



Avaya Solution & Interoperability Test Lab

Application Notes for configuring the Allworx 6x System with Allworx IP phones to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager via a SIP Trunk – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Allworx 6x with Allworx IP phones to interoperate with Aura® Session Manager and Avaya Aura® Communication Manager via a SIP trunk. The Allworx 6x system is an all-in-one communication system integrating a feature-rich phone system, advanced IP phones, and powerful software features.

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 the Allworx 6x system with Allworx IP phones to interoperate with Aura® Session Manager and Avaya Aura® Communication Manager via a SIP trunk.

Designed for companies with up to 60 users, the Allworx 6x system is an all-in-one communication system integrating a feature-rich phone system, advanced IP phones, and powerful software features. Key features of the 6x system include:

- Supports VoIP and traditional phone lines
- PBX and/or Key system features
- Built-in nine (9) auto attendants
- Unified messaging
- Unlimited call routes
- Presence management
- One (1) eight (8) seat conference bridge
- Advanced software features support

2. General Test Approach and Test Results

The compliance testing focused on verifying the ability of the Allworx 6x system with Allworx IP phones to interoperate with an Avaya SIP-enabled IP Telephony Environment comprised of Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and Avaya phones (both SIP and H.323).

2.1. Interoperability Compliance Testing

The interoperability compliance test plan included feature and serviceability test cases. The feature testing focused on the following areas:

- Basic calls [verifying proper set up and tear down of the calls, the phones display information (i.e. Caller ID), and the voice paths/quality]
- Audio codec negotiation using G.711 and G.729 codecs
- Direct IP-IP Audio Connections (media shuffling)
- DTMF transmission using RFC 2833
- Voicemail with message waiting indicators (MWI)
- Telephony features such as hold/resume, call coverage, transfers, and conferences.

Serviceability testing focused on verifying the ability of Allworx 6x system and Allworx IP phones to recover from adverse conditions such as network and server (i.e. Allworx 6x, Session Manager, and Communication Manager) outages.

2.2. Test Results

All test cases were executed and passed with the following exceptions/observations noted:

- If an Allworx phone dials an enterprise Avaya phone, and the call is then forwarded back an Allworx phone, the call is established; however, there is no audio on the call.
- The Allworx phones do not update display information for various call scenarios (e.g. call forwarding, transfers, and conferences). Instead of updating the display information to indicate the Allworx phone is now in a conference or connected to a new party (e.g. after the call has been forwarded or transferred), the Allworx phone continues to display the original connected party information.
- Bridging between Allworx phones and Avaya phones is not currently supported.
- For calls from Allworx phones to enterprise Avaya phones, leading digits (e.g. *8) had to be dialed first to indicate that the call would be an enterprise call and should be routed to Session Manager.
- When an Allworx phone receives a call from an enterprise Avaya phone, the Allworx 6x system adds an “outside line” prefix (e.g. 9) to the Avaya phone’s extension. For example, if Avaya phone extension 53005 calls an Allworx phone, 953005 will appear on the Allworx phone’s display and in the phone’s call history log file. The Allworx 6x system does not designate calls it receives from Session Manager as internal enterprise calls. Therefore, if the extension of the caller does not match the Allworx 6x system’s own dial plan for the Allworx phones, it assumes the call is from outside the enterprise. As a result, it adds a 9 with the logic that the Allworx phone would have to dial a leading 9 in order to call the caller back. However, using the example above, dialing 953005 from an Allworx phone would not be routed to the Avaya phone. Rather *853005 would need to be dialed, as described in the previous bullet.

2.3. Support

Technical support on the Allworx 6x system and Allworx IP phones can be obtained through the following:

- **Phone:** 1-866-ALLWORX, option 1, option 3
- **Web:** http://www.allworx.com/support/support_overview.aspx
- **Email:** support@allworx.com

3. Reference Configuration

As shown in **Figure 1**, a simulated enterprise site was configured consisting of the following:

- Avaya Aura® Session Manager (configured using Avaya Aura® System Manager)
- Avaya Aura® Communication Manager with an Avaya G450 Media Gateway
- Avaya SIP and non-SIP phones (extensions 5xxxx)
- Allworx 6x system
- Allworx IP Phones (extensions 7xxx)

The Allworx 6x system's WAN interface was used to establish a SIP trunk to Session Manager. The Allworx 6x system's LAN interface was used to establish network connectivity to the Allworx IP Phones. Note, the Allworx IP Phones registered with the Allworx 6x system, not with Session Manager or Communication Manager.

The enterprise also had connectivity to the PSTN via Communication Manager.

The administration routing and basic connectivity between Communication Manager and Session Manager are not the focus of these Application Notes; however, some details are provided only for reference and completeness.

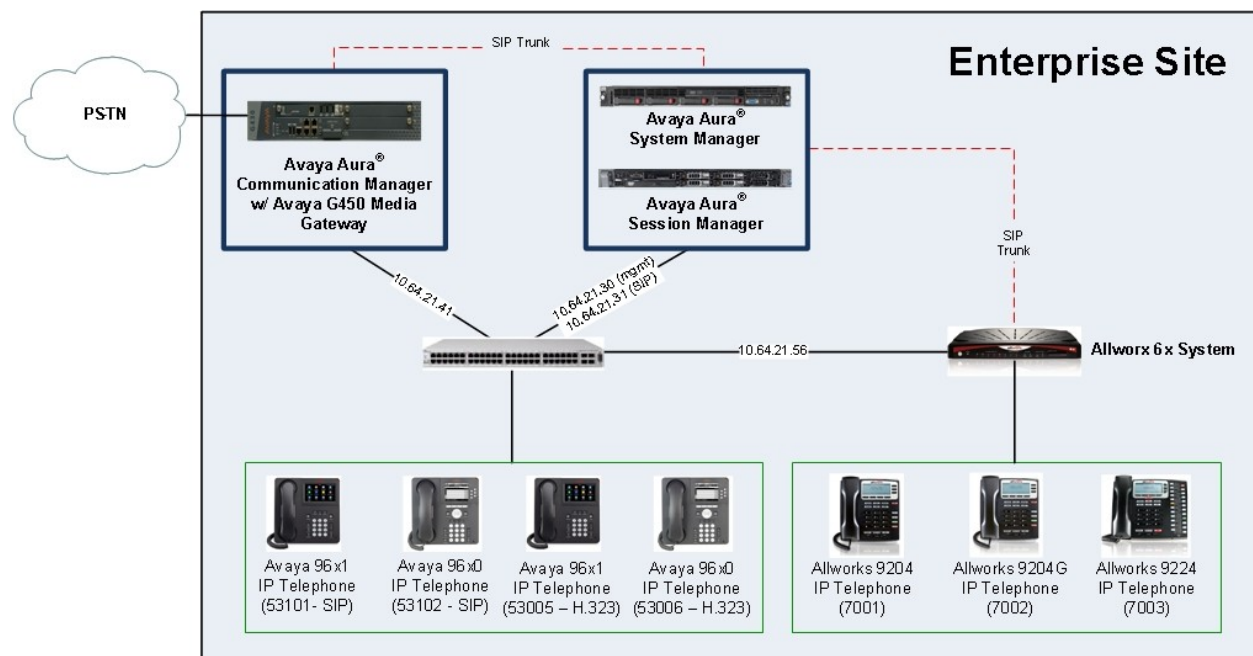


Figure 1: Allworx 6x system with Allworx IP Phones, Session Manager, and Avaya Communication Manager

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8300D Server with an Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.0.1, R016x.00.1.510.1, Update 19303 (Avaya Aura® System Platform: 6.0.3.4.3)
Dell™ PowerEdge™ R610 Server	Avaya Aura® System Manager: 6.1.0 (Build No. – 6.1.0.0.7345-6.1.5.502), Software Update Revision No : 6.1.9.1.1634 (Avaya Aura® System Platform: 6.0.3.4.3)
HP ProLiant DL360 G7 Server	Avaya Aura® Session Manager 6.1.5.0.615006
Avaya 9600 Series IP Telephones (96x0) <ul style="list-style-type: none">• H.323• SIP	3.1 Service Pack 2 2.6 Service Pack 5
Avaya 9600 Series IP Telephones (96x1) <ul style="list-style-type: none">• H.323• SIP	6.0 Service Pack 5 6.0 Service Pack 1
Allworx 6x	7.3.7.2
Allworx IP Phones <ul style="list-style-type: none">• 9204• 9204G• 9224	2.3.2.9

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify Communication Manager License
- Administer IP Codec Set
- Administer IP Network Region
- Administer IP Node Names
- Administer SIP Signaling Group
- Administer SIP Trunk Group
- Administer Route Pattern
- Administer Private Numbering
- Administer AAR Analysis

5.1. Verify Avaya Aura® Communication Manager License

Log in to 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.

display system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES	USED	
Maximum Administered H.323 Trunks:	12000	32
Maximum Concurrently Registered IP Stations:	18000	12
Maximum Administered Remote Office Trunks:	12000	0
Maximum Concurrently Registered Remote Office Stations:	18000	0
Maximum Concurrently Registered IP eCons:	414	0
Max Concur Registered Unauthenticated H.323 Stations:	100	0
Maximum Video Capable Stations:	18000	0
Maximum Video Capable IP Softphones:	18000	1
Maximum Administered SIP Trunks:	24000	170
Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0
Maximum Number of DS1 Boards with Echo Cancellation:	522	0
Maximum TN2501 VAL Boards:	128	0
Maximum Media Gateway VAL Sources:	250	1
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	0
Maximum Number of Expanded Meet-me Conference Ports:	300	0
(NOTE: You must logoff & login to effect the permission changes.)		

5.2. Administer IP Codec Set

Use the “change ip-codec-set x” command, where “x” is an existing codec set number that will be used for integration with the Allworx 6x system and Allworx IP Phones. Enter the audio codec types **G.711MU** and **G.729A** in the **Audio Codec** fields. Refer to the Allworx documentation for details on how to configure matching codecs with the Allworx equipment.

change ip-codec-set 1

Page 1 of 2

IP Codec Set

Codec Set: 1

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1:	G.711MU	n	2	20
2:	G.729A	n	2	20
3:				
4:				
5:				
6:				
7:				

Media Encryption

1: none

2:

3:

5.3. Administer IP Network Region

Use the “change ip-network-region x” command, where “x” is an existing network region that will be used for integration with the Allworx 6x system and Allworx IP Phones. Enter an **Authoritative Domain** (e.g. *avaya.com*). For the **Codec Set** field, enter the codec set number from **Section 5.2**. Set **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** to *yes*.

change ip-network-region 1		Page 1 of 20
IP NETWORK REGION		
Region: 1		
Location: Authoritative Domain: avaya.com		
Name:		
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		
Call Control PHB Value: 46		
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5		
H.323 IP ENDPOINTS		AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y		RSVP Enabled? n
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

5.4. Administer IP Node Names

Use the “change node-names ip” command, and add an entry for the Session Manager signaling interface. In this case, *SM_21_31* and *10.64.21.31* are entered as **Name** and **IP Address**. Note the *procr* / *10.64.21.41* entries, which are the node name / IP address for the processor board. It will be used later to configure the SIP trunk to Session Manager.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
AES_21_46	10.64.21.46	
CM_20_40	10.64.20.40	
CM_22_12_CLAN1A	10.64.22.16	
CM_22_12_CLAN2A	10.64.22.19	
IPO_21_64	10.64.21.64	
SM_20_31	10.64.20.31	
SM_21_31	10.64.21.31	
default	0.0.0.0	
msgserver	10.64.21.41	
procr	10.64.21.41	
procr6	::	

5.5. Administer SIP Signaling Group

Administer a SIP signaling group for a new trunk that will be created for the connection between Communication Manager and Session Manager. Use the “add signaling-group x” command, where “x” is an available signaling group number. Enter the following values for the specified fields and the default values may be used for the remaining fields.

- **Group Type:** *sip*
- **IMS Enabled?** *n*
- **Transport Method:** *tls*
- **Peer Detection Enabled?** *y*
- **Peer Server:** *SM* (this field will be automatically populated)
- **Near-end Node Name:** Processor node name from **Section 5.4**, i.e. *procr*
- **Near-end Listen Port:** *5061*
- **Far-end Node Name:** Session Manager node name from **Section 5.4**, i.e. *SM_21_31*
- **Far-end Listen Port:** *5061*
- **Far-end Network Region:** The IP network region number from **Section 5.4**, i.e. *1*
- **DTMF over IP:** *rtp-payload*
- **Direct IP-IP Audio Connections:** *y*

add signaling-group 1		Page 1 of 1
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n	SIP Enabled LSP? n	
IP Video? y	Priority Video? n	Enforce SIPS URI for SRTP? n
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: procr	Far-end Node Name: SM_21_31	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
Far-end Network Region: 1		
Far-end Domain:		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? y	
Alternate Route Timer(sec): 20		

5.6. Administer SIP Trunk Group

Administer a SIP trunk group to interface with the Session Manager. Use the “add trunk-group x” command, where “x” is an available trunk group number. Set the **Group Type** to *sip*, and **Service Type** to *tie*. Enter a descriptive **Group Name**, and an available trunk access code for the **TAC** field. Set **Member Assignment Method** to *auto*, **Signaling Group** to the signaling group number from **Section 5.5**, and enter a desired value for number of trunk group members for **Number of Members**.

add trunk-group 1		Page 1 of 21	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: to SM_21_31	COR: 1	TN: 1	TAC: 101
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Member Assignment Method: auto	
		Signaling Group: 1	
		Number of Members: 50	

Navigate to **Page 3**, and enter *unk-pvt* for the **Numbering Format** field as shown below.

change trunk-group 1		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: unk-pvt		UI Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			

5.7. Administer Route Pattern

Create a route pattern to use for the newly created SIP trunk group. Use the “change route-pattern x” command, where “x” is an available route pattern. Enter a descriptive **Pattern Name**, i.e. **to SM_21_31**. In the **Grp No** field, enter the trunk group number from **Section 5.6**. In the **FRL** field, enter a level that allows access to this trunk.

change route-pattern 1														Page 1 of 3		
Pattern Number: 1														Pattern Name: to SM_21_31		
SCCAN? n														Secure SIP? n		
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted							DCS/	IXC	
No			Mrk	Lmt	List	Del	Digits							QSIG		
							Dgts							Intw		
1:	1	0						0							n	user
2:															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	LAR		
		0	1	2	M	4	W	Request				Dgts	Format			
												Subaddress				
1:	y	y	y	y	y	n	n	rest				lev0-pvt	none			
2:	y	y	y	y	y	n	n	rest				none				
3:	y	y	y	y	y	n	n	rest				none				
4:	y	y	y	y	y	n	n	rest				none				
5:	y	y	y	y	y	n	n	rest				none				
6:	y	y	y	y	y	n	n	rest				none				

5.8. Administer Private Numbering

Use the “change private-numbering 0” command, to define the calling party number to send to Session Manager. Add an entry for the trunk group defined in **Section 5.6**. In the example shown below, all calls originating from a 5-digit extension beginning with 5 and routed over any trunk group, since the Trk Grp(s) field is blank, will result in a 5-digit calling number. The calling party number will be in the SIP “From” header.

change private-numbering 0					Page 1 of 2	
NUMBERING - PRIVATE FORMAT						
Ext	Ext	Trk	Private	Total		
Len	Code	Grp(s)	Prefix	Len		
5	2			5	Total Administered: 2	
5	5			5	Maximum Entries: 540	

5.9. Administer Automatic Alternate Routing Analysis

This section provides a sample Automatic Alternate Routing (AAR) routing used for routing calls with dialed digits 7xxx to Session Manager. Note that other methods of routing may be used. Use the “change aar analysis “7” command, and add an entry to specify how to route calls to 7xxx. In the example shown below, calls with digits 7xxx will be routed as an AAR call using route pattern “1” from **Section 5.7**. These calls will be routed to Session Manager and then to the Allworx 6x system for delivery to the Allworx IP Phones.

change aar analysis 7							Page 1 of 2		
AAR DIGIT ANALYSIS TABLE									
Location: all							Percent Full: 1		
	Dialed	Total		Route	Call	Node	ANI		
	String	Min	Max	Pattern	Type	Num	Reqd		
7		4	4	1	aar		n		

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager as shown in the reference configuration. All provisioning for Session Manager is performed via the System Manager web interface. System Manager delivers a set of shared, secure management services and a common console across multiple products in the Avaya Aura® network, including the central administration of routing policies, and a common format for logs and alarms.

The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

The Session Manager server provides the network interface for all inbound and outbound SIP signaling to all provisioned SIP entities. During compliance testing, the IP address assigned to the SIP signaling interface was 10.64.21.31 as specified in **Figure 1**. The Session Manager server also has a separate network interface used for connectivity to System Manager for provisioning Session Manager. The IP address assigned to the Session Manager management interface was 10.64.21.30.


The procedures described in this section include configurations for the following:

- **SIP Domains** – SIP Domains are the domains for which Session Manager is authoritative in routing SIP calls. In other words, for calls to such domains, Session Manager applies Network Routing Policies to route those calls to SIP Entities. For calls to other domains, Session Manager routes those calls to another SIP proxy (either a pre-defined default SIP proxy or one discovered through DNS).
- **Locations** – Locations define the physical and/or logical locations in which SIP Entities reside. Call Admission Control (CAC) / bandwidth management may be administered for each location to limit the number of calls to and from a particular Location.
- **Adaptations** – Adaptations are used to apply any necessary protocol adaptations, e.g., modify SIP headers, and apply any necessary digit conversions for the purpose of inter-working with specific SIP Entities.
- **SIP Entities** – SIP Entities represent SIP network elements such as Session Manager instances, Communication Manager systems, Session Border Controllers, SIP gateways, SIP trunks, and other SIP network devices.
- **Entity Links** – Entity Links define the SIP trunk/link parameters, e.g., ports, protocol (UDP/TCP/TLS), and trust relationship, between Session Manager instances and other SIP Entities.
- **Time Ranges** – Time Ranges specify customizable time periods, e.g., Monday through Friday from 9AM to 5:59PM, Monday through Friday 6PM to 8:59AM, all day Saturday and Sunday, etc. A Network Routing Policy may be associated with one or more Time Ranges during which the Network Routing Policy is in effect.
- **Routing Policies** – Routing Policies are used in conjunction with a Dial Patterns to specify a SIP Entity that a call should be routed to.
- **Dial Patterns** – A Dial Pattern specifies a set of criteria and a set of Network Routing Policies for routing calls that match the criteria. The criteria include the called party number

and SIP domain in the Request-URI, and the Location from which the call originated. For example, if a call arrives at Session Manager and matches a certain Dial Pattern, then Session Manager selects one of the Network Routing Policies specified in the Dial Pattern. The selected Network Routing Policy in turn specifies the SIP Entity to which the call is to be routed.

Access the Session Manager administration web interface by entering **https://<ip-addr>/SMGR** as the URL in an Internet browser, where <ip-addr> is the IP address of the System Manager server.

Log in using appropriate credentials. The main page for the administrative interface is shown below.



Avaya Aura® System Manager 6.1

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

Users

Administrators

Manage Administrative Users

Groups & Roles

Manage groups, roles and assign roles to users

Synchronize and Import

Synchronize users with the enterprise directory, import users from file

User Management

Manage users, shared user resources and provision users

Elements

Application Management

Manage applications and application certificates

Communication Manager

Manage Communication Manager objects

Conferencing

Conferencing

Inventory

Manage, discover, and navigate to elements, update element software

Messaging

Manage Messaging System objects

Presence

Presence

Routing

Network Routing Policy

Session Manager

Session Manager Element Manager

SIP AS 8.1

SIP AS 8.1

Services

Backup and Restore

Backup and restore System Manager database

Configurations

Manage system wide configurations

Events

Manage alarms, view and harvest logs

Licenses

View and configure licenses

Replication

Track data replication nodes, repair replication nodes

Scheduler

Schedule, track, cancel, update and delete jobs

Security

Manage Security Certificates

Templates

Manage Templates for Communication Manager and Messaging System objects

6.1. Specify SIP Domain

Navigate to **Home → Elements → Routing → Domains**, and click the **New** button (not shown) to add the SIP domain with the following:

- **Name:** *avaya.com* (as set in **Section 5.3**)
- **Type:** *sip*
- **Notes:** optional descriptive text

Click **Commit** to save the configuration.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) × [Home](#)

Home / Elements / Routing / Domains- Domain Management

[Help ?](#)

[Commit](#) [Cancel](#)

1 Item [Refresh](#) Filter: Enable

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

* Input Required [Commit](#) [Cancel](#)

6.2. Add Location

Locations identify logical and/or physical locations where SIP entities reside. Only one Location was configured for compliance testing.

Navigate to **Home → Elements → Routing → Locations** and click the **New** button (not shown) to add the Location. Enter the following information:

Under **General**:

- **Name:** a descriptive name
- **Notes:** optional descriptive text

Under **Location Pattern**, click the **Add** button to add a new line:

- **IP Address Pattern:** **10.64.21.***
- **Notes:** optional descriptive text

Click **Commit** to save the configuration.

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[Routing](#) * [Home](#)

Home / Elements / Routing / Locations- Location Details

Location Details [Help ?](#)

[Commit](#) [Cancel](#)

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

General

* **Name:**

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units:

Total Bandwidth:

Per-Call Bandwidth Parameters

* **Default Audio Bandwidth:**

Location Pattern

[Add](#) [Remove](#)

2 Items [Refresh](#) [Filter: Enable](#)

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.64.26.*	<input type="text"/>
<input type="checkbox"/>	* 10.64.21.*	<input type="text"/>

Select : All, None

* Input Required [Commit](#) [Cancel](#)

6.3. Add Allworx Adaptation

For calls from an Allworx IP Phone towards Communication Manager, an adaptation was created to change the domain in the *From* header from the Allworx 6x IP address to **avaya.com**.

To create an adaptation, navigate to **Home → Elements → Routing → Adaptations** and click the **New** button (not shown). Enter the following information:

- **Adaptation name:** a descriptive name
- **Module name:** select from the drop-down menu or enter *DigitConversionAdapter*
- **Module parameter:** enter *fromto=true iodstd=avaya.com ioscrd=avaya.com*
- **Notes:** optional descriptive text

Click the **Commit** button.

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[Routing](#) [Home](#)

Home / Elements / Routing / Adaptations - Adaptation Details

[Help ?](#) [Commit](#) [Cancel](#)

Adaptation Details

General

* **Adaptation name:**

Module name:

Module parameter:

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

[Add](#) [Remove](#)

0 Items [Refresh](#) Filter: [Enable](#)

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
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Digit Conversion for Outgoing Calls from SM

[Add](#) [Remove](#)

0 Items [Refresh](#) Filter: [Enable](#)

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
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* Input Required [Commit](#) [Cancel](#)

6.4. Add SIP Entities

A SIP Entity must be added for Avaya Aura® Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration, a SIP Entity is added for the Session Manager, Communication Manager, and the Allworx 6x system.

Note, the Session Manager SIP Entity is assumed to have already been configured. Navigate to **Home → Elements → Routing → SIP Entities**, check the checkbox for the Session Manager SIP Entity, and click the **Edit** button (not shown). Under the **Ports** section, verify the required Session Manager listening port for communication with Allworx is configured (i.e. **Port 5060** and **Protocol UDP**). If necessary, click the **Add** button to add the listening port and then click the **Commit** button when done to save the changes.

Port

6 Items | [Refresh](#)

Filter: [Enable](#)

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	UDP <input type="button" value="v"/>	avaya.com <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	TCP <input type="button" value="v"/>	avaya.com <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	TLS <input type="button" value="v"/>	avaya.com <input type="button" value="v"/>	<input type="text"/>

To add a SIP Entity, navigate to **Home → Elements → Routing → SIP Entities** and click the **New** button (not shown).

The configuration details for the SIP Entity defined for the Communication Manager are below:

Under **General**:

- **Name**: a descriptive name
- **FQDN or IP Address**: *10.64.21.41* as specified in **Figure 1**
- **Type**: select *CM*
- **Location**: select the location configured in **Section 6.2**

Default settings can be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

The screenshot displays the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help | About | Change Password | Log off admin". Below the navigation bar, there are tabs for "Routing" and "Home". The left sidebar shows a tree view with "Routing" expanded, containing sub-items: Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "Home / Elements / Routing / SIP Entities- SIP Entity Details". It features a "SIP Entity Details" section with a "General" tab. The "General" tab contains the following fields: "Name" (CM_21_41), "FQDN or IP Address" (10.64.21.41), "Type" (CM), "Notes" (Mike - Evolution Server - 8300D), "Adaptation" (dropdown), "Location" (.21 &.26 Subnets), "Time Zone" (America/Denver), "Override Port & Transport with DNS SRV" (checkbox), "SIP Timer B/F (in seconds)" (4), "Credential name" (text field), "Call Detail Recording" (both), and "SIP Link Monitoring" (Use Session Manager Configuration). Buttons for "Commit" and "Cancel" are located at the top right of the configuration area.

The following screen shows addition of the Allworx 6x SIP Entity. Note the selection of ***Other*** for **Type**.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) * [Home](#)

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

SIP Entity Details [Help ?](#)

[Commit](#) [Cancel](#)

General

* **Name:** Allworx 6x

* **FQDN or IP Address:** 10.64.21.56

Type: Other

Notes:

Adaptation: Allworx

Location: .21 &.26 Subnets

Time Zone: America/Denver

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

6.5. Add Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity link. Two Entity Links were created:

- Session Manager ↔ Communication Manger
- Session Manager ↔ Allworx 6x

Navigate to **Home → Elements → Routing → Entity Links**, and click the **New** button (not shown) to add a new Entity Link. The screen below shows the configuration details for the Entity Link connecting Session Manager with Communication Manager.

- **Name**: a descriptive name
- **SIP Entity 1**: select the Session Manager SIP Entity.
- **Protocol**: select **TLS** as the transport protocol
- **Port**: **5061**. This is the port number to which the other system sends SIP requests
- **SIP Entity 2**: select the Communication Manager SIP Entity
- **Port**: **5061**. This is the port number on which the other system receives SIP requests
- **Connection Policy**: select **Trusted**
- **Notes**: optional descriptive text

Click **Commit** to save the configuration.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Entity Links - Entity Links


Entity Links Help ? Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* CM_21_41	* SM_21_31	TLS	* 5061	* CM_21_41	* 5061	Trusted	

* Input Required Commit Cancel

The Entity Link for connecting Session Manager with Allworx 6x was similarly defined as shown in the screen below. Note the use of **UDP** and port **5060**.



Avaya Aura® System Manager 6.1
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing *
Home

- ▼ Routing
- Domains
- Locations
- Adaptations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns
- Regular Expressions
- Defaults

Home / Elements / Routing / Entity Links- Entity Links

Entity Links
[Help ?](#)

Commit
Cancel

1 Item Refresh
Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* to Allworx	* SM_21_31	UDP	* 5060	* Allworx 6x	* 5060	Trusted	

* Input Required

Commit
Cancel


6.6. Add Time Ranges

Before adding routing policies (configured in next step), time ranges must be defined during which the routing policies will be active. One time range was defined that would allow routing to occur at any time.

Navigate to **Elements → Routing → Time Ranges**, and click the **New** button to add a new Time Range. Enter the following information:

- **Name:** a descriptive name
- **Mo through Su:** check the box under each of these headings
- **Start Time:** enter **00:00**
- **End Time:** enter **23:59**
- **Notes:** optional descriptive text

Click **Commit** to save this time range. The screen below shows the configured Time Range.

 Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing

Home

Routing

Home

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Time Ranges- Time Ranges

Help ?

Time Ranges

Edit

New

Duplicate

Delete

More Actions

1 Item

Refresh

Filter: Enable

<input type="checkbox"/>	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
<input type="checkbox"/>	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

6.7. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities connected to the Session Manager. One routing policy must be added for routing calls to Communication Manager and one for routing calls to the Allworx 6x system.

Navigate to **Home → Elements → Routing → Routing Policies**, and click the **New** button (not shown) to add a new Routing Policy. Enter the following information:

Under **General**:

- **Name**: a descriptive name
- **Notes**: optional descriptive text

Under **SIP Entity as Destination**

Click **Select** to select the appropriate SIP Entity to which the routing policy applies (not shown).

Under **Time of Day**

Click **Add** to select the Time Range configured in the previous step (not shown).

Default settings can be used for the remaining fields. Click **Commit** to save the configuration.

The following screen shows the Routing Policy for routing calls to Communication Manager.

The screenshot displays the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.1', and links for 'Help | About | Change Password | Log off admin'. The breadcrumb trail is 'Home / Elements / Routing / Routing Policies - Routing Policy Details'. The left sidebar shows a tree view with 'Routing' expanded, containing sub-items like Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (highlighted), Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Routing Policy Details' and includes 'Commit' and 'Cancel' buttons. It is divided into three sections: 'General' with fields for 'Name' (set to 'to CM_21_41'), 'Disabled' (checkbox), and 'Notes'; 'SIP Entity as Destination' with a 'Select' button and a table listing the destination 'CM_21_41' with FQDN '10.64.21.41' and Type 'CM'; and 'Time of Day' with 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. Below this is a table with 1 item, showing a ranking of 1, name '24/7', and a time range from 00:00 to 23:59. The bottom of the interface has a 'Select : All, None' option.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) × [Home](#)

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

[Help ?](#)

Routing Policy Details [Commit](#) [Cancel](#)

General

* **Name:**

Disabled: ☐

Notes:

SIP Entity as Destination

[Select](#)

Name	FQDN or IP Address	Type	Notes
CM_21_41	10.64.21.41	CM	Mike - Evolution Server - 8300D

Time of Day


[Add](#) [Remove](#) [View Gaps/Overlaps](#)

1 Item [Refresh](#) [Filter: Enable](#)

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

Select : All, None

The following screen shows the Routing Policy for routing calls to the Allworx 6x system.


Avaya Aura® System Manager 6.1
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) * [Home](#)

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

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

Routing Policy Details
[Help ?](#)

Commit
Cancel

General

* Name:
to Allworx

Disabled:
☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Allworx 6x	10.64.21.56	Other	

Time of Day

Add
Remove
View Gaps/Overlaps

1 Item | Refresh
Filter: Enable

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

Select : All, None

6.8. Add Dial Patterns

Define dial patterns to direct calls to the appropriate SIP Entity.

Navigate to **Home → Elements → Routing → Dial Patterns**, and click the **New** button (not shown) to add a new Dial Pattern. Enter the following information to route calls that match the pattern **5xxxx** to Communication Manager.

Under **General**:

- **Pattern**: dialed number or prefix
- **Min**: minimum length of dialed number
- **Max**: maximum length of dialed number
- **SIP Domain**: select the SIP Domain created in **Section 6.1** (or select **–ALL–** to be less restrictive)
- **Notes**: optional descriptive text

Under **Originating Locations and Routing Policies**

Click **Add** to select the appropriate originating Location and Routing Policy from the list (not shown).

Default settings can be used for the remaining fields. Click **Commit** to save the configuration.

[Routing](#) x [Home](#)

[Home](#) / [Elements](#) / [Routing](#) / [Dial Patterns](#) - Dial Pattern Details

▼ Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

[Help ?](#)

Commit

Cancel

Dial Pattern Details

General

*** Pattern:**

*** Min:**

*** Max:**

Emergency Call:

☐

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add

Remove

Filter: Enable

1 Item
[Refresh](#)

<input type="checkbox"/>	Originating Location Name ¹ ▲	Originating Location Notes	Routing Policy Name	Rank ² ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	to CM_21_41	1	<input type="checkbox"/>	CM_21_41	

Select : All, None

Denied Originating Locations

Add

Remove

Filter: Enable

0 Items
[Refresh](#)

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

*** Input Required**

Commit

Cancel

MJH; Reviewed:
SPOC 2/20/2012

Solution & Interoperability Test Lab Application Notes
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27 of 53
Allworx_SM61

Enter the following information to route calls that match the pattern **7xxx** to the Allworx 6x system. The Allworx 6x system will then route the calls to the Allworx IP Phones.



[Routing](#) * [Home](#)

[Home](#) / [Elements](#) / [Routing](#) / [Dial Patterns](#) - Dial Pattern Details

- ▼ Routing
- Domains
- Locations
- Adaptations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns
- Regular Expressions
- Defaults

[Commit](#) [Cancel](#) [Help ?](#)

Dial Pattern Details

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

[Add](#) [Remove](#)
1 Item [Refresh](#)
Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	to Allworx	0	<input type="checkbox"/>	Allworx 6x	

Select : All, None

Denied Originating Locations

[Add](#) [Remove](#)
0 Items [Refresh](#)
Filter: Enable

<input type="checkbox"/>	Originating Location	Notes
<input type="checkbox"/>		

* Input Required
[Commit](#) [Cancel](#)

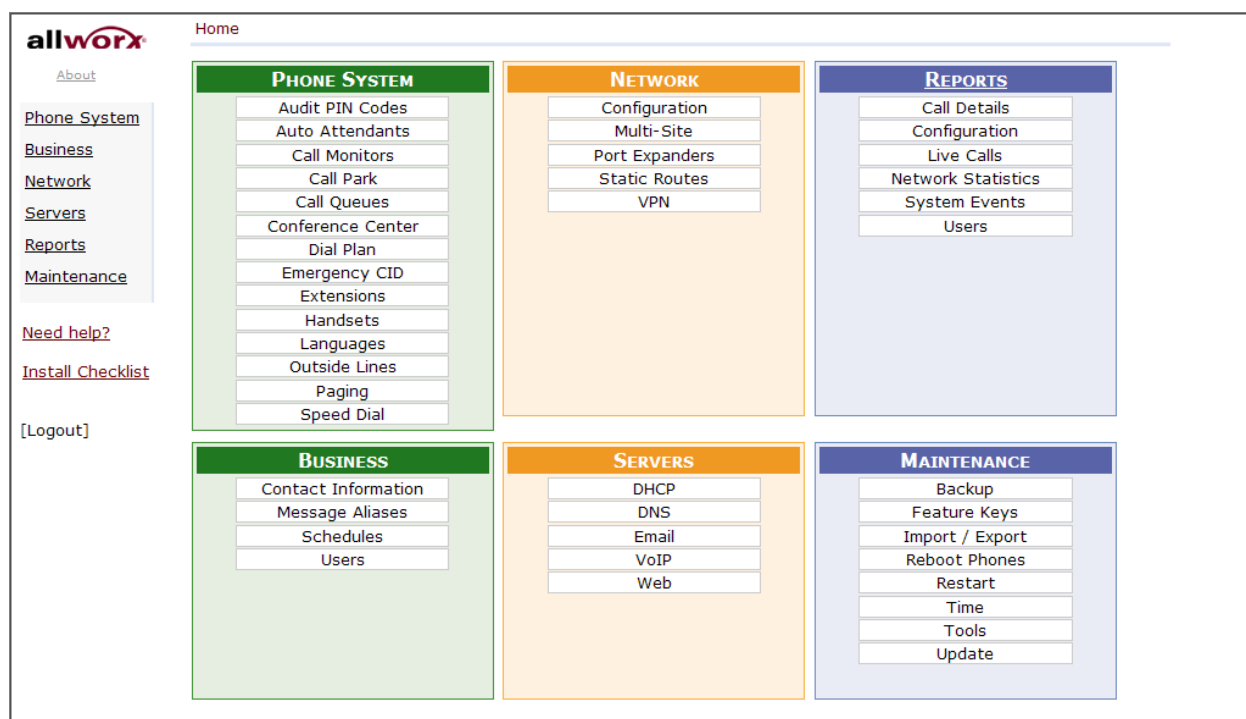
7. Configure Allworx 6x System and Allworx IP Phones

This section provides the procedures for configuring the Allworx 6x system and Allworx IP Phones as shown in the reference configuration.

7.1. Access Allworx Administration Web Pages

Access the Allworx 6x administration web interface by entering **http://<ip-addr>** as the URL in an Internet browser, where <ip-addr> is the IP address of the Allworx 6x system. Log in using appropriate credentials.

After logging in, the following **Home** screen is displayed.



7.2. Configure SIP Gateway

From the Home page, navigate to **Phone System → Outside Lines**.

allworx
[About](#)
[Phone System](#)
[Business](#)
[Network](#)
[Servers](#)
[Reports](#)
[Maintenance](#)

[Need help?](#)
[Install Checklist](#)

[Logout]

Home

PHONE SYSTEM
Audit PIN Codes
Auto Attendants
Call Monitors
Call Park
Call Queues
Conference Center
Dial Plan
Emergency CID
Extensions
Handsets
Languages
Outside Lines
Paging
Speed Dial

NETWORK
Configuration
Multi-Site
Port Expanders
Static Routes
VPN

REPORTS
Call Details
Configuration
Live Calls
Network Statistics
System Events
Users

BUSINESS
Contact Information
Message Aliases
Schedules
Users

SERVICES
DHCP
DNS
Email
VoIP
Web

MAINTENANCE
Backup
Feature Keys
Import / Export
Reboot Phones
Restart
Time
Tools
Update

The **Outside Lines** page is displayed as shown below. In the **SIP Gateways** section, click the **add new SIP Gateway** link.

allworx
About

Phone System
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[Auto Attendants](#)
[Call Monitors](#)
[Call Park](#)
[Call Queues](#)
[Conference Center](#)
[Dial Plan](#)
[Emergency CID](#)
[Extensions](#)
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Outside Lines
[Paging](#)
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[Business](#)
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[Need help?](#)
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Home > [Phone System](#) > [Outside Lines](#)

[Analog \(CO\) Lines](#) [DID Blocks](#) [DID Routing Plans](#) [SIP Gateways](#) [SIP Proxies](#)

Anonymous Call Handling [modify](#)
Anonymous Calls are routed normally.

Analog (CO) Lines

Analog (CO) Line	Type	Port	Action
	---	01	New FXO Line
	---	02	New FXO Line
	---	03	New FXO Line
	---	04	New FXO Line
	---	05	New FXO Line
	---	06	New FXO Line

Direct Inward Dial Blocks [add new DID Block](#)
No DID Blocks have been defined.

Direct Inward Dial Routing Plans
No DID Routing Plans have been defined. (new plans can be created when a DID Block is added or modified)

SIP Gateways [add new SIP Gateway](#)
No SIP Gateways have been defined.

SIP Proxies [add new SIP Proxy](#)
No SIP Proxies have been defined.

Enter the following information:

Under **SIP Gateway**:

- **Description**: enter descriptive text

Under **SIP Registration**:

- Check the radio button for **Gateway uses static IP Address**:
 - **IP Address**: enter the SIP signaling IP address of Session Manager
 - **SIP Port**: enter the SIP signaling port used by Session Manager

Click the **Add** button.

The screenshot shows the Allworx web interface for adding a new SIP Gateway. The breadcrumb trail at the top reads: Home > Phone System > Outside Lines > New SIP Gateway. On the left is a navigation menu with links for About, Phone System, Audit PIN Codes, Auto Attendants, Call Monitors, Call Park, Call Queues, Conference Center, Dial Plan, Emergency CID, Extensions, Handsets, Languages, Outside Lines, Paging, Speed Dial, Business, Network, Servers, Reports, and Maintenance. The main content area is titled "SIP Gateway" and contains two sections: "SIP Gateway" and "SIP Registration". In the "SIP Gateway" section, the "Description" field is set to "Session Manager" and the "Number of Line Appearances" is set to 0. In the "SIP Registration" section, the radio button for "Gateway uses static IP Address" is selected. Below this, the "IP Address" field is set to 10.64.21.31 and the "SIP Port" field is set to 5060. At the bottom of the form are "Add" and "Cancel" buttons. Red boxes highlight the "Description" field, the "Gateway uses static IP Address" radio button, and the "IP Address" and "SIP Port" fields.

allworx® Home > Phone System > Outside Lines > New SIP Gateway

About

Phone System

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- [Auto Attendants](#)
- [Call Monitors](#)
- [Call Park](#)
- [Call Queues](#)
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- [Maintenance](#)

[Need help?](#)

SIP Gateway

Description Session Manager

Number of Line Appearances 0 (0 to 99, should not exceed number of CO lines attached to gateway)

SIP Registration

☐ Gateway uses SIP Registration

Login ID

Password (maximum 40 characters)

☒ Gateway uses static IP Address

IP Address 10.64.21.31

SIP Port 5060

Add Cancel

On the **Outside Lines** page, click the **Modify** link (not shown) for the SIP Gateway just added.

Under the **Call Route** section at the bottom of the screen, click the radio box for **Routed using DID Block(s)** and click the **Update** button.

allworx Home > Phone System > Outside Lines > Modify SIP Gateway

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Business

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SIP Gateway ?

Description

Caller ID Name (up to 47 characters: letters digits , \ _ ' -)

☐ Use External Caller ID Name from handset (if specified)

☐ Use Caller ID Name from external sources (if received)

Caller ID Number (up to 24 digits)

☐ Use External Caller ID Number from handset (if specified)

☐ Use Caller ID Number from external sources (if received)

Number of Line Appearances (0 to 99, should not exceed number of CO lines attached to gateway)

Default Language

SIP Registration ?

☐ Gateway uses SIP Registration

Login ID

Password (maximum 40 characters)

☒ Gateway uses static IP Address

IP Address

SIP Port

Advanced Settings ?

☐ Pad DTMF RTP Packets

☒ Enable Early Media (allow audio from 183 Session Progress responses)

☒ Supports SIP REFER (when calls from this gateway are transferred back to this gateway)

☐ Supports SIP Redirect (when call requests from this gateway are routed back to the gateway)

☐ Use E.164 format for phone numbers

☒ Offer '100rel' support (RFC 3262 - PRACK)

Obtain DID/DNIS number from

Use in Request URI of outbound calls

Features ?

Prefix String (digits/characters sent by the Allworx to gateway before sending number dialed)

Default Auto Attendant

Select the attendant used to answer when calls received from this source are routed to an Auto Attendant.

Call Route

Calls received from this SIP Gateway go to:

☐ Extension

☐ Auto Attendant

☐ Voicemail for user

☒ Routed using DID Block(s): No DID Blocks have been defined!

7.3. Configure DID Block

In the **Direct Inward Dial Blocks** section of the **Outside Lines** page shown in **Section 7.2**, click the **add new DID block** link.

Enter the following:

- **Starting Phone Number:** enter the starting/first phone number of the Allworx IP Phones. Note, these are the phone numbers/extensions that will be used by Communication Manager and Session Manager to route calls to the Allworx IP phones.
- **Total number of phone number in the DID Block:** enter the number of phone numbers desired
- **DID Routing Plan:** select *make new Routing Plan* from the drop-down menu.

Click the **Add** button.

The screenshot shows the Allworx web interface for configuring a new DID block. The breadcrumb trail at the top reads: Home > Phone System > Outside Lines > New DID Block. On the left, there is a sidebar menu under the heading 'Phone System' with links to: About, Audit PIN Codes, Auto Attendants, Call Monitors, Call Park, Call Queues, Conference Center, Dial Plan, Emergency CID, Extensions, Handsets, Languages, Outside Lines, Paging, Speed Dial, Business, Network, Servers, Reports, and Maintenance. The main content area is titled 'DID Block' and contains the following fields:

- Starting Phone Number:** 7000 (include Area Code and Exchange)
- Total number of phone numbers in the DID Block:** 100
- DID Routing Plan:** make new Routing Plan (dropdown menu)

At the bottom of the form are two buttons: 'Add' and 'Cancel'.

7.4. Configure DID Routing Plan

On the **Outside Lines** page, click the **Details** link under the **Direct Inward Dialing Routing Plans** section.

allworx
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[Auto Attendants](#)
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[Analog \(CO\) Lines](#) [DID Blocks](#) [DID Routing Plans](#) [SIP Gateways](#) [SIP Proxies](#)

Anonymous Call Handling [modify](#)
Anonymous Calls are routed normally.

Analog (CO) Lines

Analog (CO) Line	Type	Port	Action
	---	01	New FXO Line
	---	02	New FXO Line
	---	03	New FXO Line
	---	04	New FXO Line
	---	05	New FXO Line
	---	06	New FXO Line

Direct Inward Dial Blocks [add new DID Block](#)

Block	Action
7000 Numbers: 100 Plan: Routing Plan 1	Modify Delete

Direct Inward Dial Routing Plans

Routing Plan	Action
Routing Plan 1	Details Delete

SIP Gateways [add new SIP Gateway](#)

Gateway	Action
Session Manager User ID: *5900 Gateway IP Address: 10.64.21.31:5060 (must assign DID Block)	Modify Delete

SIP Proxies [add new SIP Proxy](#)
No SIP Proxies have been defined.

Click the **modify** link next to **Routing Plan Information**.

allworx Home > Phone System > Outside Lines > DID Routing Plan

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Routing Plan Information [modify](#)

Description	Routing Plan 1
Default Extension	0 - Operator
Default DNIS Name	
Default Language	Use Source of call
DID Blocks using this plan	7000 / 100 numbers

Phone Number to Extension Mapping [add number to table](#)

All phone numbers in this plan use the Default Extension defined above. Click 'add number to table' to route a number to a specific extension.

Enter descriptive text for **Description** and click the **Update** button.

allworx Home > Phone System > Outside Lines > Modify DID Routing Plan

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- [Call Park](#)
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DID Routing Plan

Description Session Manager Routing

Default Extension 0 - Operator

Default DNIS Name (up to 47 characters: letters digits . , \ _ ' -)

Default Language Use Source of call

DID Blocks using this plan: 7000 / 100 numbers

Update **Cancel**

Click the **add number to table** link next to **Phone Number to Extension Mapping**.

allworx Home > Phone System > Outside Lines > DID Routing Plan

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[Need help?](#)

Routing Plan Information [modify](#)

Description	Session Manager Routing
Default Extension	0 - Operator
Default DNIS Name	
Default Language	Use Source of call
DID Blocks using this plan	7000 / 100 numbers

Phone Number to Extension Mapping [add number to table](#)

All phone numbers in this plan use the Default Extension defined above. Click 'add number to table' to route a number to a specific extension.

Select a **Phone Number(s)** and **Extension** from the drop-down menus to map the select phone number to the selected extension.

Click the **Update** button.

The screenshot shows the Allworx web interface. The breadcrumb trail at the top reads: Home > Phone System > Outside Lines > Modify DID Routing Plan. On the left is a navigation menu with the 'Phone System' section expanded, showing links to Audit PIN Codes, Auto Attendants, Call Monitors, Call Park, Call Queues, Conference Center, Dial Plan, Emergency CID, Extensions, Handsets, Languages, Outside Lines, Paging, Speed Dial, Business, Network, Servers, Reports, and Maintenance. The main content area is titled 'Add Phone Number(s) to Extension Mapping'. It contains four fields: 'Phone Number(s)' with a dropdown menu showing '(000) 000-7001', 'Extension' with a dropdown menu showing '7001 - New User One', 'DNIS Name' with a text input field and a note '(up to 47 characters: letters digits , , \ _ ' -)', and 'Default Language' with a dropdown menu showing 'Use Source of call'. At the bottom of the form are two buttons: 'Update' and 'Cancel'. The 'Update' button is highlighted with a red box.

Repeat the previous step for each phone number / extension mapping.

7.5. Modify SIP Gateway

On the **Outside Lines** page, click the **Modify** link (not shown) next to the SIP Gateway shown in **Section 7.2**. Under the **SIP Gateway** section at the top of the screen, check the **Use External Caller ID Name from handset** checkbox. Under the **Call Route** section at the bottom of the screen, check the checkbox for the DID block created in **Section 7.3**.

Click the **Update** button.

allworx

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Home > Phone System > Outside Lines > Modify SIP Gateway

SIP Gateway

DescriptionSession Manager

Caller ID Name

☒ Use External Caller ID Name from handset (if specified)

☐ Use Caller ID Name from external sources (if received)

Caller ID Number

(up to 24 digits)

☐ Use External Caller ID Number from handset (if specified)

☐ Use Caller ID Number from external sources (if received)

Number of Line Appearances0 (0 to 99, should not exceed number of CO lines attached to gateway)

Default LanguagePrimary Language

SIP Registration

☐ Gateway uses SIP Registration

Login ID

Password

☒ Gateway uses static IP Address

IP Address10.64.21.31

SIP Port5060

Advanced Settings

☐ Pad DTMF RTP Packets

☒ Enable Early Media (allow audio from 183 Session Progress responses)

☒ Supports SIP REFER (when calls from this gateway are transferred back to this gateway)

☐ Supports SIP Redirect (when call requests from this gateway are routed back to the gateway)

☐ Use E.164 format for phone numbers

☒ Offer '100rel' support (RFC 3262 - PRACK)

Obtain DID/DNIS number fromSIP To: header field

Use

dialed number

 in Request URI of outbound calls

Features

Prefix String

Default Auto Attendant

Select the attendant used to answer when calls received from this source are routed to an Auto Attendant.

Auto Attendant 1 (x*431)

Call Route

Calls received from this SIP Gateway go to:

☐ Extension

choose an extension

☐ Auto Attendant

☐ Voicemail for user

New User One (New1)

☒ Routed using DID Block(s):

check all uncheck all

☒ 7000 / 100 Numbers / Session Manager Routing

Update

Start Over

Cancel


7.6. Modify Handsets

Navigate back to the **Home** page shown below and click the **Handsets** link under **Phone System**.

The screenshot displays the Allworx Home page. On the left is a navigation sidebar with links: [About](#), [Phone System](#), [Business](#), [Network](#), [Servers](#), [Reports](#), [Maintenance](#), [Need help?](#), [Install Checklist](#), and [\[Logout\]](#). The main content area is titled "Home" and contains six categorized panels:

- PHONE SYSTEM** (Green header):
 - Audit PIN Codes
 - Auto Attendants
 - Call Monitors
 - Call Park
 - Call Queues
 - Conference Center
 - Dial Plan
 - Emergency CID
 - Extensions
 - Handsets** (highlighted with a red box)
 - Languages
 - Outside Lines
 - Paging
 - Speed Dial
- NETWORK** (Orange header):
 - Configuration
 - Multi-Site
 - Port Expanders
 - Static Routes
 - VPN
- REPORTS** (Blue header):
 - Call Details
 - Configuration
 - Live Calls
 - Network Statistics
 - System Events
 - Users
- BUSINESS** (Green header):
 - Contact Information
 - Message Aliases
 - Schedules
 - Users
- SERVERS** (Orange header):
 - DHCP
 - DNS
 - Email
 - VoIP
 - Web
- MAINTENANCE** (Blue header):
 - Backup
 - Feature Keys
 - Import / Export
 - Reboot Phones
 - Restart
 - Time
 - Tools
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The following screen is displayed. Click the **Modify** link for one of the Handsets.


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Analog Handsets

Handset	Owner	Caller ID	Port	Action
			07	New Analog Handset
			08	New Analog Handset

SIP Handsets [add new SIP handset](#) [Reboot Allworx Phones](#)

Handset	Line	Owner	Caller ID	Identification	Action
Allworx 9204G PBX Station (Default) View Configuration Add Call Appearance Reboot Replace MAC: 00-0A-DD-85-97-E9 192.168.2.4 :5060					
New User Two	1	New2	New User Two	User ID: *5100 Login ID: 5100 (expires: Nov 09, 2011 05:35 pm)	Modify Delete Ring
Allworx 9204 PBX Station (Default) View Configuration Add Call Appearance Reboot Replace MAC: 00-0A-DD-85-39-AB 192.168.2.2 :5060					
New User One	1	New1	New User One	User ID: *5101 Login ID: 5101 (expires: Nov 09, 2011 05:51 pm)	Modify Delete Ring
Allworx 9224 PBX Station (Default) View Configuration Add Call Appearance Reboot Replace MAC: 00-0A-DD-85-C8-E4 192.168.2.3 :5060					
New User Three	1	New3	New User Three	User ID: *5102 Login ID: 5102 (expires: Nov 09, 2011 05:35 pm)	Modify Delete Ring

Enter text in the **External Caller ID Name** field to match the text shown in the **Internal Caller ID Name** field. Repeat this procedure for each of the handsets.

Click the **Update** button.

allworx[Home](#) > [Phone System](#) > [Handsets](#) > [Modify Handset](#)

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Call Appearance

Call Forwarding: disabled

Phone Type: SIP Model: Allworx 9204
MAC Address: 00-0A-DD-85-39-AB
Owner: New User One (New1)
Internal Caller ID Name New User One (up to 47 characters: letters digits , , \ _ ' -)
Internal Caller ID Number use owner's extension
External Caller ID Name New User One (up to 47 characters: letters digits , , \ _ ' -)
External Caller ID Number (up to 24 digits)
Emergency Caller ID Number not specified
Description New User One
Dialing Privileges Group Dialing Privileges (Default)
Default Language Primary Language

SIP Registration

User ID: *5101
Binding: *5101@192.168.2.2:5060
Login ID 5101
Password (maximum 40 characters)
(expires: Nov 09, 2011 05:51 pm)

Call Appearance Features

☒ Provide Hold Music
☒ Can place calls
☒ Can receive calls

Update

Start Over

Cancel

After modifying the handsets, click the **Reboot Allworx Phones** button to reboot all the handsets, or click each of the individual **Reboot** links for each modified handset.

Home > Phone System > Handsets

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Analog Handsets

Handset	Owner	Caller ID	Port	Action
			07	New Analog Handset
			08	New Analog Handset

SIP Handsets [add new SIP handset](#) **Reboot Allworx Phones**

Handset	Line	Owner	Caller ID	Identification	Action
Allworx 9204G	PBX Station (Default)			View Configuration Add Call Appearance Reboot Replace	
MAC: 00-0A-DD-85-97-E9 192.168.2.4 :5060					
New User Two	1	New2	New User Two	User ID: *5100 Login ID: 5100 (expires: Nov 09, 2011 05:35 pm)	Modify Delete Ring
Allworx 9204	PBX Station (Default)			View Configuration Add Call Appearance Reboot Replace	
MAC: 00-0A-DD-85-39-AB 192.168.2.2 :5060					
New User One	1	New1	New User One	User ID: *5101 Login ID: 5101 (expires: Nov 09, 2011 05:51 pm)	Modify Delete Ring
Allworx 9224	PBX Station (Default)			View Configuration Add Call Appearance Reboot Replace	
MAC: 00-0A-DD-85-C8-E4 192.168.2.3 :5060					
New User Three	1	New3	New User Three	User ID: *5102 Login ID: 5102 (expires: Nov 09, 2011 05:35 pm)	Modify Delete Ring

7.7. Modify Dial Plan

Navigate back to the **Home** page shown below and click the **Dial Plan** link under **Phone System**.

The screenshot displays the Allworx Home page with a sidebar on the left and a main content area with six categorized panels. The 'PHONE SYSTEM' panel is highlighted in green, and the 'Dial Plan' link is enclosed in a red rectangle. The 'NETWORK' panel is orange, 'REPORTS' is blue, 'BUSINESS' is green, 'SERVERS' is orange, and 'MAINTENANCE' is blue.

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MAINTENANCE

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The following **Dial Plan** screen is displayed. Click the **modify** link next to **Internal Dial Plan**.

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[Internal Extension Length](#) [Internal Dial Plan](#) [External Dialing Rules](#) [Dialing Privileges Groups](#) [Service Groups](#)

Allworx phones must be rebooted after changes to the Internal Extension Length, Internal Dial Plan, or External Dialing Rules. [Reboot Phones](#)

Internal Extension Length [modify](#)

User and System Extensions are 4 digits in length.

Internal Dial Plan [modify](#) [view](#) the Phone Functions Reference Card

Plan	
9 + external number	External Call access (follows External Dialing Rules below)
0	Operator
1xxx 2xxx 3xxx 4xxx 5xxx 6xxx 7xxx 8xxx	User and System Extensions
*8 + enterprise number	Enterprise calling
*5nnn	Internal station access (reserved for system)
*250-*299 *24nnn	Speed dial numbers
*6 + user extension	Message Center
*700 call park *701-*709 call retrieve *7xxxx call pickup *79 + pin code	Call Functions (park/pickup/audit pin code)
*3 + user extension	Leave a voicemail for extension
*403 door relay *408 conference center *42n do not disturb *43n auto attendants *44nn call queues *4950-*4999 call retrieve *45xxxx call forwarding *46n paging	PBX Functions

External Dialing Rules

North American Numbering Plan Administration (NANPA) enabled [Modify](#)

Area Code	Exchange	Number Dialed	Service Group	Action
any		9+1+aaa-xxx-nnnn	All CO Lines & SIP Gateways	Modify

aaa - area code xxx - exchange nnnn - number

Type	Number Dialed	Service Group	Action
Emergency	9+911	see Dialing Privileges Group for source of call	Modify
Phone Services (211,311,411,511,611,711,811)	9+n11	All Trunk Devices	
Operator	9+0	All Trunk Devices	
Long Distance Services	9+1010...	All Trunk Devices	
International Calls	9+011...	All Trunk Devices	
Public SIP Directory	*8+nnnnn (5 digits)	All SIP Gateways	
PIN Code	*79+nnnnn (5 digits)	All CO Lines	

Dialing Privileges Groups

Name	Action
Dialing Privileges (Default)	View Copy

Service Groups [add new Service Group](#)

Group	Service(s)	Action
All CO Lines	(no services)	
All CO Lines & SIP Gateways	Session Manager (SIP Gateway)	
All Digital Lines	(no services)	
All SIP Gateways	Session Manager (SIP Gateway)	
All SIP Proxies	(no services)	
All Trunk Devices	Session Manager (SIP Gateway)	

Verify the **Use Extension Mode** checkbox is checked. If it isn't, check it and click the **Update** button.

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Internal Dial Plan [view the Phone Functions Reference Card](#)

TIP
It is recommended that you choose leading digits for your dial plan from the rows below using a "top-to-bottom" approach. Each time you change the digits in a row, the rows below it are automatically adjusted to only allow valid choices. As a result, some leading digits may be changed by the system in rows below the row you modified.

☒ **Use Extension Mode**

	Plan	
9	9 + external number	External Call access (follows External Dialing Rules below)
0	0	Operator
	1xxx 2xxx 3xxx 4xxx 5xxx 6xxx 7xxx 8xxx	User and System Extensions
	*8 + enterprise number	Enterprise calling
	*5nnn	Internal station access (reserved for system)
	*250-*299 *24nnn	Speed dial numbers
	*6 + user extension	Message Center
	*700 call park *701-*709 call retrieve *7xxxx call pickup *79 + pin code	Call Functions (park/pickup/audit pin code)
	*3 + user extension	Leave a voicemail for extension
	*403 door relay *408 conference center *42n do not disturb *43n auto attendants *44nn call queues *4950-*4999 call retrieve *45xxxx call forwarding *46n paging	PBX Functions

Update **Cancel**

NOTE
It is necessary to reboot all Allworx handsets for new dialplan settings to take effect.
Changing the extension mode setting will cause multi-site to be disabled.

Back on the **Dial Plan** screen, under **External Dialing Rules**, click the **modify** link to edit the **Public SIP Directory**.

For the **Public SIP Directory** row, enter the number of digits (e.g. **5**) in the **Number Dialed** column that will be used to dial enterprise Avaya phones. Select **All SIP Gateways** from the **Service Group** drop-down menu. Note, this row indicates that ***8** followed by a **5** digit extension will be used to route calls from the Allworx IP Phones to the SIP Gateway (Session Manager).

Click the **Update** button.

Home > Phone System > Dial Plan > Modify Dial Plan

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External Dialing Rules

Description	Number Dialed	Service Group
Emergency	9+911	see Dialing Privileges Group for source of call
Phone Services (211,311,411,511,611,711,811)	9+n11	All Trunk Devices
Operator	9+0	All Trunk Devices
Long Distance Services	9+1010...	All Trunk Devices
International Calls	9+011...	All Trunk Devices
Public SIP Directory	*8+5 digits	All SIP Gateways
PIN Code	*79+5 digits	All CO Lines
Outside Line Seizure	9#	All Trunk Devices

NOTE
Allworx phones must be rebooted when changes are made to the **Emergency**, **Public SIP Directory**, or **PIN Code** values.

Update **Cancel**

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, the Allworx 6x system and Allworx IP Phones.

8.1. Verify Avaya Aura® Communication Manager

On Communication Manager, verify the status of the SIP signaling group by using the “status signaling-group n” command, where “n” is the signaling group number administered in **Section 5.5**. Verify that the signaling group is *in-service* as indicated in the **Group State** field shown below.

```
status signaling-group 1
                                STATUS SIGNALING GROUP

      Group ID: 1
      Group Type: sip

      Group State: in-service
```

Verify the status of the local SIP trunk group by using the “status trunk n” command, where “n” is the trunk group number administered in **Section 5.6**. Verify that all trunks are in the *in-service/idle* state as shown below.

```
status trunk 1

                                TRUNK GROUP STATUS

Member   Port      Service State      Mtce Connected Ports
                                Busy

0001/001 T00001    in-service/idle    no
0001/002 T00002    in-service/idle    no
0001/003 T00003    in-service/idle    no
0001/004 T00004    in-service/idle    no
0001/005 T00005    in-service/idle    no
0001/006 T00006    in-service/idle    no
0001/007 T00007    in-service/idle    no
0001/008 T00008    in-service/idle    no
0001/009 T00009    in-service/idle    no
0001/010 T00010    in-service/idle    no
```

8.2. Verify Avaya Aura® Session Manager

Navigate to **Home** → **Elements** → **Session Manager** → **System Status** → **SIP Entity Monitoring** and select the Communication Manager SIP Entity (not shown). Verify the **Conn. Status** and **Link Status** are *Up*.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with categories: Session Manager, Network Configuration, Application Configuration, System Status, and System Tools. The 'System Status' category is expanded, showing 'SIP Entity Monitoring' as the selected option. The main content area displays the 'SIP Entity, Entity Link Connection Status' page. It includes a breadcrumb trail: Home / Elements / Session Manager / System Status / SIP Entity Monitoring - SIP Entity Monitoring. Below the title, a message states: 'This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.' A link 'All Entity Links to SIP Entity: CM_21_41' is provided, along with a 'Summary View' button. A table shows the connection status for one item, filtered by 'Enable'. The table has columns: Details, Session Manager Name, SIP Entity Resolved IP, Port, Proto., Conn. Status, Reason Code, and Link Status. The data row shows: SM_21_31, 10.64.21.41, 5061, TLS, Up, 200 OK, and Up.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	SM_21_31	10.64.21.41	5061	TLS	Up	200 OK	Up

Repeat the procedure above selecting the Allworx 6x SIP Entity, and verify the **Conn. Status** and **Link Status** are *Up*.

The screenshot shows the Avaya Aura System Manager 6.1 interface, similar to the previous one, but with 'Allworx 6x' selected in the breadcrumb trail. The 'All Entity Links to SIP Entity: Allworx 6x' link is present. The table shows the connection status for one item, filtered by 'Enable'. The data row shows: SM_21_31, 10.64.21.56, 5060, UDP, Up, 200 Ok, and Up.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	SM_21_31	10.64.21.56	5060	UDP	Up	200 Ok	Up

8.3. Verify Allworx 6x System and Allworx IP Phones

Make the following calls and verify the calls are set up properly, there is two-way audio with good audio quality, and the calls are torn down properly after completing the calls.

- Place a call from an Allworx IP Phone to an enterprise Avaya phone
- Place a call from an enterprise Avaya phone to an Allworx IP phone
- Place a call from an Allworx IP Phone to the PSTN
- Place a call from the PSTN to an Allworx IP Phone

9. Conclusion

These Application Notes describe the configuration steps required for the Allworx 6x system with Allworx IP phones to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager via a SIP trunk. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in **Section 2.2**.

10. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 6.0, Release 6.0, August 2010, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Document 03-603324, Issue 1.1, Release 6.1, October 2011, available at <http://support.avaya.com>

Allworx documentation is available at <http://www.allworx.com/support/resources.aspx>.

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