

Avaya Solution & Interoperability Test Lab

# Application Notes for Configuring Avaya Communication Server 1000E R7.5 with Avaya Aura<sup>®</sup> 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<sup>®</sup> 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<sup>®</sup> 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 preconfigured 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

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## 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	CSP Version: MGCC CD01
Media Gateway	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
                                                                      1
IPMGs Registered:
IPMGs Unregistered:
                                                                      Ω
IPMGs Configured/unregistered: 0
TRADITIONAL TELEPHONES 32767 LEFT 32766 USED
                                                                                                                      1
DECT USERS 32767 LEFT 32767 USED
                                                                                                                        0

        IP USERS
        32767
        LEFT 32766
        USED
        1

        BASIC IP USERS
        32767
        LEFT 32766
        USED
        1

        TEMPORARY IP USERS
        32767
        LEFT 32767
        USED
        0

        DECT VISITOR USER
        10000
        LEFT 10000
        USED
        0

        32767
        LEFT 32752
        USED
        15

IP USERS
                                                32767 LEFT 32744 USED 23
ACD AGENTS32767LEFT 10000USED0ACD AGENTS32767LEFT 32752USED15MOBILE EXTENSIONS32767LEFT 32767USED0TELEPHONY SERVICES32767LEFT 32767USED0CONVERGED MOBILE USERS32767LEFT 32767USED0NORTEL SIP LINES32767LEFT 32765USED2THIRD PARTY SIP LINES32767LEFT 32761USED6
SIP CONVERGED DESKTOPS 32767 LEFT 32767
                                                                                                    USED
                                                                                                                        0

        SIP CTI TR87
        32767
        LEFT 32767
        USED

        SIP ACCESS PORTS
        32767
        LEFT 32752
        USED
        15

                                                                                                                        0
```

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**  $\rightarrow$  **IP Telephony Nodes**  $\rightarrow$  **Node Details**  $\rightarrow$  **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		
lanaging: 192.168.51.21 Username: admin System » IP Network » IP Telephony Nodes » Node I	etails » VGW and Codecs	
lode ID: 1231 - Voice Gateway (VGW) and	d 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 Holse level.	-00 (-027 - +027 0.00m)	
Signaling options	DTMF tone detection	
	Low latency mode	
	Remove DTMF delay (squeich DTMF from TDM to IP)	
	Modem/Fax pass-through	
	V 21 Fax tone detection	
l l		

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

CS1	000	Eleme	ent M	lana	aer
					9

Managing: 192.168.51.21 System > IP Net	Username: admin work » IP Telephony Nodes » Node De	etails > VGW and Codecs	
Node ID: 1231 - V	oice Gateway (VGVV) and	Codecs	
General   Voice Code	is   Fax		
	Codec G711:	Enabled (required)	
		Maximum delay may be automatically adjust settings.	ed based on nominal
		Voice Activity Detection (VAD)	
	Codec G722:	Enabled	
	Voice paylo	ad size: 20 v imiliseconds per frame)	
	Voice playout (jitter buffe	r) delay: 40 😵 80 😪 (milliseconds)	
		Nominal Maximum	
		Maximum delay may be automatically adjust settings.	ed based on nominal
			X
		Maximum delay may be automatically adjust	ed based on nominal
		settings.	
		Voice Activity Detection (VAD)	
* Required Value.	Note: Chang transmitte	es made on this page will NOT be ed until the Node is also saved.	Save Cancel

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

### CS1000 Element Manager

Node ID: 1231	- Voice Gateway (VGW) and Codecs	
Fax		
	Codec name: T.38 FAX	
	Maximum rate: 14400 🛩 (bps)	
	Fax TCF method: 2 💌	
	Fax playout nominal delay: 100 (0 - 300 milliseconds)	
	FAX no activity timeout: 20 (10 - 32000 milliseconds)	
	Packet size: 30 V (bos)	

### 5.3. Virtual Trunk Gateway Configuration

Use Communication Server 1000E Element Manager to configure the system node properties. Navigate to the System  $\rightarrow$  IP Networks  $\rightarrow$  IP Telephony Nodes  $\rightarrow$  Node Details and verify the highlighted section is completed with the correct IP addresses and subnet masks.

#### CS1000 Element Manager

	er en Ellio, E		cutonay ( on	011,1102000	n
Node ID:	1231	(0-9999)			
Call server IP address:	192.168.51.21	TLAN address type	<ul> <li>IPv4 only</li> <li>IPv4 and IPv6</li> </ul>		
Embedded LAN (ELAN)		Telephony LAN (TLAN	)		
Gateway IP address:	192.168.51.17	Node IPv4 address	192.168.51.34	] •	
Subnet mask:	255.255.255.240	Subnet mask	255.255.255.224		
		Node IPv6 address:			
* Required Value.				Saw	e Cance
Associated Signalin	g Servers & Ca	ırds			
Select to add 💌 🗛	d Remove	Make Leader			Print   Refre
Hostname +	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
primflwr-	Signaling_Serve	LTPS, Gateway, PD, Presence Publisher, IP Media Services	192.168.51.19	192.168.51.36	Follower

Solution & Interoperability Test Lab Application Notes ©2011 Avaya Inc. All Rights Reserved. The next three screenshots show the SIP Virtual Trunk Gateway configuration, navigate to System  $\rightarrow$  IP Networks  $\rightarrow$  IP Telephony Nodes  $\rightarrow$  Node Details  $\rightarrow$  Virtual Trunk Configuration Details and fill in the highlighted areas with the relevant settings.

	al Trunk Gateway Con	figuration Details	
eneral   SIP Gateway Se	ttings   SIP Gateway Services	H.323 Gateway Settings	
	Vtrk gateway application: 🔽	Enable gateway service on this node	100
_			
nerai		Virtual Trunk Network Health Monitor	
Vtrk gateway applic	ation: SIPGw and H.323Gw	Monitor IP addresses (listed below)	
SIP domain n	ame: umlab.local	Information will be captured for the IP addresses listed	
Local SIP	port: 5060 * (1 -	65535) below.	
Gateway endpoint n	PRIM SS LEADER	Monitor IP: Add	
Gateriay endpoint in		Monitor addresses:	
Gateway pass	word	192 168 131 186 192 168 51 46	
H.32	23 ID: PRIM_SS_LEADER	Remove	
Application nor	te ID: 1231 *(0.5	9999)	
, applied ton not			
Enable failsafe	NRS:		
Gateway Settings			
rect SIP Route	of Microsoft Mediation Server.	Cheff addrend addrend X509 certificate authority Center Direct SIP Route to Microsoft Mediation Server	
	Port	(1 - 65535)	
	Transport protocol:		
oxy Or Redirect Server: Proxy Server Boute	• 1:		
rish server hour	Primary TLAN IP address:	192.168.131.186	
	T	he IP address can have either IPv4 or IPv6 format based on the value of "TLAN ddress type"	
	Port	5060 (1 . 85535)	
	FOIL	(1+05555)	
	-		
	Transport protocol:		
	Transport protocol: [ Options: [	TCP Support registration Primary CDS proxy	
	Transport protocol: [ Options: [ [	TCP  Support registration Primary CDS proxy	
	Transport protocol: Options: [ Secondary TLAN IP address: [ Ti a	TCP  Support registration Primary CDS proxy 0.0.0 he IP address can have either IPv4 or IPv6 format based on the value of "TLAN ddress type"	
	Transport protocol: [ Options: [ Secondary TLAN IP address: [ TI ar	TCP       Support registration         Primary CDS proxy         0.0.0         he IP address can have either IPv4 or IPv6 format based on the value of "TLAN ddress type"         5060       (1 - 65535)	
	Transport protocol: Options: Secondary TLAN IP address: Ti a Port	TCP  Support registration Primary CDS proxy 0.0.0 he IP address can have either IPv4 or IPv6 format based on the value of "TLAN ddress type" 5060 (1 - 65535) TCP	

CS1000 Element Manager

#### CS1000 Element Manager

Managing: 192.168.51.21 Username: admin System » P Network » P Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General   SIP	<sup>o</sup> Gateway Settings	SIP Gateway Services	H.323 Ga	teway Settin	as		
		Options:	Suppo	nt registratio	n		
			Secon	dary CDS pr	oxy		
		Tertiary IP address:	192.168	51.169			
		Port:	5060	(1	65535)		
		Transport protocol:	TCP 👻	1			
		Options:	Suppo	rt registratio	n		
			Tertiar	y CDS proxy			
Proxy	Server Route 2:	imary TI AN IP address:	192 168	131 186			
		indiy restriction address.	The IP addre address typ	ess can have	either IPv4 or IPv6 form	nat based on the value of "TLAN	
		Port	5060	(1	65535)		
		Transport protocol:	TCP 🗸				
		Options:	Regis	tration not su	upported		
			Prima	ry CDS proxy	r.		
CLID Presen	tation:						
		Country code (CCC):	44				
		Area code:	113	NPA in Nor	th America		
		Number translation:	Strip:	Prefix	CLID display form	at	
		Subscriber (SN):	0		<ccc>-<area cod<="" td=""/><td>e&gt;<sn></sn></td><td></td></ccc>	e> <sn></sn>	
		National (NN):	0	1	<ccc><nn></nn></ccc>		
		International:	0		<pre></pre>	nber>	
SIP URI Map:	8						
	Public E.164	domain names			Private do	main names	
	National:	E164 Nat			UDP:	udp	
	Subscriber:	E164.Sub	ľ		CDP:	cdp.udp	
	Special number:	PublicSpecial			Special number:	PrivateSpecial	
	Unknown:	PublicUnknown			Vacant number:	PrivateUnknown	
					Unknown:	UnknownUnknown	
IP Gateway	Services						
SIP Converg	ed Desktop: 🔲 En	able CD service					
		Service DN:		Use	ed for making VTRK ca	Il from agent.	
	Converged tele	phone call forward DN:					
	R	AN route for announce:	1	(roi	ute number 0 - 511)		
	Wait ti	me before RAN queue:	1	(-1	- 32767 msec)		
	Timeo	ut for ringing indication:	10	(5 -	60 seconds)		
		Timeout for CD server:	5	(1 -	30 seconds)		
	Tim	eout for non-CD server:	2	(2 -	60 seconds)		

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#### CS1000 Element Manager

the second s		
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

			5. C				1
	User Invite me Invite me Notify message for c	information fields essage for MO set essage for MV set onverged desktop:	sip:conve sip:conve sip:conve	ergeddeskto ergeddeskto ergeddeskto	pp@umlab.local;nor pp@umlab.local;nor pp@umlab.local	telconverged=continueforce telconverged=conditionalfork	
SIP CTI Se	ervice: 🔽 Enable CTI servi	ce					
			TLS er	ndpoints onl	У		
	CTI setting	gs			Dial pla	an prefixes	
	Customer number.	0	]		National	: 90	
Maxii	mum associations per DN	1 💌			International	: 900	
	International calls	Place as natio	nal		Location code call	2	
		For calls within th	is country.		Special number		
					Cubesciber		
					Subscriber	·	
	CTI CLID prese	ntation	000				
	0.11	Dialing plan	CDP V	antaute (O)	DUDIMA Card		1
	Calling d	evice UKI format:	phone-c	ontext= <si< td=""><td>P URI Map Entries</td><td>&gt; 📉</td><td></td></si<>	P URI Map Entries	> 📉	
	He	ome location code:	/50				
	Co	ountry code (CCC):	44	_			
		Area code:	113	NPA in Nor	th America		
	N	umber translation:	Strip:	Prefix	CLID display form	nat	
		Subscriber (SN):	0		<ccc><area cod<="" th=""/><th>le&gt;<sn></sn></th><th></th></ccc>	le> <sn></sn>	
		National (NN):	0		<ccc><nn></nn></ccc>		
		International:	0		<international nui<="" td=""><td>mber&gt;</td><td></td></international>	mber>	
icrosoft	Unified Messaging:	M application DN:	7400				
		MAR dialian plan	000	1			
		Options:	CDP V	e e officeure			
to Attor	adapt Sanuisa	Contraction of the second seco	1 I Ellapie	e somers			-1
Add.							-
Add		dude Mumb			law at	hank as	
	Auto Number	Auto Numb	erUse		Insertio	lumper	

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## 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  $\rightarrow$  IP Network  $\rightarrow$  Zones  $\rightarrow$  Bandwidth Zones and add new zones as required.

CS100	CS1000 Element Manager							
Managing: <u>192.168.51.</u> System » IP I	<u>21</u> Username: admin Network » <u>Zones</u> » Bandwidth 2	Cones						
Bandwidth 2	Zones						1	
Add Edit	Import Export	Maintenance	Delete				Refresh	
Zone +	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description	
101	100000	BQ	100000	BQ	SHARED	MO	GR_PRIM	
202	100000	BQ	100000	BB	SHARED	MO	GR_SEC	
3 🔾 3	100000	BQ	10000	88	SHARED	MO	SURV_MG1000	
404	1000000	BQ	1000000	BQ	SHARED	VTRK	SIPLINEZONE	
5 () 253	1000000	BQ	1000000	BB	SHARED	VTRK	SIP_VTRK_NOAS	
6 () 254	100000	BQ	10000	BQ	SHARED	MO	VIRTUALSETS	
7 () 255	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	CS1000 Element Manager							
Managing: <u>192.168.51.21</u> Username: ad Dialing and Numbering Plans » <u>b</u>	imin ncoming Digit Translation » Custome	r 00 » Digit Conversion Tree 10 Configuration						
Digit Conversion Tree	e 10 Configuration							
Regular IDC tree								
Send calling party DID disabled								
Add Delete IDC	Delete IDC tree			Refresh				
Incomina Diaits +	Converted Digits	CPND Name	CPND language					
1 0 0207960	52201							
2 0 0207960	52000							
3 0 0207960	52200							
4 0 0207960	52200							
5 O 0207960	52000							
6 O 0207960	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.

<b>Overlay</b>	17
ADAN	DCH 50
СТҮР	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	
OVI	LR NO
OVI	LS 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.

Overlay 16		
TYPE: rdb	ACOD 1600	CPDC NO
CUST 00	TCPP NO	DLTN NO
ROUT 100	PII NO	HOLD 02 02 40
TYPE RDB	AUXP NO	SEIZ 02 02
CUST 00	TARG	SVFL 02 02
ROUT 100	CLEN 1	DRNG NO
DES VIR TRK	BILN NO	CDR NO
	OABS	NATL YES
NPTD TRL NUM ()	INST	SSL
ESN NO	TDC VES	CFWR NO
RPA NO	DCNO 10	TDOP NO
CNUT NO	NDNO 10 *	VRAT NO
	DEXT NO	MUS VES
DOLO EVE	DIAI NO	MRT 21
	SICO STD	PANS YES
	STGO STD	RACD NO
20NE 00253	MEC NO	MANO NO
	TOTO VEO	FRI. 0.0
CRID NO	ICIS IES	FRE 0 0
NODE 1231	UGIS IES	FRL I U
DTRK NO	TIMR ICF 1920	FRL 2 U
ISDN YES	OGF 1920	FRL SU
MODE ISLD	EOD 13952	FRL 4 0
DCH 50	LCT 256	FRL 5 U
IFC SL1	DSI 34944	FRL 6 U
PNI 00001	NRD 10112	FRL / U
NCNA YES	DDL 70	OHQ NO
NCRD YES	ODT 4096	OHQT UU
TRO NO	RGV 640	CBQ NO
FALT NO	GTO 896	AUTH NO
CTYP UKWN	GTI 896	TTBL 0
INAC NO	SFB 3	ATAN NO
ISAR NO	PRPS 800	OHTD NO
DAPC NO	NBS 2048	PLEV 2
MBXR NO	NBL 4096	OPR NO
MBXOT NPA	IENB 5	ALRM NO
MBXT 0	TFD 0	ART 0
PTYP ATT	VSS 0	PECL NO
CNDP UKWN	VGD 6	DCTI 0
AUTO NO	EESD 1024	TIDY 1600 100
DNIS NO	SST 5 0	ATRR NO
DCDR NO	DTD NO	TRRL NO
ICOG IAO	SCDT NO	SGRP 0
SRCH LIN	2 DT NO	ARDN NO
TRMB YES	NEDC ORG	CTBL 0
STEP	FEDC ORG	AACR NO

Next, configure virtual trunk members using the Communication Server 1000E system terminal and overlay 14. Configure sufficient trunk members to carry both incoming and outgoing PSTN calls. The following example shows a single SIP trunk member configuration. Load overlay 14 at the system terminal and type **new X**, where X is the required number of trunks. Continue entering data until the overlay exits. The **RTMB** value is a combination of the **ROUT** value entered in the previous step and the first trunk member (usually 1). The remaining highlighted values are important for correct SIP trunk operation.

Overlay 14
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.

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

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

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

### 5.7. 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 RCBD
     ICDA CDMD LLCN MCTD CLBD AUTR
     GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTA AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXRO
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA PKCH MUTA MWTD
---continued on next page----
```

```
---continued from previous page----
DVLD CROD CROD
CPND_LANG ENG
RCO 0
hunt 0
LHK O
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
```

Overlay 20 - Digital Set configuration TYPE: 3904 DES 3904 TN 000 0 09 08 VIRTUAL TYPE 3904 CDEN 8D CTYP XDLC CUST 0 MRT ERL 0 FDN 0 TGAR 0 LDN NO NCOS 0 SGRP 0 RNPG 1 SCI 0 SSU LNRS 16 XLST SCPW SFLT NO CAC MFC 0 CLS UNR FBD WTA LPR PUA MTD FND HTD TDD HFA GRLD CRPA STSD MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1 POD SLKD CCSD SWD LNA CNDA CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD ICDA CDMA LLCN MCTD CLBD AUTU GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD DRDD EXRO USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN FDSD NOVD CDMR PRED RECA MCDD T87D SBMD PKCH CROD CROD CPND LANG ENG RCO 0 HUNT PLEV 02 PUID DANI NO SPID NONE AST IAPG 1 AACS ACQ ASID SFNB SFRB USFB CALB FCTB ITNA NO DGRP PRI 01 MLWU LANG 0 ---continued on next page----

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.

```
---continued from previous page----
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 NRCD NROD SPKD CRD PRSD MCRD
    EXRO SHL SMSD ABDD CFHD DNDY DNO3
     CWND USMD USRD CCBD BNRD OCBD RTDD RBDD RBHD FAXA CNUD CNAD PGND FTTC
    FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTD
PLEV 02
PUTD
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  $\rightarrow$  IP Telephony Nodes  $\rightarrow$  Node Details  $\rightarrow$  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.

eneral   SIP Line Gateway	Settings   SIP Line Gatewa	av Service		
SIP	Line Gateway Application:	🗹 Enab	le gateway service on this node	
neral			Virtual Trunk Network Health Mon	nitor
SIP domain name:	umlab.local *		Monitor IP addresses (liste	d below)
			Information will be captured	d for the IP addresses listed
SLG endpoint name:			below.	
SLG Group ID:			Monitor IP:	Add
			Monitor addresses:	
SLG Local Sip port	5070 (1 - 655	35)	192.168.131.186	
SLG Local TIs port	<b>5071</b> (1 - 655	35)	192.168.51.46	Remove
P Line Gateway Settings				
	Security policy:	Securit	v Disabled 💌	
Nun	ber of byte re-negotiation:	0		
	Options:	Clien	t authentication	
		x509	Certificate authentication enabled	
Line Gateway Service				
anch / GR Office Settings:		Francisco and	1	
	SLG role:	MO 💌		
	SLG mode:	S1/S2	~	
	MO SLG IPv4 address:	192.168	1.51.34	
		address ty	r <del>ess can have other inve or involf</del> ormat 'pe"	based on the value of "ILAN
	MO SLG IPv6 address:			
	MO SLG port	5070	(1 - 65535)	
	MO SI G transport	TCP		
	GR SI G IPut address:	0.0.0.0	2)	
	GR 020 II 14 8001855.	The IP add	ress can have either IPv4 or IPv6 format	based on the value of 'TLAN
		a second second		

#### CS1000 Element Manager

## 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
    096 0 01 15 VIRTUAL
TN
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
    UNR FBD WTA LPR MTD FNA HTA TDD HFD CRPD
CLS
     MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LND CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
```

---continued from previous page---UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD DRDD EXR0 USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD MSNV FRA PKCH MWTD DVLD CROD CROD CPND\_LANG ENG RCO 0 HUNT LHK 0 PLEV 02 PUID DANI NO AST IAPG 0 \* AACS NO ITNA NO DGRP MLWU LANG 0 MLNG ENG DNDR 0 **KEY 00 MCR 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<sup>®</sup> 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<sup>®</sup> 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.



## 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

GOR; Reviewed: SPOC 12/15/2011 Solution & Interoperability Test Lab Application Notes ©2011 Avaya Inc. All Rights Reserved. name (e.g., **umlab.local**) and optionally a description for the domain in the **Notes** field. Click **Commit** to save changes.

AVAYA	Avaya Aura™ S	ystem Manager 6	.1		Help   About   Char	nge Password   Log off admin
						Routing * Home
* Routing	Home / Elements / Routing /	/ Domains - Domain Manage	ment			
Domains						Help 1
Locations	Domain Management					Commit Cance
Adaptations						
SIP Entities						
Entity Links	1 Item Refresh		108000	10000000		Filter: Enable
Time Ranges	Name		Туре	Default	Notes	
Routing Policies	umlab.local		sip +		Avaya Blue CSLabs SIP Domain	
Dial Patterns						
Regular Expressions						Commit Cance
Defaults						

## 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.

AVALYA	Avaya Aura System Manager 6.1	Help   About   Change Password   Log off admi
		Routing * Home
Routing	Home / Elements / Routing / Locations - Location Details	
Domains		Help
Locations	Location Details	Commit Cano
Adaptations		
SIP Entities	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	
Entity Links		
Time Ranges	General	
Routing Policies	* Name: Marlborough Street Lab	
Dial Patterns	Notes: Leeds	
Regular Expressions	Sector Provide Independent	
Defaults	Overall Managed Bandwidth	
	Managed Bandwidth Units: Mbit/sec 👻	
	Total Bandwidth: 1000	
	Per-Call Bandwidth Parameters	
	* Default Audio Bandwidth: 80 Kbit/sec 💌	
	Location Pattern	
	Add Remove	
	1 Item Refresh	Filter: Enab
	IP Address Pattern Notes	

## 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.

AVAYA	Avaya Aura™ System Manager 6.1	Help   About   Change Password   Log off admin
✓ Routing	Home / Elements / Routing / Adaptations - Adaptation Details	Routing × Home
Domains Locations Adoptations SIP Entities Entity Links Time Ranges Routing Policies Dial Patterns	Adaptation Details General Adaptation name: adapt_PRIM_SS_LEADER Module name: CS1000Adapter Module parameter: Egress URI Parameters:	Helo ? Commit Cancel
Regular Expressions Defaults	Notes:	

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.

Add	d Remove										
2 Ite	ms Refresh							14 C	Filter: En		
	Matching Pattern	-	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes		
2	* 003		* 3	* 36	PrivateSpecia	• 2	+	destination 💌	Ireland IDD Code		
E1	* 0113		* 4	* 36	PrivateSpecia	• 1	+44	destination 💌	Leeds Area STD Code		
B	* 0121		* 4	* 36	PrivateSpecia	• 1	+44	destination 💌	Birmingham Area STD Code		
E	* 0131		* 4	* 36	PrivateSpecia	• 1	+44	destination 💌	Edinburgh Area STD Code		
凹	* 01903		* 5	* 36	PrivateSpecia	• 1	+44	destination 💌	Worthing Area STD Code		
E	* 0191		+ 4	* 36	PrivateSpecia	• 1	+44	destination 💌	Tyneside Area STD Code		
8	* 020		* 3	• 36	PrivateSpecia	• 1	+44	destination 💌	London Area STD Code		
63	* 05		* 2	* 36		• 0	+	both 💌	Type:E164 Local, special rule		
8	* 07		* 2	* 36	PrivateSpecia	• 1	+44	destination 💌	UK Mobile Services		
0	* ×		* 1	* 36	cdp.udp	• o	55	both 💌	Type:Level 0 Regional, special rule		
	* x		* 1	* 36	PrivateSpecia	• 0	56	both 💌	Type:Special, general rule		
13	* ×		* 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.

Add	Remove								
3 Item	s Refresh				Q				Filter: Er
	Matching Pattern	-	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
23	• #		* 1	* 36	udp	* 0		both 💌	Type:Level 1 Regional Entity:PRIM
8	* +4420		* 5	* 36		* 3	0	destination 💌	IC BT NOAS Call translation
171	* 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<sup>®</sup> 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.

Routing	Home / Elements / Rout	ting / SIP Entities - SIP Entity I	Petails			
Domains						Help
Locations	SIP Entity Details				Comm	ut Can
Adaptations	General					
SIP Entities		* Name:	Leeds SM 6.1			
Entity Links		* FQDN or IP Address:	192.168.51.46			
Time Ranges		Type:	Session Manager 🐳			
Routing Policies		Notes	1			
Dial Patterns		notes.				
<b>Regular Expressions</b>		Location:	Marlbourgh Street Lab	a		
Defaults		Outbound Prover				
		Time Zone:	Europe/London			
		Credential name:		. Second		

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.

Iten	ns Refresh				Filter: Enabl
	Port	Protocol	Default Domain	Notes	
ř.	5060	TCP	umlab.local 💌		
3	5060	UDP -	umlab.local		
2	5061	TLS -	umlab.local		

### 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**.

Routing	Home / Elements / Routing / SIP Entities - SIP Entity	Details	
Domains			He
Locations	SIP Entity Details		Commit Ca
Adaptations	General		
SIP Entities	* Name:	PRIM_SS_LEADER	
Entity Links	* FQDN or IP Address:	192.168.51.34	
Time Ranges	Type:	Other	
Routing Policies	Notes	CP DRIME SITE	
Dial Patterns	notes.	SK PADIE STE	
Regular Expressions	Adaptation:	adapt PRIM SS LEADER	
Defaults	Location		
	Time Zone	Europed and an	
	Querride Bert & Transport with DNE SBV		
	overlide Port a multiport with Didd sky.		
	* SIP Timer B/F (in seconds):	4	
	Credential name:		
	Call Detail Recording:	none 💌	

### 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).

	Home / Elements / Routing / SIP Entities - SIP Entity Details	
Domains Locations	SIP Entity Details	He Commit) Ca
Adaptations SIP Entities	General  Name: NOAS SBC Birm2	
Entity Links Time Ranges	* FQDN or IP Address: xxx.yvy.113.62	
Routing Policies Dial Patterns	Notes: Primary SIP inbound / outbound ca	
Regular Expressions Defaults	Adaptation:	
	Time Zone: Europe/London	
	Override Port & Transport with DNS SRV:	
	Credential name:	
	Call Detail Recording: none 💌	

## 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.

AVAYA	Avaya Aura'	System Man	ager 6	.1		Help	About ( Cha	nge Password	Log off admi
								Rout	ing × Hom
Routing	Home / Elements / Rou	iting / Entity Links - En	tity Links						
Domains									Help
Locations	Entity Links								Commit Can
Adaptations									
SIP Entities									
Entity Links	1 Item Refresh		har and the second			 Lange and the second			Filter: Ena
Time Ranges	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	 Port	Trusted	Notes	
Routing Policies	* Leeds SM6.1_NOAS S	* Leeds SM6.1	UDP 💌	* 5060	NOAS SBC Birm1	* \$060	N.		
Dial Patterns									
Regular Expressions									
Defaults									
	. Input Paguirad								(Course)

## 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.

					Routing * Hom
Routing	Home / Elements / Routing / Routing Police	cles - Routing Policy Details			
Domains Locations	Routing Policy Details				Commit) Can
Adaptations	General				
SIP Entities		Names Incoming to Londs (\$1000 Direct			
Entity Links		- Name: Incoming to Leeds CS1000 Direct			
Time Ranges		Disabled:			
Routing Policies		Notes: Calls to Prim_SS_Leader			
Dial Dattages					
Dial Patterns					
Regular Expressions	SIP Entity as Destination				
Regular Expressions Defaults	SIP Entity as Destination				_
Regular Expressions Defaults	SIP Entity as Destination Select Name	FQDN or IP Address	Туре	Notes	1
Regular Expressions Defaults	SIP Entity as Destination Select Name PRIM_SS_LEADER	FQDN or IP Address 192.168.51.34	Type Other	Notes GR PRIME SITE	
Defaults	SIP Entity as Destination Select Name PRIM_SS_LEADER Time of Day Add Remove View Gaps/Overlaps	FQDN or IP Address 192.168.51.34	Type Other	Notes GR PRIME SITE	]
Regular Expressions Defaults	SIP Entity as Destination Select Name PRIM_SS_LEADER Time of Day Add Remove View Gaps/Overlaps 1 Item Refresh	FQDN or IP Address 192.168.51.34	Type Other	Notes GR PRIME SITE	Filter: Ena
Defaults	SIP Entity as Destination Select Name PRIM_SS_LEADER Time of Day Add Remove View Gaps/Overlaps 1 Item Refresh Rankin Refresh Table Name 2	FQDN or IP Address           192.168.51.34           Mon         Tue         Wed         Thu         Fri         Sat           (2)	Type Other Sun Start Time	Notes GR PRIME SITE End Time	Filter: Ena Notes

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.

										ALC: NO
Domains	Pouting Policy Details								Commit	Help
Locations	Routing Poncy Details								(comme) (c	Cance
Adaptations	General									
SIP Entities	ounciu									
Entity Links		• Name:	SIP Trunk Calls to	sirm2	-					
Time Ranges		Disabled:	0							
Routing Policies		Notes:								
Dial Patterns										
Regular Expressions	SIP Entity as Destinat	ion								
Defaults	Select									
	Manua	FQDN or IP Address		Туре		Notes				
	name									
	NOAS SBC Birm2	xxx.yyy.113.62		Other	3	Primary SIP	inbound / outbound	calls		
	NOAS SBC Birm2 Time of Day Add Remove View Gap: 1 Item Refre Selected Time of Resting Policy.	xxx.yyy.113.62 s/Overlaps Day entries will be deleted from this		Other		Primary SIP	inbound / outbound	calls	Filter: E	Enab
	NOAS SBC Birm2 Time of Day Add Remove View Gap: 1 Item Refre Selected Time of Ranking	xxx.yyy.113.62 s/Overlaps Day entries will be deleted from this name non Tu	se Wed Ti	Other nu Fri	Sat	Primary SIP Sun	inbound / outbound Start Time	calls End Time	Filter: E Notes	Enab

## 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.

	* Montellan	le / Elements / Routing / Dial	Patterns - Dial Pattern	Details				
Domains								Help
Locations	Dial Pattern Details							Commit Can
Adaptations								
SIP Entities	Gene	eral				٦		
Entity Links			* Pattern: +	44113				
Time Ranges			* Min: 6					
Routing Policies			* Max: 3	6				
Dial Patterns			Emergency Call:	7				
Regular Expressions			Emergency can.					
Defaults			SIP Domain:	ALL-				
	0.14		Notes: L	eeds PSTN Area Code via SIP	Trunk			
	Origi	Remove	ing Policies	eeds PSTN Area Code via SIP	ITUIK			
	Origi Add 5 Iten	nating Locations and Rout Remove ns Refresh	ing Policies	eeds PSTN Area Code via SIP	Trunk	Routing Policy	Routing Policy	Filter: Ena
	Add 5 Iten	Inating Locations and Rout Remove ns. Refresh Originating Location Name 1	Originating Location	eeds PSTN Area Code via SIP Routing Policy Name	Rank 2	Routing Policy Disabled	Routing Policy Destination	Filter: Ena Routing Policy Notes
	Origi Add 5 Iten	Remove s Refresh Originating Location Name <sup>1</sup>	Originating Location Notes	Routing Policy Name SIP Calls to Romford Acres SBC	Rank 2	Routing Policy Disabled	Routing Policy Destination Romford SBC Acme 4500 net-net	Filter: En. Routing Policy Notes
	Origi Add 5 Iten	Remove s Refresh Originating Location Name 1	Originating Location Notes Any Locations Any Locations	Routing Policy Name SIP Calls to Romford Acres SBC SIP Trunk Calls to Birm2	Rank 20 10	Routing Policy Disabled	Routing Policy Destination Romford SBC Acme 4500 net-net NOAS SBC Birm2	Filter: Ent Routing Policy Notes
	Add 5 Iten	ALL-	Any Locations Any Locations Any Locations	Routing Policy Name SIP Calls to Romford Acme.SBC SIP Trunk Calls to Rirm2 SIP Trunk Calls to Rirm1	Rank 2	Routing Policy Disabled	Routing Policy Destination Romford SBC Acme 4500 net-net NOAS SBC Birm2 NOAS SBC Birm1	Filter: En. Routing Policy Notes
	Origi	ALL-	Any Locations Any Locations Any Locations Any Locations Any Locations	Routing Policy Name SIP Calls to Romford Acme SBC SIP Trunk Calls to Birm2 SIP Trunk calls to Birm1 SIP Trunk calls to Man2	Rank 20 10 20 30	Routing Policy Disabled	Routing Policy Destination Romford SBC Acme 4500 net-net NOAS SBC Birm2 NOAS SBC Birm1 NOAS SBC Man2	Filter: En Routing Policy Notes

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

Routing	<ul> <li>Home / Elements / Routing / Dial</li> </ul>	Patterns - Dial Patter	m Details					
Domains								Help
Locations	Dial Pattern Details							Commit Cano
Adaptations								
SIP Entities	General		And and the Article Article International					
Entity Links		* Pattern:	+44207960325					
Time Ranges		* Min:	12					
Routing Policies		* Max:	36					
Dial Patterns		Emergency Call:	19					
Regular Expressions		EID Domainu	ALL					
Defaults		SIP Domain.	"ALL"					
		Notes:	Inbound DDI +44207 96325X	from NOAS S	Serv			
	Originating Locations and Rout	ing Policies						
	Add Remove							
	1 Item Refresh				_			Filter: Enab
	Originating Location Name <sup>1</sup>	Originating Location Notes	Routing Policy Name	Rank	2	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
	NOAS SIP Service		Incoming to Leeds CS1000 Direct	0			PRIM_SS_LEADER	Calls to Prim_SS_Leader
	Select : All None							

## 7. Verification Steps

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

 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.

AVAYA	Ava	aya Aura™ Syste	em Manager 6.:	1		Help   About	Change Password	Log off admin
							Session Mana	ger × Home
* Session Manager	Home /	Elements / Session Mana	iger / System Status / SII	P Entity M	Monitorin	g – SIP Entity M	onitoring	
Dashboard	[							Help ?
Session Manager Administration	SIP EI This page d	ntity, Entity Link lisplays detailed connection statu	connection State	IS ision Mana	ger instance	es to a single SIP er	itity.	
Communication Profile Editor	All En	tity Links to SIP Entity:	: Leeds SM6.1					
Network Configuration	Sum	nary view						
Device and Location	1 Item	Refresh						Filter: Enable
Configuration	Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Application Configuration	Show	Leeds SM6.1	192.168.51.45	5060	TCP	Up	200 OK	Up

2. From the Communication Server 1000Esystem 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<sup>®</sup> 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 <u>http://support.avaya.com</u>.

- [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

0IPMGs Configured/unregistered:

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:

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 ISS1:10F1 p30550\_1 14/12/2010 p30550\_1.cpm NO 001 wi00835093 ISS1:10F1 p30553 1 14/12/2010 p30550\_1.cpm YES 002 wi00832626 ISS2:10F1 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: 7.5	50.17.00			
In syst	em patches: (	)			
In Syst	em service up	dates: 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 i	s no SP in lo	aded status	3.		
The las	t applied SP:	: Service Pa	ack Linux 7	7.50 17 20110	)118.ntl, It is a STANDARD SP.
Has bee	n applied by	user nortel	on Mon Fe	eb 7 14:59:0	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	.1 .
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	<u> </u>	
Configuration version:	7 50 17-00	
dhcom	7 50 17	
cslogin	7 50 17	
sigerverShare	7 50 17	[natched]
Cev	7.50.17	[pacened]
+ne	7.50.17	
utrk	7.50.17	[natched]
nd	7.50.17	[pacened]
pu	7.50.17	[natched]
525	7.50.17	[pacened]
	7.30.17	
yk. En Confin	7.30.17	
Emconing	7.50.17	
	7.50.17	
emweblocal_6-0	7.50.17	
csmweb	7.50.17	
	7.50.17	[patched]
ftrpkg	7.50.17	
cs1000WebService_6-0	7.50.17	
managedElementWebService	2.50.17	
mscAnnc	7.50.17	
mscAttn	7.50.17	
mscConf	7.50.17	
mscMusc	7.50.17	
mscTone	7.50.17	

GOR; Reviewed: SPOC 12/15/2011

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