.cpl	YES
133	wi00900668	ISS1:10F1	p30456_1	30/04/2012	p30456_1.cpl	NO
134	wi00862574	iss1:10f1	p30870_1	30/04/2012	p30870_1.cpl	NO
135	wi00894243	ISS1:10F1	p31087_1	30/04/2012	p31087_1.cpl	NO
136	wi00959820	ISS1:10F1	p31562_1	30/04/2012	p31562_1.cpl	NO
137	WI00889786	ISS1:10F1	p30750_1	30/04/2012	p30750_1.cpl	NO
138	wi00943748	ISS1:10F1	p31516_1	30/04/2012	p31516_1.cpl	NO
139	wi00959463	ISS1:10F1	p31528_1	30/04/2012	p31528_1.cpl	NO
140	WI00928455	ISS1:10F1	p31297_1	30/04/2012	p31297_1.cpl	NO
141	wi00896680	ISS1:10F1	p30357_1	30/04/2012	p30357_1.cpl	NO
142	wi00925141	ISS1:10F1	p30802_1	30/04/2012	p30802_1.cpl	NO
143	wi00968157	ISS1:10F1	p31637_1	30/04/2012	p31637_1.cpl	NO
144	wi00884699	ISS1:10F1	p31000_1	30/04/2012	p31000_1.cpl	YES
145	wi00932958	ISS1:10F1	p31115_1	30/04/2012	p31115_1.cpl	NO
146	wi00921295	ISS1:10F1	p31265_1	30/04/2012	p31265_1.cpl	NO
147	wi00906163	ISS1:10F1	p31205_1	30/04/2012	p31205_1.cpl	NO
148	wi00903437	ISS1:10F1	p31167_1	30/04/2012	p31167_1.cpl	NO
149	wi00960133	ISS2:10F1	p31557_2	30/04/2012	p31557_2.cpl	NO
150	wi00879322	ISS1:10F1	p30954_1	30/04/2012	p30954_1.cpl	NO
151	wi00896420	ISS1:10F1	p30867_1	30/04/2012	p30867_1.cpl	NO
152	wi00924886	ISS1:10F1	p31062_1	30/04/2012	p31062_1.cpl	YES
153	wi00877592	ISS1:10F1	p30880_1	30/04/2012	p30880_1.cpl	NO

154	wi00958682	ISS1:10F1	p31540_1	30/04/2012	p31540_1.cpl	NO
155	wi00882293	ISS1:10F1	p31010_1	30/04/2012	p31010_1.cpl	NO
156	wi00905297	ISS1:10F1	p31195_1	30/04/2012	p31195_1.cpl	NO
157	wi00968353	ISS1:10F1	p31412_1	30/04/2012	p31412_1.cpl	NO
158	wi00975133	ISS1:10F1	p31731_1	30/04/2012	p31731_1.cpl	NO
159	wi00897096	ISS1:10F1	p30676_1	30/04/2012	p30676_1.cpl	NO
160	wi00969890	ISS1:10F1	p31664_1	30/04/2012	p31664_1.cpl	YES
161	wi00967510	ISS1:10F1	p31147_1	30/04/2012	p31147_1.cpl	NO
162	wi00891621	ISS1:10F1	p31037_1	30/04/2012	p31037_1.cpl	NO
163	wi00968448	ISS1:10F1	p31648_1	30/04/2012	p31648_1.cpl	YES
164	wi00945997	ISS1:10F1	p31641_1	30/04/2012	p31641_1.cpl	NO
165	wi00967509	ISS1:10F1	p31294_1	30/04/2012	p31294_1.cpl	NO
166	wi00969208	ISS1:10F1	p31656_1	30/04/2012	p31656_1.cpl	NO
167	wi00976209	ISS1:10F1	p31717_1	30/04/2012	p31717_1.cpl	YES
168	wi00969039	ISS1:10F1	p31643_1	30/04/2012	p31643_1.cpl	NO
169	wi00973858	ISS1:10F1	p31691_1	30/04/2012	p31691_1.cpl	NO
170	wi00950575	ISS1:10F1	p31724_1	30/04/2012	p31724_1.cpl	NO
171	wi00975659	ISS1:10F1	p31707_1	30/04/2012	p31707_1.cpl	NO
172	wi00949410	ISS1:10F1	p31248_1	30/04/2012	p31248_1.cpl	NO
173	wi00967754	ISS1:10F1	p31653_1	30/04/2012	p31653_1.cpl	YES
174	wi00965285	ISS1:10F1	p31476_1	30/04/2012	p31476_1.cpl	NO
175	wi00978007	ISS1:10F1	p31737_1	30/04/2012	p31737_1.cpl	NO
176	wi00982243	ISS1:10F1	p31797_1	30/04/2012	p31797_1.cpl	NO
177	wi00960809	ISS1:10F1	p31564_1	30/04/2012	p31564_1.cpl	NO
178	wi00964006	ISS1:10F1	p31595_1	30/04/2012	p31595_1.cpl	YES
179	wi00965838	ISS1:10F1	p31623_1	30/04/2012	p31623_1.cpl	NO
180	wi00977393	ISS1:10F1	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)

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	
c ppmUtil	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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