

Avaya Solution & Interoperability Test Lab

# Application Notes for IPC System Interconnect with Avaya Aura<sup>TM</sup> Communication Manager Using Avaya Aura<sup>TM</sup> SIP Enablement Services – Issue 1.0

# Abstract

These Application Notes describe the configuration steps required for IPC System Interconnect 16.1 to interoperate with Avaya Aura<sup>TM</sup> Communication Manager 5.2.1 using Avaya Aura<sup>TM</sup> SIP Enablement Services 5.2.1.

IPC System Interconnect is a trading communication solution. In the compliance testing, IPC System Interconnect used SIP trunks to Avaya Aura<sup>TM</sup> SIP Enablement Services, for turret users on IPC to reach users on Avaya Aura<sup>TM</sup> Communication Manager and on the PSTN.

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 configuration steps required for IPC System Interconnect 16.1 to interoperate with Avaya Aura<sup>TM</sup> Communication Manager 5.2.1 using Avaya Aura<sup>TM</sup> SIP Enablement Services (SES) 5.2.1.

IPC System Interconnect is a trading communication solution. In the compliance testing, IPC System Interconnect used SIP trunks to Avaya Aura<sup>TM</sup> SES, for turret users on IPC to reach users on Avaya Aura<sup>TM</sup> Communication Manager and on the PSTN.

# 1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included basic call, display, G.711MU, G.729AB, codec negotiation, media shuffling, hold/reconnect, DTMF, call forwarding unconditional/ring-no-answer/busy, blind/attended transfer, and attended conference.

The serviceability testing focused on verifying the ability of IPC System Interconnect to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cables to IPC System Interconnect.

# 1.2. Support

Technical support on IPC System Interconnect can be obtained through the following:

- Phone: (800) NEEDIPC, (203) 339-7800
- Email: <u>systems.support@ipc.com</u>

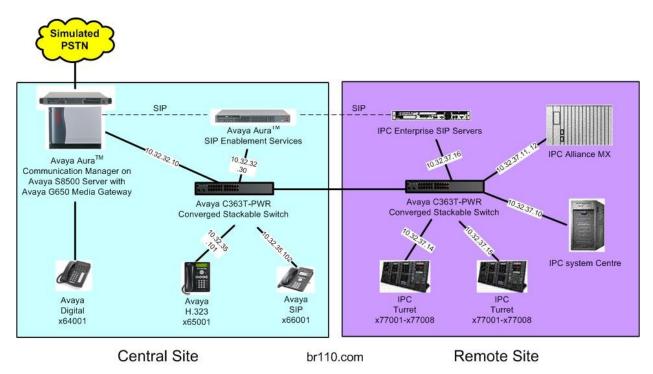
# 2. Reference Configuration

As shown in the test configuration below, IPC System Interconnect at the Remote Site consists of the Enterprise SIP Server (ESS), Alliance MX, System Center, and Turrets. SIP trunks are used from System Interconnect to Avaya Aura<sup>TM</sup> SES, to reach users on Avaya Aura<sup>TM</sup> Communication Manager and on the PSTN.

IPC System Interconnect supports only one SIP domain, which will be used in both the SIP "From" and "To" headers. Therefore, the same domain must be used for the two sites. In the compliance testing, the "br110.com" domain was used for all users on both sites.

A five digit Uniform Dial Plan (UDP) was used to facilitate dialing between the Central and Remote sites. Unique extension ranges were associated with Avaya Aura<sup>TM</sup> Communication Manager users at the Central site (6xxxx), and IPC turret users at the Remote site (7xxxx).

The detailed administration of basic connectivity between Avaya Aura<sup>TM</sup> Communication Manager and Avaya Aura<sup>TM</sup> SIP Enablement Services is not the focus of these Application Notes and will not be described.



# 3. Equipment and Software Validated

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

Equipment	Software
Avaya Aura <sup>TM</sup> Communication Manager on Avaya S8500 Server	5.2.1 (R015x.02.1.016.4-18433)
<ul> <li>Avaya G650 Media Gateway</li> <li>TN799DP C-LAN Circuit Pack</li> <li>TN2302AP IP Media Processor</li> </ul>	HW01 FW038 HW20 FW121
Avaya Aura <sup>TM</sup> SIP Enablement Services	5.2.1 (SES-5.2.1.0-016.4)
Avaya 1608 IP Telephone (H.323)	1.3
Avaya 9630 IP Telephone (H.323)	3.1
Avaya 9630 IP Telephone (SIP)	2.6.2
<ul> <li>IPC System Interconnect</li> <li>Alliance MX</li> <li>Enterprise SIP Server</li> <li>System Center <ul> <li>SIPX Line Card</li> <li>Turrets</li> </ul> </li> </ul>	SipProxy-2.00.01-13 16.01.01.03.0007 16.01.01.03.0007 16.01.01.03.0007 16.01.01.03.0007 16.01.01.03.0007

# 4. Configure Avaya Aura<sup>TM</sup> Communication Manager

This section provides the procedures for configuring Avaya Aura<sup>TM</sup> Communication Manager. The procedures include the following areas:

- Verify Communication Manager license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer public unknown numbering
- Administer uniform dial plan
- Administer AAR analysis
- Administer ISDN trunk group
- Administer tandem calling party number

In the compliance testing, the same set of codec set, network region, trunk group, and signaling group were used for the Avaya SIP and IPC turret users, which enabled IPC turret users to use the same digits dialing as Avaya SIP users, to reach other users on Communication Manager and on the PSTN.

#### 4.1. Verify Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

change system-parameters customer-options		Page	2 of	11
OPTIONAL FEATURES				
IP PORT CAPACITIES		USED		
Maximum Administered H.323 Trunks:	100	6		
Maximum Concurrently Registered IP Stations:	18000	4		
Maximum Administered Remote Office Trunks:	8000	0		
Maximum Concurrently Registered Remote Office Stations:	18000	0		
Maximum Concurrently Registered IP eCons:	10	0		
Max Concur Registered Unauthenticated H.323 Stations:	10	0		
Maximum Video Capable H.323 Stations:	100	0		
Maximum Video Capable IP Softphones:	100	0		
Maximum Administered SIP Trunks:	100	10		
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0		
Maximum Number of DS1 Boards with Echo Cancellation:	0	0		

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## 4.2. Administer System Parameters Features

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers.

This feature is needed to be able to transfer an incoming call from IPC back out to IPC (incoming trunk to outgoing trunk), and to transfer an outgoing call to IPC to another outgoing call to IPC (outgoing trunk to outgoing trunk). For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class Of Restriction or Class Of Service levels. Refer to [1] for more details.

```
change system-parameters features
                                                               Page 1 of 18
                           FEATURE-RELATED SYSTEM PARAMETERS
                              Self Station Display Enabled? y
                                   Trunk-to-Trunk Transfer: all
              Automatic Callback with Called Party Queuing? n
   Automatic Callback - No Answer Timeout Interval (rings): 3
                      Call Park Timeout Interval (minutes): 10
       Off-Premises Tone Detect Timeout Interval (seconds): 20
                                AAR/ARS Dial Tone Required? y
             Music (or Silence) on Transferred Trunk Calls? no
                     DID/Tie/ISDN/SIP Intercept Treatment: attd
   Internal Auto-Answer of Attd-Extended/Transferred Calls: none
                 Automatic Circuit Assurance (ACA) Enabled? n
            Abbreviated Dial Programming by Assigned Lists? n
      Auto Abbreviated/Delayed Transition Interval (rings): 2
                   Protocol for Caller ID Analog Terminals: Bellcore
   Display Calling Number for Room to Room Caller ID Calls? n
```

#### 4.3. Administer SIP Trunk Group

Use the "change trunk-group n" command, where "n" is the existing SIP trunk group number used to reach Avaya SES, in this case "5".

For **Group Name**, update as desired to reflect the same trunk group used to reach SES and IPC. For **Number of Members**, enter sufficient number for simultaneous calls to Avaya SIP and IPC users. Note that a call between an Avaya SIP user and an IPC user uses two SIP trunks, whereas a call between an Avaya non-SIP user and an IPC user uses one SIP trunk. Make a note of the **Signaling Group** number.

change trunk-group 5	Page 1 of 21
TRUNK GROUP	
Group Number: 5 Group Type Group Name: SIP Trunk to SES/IPC COR Direction: two-way Outgoing Display Dial Access? n	: 1 TN: 1 TAC: 1005
Queue Length: 0 Service Type: tie Auth Code	-
	Signaling Group: 5 Number of Members: 10

Navigate to Page 3, and enter "public" for Numbering Format.

change trunk-group 5 TRUNK FEATURES ACA Assignment? n	Measured	Page 3 of 21
	neabarea	Maintenance Tests? y
Numbering Format:	public	UUI Treatment: service-provider
		Replace Restricted Numbers? n Replace Unavailable Numbers? N

Navigate to Page 4, and enter "101" for Telephone Event Payload Type, as shown below.

change trunk-group 5		Page	4 of	21
PROTOCOL VARIA	ATIONS			
Mark Users as Phone? n	n			
Prepend '+' to Calling Number? n	n			
Send Transferring Party Information? r	n			
Network Call Redirection? r	n			
Send Diversion Header? n	n			
Support Request History?	У			
Telephone Event Payload Type: 1	101			

#### 4.4. Administer SIP Signaling Group

Use the "change signaling-group n" command, where "n" is the existing SIP signaling group number used by the SIP trunk group from **Section 4.3**.

For **DTMF over IP**, enter "rtp-payload". For **Direct IP-IP Audio Connections**, enter "y". Make a note of the **Far-end Network Region** number.

```
change signaling-group 5
                                                                Page 1 of 1
                                SIGNALING GROUP
Group Number: 5
                            Group Type: sip
                        Transport Method: tls
  IMS Enabled? n
  Near-end Node Name: Clan-1
                                            Far-end Node Name: SES
                                         Far-end Listen Port: 5061
Near-end Listen Port: 5061
                                      Far-end Network Region: 1
Far-end Domain: br110.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate
                                                    RFC 3389 Comfort Noise? n
        DTMF over IP: rtp-payload
                                            Direct IP-IP Audio Connections? y
                                             IP Audio Hairpinning? n
Direct IP-IP Early Media? n
Session Establishment Timer(min): 3
Enable Laver 3 Test? n
      Enable Layer 3 Test? n
H.323 Station Outgoing Direct Media? n
                                                 Alternate Route Timer(sec): 6
```

#### 4.5. Administer IP Network Region

Use the "change ip-network-region n" command, where "n" is the existing far-end network region number used by the SIP signaling group from **Section 4.4**.

For **Name**, update as desired to reflect the same network region used to reach SES and IPC. Enter "yes" for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. In the compliance testing, the same network region was used for all Avaya users. Make a note of the **Codec Set** number. Also make a note of the **Authoritative Domain**, which should match the SIP domain name of the SES server, and will be used later to configure IPC.

```
change ip-network-region 1
                                                                   Page 1 of 19
                                IP NETWORK REGION
 Region: 1
                 Authoritative Domain: br110.com
Location:
   Name: SES/IPC Region
MEDIA PARAMETERS
                                Intra-region IP-IP Direct Audio: yes
     Codec Set: 1
                                Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048
                                            IP Audio Hairpinning? n
  UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
                                          RTCP Reporting Enabled? y
Call Control PHB Value: 46
Audio PHB Value: 46
RTCP MONITOR SERVER PARAMETERS
Use Default Server Parameters
                                 Use Default Server Parameters? y
       Video PHB Value: 26
```

#### 4.6. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the existing codec set number used by the IP network region from **Section 4.5**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that IPC System Interconnect supports the G.711 and G.729 codec variants. For **Media Encryption**, make certain "none" is specified.

In the compliance testing, the same codec set was used for all Avaya users.

```
change ip-codec-set 1

IP Codec Set

Codec Set: 1

Audio Silence Frames Packet

Codec Suppression Per Pkt Size(ms)

1: G.711MU n 2 20

2: G.729AB n 2 20

3:

4:

5:

6:

7:

Media Encryption

1: none

2:
```

## 4.7. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is the existing route pattern number to reach SES, in this case "5". For **Pattern Name**, update as desired to reflect the same route pattern used to reach SES and IPC. For **Secure SIP**, make certain the value is "n".

chai	nge route-pattern 5	Page 1 o	of 3
	Pattern M	Number: 5 Pattern Name: To SES/IPC	
		SCCAN? n Secure SIP? n	
	Grp FRL NPA Pfx Hop Toll	No. Inserted DC:	S/ IXC
	No Mrk Lmt List		TG
		Dgts In	
1:	5 0	2900 II. n	user
2:	5 0	n	user
3:			
		n	user
4:		n	user
5:		n	user
6:		n	user
	BCC VALUE TSC CA-TSC	ITC BCIE Service/Feature PARM No. Numbering	g LAR
	0 1 2 M 4 W Request	Dgts Format	
	-	Subaddress	
1:	yyyyn n	rest	none

## 4.8. Administer Public Unknown Numbering

Use the "change public-unknown-numbering 0" command, to define the calling party number to send to IPC. Add an entry for the trunk group defined in **Section 4.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 5 will result in a 5-digit calling number. The calling party number will be in the SIP "From" header.

char	nge public-unki	nown-numbe	ring O		Page 1	of	2
		NUMBE	RING - PUBLIC/UN	JKNOWN FO	ORMAT		
				Total			
Ext	Ext	Trk	CPN	CPN			
Len	Code	Grp(s)	Prefix	Len			
					Total Administered:	3	
5	6	5		5	Maximum Entries:	9999	

## 4.9. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 7xxxx to IPC. Note that other methods of routing may be used. Use the "change uniform-dialplan 0" command, and add an entry to specify the use of AAR for routing digits 7xxxx, as shown below.

ſ	change uniform-dialplan 0					Page	1 of	2		
	UNIFORM DIAL PLAN TABLE				rage	I OI	2			
			UNIFO.	RM DIAL FLAI	N IAD			Percent	Full:	0
	Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num			
	7	5	0		aar	n				

#### 4.10. Administer AAR Analysis

Use the "change aar analysis 0" command, and add an entry to specify how to route calls to 7xxxx. In the example shown below, calls with digits 7xxxx will be routed as an AAR call using route pattern "5" from **Section 4.7**.

change aar analysis 0					Page 1 of	2
	AAR D	IGIT ANALYS	SIS TABI	ΞE		
		Location:	all		Percent Full:	2
Dialed	Total	Route	Call	Node	ANI	
String	Min Max	Pattern	Туре	Num	Reqd	
7	55	5	aar		n	

## 4.11. Administer ISDN Trunk Group

Use the "change trunk-group n" command, where "n" is the existing ISDN trunk group number used to reach the PSTN, in this case "500".

For **Modify Tandem Calling Number**, enter "y" to allow for the calling party number from IPC to be modified.

```
Page 3 of 21
change trunk-group 500
          TURES
ACA Assignment? n Measured: None
Internal Alert? n Maintenance Tesco. 1
Data Restriction? n NCA-TSC Trunk Member:
Send Name: n Send Calling Number: y
Send EMU Visitor CPN? y
TRUNK FEATURES
   Used for DCS? n
Suppress # Outpulsing? n Format: public
 Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider
                                                       Replace Restricted Numbers? n
                                                       Replace Unavailable Numbers? n
                                                             Send Connected Number: y
Network Call Redirection: none
                                                         Hold/Unhold Notifications? n
            Send UUI IE? y
                                                     Modify Tandem Calling Number? y
               Send UCID? n
Send Codeset 6/7 LAI IE? y
                                                           Ds1 Echo Cancellation? n
                                              US NI Delayed Calling Name Update? n
    Apply Local Ringback? n
Show ANSWERED BY on Display? y
                                 Network (Japan) Needs Connect Before Disconnect? n
```

# 4.12. Administer Tandem Calling Party Number

Use the "change tandem-calling-party-num" command, to define the calling party number to send to the PSTN for tandem calls from IPC turret users.

In the example shown below, all calls originating from a 5-digit extension beginning with 7 and routed to trunk group 500 will result in a 10-digit calling number. For **Number Format**, use an applicable format, in this case "pub-unk".

change	tandem-calling-par CALLING	ty-num PARTY NUMBER C	ONVERSION	Page	1 of	8
		LS				
CP	N Trk			Number		
Len Pr	refix Grp	(s) Delete	Insert	Format		
57	500		90884	pub-unk		

# 5. Configure Avaya Aura<sup>TM</sup> SIP Enablement Services

This section provides the procedures for configuring Avaya Aura<sup>TM</sup> SES. The procedures include the following areas:

- Launch SES administration
- Administer host address map
- Administer host contact
- Administer trusted host

#### 5.1. Launch SES Administration

Access the SES web interface by using the URL "http://ip-address/admin" in an Internet browser window, where "ip-address" is the IP address of the SES server. Log in using the appropriate credentials.

	SIP Enablement Services (SES System Management Interface (SM)	
Help Exit		
	Logon	
	Logon ID:	
	Logon	
1		
	© 2001-2009 Avaya Inc. All Rights Reserved.	

In the subsequent screen, select **Administration > SIP Enablement Services** from the top menu.



The **Top** screen is displayed next.

AVAYA			Integrated Management SIP Server Management
Help Exit			This Server: [1] brses1
Top © Users	🖡 Тор		
Address Map Priorities Adjunct Systems	Manage Users	Add and delete Users.	
• Aggregator	Manage Address Map Priorities	Adjust Address Map Priorities.	
<ul> <li>Certificate Management</li> <li>Conferences</li> </ul>	Manage Adjunct Systems	Add and delete Adjunct Systems.	
Emergency Contacts Export/Import to ProVision	Manage Event Aggregators	Add/Delete Event Aggregators.	
Hosts	Certificate Management	Manage Certificates.	
List Migrate Home/Edge	Manage Conferencing	Add and delete Conference Extensions.	
IM logs Communication Manager Servers	Manage Emergency Contacts	Add and delete Emergency Contacts.	
■ Communication Manager Extensions	Export Import to ProVision	Export and import data using ProVision on this host.	
Server Configuration	Manage Hosts	Add and delete Hosts.	
SIP Phone Settings	IM logs	Download IM Logs.	
Survivable Call Processors	Manage	Add and delete Communication	
System Status	Communication Manager Servers	Manager Servers.	
<ul> <li>Trace Logger</li> <li>Trusted Hosts</li> </ul>	Manage Communication Manager Extensions	Add and delete Communication Manager Extensions.	

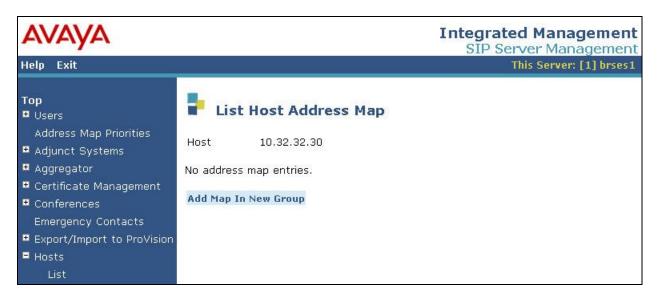
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# 5.2. Administer Host Address Map

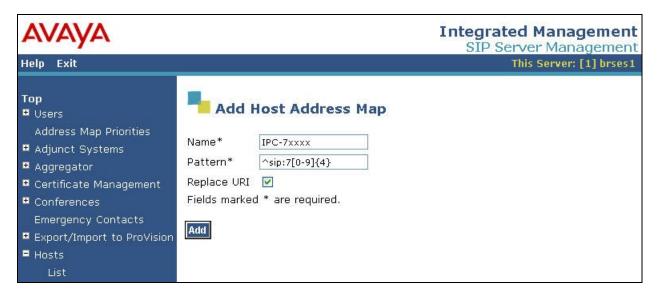
Select Hosts > List from the left pane. The List Hosts screen is displayed. Click on the Map link.



In the List Host Address Map screen below, click Add Map In New Group in the right pane.



The Add Host Address Map screen is displayed next. This screen is used to specify which calls are to be routed to IPC. For Name, enter a descriptive name to denote the routing. For Pattern, enter an appropriate syntax for address mapping. For the compliance testing, a pattern of "^sip:7[0-9]{4}" is used to match to any IPC turret user extensions of 7xxxx. Maintain the check in **Replace URI**.



## 5.3. Administer Host Contact

The List Host Address Map screen is displayed again, and updated with the newly created address map. Click Add Another Contact in the right pane.



In the Add Host Contact screen, enter the contact "sip:\$(user)@<destination-IP-address> :5060;transport=tcp", where the <destination-IP-address> is the IP address of the IPC ESS server. Avaya SES will substitute "\$(user)" with the user portion of the request URI before sending the message.



## 5.4. Administer Trusted Host

Select **Trusted Hosts > Add** from the left pane. The **Add Trusted Host** screen is displayed. For the **IP Address** field, enter the IP address of the IPC ESS server from **Section 5.3**. Enter a desired description for **Comment**.

avaya			Integrated Management SIP Server Management
Help Exit			This Server: [1] brses1
Top ■ Users Address Map Priorities	Add Trus	sted Host	
<ul> <li>Adjunct Systems</li> </ul>	IP Address*:	10.32.37.16	
• Aggregator	Host*:	10.32.32.30 💌	
• Certificate Management	Comment:	IPC ESS	
• Conferences	Perform Originatio	on Processing:	
Emergency Contacts	Fields marked * a	2011년21년21년21년21년21년21년21년21년21년21년21년21년2	
▪ Export/Import to ProVision	Add		
▪ Hosts			
IM logs			
<ul> <li>Communication Manager Servers</li> </ul>			
Extensions			
Server Configuration			
• SIP Phone Settings			
Survivable Call Processors			
System Status			
Trace Logger			
Trusted Hosts			
Add			

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# 6. Configure IPC System Interconnect

This section provides the procedures for configuring IPC System Interconnect. The procedures include the following areas:

- Launch One Management System
- Administer SIP configuration
- Administer routing plan
- Administer wire groups
- Administer trusted host

The configuration of System Interconnect is typically performed by IPC installation technicians. The procedural steps are presented in these Application Notes for informational purposes.

## 6.1. Launch One Management System

Access the One Management System web interface by using the URL "http://ipaddress/oneview" in an Internet browser window, where "ip-address" is the IP address of IPC System Center. Log in using the appropriate credentials.

The Login screen is displayed. Enter the appropriate credentials. Check I agree to the terms and conditions, and click Login.

The License Login screen is displayed next (not shown). Enter the appropriate password and click Login. In the subsequent Login Information screen (not shown), click Continue.

One Management System	Login En Username Password	nglish 🔻
	Reset	Login
TERMS AND CONDITIONS	🗹 I agree to the tern	ns and conditions.
Access to this system and/or netw in it are lawfully available only fo employees of IPC or other users than where prohibited by law and requirements, IPC reserves the r in any form on this system and/o	r approved purposes by authorized by IPC. Oth I subject to legal ight to review any inform	er
This system is for the use of aut individuals using this computer s their activities on this system mo using this system expressly cons	ystem are subject to hav onitored and recorded. Ar	nyone

# 6.2. Administer SIP Configuration

The screen below is displayed next, with the **Main Menu** screen in the forefront. Select **NEXUS** > **SIP Trunk Parameters** > **Edit SIP Config**, as shown below.

Alarm	_				- 🗆 🗙
Red Alarms	P	ink Alarms			
				DDI Exte	Time Rep
1	Unk	Main Menu	SIP Sites	0	2010-09-20
	(0)	Hum Hend	SIP Servers		11:32:24
2	Unk	TRADER CONFIG	▶ SIP Authentication	0	2010-09-20
	(0)	BUTTON CONFIG	🔻 SIP Trunk Parameters		11:32:24
	1000000	ICM CONFIG	Edit SIP Config		-
3	Unk (0)	LINE CONFIG	▶ Update ESS with SIP Trunk Info	0	2010-09-20
		STATION CONFIG	▶ Routing Plan		
ŧ	Unk	GROUPS	Enterprise Lines	0	2010-09-20
	(0)		▶ Enterprise Reach		11:37:24
5	Unk	SYSTEM STATUS	SIP Security Config	0	2010-09-20
~	(0)	SYSTEM SETTINGS			11:37:24
	1000	VOICE RECORDING			
5	Unk (0)	LINE NETWORKING		0	2010-09-20 11:37:24
		NEXUS			_
7	Unk (0)	MAXaccess 1000		0	2010-09-20
	(0)	TOOLS			11/37/24
3	Unk	REPORTS		0	2010-09-20
	(0)	ONEMS ADMIN			11:42:24

The Edit SIP Config screen is displayed. For DDI Group ID/ DDI Group Name, select the relevant SIP trunk card number from the drop-down list, in this case "5". Click Submit.

OneView Log out Main Menu 2 Work Areas		C
dit SIP Config	-	×
DDI Group ID/ DDI Group Name 5 [?] 🔻		
Submit		

The Edit SIP Config screen is updated with the located DDI Group ID entry. Double click on the Outbound URL field corresponding to the located entry, and enter the SIP domain from Section 4.5. IPC will use this SIP domain in the SIP "From" and "To" headers.

Ed	lit SIP Config			EDIT	ACTION 🔳 ·	- 🗆 ×
Sele	ect column :		Go			
	DDI Group ID	Outbound URL	Usemame	Password	Confirm Password	DNS1 IP Address
1	5	br110.com	avaya	****	****	

# 6.3. Administer Routing Plan

Select MAIN MENU from the top menu to display the Main Menu screen. Select NEXUS > Routing Plan > View/Edit/Delete Routing Plan, as shown below. Click Submit in the subsequent screen (not shown) to search for all routing plans.

Alarm				,
Red Alarms	Pink Alarms			
	1		- × DDI Exte	Time Rep
	Main Menu	SIP Sites	0	2010-09-20
(		► SIP Servers		11:17:23
2	TRADER CONFIG	SIP Authentication	0	2010-09-20
	0) BUTTON CONFIG	SIP Trunk Parameters		11:17:23
	ICM CONFIG	🔻 Routing Plan		
	0) LINE CONFIG	Add Routing Plan	0	2010-09-20
	STATION CONFIG	View/Edit/Delete Routing Plan		- Contraction of the second se
	Jok	▶ Enterprise Lines	0	2010-09-20
C	<b>0</b> )	▶ Enterprise Reach		11:17:23
;	SYSTEM STATUS	SIP Security Config	0	2010-09-20
(	0) SYSTEM SETTINGS			11:22:23
	VOICE RECORDING			
	0) LINE NETWORKING		0	2010-09-20
	NEXUS			
	MAXaccess 1000		0	2010-09-20
	0) TOOLS			11:22:23
	Ink REPORTS		0	2010-09-20
(	0) ONEMS ADMIN			11:22:23

The **View/Edit/Delete Routing Plan** screen is displayed. Follow [3] to add two routing entries shown below.

The entry with **Sequence Number 2** was used for routing of inbound calls to IPC. Note that the **Destination** URL contains the internal default value for the SIP trunk card, in this case "group5.com".

The entry with **Sequence Number 3** was used for routing of outbound calls to Avaya SES. Note the **Destination** URL includes the IP address of Avaya SES, and the transport method from **Section 5.3**.

IPO	OneVie	ew log ou	T MAIN MENU	J 2 WORK ARE	AS 🖵 🌲 C:\ 🕑 \$xdb1	
Vie	w/Edit/Delet	e Routing Pla	in		EDIT ACTION	1
Sele	ct column :		Go	· · ·		
	Sequence Number	Action	From	То	Destination	•
2	2	Forward	sip:*	sip:77\$\$\$@*	sip:{user}@group5.com	
3	3	Forward	sip:*	sip:*	sip:{user}@10.32.32.30;transport=TCP	

## 6.4. Administer Wire Groups

Select MAIN MENU from the top menu to display the Main Menu screen. Select GROUPS > Engineering Groups > Wire Groups, as shown below.

Red Alarms	Pink Alarms			
	1	- ×	DDI Exte	Time Rep
500 Barrier (* 1997)	Main Menu	▶ Trader Group	0	2010-09-20
(0		Billing Group		12:02:50
2 U	TRADER CONFIG	▶ Hunt Group	0	2010-09-20
	) BUTTON CONFIG	Engineering Groups		12:02:50
	ICM CONFIG	Line Groups		
2.5	LINE CONFIG	Wire Groups		2010-09-20
	STATION CONFIG	Station Groups		12.02.00
4 Unk	nk	Module Groups	0	2010-09-20
0	) GROUPS	Port Groups		12:02:50
5 A	SYSTEM STATUS		-1	2010-09-20
	at SYSTEM SETTINGS		-1	12:14:55
3	VOICE RECORDING			
22.5 E	ar LINE NETWORKING			2010-09-20
			-1	18:42:09
	es MAXaccess 1000		-1	18:43:31
)		_		
	ar TOOLS			2010-09-20
R	es REPORTS			18:43:27
) P	ONEMS ADMIN		-1	2010-09-20

The **Wire Groups** screen is displayed next. Select "SIP" from the **Select Wire Group** dropdown list, and "Edit" from the **Select Operation** drop-down list, as shown below.

🖭 OneVie	W LOG OUT M	IAIN MENU	2 WORK AREAS	Ţ		C:\	1	sxdb1
Wire Groups				-	□ ×		-)	n ×
Select Wire Group	SIP	•				-		
Select Operation	Edit 🛛 🔻					Exte	Time F	No. of Concession, Name
	Submit						2010-	
							2010- 11:17	09-20 123

The **Edit Wire Groups** screen is displayed. Scroll down the screen as necessary to locate the entry with **Param ID** of "365". Double click on the corresponding **Param Value** field, and enter "2" to denote Avaya as the PBX provider.

Locate the entry with **Param ID** of "370". Double click on the corresponding **Param Value** field, and enter "4" to enable Forward Switching.

Edi	it Wire Groups	5					EDI	T ACTION	0 - 5
Sele	ct column :		Go						
1	Group	Param Value	Param Min	Param Max	Param	Param	Param Type	Param ID	Group ID
72	SIP Line Card	32767	1	32767	DSP_TERM_ATTE	DSP TERM thresh	number	141	27
73	SIP Line Card	0	-5	5	TERM_SHIFT	gain/loss into ip	number	362	27
74	SIP Line Card	0	-5	5	PERIPH_SHIFT	gain/loss into pu	number	363	27
75	SIP Line Card	6	0	32	INTERDIGIT_TO	interdigit timeou	number	364	27
76	SIP Line Card	2	1	7	PBX_PROVIDER	1-7/DEF,AVYA,NF	enum	365	27
77	SIP Line Card	6	1	15	MAX_DIVERTS	Max Number of I	number	369	27
78	SIP Line Card	4	0	4	FS_ENABLE	0-4/Off, Imm&B	number	370	27
79	SIP Line Card	200	200	10000	FS_DELAY	Time(msec) to V	number	371	27
80	SIP Line Card	1	1	5	LN_RECORDS	1-5/NONE,MX_PE	number	375	27

Scroll down the screen as necessary to locate the entry with **Param ID** of "661". Double click on the corresponding **Param Value** field, and enter "1" to activate detection for G729.

Locate the entry with **Param ID** of "666". Double click on the corresponding **Param Value** field, and enter "1" to enable SIP Provisional Acknowledgement (PRACK).

Locate the entry with **Param ID** of "668". Double click on the corresponding **Param Value** field, and enter "0" to disable SIP Remote Party ID (RPI).

Edi	t Wire Groups	5					EDI	T ACTION	0 - 8
Selec	t column :		Go						
1	Group	Param Value	Param Min	Param Max	Param	Param	Param Type	Param ID	Group ID
95	SIP Line Card	1209	0	3000	SPLSHTONELO_F	Splash tone LO I	number	656	27
96	SIP Line Card	1477	0	3000	SPLSHTONEHI_F	Splash tone HI f	number	657	27
97	SIP Line Card	1400	0	3000	RECWARNTONE_	Record warning f	number	658	27
98	SIP Line Card	0	0	10000	MRD Ringback T	Ringback Tone [	number	659	27
99	SIP Line Card	1	0	1	VAD	Voice Activity De	number	661	27
100	SIP Line Card	0	0	1	MWI Subscribe	Send MWI Subso	number	663	27
101	SIP Line Card	0	0	1	SIP Divert	HistoryInfo = 0,	number	664	27
102	SIP Line Card	1	0	1	SIP PRACK	Enable SIP Provi	number	666	27
103	SIP Line Card	1	0	1	SIP PAI	Enable SIP P-As:	number	667	27
104	SIP Line Card	0	0	1	SIP RPID	Enable SIP Rem	number	668	27
105	SIP Line Card	0	0	1	AEC_Enable	Enable AEC Cont	number	669	27
106	SIP Line Card	0	-3	3	AEC_Control	AEC Aggression	number	670	27

Follow [3] to reboot the SIP trunk card.

# 6.5. Administer Trusted Host

From the Linux shell of the ESS server, navigate to the /usr/local/SipProxy/ directory, and issue the command shown below with the "-add" option to add Avaya SES as a trusted host. Note that 10.32.32.30 is the IP address of Avaya SES.

The same command can be used with the "-view" option to make certain Avaya SES is displayed as a trusted host.

```
[root@esshost ~]# cd /usr/local/SipProxy/
[root@esshost SipProxy]# ./trusted_hosts.pl -add=10.32.32.30
[root@esshost SipProxy]# ./trusted_hosts.pl -view
ip_address last_modified
10.32.32.30 2010-09-21 16:48:09
```

# 7. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established among IPC turret users with Avaya SIP, Avaya H.323, Avaya Digital, and/or PSTN users. Call controls were performed from the various users to verify the call scenarios.

The serviceability test cases were performed manually by disconnecting and reconnecting the LAN cables to the IPC ESS and IPC System Center servers.

All test cases were executed and passed.

# 8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura<sup>TM</sup> Communication Manager, Avaya Aura<sup>TM</sup> SIP Enablement Services, and IPC System Interconnect.

# 8.1. Verify Avaya Aura<sup>™</sup> Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 4.3**. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 5
                           TRUNK GROUP STATUS
Member Port Service State
                                Mtce Connected Ports
                                  Busy
0005/001 T00083 in-service/idle
                                  no
0005/002 T00084 in-service/idle
                                  no
0005/003 T00085 in-service/idle
                                 no
0005/004 T00086 in-service/idle
                                 no
0005/005 T00087 in-service/idle
                                 no
0005/006 T00082 in-service/idle
                                 no
0005/007 T00088 in-service/idle
                                 no
0005/008 T00089 in-service/idle
                                 no
0005/009 T00090 in-service/idle
                                  no
0005/010 T00091 in-service/idle
                                  no
```

Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 4.4**. Verify that the signaling group is "in-service" as indicated in the **Group State** field shown below.

```
status signaling-group 5

STATUS SIGNALING GROUP

Group ID: 5

Group Type: sip

Signaling Type: facility associated signaling

Group State: in-service
```

# 8.2. Verify Avaya Aura<sup>™</sup> SIP Enablement Services

From the SES web interface, select **Trusted Hosts > List** from the left pane, to display the **List Trusted Hosts** screen. Verify that IPC ESS is listed as a trusted host.

Αναγα	Integrated Management SIP Server Management					
Help Exit This Server: [1] brse						
Top Users Address Map Priorities		Trusted Ho			<u>Perform</u>	
<ul> <li>Adjunct Systems</li> <li>Aggregator</li> </ul>	<u>Commands</u>	IP Address	Trusted by Host	Comment	Origination Processing	
<ul> <li>Aggregator</li> <li>Certificate Management</li> <li>Conferences</li> </ul>	Edit Delete	10.32.37.16	10.32.32.30	IPC ESS		
Emergency Contacts Export/Import to ProVision	Add Another	Trusted Host				
# Hosts						
IM logs						
Communication Manager						
Servers Communication Manager Extensions						
Server Configuration						
SIP Phone Settings						
Survivable Call Processors						
System Status						
• Trace Logger						
Trusted Hosts						
Add						
List						

## 8.3. Verify IPC System Interconnect

From the One Management System web interface, select MAIN MENU from the top menu to display the Main Menu screen. Select NEXUS > SIP Trunk Parameters > Update ESS with SIP Trunk Info > View/Delete SIP Cards to Trunks, as shown below.

Alarm					- 🗆 🗙
Red Alarms	Pi	nk Alarms			
1				DDI Exte	Time Rep
1	Unk (0)	Main Menu	SIP Sites	0	2010-09-20 4
		Hum Henu	SIP Servers		
0	Unk	TRADER CONFIG	SIP Authentication	0	2010-09-20 11:32:24
	(0)	BUTTON CONFIG	V SIP Trunk Parameters		
		ICM CONFIG	Edit SIP Config		
97.00 E	Unk	LINE CONFIG	🔻 Update ESS with SIP Trunk Info	0	2010-09-20 11:32:24
			Add SIP Cards to Trunks		
	Unk (0)	STATION CONFIG	View/Delete SIP Cards to Trunks	0	2010-09-20 11:37:24 2010-09-20 11:37:24
		GROUPS	Routing Plan		
		SYSTEM STATUS	Enterprise Lines		
97.00	Unk (0)	SYSTEM SETTINGS	▶ Enterprise Reach	a	
		VOICE RECORDING	SIP Security Config		
(#)))	Unk (0)	LINE NETWORKING		0	2010-09-20 11:37:24
		NEXUS			
7	Unk	MAXaccess 1000		0	2010-09-20

The **View/Delete SIP Cards to Trunks** screen is displayed. Verify that there is an entry that corresponds to SIP card number 5. Verify that the **Status** is "Online", as shown below.

IP	• OneVi	ew LOG OU	T MAIN MENU	2 WORK AREAS	Q	▲ C:\	③ sxdb1
Vie	ew/Delete SI	P Cards to Tru	nks	EDIT ACTION 🔳 –	= ×		×
Sele	ct column :		Go				
1	Domain	IP Address		Status	×	DDI Exte	Time Rep
1	group5.com	10.32.37.12	Online			ő	2010-09-20
2							11:32:24
						0	2010-09-20

# 9. Conclusion

These Application Notes describe the configuration steps required for IPC System Interconnect 16.1 to successfully interoperate with Avaya Aura<sup>TM</sup> Communication Manager 5.2.1 using Avaya Aura<sup>TM</sup> SIP Enablement Services 5.2.1. All feature and serviceability test cases were completed.

# 10. Additional References

This section references the product documentation relevant to these Application Notes.

- 1. Administrator Guide for Avaya Aura<sup>TM</sup> Communication Manager, Document 03-300509, Issue 8.0, Release 5.2, May 2009, available at <u>http://support.avaya.com</u>.
- **2.** *Installing, Administering, Maintaining, and Troubleshooting Avaya Aura<sup>TM</sup> SIP Enablement Services*, Document ID 03-600768, Issue 8.0, November 2009, available at <u>http://support.avaya.com</u>.
- **3.** *Nexus Suite 2.0 SP1 Patch11 or Higher Deployment Guide*, Part Number B02200161, Revision Number 01, upon request to IPC Support.

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