



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.5, Avaya Aura® Session Manager R6.2, Avaya Session Border Controller for Enterprise R4.0.5 to support TDC Business Trunk - Issue 1.0

Abstract

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

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

1. Introduction

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

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Session Manager, Avaya SBCE and Communication Server 1000E. The enterprise site was configured to use the SIP Trunk to TDC Business Trunk. This configuration (shown in Figure 1) was used to exercise the features and functionality listed in Section 2.1.

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

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming PSTN calls were made to Unistim, SIP, Digital and Analog telephones and one-X® Communicator softphones at the enterprise
- Incoming calls to the enterprise site from the PSTN routed to the DDI numbers assigned by TDC
- Outgoing calls from the enterprise to the PSTN were made from Unistim, SIP, Digital and Analog telephones and one-X® Communicator softphones
- Outgoing calls from the enterprise site completed via TDC to PSTN destinations
- Calls using the G.711A, G.711MU and G.729A codecs supported by TDC (G.729A was never selected when G.711 was present in the SDP)
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using the T.38 mode
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls
- User features such as hold and resume, transfer, conference, call forwarding, etc
- Caller ID Presentation and Caller ID Restriction
- Call coverage and call forwarding for endpoints at the enterprise site

- Off-net call forwarding and mobility (extension to mobile)

2.2. Test Results

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

- No inbound toll free numbers were tested, however routing of inbound DID 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.
- SIP OPTIONS messages from the network contained a user in the URI which the Session Manager attempted to analyse. A “404 Not Found” message was returned.
- When an unassigned PSTN number was dialled, the network responded with a “500 Server Internal Error”. A more commonly used and informative response is “404 Not Found”.
- Codec Testing was limited as the network always selects G.711A/MU if available which is always the case on CS1000E.
- When testing blind call transfer to the PSTN, no ring-back was heard on the calling phone. Provisional reliable responses weren’t used on leg 2 of the call, in which case CS1000E does not send UPDATE messages. Without UPDATE, the backwards speech path is not established meaning the caller does not hear ring-back.
- One-X Communicator uses Payload Type 120 for DTMF and when this was sent in the re-INVITE when the call was put on hold, the network cleared the call. A script is required on the Avaya SBCE as a workaround.
- Calls to the mobile extension require two numbers in the To header, these are the Angöringsnummer (ANG) and the Calling Party Number. This could only be achieved with a script on the Avaya SBCE. For details of the mobile extension service, refer to the documentation for TDC Business Trunk

2.3. Support

For technical support on TDC products please contact the following website:

<http://www.tdc.se>

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the TDC Business Trunk Service. Located at the enterprise site are Session Manager, Avaya SBCE and 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, Analog 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.

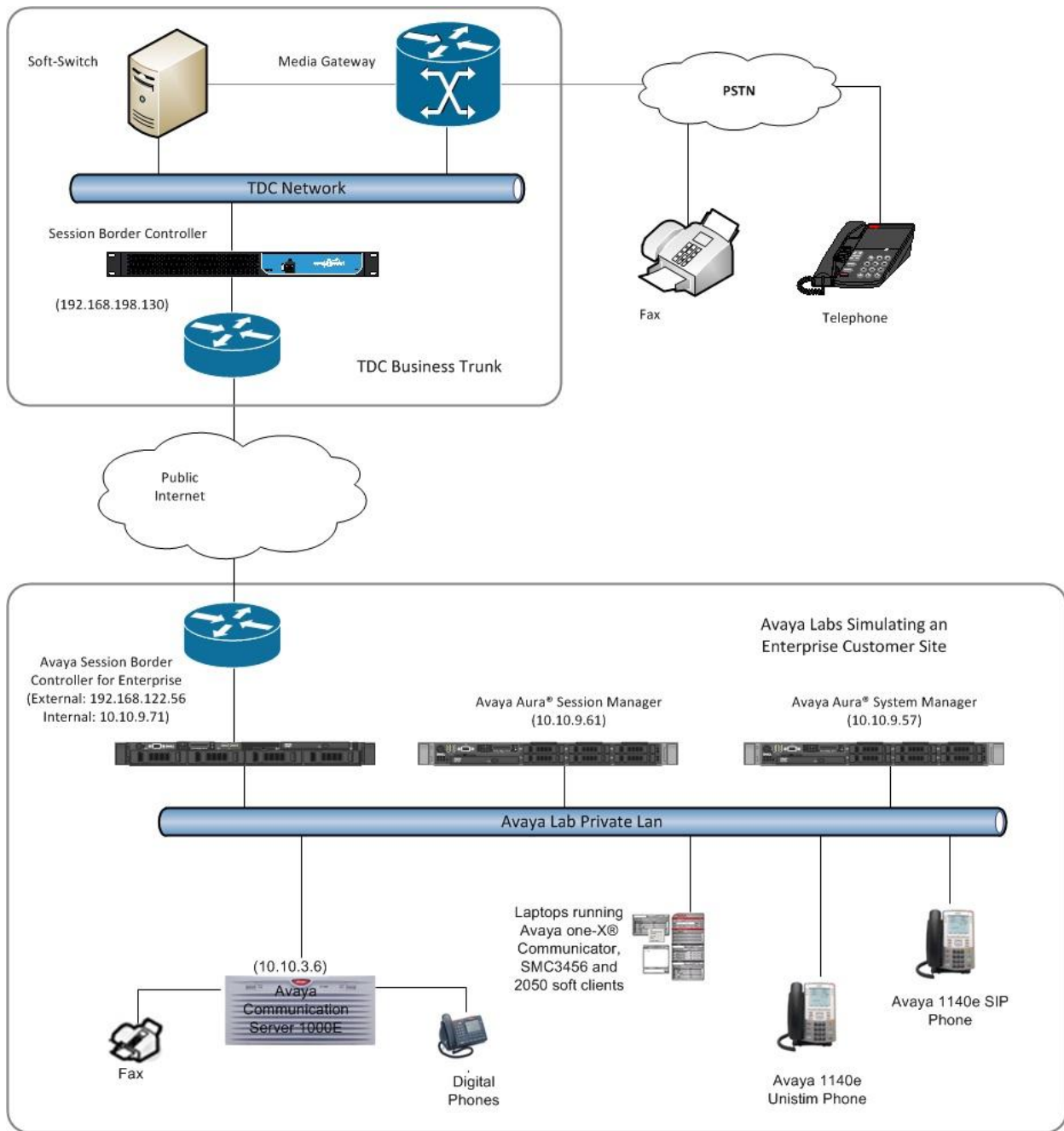


Figure 1: TDC SIP Trunk Topology

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura [®] Session Manager running on Avaya S8800 Server	R6.2 Build 6.2.0.0.620110
Avaya Aura [®] System Manager running on Avaya S8800 Server	R6.2 (System Platform 6.2.0.0.27, Template 6.2.12.0)
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	R7.5, Version 7.50.17 Service Update: 7.50_17Jan11 Deplst: X21 07.50Q
Avaya Session Border Controller for Enterprise on Dell R210 V2 server	Build: 4.0.5.Q09
Avaya Communication Server 1000E Media Gateway	CSP Version: MGCC CD01 MSP Version: MGCM AB01 APP Version: MGCA BA07 FPGA Version: MGCF AA18 BOOT Version: MGCB BA07 DSP1 Version: DSP1 AB03
Avaya 1140e and 1230 Unistim Telephones	FW: 0625C8A
Avaya 1140e and 1230 SIP Telephones	FW: 04.01.13.00.bin
Avaya SMC 3456	Version 2.6 build 53715
Avaya one-X [®] Communicator	Version cs6.1.0.10
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
TDC	
Acme Packet SD3820	6.1
Ericsson IMS	11B
Broadsoft Broadworks	R17
Cisco PGW2200	9.8

5. Configure Avaya Aura[®] Communication Manager 1000E

This section describes the steps required to configure Communication Server 1000E for SIP Trunking and also the necessary configuration for terminals (analog, SIP and IP phones). SIP trunks are established between Communication Server 1000E and Session Manager. These SIP trunks carry SIP Signalling associated with TDC Business Trunk. For incoming calls, the Session Manager receives SIP messages from the Avaya SBCE through which TDC's 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 signalling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the Avaya SBCE and on to TDC'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. Log in to the Avaya Communication Server 1000E

Log in using SSH to the ELAN IP address of the Call Server using a user with correct privileges. Once logged in type **csconsole**, this will take the user into the vxworks shell of the call server. Next type **logi**, 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 TDC'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      2000 LEFT 1970 USED 30
```

Load **overlay 21**, and confirm the Communication Server 1000E 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

TDC's SIP Trunk service supports G.711A, G.711MU and G.729A voice codecs and T.38 FAX transmissions. Use the Communication Server 1000E element manager to configure the Voice and Fax properties. Navigate to the **IP Network → IP Telephony Nodes → Node Details → VGW Gateway (VGW) and Codecs** (not shown) property page and configure the Communication Server 1000E General codec settings as in the next screenshot.

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

General | Voice Codecs | Fax

General

Echo cancellation: ☒ Use canceller, with tail delay: 128
☒ Dynamic attenuation

Voice activity detection threshold: -17 (-20 - +10 DBM)

Idle noise level: -65 (-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

Voice Codecs

Codec G711: ☒ Enabled (required)

Voice payload size: 20 (milliseconds per frame)

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

* Required Value.

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

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

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

General | **Voice Codecs** | Fax

Voice Codecs

Codec G711: ☒ Enabled (required)

Voice payload size: 20 (milliseconds per frame)

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

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

☐ Voice Activity Detection (VAD)

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: 20 (milliseconds per frame)

* Required Value.

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

Save Cancel

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

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

General | **Voice Codecs** | Fax

Voice Codecs

Codec G729: ☒ Enabled

Voice payload size: 20 (milliseconds per frame)

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

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

☒ Voice Activity Detection (VAD)

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

* Required Value.

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

Save Cancel

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

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

General | Voice Codecs | Fax

Codec G723.1: ☐ Enabled

Voice payload size: 30 (milliseconds per frame)

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

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

Coding rate: 5.3 (kbps)

Fax

Codec name: T.38 FAX

Maximum rate: 14400 (bps)

Fax TCF method: 2

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

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

Packet size: 30 (bps)

* 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 Communication Server 1000E, it is the Node IP that is used (please see **Section 6.5 – Define SIP Entities** for more details).

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

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

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

* Required Value. Save Cancel

Associated Signaling Servers & Cards

Select to add ▼ Add Remove Make Leader Print Refresh

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

Show: ☐ IPv6 address

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 Gate way (SIPGw)**, **H.323Gw**, and **SIPGw and H.323Gw**. **SIPGw** was used in the test configuration
- **SIP domain name:** The SIP domain name configured in this section must match the SIP domain name configured in the Session Manager **Section 6.2**, in this case **avaya.com**
- **Local SIP port:** The Local SIP Port is the port to which the gateway listens. The default value is **5060**

- **Gateway endpoint name:** This field cannot be left blank so a value is needed here. This field is used when a Network Routing Server is used for registration of the endpoint. In this network a Session Manager is used so any value can be put in here and will not be used
- **Application node ID:** This is a unique value that can be alphanumeric and is for the new Node that is being created, in this case **100**
- **Proxy or Redirect Server:** Primary TLAN IP address is the Security Module IP address of the Session Manager. The **Transport protocol** used for SIP, in this case is **TCP**
- **SIP URI Map:** **Public National** and **Private Unknown** are left blank. All other fields in the SIP URI Map are left with default values

Managing: 192.168.1.5 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 100 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Vtrk gateway application: ☒ Enable gateway service on this node

General

Vtrk gateway application: SIP Gateway (SIPGw) ▼

SIP domain name: avaya.com *

Local SIP port: 5060 * (1 - 65535)

Gateway endpoint name: cs1kv3 *

Gateway password: *

Application node ID: 100 * (0-9999)

Enable failsafe NRS: ☐

SIP ANAT: ☒ IPv4

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)

Information will be captured for the IP addresses listed below.

Monitor IP: Add

Monitor addresses: Remove

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 10.10.9.61

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, not shared with other resources and best practice dictates that IP telephones and Media Gateways are all placed in separate zones. Use Element Manager to define bandwidth zones as in the following highlighted example. Use Element Manager and navigate to **System → IP Network → Zones → Bandwidth Zones** and add new zones as required.

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

Bandwidth Zones

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

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

5.6. Configure SIP Trunks

Communication Server 1000E virtual trunks will be used for all inbound and outbound PSTN calls to TDC's SIP Trunk Service. Six 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 Digit Manipulation Data Block (**DGT**); configure using the Communication Server 1000E system terminal and overlay 86
- Configure a Route List Block (**RLB**); configure using the Communication Server 1000E system terminal and overlay 86
- Configure Co-ordinated Dialling Plan(s) (**CDP**); configure using the Communication Server 1000E system terminal and overlay 87

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 1**

CTYP DCIP

DES VIR TRK

USR ISLD

ISLM 4000

SSRC 3700

OTBF 32

NASA YES

IFC SL1

CNEG 1

RLS ID 4

RCAP ND2

MBGA NO

H323

 OVLR NO

 OVLS NO

Next, configure the SIP trunk Route Data Block (**RDB**) using the 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 **VTRK**. The remaining highlighted values are important for correct SIP trunk operation.

Overlay 16 TYPE: RDB CUST 00 ROUT 1 TYPE RDB CUST 00 ROUT 1 DES VIR_TRK TKTP TIE NPID_TBL_NUM 0 ESN NO RPA NO CNVT NO SAT NO RCLS EXT VTRK YES ZONE 00001 PCID SIP CRID NO NODE 100 DTRK NO ISDN YES MODE ISLD DCH 1 IFC SL1 PNI 00000 NCNA YES NCRD YES TRO NO FALT NO CTYP UKWN INAC NO ISAR NO DAPC NO MBXR NO MBXOT NPA MBXT 0 PTYP ATT CNDP UKWN AUTO NO DNIS NO DCDR NO ICOG IAO SRCH LIN TRMB YES STEP	ACOD 1111 TCPP NO PII NO AUXP NO TARG CLEN 1 BILN NO OABS INST IDC YES DCNO 0 NDNO 0 * DEXT NO DNAM NO SIGO STD STYP SDAT MFC NO ICIS YES OGIS YES TIMR ICF 1920 OGF 1920 EOD 13952 LCT 256 DSI 34944 NRD 10112 DDL 70 ODT 4096 RGV 640 GTO 896 GTI 896 SFB 3 PRPS 800 NBS 2048 NBL 4096 IENB 5 TFD 0 VSS 0 VGD 6 EESD 1024 SST 5 0 DTD NO SCDT NO 2 DT NO NEDC ORG FEDC ORG	CPDC NO DLTN NO HOLD 02 02 40 SEIZ 02 02 SVFL 02 02 DRNG NO CDR NO NATL YES SSL CFWR NO IDOP NO VRAT NO MUS YES MRT 21 PANS YES RACD NO MANO NO FRL 0 0 FRL 1 0 FRL 2 0 FRL 3 0 FRL 4 0 FRL 5 0 FRL 6 0 FRL 7 0 OHQ NO OHQT 00 CBQ NO AUTH NO TTBL 0 ATAN NO OHTD NO PLEV 2 OPR NO ALRM NO ART 0 PECL NO DCTI 0 TIDY 1600 100 ATRR NO TRRL NO SGRP 0 ARDN NO CTBL 0 AACR NO
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Next, configure virtual trunk members using the 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
TN 100 0 0 0
DATE
PAGE
DES VIR_TRK
TN 100 0 00 00 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 00001
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK ANLG
NCOS 0
RTMB 1 1
CHID 1
TGAR 1
STRI/STRO IMM IMM
SUPN YES
AST NO
IAPG 0
CLS UNR DIP CND ECD WTA LPR APN THFD XREP SPCD MSBT
P10 NTC
TKID
AACR NO
```

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

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


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

Overlay 86 CUST 0 FEAT rlb RLI 10 ELC NO ENTR 0 LTER NO ROUT 1 TOD 0 ON 1 ON 2 ON 3 ON 4 ON 5 ON 6 ON 7 ON VNS NO SCNV NO CNV NO EXP NO FRL 0 DMI 10 CTBL 0 ISDM 0	FCI 0 FSNI 0 BNE NO DORG NO SBOC NRR PROU 1 IDBB DBD IOHQ NO OHQ NO CBQ NO ISET 0 NALT 5 MFRL 0 OVLL 0
--	---

Next, configure Co-ordinated Dialling Plan(s) (CDP) which users will dial to reach PSTN numbers. Use the Communication Server 1000E system terminal and **Overlay 87**. The following are some example CDP entries used. The highlighted **RLI** value previously configured in overlay 86 is used as the Route List Index (**RLI**); this is the default PSTN route to the SIP Trunk service.

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

5.7. Configure Analog, Digital and IP Telephones

A variety of telephone types were used during the testing, the following is the configuration for the Avaya 1140e Unistim IP telephone. Load **Overlay 20** at the system terminal and enter the following values. 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 5.5** for **MO**

Overlay 20 IP Telephone configuration

```
DES 1140
TN 100 0 01 0 VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00002
CUR_ZONE 00002
ERL 0
ECL 0
FDN 0
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 1
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED 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
FDSD NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA PKCH MUTA MWTD
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
```

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DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 MCR 5000 0 MARP
 CPND
 CPND_LANG ROMAN
 NAME IP1140
 XPLN 10
 DISPLAY FMT FIRST, LAST
01 MCR 5000 0
 CPND
 CPND_LANG ROMAN
 NAME IP1140
 XPLN 10
 DISPLAY FMT FIRST, LAST
02
03 BSY
04 DSP
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN

Digital telephones are configured using the **Overlay 20**; the following is a sample **3904** digital set configuration. Again, a unique number is entered for the **KEY00** and **KEY01** value.

Overlay 20 - Digital Set configuration

```
TYPE: 3904
DES 3904
TN 04 0 02 00 VIRTUAL
TYPE 3904
CDEN 8D
CTYP XDLC
CUST 0
MRT
ERL 0
FDN 0
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 1
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBD WTA LPR PUA MTD FND HTD TDD HFA GRLD CRPA STSD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDA CDMA LLCN MCTD CLBD AUTU
GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD
CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXRO
USMD USRD ULAD CCBF 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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MLNG ENG

DNDR 0

KEY 00 MCR 5008 0 MARP

CPND

CPND LANG ROMAN

NAME Digital Set

XPLN 10

DISPLAY_FMT FIRST, LAST

01 MCR 5008 0

CPND

CPND LANG ROMAN

NAME Digital Set

XPLN 10

DISPLAY_FMT FIRST, LAST

02

03

04

05

06

07

08

09

10

11

12

13

14

15

16

17 TRN

18 AO6

19 CFW 16

20 RGA

21 PRK

22 RNP

23

24 PRS

25 CHG

26 CPN

27 CLT

28 RLT

29

30

31

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

Overlay 20 – Analog Telephone Configuration

```
DES 500
TN 04 0 03 00
TYPE 500
CDEN 4D
CUST 0
MRT
ERL 00000
WRLS NO
DN 5015
AST NO
IAPG 0
HUNT
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
XLST
SCI 0
SCPW
SFLT NO
CAC MFC 0
CLS UNR DTN FBD XFD WTA THFD FND HTD ONS
    LPR XRD AGRD CWD SWD MWD RMMD SMWD LPD XHD SLKD CCSD LND TVD
    CFTD SFD MRD C6D CNID CLBD AUTU
    ICDD CDMD LLCN EHTD MCTD
    GPUD DPUD CFXD ARHD OVDD AGTD CLTD LDTD ASCD SDND
    MBXD CPFA CPTA UDI RCC HBTD IRGD DDGA NAMA MIND
    NRWD NRCD NROD SPKD CRD PRSD MCRD
    EXR0 SHL SMSD ABDD CFHD DNDY DNO3
    CWND USMD USRD CCBD BNRD OCBD RTDD RBDD 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
```

5.8. 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 11
NMME NO
```

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

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

AVAYA CS1000 Element Manager

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

Node ID: 100 - SIP Line Configuration Details

General | SIP Line Gateway Settings | SIP Line Gateway Service

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

General

SIP domain name: *

SLG endpoint name:

SLG Group ID:

SLG Local Sip port: (1 - 65535)

SLG Local Tls port: (1 - 65535)

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)
Information will be captured for the IP addresses listed below.

Monitor IP:

Monitor addresses:

5.9. 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 **VTRK** in **Section 5.5**. A unique telephone number is entered for value **KEY 00**. The value for **KEY 01** is comprised of the **UAPR** value and the telephone number used in **KEY 00**.

Overlay 20 – SIP Telephone Configuration

```
DES  SIPD
TN    100 0 01 10  VIRTUAL
TYPE  UEXT
CDEN  8D
CTYP  XDLC
CUST  0
UXTY  SIPL
MCCL  YES
SIPN  1
SIP3  0
FMCL  0
TLSV  0
SIPU  5003
NDID  100
SUPR  NO
SUBR  DFLT MWI RGA CWI MSB
UXID
NUID  100
NHTN  100 0 01 10
CFG_ZONE 00002
CUR_ZONE 00002
ERL   0
ECL   0
VSIT  NO
FDN
TGAR  0
LDN   NO
NCOS  0
SGRP  0
RNPG  0
SCI   0
SSU
XLST
SCPW  1234
SFLT  NO
CAC_MFC 0
CLS   UNR FBD WTA LPR MTD FNA HTA TDD HFD CRPD
      MWD LMPN RMDM 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
```

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```

    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 SCR 5003 0      MARP
    CPND
    CPND_LANG ROMAN
    NAME Sigma 1140
    XPLN 11
    DISPLAY_FMT FIRST, LAST*
01 HOT U 115003 MARP 0
02
03
04
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23 *
24 PRS
25 CHG
26 CPN
27
28
29
30
31
```

5.10. Save Configuration

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

The screenshot shows the AVAYA CS1000 Element Manager web interface. On the left is a navigation tree with categories like 'Tools', '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 includes a breadcrumb trail: 'Tools » Backup and Restore » Call Server Backup and Restore » Call Server Backup'. Below the title, there is an 'Action' dropdown menu set to 'Backup', and 'Submit' and 'Cancel' buttons. At the top right, it shows 'Managing: 192.168.1.5' and 'Username: admin'.

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. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer Locations
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns

6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by entering **http://<FQDN>/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the **Home** tab will be presented with menu options shown below.

AVAYA Avaya Aura® System Manager 6.2 Last Logged on at January 17, 2013 2:08 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Users	Elements	Services
Administrators Manage Administrative Users	B5800 Branch Gateway Manage B5800 Branch Gateway 6.2 elements	Backup and Restore Backup and restore System Manager database
Directory Synchronization Synchronize users with the enterprise directory	Communication Manager Manage Communication Manager 5.2 and higher elements	Bulk Import and Export Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
Groups & Roles Manage groups, roles and assign roles to users	Conferencing Manage Conferencing Multimedia Server objects	Configurations Manage system wide configurations
UCM Roles Manage UCM Roles, assign roles to users	Inventory Manage, discover, and navigate to elements, update element software	Events Manage alarms, view and harvest logs
User Management Manage users, shared user resources and provision users	Meeting Exchange Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements	Licenses View and configure licenses
	Messaging Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging	Replication Track data replication nodes, repair replication nodes
	Presence Presence	Scheduler Schedule, track, cancel, update and delete jobs
	Routing Network Routing Policy	Security Manage Security Certificates
	Session Manager	Templates Manage Templates for Communication Manager, Messaging System and B5800

6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name agreed with TDC; this will be the same as specified in the Authoritative Domain specified for the CS1000E SIP Gateway. Refer to **Section 5.4** for details. In test, **avaya.com** was used. Optionally, a description for the domain can be entered in the Notes field. Click **Commit** to save changes.

The screenshot shows the Avaya Aura System Manager 6.2 web interface. The top header includes the Avaya logo, the system name "Avaya Aura® System Manager 6.2", and a user status bar indicating "Last Logged on at January 17, 2013 2:08 PM" with links for "Help", "About", "Change Password", and "Log off admin". The left-hand navigation menu is expanded to show the "Routing" section, with "Domains" selected. The main content area is titled "Domain Management" and includes a "Home / Elements / Routing / Domains" breadcrumb trail. Below the title, there are buttons for "Edit", "New", "Duplicate", "Delete", and "More Actions". A table displays the domain entries, with one item shown: "avaya.com" of type "sip". The table has columns for "Name", "Type", "Default", and "Notes". A "Filter: Enable" link is present in the top right of the table area. The bottom of the table shows a "Select : All, None" option.

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

6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General**, in the **Name** field, enter an informative name for the location. Scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the test enterprise.

Home / Elements / Routing / Locations

Location Details

Help ?

Commit

Cancel

General

* Name:

Galway

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units:

Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth:

☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location):

1000

Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location):

1000

Kbit/Sec

* Minimum Multimedia Bandwidth:

64

Kbit/Sec

* Default Audio Bandwidth:

80

Kbit/sec

Alarm Threshold

Overall Alarm Threshold:

80

%

Multimedia Alarm Threshold:

80

%

* Latency before Overall Alarm Trigger:

5

Minutes

* Latency before Multimedia Alarm Trigger:

5

Minutes

Location Pattern

Add

Remove

2 Items

Refresh

Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.10.9.*	
<input type="checkbox"/>	* 10.10.3.*	

6.4. Administer Adaptations

Adaptations can be used to modify the called and calling party numbers to meet the requirements of the service. The called party number present in the SIP INVITE Request URI is modified by the **Digit Conversion** in the Adaptation. Additionally, the called and calling party numbers can be modified using **Digit Conversion** when **fromto=true** is entered in the **Module Parameters**.

The example shown uses **Digit Conversion for Incoming Calls to SM** to convert the calling number from the 4 digit extension to E.164; this applies to calls from the CS1000E to the Session Manager. It also uses **Digit Conversion for Outgoing Calls from SM** to convert the called number from E.164 to the 4 digit extension; this applies to calls from the Session Manager to the CS1000E. The module **CS1000Adaptor** is used, significant digits of the test DDI range have been obscured.

Home / Elements / Routing / Adaptations

Adaptation Details Help ?

Commit Cancel

General

* Adaptation name: TDC
 Module name: CS1000Adaptor
 Module parameter: fromto=true
 Egress URI Parameters:
 Notes:

Digit Conversion for Incoming Calls to SM

Add Remove

9 Items Refresh Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	* 5000	* 4	* 4		* 4	+46851nnnnn5	origination		
<input type="checkbox"/>	* 5001	* 4	* 4		* 4	+46851nnnnn6	origination		
<input type="checkbox"/>	* 5003	* 4	* 4		* 4	+46851nnnnn7	origination		
<input type="checkbox"/>	* 5004	* 4	* 4		* 4	+46851nnnnn8	origination		
<input type="checkbox"/>	* 5005	* 4	* 4		* 4	+46851nnnnn3	origination		
<input type="checkbox"/>	* 5006	* 4	* 4		* 4	+46851nnnnn6	origination		
<input type="checkbox"/>	* 5015	* 4	* 4		* 4	+46851nnnnn8	origination		
<input type="checkbox"/>	* 5500	* 4	* 4		* 4	+46851nnnnn8	origination		

Select : All, None

Digit Conversion for Outgoing Calls from SM

Add Remove

4 Items Refresh Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	* +46851nnnnn5	* 12	* 12		* 12	5000	destination		
<input type="checkbox"/>	* +46851nnnnn6	* 12	* 12		* 12	5001	destination		
<input type="checkbox"/>	* +46851nnnnn7	* 12	* 12		* 12	5003	destination		
<input type="checkbox"/>	* +46851nnnnn8	* 12	* 12		* 12	5004	destination		

The next example shown uses “**MIME=no**” to strip MIME message bodies on egress from Session Manager to the Avaya SBCE.

Home / Elements / Routing / Adaptations

Adaptation Details Help ? Commit Cancel

General

* Adaptation name:

Module name:

Module parameter:

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

Add Remove

0 Items Refresh Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes

Digit Conversion for Outgoing Calls from SM

Add Remove

0 Items Refresh Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes

6.5. Administer 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 CS1000E SIP entity and **SIP Trunk** 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 CS1000E SIP Entity
- Avaya Session Border Controller for Enterprise (Avaya SBCE) SIP Entity

6.5.1. 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.

Home / Elements / Routing / SIP Entities

SIP Entity Details Help ?

Commit Cancel

General

* Name: Session Manager

* FQDN or IP Address: 10.10.9.61

Type: Session Manager

Notes:

Location: Galway

Outbound Proxy:

Time Zone: Europe/Dublin

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

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 the domain added in **Section 6.2** as the default domain

Port

TCP Failover port:

TLS Failover port:

Add Remove

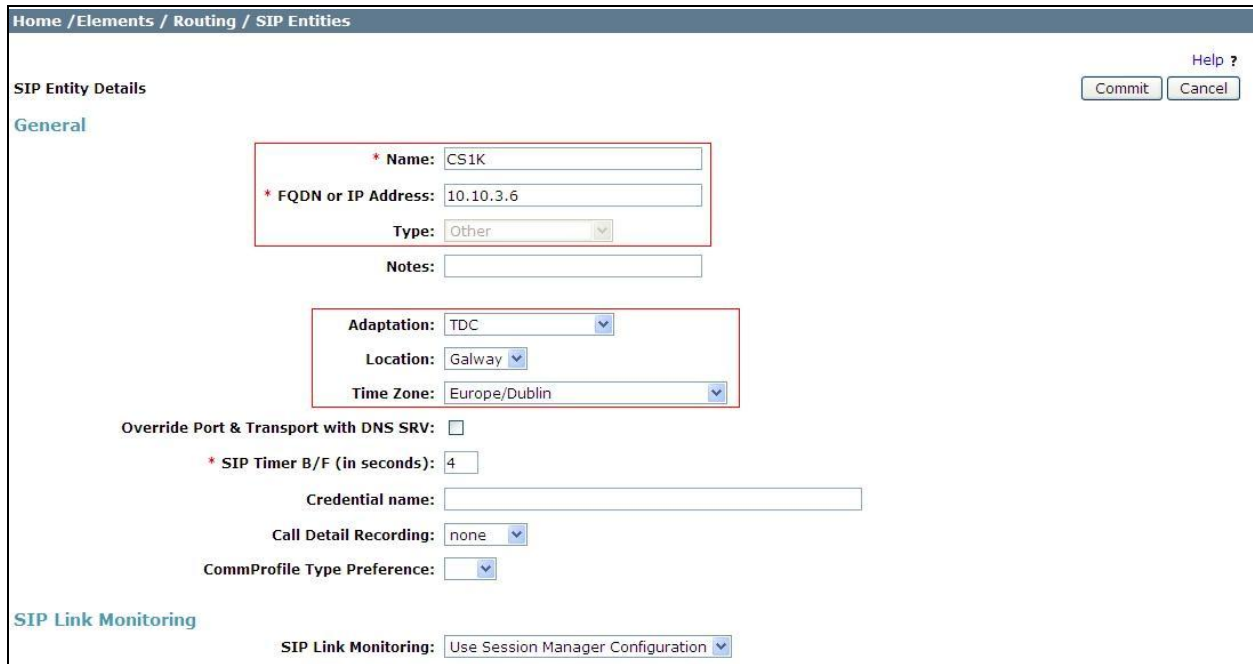
3 Items Refresh Filter: Enable

<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

6.5.2. Avaya Communication Server 1000E SIP Entity

The following screen shows the SIP entity for CS1000E. The FQDN or IP Address field is set to the TLAN Node IP address defined in **Section 5.4**. Set the **Adaptation** to the appropriate Adaptation defined in **Section 6.4** for traffic to and from the CS1000E and set the location to that defined in **Section 6.3**.



Home / Elements / Routing / SIP Entities

SIP Entity Details Help ?

Commit Cancel

General

* Name: CS1K

* FQDN or IP Address: 10.10.3.6

Type: Other

Notes:

Adaptation: TDC

Location: Galway

Time Zone: Europe/Dublin

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

CommProfile Type Preference:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

6.5.3. Avaya Aura® Session Border Controller SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface (see **Figure 1**). Set the **Adaptation** to the appropriate Adaptation defined in **Section 6.4** for outbound traffic and set the location defined for use with Avaya SBCE.

Home / Elements / Routing / SIP Entities

SIP Entity Details Help ?

Commit Cancel

General

* Name: ASBCE

* FQDN or IP Address: 10.10.9.71

Type: SIP Trunk

Notes:

Adaptation: No_MIME

Location: Galway

Time Zone: Europe/Dublin

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

6.6. Administer 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 (not shown). 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**
- 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 6.5**
- 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

Click **Commit** to save changes. The following screen shows the Entity Links used in this configuration.

Home / Elements / Routing / Entity Links Help ?

Entity Links

4 Items Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
<input type="checkbox"/>	ASBCE Link	Session Manager	TCP	5060	ASBCE	5060	Trusted	
<input type="checkbox"/>	CS1K Link	Session Manager	TCP	5060	CS1K	5060	Trusted	
<input type="checkbox"/>	Msg Link	Session Manager	TCP	5060	Messaging	5060	Trusted	
<input type="checkbox"/>	Session Manager Communication Manager 5061 TLS	Session Manager	TCP	5060	Communication Manager	5060	Trusted	

6.7. Administer Routing Policies

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 CS1000E.

Home / Elements / Routing / Routing Policies

Routing Policy Details

CommitCancelHelp ?

General

* Name: CS1K

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CS1K	10.10.3.6	Other	

Time of Day

AddRemoveView Gaps/Overlaps

1 Item RefreshFilter: Enable

	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

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

Home / Elements / Routing / Routing Policies

Routing Policy Details Help ?

Commit Cancel

General

* Name:

Disabled: ☐

* Retries:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
ASBCE	10.10.9.71	SIP Trunk	

Time of Day

Add Remove View Gaps/Overlaps

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	Ranking	1 ▲	Name	2 ▲	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0		24/7		<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

6.8. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. 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 dialled 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**

Configuration is continued on the next page.

Under **Originating Locations and Routing Policies**, click **Add**. In the resulting screen (not shown), under **Originating Location** select the location defined in **Section 6.3** or **ALL** and under **Routing Policies** select one of the routing policies defined in **Section 6.7**. 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 TDC Business Trunk service.

Home /Elements / Routing / Dial Patterns

Dial Pattern Details Help ?
Commit Cancel

General

* Pattern: 00353
 * Min: 13
 * Max: 14

Emergency Call: ☐
 Emergency Priority: 1
 Emergency Type:
 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"/>	Galway		External	0	<input type="checkbox"/>	ASBCE	

The following screen shows the test dial pattern configured for CS1000E.

Home /Elements / Routing / Dial Patterns

Dial Pattern Details Help ?
Commit Cancel

General

* Pattern: +46851nnnn
 * Min: 11
 * Max: 12

Emergency Call: ☐
 Emergency Priority: 1
 Emergency Type:
 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"/>	Galway		CS1K	0	<input type="checkbox"/>	CS1K	

Note: The pattern to be matched has been obscured.

7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Session Border Controller for Enterprise. At the time of writing the Avaya Session Border Controller for Enterprise was badged as the Sipera E-SBC (Enterprise Session Border Controller) developed for Unified Communications Security (UC-Sec). The Avaya Session Border Controller for Enterprise is administered using the UC-Sec Control Center.

7.1. Access Avaya Session Border Controller for Enterprise

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



Log in with the appropriate credentials.

The UC-Sec™ family of products from Sipera Systems delivers comprehensive VoIP security by adapting the best practices of internet security and by using unique, sophisticated techniques such as VoIP protocol misuse & anomaly detection, behavioral learning based anomaly detection and voice spam detection to protect VoIP networks.

[Visit the Sipera Systems website to learn more.](#)

NOTICE TO USERS: This system is for authorized use only. Unauthorized use of this system is strictly prohibited. Unauthorized or improper use of this system may result in civil and/or criminal penalties. Use of this system constitutes consent to security monitoring. All activity is logged with login info, host name and IP address.

The following screenshot shows the opening screen. Navigation of the GUI is done in the **UC-Sec Control Center** menu on the left hand side.



So that screenshots can be focused on the areas of the GUI where configuration takes place, the **UC-Sec Control Center** menu is not shown in subsequent screenshots

7.2. Define Network Information

Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one interface assigned.

To define the network information, navigate to **Device Specific Settings → Network Management** in the **UC-Sec Control Center** menu on the left hand side and click on **Add IP**. Enter details in the blank box that appears at the end of the list

- Define the internal IP address with screening mask and assign to interface **A1**
- Select **Save Changes** to save the information
- Click on **Add IP**
- Define the external IP address with screening mask and assign to interface **B1**
- Select **Save Changes** to save the information
- Click on **System Management** in the main menu
- Select **Restart Application** indicated by an icon in the status bar (not shown)

Device Specific Settings > Network Management: GSSCP_V9

UC-Sec Devices
GSSCP_V9

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
10.10.9.71		10.10.9.1	A1
192.168.122.56		192.168.122.7	B1

Select the **Interface Configuration** tab and click on **Toggle State** to enable the interfaces.

Device Specific Settings > Network Management: GSSCP_V9

UC-Sec Devices
GSSCP_V9

Network Configuration | Interface Configuration

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

7.3. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces.

7.3.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings** → **Signaling Interface** in the **UC-Sec Control Center** menu on the left hand side. Details of transport protocol and ports for the internal and external SIP signalling are entered here

- Select **Add Signaling Interface** and enter details in the pop-up menu (not shown)
- In the **Name** field enter a descriptive name for the internal signalling interface
- For **Signaling IP**, select an **internal** signalling interface IP address defined in **Section 7.2**
- Select **UDP** and **TCP** port numbers, **5060** is used for the Session Manager
- Select **Add Signaling Interface** and enter details in the pop-up menu (not shown)
- In the **Name** field enter a descriptive name for the external signalling interface
- For **Signaling IP**, select an **external** signalling interface IP address defined in **Section 7.2**
- Select **UDP** and **TCP** port numbers, **5060** is used for TDC

Device Specific Settings > Signaling Interface: GSSCP_V9

UC-Sec Devices

GSSCP_V9

Signaling Interface

Add Signaling Interface

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile		
Int_Sig	10.10.9.71	5060	5060	---	None		
Ext_Sig	192.168.122.56	5060	5060	---	None		

7.3.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings** → **Media Interface** in the **UC-Sec Control Center** menu on the left hand side. Details of the RTP and SRTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

- Select **Add Media Interface** and enter details in the pop-up menu
- In the **Name** field enter a descriptive name for the internal media interface
- For **Media IP**, select an **internal** media interface IP address defined in **Section 7.2**
- Select **RTP port** ranges for the media path with the enterprise end-points
- Select **Add Media Interface** and enter details in the pop-up menu
- In the **Name** field enter a descriptive name for the external media interface
- For **Media IP**, select an **external** media interface IP address defined in **Section 7.2**
- Select **RTP port** ranges for the media path with the TDC network

Device Specific Settings > Media Interface: GSSCP_V9

UC-Sec Devices

GSSCP_V9

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_Med	10.10.9.71	2048 - 3329		
Ext-Med	192.168.122.56	50000 - 60000		

Note: During test the port ranges for the external media interface were left at the default values

7.4. Define Server Interworking

Server interworking is defined for each server connected to the Avaya SBCE. In this case, the TDC SBC is connected as the Trunk Server and the Session Manager is connected as the Call Server. Configuration of interworking includes Hold support, T.38 fax support and SIP extensions. Also included in this configuration is Request-URI header manipulation on the trunk Server to remove the international dialling prefix of “00” and insert a “+”. Although this can be achieved using digit manipulation on the Session manager, it is included here for information.

To define server interworking on the Avaya SBCE, navigate to **Global Profiles → Server Interworking** in the **UC-Sec Control Center** menu on the left hand side. To define Server Interworking for the Session Manager, highlight the **avaya-ru** profile which is a factory setting appropriate for Avaya equipment and select **Clone Profile**. A pop-up menu is generated headed **Clone Profile** (not shown)

- In the **Clone Name** field enter a descriptive name for the Session Manager and click **Finish** – in test **ASM9** was used
- In the **General** tab (not shown) Select **Edit** and enter details in the pop-up menu.
- Check the **T.38** box
- Change the **Hold Support** RFC to **RFC2543** then click **Next** and **Finish**

General	
	<input type="radio"/> None
Hold Support	<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

- In the **Advanced** tab (not shown) Select **Edit** and enter details in the pop-up menu.
- Uncheck the **AVAYA Extensions** box

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

Finish

To define Server Interworking for the TDC SBC, highlight the previously defined profile for the Session Manager and select **Clone Profile**. A pop-up menu is generated headed **Clone Profile** (not shown)

- In the **Clone Name** field enter a descriptive name for server interworking profile for the TDC SBC and click **Finish** – in test **TDC** was used
- Select **Edit** and enter details in the pop-up menu
- Check the **T.38** box
- Select **Next** three times and **Finish**

The Trunk Server interworking includes the Request-URI header manipulation

- In the **URI Manipulation** tab (not shown) Select **Add Regex** and enter details in the pop-up menu.
- Enter **00.*** in the **User Regex** box (the “.” denotes any character and the “*” allows any subsequent characters).
- Select **Replace [Value 1] with [Value 2]** in the **User Action** drop down menu and enter **00** as Value 1 and **+** as Value 2 in the **User Values** boxes
- Select **Finish**

Add Regex

URI Manipulation

Invalid or incorrectly entered regular expressions may cause unexpected results.
Ex: [0-9]{3,5}\\\\user, (simple|advanced)\\\\-user[A-Z]{3}

When a URI [user@domain] matches the following:

User Regex: 00.* (Blank is wildcard)

Domain Regex: (Blank is wildcard)

Do this with the user section:

User Action: Replace [Value 1] with [Value 2]

User Values: 00 +

Do this with the domain section:

Domain Action: None

Domain Values:

Finish

The resultant URI manipulation appears under the **URI manipulation** tab as follows:

Global Profiles > Server Interworking: TDC

Interworking Profiles

cs2100

avaya-ru

OCS-Edge-Server

cisco-ccm

cups

Click here to add a description.

General Timers URI Manipulation Header Manipulation Advanced

Add Regex

User Regex	Domain Regex	User Action	Domain Action
00.*		Replace 00 with +	None

7.5. Define Signalling Manipulation

Signalling manipulation is required in some cases to ensure effective interworking. During test, some issues were found in the interworking between the TDC Business Trunk service and the enterprise. Two of these issues could not be resolved by other methods such as **Server Interworking** and **Signaling Rules**. The first issue is that re-INVITEs from One-X Communicator, e.g. for call hold, included Payload Type 120 for DTMF. These re-INVITEs were resulting in the call being cleared by the network.

The second issue is that calls to the mobile extension for the Mobile Extension (MEX) service require two numbers in the To header, these are the Angöringsnummer (ANG) and the Calling Party Number. This could only be achieved with a script on the Avaya SBCE.

To define the signalling manipulation to change the Payload Type for DTMF in the re-INVITE sent for call hold, navigate to **Global Profiles → Signaling Manipulation** in the **UC-Sec Control Center** menu on the left hand side. Click on **Add Script** and enter a title and the script in the script editor (not shown). The title use in test was **Video_Removal**. The script text is as follows:

```
within session "ALL"
{
  act on message where %DIRECTION="INBOUND" and %ENTRY_POINT="AFTER_NETWORK" and
  %METHOD="INVITE"
  {
    if(exists(%SDP[1]["s"]["m"][2].ATTRIBUTES["video"][1]))then
    {
      %BODY[1].regex_replace("b=TIAS:13952000","");
      %SDP[1]["s"]["m"][1].FORMATS[4]="101";
      %SDP[1]["s"]["m"][1].ATTRIBUTES["rtpmap"][4]="101 telephone-event/8000/1";
      remove(%SDP[1]["s"]["m"][2]);
    }
  }
}
```

Note: This script also removes video attributes present in the SDP for call hold, hence the name.

To define the signalling manipulation to reformat the To header in line with the requirements of the MEX service, navigate to **Global Profiles → Signaling Manipulation** in the **UC-Sec Control Center** menu on the left hand side. Click on **Add Script** and enter a title and the script in the script editor. The title used in test was **MEX_Mobile_From**. The script text is as follows:

```
within session "ALL"
{
  act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and
  %METHOD="INVITE"
  {
    if(%HEADERS["To"][1].URI.USER.regex_match("^46394980"))then
    {
      %HEADERS["From"][1].URI.USER.regex_replace("^\\+", "00");
      %FromUser = %HEADERS["From"][1].URI.USER;
      %HEADERS["From"][1].URI.USER = "+46752468911";
      append(%HEADERS["From"][1].URI.USER, %FromUser);
    }
  }
}
```

Note: The above script prefixes the ANG to the calling party number in the From header, and also reformats the calling party number to insert the international dialling prefix. This avoids a “+” appearing in the middle of the number.

7.6. Define Servers

Servers are defined for each server connected to the Avaya SBCE. In this case, the TDC SBC is connected as the Trunk Server and the Session Manager is connected as the Call Server. To define the Session Manager, navigate to **Global Profiles → Server Configuration** in the **UC-Sec Control Center** menu on the left hand side. Click on **Add Profile** (not shown) and enter details in the pop-up menu

- In the **Profile Name** field enter a descriptive name for the Session Manager and click **Next** (not shown)
- In the **Server Type** drop down menu, select **Call Server**
- In the **IP Addresses / Supported FQDNs** box, type the Session Manager SIP interface address which is the same as that defined on Session Manager in **Section 6.5**
- Check **TCP** in **Supported Transports**
- Define the **TCP** port for SIP signalling, **5060** is used for the Session Manager
- Click **Next** three times then select the **Interworking Profile** for the Session Manager defined in **Section 7.4** from the drop down menu
- Select the **Video_Removal** Signaling Manipulation Script defined in **Section 7.5** from the drop down menu and click **Finish**

Edit Server Configuration Profile - General		Edit Server Configuration Profile - Advanced	
Server Type	Call Server	Enable DoS Protection	<input type="checkbox"/>
IP Addresses / Supported FQDNs <small>Comma separated list</small>	10.10.9.61	Enable Grooming	<input type="checkbox"/>
Supported Transports	<input checked="" type="checkbox"/> TCP <input type="checkbox"/> UDP <input type="checkbox"/> TLS	Interworking Profile	ASM9
TCP Port	5060	Signaling Manipulation Script	Video_Removal
UDP Port		TCP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
TLS Port			
Finish		Finish	

To define the TDC SBC as a Trunk Server, navigate to **Global Profiles → Server Configuration** in the **UC-Sec Control Center** menu on the left hand side. Click on **Add Profile** (not shown) and enter details in the pop-up menu

- In the **Profile Name** field enter a descriptive name for the TDC SBC and click **Next** (not shown)
- In the **Server Type** drop down menu, select **Trunk Server**
- In the **IP Addresses / Supported FQDNs** box, type the IP address of the TDC SBC
- Check **UDP** in **Supported Transports**
- Define the **UDP** port for SIP signaling, **5060** is used for TDC
- Click **Next** three times then select the **Interworking Profile** for the TDC SBC defined in **Section 7.4** from the drop down menu
- Select the **MEX_Mobile_From** Signaling Manipulation Script defined in **Section 7.5** from the drop down menu and click **Finish**

Edit Server Configuration Profile - General		Edit Server Configuration Profile - Advanced	
Server Type	Trunk Server	Enable DoS Protection	<input type="checkbox"/>
IP Addresses / Supported FQDNs <small>Comma seperated list</small>	192.168.198.130	Enable Grooming	<input type="checkbox"/>
Supported Transports	<input checked="" type="checkbox"/> TCP <input checked="" type="checkbox"/> UDP <input type="checkbox"/> TLS	Interworking Profile	TDC
TCP Port	5060	Signaling Manipulation Script	MEX_Mobile_From
UDP Port	5060	TCP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
TLS Port		UDP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
<input type="button" value="Finish"/>		<input type="button" value="Finish"/>	

7.7. Define Routing

Routing information is required for routing to the Session Manager on the internal side and the TDC SBC on the external side. The IP addresses and ports defined here will be used as the destination addresses for signalling. If no port is specified in the **Next Hop IP Address**, default 5060 is used. To define routing to the Session Manager, navigate to **Global Profiles → Routing** in the **UC-Sec Control Center** menu on the left hand side. Click on **Add Profile** and enter details in the **Routing Profile** pop-up menu (not shown).

- In the **Profile Name** field enter a descriptive name for the Session Manager, in this case **ASM9**, and click **Next**
- Enter the Session Manager SIP interface address and port in the **Next Hop Server 1** field
- Select **TCP** for the **Outgoing Transport**
- Click **Finish**

Global Profiles > Routing: ASM9

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

Routing Profiles: default, **ASM9**, Trunk Server

Click here to add a description.

Routing Profile

Add Routing Rule

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	10.10.9.61	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	TCP

To define routing to the TDC SBC, navigate to **Global Profiles → Routing** in the **UC-Sec Control Center** menu on the left hand side. Click on **Add Profile** and enter details in the **Routing Profile** pop-up menu.

- In the **Profile Name** field enter a descriptive name for the TDC SBC, in this case a generic name of **Trunk Server** was used, and click **Next**
- Enter the TDC SBC IP address and port in the **Next Hop Server 1** field
- Select **UDP** for the **Outgoing Transport**
- Click **Finish**

Global Profiles > Routing: Trunk Server

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

Routing Profiles: default, ASM9, **Trunk Server**

Click here to add a description.

Routing Profile

Add Routing Rule

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	192.168.198.130	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	UDP

7.8. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**; this replaces local information with IP addresses, generally the next hop. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise, particularly from the Session Manager. In some cases where Topology Hiding can't be applied, in particular the Contact header, IP addresses are translated to the Avaya SBCE external addresses using NAT.

To define Topology Hiding for the Session Manager, navigate to **Global Profiles → Topology Hiding** in the **UC-Sec Control Center** menu on the left hand side. Click on **Add Profile** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- In the **Profile Name** field enter a descriptive name for the Session Manager and click **Next**
- If the **Request-Line**, **Record-Route** and **Via** Headers aren't shown, click on **Add Header** and select from the **Header** drop down menu
- For each of the above headers, leave the **Replace Action** at the default value of **Auto**
- If the **From**, **To** and **SDP** Headers aren't shown, click on **Add Header** and select from the **Header** drop down menu
- For each of the above headers, select **IP** from the **Criteria** drop down menu (important for the **From** header so that the "anonymous.invalid" domain name for restricted CLI is not overwritten)
- For each of the headers leave the **Replace Action** at the default value of **Auto**

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

Note: The use of **Auto** results in an IP address being inserted in the host portion of the Request-URI as opposed to a domain name. If a domain name is required, the action **Overwrite** must be used where appropriate, and the required domain names entered in the **Overwrite Value** field. Different domain names could be used for the enterprise and the TDC network.

To define Topology Hiding for the TDC SBC, navigate to **Global Profiles → Topology Hiding** in the **UC-Sec Control Center** menu on the left hand side. Click on **Add Profile** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- In the **Profile Name** field enter a descriptive name for the TDC SBC and click **Next**
- If the **Request-Line**, **From** and **To** Headers aren't shown, click on **Add Header** and select from the **Header** drop down menu
- For each of the above headers, change the **Replace Action** to **Overwrite** and define the required domain name in the **Overwrite Value** field
- If the **Record-Route** and **Via** Headers aren't shown, click on **Add Header** and select from the **Header** drop down menu
- For each of the above headers, select **IP** from the **Criteria** drop down menu
- For each of the headers leave the **Replace Action** at the default value of **Auto**
- Repeat the above steps for the **SDP** if required and set the **Criteria** to **IP**

Global Profiles > Topology Hiding: TDC

Add Profile

Rename Profile

Clone Profile

Delete Profile

Topology Hiding Profiles

Click here to add a description.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
Via	IP/Domain	Auto	---
To	IP/Domain	Overwrite	test06.btrunk.se
SDP	IP	Auto	---
Record-Route	IP/Domain	Auto	---
From	IP/Domain	Overwrite	test06.btrunk.se
Request-Line	IP/Domain	Overwrite	test06.btrunk.se

Edit

default

cisco_th_profile

ASM9

TDC

7.9. Server Flows

Server Flows combine the previously defined profiles into an outgoing flow from the Session Manager to the TDC SBC and an incoming flow from the TDC SBC to the Session Manager. This configuration ties all the previously entered information together so that calls can be routed from the Session Manager to the TDC SBC and vice versa. The information for all Server Flows is shown on a single screen on the Avaya SBCE.

Device Specific Settings > End Point Flows: GSSCP_V9

UC-Sec Devices

GSSCP_V9

Subscriber Flows

Server Flows

Add Flow

Hover over a row to see its description.

Server Configuration: ASM9_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	Call_Server	*	*	*	Ext_Sig	Int_Sig	Int_Med	default-low	Trunk_Server	ASM9	None			

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	Trunk_Server	*	*	*	Int_Sig	Ext_Sig	Ext-Med	default-low	ASM9	TDC	None			




To define an outgoing Server Flow, navigate to **Device Specific Settings → End Point Flows**.

- Click on the **Server Flows** tab
- Select **Add Flow** and enter details in the pop-up menu (not shown)
- In the **Name** field enter a descriptive name for the outgoing server flow to the TDC SBC, in this case a generic name of **Trunk_Server** was used
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.3**
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.3**
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 7.3**
- In the **Routing Profile** drop-down menu, select the routing profile of the Session Manager defined in **Section 7.7**
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the TDC SBC defined in **Section 7.8** and click **Finish**

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	Trunk_Server	*	*	*	Int_Sig	Ext_Sig	Ext-Med	default-low	ASM9	TDC	None	  

An incoming Server Flow is defined as a reversal of the outgoing Server Flow

- Click on the **Server Flows** tab
- Select **Add Flow** and enter details in the pop-up menu (not shown)
- In the **Name** field enter a descriptive name for the incoming server flow to the Session Manager, in this case a generic name of **Call_Server** was used
- In the **Received Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.3**
- In the **Signaling Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.3**
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 7.3**
- In the **Routing Profile** drop-down menu, select the routing profile of the TDC SBC defined in **Section 7.7**
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Session Manager defined in **Section 7.8** and click **Finish**

Server Configuration: ASM9_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	Call_Server	*	*	*	Ext_Sig	Int_Sig	Int_Med	default-low	Trunk_Server	ASM9	None	  

8. Configure TDC Equipment

The configuration of the TDC equipment used to support the TDC Business Trunk service is outside of the scope of these Application Notes and will not be covered. To obtain further information on TDC equipment and system configuration please contact an authorised TDC representative.

9. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager **Home** tab click on **Session Manager** and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entity from the list and observe if the **Conn Status** and **Link Status** are showing as **up**.

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

Help ?

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: ASBCE

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	10.10.9.71	5060	TCP	Up	404 Called User Unknown	Up

2. From CS1000E Element Manager, 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.

AVAYA CS1000 Element Manager

Help | Logout

Managing: 192.168.1.5 Username: admin
System > Maintenance

Maintenance

☒ Select by Overlay ☐ Select by Functionality

<Select by Overlay>

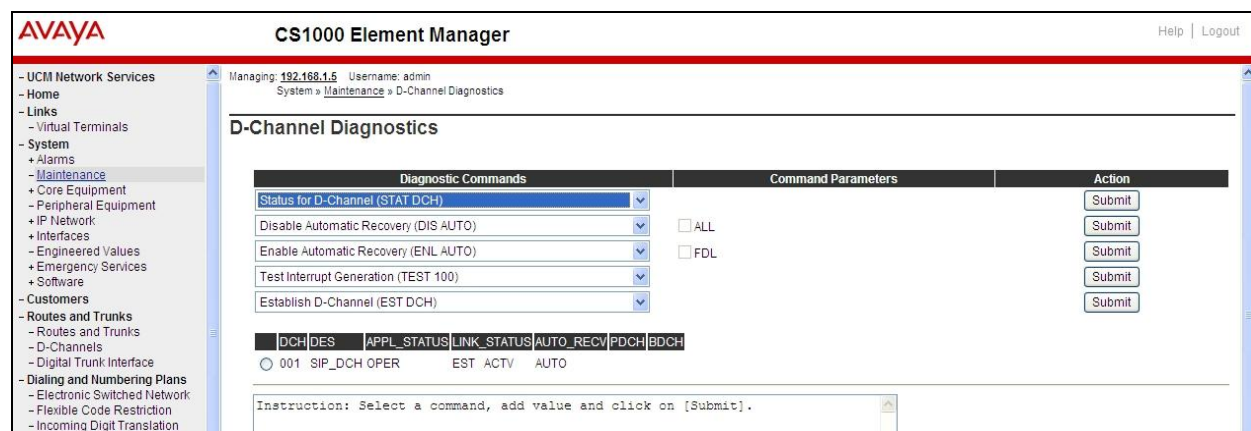
LD 30 - Network and Signaling
LD 32 - Network and Peripheral Equipment
LD 34 - Tone and Digit Switch
LD 36 - Trunk
LD 37 - Input/Output
LD 38 - Conference Circuit
LD 39 - Intergroup Switch and System Clock
LD 45 - Background Signaling and Switching
LD 46 - Multifrequency Sender
LD 48 - Link
LD 54 - Multifrequency 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**



AVAYA CS1000 Element Manager

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

D-Channel Diagnostics

Diagnostic Commands	Command Parameters	Action
Status for D-Channel (STAT DCH)		<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_REC ☐ PDCH ☐ BDCH

☐ 001 SIP_DCH OPER EST ACTV AUTO

Instruction: Select a command, add value and click on [Submit].

- Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active
- Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active
- Verify that the user on the PSTN can end an active call by hanging up
- Verify that an endpoint at the enterprise site can end an active call by hanging up.
- Should issues arise with the SIP trunk, check from the Avaya SBCE using **OPTIONS**. This is done by defining the heartbeat in the Server configuration then running a trace. To define the heartbeat, navigate to **Global Profiles → Server Configuration** in the **UC-Sec Control Center** menu on the left hand side and click on the Trunk Server profile. Select the **Heartbeat** tab and click on **Edit**
 - Check the **Enable Heartbeat** box
 - Select **OPTIONS** from the **Method** drop down menu
 - Enter the **Frequency** in seconds, for convenience this can be set to the minimum value of **60** seconds
 - Enter the **From URI** in Fully Qualified Domain Name format
 - Enter the **To URI** in FQDN
 - Click on **Finish**

Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS
Frequency	60 seconds
From URI	ping@192.168.122.56
To URI	ping@192.168.198.130
TCP Probe	<input type="checkbox"/>
TCP Probe Frequency	seconds
Finish	

To define the trace, navigate to **Troubleshooting → Trace Settings** in the **UC-Sec Control Center** menu on the left hand side and select the **Packet Capture** tab.

- Select the SIP Trunk interface from the **Interface** drop down menu
- Select the signalling interface IP address from the **Local Address** drop down menu
- Enter the IP address of the Service Provider's SBC in the **Remote Address** field or enter a * to capture all traffic
- Specify the **Maximum Number of Packets to Capture**, 10000 is shown as an example
- Specify the filename of the resultant pcap file in the **Capture Filename** field
- Click on **Start Capture**

Troubleshooting > Trace Settings: GSSCP_V9	
UC-Sec Devices	Packet Trace Call Trace Packet Capture Captures
GSSCP_V9	<div> <div>Packet Capture Configuration</div> <div> <div>Currently capturing</div> <div>No</div> </div> <div> <div>Interface</div> <div>B1</div> </div> <div> <div>Local Address (ip:port)</div> <div>All</div> </div> <div> <div>Remote Address (*, *:port, ip, ip:port)</div> <div>*</div> </div> <div> <div>Protocol</div> <div>All</div> </div> <div> <div>Maximum Number of Packets to Capture</div> <div>10000</div> </div> <div> <div>Capture Filename</div> <div>OPTIONS.pcap</div> </div> <div>Existing captures with the same name will be overwritten</div> </div> <div> <div>Start Capture</div> <div>Clear</div> </div>

To view the trace, select the **Captures** tab and click on the relevant filename in the list of traces.

Troubleshooting > Trace Settings: GSSCP_V9								
UC-Sec Devices	Packet Trace Call Trace Packet Capture Captures							
GSSCP_V9	<div>Refresh</div> <table> <tr> <th>File Name</th> <th>File Size (bytes)</th> <th>Last Modified</th> </tr> <tr> <td>OPTIONS_20130219093000.pcap</td> <td>0</td> <td>February 19, 2013 9:30:00 AM GMT</td> </tr> </table>		File Name	File Size (bytes)	Last Modified	OPTIONS_20130219093000.pcap	0	February 19, 2013 9:30:00 AM GMT
File Name	File Size (bytes)	Last Modified						
OPTIONS_20130219093000.pcap	0	February 19, 2013 9:30:00 AM GMT						

The trace is viewed as a standard pcap file in Wireshark. If the SIP trunk is working correctly, a SIP response will be seen from the Service Provider.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Communication Server 1000E R7.5, Avaya Aura® Session Manager R6.2 and Avaya Session Border Controller for Enterprise R4.0.5 to the TDC Business Trunk service. TDC Business Trunk is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. 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] *Installing and Configuring Avaya Aura® System Platform Release 6.2*, March 2012.
- [2] *Administering Avaya Aura® System Platform Release 6.2*, February 2012.
- [3] *Co-resident Call Server and Signaling Server Fundamentals, Avaya Communication Server 1000E Release 7.5*, Document Number NN43001-509
- [4] *Signaling Server and IP Line Fundamentals, Avaya Communication Server 1000E Release 7.5*, Document Number NN43001-125
- [5] *Implementing Avaya Aura® System Manager Release 6.2*, March 2012.
- [6] *Implementing Avaya Aura® Session Manager*, February 2012, Document Number 03-603473
- [7] *Administering Avaya Aura® Session Manager*, February 2012, Document Number 03-603324.
- [8] *E-SBC (Avaya Session Border Controller for Enterprise) Administration Guide*, November 2011
- [9] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [10] *TDC Business Trunk Technical Specification*, Release A5, September 2012

Appendix A – Avaya Communication Server 1000E Software

Avaya Communication Server 1000E call server patches and plug ins

TID: 46379

VERSION 4121

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

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

RELEASE 7
ISSUE 50 Q +
IDLE SET DISPLAY NORTEL
DepList 1: core Issue: 01 ALTERED(created: 2012-07-16 17:52:47 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2012-10-02 13:46:39(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2012-08-20 11:29:05(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	wi00832543	ISS1:10F1	DSP1AB04	10/08/2012	DSP1AB04.LW
01	wi00946113	ISS1:10F1	MGCBB15	24/04/2012	MGCBB15.LW
02	wi00890367	ISS1:10F1	MGCCCD02	24/04/2012	MGCCCD02.LW

Avaya Communication Server 1000E call server deplists

VERSION 4121
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 01 (created: 2012-07-16 17:52:47 (est)) ALTERED

IN-SERVICE PEPS

PAT#	CR #	PATCH REF #	NAME	DATE	FILENAME	SPECINS
000	wi00891626	ISS1:10F1	p31051_1	11/10/2012	p31051_1.cpl	YES
001	wi00951837	ISS1:10F1	p31485_1	11/10/2012	p31485_1.cpl	NO
002	wi00946477	ISS1:10F1	p31426_1	11/10/2012	p31426_1.cpl	NO
003	wi00906163	ISS1:10F1	p31205_1	11/10/2012	p31205_1.cpl	NO
004	wi00962211	ISS1:10F1	p31580_1	11/10/2012	p31580_1.cpl	NO
005	wi00877592	ISS1:10F1	p30880_1	11/10/2012	p30880_1.cpl	NO
006	wi00839134	ISS1:10F1	p30698_1	11/10/2012	p30698_1.cpl	YES
007	wi00984888	ISS1:10F1	p31795_1	11/10/2012	p31795_1.cpl	NO
008	wi00868729	ISS1:10F1	p31163_1	11/10/2012	p31163_1.cpl	NO
009	wi00886321	ISS1:10F1	p31009_1	11/10/2012	p31009_1.cpl	NO
010	wi00946282	ISS1:10F1	p31204_1	11/10/2012	p31204_1.cpl	NO
011	wi00841980	ISS1:10F1	p30618_1	11/10/2012	p30618_1.cpl	NO
012	wi00968448	ISS1:10F1	p31648_1	11/10/2012	p31648_1.cpl	YES
013	wi00977002	ISS2:10F1	p30786_2	11/10/2012	p30786_2.cpl	NO
014	wi00843623	ISS1:10F1	p30731_1	11/10/2012	p30731_1.cpl	YES
015	wi00958776	ISS1:10F1	p31542_1	11/10/2012	p31542_1.cpl	YES
017	wi00865477	ISS1:10F1	p30893_1	11/10/2012	p30893_1.cpl	YES
018	wi00879526	ISS1:10F1	p31007_1	11/10/2012	p31007_1.cpl	NO
019	wi00894243	ISS1:10F1	p31087_1	11/10/2012	p31087_1.cpl	NO
020	wi00890475	p30952	p31048_1	11/10/2012	p31048_1.cpl	NO
021	WI00927300	ISS1:10F1	p30999_1	11/10/2012	p30999_1.cpl	NO
022	wi00856991	ISS1:10F1	p17588_1	11/10/2012	p17588_1.cpl	NO
023	wi00688381	ISS1:10F1	p30104_1	11/10/2012	p30104_1.cpl	NO
024	wi00881777	ISS1:10F1	p25747_1	11/10/2012	p25747_1.cpl	NO
025	WI00853473	ISS1:10F1	p30625_1	11/10/2012	p30625_1.cpl	NO
026	wi00855423	ISS1:10F1	p31328_1	11/10/2012	p31328_1.cpl	YES
027	wi00943172	ISS1:10F1	p31402_1	11/10/2012	p31402_1.cpl	NO

028	wi00865477	ISS1:10F1	p30898_1	11/10/2012	p30898_1.cpl	YES
029	wi00850521	ISS1:10F1	p30709_1	11/10/2012	p30709_1.cpl	YES
030	wi00898327	ISS1:10F1	p31136_1	11/10/2012	p31136_1.cpl	NO
031	wi00871739	ISS1:10F1	p30856_1	11/10/2012	p30856_1.cpl	NO
032	wi00984178	ISS1:10F1	p31786_1	11/10/2012	p31786_1.cpl	NO
033	wi00839821	ISS1:10F1	p30619_1	11/10/2012	p30619_1.cpl	NO
034	wi00854130	ISS1:10F1	p30443_1	11/10/2012	p30443_1.cpl	NO
035	wi00871969	ISS1:10F1	p30768_1	11/10/2012	p30768_1.cpl	NO
036	wi00973241	ISS1:10F1	p31715_1	11/10/2012	p31715_1.cpl	NO
037	wi00946876	ISS1:10F1	p31430_1	11/10/2012	p31430_1.cpl	NO
038	wi01008943	ISS1:10F1	p31382_1	11/10/2012	p31382_1.cpl	NO
039	wi00969890	ISS1:10F1	p31664_1	11/10/2012	p31664_1.cpl	YES
040	wi00937672	ISS1:10F1	p31276_1	11/10/2012	p31276_1.cpl	NO
041	wi00875425	ISS1:10F1	p30943_1	11/10/2012	p30943_1.cpl	NO
042	wi00862574	iss1:10f1	p30870_1	11/10/2012	p30870_1.cpl	NO
043	wi00859499	ISS1:10F1	p30694_1	11/10/2012	p30694_1.cpl	NO
044	wi00925208	ISS1:10F1	p30986_1	11/10/2012	p30986_1.cpl	NO
045	wi00965285	ISS1:10F1	p31476_1	11/10/2012	p31476_1.cpl	NO
046	wi00900668	ISS1:10F1	p30456_1	11/10/2012	p30456_1.cpl	NO
047	wi00967509	ISS1:10F1	p31294_1	11/10/2012	p31294_1.cpl	NO
048	wi00879322	ISS1:10F1	p30954_1	11/10/2012	p30954_1.cpl	NO
049	wi00976209	ISS1:10F1	p31717_1	11/10/2012	p31717_1.cpl	YES
050	wi00956788	ISS1:10F1	p31638_1	11/10/2012	p31638_1.cpl	NO
051	wi00865477	ISS1:10F1	p30894_1	11/10/2012	p30894_1.cpl	YES
052	wi00991523	ISS1:10F1	p31603_1	11/10/2012	p31603_1.cpl	NO
053	wi00865477	ISS1:10F1	p30892_1	11/10/2012	p30892_1.cpl	YES
054	wi01007604	ISS1:10F1	p31983_1	11/10/2012	p31983_1.cpl	NO
055	wi00931028	ISS1:10F1	p31354_1	11/10/2012	p31354_1.cpl	YES
056	wi00932948	ISS1:10F1	p31077_1	11/10/2012	p31077_1.cpl	NO
057	wi01001911	ISS1:10F1	p31920_1	11/10/2012	p31920_1.cpl	NO
058	wi00838073	ISS1:10F1	p30588_1	11/10/2012	p30588_1.cpl	NO
059	wi00852365	ISS1:10F1	p30707_1	11/10/2012	p30707_1.cpl	NO
060	wi00927321	ISS1:10F1	p31286_1	11/10/2012	p31286_1.cpl	YES
061	wi00937114	ISS1:10F1	p31310_1	11/10/2012	p31310_1.cpl	NO
062	wi00877367	ISS1:10F1	p30534_1	11/10/2012	p30534_1.cpl	NO
063	wi00900096	ISS1:10F1	p31006_1	11/10/2012	p31006_1.cpl	NO
064	wi00905660	ISS1:10F1	p27968_1	11/10/2012	p27968_1.cpl	NO
065	wi00925141	ISS1:10F1	p30802_1	11/10/2012	p30802_1.cpl	NO
066	wi00943748	ISS1:10F1	p31516_1	11/10/2012	p31516_1.cpl	NO
067	wi00827950	ISS2:10F1	p30471_2	11/10/2012	p30471_2.cpl	NO
068	wi00930649	ISS1:10F1	p31570_1	11/10/2012	p31570_1.cpl	NO
069	wi00897279	ISS1:10F1	p31129_1	11/10/2012	p31129_1.cpl	NO
070	wi00961267	ISS1:10F1	p30288_1	11/10/2012	p30288_1.cpl	NO
071	wi00936714	ISS1:10F1	p31379_1	11/10/2012	p31379_1.cpl	NO
072	wi00906022	ISS1:10F1	p31202_1	11/10/2012	p31202_1.cpl	NO
073	wi00852389	ISS1:10F1	p30641_1	11/10/2012	p30641_1.cpl	NO
074	wi00857566	ISS1:10F1	p30766_1	11/10/2012	p30766_1.cpl	NO
075	wi00932204	ISS2:10F1	p31305_2	11/10/2012	p31305_2.cpl	NO
077	wi00891621	ISS1:10F1	p31037_1	11/10/2012	p31037_1.cpl	NO
078	wi00957235	ISS1:10F1	p31798_1	11/10/2012	p31798_1.cpl	NO
079	wi00948274	ISS1:10F1	p31365_1	11/10/2012	p31365_1.cpl	NO
080	wi00923899	ISS1:10F1	p31270_1	11/10/2012	p31270_1.cpl	NO
081	wi00856410	ISS1:10F1	p30749_1	11/10/2012	p30749_1.cpl	NO
082	wi00854415	ISS1:10F1	p30593_1	11/10/2012	p30593_1.cpl	NO
083	wi00896394	ISS1:10F1	p30807_1	11/10/2012	p30807_1.cpl	NO
084	wi00826075	ISS1:10F1	p30452_1	11/10/2012	p30452_1.cpl	NO
085	wi00863876	ISS1:10F1	p30787_1	11/10/2012	p30787_1.cpl	NO
086	wi00880386	ISS1:10F1	p30977_1	11/10/2012	p30977_1.cpl	NO
087	wi00840590	ISS1:10F1	p30767_1	11/10/2012	p30767_1.cpl	NO
088	wi00949627	ISS1:10F1	p31462_1	11/10/2012	p31462_1.cpl	NO
089	wi00842409	ISS1:10F1	p30621_1	11/10/2012	p30621_1.cpl	NO
090	wi00865477	ISS1:10F1	p30896_1	11/10/2012	p30896_1.cpl	YES
091	wi00897096	ISS1:10F1	p30676_1	11/10/2012	p30676_1.cpl	NO
092	wi00899584	ISS1:10F1	p30809_1	11/10/2012	p30809_1.cpl	NO
093	wi01007960	ISS1:10F1	p31965_1	11/10/2012	p31965_1.cpl	NO
094	wi00949273	ISS1:10F1	p31411_1	11/10/2012	p31411_1.cpl	NO
095	wi00839255	ISS1:10F1	p30591_1	11/10/2012	p30591_1.cpl	NO
096	wi00945997	ISS1:10F1	p31641_1	11/10/2012	p31641_1.cpl	NO
097	wi00903369	ISS1:10F1	p31165_1	11/10/2012	p31165_1.cpl	NO
098	wi00875701	ISS1:10F1	p30942_1	11/10/2012	p30942_1.cpl	NO

099	wi00884699	ISS1:10F1	p31000_1	11/10/2012	p31000_1.cpl	YES
100	wi00834382	ISS1:10F1	p30548_1	11/10/2012	p30548_1.cpl	NO
101	wi00960133	ISS2:10F1	p31557_2	11/10/2012	p31557_2.cpl	NO
102	wi00929140	ISS1:10F1	p31284_1	11/10/2012	p31284_1.cpl	NO
103	wi00948931	ISS1:10F1	p31407_1	11/10/2012	p31407_1.cpl	NO
104	wi00887744	ISS2:10F1	p31026_2	11/10/2012	p31026_2.cpl	NO
105	wi00905600	ISS1:10F1	p31201_1	11/10/2012	p31201_1.cpl	NO
106	wi00869243	ISS1:10F1	p30848_1	11/10/2012	p30848_1.cpl	NO
107	WI00854150	ISS1:10F1	p30468_1	11/10/2012	p30468_1.cpl	NO
108	wi00897176	ISS1:10F1	p30418_1	11/10/2012	p30418_1.cpl	NO
109	wi00903381	ISS1:10F1	p30421_1	11/10/2012	p30421_1.cpl	NO
110	wi00950575	ISS1:10F1	p31724_1	11/10/2012	p31724_1.cpl	NO
111	wi00908598	ISS1:10F1	p31235_1	11/10/2012	p31235_1.cpl	NO
112	wi00903437	ISS1:10F1	p31167_1	11/10/2012	p31167_1.cpl	NO
113	wi00900766	ISS1:10F1	p31159_1	11/10/2012	p31159_1.cpl	NO
114	wi00946558	ISS1:10F1	p31358_1	11/10/2012	p31358_1.cpl	NO
115	wi00932958	ISS1:10F1	p31115_1	11/10/2012	p31115_1.cpl	NO
116	wi00895090	ISS1:10F1	p31105_1	11/10/2012	p31105_1.cpl	NO
117	wi00824257	ISS1:10F1	p30447_1	11/10/2012	p30447_1.cpl	NO
118	wi00895181	ISS1:10F1	p31106_1	11/10/2012	p31106_1.cpl	NO
119	WI00928455	ISS1:10F1	p31297_1	11/10/2012	p31297_1.cpl	NO
120	wi00832106	ISS1:10F1	p30550_1	11/10/2012	p30550_1.cpl	NO
121	wi00953900	ISS1:10F1	p31494_1	11/10/2012	p31494_1.cpl	NO
122	wi00942734	ISS1:10F1	p31409_1	11/10/2012	p31409_1.cpl	NO
123	wi00986337	ISS1:10F1	p31803_1	11/10/2012	p31803_1.cpl	NO
124	wi00882293	ISS1:10F1	p31010_1	11/10/2012	p31010_1.cpl	NO
125	WI00843571	ISS1:10F1	p30627_1	11/10/2012	p30627_1.cpl	NO
126	wi00835294	ISS1:10F1	p30565_1	11/10/2012	p30565_1.cpl	NO
127	WI00836292	ISS1:10F1	p30554_1	11/10/2012	p30554_1.cpl	NO
128	wi00969581	ISS1:10F1	p31661_1	11/10/2012	p31661_1.cpl	YES
129	wi00921295	ISS1:10F1	p31265_1	11/10/2012	p31265_1.cpl	NO
130	wi00964006	ISS1:10F1	p31595_1	11/10/2012	p31595_1.cpl	YES
131	WI00836334	ISS1:10F1	p30481_1	11/10/2012	p30481_1.cpl	NO
132	wi00858335	ISS1:10F1	p30819_1	11/10/2012	p30819_1.cpl	NO
133	wi00859123	ISS1:10F1	p30648_1	11/10/2012	p30648_1.cpl	NO
134	wi00959820	ISS1:10F1	p31562_1	11/10/2012	p31562_1.cpl	NO
135	wi00905297	ISS1:10F1	p31195_1	11/10/2012	p31195_1.cpl	NO
136	wi00907697	ISS1:10F1	p31227_1	11/10/2012	p31227_1.cpl	NO
137	wi00951427	ISS1:10F1	p31478_1	11/10/2012	p31478_1.cpl	NO
138	wi00883604	ISS1:10F1	p30973_1	11/10/2012	p30973_1.cpl	NO
139	wi00962955	ISS1:10F1	p31585_1	11/10/2012	p31585_1.cpl	NO
140	wi00860279	ISS1:10F1	p30789_1	11/10/2012	p30789_1.cpl	NO
141	wi00909476	ISS1:10F1	p31340_1	11/10/2012	p31340_1.cpl	NO
142	wi00925218	ISS1:10F1	p30675_1	11/10/2012	p30675_1.cpl	NO
143	wi00836182	ISS1:10F1	p30450_1	11/10/2012	p30450_1.cpl	NO
144	wi00841273	ISS1:10F1	p30713_1	11/10/2012	p30713_1.cpl	NO
145	WI00889786	ISS1:10F1	p30750_1	11/10/2012	p30750_1.cpl	NO
146	wi00894443	ISS1:10F1	p31093_1	11/10/2012	p31093_1.cpl	NO
147	wi00896420	ISS1:10F1	p30867_1	11/10/2012	p30867_1.cpl	NO
148	wi00971029	ISS1:10F1	p31794_1	11/10/2012	p31794_1.cpl	NO
149	wi00955753	ISS1:10F1	p31733_1	11/10/2012	p31733_1.cpl	NO
150	wi00968531	ISS1:10F1	p31645_1	11/10/2012	p31645_1.cpl	NO
151	wi00930864	ISS1:10F1	p31325_1	11/10/2012	p31325_1.cpl	NO
152	wi00957252	ISS1:10F1	p31530_1	11/10/2012	p31530_1.cpl	NO
153	wi00880836	ISS1:10F1	p30976_1	11/10/2012	p30976_1.cpl	NO
154	wi00959457	ISS1:10F1	p31551_1	11/10/2012	p31551_1.cpl	NO
155	wi00896680	ISS1:10F1	p30357_1	11/10/2012	p30357_1.cpl	NO
156	wi00856702	ISS1:10F1	p30573_1	11/10/2012	p30573_1.cpl	NO
157	wi00897082	ISS1:10F1	p31124_1	11/10/2012	p31124_1.cpl	NO
158	wi00853178	ISS1:10F1	p30719_1	11/10/2012	p30719_1.cpl	NO
159	wi00938555	ISS1:10F1	p30881_1	11/10/2012	p30881_1.cpl	YES
160	WI00839794	ISS1:10F1	p28647_1	11/10/2012	p28647_1.cpl	NO
161	wi00965838	ISS1:10F1	p31623_1	11/10/2012	p31623_1.cpl	NO
162	wi00977393	ISS1:10F1	p31744_1	11/10/2012	p31744_1.cpl	YES
163	wi00959284	ISS1:10F1	p31531_1	11/10/2012	p31531_1.cpl	NO
164	wi00968353	ISS1:10F1	p31412_1	11/10/2012	p31412_1.cpl	NO
165	wi00998121	ISS1:10F1	p31897_1	11/10/2012	p31897_1.cpl	NO
166	wi00968157	ISS1:10F1	p31637_1	11/10/2012	p31637_1.cpl	NO
167	wi00967510	ISS1:10F1	p31147_1	11/10/2012	p31147_1.cpl	NO
168	wi00949410	ISS1:10F1	p31248_1	11/10/2012	p31248_1.cpl	NO

169	wi00969039	ISS1:10F1	p31643_1	11/10/2012	p31643_1.cpl	NO
170	wi00959463	ISS1:10F1	p31528_1	11/10/2012	p31528_1.cpl	NO
171	wi00983505	ISS1:10F1	p31758_1	11/10/2012	p31758_1.cpl	NO
172	wi00924886	ISS1:10F1	p31062_1	11/10/2012	p31062_1.cpl	YES
173	wi00969208	ISS1:10F1	p31656_1	11/10/2012	p31656_1.cpl	NO
174	wi00974272	ISS1:10F1	p31690_1	11/10/2012	p31690_1.cpl	YES
175	wi00988285	ISS1:10F1	p31824_1	11/10/2012	p31824_1.cpl	NO
176	wi00975659	ISS1:10F1	p31707_1	11/10/2012	p31707_1.cpl	NO
177	wi00960809	ISS1:10F1	p31564_1	11/10/2012	p31564_1.cpl	NO
178	wi00936935	ISS1:10F1	p31362_1	11/10/2012	p31362_1.cpl	NO
179	wi01012229	ISS1:10F1	p31993_1	11/10/2012	p31993_1.cpl	NO
180	wi00989828	ISS1:10F1	p31836_1	11/10/2012	p31836_1.cpl	NO
182	wi00985760	ISS1:10F1	p31913_1	11/10/2012	p31913_1.cpl	NO
183	wi01003896	ISS1:10F1	p31631_1	11/10/2012	p31631_1.cpl	NO
184	wi00978064	ISS1:10F1	p31760_1	11/10/2012	p31760_1.cpl	NO
185	wi00996889	ISS1:10F1	p31933_1	11/10/2012	p31933_1.cpl	NO
186	wi00991907	iss1:10f1	p31907_1	11/10/2012	p31907_1.cpl	NO
187	wi01005653	ISS1:10F1	p31952_1	11/10/2012	p31952_1.cpl	NO
188	wi01005513	ISS1:10F1	p31951_1	11/10/2012	p31951_1.cpl	NO
189	wi00967512	ISS1:10F1	p31384_1	11/10/2012	p31384_1.cpl	NO
190	wi01003814	ISS1:10F1	p31940_1	11/10/2012	p31940_1.cpl	NO
191	wi00984652	ISS1:10F1	p31792_1	11/10/2012	p31792_1.cpl	NO
192	wi00967514	ISS1:10F1	p31351_1	11/10/2012	p31351_1.cpl	NO
193	wi00978818	ISS1:10F1	p31919_1	11/10/2012	p31919_1.cpl	NO
194	WI00980321	ISS1:10F1	p31912_1	11/10/2012	p31912_1.cpl	YES
195	wi00999802	ISS1:10F1	p31577_1	11/10/2012	p31577_1.cpl	NO
196	wi01008316	ISS1:10F1	p32026_1	11/10/2012	p32026_1.cpl	YES
197	wi01003384	ISS1:10F1	p31479_1	11/10/2012	p31479_1.cpl	NO
198	wi01003999	ISS1:10F1	p31946_1	11/10/2012	p31946_1.cpl	YES
199	wi01011078	ISS1:10F1	p31996_1	11/10/2012	p31996_1.cpl	NO
200	wi01016398	ISS1:10F1	p32019_1	11/10/2012	p32019_1.cpl	NO
201	wi00973270	ISS1:10F1	p31751_1	11/10/2012	p31751_1.cpl	NO
202	wi00981711	ISS1:10F1	p31766_1	11/10/2012	p31766_1.cpl	NO
203	wi00977978	ISS1:10F1	p31831_1	11/10/2012	p31831_1.cpl	NO
204	wi00992974	ISS1:10F1	p31889_1	11/10/2012	p31889_1.cpl	NO
205	wi00980476	ISS1:10F1	p31387_1	11/10/2012	p31387_1.cpl	NO
206	wi01008106	ISS1:10F1	p31861_1	11/10/2012	p31861_1.cpl	NO
207	wi00906350	ISS1:10F1	p31219_1	11/10/2012	p31219_1.cpl	NO
208	wi01006063	ISS1:10F1	p31957_1	11/10/2012	p31957_1.cpl	NO
209	wi00971980	ISS1:10F1	p31863_1	11/10/2012	p31863_1.cpl	NO
210	wi01020959	ISS1:10F1	p32062_1	11/10/2012	p32062_1.cpl	NO
211	wi01011537	ISS1:10F1	p32024_1	11/10/2012	p32024_1.cpl	NO
212	wi01008505	ISS1:10F1	p31968_1	11/10/2012	p31968_1.cpl	NO
213	wi01014835	ISS1:10F1	p32015_1	11/10/2012	p32015_1.cpl	NO
214	wi00983007	ISS1:10F1	p31778_1	11/10/2012	p31778_1.cpl	YES
215	wi00987424	ISS1:10F1	p31815_1	11/10/2012	p31815_1.cpl	NO
216	wi00997559	ISS1:10F1	p31898_1	11/10/2012	p31898_1.cpl	NO
217	wi01012638	ISS1:10F1	p32008_1	11/10/2012	p32008_1.cpl	NO
218	wi00985153	ISS1:10F1	p31859_1	11/10/2012	p31859_1.cpl	NO
219	wi00979591	ISS1:10F1	p31746_1	11/10/2012	p31746_1.cpl	NO
220	wi00978892	ISS1:10F1	p31894_1	11/10/2012	p31894_1.cpl	NO
221	wi00996639	ISS1:10F1	p31886_1	11/10/2012	p31886_1.cpl	NO
222	wi00994044	ISS1:10F1	p31871_1	11/10/2012	p31871_1.cpl	NO
223	wi00991892	ISS1:10F1	p31853_1	11/10/2012	p31853_1.cpl	NO
224	wi00974856	ISS1:10F1	p31823_1	11/10/2012	p31823_1.cpl	NO
225	wi00993377	ISS1:10F1	p31860_1	11/10/2012	p31860_1.cpl	NO
226	wi00982566	ISS1:10F1	p31774_1	11/10/2012	p31774_1.cpl	NO
227	wi00993743	ISS1:10F1	p31865_1	11/10/2012	p31865_1.cpl	NO
228	wi00944019	ISS1:10F1	p31874_1	11/10/2012	p31874_1.cpl	NO
229	wi00998328	ISS1:10F1	p31899_1	11/10/2012	p31899_1.cpl	NO
230	wi01008188	ISS1:10F1	p32020_1	11/10/2012	p32020_1.cpl	NO
231	wi00987089	ISS1:10F1	p31809_1	11/10/2012	p31809_1.cpl	NO
232	wi00979414	ISS1:10F1	p31748_1	11/10/2012	p31748_1.cpl	YES
233	wi01006811	ISS1:10F1	p31967_1	11/10/2012	p31967_1.cpl	YES
234	wi01012289	p31274	p31999_1	11/10/2012	p31999_1.cpl	NO
235	wi00971209	ISS1:10F1	p31750_1	11/10/2012	p31750_1.cpl	NO
236	wi00997316	ISS1:10F1	p31870_1	11/10/2012	p31870_1.cpl	NO
237	wi00990993	ISS1:10F1	p31825_1	11/10/2012	p31825_1.cpl	NO
238	wi00977436	ISS1:10F1	p31834_1	11/10/2012	p31834_1.cpl	NO
239	wi01001938	ISS1:10F1	p31921_1	11/10/2012	p31921_1.cpl	YES

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240 wi01012423 ISS1:10F1 p26155_1 11/10/2012 p26155_1.cpl NO
241 wi01010472 ISS1:10F1 p31975_1 11/10/2012 p31975_1.cpl NO
242 wi01000796 ISS1:10F1 p31800_1 11/10/2012 p31800_1.cpl NO
243 wi00981928 ISS1:10F1 p31869_1 11/10/2012 p31869_1.cpl NO
244 wi00992921 ISS1:10F1 p31878_1 11/10/2012 p31878_1.cpl NO
245 wi01001588 ISS1:10F1 p31976_1 11/10/2012 p31976_1.cpl NO
246 wi00976951 ISS1:10F1 p30112_1 11/10/2012 p30112_1.cpl NO

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MDP>LAST SUCCESSFUL MDP REFRESH :2012-10-02 13:46:39(Local Time)

MDP>USING DEPLIST ZIP FILE DOWNLOADED :2012-08-20 11:29:05(est)

Avaya Communication Server 1000E signaling server service updates

In System service updates: 33

PATCH#	IN SERVICE	DATE	SPECINS	REMOVABLE	NAME
3	Yes	20/01/12	NO	YES	cs1000-dbcom-7.50.17-02.i386.000
4	Yes	18/12/12	NO	yes	tzdata-2011h-2.el5.i386.000
5	Yes	20/01/12	NO	YES	cs1000-shared-pbx-7.50.17.16-1.i386.000
6	Yes	20/01/12	NO	YES	cs1000-kcv-7.50.17.16-1.i386.000
7	Yes	20/01/12	NO	YES	cs1000-nrsmWebService-7.50.17.16-1.i386.000
9	Yes	02/10/12	YES	YES	cs1000-baseWeb-7.50.17.16-2.i386.000
10	Yes	20/01/12	NO	YES	cs1000-ipsec-7.50.17.16-1.i386.000
11	Yes	02/10/12	NO	yes	avaya-cs1000-cnd-4.0.20-00.i386.000
12	Yes	02/10/12	NO	YES	cs1000-pd-7.50.17.16-1.i386.000
13	Yes	02/10/12	NO	YES	cs1000-ncs-7.50.17.16-1.i386.000
14	Yes	20/01/12	NO	YES	ipsec-tools-0.6.5-14.el5.3 avaya 1.i386.000
15	Yes	20/01/12	NO	YES	spiritAgent-6.1-1.0.0.108.208.i386.000
16	No	18/12/12	NO	YES	cs1000-tps-7.50.17.16-24.i386.000
17	Yes	02/10/12	NO	YES	cs1000-EmCentralLogic-7.50.17.16-2.i386.000
20	Yes	02/10/12	NO	YES	cs1000-cs1000WebService_6-0-7.50.17.16-1.i386.000
21	Yes	02/10/12	NO	YES	cs1000-mscMusc-7.50.17.16-11.i386.000
22	Yes	02/10/12	NO	YES	cs1000-mscAnnc-7.50.17.16-10.i386.000
23	No	18/12/12	NO	YES	cs1000-sps-7.50.17.16-10.i386.000
24	Yes	27/03/12	NO	YES	cs1000-mscTone-7.50.17.16-1.i386.000
25	No	18/12/12	NO	YES	cs1000-fttrpkg-7.50.17.16-11.i386.000
26	Yes	18/12/12	NO	YES	cs1000-dmWeb-7.50.17.16-6.i386.000
27	Yes	02/10/12	NO	YES	cs1000-csoneksvrmgr-7.50.17.16-1.i386.000
28	No	18/12/12	NO	YES	cs1000-dbcom-7.50.17.16-1.i386.000
29	No	18/12/12	NO	YES	cs1000-vtrk-7.50.17.16-131.i386.001
30	Yes	27/03/12	NO	YES	cs1000-sps-7.50.17.16-4.i386.000
31	Yes	18/12/12	NO	YES	cs1000-linuxbase-7.50.17.16-13.i386.000
32	Yes	18/12/12	NO	YES	cs1000-mscAttn-7.50.17.16-3.i386.000
35	Yes	02/10/12	YES	YES	cs1000-nrsm-7.50.17.16-4.i386.000
36	Yes	02/10/12	NO	YES	cs1000-csmWeb-7.50.17.16-6.i386.000
37	Yes	02/10/12	NO	YES	cs1000-mscConf-7.50.17.16-1.i386.000
38	Yes	02/10/12	NO	YES	cs1000-emWeb_6-0-7.50.17.16-34.i386.000
40	Yes	02/10/12	NO	YES	cs1000-Jboss-Quantum-7.50.17.16-30.i386.000
42	Yes	02/10/12	NO	YES	cs1000-emWebLocal_6-0-7.50.17.16-3.i386.000

Avaya Communication Server 1000E system software

Product Release: 7.50.17.00

Base Applications

base	7.50.17	[patched]
NTAFS	7.50.17	
sm	7.50.17	
cs1000-Auth	7.50.17	
Jboss-Quantum	7.50.17	[patched]
cnd	n/a	[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	[patched]
ISECSH	7.50.17	
patchWeb	n/a	
EmCentralLogic	n/a	[patched]

Application configuration: CS+SS+NRS+EM

Packages:

CS+SS+NRS+EM

Configuration version: 7.50.17-00

cs	7.50.17	
dbcom	7.50.17.16	[patched]
cslogin	7.50.17	
sigServerShare	7.50.17	[patched]
csv	7.50.17	
tps	7.50.17.16	
vtrk	7.50.17.16	
pd	7.50.17.16	[patched]
sps	7.50.17.16	[patched]
ncs	7.50.17.16	[patched]
gk	7.50.17	
nrsm	7.50.17	[patched]
nrsmWebService	7.50.17	[patched]
managedElementWebService	7.50.17	
EmConfig	7.50.17	
emWeb_6-0	7.50.17	[patched]
emWebLocal 6-0	7.50.17	[patched]
csmWeb	7.50.17	[patched]
bcc	7.50.17	
ftrpkg	n/a	
cs1000WebService_6-0	7.50.17	[patched]
mscAnnc	7.50.17.16	[patched]
mscAttn	7.50.17.16	[patched]
mscConf	7.50.17.16	[patched]
mscMusc	7.50.17.16	[patched]
mscTone	7.50.17.16	[patched]

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