



Avaya Solution & Interoperability Test Lab

Application Notes for Lyrix Mobiso Speech Assistant Enterprise with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Lyrix Mobiso Speech Assistant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using Session Initiation Protocol (SIP) in an enterprise network. Lyrix Mobiso allows callers to speak the name of a person and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to transfer to the requested party. The integration between Mobiso and Session Manager was a SIP trunk.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps required to integrate the Lyrix Mobiso Speech Assistant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using Session Initiation Protocol (SIP) in an enterprise network. Lyrix Mobiso allows callers to speak the name of a person and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to transfer to the requested party. Lyrix Mobiso supports supervised and trombone (hairpin) transfers. A trombone transfer results in the media between two parties to flow through Mobiso. If the transfer call fails, Mobiso can redirect the call to an operator. The integration between Mobiso and Session Manager was a SIP trunk.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

The interoperability compliance test included feature and serviceability testing. Feature testing focused on Lyrix Mobiso successfully recognizing spoken names and extensions entered via DTMF and transfer the call to the proper destination. Supervised and trombone (hairpin) transfers were verified. In addition, routing failed transfer attempts to an operator were also verified.

Serviceability testing focused on verifying the ability of Lyrix Mobiso to recover from adverse conditions, such as server restart and loss of network connectivity.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Incoming calls to Mobiso, including multiple calls to Mobiso.
- Supervised and trombone (hairpin) call transfers.
- Call transfers based on spoken names (speech recognition).
- Call transfers based on dialed extension numbers (DTMF).
- Failed transfer attempts (e.g., busy station or invalid number) routed to operator.
- Recording of user name using the Name Collect feature.
- Enhanced disambiguation for people with similar names.
- Proper SIP call disconnection.
- G.711mu-law codec support and direct IP-IP media (i.e., Shuffling).
- Proper system recovery after a reboot of the Mobiso server and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations:

1. When a trombone (hairpin) transfer is performed, the caller does not hear ringback while the transfer-to station is ringing. For a supervised transfer, ringback is heard by the caller.
2. For supervised transfers to a busy station or invalid number, the call is transferred to the operator, but the caller does not receive an announcement indicating who will receive the call. For trombone transfers, Mobiso plays an announcement to caller indicating that the call will be transferred to the operator.

2.3. Support

For technical support on Mobiso, contact Lyrix Support via phone or website.

- **Phone:** 1-877-75-LYRIX (1-877-755-9749)
- **Web:** <http://www.mobiso.com/support-overview.htm>

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Communication Manager running on an Avaya S8800 Server with a G650 Media Gateway.
- Session Manager connected to Communication Manager via a SIP trunk and serving SIP telephones and the Lyrix Mobiso Speech Assistant. Session Manager was configured using Avaya Aura® System Manager.
- Avaya H.323 and SIP telephones.

Lyrix Mobiso interfaced to Session Manager via a SIP trunk. Mobiso was configured using a console and a web browser.

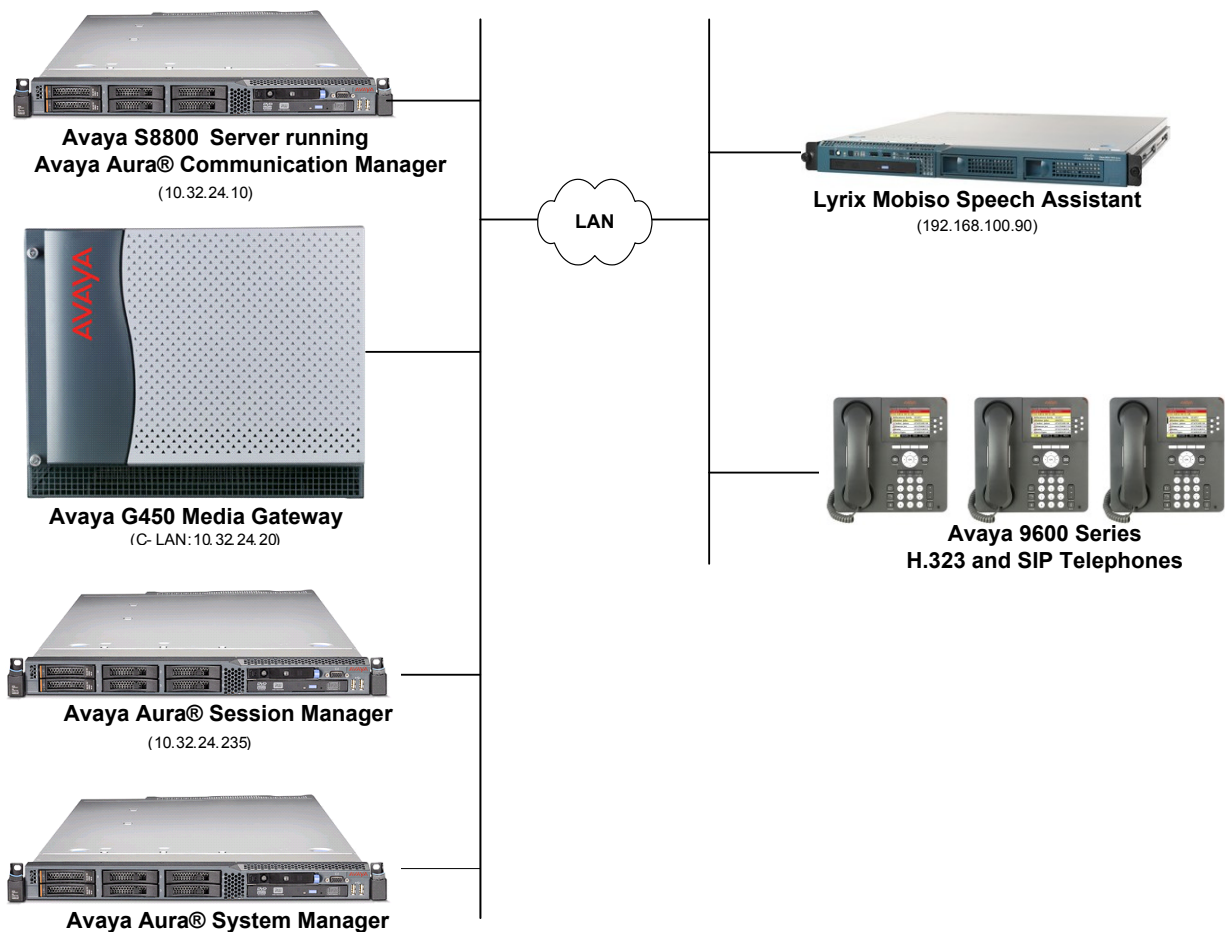


Figure 1: Avaya SIP Network with Lyrix Mobiso Speech Assistant

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8800 Server	6.0.1 (R016x.00.1.510.1) with Service Pack 5.01 (Patch 19303)
Avaya G650 Media Gateway <ul style="list-style-type: none">▪ TN799DP C-LAN Board▪ TN2602AP Media Processor Board	HW01 FW031 HW02 FW047
Avaya Aura® Session Manager	6.1 (6.1.5.0.615006)
Avaya Aura® System Manager	6.1.0 (6.1.0.07345-6.1.5.502) with Software Update Revision 6.1.9.1.1634
Avaya 9600 Series IP Telephones	3.1 SP 2 (H.323) 2.6.6 (SIP)
Lyrinx Mobiso Speech Assistant	6.5.1-3

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring a SIP trunk to Session Manager, which in turn will route calls to Mobiso. Administration of Communication Manager was performed using the System Access Terminal (SAT). The SAT is accessed by establishing a telnet session to Communication Manager using a terminal emulation application.

This section covers the following configuration:

- Verify OPS and SIP Trunk capacity.
- **IP Node Names** to associate names with IP addresses.
- **IP Network Region** to specify the domain name and the IP codec set, to enable IP-IP direct audio (i.e., Shuffling), and to specify the UDP port range.
- **IP Codec Set** to specify the codec type used for calls to Mobiso.
- **SIP trunks** for outgoing calls to Mobiso.
- **Private Numbering** to allow the caller's extension to be sent to Mobiso.
- **Call Routing** to route calls to Mobiso using AAR.

5.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks options are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of **Maximum Off-PBX Telephones - OPS** stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                                Page 1 of 11
                                OPTIONAL FEATURES

G3 Version: V16                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                         Module ID (MID): 1

                                USED
Platform Maximum Ports: 65000 170
Maximum Stations: 41000 87
Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 8
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 0 0
Maximum Survivable Processors: 313 0

(NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2** of the **system-parameters customer-options** form, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

display system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	12000	30
Maximum Concurrently Registered IP Stations:	18000	7
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	1
Maximum Video Capable IP Softphones:	18000	4
Maximum Administered SIP Trunks:	24000	30
Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0
Maximum Number of DS1 Boards with Echo Cancellation:	522	0
Maximum TN2501 VAL Boards:	128	1
Maximum Media Gateway VAL Sources:	250	0
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	1
Maximum Number of Expanded Meet-me Conference Ports:	300	0
(NOTE: You must logoff & login to effect the permission changes.)		

5.2. Configure IP Node Names

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the G650 Media Gateway (*clancrm*) and Session Manager (*devcon-asm*). The host names will be used in other configuration screens of Communication Manager.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
Gateway001	10.32.24.1	
ModMsg	192.50.10.45	
clancrm	10.32.24.20	
default	0.0.0.0	
devcon-asm	10.32.24.235	
medpro-1a13	10.32.24.26	
procr	10.32.24.10	
procr6	::	

(8 of 8 administered node-names were displayed)
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

5.3. Configure IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

change ip-network-region 1		Page 1 of 19
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? y	
UDP Port Max: 65535		
DIFFSERV/TOS PARAMETERS	RTCP Reporting Enabled? y	
Call Control PHB Value: 34	RTCP MONITOR SERVER PARAMETERS	
Audio PHB Value: 46	Use Default Server Parameters? y	
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 7		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5	AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 IP ENDPOINTS	RSVP Enabled? n	
H.323 Link Bounce Recovery? y		
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

5.4. Configure IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Mobiso. The form is accessed via the **change ip-codec-set 1** command. Testing was performed with G.711mu.

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:				
3:				
4:				
5:				
6:				
7:				

5.5. Configure SIP Trunk

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*.
- Specify the C-LAN board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Disable **Initial IP-IP Direct Media**.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 50		Page 1 of 1
SIGNALING GROUP		
Group Number: 50	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n	SIP Enabled LSP? n	
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: clancrm	Far-end Node Name: devcon-asm	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
	Far-end Secondary Node Name:	
Far-end Domain: avaya.com		
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? n	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. This trunk group is used for routing calls over SIP trunks to Session Manager. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

add trunk-group 50		Page 1 of 21	
TRUNK GROUP			
Group Number: 50	Group Type: sip	CDR Reports: y	
Group Name: To devcon-asm	COR: 1	TN: 1	TAC: 1050
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Member Assignment Method: auto	
		Signaling Group: 50	
		Number of Members: 10	

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 50		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private			
UUI Treatment: service-provider			
Replace Restricted Numbers? n			
Replace Unavailable Numbers? n			
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? Y			
DSN Term? n			

On **Page 4** of the trunk group form, set the **Identity for Calling Party Display** to *From*. Mobiso does not support the P-Asserted-Identity header in the SIP INVITE message. Therefore, when using Trombone transfers, this field must be set in order to display the proper caller ID on the transfer-to station.

add trunk-group 50	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? y Prepend '+' to Calling Number? n Send Transferring Party Information? n Network Call Redirection? n Send Diversion Header? n Support Request History? y Telephone Event Payload Type: 101	
Convert 180 to 183 for Early Media? n Always Use re-INVITE for Display Updates? n Identity for Calling Party Display: From Enable Q-SIP? n	

5.6. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with 7 whose calls are routed over any trunk group, including SIP trunk group 50, have the extension sent to the far-end for display purposes.

change private-numbering 0	Page 1 of 2										
NUMBERING - PRIVATE FORMAT											
<table border="1"> <thead> <tr> <th>Ext Len</th> <th>Ext Code</th> <th>Trk Grp(s)</th> <th>Private Prefix</th> <th>Total Len</th> </tr> </thead> <tbody> <tr> <td>5</td> <td>7</td> <td></td> <td></td> <td>5</td> </tr> </tbody> </table>	Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len	5	7			5	Total Administered: 1 Maximum Entries: 540
Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len							
5	7			5							

5.7. Configure Call Routing

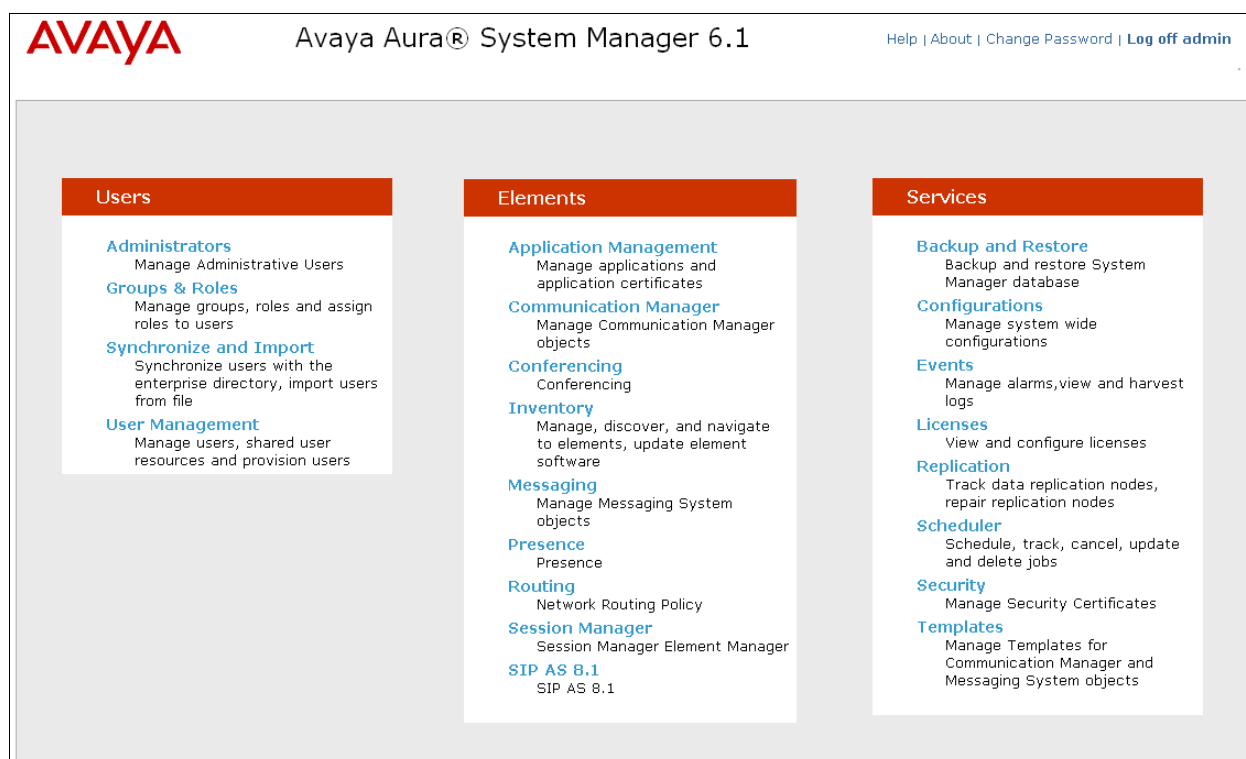
In this configuration, AAR was used to route calls to Mobiso. The extension assigned to Mobiso was 22005. This extension was included in the **Uniform Dial Plan Table** so that when it is dialed it is routed using AAR. The **AAR analysis** table then routed calls destined for Mobiso (extension 22005) over the SIP trunk configured in **Section 5.5** as specified on the **Route Pattern** form. For information in configuring AAR or ARS, refer to [1].

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager, Communication Manager, and Mobiso
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Session Manager server to be managed by System Manager

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL “https://<ip-address>/SMGR”, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials. The initial screen is displayed as shown below. The configuration in this section will be performed under **Routing and Session Manager** listed within the **Elements** box.



6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *avaya.com*)
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.

The screenshot shows 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", and "Log off admin". Below the navigation bar, there are tabs for "Routing" and "Home". The left sidebar contains a tree view with "Routing" expanded, showing sub-items: "Domains", "Locations", "Adaptations", "SIP Entities", "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns", "Regular Expressions", and "Defaults". The main content area is titled "Domain Management" and shows a breadcrumb trail: "Home / Elements / Routing / Domains - Domain Management". There are "Commit" and "Cancel" buttons at the top right of the main area. Below the breadcrumb, there is a table with one item: "avaya.com". The table has columns for "Name", "Type", "Default", and "Notes". The "Name" column contains "avaya.com" with a red asterisk indicating required input. The "Type" column contains "sip" with a dropdown arrow. The "Default" column contains an unchecked checkbox. The "Notes" column contains "Enterprise Domain". At the bottom of the table, there is a "Filter: Enable" link. Below the table, there is a red asterisk and the text "Input Required". At the bottom right of the main area, there are "Commit" and "Cancel" buttons.

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

6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *BR-DevConnect* location, which includes Communication Manager and Session Manager. Click **Commit** to save the Location definition.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Home /Elements / Routing / Locations- Location Details

Location Details

Commit Cancel Help ?

General

* Name: BR-DevConnect

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): Kbit/Sec

Minimum Multimedia Bandwidth: 64 Kbit/Sec

* Default Audio Bandwidth: 80 Kbit/sec

Location Pattern

Add Remove

2 Items | Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 192.168.100.*	
<input type="checkbox"/>	* 10.32.24.*	

Select : All, None

6.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, the C-LAN in the G650 Media Gateway, and Mobiso.

6.3.1. Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *avaya.com*).

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

SIP Entity Details

[Help ?](#)

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Location:

Outbound Proxy:

Time Zone:

Credential name:

SIP Link Monitoring

SIP Link Monitoring:

Entity Links

Entity Links can be modified after SIP Entity is committed.

Port

3 Items | [Refresh](#)

Filter: [Enable](#)

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

Select : [All](#), [None](#)

* Input Required

6.3.2. Communication Manager

A SIP Entity must be added for Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., C-LAN board) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults may 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", and "Log off admin". A breadcrumb trail shows "Home /Elements / Routing / SIP Entities- SIP Entity Details". The left sidebar contains a tree view with "Routing" expanded, showing sub-items like Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "SIP Entity Details" and has a "General" tab selected. It contains several form fields: "Name" (devcon13), "FQDN or IP Address" (10.32.24.20), "Type" (CM), "Notes" (empty), "Adaptation" (empty), "Location" (BR-DevConnect), "Time Zone" (America/Denver), "Override Port & Transport with DNS SRV" (unchecked), "SIP Timer B/F (in seconds)" (4), "Credential name" (empty), "Call Detail Recording" (none), and "SIP Link Monitoring" (Use Session Manager Configuration). There are "Commit" and "Cancel" buttons at the top right and bottom right. A red asterisk indicates required fields. A message at the bottom states "Entity Links can be modified after SIP Entity is committed."

6.3.3. Mobiso

A SIP Entity must be added for Mobiso. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., Mobiso) on the telephony system.
- **Type:** Select *Other*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

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'. A breadcrumb trail shows 'Home / Elements / Routing / SIP Entities- SIP Entity Details'. The left sidebar contains a tree view with 'Routing' expanded, showing sub-items like Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and has a 'General' tab selected. It contains several input fields: 'Name' (filled with 'Mobiso'), 'FQDN or IP Address' (filled with '192.168.100.90'), 'Type' (dropdown menu set to 'Other'), 'Notes' (empty text area), 'Adaptation' (dropdown menu), 'Location' (dropdown menu set to 'BR-DevConnect'), and 'Time Zone' (dropdown menu set to 'America/New_York'). There is an unchecked checkbox for 'Override Port & Transport with DNS SRV:'. Below these are 'SIP Timer B/F (in seconds):' (filled with '4'), 'Credential name:' (empty text field), and 'Call Detail Recording:' (dropdown menu set to 'none'). A section for 'SIP Link Monitoring' has a dropdown set to 'Use Session Manager Configuration'. At the bottom, there is a red warning message: 'Entity Links can be modified after SIP Entity is committed.' and a note '* Input Required'. 'Commit' and 'Cancel' buttons are present at the top right and bottom right of the form area.

6.4. Add Entity Links

In the sample configuration, two Entity links were added, one for Communication Manager and another one for Mobiso.

6.4.1. Communication Manager

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *devcon13 Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Set to *Trusted*. *Note: If this setting is incorrect, calls from the associated SIP Entity specified in Section 6.3.2 will be denied.*

Click **Commit** to save the Entity Link definition.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Entity Links - Entity Links

Entity Links

Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* devcon13 Link	* devcon-asm	TCP	* 5060	* devcon13	* 5060	Trusted	

* Input Required

Commit Cancel

6.4.2. Mobiso

The SIP trunk from Session Manager to Mobiso is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *Mobiso Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the Mobiso SIP entity.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Set to *Trusted*. *Note: If this setting is incorrect, calls from the associated SIP Entity specified in Section 6.3.3 will be denied.*

Click **Commit** to save the Entity Link definition.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Entity Links

Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* Mobiso Link	* devcon-asm	UDP	* 5060	* Mobiso	* 5060	Trusted	

* Input Required

Commit Cancel

6.5. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3**. Two routing policies were added – one for Communication Manager and one for Mobiso. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.

The screenshot displays the Avaya Aura System Manager 6.1 web interface. The left sidebar shows a navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a breadcrumb trail: 'Home / Elements / Routing / Routing Policies - Routing Policy Details'. The 'General' section contains fields for 'Name' (set to 'To devcon13'), 'Disabled' (unchecked), and 'Notes'. The 'SIP Entity as Destination' section has a 'Select' button. Below this is a table with columns 'Name', 'FQDN or IP Address', 'Type', and 'Notes', containing one entry: 'devcon13' with FQDN '10.32.24.20' and Type 'CM'. The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. It shows a table with one item, '24/7', with a ranking of 0 and checkboxes for all days of the week. The 'Start Time' is 00:00 and the 'End Time' is 23:59. The interface also includes a 'Commit' button and a 'Help ?' link.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

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

Routing Policy Details

Commit Cancel

Help ?

General

* Name: To devcon13

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
devcon13	10.32.24.20	CM	

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

The following screen shows the Routing Policy for Mobiso.

Avaya Aura® System Manager 6.1

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

Routing

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

Commit

Cancel

Help ?

General

* Name:

To Mobiso

Disabled:

☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Mobiso	192.168.100.90	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

6.6. Add Dial Patterns

Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “77” reside on Communication Manager, extension “22005” is assigned to Mobiso. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definitions for local extensions on Communication Manager.

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

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

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

[Help ?](#)

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

Dial Pattern Details

General

* **Pattern:** 77

* **Min:** 5

* **Max:** 5

Emergency Call: ☐

SIP Domain: avaya.com

Notes: Avaya CM

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"/>	BR-DevConnect	Juan's Subnet(s)	To devcon13	0	<input type="checkbox"/>	devcon13	

Select : All, None

The following screen shows the dial pattern definition for Mobiso.

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

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

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

Dial Pattern Details [Help ?](#)
[Commit](#) [Cancel](#)

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call: ☐

SIP Domain:

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"/>	BR-DevConnect	Juan's Subnet(s)	To Mobiso	0	<input type="checkbox"/>	Mobiso	

Select : [All](#), [None](#)

6.7. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Click **Add** (not shown) and fill in the fields as described below and shown in the following screen:

Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager.
- **Description:** Descriptive comment (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields. Click **Commit** to add this Session Manager.

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

[Session Manager](#) × [Home](#)

Home / Elements / Session Manager / Session Manager Administration - Session Manager Administration [Help ?](#)

Edit Session Manager

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

[General](#) | [Security Module](#) | [NIC Bonding](#) | [Monitoring](#) | [CDR](#) | [Personal Profile Manager \(PPM\)](#) - [Connection Settings](#) | [Event Server](#) | [Expand All](#) | [Collapse All](#)

General

SIP Entity Name

Description

*Management Access Point Host Name/IP

*Direct Routing to Endpoints

Security Module

SIP Entity IP Address

*Network Mask

*Default Gateway

*Call Control PHB

*QOS Priority

*Speed & Duplex

VLAN ID

7. Configure Lyrix Mobiso Speech Assistant

This section covers the procedure for configuring Mobiso. Mobiso is configured using the console and Mobiso web admin. The steps include:

- Configure the IP network parameters of the server via the Mobiso console.
- Launch the Mobiso web admin interface.
- Configure the DNIS for Mobiso.
- Configure the dialing rules and transfer type.
- Specify the operator number in the Mobiso application.
- Enable name collect, if desired.
- Add a person to the company directory.
- Configure other settings.

7.1. Configure IP Network Parameters


Log into the Mobiso console using the appropriate credentials. At the command prompt, run **pf_sys reip** as shown below. At the prompt asking, “What do you want to do? (S/M/Q)”, enter *M* to modify the IP settings. Continue by entering the appropriate information as shown below. When the next prompt is provided, type *S* to save the settings and then reboot the server.

```
root@mobiso:~  
root@mobiso.avaya.com#pf_sys reip  
  
Lyrix PeopleFind  
  
grep: /etc/resolv.conf: No such file or directory  
  
Network Configuration  
-----  
IP assignment method: static  
Host name: mobiso  
Domain name: avaya.com  
IP Address: 192.168.100.90  
Netmask: 255.255.255.0  
Gateway Address: 192.168.100.1  
Network Address: 192.168.100.0  
Broadcast Address: 192.168.100.255  
DNS address #1:  
DNS address #2:  
DNS address #3:  
  
Answer 'S' to save the settings shown above, stop PeopleFind services  
on this system, rewrite the OS configuration files and  
reboot the system.  
Answer 'M' to modify the above settings.  
Answer 'Q' to quit without saving changes.  
What do you want to do? (S/M/Q): M  
Host name of this system (do not include domain): mobiso  
Domain name: avaya.com  
System IP Address: 192.168.100.90  
Netmask: 255.255.255.0  
Gateway: 192.168.100.1  
DNS address #1:  
  
Network Configuration  
-----  
IP assignment method: static  
Host name: mobiso  
Domain name: avaya.com  
IP Address: 192.168.100.90  
Netmask: 255.255.255.0  
Gateway Address: 192.168.100.1  
Network Address: 192.168.100.0  
Broadcast Address: 192.168.100.255  
DNS address #1:  
DNS address #2:  
DNS address #3:  
  
Answer 'S' to save the settings shown above, stop PeopleFind services  
on this system, rewrite the OS configuration files and  
reboot the system.  
Answer 'M' to modify the above settings.  
Answer 'Q' to quit without saving changes.  
What do you want to do? (S/M/Q): S
```

```
root@mobiso:~  
root@mobiso.avaya.com#cat /etc/hosts  
# Do not remove the following line, or various programs  
# that require network functionality will fail.  
127.0.0.1    localhost.localdomain localhost DB-SERVER-SHARED-IP  
192.168.100.90  mobiso.avaya.com mobiso WEB-PROXY-SHARED-IP CALL-PROXY-SHARED-IP  
10.32.24.235   avaya.com  
root@mobiso.avaya.com#
```

7.2. Launch Mobiso Web Admin Interface

Configuration is accomplished by accessing the browser-based Mobiso admin interface using the URL “https://<ip-address>”, where <ip-address> is the IP address of the Mobiso server. Log in with the appropriate credentials as shown below.



mobiso
MAKING MOBILITY WORK

Login

Authorized access only. Access must be in accordance with the [Terms & Conditions](#)

Username

Password

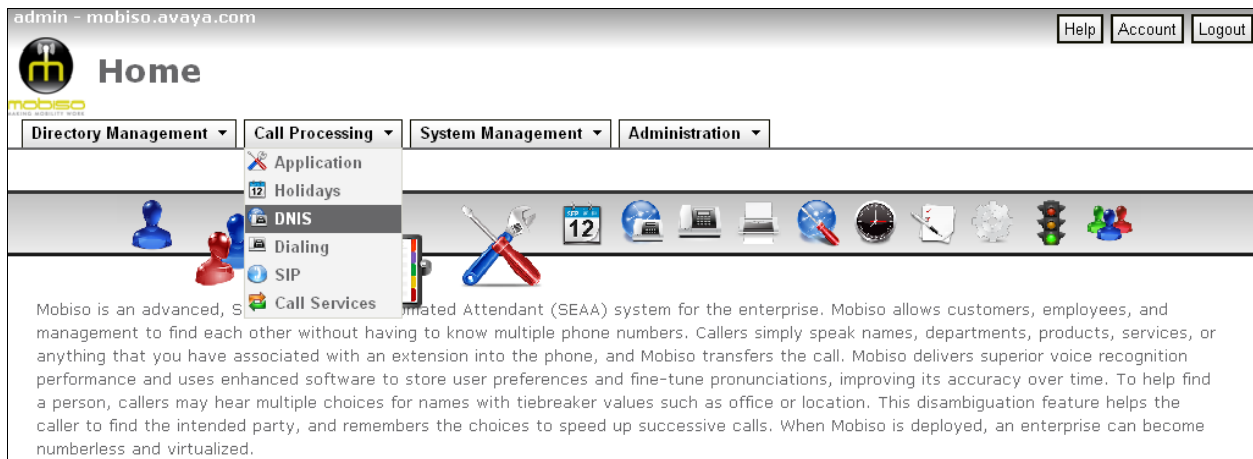
Forgot your password?

Best viewed in Firefox.

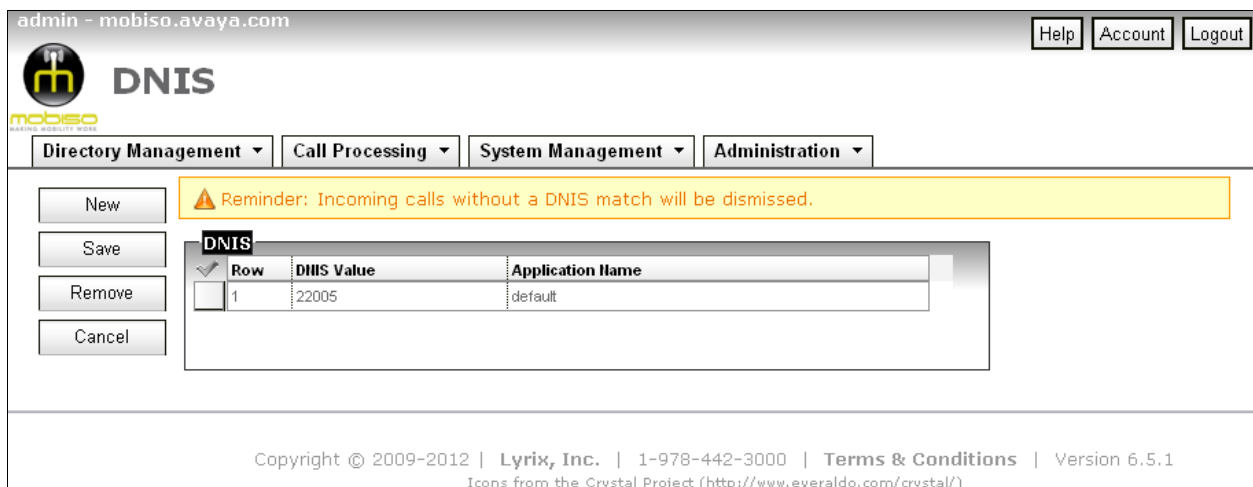
Copyright © 2009-2012 | Lyrix, Inc. | 1-978-442-3000

7.3. Configure the Mobiso DNIS

From the initial admin screen displayed below navigate to **Call Processing → DNIS**.



At the **DNIS** screen, click the **New** button and add the Mobiso DNIS in the **DNIS Value** field. In this example, Mobiso was assigned extension **22005**. The **Application Name** field should be left at *default*. Click **Save**.



7.4. Configure the Dialing Rules and Transfer Type

Navigate to **Call Processing → Dialing** to display the screen below. Set the **Maximum Number of Digits for Internal Calls** to 5 since a 5-digit dial plan was used. Accept the defaults for the **Outbound ANI Format**, **Internal Call Format**, and **Internal Transfer Format** fields. These settings will allow Mobiso to use information that arrives in the incoming SIP INVITE message. The **Allow Transfers for Internal Calls** should be set to *Always* for supervised call transfers and to **Never** for Trombone (hairpin) transfers where the call media would still flow through the Mobiso server. Click **Save**.

Note: If Trombone transfers are used, note that ringback tone will not be provided to the caller. For supervised transfers, ringback will be heard by the caller.

The screenshot displays the 'Dialing' configuration page in the Mobiso administration interface. The page is titled 'admin - mobiso.avaya.com' and includes navigation tabs for 'Directory Management', 'Call Processing', 'System Management', and 'Administration'. The 'Call Processing' tab is selected, and the 'Dialing' sub-tab is active. On the left, there are buttons for 'Save', 'Show Test', and 'Cancel'. The main content area is divided into three sections:

- System Dialing Rules:** This section contains fields for '* Country Code' (set to 1), 'International Dialing Prefix' (set to 011), '* Long Distance Prefix' (set to 1), and '* Longest Number' (set to 11). It also has a dropdown for 'ANI Matching Mode' set to 'Strict' and a text field for '* Outbound ANI Format' set to 'sip:%TO_USER@%FROM_HOST'.
- Phone Number Translation:** This section contains a list of 'Translation Rules' with five numbered dropdown menus, all of which are currently empty.
- Dialing Rules for Internal Calls:** This section contains fields for '* Maximum Number of Digits for Internal Calls' (set to 5), '* Internal Call Format' (set to 'sip:%DESTINATION@%FROM_HOST'), '* Internal Transfer Format' (set to 'sip:%DESTINATION@%FROM_HOST'), and 'Internal Outbound Proxy'. At the bottom, there are radio buttons for 'Allow Transfers for Internal Calls', with 'Always' selected.

7.5. Configure Operator Number

Navigate to **Call Processing** → **Application** to display the screen below. Specify the **Operator Number** and the **Fail Phone Number** by the appropriate application. In this example, the same extension (77304) was used by the “Mobiso” application that is run during the day session. The **Fail Phone Number** is called if the transfer fails for whatever reason. Click **Save**.

admin - mobiso.avaya.com Help Account Logout

Application

Directory Management ▾ Call Processing ▾ System Management ▾ Administration ▾

Previous
Next
New
Copy
Save
Remove
Cancel

* Application Name: default Assigned to DNIS: 22005

Prompts Recognition Disambiguation Other

Timezone: America/New_York

	Settings	Welcome Prompt	Operator Number Fail Phone Number	Route Failed Calls to Operator	Play Office Number
Day		[3] Mobiso	77304 77304		<input type="checkbox"/>
Night		[1] Thank you for		<input checked="" type="checkbox"/>	<input type="checkbox"/>
Weekend	Use <input type="checkbox"/>	[1] Thank you for		<input checked="" type="checkbox"/>	<input type="checkbox"/>
Holiday	Use <input type="checkbox"/>	[1] Thank you for		<input checked="" type="checkbox"/>	<input type="checkbox"/>

Main Prompt: [1] Who would yc
Retry Prompt: [1] At any time, ε
Hold Prompt: [1] Connecting nc
Help Prompt: [1] The system c

7.6. Enable Name Collect (Optional)

Navigate to **Call Processing** → **Application** and then select the **Other** tab. Enable name collect as shown below if people should record their own name to replace their name using TTS. Click **Save**.

The screenshot shows the 'Application' configuration page for 'mobiso.avaya.com'. The page has a top navigation bar with 'Help', 'Account', and 'Logout' buttons. Below this is a breadcrumb trail: 'Directory Management' > 'Call Processing' > 'System Management' > 'Administration'. The main content area is divided into a left sidebar with buttons for 'Previous', 'Next', 'New', 'Copy', 'Save', 'Remove', and 'Cancel'. The main panel has tabs for 'Prompts', 'Recognition', 'Disambiguation', and 'Other'. The 'Other' tab is selected. In this tab, there is a 'Corporate Location Dialing' section with a dropdown set to 'Disabled'. Below this is the 'Enable Name Collect' checkbox, which is checked. A text block explains that enabling this feature prompts users to record their names, which will replace TTS. It also mentions a 'Voicemail Number' field and a 'Test Id' field. The 'Test Id' field is currently empty.

admin - mobiso.avaya.com

Help Account Logout

Application

Directory Management Call Processing System Management Administration

Previous Next New Copy Save Remove Cancel

* Application Name default Assigned to DNIS 22005

Prompts Recognition Disambiguation **Other**

Corporate Location Dialing
Disabled

☒ **Enable Name Collect**

If you enable Name Collect for the application, people without a spoken name will be prompted to record their names the next time they use Mobiso. The spoken name recording will then replace the TTS. In order for people to rerecord previous recordings that they made, Name Collect must be active for the application, and you must click the Record Name field in a person's record page. The person will then be prompted to record their name the next time they call the system. Alternatively, you can record a spoken name on your PC or have the spoken names recorded by a third party and then upload it using the Browse button in the Greeting box. The uploaded audio can be deleted by clicking the red 'x' button in the Greeting box.

Voicemail Number

Test Id

The test id is a number you can call into to test the system.

7.7. Add Person to Company Directory

Navigate to **Directory Management** → **People** to display the screen below. Click **New** to add a new person. Specify an **Id**, such as *1JohnSmith*. Specify the **First Name**, **Last Name**, and **Office Extension**. Since the **Office Extension** field was used, **Section 7.8** must be completed too.

The screenshot shows the 'People' management page in the Mobiso Avaya admin console. The page has a header with the Mobiso logo and 'People' title. Below the header are navigation tabs: Directory Management, Call Processing, System Management, and Administration. The main content area is divided into several sections:

- Person**: Fields for * Id (1JohnSmith), Date Created (2012-03-16 11:46:24 EDT), * First Name (John), Middle Name, * Last Name (Smith), Nickname, Voice Portal PIN, and Email. There are also dropdowns for Tiebreaker 1 and Tiebreaker 2.
- Phone Numbers**: Fields for * Find Me (Office), Office Extension (77308), Office Phone, Other Office/Skype, Mobile Phone, Home Phone, and Other Phone.
- Settings**: Checkboxes for List in External Directory and Disambiguation, a checkbox for Record Name, and a dropdown for * Language (None).
- Activity**: Fields for Last Inbound Call, Last Authenticated Call, and Last Outbound Contact, each with an EDT (Edit) button.
- Greeting**: A section for uploading an audio file, with a red 'X' icon and a 'Browse...' button.

On the left side of the 'Person' section, there are buttons for Previous, Next, New, Save, Remove, Search, Clear, File Import, AD Import, and Cancel.

Navigate to **Directory Management** → **Company Directories** and click **Rebuild** to update the company directory.

The screenshot shows the 'Company Directories' management page in the Mobiso Avaya admin console. The page has a header with the Mobiso logo and 'Company Directories' title. Below the header are navigation tabs: Directory Management, Call Processing, System Management, and Administration. The main content area includes:

- Buttons for New, Save, Remove, Rebuild, and Cancel.
- A checkbox for Email execution output to administrator, which is checked.
- A table with the following data:

Row	Directory Name	Include All People	Include Departments	Application(s)
1	Company	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	default

7.8. Configure Other Settings

Navigate to **System Management** → **Settings** and enable **Use Office Extension**. Click **Save**.

admin - mobiso.avaya.com Help Account Logout

Settings

Directory Management Call Processing System Management Administration

Save Cancel

Call Parameters

- ☒ Allow Find-Me
- ☐ Allow International Calls
- ☐ Allow Non-Directory Dialing
- ☒ Allow Messaging
- ☐ Allow Messaging After Find-Me
- ☐ Allow Messaging To All
- ☒ Use Office Extension
- ☐ Auto Login for Known Callers

Disambiguation History

	Maximum # of Entries	Number of Days Retained
ED Person	25	0
ED Extension	10	0
ED External	10	0

Miscellaneous

Reporting email

Test email Address

Default Voice Portal PIN

Postmaster Address

SMTP Server

* Timezone
America/New_York ▼

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Mobiso with Communication Manager and Session Manager. The following steps can be used to verify installations in the field.

1. Verify that the Mobiso link status is up. From System Manager, navigate to **System Status → SIP Entity Monitoring** and then click on the hyperlink associated with Mobiso. The following screen should be displayed indicating that the **Link Status** is *Up*.

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager, Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, and System Status. The main content area is titled 'SIP Entity, Entity Link Connection Status' and displays 'All Entity Links to SIP Entity: Mobiso'. A table shows the connection status for a single item, 'devcon-asm', with a Link Status of 'Up'.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	devcon-asm	192.168.100.90	5060	UDP	Up	200 OK	Up

2. From the Mobiso console or SSH, run the command **runstat** to verify that the Mobiso channels are in the *IDLE* state as shown below. When there is an incoming call, the channel status will change to *CALL*.

The screenshot shows a terminal window with the command 'runstat' executed. The output displays a table of Mobiso channel status information, including channel number, active status, application, state, and time.

```
root@mobiso:~  
mobiso.avaya.com Fri Mar 16 11:50:38 EDT 2012  
Active Calls: 1 Mobiso Channel Status Monitor (c)ycle (e)xit  
CH ACTIVE APP STAT SP TNT DATE | CH ACTIVE APP STAT SP TNT DATE |  
0 default CALL 1 1 11:50:31 |  
1 IDLE 0 0 11:16:41 |  
2 IDLE 0 0 11:16:41 |  
3 IDLE 0 0 11:16:41 |  
4 IDLE 0 0 11:16:41 |  
5 IDLE 0 0 11:16:41 |  
6 IDLE 0 0 11:16:41 |  
7 IDLE 0 0 11:16:41 |
```

3. Place a call to Mobiso and verify that the greeting is heard.
4. Speak the name of the person to which Mobiso should transfer the call.
5. Verify that the call transfer can be completed successfully.

9. Conclusion

These Application Notes have described the configuration steps required to integrate Lyrix Mobiso Speech Assistant with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All test cases passed with observations noted in **Section 2.2**.

10. References

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

- [1] *Administering Avaya Aura® Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.

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