



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.5, Avaya Aura[®] Session Manager R6.1 to Support BT Wholesale/HIPCOM SIP Trunk Service – Issue 1.0

Abstract

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

BT is a member of the DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect lab.

1. Introduction

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

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of CS1000E and Session Manager. The enterprise site was configured to use the SIP Trunk Service provided by BTW/HIPCOM.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

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

2.2. Test Results

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

- Incoming call to busy trunks or SIP Trunk signaling failure the following was observed - PSTN user hears Number Unobtainable tone after several seconds and 500 Service Unavailable SIP message. The global parameter set on BTW/HIPCOM's SBC is 4 hunts per call, so if the call doesn't set up on the first try BTW/HIPCOM's SBC will re-try a further 3 times.

2.3. Support

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

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to BTW/HIPCOM using SIP Trunks. Located at the enterprise site are Session Manager and a Communication Server 1000E. Endpoints are Avaya 1140 series IP telephones, Avaya 1200 series (not shown in **Figure 1**) IP telephones (with Unistim and SIP firmware), Avaya IP Softphones (SMC3456, 2050 and one-X Communicator), Avaya Digital telephone, Analogue telephone and fax machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

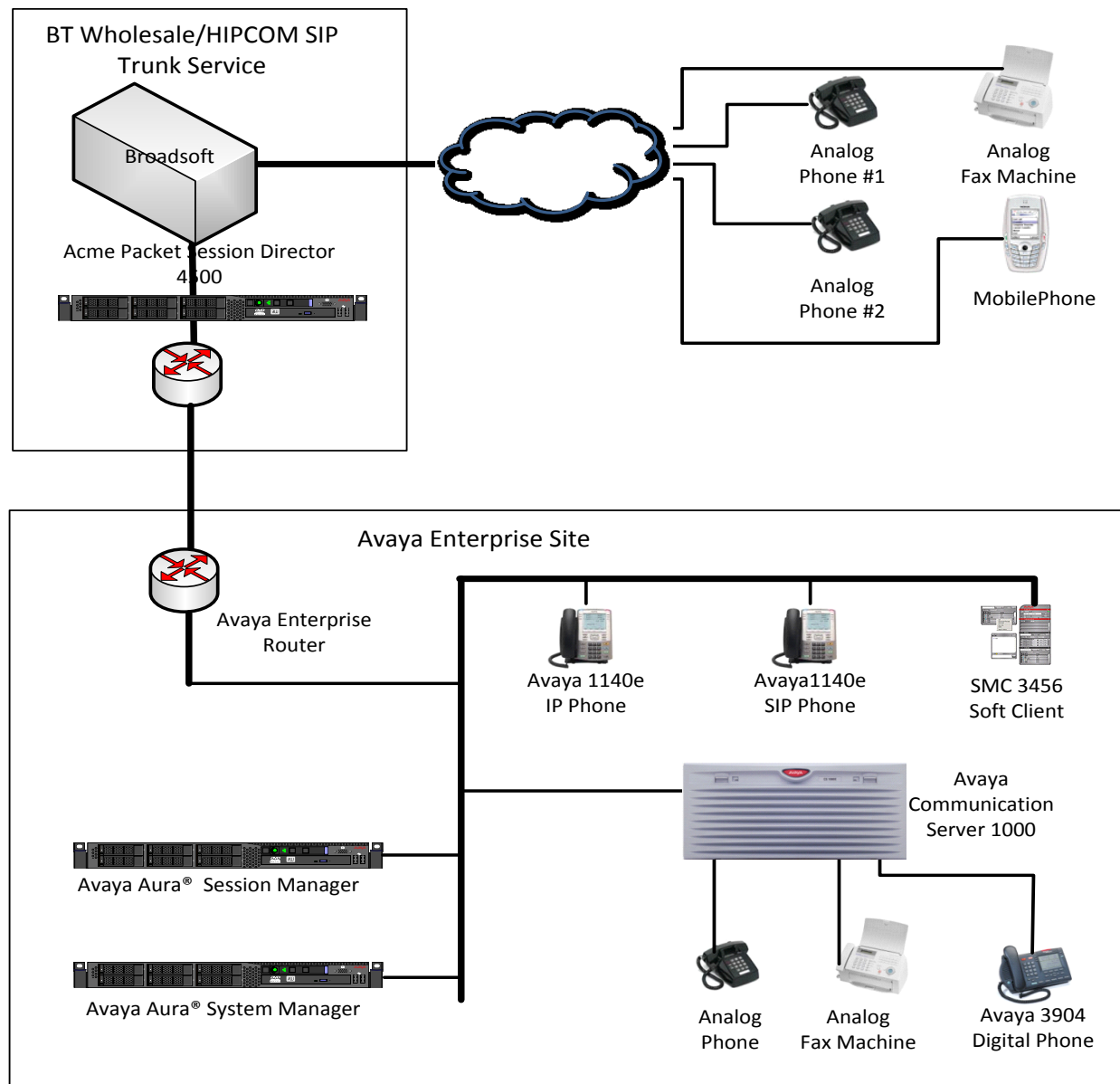


Figure 1: Avaya Communication Server 1000E with BT Wholesale/HIPCOM SIP Trunk Topology

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 server	Avaya Aura® Session Manager R6.1 Build: 6.1.0.0.610023
Avaya S8800 server	Avaya Aura® System Manager R6.1 Load: 6.1.0.0.7345 Service Pack 0
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	Avaya Communication Server 1000E R7.5 Version 7.50.17 Service Update: 7.50_17Nov23 Deplist: X21 07.50Q
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 AB04
Avaya 1140e and 1230 Unistim Telephones	FW: 0625C8A
Avaya 1140e and 1230 SIP Telephones	FW: 04.01.13.00.bin
Avaya SMC 3456	Version 2.6 build 57666
Avaya one-X® Communicator	Avaya one-X® Communicator - cs6.1.0.10
Avaya 2050 IP Softphone	Release 4.0.2.0062
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
BTW/HIPCOM SIP Trunk Service	Acme Packet 4500 Net-Net SBC ver SCX6.1.0 Broadsoft - ver 14 Sevice Pack 9 Configuration version - HIPCOM v8.1

5. Configure Avaya Communication Server 1000E

This section describes the steps required to configure Communication Server 1000E for SIP Trunking and also the necessary configuration for terminals (Analogue, SIP and IP phones). SIP trunks are established between Communication Server 1000E and Session Manager. These SIP trunks carry SIP Signaling associated with BTW/HIPCOM's SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from BTW/HIPCOM Acme Packet Net-Net Session Director. Incoming SIP messages are then directed to Communication Server 1000E (see **Figure 1**). Once a SIP message arrives at Communication Server 1000E, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Server 1000E and may be first subject to outbound features such as route selection, digit manipulation and class of service restrictions. Once Communication Server 1000E selects a SIP trunk, the SIP signaling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to BTW/HIPCOM's network. Specific Communication Server 1000E configuration was performed using Element Manager and the system terminal interface. The general installation of the

Communication Server 1000E, System Manager and Session Manager is presumed to have been previously completed and is not discussed here.

5.1. Logging into the Avaya Communication Server 1000E

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 BTW/HIPCOM's network, and any other SIP trunks needed. See the following screenshot for a typical System Limits printout. The value of **SIP ACCESS PORTS** defines the maximum number of SIP trunks for the Communication Server 1000E.

```
load Overlay 22
req: SLT

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

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

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

Load overlay 21 and confirm the customer is setup to use **ISDN** trunks by typing the **PRT** and **NET_DATA** commands as shown below.

```
Load Overlay 21
REQ: PRT
TYPE: net
TYPE NET_DATA
CUST 0

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

5.3. Configure Codec's for Voice and FAX operation

BTW/HIPCOM SIP Trunk service supports G.711A/G.729A voice codec's and T.38 FAX transmissions. Using the Communication Server 1000E element manager sidebar, navigate to the **IP Network → IP Telephony Nodes → Node Details → Voice Gateway (VGW) and Codecs** property page and configure the Communication Server 1000E General codec settings as shown in the screenshot below. The values highlighted are required for correct operation; most of the options are turned on by default but its good practice to ensure that they are set as shown below.

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

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

General

Echo cancellation: ☒ Use canceller, with tail delay: 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 (squellch DTMF from TDM to IP)
☒ Modem/Fax pass-through
☒ V.21 Fax tone detection
☐ R factor calculation

Next, scroll down and configure the CS1000E to use **Codec G.711** and **Codec G.729** so as to align with what the Service Provider supports on their network. The settings configured were set to default, e.g. if G.729 is enabled the voice payload size and jitter buffer delay values are not changed and are left as is. The relevant settings are highlighted in the following screenshot.

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

General | Voice Codecs | Fax

Codec G711: ☒ Enabled (required)

Voice payload size: 20 (milliseconds per frame)

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

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

☐ Voice Activity Detection (VAD)

Codec G722: ☐ Enabled

Voice payload size: 20 (milliseconds per frame)

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

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

Codec G729: ☒ Enabled

Voice payload size: 30 (milliseconds per frame)

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

* Required Value.

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

Save Cancel

Finally, configure the **Fax** settings as in the highlighted section of the next screenshot with system defaults as shown below.

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

General | Voice Codecs | Fax

Codec G723.1: ☐ Enabled

Voice payload size: 30 (milliseconds per frame)

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

Nominal Maximum

Maximum delay may be automatically adjusted based on nominal settings.

Coding rate: 5.3 (kbps)

Fax

Codec name: T.38 FAX

Maximum rate: 14400 (bps)

Fax TCF method: 2

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

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

Packet size: 20 (bps)

* Required Value.

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

Save Cancel

5.4. Virtual Trunk Gateway Configuration

Use Communication Server 1000E Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and verify the highlighted section is completed with the correct IP addresses and subnet masks of the Node. At this stage the call server has an ip address and so too does the signalling server. The Node ip is the ip address that the IP phones use to register. This is also where the SIP trunk connection is made to the Session Manager. When an entity link is added in Session Manager for the CS1000E it is the Node ip that is used (see **Section 6.4 – Define SIP Entities** for more details).

CS1000 Element Manager

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

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

Node ID: * (0-9999)

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

Embedded LAN (ELAN) Telephony LAN (TLAN)

Gateway IP address: * Node IPv4 address: *

Subnet mask: * Subnet mask: *

Node IPv6 address:

* Required Value.

Associated Signaling Servers & Cards

Select to add Add Remove Make Leader [Print](#) [Refresh](#)

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

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

- **Vtrk gateway application:** Provides option to select Gateway applications. The three supported modes are **SIP Gateway (SIPGw)**, **H.323Gw** and **H.323Gw/SIPGw**
- **SIP domain name:** The SIP Domain Name is the SIP Service Domain. The SIP Domain Name configured in the Signaling Server properties must match the Service Domain name configured in the Session Manager, in this case **lab.ic.static.hipcom.co.uk**
- **Local SIP port:** The Local SIP Port is the port to which the gateway listens. The default value is **5060**
- **Gateway endpoint name:** This field cannot be left blank so a value is needed here. This field is used when a Network Routing Server 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 **5000**

- **Proxy or Redirect Server:** Primary TLAN ip address is the SM100 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

Node ID: 5000 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Vtrk gateway application: ☒ Enable gateway service on this node

General

Vtrk gateway application: SIP Gateway (SIPGw) ▼

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

Local SIP port: 5060 * (1 - 65535)

Gateway endpoint name: spcs1k *

Gateway password: *

Application node ID: 5000 * (0-9999)

Enable failsafe NRS: ☐

SIP ANAT: ☒ IPv4

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)

Information will be captured for the IP addresses listed below.

Monitor IP: Add

Monitor addresses:

Remove

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 10.10.8.56

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

Port: 5060 (1 - 65535)

Transport protocol: TCP ▼

Options: ☐ Support registration

☐ Primary CDS proxy

SIP URI Map:

Public E.164 domain names	Private domain names
National: <input type="text"/>	UDP: <input type="text"/>
Subscriber: <input type="text"/>	CDP: <input type="text"/>
Special number: <input type="text"/>	Special number: <input type="text"/>
Unknown: <input type="text"/>	Vacant number: <input type="text"/>
	Unknown: <input type="text"/>

5.5. Configure Bandwidth Zones

Bandwidth Zones are used for alternate call routing between IP stations and for Bandwidth Management. SIP trunks require a unique zone, not shared with other resources and best practice dictates that IP telephones, 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.

The screenshot shows the CS1000 Element Manager interface. The left sidebar contains a navigation tree with 'System' expanded, and 'IP Network' and 'Zones' highlighted. The main content area is titled 'Bandwidth Zones' and contains a table with the following data:

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

5.6. Configure Incoming Digit Conversion Table

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

The screenshot shows the CS1000 Element Manager interface for 'Digit Conversion Tree 0 Configuration'. The left sidebar contains a navigation tree with 'Dialing and Numbering Plans' expanded, and 'Incoming Digit Translation' and 'Customer 00' highlighted. The main content area is titled 'Digit Conversion Tree 0 Configuration' and contains a table with the following data:

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

5.7. Configure SIP Trunks

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

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

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

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

Next, configure the SIP trunk Route Data Block (RDB) using the Communication Server 1000E system terminal and overlay 16. Load **Overlay 16**, enter **RDB** at the prompt, press return and commence configuration. The value for **DCH** is the same as previously entered in overlay 17. The value for **NODE** should match the node value in **Section 5.3**. The value for **ZONE** should match that used in **Section 5.5** for **SIP_VTRK**. The remaining highlighted values are important for correct SIP trunk operation.

Overlay 16 TYPE: RDB CUST 00 ROUT 100 TYPE RDB CUST 00 ROUT 100 DES VIR_TRK TKTP TIE NPID_TBL_NUM 0 ESN NO RPA NO CNVT NO SAT NO RCLS EXT VTRK YES ZONE 0020 PCID SIP CRID NO NODE 5000 DTRK NO ISDN YES MODE ISLD DCH 10 IFC SL1 PNI 00001 NCNA YES NCRD YES TRO NO FALT NO CTYP UKWN INAC NO ISAR NO DAPC NO MBXR NO MBXOT NPA MBXT 0 PTYP ATT CNDP UKWN AUTO NO DNIS NO DCDR NO ICOG IAO SRCH LIN TRMB YES STEP	ACOD 1600 TCPP NO PII NO AUXP NO TARG CLEN 1 BILN NO OABS INST IDC YES DCNO 0 NDNO 0 * DEXT NO DNAM NO SIGO STD STYP SDAT MFC NO ICIS YES OGIS YES TIMR ICF 1920 OGF 1920 EOD 13952 LCT 256 DSI 34944 NRD 10112 DDL 70 ODT 4096 RGV 640 GTO 896 GTI 896 SFB 3 PRPS 800 NBS 2048 NBL 4096 IENB 5 TFD 0 VSS 0 VGD 6 EESD 1024 SST 5 0 DTD NO SCDT NO 2 DT NO NEDC ORG FEDC ORG	CPDC NO DLTN NO HOLD 02 02 40 SEIZ 02 02 SVFL 02 02 DRNG NO CDR NO NATL YES SSL CFWR NO IDOP NO VRAT NO MUS YES MRT 21 PANS YES RACD NO MANO NO FRL 0 0 FRL 1 0 FRL 2 0 FRL 3 0 FRL 4 0 FRL 5 0 FRL 6 0 FRL 7 0 OHQ NO OHQT 00 CBQ NO AUTH NO TTBL 0 ATAN NO OHTD NO PLEV 2 OPR NO ALRM NO ART 0 PECL NO DCTI 0 TIDY 1600 100 ATRR NO TRRL NO SGRP 0 ARDN NO CTBL 0 AACR NO
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Next, configure virtual trunk members using the Communication Server 1000E system terminal and **Overlay 14**. Configure sufficient trunk members to carry both incoming and outgoing PSTN calls. The following example shows a single SIP trunk member configuration. Load **Overlay 14** at the system terminal and type **new X**, where X is the required number of trunks. Continue entering data until the overlay exits. The **RTMB** value is a combination of the **ROUT** value entered in the previous step and the first trunk member (usually 1). The remaining highlighted values are important for correct SIP trunk operation.

```
Overlay 14
new 30
TN 160 0 0 0
DATE
PAGE
DES VIR_TRK
TN 160 0 00 00 VIRTUAL
TYPE IPTI
CUST 0
XTRK VTRK
ZONE 0020
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK ANLG
NCOS 0
RTMB 100 1
CHID 1
TGAR 1
STRI/STRO WNK WNK
SUPN YES
AST NO
IAPG 0
CLS TLD DTN CND ECD WTA LPR APN THFD XREP SPCD MSBT
P10 NTC
TKID
AACR NO
```

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

Overlay 86 new CUST 0 FEAT rlb RLI 24 ELC NO ENTR 0 LTER NO ROUT 100 TOD 0 ON 1 ON 2 ON 3 ON 4 ON 5 ON 6 ON 7 ON VNS NO SCNV NO CNV NO EXP NO FRL 0 DMI 0 CTBL 0 ISDM 0	FCI 0 FSNI 0 BNE NO DORG NO SBOC NRR PROU 1 IDBB DBD IOHQ NO OHQ NO CBQ NO ISET 0 NALT 5 MFRL 0 OVL 0
---	--

Next, configure Special Prefix Number(s) (SPN) which users will dial to reach PSTN numbers. Use the Communication Server 1000E system terminal and overlay 90. The following are some example SPN entries used. The highlighted **RLI** value previously configured in overlay 86 is used as the Route List Index (**RLI**), this is the default PSTN route to the SIP Trunk service.

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

5.8. Configure 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 5.4** for **VIRTUALSETS**.

```

Overlay 20 IP Telephone configuration
DES 1140
TN 096 0 01 16 VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
ERL 0
ECL 0
FDN 0
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 1
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDA CDMD LLCN MCTD CLBD AUTR
GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTA AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBF RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
FDSF NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA PKCH MUTA MWTD
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```

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```
DVLD CROD CROD
CPND_LANG ENG
RCO 0
HUNT 0
LHK 0
PLEV 02
PUID
DANI NO
AST 00
IAPG 1
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 MCR 8000 0      MARP
      CPND
        CPND_LANG ROMAN
          NAME IP1140
          XPLN 10
          DISPLAY_FMT FIRST, LAST
01 MCR 8000 0
      CPND
        CPND_LANG ROMAN
          NAME IP1140
          XPLN 10
          DISPLAY_FMT FIRST, LAST
02
03 BSY
04 DSP
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
```

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

Overlay 20 - Digital Set configuration

```
TYPE: 3904
DES 3904
TN 000 0 09 08 VIRTUAL
TYPE 3904
CDEN 8D
CTYP XDLC
CUST 0
MRT
ERL 0
FDN 0
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 1
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBD WTA LPR PUA MTD FND HTD TDD HFA GRLD CRPA STSD
    MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD SLKD CCSD SWD LNA CNDA
    CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
    ICDA CDMA LLCN MCTD CLBD AUTU
    GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD
    CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
    UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
    DRDD EXR0
    USMD USRD ULAD CCBF RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
    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 8866 0 MARP

CPND

CPND_LANG ROMAN

NAME Digital Set

XPLN 10

DISPLAY_FMT FIRST, LAST

01 MCR 8866 0

CPND

CPND_LANG ROMAN

NAME Digital Set

XPLN 10

DISPLAY_FMT FIRST, LAST

02 DSP

03 MSB

04

05

06

07

08

09

10

11

12

13

14

15

16

17 TRN

18 AO6

19 CFW 16

20 RGA

21 PRK

22 RNP

23

24 PRS

25 CHG

26 CPN

27 CLT

28 RLT

29

30

31

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

Overlay 20 - Analogue Telephone Configuration

```
DES 500
TN 100 0 00 03
TYPE 500
CDEN 4D
CUST 0
MRT

ERL 00000
WRLS NO
DN 8888
AST NO
IAPG 0
HUNT
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
XLST
SCI 0
SCPW
SFLT NO
CAC_MFC 0
CLS UNR DTN FBD XFD WTA THFD FND HTD ONS
LPR XRD AGRD CWD SWD MWD RMMD SMWD LPD XHD SLKD CCSD LND TVD
CFTD SFD MRD C6D CNID CLBD AUTU
ICDD CDMD LLCN EHTD MCTD
GPUD DPUD CFXD ARHD OVDD AGTD CLTD LDTD ASCD SDND
MBXD CPFA CPTA UDI RCC HBTD IRGD DDGA NAMA MIND
NRWD NRCD NROD SPKD CRD PRSD MCRD
EXR0 SHL SMSD ABDD CFHD DNDY DNO3
CWND USMD USRD CCBF BNRD OCBF RTDD RBDD RBHD FAXA CNUD CNAD PGND FTTC
FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTD
PLEV 02
PUID
AACS NO
MLWU_LANG 0
FTR DCFW 4
```

5.9. Configure the SIP Line Gateway Service

SIP terminal operation requires the Communication Server node to be configured as a SIP Line Gateway (SLG) before SIP telephones can be configured. Prior to configuring the SIP Line node properties, the SIP Line service must be enabled in the customer data block. Use the Communication Server 1000E system terminal and overlay 15 to activate SIP Line services, as in the following example where **SIPL_ON** is set to **YES**.

```
SLS_DATA
SIPL_ON YES
UAPR 78
NMME NO
```

If a numerical value is entered against the **UAPR** setting, this number will be pre appended to all SIP Line configurations, and is used internally in the SIP Line server to track SIP terminals. Use Element Manager and navigate to the **IP Network → IP Telephony Nodes → Node Details → SIP Line Gateway Configuration** page. See the following screenshot for highlighted critical parameters. The value for **SIP Domain Name** must match that configured in **Section 6.1**.

- **SIP Line Gateway Application** ☐ **Enable the SIP line service on the node**, check the box to enable
- **SLG endpoint name** The endpoint name is the same endpoint name as the SIP Line Gateway and will be used for SIP gateway registration
- **SLG Local Sip port** Default value is **5070**
- **SLG Local TLS port** Default value is **5071**

AVAYA CS1000 Element Manager

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

Node ID: 5000 - SIP Line Configuration Details

General | SIP Line Gateway Settings | SIP Line Gateway Service

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

General

SIP domain name: lab.ic.static.hipc *

SLG endpoint name: spcs1k

SLG Group ID:

SLG Local Sip port: 5070 (1 - 65535)

SLG Local Tls port: 5071 (1 - 65535)

Virtual Trunk Network Health Monitor

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

Monitor IP: Add

Monitor addresses: Remove

5.10. Configure SIP Line Telephones

When SIP Line service configuration is completed, use the Communication Server 1000E system terminal and **Overlay 20** to add a Universal Extension (UEXT). See the following example of a SIP Line extension. The value for **UXTY** must be **SIPL**. This example is for an Avaya SIP telephone, so the value for **SIPN** is 1. The **SIPU** value is the username, **SCPW** is the logon password and these values are required to register the SIP telephone to the SLG. The value for **CFG_ZONE** is the value set for **SIPLINEZONE** in **Section 5.4**. A unique telephone number is entered for value **KEY 00**. The value for **KEY 01** is comprised of the **UAPR** value (set to 78 previously in this section) and the telephone number used in **KEY 00**.

Overlay 20 - SIP Telephone Configuration

```
DES SIPD
TN 096 0 01 15 VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TLSV 0
SIPU 8889
NDID 5
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
ERL 0
ECL 0
VSIT NO
FDN
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW 1234
SFLT NO
CAC MFC 0
CLS UNR FBD WTA LPR MTD FNA HTA TDD HFD CRPD
MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LND CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
```

---continued on next page---

---continued from previous page---

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

AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 MCR 8889 0 MARP
    CPND
        CPND_LANG ROMAN
        NAME Sigma 1140
        XPLN 11
        DISPLAY_FMT FIRST, LAST*
01 HOT U 788889 MARP 0
02
03
04
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23 *
24 PRS
25 CHG
26 CPN
27
28
29
30
31
```


5.11. Save Configuration

Expand **Tools** → **Backup and Restore** on the left navigation panel and select **Call Server**. Select **Backup** (not shown) and in the window below click **Submit** to save configuration changes as shown below.

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

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

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

Configuration of Communication Server 1000E is complete.

6. Configure Avaya Aura® Session Manager

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

- Define SIP Domain
- Define Location for Avaya Communication Server 1000E
- Define SIP Entities
- Define Entity Links
- Define Routing Policy
- Define Dial Patterns

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

6.1. Define SIP domains

Expand **Elements** → **Routing** and select **Domains** from the left navigation menu, click **New** . Enter the following values and use default values for remaining fields.

- **Name** Enter the Domain Name specified for the SIP Gateway in **Section 5.4**. In the sample configuration, **lab.ic.static.hipcom.co.uk** was used
- **Type** Verify **sip** is selected
- **Notes** Add a brief description [Optional]

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

The screenshot displays the Avaya Aura System Manager 6.1 interface. The left navigation pane shows the 'Routing' section expanded, with 'Domains' selected. The main content area is titled 'Domain Management' and includes buttons for 'Edit', 'New', 'Duplicate', 'Delete', and 'More Actions'. Below these buttons is a table with one item, 'lab.ic.static.hipcom.co.uk', which is of type 'sip'. The 'Default' checkbox for this domain is unchecked. The table has columns for 'Name', 'Type', 'Default', and 'Notes'. A red box highlights the table content.

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

6.2. Define Location for Avaya Communication Server 1000E

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

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

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

- **IP Address Pattern** Enter the logical pattern used to identify the location. For the sample configuration, **10.10.8.*** was used
- **Notes** Add a brief description [Optional]

Click **Commit** to save (not shown). The screenshot below shows the Location defined for Communication Server 1000E in the sample configuration.

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth.
See Session Manager -> Session Manager Administration -> Global Setting

General

* Name: SipLab8

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Per-Call Bandwidth Parameters

* Default Audio Bandwidth: 80 Kbit/sec

Location Pattern

Add Remove

2 Items Refresh Unit of Measurement: Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.10.2.*	
<input type="checkbox"/>	* 10.10.8.*	

6.3. Define SIP Entities

A SIP Entity must be added for Communication Server 1000E and also for BTW/HIPCOM's SBC. Expand **Elements** → **Routing** and select **SIP Entities** from the left navigation menu. 2 new SIP Entities will need to be added as noted above. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Name** Enter an identifier for the SIP Entity
- **FQDN or IP Address** Enter TLAN IP address of Communication Server 1000E Node identified in **Section 5.4**. For BTW/HIPCOM's SBC enter the IP address of the public interface
- **Type** Select **Other** for the Communication Server 1000E
- **Notes** Enter a brief description [Optional]
- **Location** Select the Location defined for Communication Server 1000E in **Section 6.2** and also apply this same location to the BTW/HIPCOM's SBC

In the **SIP Link Monitoring** section.

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

Click **Commit** to save the definition of the new SIP Entity. The following screenshot shows the SIP Entity defined for Communication Server 1000E in the sample configuration.

The screenshot shows the 'SIP Entity Details' configuration page for a SIP Entity named 'CS1K'. The page is divided into two main sections: 'General' and 'SIP Link Monitoring'. The 'General' section contains fields for Name, FQDN or IP Address, Type, Notes, Adaptation, Location, and Time Zone. The 'SIP Link Monitoring' section contains a dropdown for SIP Link Monitoring. The 'SIP Entities' menu item in the left navigation pane is highlighted with a red box. The 'SIP Entity Details' title bar is also highlighted with a red box. The 'SIP Link Monitoring' section is highlighted with a red box.

Field	Value
Name	CS1K
FQDN or IP Address	10.10.8.3
Type	Other
Notes	
Adaptation	
Location	SipLab8
Time Zone	Etc/GMT
Override Port & Transport with DNS SRV	<input type="checkbox"/>
SIP Timer B/F (in seconds)	4
Credential name	
Call Detail Recording	none
SIP Link Monitoring	Use Session Manager Configuration

The following screenshot shows the SIP Entity defined for BTW/HIPCOM's SBC in the sample configuration, the ip address has been blanked out for security purposes.

The screenshot displays the 'SIP Entity Details' configuration page for the entity 'BTW/HIPCOM'. The left sidebar shows a navigation menu with 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and includes a 'Commit' button. The 'General' tab is active, showing the following fields:

- * Name: BTW/HIPCOM
- * FQDN or IP Address: [Redacted] (The IP address has been blanked out for security purposes.)
- Type: Gateway
- Notes:
- Adaptation:
- Location: SipLab8
- Time Zone: Etc/GMT

Below these fields, there is a checkbox for 'Override Port & Transport with DNS SRV' which is unchecked. Further down, the 'SIP Timer B/F (in seconds)' is set to 4. The 'Credential name' field is empty. The 'Call Detail Recording' is set to 'none'. At the bottom, the 'SIP Link Monitoring' section shows 'SIP Link Monitoring' set to 'Use Session Manager Configuration'.

A SIP Entity link must also be defined for your Session Manager but that is not shown in this document.

6.4. Define Entity links

The SIP trunk between the Session Manager and the Communication Server 1000E is described by an Entity link. The same is needed between the Session Manager and BTW/HIPCOM's SBC. Expand **Elements** → **Routing** and select **Entity Links** from the left navigation menu. Click **New** (not shown). Enter the following values.

- **Name** Enter an identifier for the link to each telephony system
- **SIP Entity 1** Select SIP Entity defined for **Session Manager**
- **SIP Entity 2** Select the SIP Entity defined for Avaya Communication Server 1000E and BTW/HIPCOM's SBC in **Section 6.3** i.e. **CS1000E** and **BTW/HIPCOM**
- **Protocol** After selecting both SIP Entities, select **TCP** as the required protocol
- **Port** Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is **5060**
- **Trusted** Enter a tick in the box
- **Notes** Enter a brief description [Optional]

Click **Commit** to save **Entity Link** definition. The following screen shows the entity link defined for the SIP trunk between Session Manager and Communication Server 1000E.

The screenshot shows the 'Entity Links' configuration page in the Avaya Aura System Manager 6.1. The left navigation pane has 'Entity Links' selected under the 'Routing' section. The main area displays a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The values in the table are: Name: CS1K, SIP Entity 1: Session Manager, Protocol: TCP, Port: 5060, SIP Entity 2: CS1K, Port: 5060, Trusted: checked, Notes: toCS1K. There are 'Commit' and 'Cancel' buttons at the top right.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* CS1K	* Session Manager	TCP	* 5060	* CS1K	* 5060	<input checked="" type="checkbox"/>	toCS1K

The following screen shows the entity link defined for the SIP trunk between Session Manager and BTW/HIPCOM's SBC.

The screenshot shows the 'Entity Links' configuration page in the Avaya Aura System Manager 6.1. The left navigation pane has 'Entity Links' selected under the 'Routing' section. The main area displays a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The values in the table are: Name: BTW/HIPCOM, SIP Entity 1: Session Manager, Protocol: UDP, Port: 5060, SIP Entity 2: BTW/HIPCOM, Port: 5060, Trusted: checked, Notes: . There are 'Commit' and 'Cancel' buttons at the top right.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* BTW/HIPCOM	* Session Manager	UDP	* 5060	* BTW/HIPCOM	* 5060	<input checked="" type="checkbox"/>	

6.5. Define Routing Policy

Routing policies describe the conditions under which calls will be routed to CS1000E from either SIP endpoint registered to Session Manager or from other telephony system. It also describes the routing policies for which calls will be routed to BTW/HIPCOM's SBC. To add a routing policy, expand **Elements** → **Routing** and select **Routing Policies**. Click **New** (not shown). In the **General** section, enter the following values.

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

In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). For routing policy to the Communication Server 1000E, select the SIP Entity associated with Communication Server 1000E defined in **Section 6.4** and click **Select** (not shown). The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition.

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

The following screenshot shows the Routing Policy for CS1000E:

Routing Policy Details

General

* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
CS1K	10.10.8.3	Other	

For routing policy to BTW/HIPCOM's SBC, select the SIP Entity associated with this, defined in **Section 6.4** and click **Select** (not shown). The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition. The following screenshot shows the Routing Policy for BTW/HIPCOM's SBC, the ip address has been blanked out for security purposes.

AVAYA
Avaya Aura® System Manager 6.1

[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Routing

Session Manager

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

Routing Policy Details

Commit

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

General

* Name:

to BTW/HIPCOM

Disabled:

☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
BTW/HIPCOM	[REDACTED], 5	Gateway	

6.6. Define Dial Pattern

Dial patterns are used to route calls to appropriate SIP Entities. In the sample configuration, since the DDI range given for the testing all numbers that start with **44203** will be routed to the Communication Server 1000E for terminating to test sets. Alternately calls that are originated on the Communication Server 1000E that start with digits **00353** will be routed to BTW/HIPCOM's SBC, there is a dialing pattern added for this as well. To define a dial pattern, expand **Elements** → **Routing** and select **Dial Patterns** (not shown). Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Pattern** Enter dial pattern for calls to Avaya Communication Server 1000E
- **Min** Enter the minimum number digits that must be dialed
- **Max** Enter the maximum number digits that may be dialed
- **SIP Domain** Select the SIP Domain from drop-down menu or select **All** if Session Manager should accept incoming calls from all SIP domains
- **Notes** Enter a brief description [Optional]

In the **Originating Locations and Routing Policies** section, click **Add**. The **Originating Locations and Routing Policy List** page opens (not shown).

- **Originating Locations** Select **ALL**
- **Routing Policies** Select the Routing Policy defined for Communication Server 1000E in **Section 6.5**

Click **Select** to save these changes and return to **Dial Pattern Details** page. Click **Commit** to save. The following screen shows the Dial Pattern defined for sample configuration. The following screenshot shows the Routing Policy for Communication Server 1000E.

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

Dial Pattern Details

General

* Pattern: 44203

* Min: 5

* Max: 12

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

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

Repeat the above steps to add the dial Pattern to BTW/HIPCOM's SBC; select the routing policy defined for this in **Section 6.5**. The following screenshot shows the Routing Policy for BTW/HIPCOM's SBC.

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

Dial Pattern Details

General

* Pattern: 00353

* Min: 5

* Max: 16

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh

<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"/>	SipLab8		to BTW/HIPCOM	0	<input type="checkbox"/>	BTW/HIPCOM	

7. BT Wholesale/HIPCOM Service Provider Configuration

The configuration of BTW/HIPCOM's equipment used to support the SIP trunk service is outside of the scope for these application notes and will not be covered. To obtain further information on BTW/HIPCOM's equipment and system configuration please contact an authorised BTW/HIPCOM representative.

8. Verification

8.1. Verify Avaya Communication Server 1000E Operational Status

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

The screenshot displays the AVAYA CS1000 Element Manager web interface. The top header shows the AVAYA logo, the title 'CS1000 Element Manager', and links for 'Help' and 'Log'. Below the header, the left navigation pane is expanded to show the 'System' menu, with 'Maintenance' selected. The main content area is titled 'Maintenance' and contains two radio buttons: 'Select by Overlay' (selected) and 'Select by Functionality'. Below these buttons are two tables. The first table, titled '<Select by Overlay>', lists various LDs (LD 30, LD 32, LD 34, LD 36, LD 37, LD 38, LD 39, LD 45, LD 46, LD 48, LD 54, LD 60, LD 75, LD 80, LD 96, LD 117, LD 135, LD 137, LD 143). The second table, titled '<Select Group>', lists three groups: 'D-Channel Diagnostics', 'MSDL Diagnostics', and 'TMDI Diagnostics'. The 'LD 96 - D-Channel' entry in the first table and the 'D-Channel Diagnostics' entry in the second table are highlighted with blue backgrounds.

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

Select **Status for D-Channel (STAT DCH)** command and click **Submit** (not shown) to verify status of virtual D-Channel as shown below. Verify the status of the following fields:

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

D-Channel Diagnostics

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

DCH
DES
APPL_STATUS
LINK_STATUS
AUTO_RECV
PDCH
BDCH

☐ 010
Vtrk
OPER
EST
ACTV
AUTO




STAT DCH 010

Command executed successfully.

8.2. Verify Avaya Aura® Session Manager Operational Status

8.2.1. Verify Avaya Aura® Session Manager is Operational

Navigate to **Elements → Session Manager → Dashboard** to verify the overall system status for Session Manager. Specifically, verify the status of the following fields as shown below in the screenshot.

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

Home / Elements / Session Manager- Session Manager

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

Session Manager Dashboard

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

Session Manager Instances

Service State
Shutdown System
As of 9:22 AM

1 Item
Refresh
Show ALL
Filter: Enable

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

Select : All, None

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

Reset

Synchronize

Certificate Management ▾

Connection Status

1 Item

Refresh

Show

ALL ▾

Filter: Enable

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

Select : None

8.2.2. Verify SIP Entity Link Status

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

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

Verify the SIP link is **Up** between the Session Manager and BTW/HIPCOM's SBC by going through the same process as outlined above but selecting the SIP Entity for BTW/HIPCOM in the **All Monitored SIP Entities** table.

SIP Entity, Entity Link Connection Status								
This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.								
All Entity Links to SIP Entity: BTW/HIPCOM								
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	85.119.57.5	5060	UDP	Up	200 OK	Up

9. Conclusion

These Application Notes describe the configuration necessary to connect the Avaya Communication Server 1000E, Avaya Aura® Session Manager to BTW/HIPCOM's SIP Service. The testing was successfully performed with BTW/HIPCOM, refer to **Section 2.2** for more details.

10. Additional References

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

- [1] Avaya Aura® Session Manager Overview, Doc ID 03-603323, available at <http://support.avaya.com>.
- [2] Installing and Configuring Avaya Aura® Session Manager, available at <http://support.avaya.com>.
- [3] Avaya Aura® Session Manager Case Studies, available at <http://support.avaya.com>
- [4] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, available at <http://support.avaya.com>.
- [5] Administering Avaya Aura® Session Manager, Doc ID 03-603324, available at <http://support.avaya.com>
- [6] IP Peer Networking Installation and Commissioning, Release 7.5, Document Number NN43001-313, available at <http://support.avaya.com>
- [7] Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-116, available at <http://support.avaya.com>
- [8] Network Routing Service Fundamentals, Release 7.5, Document Number NN43001-130, Issue 03.02, available at <http://support.avaya.com>
- [9] Co-resident Call Server and Signaling Server Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-509, available at <http://support.avaya.com>
- [10] Signaling Server and IP Line Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-125, available at <http://support.avaya.com>

Appendix A

Avaya Communication Server 1000E Software

Avaya Communication Server 1000E call server patches and plug ins

```

08/04/11 10:25:28
TID: 008808096

VERSION 4021

System type is - Communication Server 1000E/CP PM

CP PM - Pentium M 1.4 GHz

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

RELEASE 7
ISSUE 50 Q +
IDLE_SET_DISPLAY Avaya 7.5
DepList 1: core Issue: 02(created: 2010-11-30 15:12:45 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2010-12-06 15:33:54(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2010-12-01 08:31:36(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE

LOADWARE VERSION: PSWV 100
INSTALLED LOADWARE PEPS : 0
ENABLED PLUGINS : 0

```

Avaya Communication Server 1000E call server deplists

```

VERSION 4121
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 01 (created: 2011-05-24 10:13:35 (est)) ALTERED

IN-SERVICE PEPS
PAT# CR #          PATCH REF #    NAME          DATE          FILENAME        SPECINS
012 wi00843623     ISS1:10F1     p30731_1    16/06/2011    p30731_1.cpl    YES
013 WI00843571     ISS1:10F1     p30627_1    16/06/2011    p30627_1.cpl    NO
014 wi00871739     ISS1:10F1     p30856_1    16/06/2011    p30856_1.cpl    NO
015 wi00852365     ISS1:10F1     p30707_1    16/06/2011    p30707_1.cpl    NO
016 wi00852389     ISS1:10F1     p30641_1    16/06/2011    p30641_1.cpl    NO
017 wi00839134     ISS1:10F1     p30698_1    16/06/2011    p30698_1.cpl    YES
018 wi00856702     ISS1:10F1     p30573_1    16/06/2011    p30573_1.cpl    NO
019 wi00857566     ISS1:10F1     p30766_1    16/06/2011    p30766_1.cpl    NO
020 wi00850521     ISS1:10F1     p30709_1    16/06/2011    p30709_1.cpl    YES
021 wi00860722     ISS1:10F1     p30784_1    16/06/2011    p30784_1.cpl    YES
022 wi00863876     ISS1:10F1     p30787_1    16/06/2011    p30787_1.cpl    NO
023 WI00853473     ISS1:10F1     p30625_1    16/06/2011    p30625_1.cpl    NO
024 wi00854130     ISS1:10F1     p30443_1    16/06/2011    p30443_1.cpl    NO
025 wi00875425     ISS1:10F1     p30943_1    16/06/2011    p30943_1.cpl    NO
026 wi00853658     ISS1:10F1     p30990_1    16/06/2011    p30990_1.cpl    NO
027 wi00875701     ISS1:10F1     p30942_1    16/06/2011    p30942_1.cpl    NO
028 wi00853031     ISS1:10F1     p30531_1    16/06/2011    p30531_1.cpl    NO
029 wi00877367     ISS1:10F1     p30534_1    16/06/2011    p30534_1.cpl    NO
030 wi00871969     ISS1:10F1     p30768_1    16/06/2011    p30768_1.cpl    NO
031 wi00886321     ISS1:10F1     p31009_1    16/06/2011    p31009_1.cpl    NO
032 WI00836334     ISS1:10F1     p30481_1    16/06/2011    p30481_1.cpl    NO
033 wi00836182     ISS1:10F1     p30450_1    16/06/2011    p30450_1.cpl    NO
034 wi00858335     ISS1:10F1     p30819_1    16/06/2011    p30819_1.cpl    NO
035 wi00860279     ISS1:10F1     p30789_1    16/06/2011    p30789_1.cpl    NO
036 wi00866570     ISS1:10F1     p30477_1    16/06/2011    p30477_1.cpl    NO

```


037	wi00854415	ISS1:10F1	p30593_1	16/06/2011	p30593_1.cpl	NO
038	WI00836292	ISS1:10F1	p30554_1	16/06/2011	p30554_1.cpl	NO
039	WI00839794	ISS1:10F1	p28647_1	16/06/2011	p28647_1.cpl	NO
040	wi00824257	ISS1:10F1	p30447_1	16/06/2011	p30447_1.cpl	NO
041	wi00827950	ISS2:10F1	p30471_2	16/06/2011	p30471_2.cpl	NO
042	wi00879814	ISS1:10F1	p30970_1	16/06/2011	p30970_1.cpl	NO
043	WI00854150	ISS1:10F1	p30468_1	16/06/2011	p30468_1.cpl	NO
044	wi00873382	ISS1:10F1	p30832_1	16/06/2011	p30832_1.cpl	NO
045	wi00853178	ISS1:10F1	p30719_1	16/06/2011	p30719_1.cpl	NO
046	wi00869695	ISS1:10F1	p30654_1	16/06/2011	p30654_1.cpl	NO
047	wi00834382	ISS1:10F1	p30548_1	16/06/2011	p30548_1.cpl	NO
048	wi00836472	ISS1:10F1	p30626_1	16/06/2011	p30626_1.cpl	NO
049	wi00854409	ISS1:10F1	p30479_1	16/06/2011	p30479_1.cpl	NO
050	WI00728461	ISS1:10F1	p30346_1	16/06/2011	p30346_1.cpl	NO

MDP>LAST SUCCESSFUL MDP REFRESH :2011-05-25 10:18:44 (Local Time)

MDP>USING DEPLIST ZIP FILE DOWNLOADED :2011-05-25 04:41:04 (est)

Avaya Communication Server 1000E signaling server service updates

Product Release: 7.50.17.00

In system patches: 0

In System service updates: 8

PATCH#	IN_SERVICE	DATE	SPECINS	REMOVABLE	NAME
0	Yes	07/02/11	NO	YES	cs1000-baseWeb-7.50.17.01-1.i386.000
1	Yes	07/02/11	NO	YES	cs1000-linuxbase-7.50.17.04-00.i386.000
2	Yes	07/02/11	NO	YES	cs1000-sps-7.50.17-01.i386.000
3	Yes	07/02/11	NO	YES	cs1000-shared-pbx-7.50.17-01.i386.000
4	Yes	07/02/11	NO	YES	cs1000-bcc-7.50.17.03-00.i386.000
5	Yes	07/02/11	NO	YES	cs1000-Jboss-Quantum-7.50.17.01-1.i386.000
6	Yes	07/02/11	NO	YES	cs1000-vtrk-7.50.17-11.i386.000
7	Yes	07/02/11	NO	YES	cs1000-dmWeb-7.50.17.04-00.i386.001

There is no SP in loaded status.

The last applied SP: Service Pack Linux 7.50 17 20110118.ntl, It is a STANDARD SP.

Has been applied by user nortel on Mon Feb 7 14:59:01 2011

Avaya Communication Server 1000E system software

Product Release: 7.50.17.00

Base Applications

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

Application configuration: CS+SS+EM

Packages: CS+SS+EM

Configuration version: 7.50.17-00

cs	7.50.17	
dbcom	7.50.17	[patched]
cslogin	7.50.17	
sigServerShare	7.50.17	[patched]
csv	7.50.17	
tps	7.50.17	

vtrk	7.50.17	[patched]
pd	7.50.17	
sps	7.50.17	[patched]
ncs	7.50.17	
gk	7.50.17	
EmConfig	7.50.17	
emWeb_6-0	7.50.17	[patched]
emWebLocal 6-0	7.50.17	
csmWeb	7.50.17	
bcc	7.50.17	[patched]
ftrpkg	7.50.17	
cs1000WebService_6-0	7.50.17	
managedElementWebService	7.50.17	
mscAnnc	7.50.17	
mscAttn	7.50.17	
mscConf	7.50.17	
mscMusc	7.50.17	
mscTone	7.50.17	

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