



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.5 with Avaya Aura[®] Session Manager 6.1 to support BT Global Services NOAS SIP Trunk - Issue 1.1

Abstract

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

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Session Manager and Communication Server 1000E. The enterprise site was configured to use the SIP Trunk Service provided by BT, with all incoming and outgoing PSTN calls via the BT SIP Trunk Service.

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 BT. Incoming PSTN calls were terminated on Digital, Unistim, SIP and Analog telephones at the enterprise side.
- Outgoing calls from the enterprise site were completed via BT to PSTN telephones. Outgoing calls from the enterprise to the PSTN were made from Digital, Unistim, SIP and Analog telephones.
- Calls were made using G.729A, and G.711A codecs.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using the T.38 transmission mode.
- DTMF transmission using RFC 2833 with successful IVR menu progression.
- 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.
- Transmission and response of SIP OPTIONS messages sent by BT requiring Avaya response and sent by Avaya requiring BT response.

2.2. Test Results

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

- The Calling Line Identity (CLI) presented to a PSTN called party is set to a pre-configured trunk number if the CLI is withheld at the enterprise side.
- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.
- Routing to emergency numbers (such as 112) was tested.
- G729 annex b (silence suppression) is not supported by BT SIP Trunk Service and thus was not tested.
- G711mu is not supported by BT SIP Trunk Service and thus was not tested.
- Early media is only supported for UEXT type phones on Communication Server 1000.
- PSTN called party hangup during an active call did not cause the call to drop. The Communication Server 1000E caller must hangup first, or wait for the PSTN T2ISUP timer to expire.
- Unsupervised transfer of incoming or outgoing PSTN calls to PSTN called parties is not permitted; this is a PSTN imposed restriction. The same restriction exists for supervised transfers of an existing PSTN call to a PSTN called party.
- Call hold has a time limit of less than 16 minutes. If this time limit is exceeded, the call drops. This is a PSTN imposed restriction.
- Calls to/from SMC 3456 soft clients using unsupported codecs failed, most likely because the call server was unable to determine the set capabilities and the SMC 3456 not correctly handling the calls.
- The BT SIP Trunk Service did not handle some SIP 5xx messages correctly, causing Call Admission Control (CAC) issues on PSTN calls, with the effect of reducing the pool of available SIP trunks. A workaround was to manually clear the CAC counters. This will be resolved with a software patch to the BT SIP Trunking Service.
- T.38 outgoing Fax calls (either single or multiple page, G.711 setup) only transmitted as clear channel Fax calls. T.38 outgoing Fax does not work with NOAS.
- T.38 outgoing Fax calls (either single or multiple pages, G.729 setup) fail. T.38 outgoing Fax does not work with NOAS.

2.3. Support

For technical support on BT products please use the following web link.

<http://btbusiness.custhelp.com/app/contact>

3. Reference Configuration

Figure 1 illustrates the tested configuration. The test configuration shows an Avaya enterprise site connected to the BT SIP Trunk Service. Located at the enterprise site are a Session Manager and Communication Server 1000E. Endpoints are Avaya 1140e series IP telephones (one with SIP firmware), Avaya 3904 series Digital telephones, an SMC 3456 Soft Client, an Analog Telephone and a Fax Machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

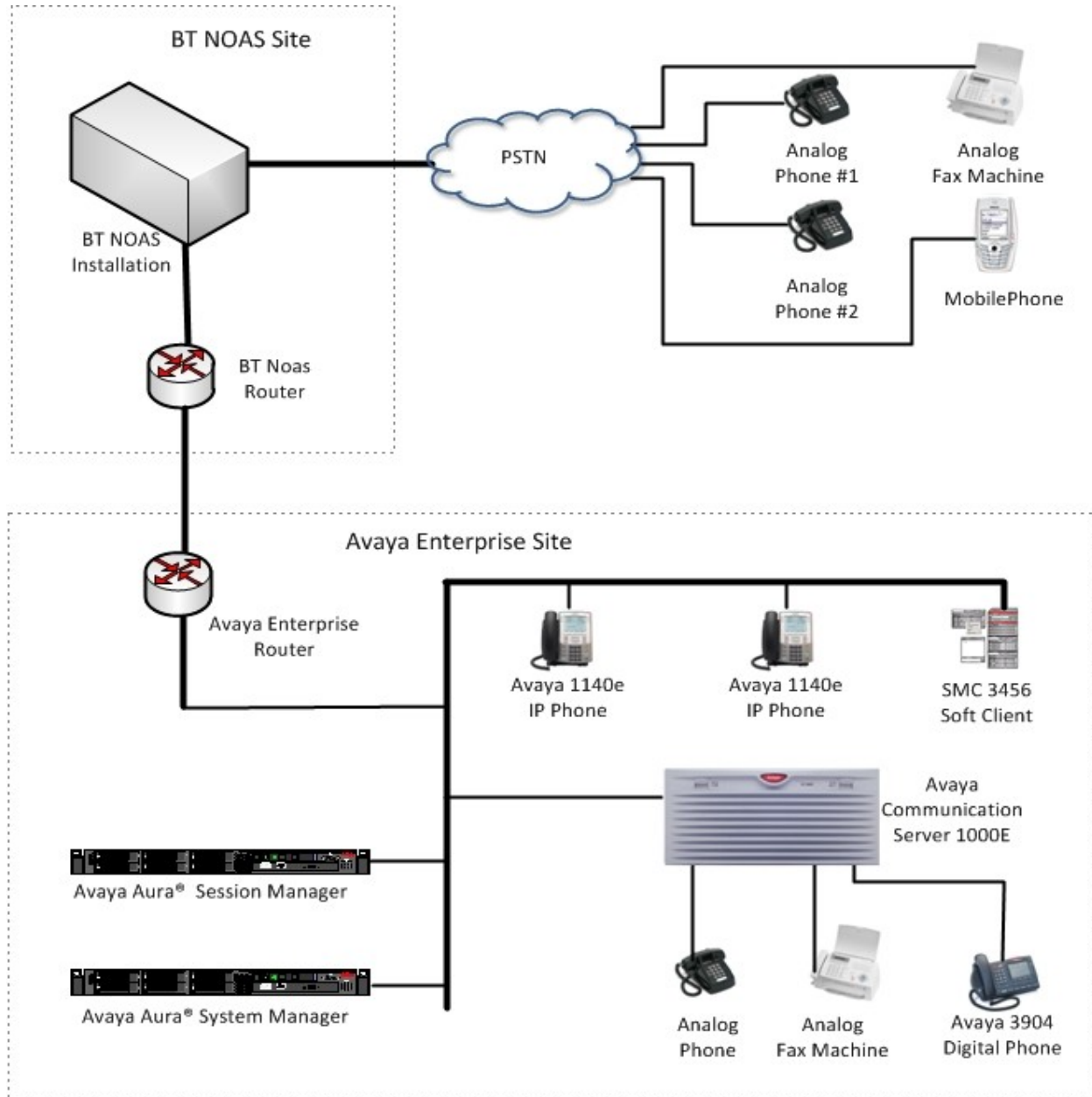


Figure 1: BT Test Configuration

4. Equipment and Software Validated

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

Equipment	Software
Avaya Communication Server 1000E	Avaya Communication Server 1000E 007.50Q/ 7.50.17 (PSWV 100 with latest Patches and Deplist)
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 DSP2 Version: DSP2 AB03
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 (6.1.0.0.610023)
Avaya S8800 Server	Avaya Aura® System Manager 6.1 (6.1.4.0 Build Number 6.1.0.4.5072)
Avaya 1140e Unistim Phone	5.0
Avaya 1140e SIP Phone	4.00.03.00
Analog Phone	N/A
BT SIP Trunk Service	2.1.0.8

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 (digital, analog, SIP and IP phones). SIP trunks are established between Communication Server 1000E and Session Manager. These SIP trunks carry SIP Signaling associated with BT SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from the BT Global Services NOAS SIP Trunk router, through which the BT Global Services NOAS SIP Trunk service directs incoming SIP messages to Communication Server 1000E (see **Figure 1**). Once a SIP message arrives at Communication Server 1000E, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Server 1000E and may be first subject to outbound features such as route selection, digit manipulation and class of service restrictions. Once Communication Server 1000E selects a SIP trunk, the SIP signaling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the Avaya Enterprise router and on to the BT network. Specific Communication Server 1000E configuration was performed using Element Manager and the system terminal interface. The general installation of the Avaya Communication Server 1000E and System Manager and Session Manager is presumed to have been previously completed and is not discussed here.

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

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

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

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

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

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

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

5.2. Configure Codecs for Voice and FAX operation

The BT Global Services NOAS SIP Trunk service supports G.711A and G.729A voice codecs and T.38 FAX transmissions. Using the Communication Server 1000E element manager sidebar, navigate to the **IP Network** → **IP Telephony Nodes** → **Node Details** → **VGW and Codecs** property page and configure the Communication Server 1000E General codec settings as in the next screenshot. The values highlighted are required for correct operation.

CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 1231 - 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

Next, scroll down and configure the G.711 and G.729 codec settings. The relevant settings are highlighted in the following screenshot.

CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 1231 - Voice Gateway (VGW) and Codecs

General | **Voice Codecs** | Fax

Codec G711: Enabled (required)

Maximum delay may be automatically adjusted based on nominal settings.

Voice Activity Detection (VAD)

Codec G722: Enabled

Voice payload size: (milliseconds per frame)

Voice playout (jitter buffer) delay: (milliseconds)

Nominal Maximum

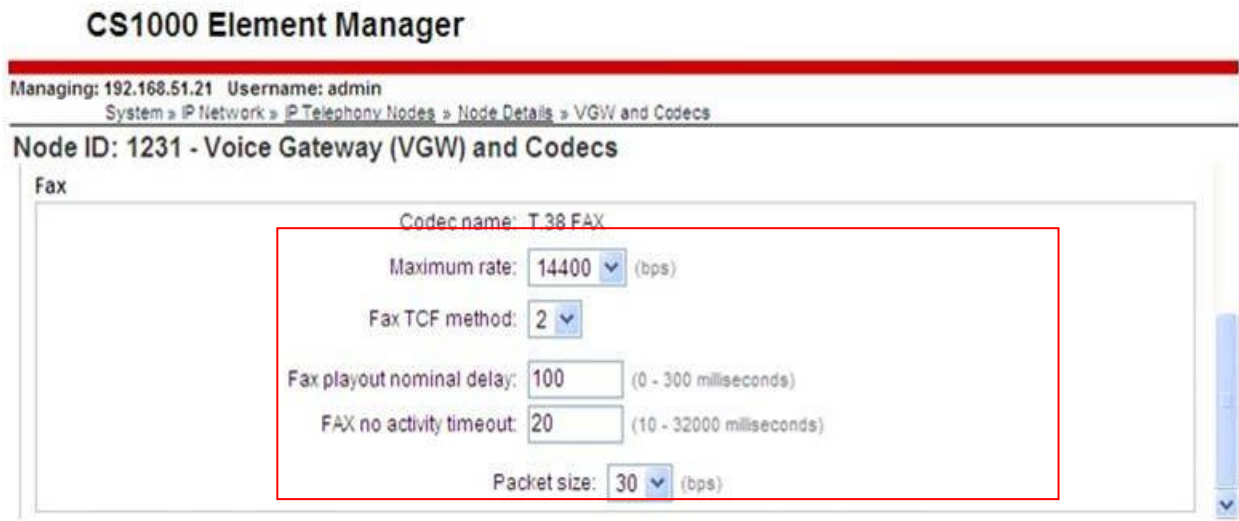
Maximum delay may be automatically adjusted based on nominal settings.

Maximum delay may be automatically adjusted based on nominal settings.

Voice Activity Detection (VAD)

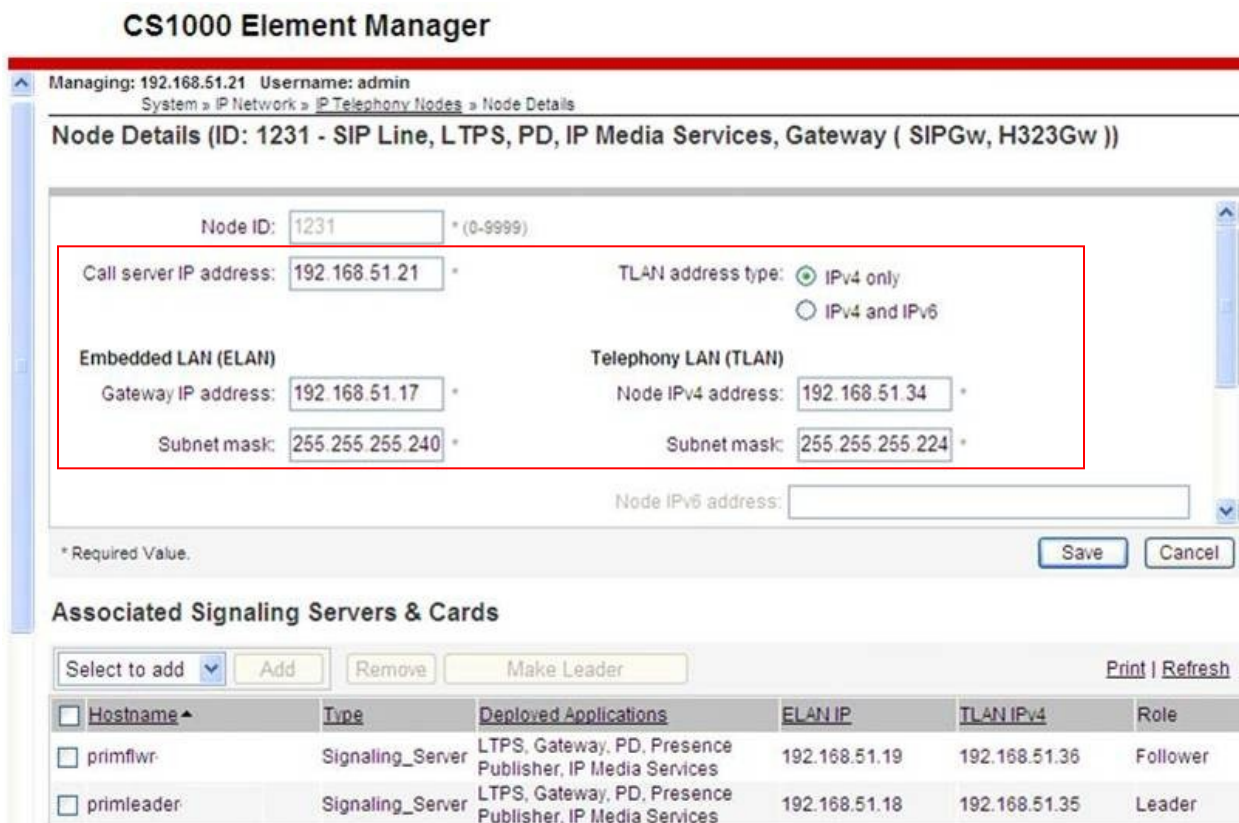
* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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



5.3. 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.



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

CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System > IP Network > IP Telephony Nodes > Node Details > Virtual Trunk Gateway Configuration

Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Vtrk gateway application: Enable gateway service on this node

General

Vtrk gateway application: SIPGw and H.323Gw
SIP domain name: umlab.local
Local SIP port: 5060 * (1 - 65535)
Gateway endpoint name: PRIM_SS_LEADER
Gateway password: *
H.323 ID: PRIM_SS_LEADER
Application node ID: 1231 * (0-9999)
Enable failsafe NRS:

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:
192.168.131.186
192.168.51.46
Remove

SIP Gateway Settings

TLS Security: Security Disabled
Port: 5061 (1 - 65535)
Number of byte re-negotiation: 0
Options: Client authentication
 X509 certificate authority

Direct SIP Route

Enforce Direct SIP Route to Microsoft Mediation Server
FQDN of Microsoft Mediation Server:
Port: (1 - 65535)
Transport protocol: TCP

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 192.168.131.186
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

Secondary TLAN IP address:

Secondary TLAN IP address: 0.0.0.0
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

CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Options: Support registration
 Secondary CDS proxy

Tertiary IP address:
Port: (1 - 65535)
Transport protocol:
Options: Support registration
 Tertiary CDS proxy

Proxy Server Route 2:

Primary TLAN IP address:
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"
Port: (1 - 65535)
Transport protocol:

Options: Registration not supported
 Primary CDS proxy

CLID Presentation:

Country code (CCC):
Area code: NPA in North America

Number translation: Strip: Prefix: CLID display format:

Subscriber (SN): <CCC><Area code><SN>
National (NN): <CCC><NN>
International: <International number>

SIP URI Map:

Public E.164 domain names	Private domain names
National: <input type="text" value="E164.Nat"/>	UDP: <input type="text" value="udp"/>
Subscriber: <input type="text" value="E164.Sub"/>	CDP: <input type="text" value="cdp.udp"/>
Special number: <input type="text" value="PublicSpecial"/>	Special number: <input type="text" value="PrivateSpecial"/>
Unknown: <input type="text" value="PublicUnknown"/>	Vacant number: <input type="text" value="PrivateUnknown"/>
	Unknown: <input type="text" value="UnknownUnknown"/>

SIP Gateway Services

SIP Converged Desktop: Enable CD service

Service DN: Used for making VTRK call from agent.
Converged telephone call forward DN:
RAN route for announce: (route number 0 - 511)
Wait time before RAN queue: (-1 - 32767 msec)
Timeout for ringing indication: (5 - 60 seconds)
Timeout for CD server: (1 - 30 seconds)
Timeout for non-CD server: (2 - 60 seconds)

CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

User information fields
Invite message for MO set: sip:convergeddesktop@umlab.local;nortelconverged=continueforce
Invite message for MV set: sip:convergeddesktop@umlab.local;nortelconverged=conditionalfork
Notify message for converged desktop: sip:convergeddesktop@umlab.local

SIP CTI Service: Enable CTI service TLS endpoints only

CTI settings
Customer number: 0
Maximum associations per DN: 1

Dial plan prefixes
National: 90
International: 900

International calls: Place as national
For calls within this country.

Location code call:
Special number:
Subscriber:

CTI CLID presentation
Dialing plan: CDP
Calling device URI format: phone-context=<SIP URI Map Entries>
Home location code: 750
Country code (CCC): 44
Area code: 113 NPA in North America

Number translation: Strip: Prefix: CLID display format
Subscriber (SN): 0 <CCC><Area code><SN>
National (NN): 0 <CCC><NN>
International: 0 <International number>

Microsoft Unified Messaging:
MWI application DN: 7400
MWI dialing plan: CDP
Options: Enable softkeys

Auto Attendant Service
Add Remove

Auto Number	Auto Number Use	Insert Number
<input type="checkbox"/>		

H.323 Gateway Settings
Primary gatekeeper (TLAN) IP address: 192.168.51.169
Alternate gatekeeper (TLAN) IP address:

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

5.4. 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.

CS1000 Element Manager Help | Logout

Managing: 192.168.51.21 Username: admin
System > IP Network > Zones > Bandwidth Zones

Bandwidth Zones

Zone *	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description
1 <input type="radio"/>	100000	BQ	100000	BQ	SHARED	MO	GR_PRIM
2 <input type="radio"/>	100000	BQ	100000	BB	SHARED	MO	GR_SEC
3 <input type="radio"/>	100000	BQ	10000	BB	SHARED	MO	SURV_MG1000
4 <input type="radio"/>	1000000	BQ	1000000	BQ	SHARED	VTRK	SIPLINEZONE
5 <input type="radio"/>	1000000	BQ	1000000	BB	SHARED	VTRK	SIP_VTRK_NOAS
6 <input type="radio"/>	100000	BQ	10000	BQ	SHARED	MO	VIRTUALSETS
7 <input type="radio"/>	100000	BQ	100000	BQ	SHARED	VTRK	VIRTUAL TRKS

5.5. Configure Incoming Digit Conversion Table

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

CS1000 Element Manager Help | Logout

Managing: 192.168.51.21 Username: admin
Dialing and Numbering Plans > Incoming Digit Translation > Customer 00 > Digit Conversion Tree 10 Configuration

Digit Conversion Tree 10 Configuration

Regular IDC tree
Send calling party DID disabled

	Incoming Digits *	Converted Digits	CPND Name	CPND language
1 <input type="radio"/>	0207960 [REDACTED]	52201		
2 <input type="radio"/>	0207960 [REDACTED]	52000		
3 <input type="radio"/>	0207960 [REDACTED]	52200		
4 <input type="radio"/>	0207960 [REDACTED]	52200		
5 <input type="radio"/>	0207960 [REDACTED]	52000		
6 <input type="radio"/>	0207960 [REDACTED]	52201		

5.6. Configure SIP Trunks

Communication Server 1000E virtual trunks will be used for all inbound and outbound PSTN calls to the BT 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 (SPNs); 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 50
  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
    OVLN 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 5.4** for **SIP_VTRK_NOAS**. The remaining highlighted values are important for correct SIP trunk operation.

<pre> 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 00253 PCID SIP CRID NO NODE 1231 DTRK NO ISDN YES MODE ISLD DCH 50 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 </pre>	<pre> ACOD 1600 TCPP NO PII NO AUXP NO TARG CLEN 1 BILN NO OABS INST IDC YES DCNO 10 NDNO 10 * 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 </pre>	<pre> 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 </pre>
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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 160 0 0 0
PAGE
DES VIR_TRK
TN 160 0 00 00 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 00253
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.

<pre> Overlay 86 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 CTBL 0 ISDM 0 </pre>	<pre> DMI 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 </pre>
---	---

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

5.7. 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 5.4** 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 00254
CUR_ZONE 00254
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 CCBD RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
FDSO NOVU 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 52000 0      MARP
      CPND
        CPND_LANG ROMAN
          NAME IP1140
          XPLN 10
          DISPLAY_FMT FIRST, LAST
01 MCR 52000 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 HBTB AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBF RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
FDSB 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 52001 0 MARP

CPND

CPND_LANG ROMAN

NAME Digital Set

XPLN 10

DISPLAY_FMT FIRST, LAST

01 MCR 52001 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

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

Overlay 20 - Analog Telephone Configuration

```

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

ERL 00000
WRLS NO
DN 52002
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 NRCN NROD SPKD CRD PRSD MCRD
EXR0 SHL SMSD ABDD CFHD DNDY DNO3
CWND USMD USRD CCBP BNRD OCBP 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

```

5.8. 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 78
NMME NO

```

If a numerical value is entered against the **UAPR** setting, this number will be prepended 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 6.5.1**. The IP address configured in **MO SLG IPv4 address** is the system **NODE IP address**, as previously configured in **Section 5.3**.

CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
 System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration

Node ID: 1231 - 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: Add

Monitor addresses:
 192.168.131.186
 192.168.51.46 Remove

SIP Line Gateway Settings

Security policy: Security Disabled

Number of byte re-negotiation: 0

Options: Client authentication
 x509 Certificate authentication enabled

SIP Line Gateway Service

Branch / GR Office Settings:

SLG role: MO

SLG mode: S1/S2

MO SLG IPv4 address: 192.168.51.34
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

MO SLG IPv6 address:

MO SLG port: 5070 (1 - 65535)

MO SLG transport: TCP

GR SLG IPv4 address: 0.0.0.0
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

GR SLG IPv6 address:

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

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 **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 52003
NDID 5
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 00004
CUR_ZONE 00004
ERL 0
ECL 0
VSIT NO
FDN
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW 52003
SFLT NO
CAC_MFC 0
CLS UNR FBD WTA LPR MTD FNA HTA TDD HFD CRPD
MWD LMPN RMD 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 CLBDAUTU
GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
```

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```
UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBD RTDD RBDD RBHD PGND 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 52003 0 MARP
    CPND
        CPND_LANG ROMAN
            NAME Sigma 1140
            XPLN 11
            DISPLAY_FMT FIRST, LAST*
01 HOT U 7852003 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
```


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® Session Manager
- Administer SIP domain
- Administer Locations
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Avaya Communication Server 1000E as Managed Element

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.

The screenshot displays the Avaya Aura System Manager 6.1 interface. At the top left is the AVAYA logo, followed by the text "Avaya Aura™ System Manager 6.1". On the top right, there are links for "Help | About | Change Password | Log off admin". Below these links are two tabs: "Session Manager" (with a close button 'x') and "Home" (which is the active tab). The main content area is divided into three vertical panels:

- Users:** Contains links for Administrators, Groups & Roles, Subscribers, Synchronize and Import, UCM Roles, and User Management.
- Elements:** Contains links for Application Management, Communication Manager, Conferencing, Inventory, Messaging, Presence, and Routing (highlighted with a red box). Below these are links for SIP AS 8.1 and Session Manager.
- Services:** Contains links for Backup and Restore, Configurations, Events, Licenses, Replication, Scheduler, Security, Templates, and UCM Services.

6.2. Administer SIP Domain

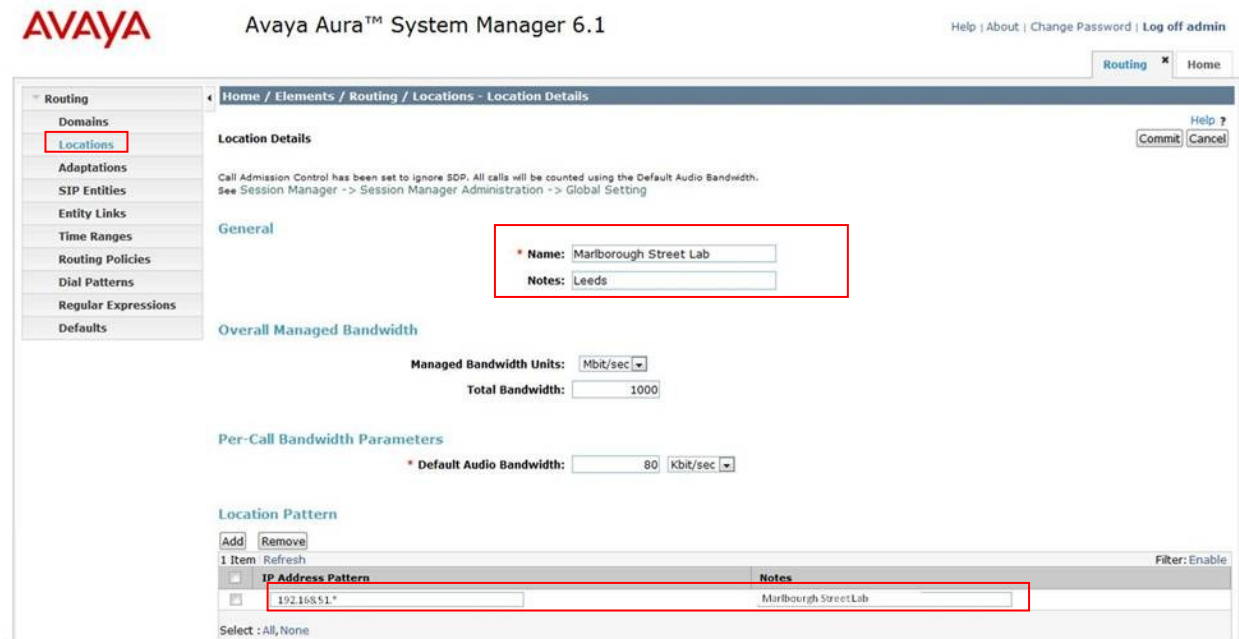
To add the SIP domain that will be used with Session Manager, select **Routing** from the Elements Home tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button (not shown) to create a new SIP domain entry. In the **Name** field, enter the domain

name (e.g., **umlab.local**) and optionally a description for the domain in the **Notes** field. Click **Commit** to save changes.



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 the enterprise SIP entities. Under 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 simulated Enterprise site.



6.4. Administer Adaptations

To ensure that the E.164 numbering format is used between the enterprise and BT SIP Trunk Service, an adaptation module is used to perform some digit manipulation. This adaptation is applied to the Communication Server 1000E SIP entity. To add an adaptation, under the **Routing** tab, select **Adaptations** on the left hand menu and then click on the **New** button (not shown).

Under **Adaption Details** → **General**:

- In the **Adaptation name** field enter an informative name.
- In the **Module name** field, click on the down arrow and then select the **<click to add module>** entry from the drop down list and type **CS1000Adapter** in the resulting New Module Name field.



The screenshot shows the Avaya Aura System Manager 6.1 interface. The left navigation pane has 'Adaptations' highlighted. The main content area is titled 'Adaptation Details' and shows the 'General' tab. The 'Adaptation name' field contains 'adapt_PRIM_SS_LEADER'. The 'Module name' dropdown menu is open, showing 'CS1000Adapter' selected. Below these fields are 'Module parameter:', 'Egress URI Parameters:', and 'Notes:' fields, all of which are currently empty.

Scroll down the page and under **Digit Conversion for Incoming Calls to SM**, click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the Matching Pattern field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so destination has been selected.

This will ensure any destination numbers received from Communication Server 1000E are converted to the E.164 numbering format before being processed by Session Manager. The following screenshot shows the settings used.

Digit Conversion for Incoming Calls to SM										
Add Remove										Filter: Enable
12 Items Refresh										
<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes		
<input type="checkbox"/>	*003	*3	*36	PrivateSpecia	*2	+	destination	Ireland IDD Code		
<input type="checkbox"/>	*0113	*4	*36	PrivateSpecia	*1	+44	destination	Leeds Area STD Code		
<input type="checkbox"/>	*0121	*4	*36	PrivateSpecia	*1	+44	destination	Birmingham Area STD Code		
<input type="checkbox"/>	*0131	*4	*36	PrivateSpecia	*1	+44	destination	Edinburgh Area STD Code		
<input type="checkbox"/>	*01903	*5	*36	PrivateSpecia	*1	+44	destination	Worthing Area STD Code		
<input type="checkbox"/>	*0191	*4	*36	PrivateSpecia	*1	+44	destination	Tyneside Area STD Code		
<input type="checkbox"/>	*020	*3	*36	PrivateSpecia	*1	+44	destination	London Area STD Code		
<input type="checkbox"/>	*05	*2	*36		*0	+	both	Type:E164 Local, special rule		
<input type="checkbox"/>	*07	*2	*36	PrivateSpecia	*1	+44	destination	UK Mobile Services		
<input type="checkbox"/>	*x	*1	*36	cdp.udp	*0	55	both	Type:Level 0 Regional, special rule		
<input type="checkbox"/>	*x	*1	*36	PrivateSpecia	*0	56	both	Type:Special, general rule		
<input type="checkbox"/>	*x	*1	*36	+1	*0	+1	both	Type:E164 National, special rule		

Under **Digit Conversion for Outgoing Calls from Session Manager** click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the **Matching Pattern** field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so destination has been selected.

This will ensure any destination numbers will have the + symbol and international dialing code removed before being presented to Communication Server 1000E. See the following screenshot for the settings used.

Digit Conversion for Outgoing Calls from SM										
Add Remove										Filter: Enable
3 Items Refresh										
<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes		
<input type="checkbox"/>	*#	*1	*36	udp	*0		both	Type:Level 1 Regional Entity:PRIM		
<input type="checkbox"/>	*+4420	*5	*36		*3	0	destination	IC BT NOAS Call translation		
<input type="checkbox"/>	*55	*2	*36	cdp.udp	*2		both	Type:Level 0 Regional Entity:PRIM		

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 (see the following screenshot) and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity.

Under **SIP Entity Details** → **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 signaling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **Other** for a Communication Server 1000E SIP entity and **Other** for the NOAS SBC Birm2 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 enterprise site configuration there are three SIP Entities configured.

- Session Manager SIP Entity
- Communication Server 1000E SIP Entity
- NOAS SBC Birm2 SIP Entity

6.5.1. Avaya Aura® Session Manager SIP Entity

The following two 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 signaling interface.

AVAYA Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

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

SIP Entity Details

General

* Name: Leeds SM 6.1

* FQDN or IP Address: 192.168.51.46

Type: Session Manager

Notes:

Location: Marlborough Street Lab

Outbound Proxy:

Time Zone: Europe/London

Credential name:

SIP Link Monitoring: Use Session Manager Configuration

The Session Manager must be configured with the port numbers of 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 **umlab.local** as the default domain.

The screenshot shows a configuration interface for SIP ports. At the top left, there is a 'Port' section with 'Add' and 'Remove' buttons. Below this, there is a table with 3 items and a 'Refresh' button. The table has columns for 'Port', 'Protocol', 'Default Domain', and 'Notes'. The first three columns of the first row are highlighted with a red box. The table contains the following data:

Port	Protocol	Default Domain	Notes
5060	TCP	umlab.local	
5060	UDP	umlab.local	
5061	TLS	umlab.local	

At the bottom left of the interface, there is a 'Select : All, None' option. At the top right, there is a 'Filter: Enable' option.

6.5.2. Avaya Communication Server 1000E SIP Entity

The following screenshot shows the SIP entity for Communication Server 1000E which is configured as **Type Other**. The **FQDN or IP Address** field is set to the Communication Server 1000E node IP address. For the **Adaptation** field, select the adaptation module previously defined for dial plan digit manipulation in **Section 6.4**.

The screenshot displays the Avaya Aura System Manager 6.1 interface. The left-hand navigation pane shows the 'SIP Entities' menu item highlighted with a red box. The main content area is titled 'SIP Entity Details' and shows the 'General' tab. Two red boxes highlight specific configuration fields: the first box encloses the 'Name' field (set to 'PRIM_SS_LEADER'), the 'FQDN or IP Address' field (set to '192.168.51.34'), and the 'Type' dropdown (set to 'Other'); the second box encloses the 'Adaptation' dropdown (set to 'adapt_PRIM_SS_LEADER'), the 'Location' dropdown, and the 'Time Zone' dropdown (set to 'Europe/London'). Other visible fields include 'Notes' (set to 'GR PRIME SITE'), 'SIP Timer B/F (in seconds)' (set to '4'), 'Call Detail Recording' (set to 'none'), and 'SIP Link Monitoring' (set to 'Link Monitoring Enabled').

6.5.3. BT NOAS Birmingham Node2 SIP Entity

The following screen shows the SIP Entity for the BT NOAS Birmingham Node2. The **FQDN or IP Address** field is set to the IP address of the NOAS SBC Birm2 public network interface (altered in this document for security reasons).

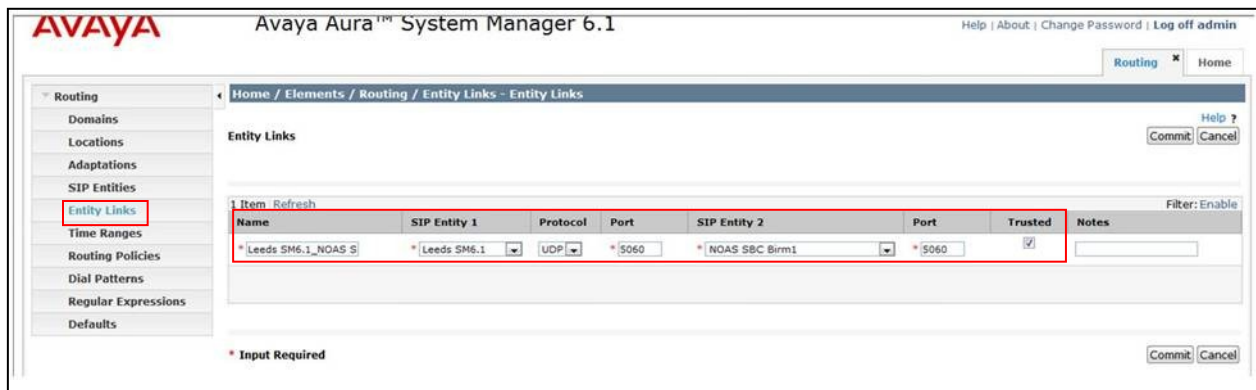
The screenshot displays the Avaya Aura System Manager 6.1 interface. The left-hand navigation pane shows the 'SIP Entities' menu item highlighted with a red box. The main content area is titled 'SIP Entity Details' and shows the 'General' tab. Two red boxes highlight specific configuration fields: the first box encloses the 'Name' field (set to 'NOAS SBC Birm2'), the 'FQDN or IP Address' field (set to 'xxx.vvv.113.62'), and the 'Type' dropdown (set to 'Other'); the second box encloses the 'Adaptation' dropdown, the 'Location' dropdown (set to 'NOAS SIP Service'), and the 'Time Zone' dropdown (set to 'Europe/London'). Other visible fields include 'Notes' (set to 'Primary SIP inbound / outbound ca'), 'SIP Timer B/F (in seconds)' (set to '4'), 'Call Detail Recording' (set to 'none'), and 'SIP Link Monitoring' (set to '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 **Protocol** field enter the transport protocol to be used to send SIP requests.
- 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.

Click **Commit** to save changes. The following screen shows an example Entity Link used in this configuration.



The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar has a menu with 'Entity Links' highlighted. The main content area shows the 'Entity Links' configuration page. A table with one row is displayed, with the following data:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* Leeds SM6.1_NOAS S	* Leeds SM6.1	UDP	* 5060	* NOAS SBC Birm1	* 5060	<input checked="" type="checkbox"/>	

Buttons for 'Commit' and 'Cancel' are visible at the top right and bottom right of the configuration area.

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 (see next screenshot) and then click on the **New** button (not shown).

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

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

The **SIP Entity as Destination** value is set to PRIM_SS_LEADER, as entered in **Section 6.5.2**.

The **Time of Day** is set to 24 hour by 7 day operation.

The screenshot displays the Avaya Aura System Manager 6.1 interface. The left sidebar shows the navigation menu with 'Routing Policies' highlighted. The main content area is titled 'Routing Policy Details' and includes sections for 'General', 'SIP Entity as Destination', and 'Time of Day'. The 'General' section shows the policy name 'Incoming to Leeds CS1000 Direct', a 'Disabled' checkbox, and a note 'Calls to Prim_SS_Leader'. The 'SIP Entity as Destination' section shows a table with one entry: PRIM_SS_LEADER, 192.168.51.34, Other, GR PRIME SITE. The 'Time of Day' section shows a table with one entry: 24/7, 00:00, 23:59, Time Range 24/7.

Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

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

Routing Policy Details

Help ?

Commit Cancel

General

* Name: Incoming to Leeds CS1000 Direct

Disabled:

Notes: Calls to Prim_SS_Leader

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
PRIM_SS_LEADER	192.168.51.34	Other	GR PRIME SITE

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
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

Select : All, None

The following screen shows the routing policy for BT SBC Birm2. A routing policy must be added for each NOAS node. Note the **Ranking** given to the time range in this routing policy is set to 10. Each NOAS node routing policy will have a different ranking; this is to define a priority order for the routing policies when they are added to a dial pattern in **Section 6.8**. The rankings are set in blocks of ten for clarity in this Application Note. Lower number means higher ranking.

Routing Policy Details

General

Name: SIP Trunk Calls to Birm2

Disabled:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
NOAS SBC Birm2	xxx.yyy.113.82	Other	Primary SIP inbound / outbound calls

Time of Day

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
10	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 (see below) and then click on the **New** button (not shown).

Under **Dial Pattern Details** → **General**:

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

Under **Originating Locations and Routing Policies**, click **Add**, in the resulting screen (not shown) under **Originating Location** select **ALL** and under **Routing Policies** select one of the routing policies defined in **Section 6.7**. Click **Select** button to save. The following screen shows an example dial pattern configured for BT SIP Trunk Service. Note the ranking for each routing policy as applied in **Section 6.7**. The routing policy with the lowest rank will be selected first, if this route is unavailable or does not respond then the routing policy with the next lowest rank will be selected and so on. This allows for redundant routing within Session Manager.

The screenshot displays the 'Dial Pattern Details' configuration page. The left sidebar shows a navigation menu with 'Dial Patterns' highlighted. The main content area is divided into two sections: 'General' and 'Originating Locations and Routing Policies'.

General Section:

- Pattern:** +44113
- Min:** 6
- Max:** 36
- Emergency Call:**
- SIP Domain:** -ALL-
- Notes:** Leeds PSTN Area Code via SIP Trunk

Originating Locations and Routing Policies Section:

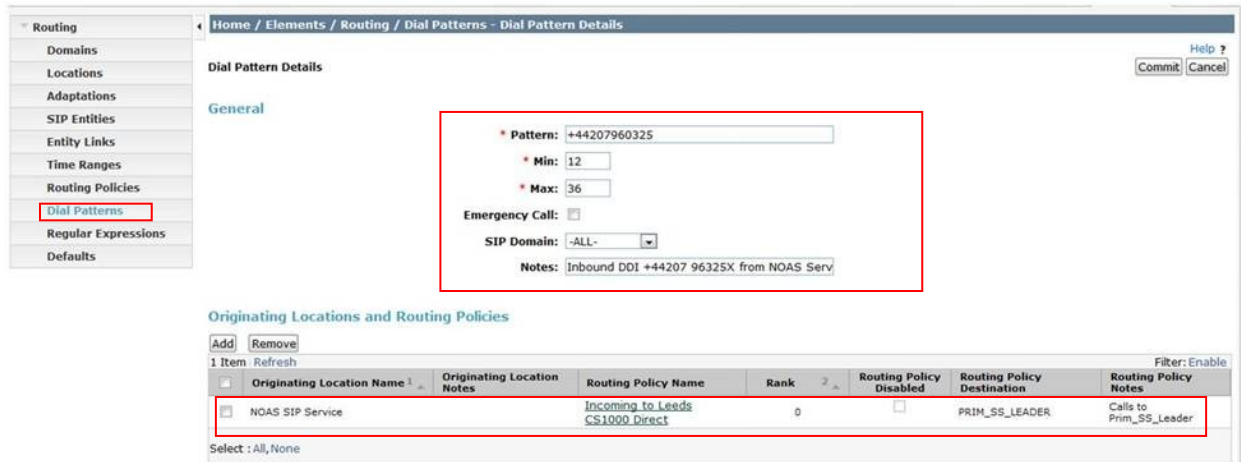
Buttons: Add, Remove

5 Items Refresh

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	SIP Calls to Romford Acme SBC	0	<input type="checkbox"/>	Romford SBC Acme 4500 net-net	
<input type="checkbox"/>	-ALL-	Any Locations	SIP Trunk Calls to Birm2	10	<input type="checkbox"/>	NOAS SBC Birm2	
<input type="checkbox"/>	-ALL-	Any Locations	SIP Trunk calls to Birm1	20	<input type="checkbox"/>	NOAS SBC Birm1	
<input type="checkbox"/>	-ALL-	Any Locations	SIP Trunk calls to Man2	30	<input type="checkbox"/>	NOAS SBC Man2	
<input type="checkbox"/>	-ALL-	Any Locations	SIP Trunk calls to Man1	40	<input type="checkbox"/>	NOAS SBC Man1	

Select : All, None

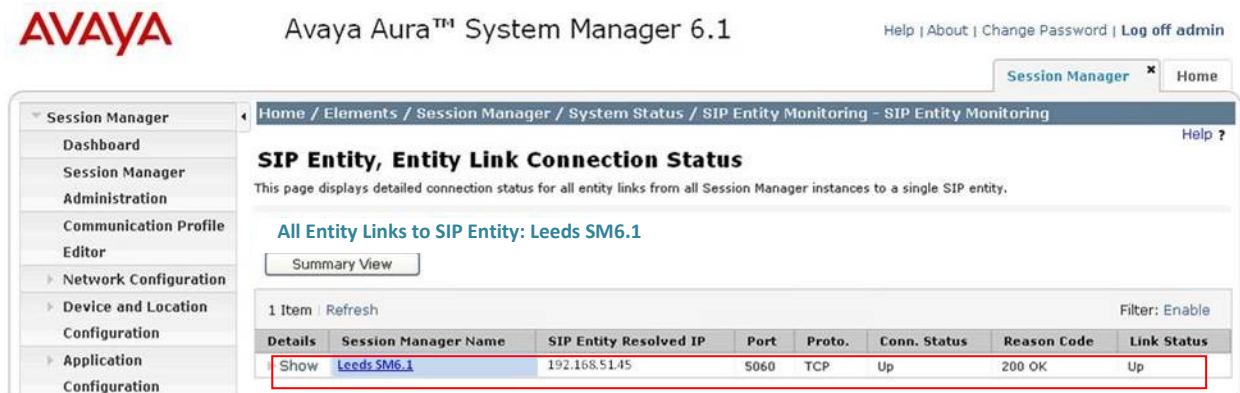
The following screen shows an example dial pattern configured for Communication Server 1000E.



7. Verification Steps

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

1. From System Manager **Home** Tab (see **Section 6.1**), 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. See the following for an example.



2. From the Communication Server 1000E system terminal; load overlay 32 and run the command 'stat vtrm <cust> <x>' where 'cust' is the customer number (usually 0) and 'x' is a previously configured SIP trunk route. Confirm all channels on the trunk group display idle registered.

```
stst vtrm 0 100

*****
STATUS OF VTRL IP TRUNK ROUTE AND MBRS
*****

=====
CUST ROUTE PROTOCOL CALL_DIRCTN
0 100 SIP IN AND OUT

DCH 50 SSRC TOTAL 2048 SSRC USED 77 SSRC AVAILABLE 1971

MBR STATUS

IDLE UNREGISTERED 0
IDLE REGISTERED 15
BUSY 0
MBSY 0
DSBL UNREGISTERED 0
DSBL REGISTERED 0
LCKO 0
```

- 3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
- 4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call remains active.
- 5. Verify that the user on the PSTN can end an active call by hanging up.
- 6. Verify that an endpoint at the enterprise site can end an active call by hanging up.

8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Communication Server 1000E and Avaya Aura® Session Manager to BT SIP Trunk Service. BT SIP Trunk Service is a SIP-based Voice over IP solution providing businesses with a flexible, cost-saving alternative to traditional hardwired telephony trunks.

9. 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, June 2010.
- [2] *Administering Avaya Aura® System Platform*, Release 6, June 2010.
- [3] *Avaya Communication Server 1000E Installation and Commissioning*, November 2010, Document Number NN43041-310.
- [4] *Feature Listing Reference Avaya Communication Server 1000*, November 2010, Document Number NN43001-111, 05.01.
- [5] *Installing and Upgrading Avaya Aura® System Manager Release 6.1*, November 2010.
- [6] *Installing and Configuring Avaya Aura® Session Manager*, January 2011, Document Number 03-603473
- [7] *Administering Avaya Aura® Session Manager*, March 2011, Document Number 03-603324.
- [8] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

Appendix A – Avaya Communication Server 1000 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:          4IPMGs Unregistered:          0IPMGs Configured/unregistered:
2
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 4021
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 02 (created: 2010-11-30 15:12:45 (est))

IN-SERVICE PEPS
PAT# CR #          PATCH REF #      NAME      DATE      FILENAME      SPECINS
000 wi00832106      ISS1:1OF1       p30550_1  14/12/2010 p30550_1.cpm  NO
001 wi00835093      ISS1:1OF1       p30553_1  14/12/2010 p30553_1.cpm  YES
002 wi00832626      ISS2:1OF1       p30560_2  14/12/2010 p30560_2.cpm  NO
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)
```

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: SS_EM

Packages: SS+EM

Configuration version: 7.50.17-00

dbcom	7.50.17	
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	
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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