Avaya Solution & Interoperability Test Lab

Application Notes for Configuration of Avaya Aura® Session Manager Version 6.1 and Lyrix Mobiso Version 6.5 as a Hosted Solution - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate Lyrix Mobiso as a Hosted Solution version 6.5 with Avaya Aura® SIP system Release 6.1 as a SIP endpoint on the Avaya Aura® Session Manager over the internet.

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 Lyrix Mobiso Speech Enabled Automated Attendant as a Hosted Solution (hereafter referred to as Mobiso) with Avaya Aura® System Release 6.1 as a SIP endpoint on the Avaya Session Manager over the internet. Mobiso Hosted Solution is an advanced, Speech-Enabled Automated Attendant (SEAA) over the internet. It allows customers, employees, and management to connect with each other without having to know multiple phone numbers. Callers simply speak names, departments, products, services, or anything associated with a phone number, and Mobiso connects the call.

2. General Test Approach and Test Results

The compliance testing verified interoperability between Mobiso, Communication Manager and Session Manager. Testing was performed manually using H.323 and SIP phones calling into the Mobiso test system that had been configured to operate in the Avaya SIP network by registering to it over the internet as a SIP endpoint. The test coverage included the following:

- Answer incoming calls
- Perform speech recognition
- Call transfer using SIP REFER Call trombone (hairpin) using SIP INVITE instead of REFER transfer
- Recognize DTMF digits
- Use caller identification to record name response (Mobiso Name Collect) and verify Enhanced Disambiguation functionality in Mobiso
- Handle SIP disconnect events at various points in the call session
- Basic network failure and re-establishment (unplugged cable, rebooted server)

2.1 Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Feature/functionality: demonstrates the ability to respond properly to dialer events.
  - Basic Call
  - Blind Transfer Call based on Speech Recognition
  - Blind Transfer Call based on DTMF (RFC2833) Recognition
  - Supervised (Trombone) Transfer Call based on Speech Recognition
  - Supervised (Trombone) Transfer Call based on DTMF Recognition
  - Mobiso Name Collect and Enhanced Disambiguation
  - Invalid Blind Transfer Call
  - Invalid Supervised (Trombone) Transfer Call
  - Hang up Functionality
  - Multiple Calls
- Serviceability: demonstrates the ability to operate properly to external events.

2.2 Test Results

All test cases were passed with the following observations,

1. Transferring a call to an invalid, unregistered or unplugged phone from a user who is registered on the SM which has profile on CM:
When the first transfer failed, the originator will get transferred to an operator. But the originator will not hear the prompt notifying them that they will get transferred to operator.

2. Transferring a call to an invalid, unregistered or unplugged phone from a user who is registered on the SM only:
When the first transfer failed, the originator will hear fast busy tone. But the originator will not get transferred to operator.

2.3 Support

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

- **Phone:** 1-978-442-3400
- **Email:** support@lyrix.com
- **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:

- Avaya Aura® Communication Manager running on an Avaya S8800 Server with a G650 Media Gateway.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones and video endpoints.
- Avaya Aura® System Manager used to configure Session Manager.

In addition, there were SIP, H.323 phones and Avaya Flare used for voice calls. All SIP devices registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

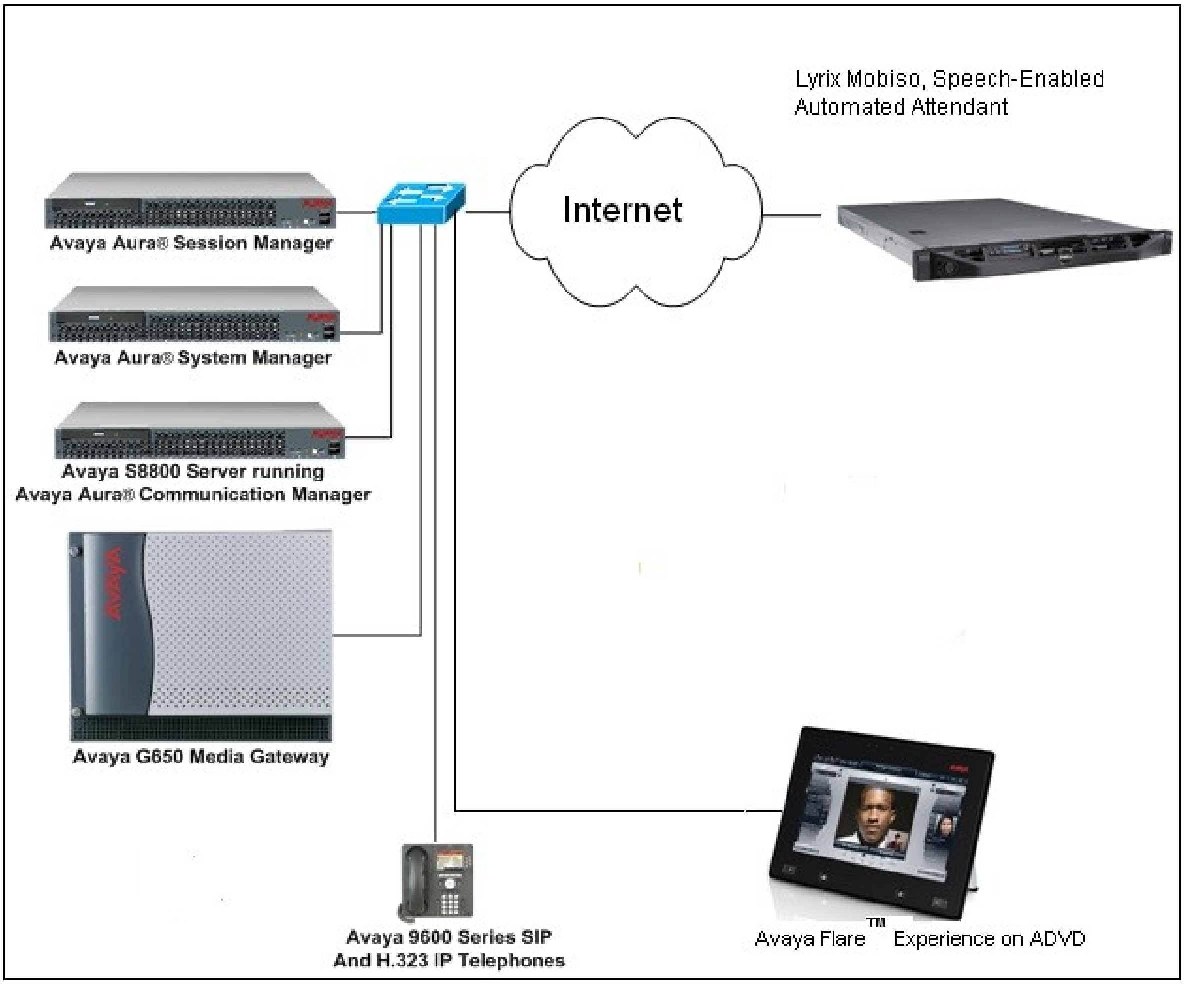

Figure 1: Avaya Network with the Lyrix Mobiso as Hosted Solution integrated as SIP endpoint connecting to Avaya Session Manager over the internet.

4. Equipment and Software Validated

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

Equipment	Software Version
Avaya S8800 server	Avaya Aura® Communication Manager R016x.00.1.510.1
Avaya S8800 server	Avaya Aura® Session Manager 6.1 (6.1.5.0.615006)
Avaya G650 Media Gateway IPSI TN2312BP CLAN TN799DP IP Media Processor TN2302AP Digital Line TN2224	 HW06, FW043 HW01, FW026 HW20, FW095 000006
Avaya One-X® Communicator	6.1.1.02-SP1-32858
Avaya Flare ™ Experience on ADVD (SIP)	1.0.3
Avaya 9611G (H323) IP Phone	6.0.1
Avaya 9650C (SIP) IP Phone	2.6.4
Avaya 1608 (SIP) IP Phone	3.1
Lyrix Mobiso (Linux Virtual Machine on Dell R710) (SIP)	6.5.1-3

5. Configure Avaya Aura®

These Application Notes assumes that Avaya Aura® System namely Communication Manager (CM) and Session Manager are configured and operational. For detailed information on how to configure and administer the Avaya Aura® System, please refer to the **Section 9 [1]**.

The following section will describe how to configure the Mobiso as a SIP endpoint to the Session Manger.

5.1 Configure the Avaya Aura® Session Manager

Log in to the **System Manger** with appropriate credentials (not shown), the **System Manger** home page is seen as shown in **Figure 2** below:

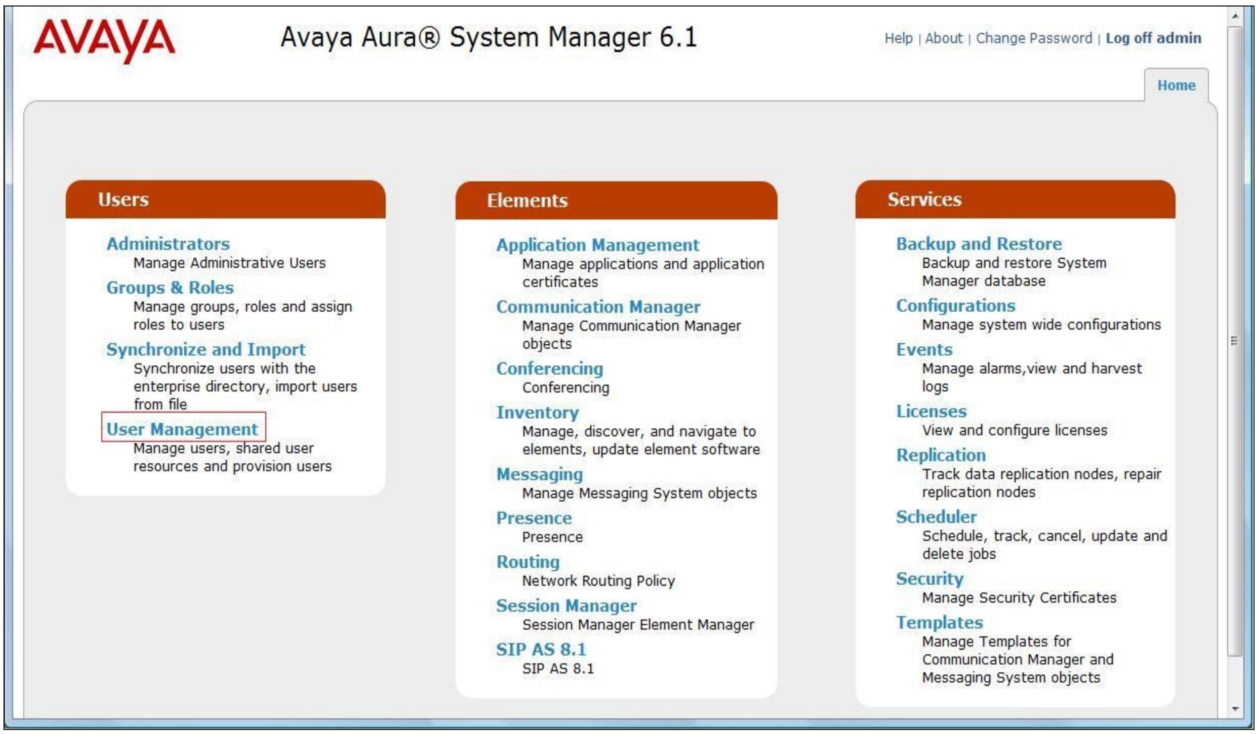

Figure 2: System Manger Home Page

Navigate to **User Management** > **Manage Users**, click on the **New** (not shown) button. At the **Identity** tab, fill in the new user information as shown in the red boxes in **Figure 3.**

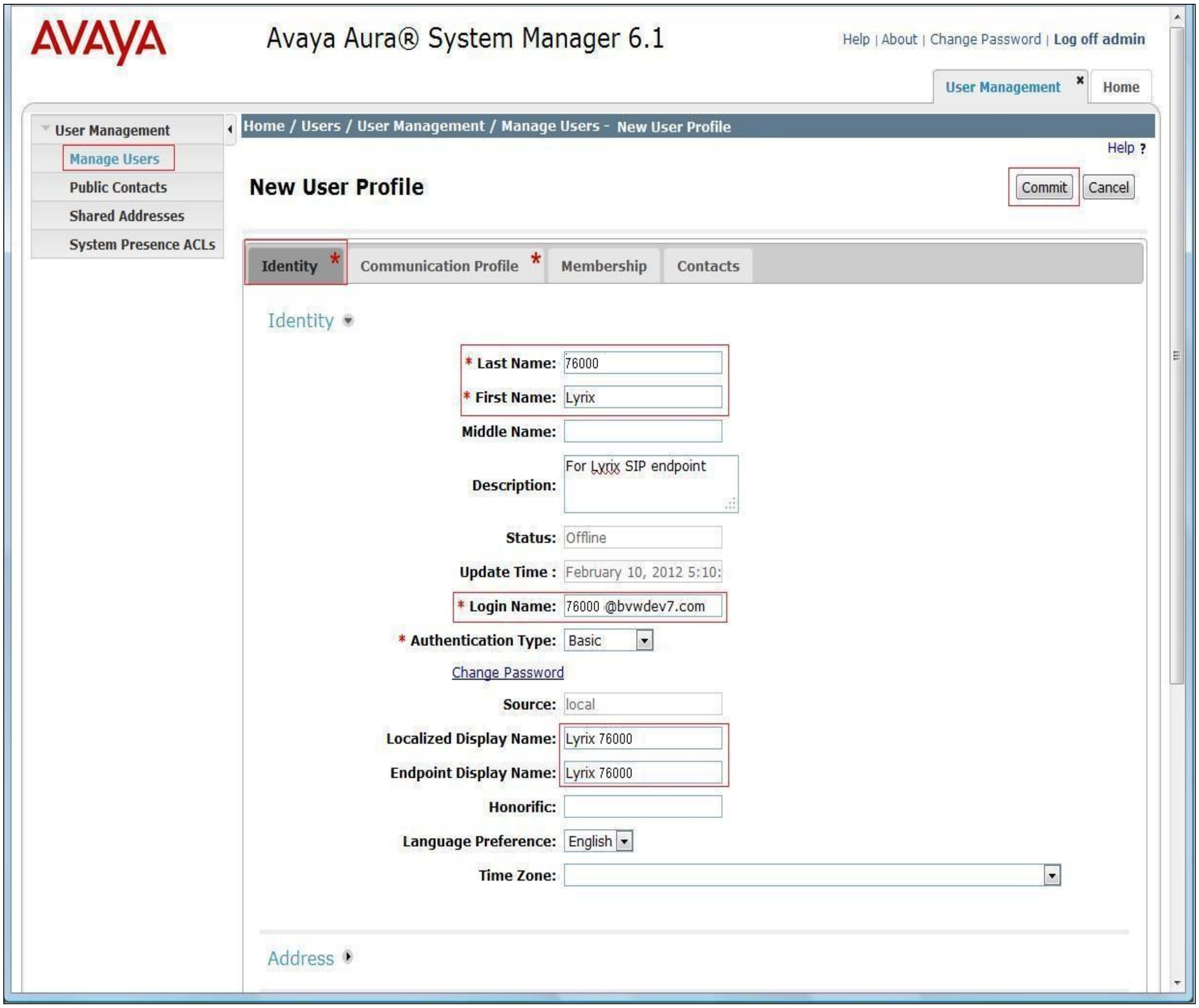

Figure 3: New User Profile

Click on the **Communication Profile** tab to enter the information as shown in **Figure 4.** At the **Communication Address,** click on **New** to create a type of user with associate SIP domain then click **Add** to save the user. Click **Commit (not shown)** button to save user profile.

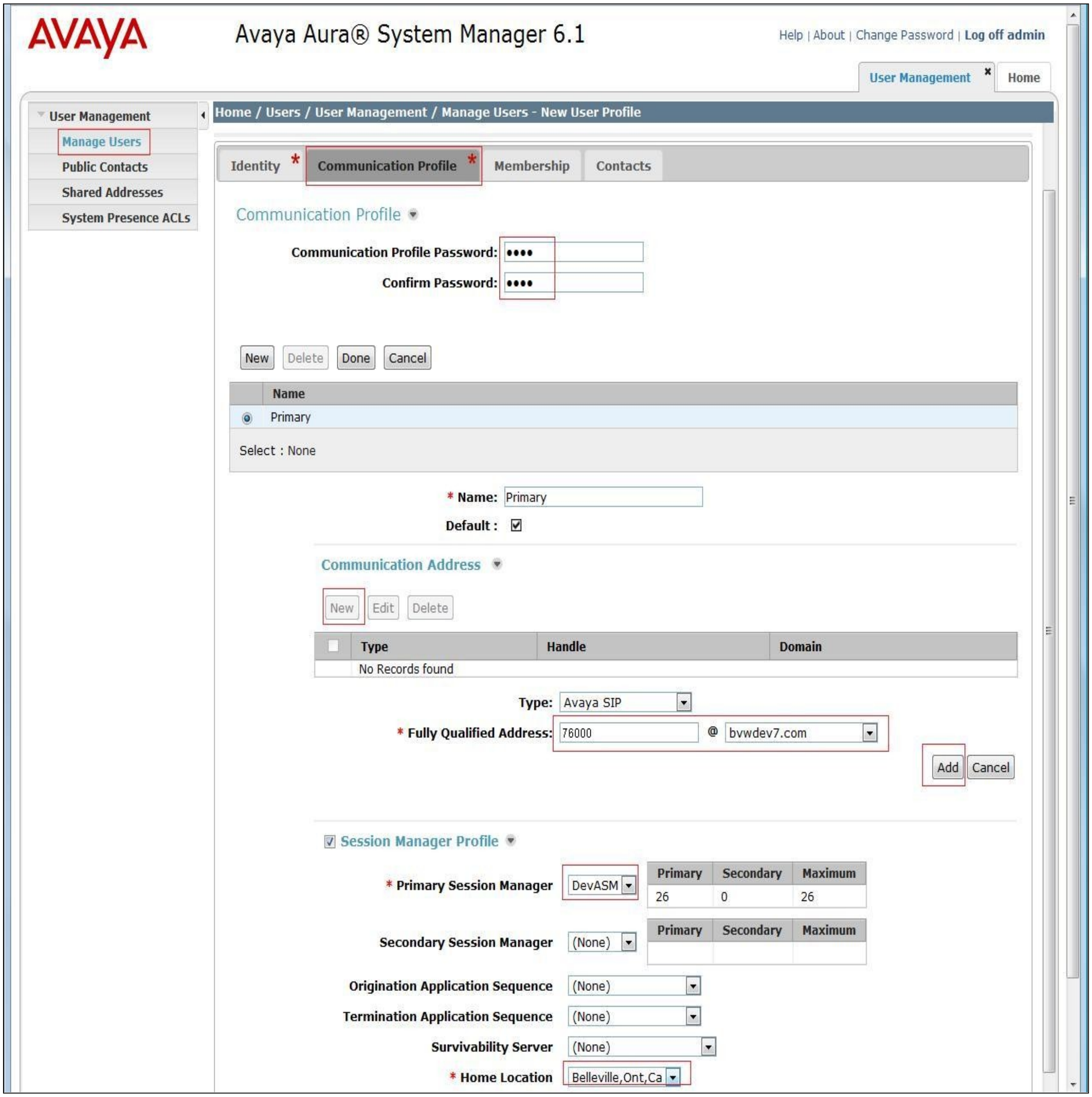

Figure 4: Communication Profile

5.2 Configure Lyrix Mobiso

This section shows the sample steps to configure the Mobiso to inter-operate with the Avaya Aura® SIP network system.

Assumption: The customer has signed up for Mobiso hosted service and has been given access to the Mobiso system through a Service Provider level account. Depending on the contract agreement, the account may only be Tenant level which does not have access to some of the screens shown below. For those cases, Lyrix is responsible for configuration of the Tenant.

Step 1: Open a web browser and go to the Mobiso administration URL that was provided when signing up for the service.

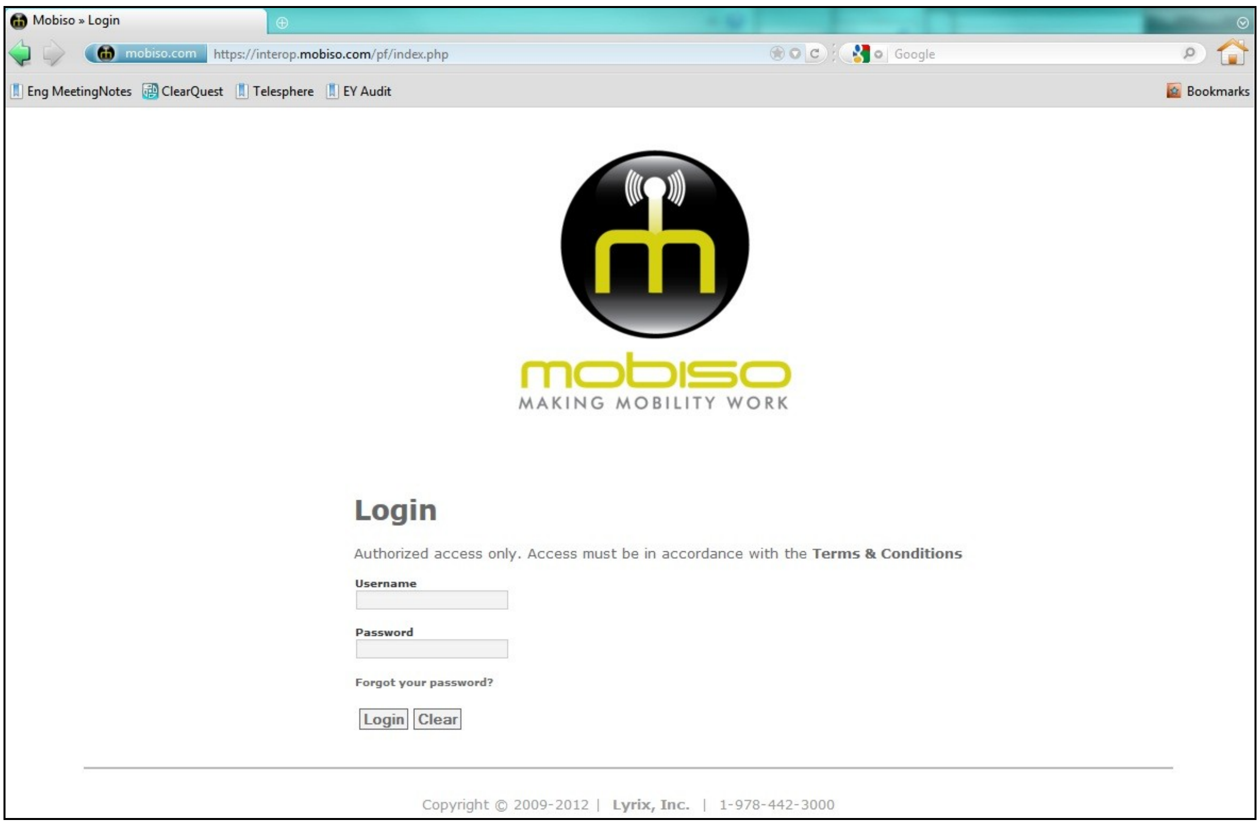

Figure 5: Mobiso Login Screen

Step 2: Enter the **username** and **password** that were provided when signing up for the service and login to the Mobiso web administration interface.

Step 3: From the main page, go to the **SIP Registration** page by opening the **Call Processing** tab and selecting **SIP**.

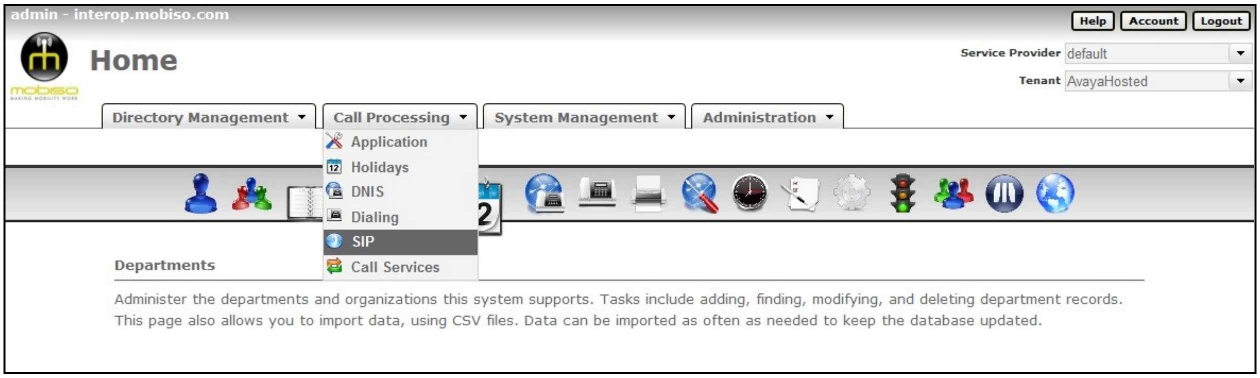

Figure 6: Navigating to SIP page

Step 4: Enter the required **SIP registration** information into the fields on the page. Note that verification of field values and testing must be performed per ***Section 6 Verification Steps*** before the information can be saved to the Mobiso database and activated on the system.

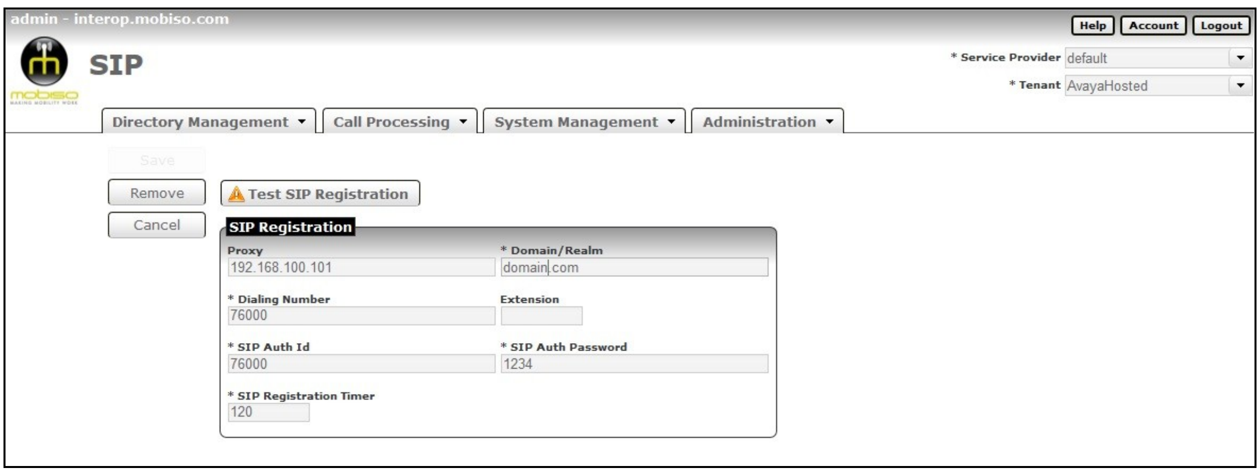

Figure 7: SIP Registration page

Step 5: Navigate to the Mobiso **DNIS** page by opening the **Call Processing** tab and selecting **DNIS**.

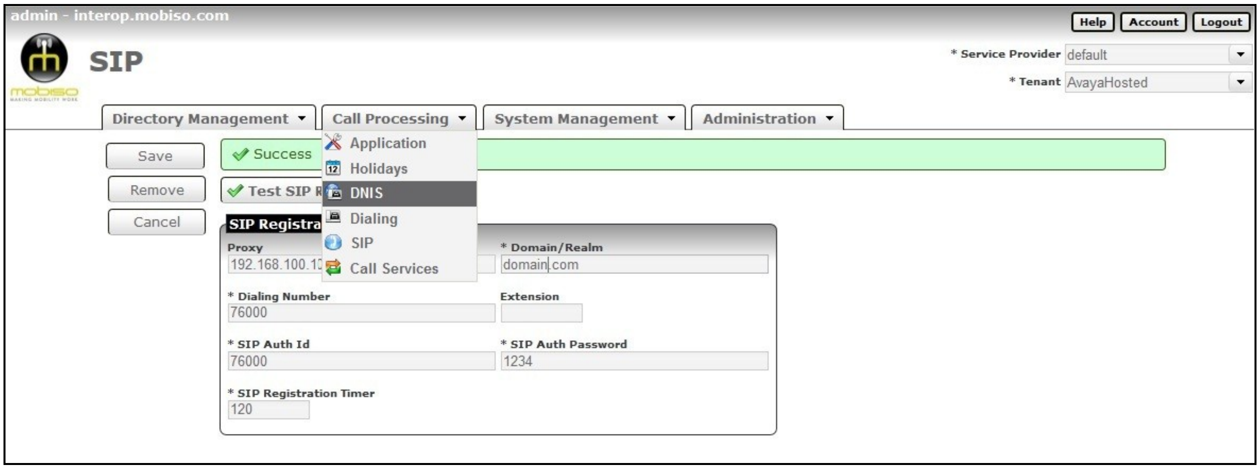

Figure 8: Navigate to DNIS page

Step 6: On the **DNIS** page, click the **New** button and enter the **DNIS number** in the grid, then click the **Save** button. **Figure 9** shows the DNIS number 76000 being configured. This matches the SIP Registration number, and is required in order for Mobiso to answer calls.

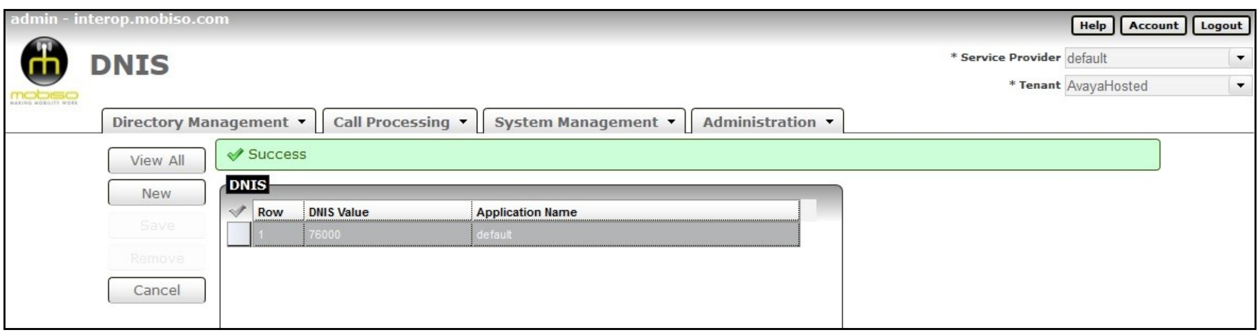

Figure 9: DNIS page

Step 7: Navigate to the **Dialing** page by opening the **Call Processing** tab and selecting **Dialing**.

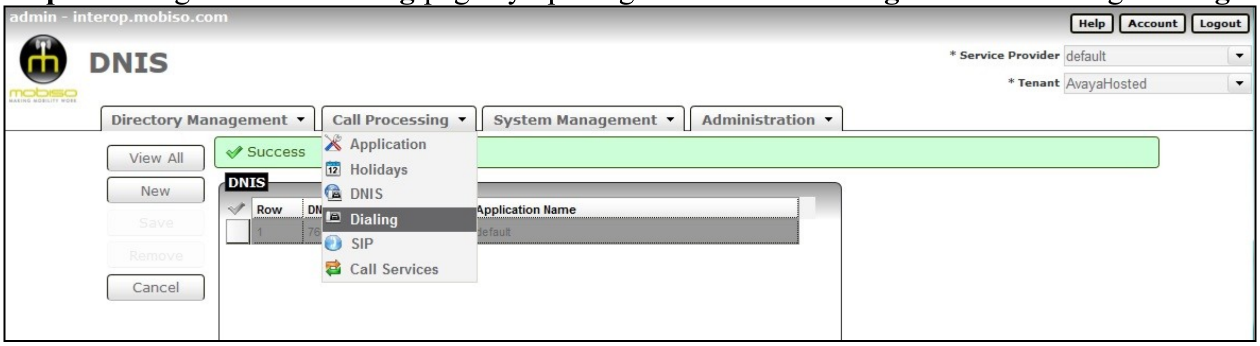

Figure 10: Navigate to Dialing page

Step 8: On the **Dialing** page, configure any required settings as needed. Normally the default settings are sufficient. **Figures 11** and **12** shows the Dialing setup used during testing.

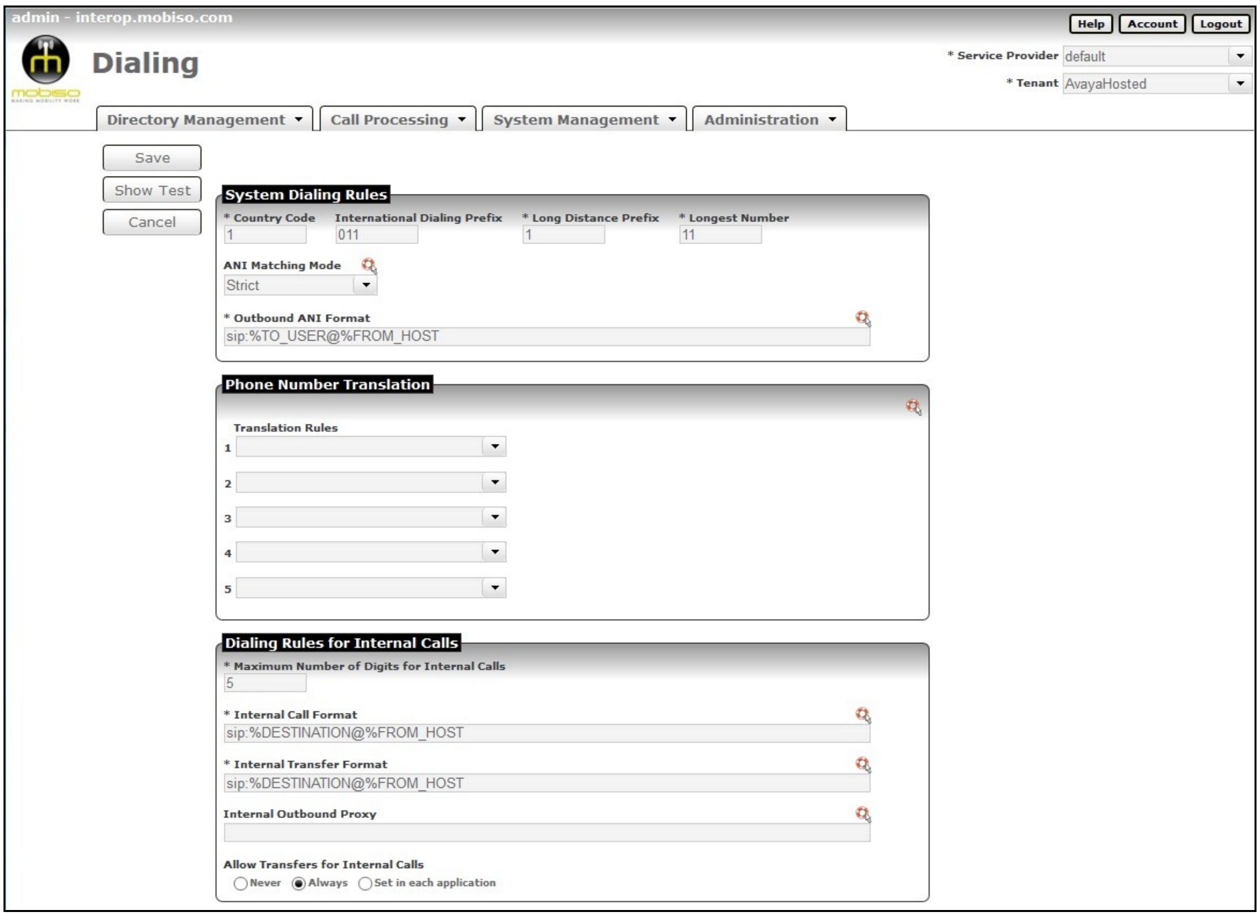

Figure 11: Dialing page

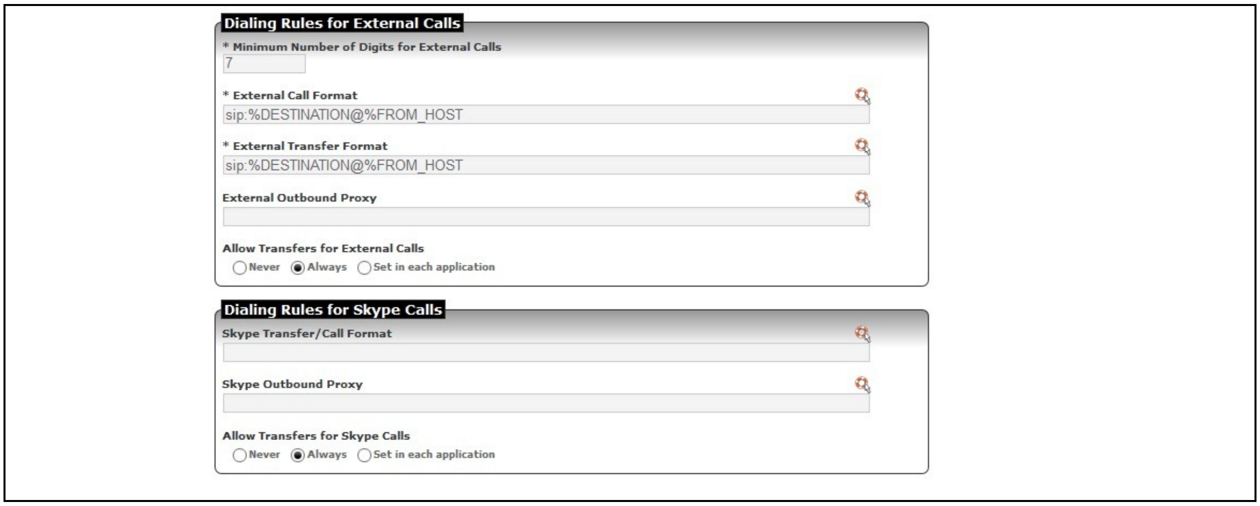

Figure 12: Dialing page (cont'd)

Step 9: Navigate to the **People** page by opening the **Directory Management** tab and selecting **People**.

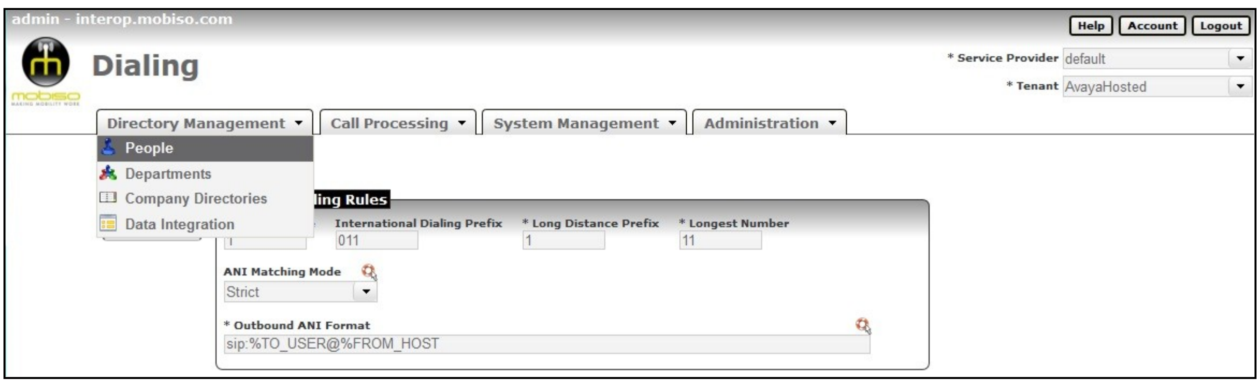

Figure 13: Navigate to People page

Step 10: Add people to the Mobiso directory.

Figure 14 shows the **People** page where employee directory information may be entered manually by clicking on the **New** button on the left. The minimum set of the information required for a single record is the unique "Id", **First Name**, and **Last Name**. For the purpose of this testing, a phone number entered into the Office Extension field is also required. A Department page is also available. Alternative automated methods of loading the employee and department directories are available, including CSV file loading and Active Directory. Refer to the Mobiso Administration Guide for more information about how to import employee and departments into Mobiso. In this screenshot, the person Mike Jones at office phone number 76001 has been added to the system.

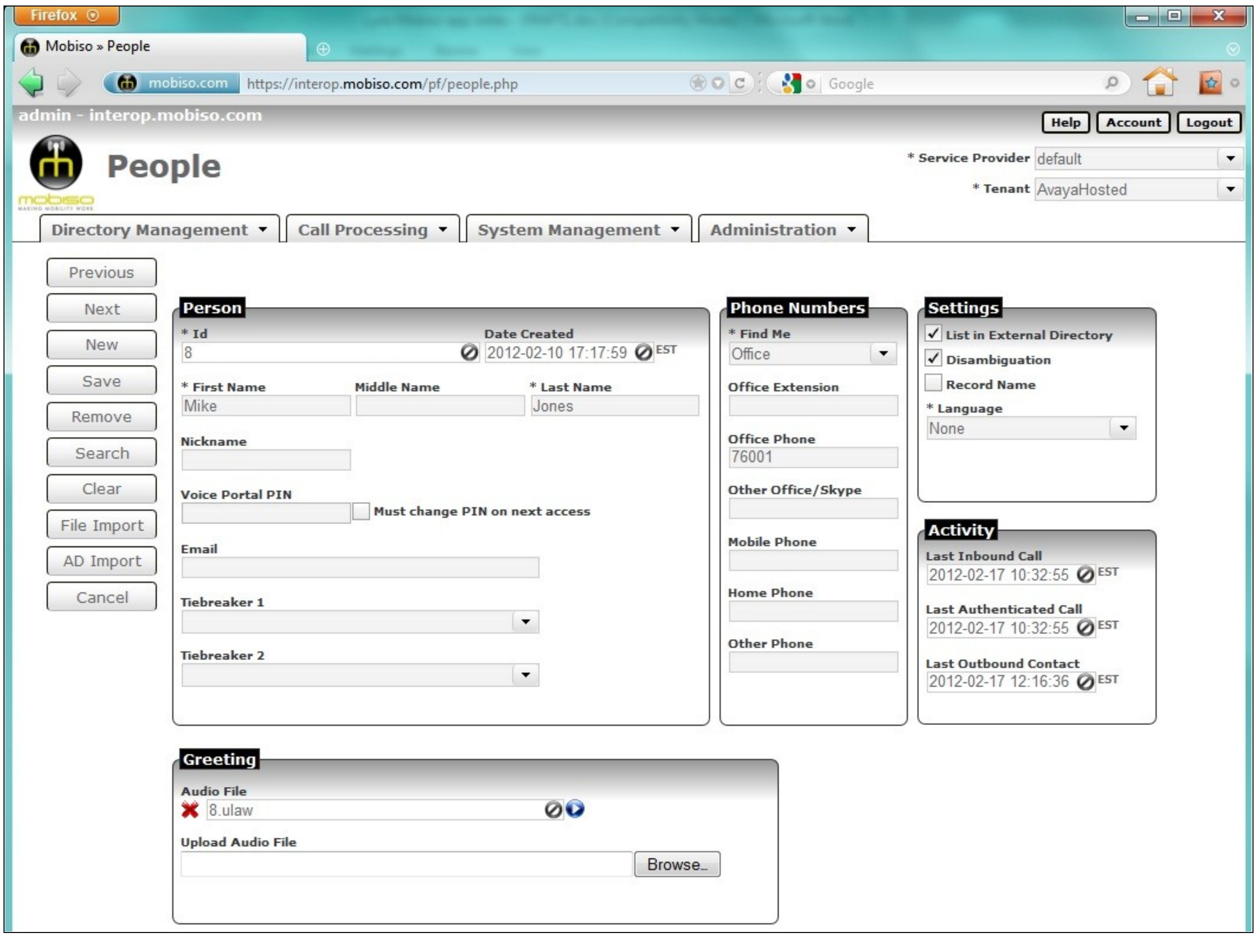

Figure 14: People page

Step 11: After all People and Departments have been added to the system, the Company Directories must be rebuilt. Navigate to the **Company Directory** page by opening the **Directory Management** tab and choosing **Company Directories**

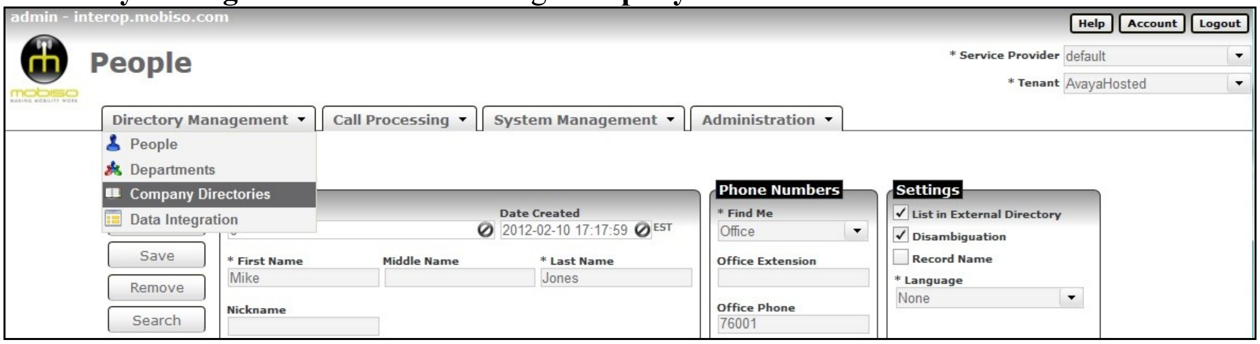

Figure 15: Navigate to Company Directories page

Step 12: Click on the **Rebuild** button to rebuild the directories for the Tenant. This must be done after making any name changes to People or Departments, and also after adding or removing People or Departments.

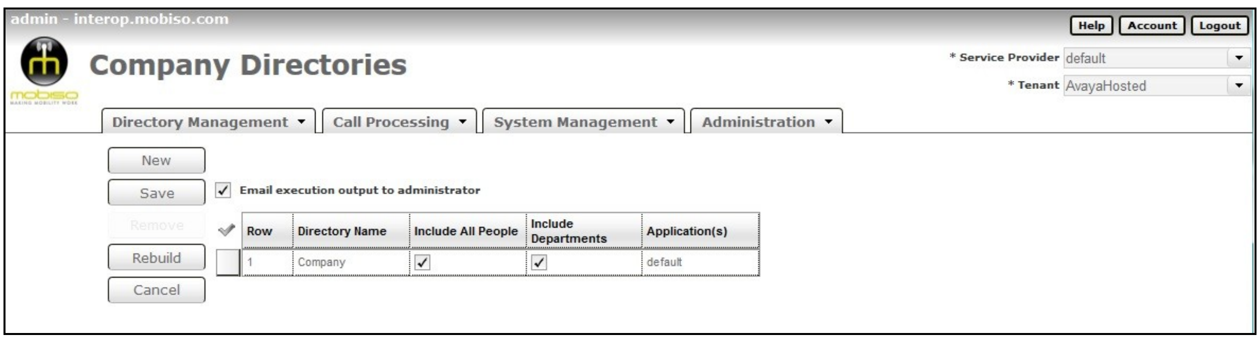

Figure 16: Company Directories page

6. Verification Steps

This section provides the steps that may be performed to verify proper configuration of Mobiso and Avaya Aura® SIP system network.

1. Verify that when SIP registration set up is completed, the Mobiso system is able to register to Session Manager as a SIP endpoint by clicking on the Test SIP Registration button. A successful registration is shown in **Figure 17**.

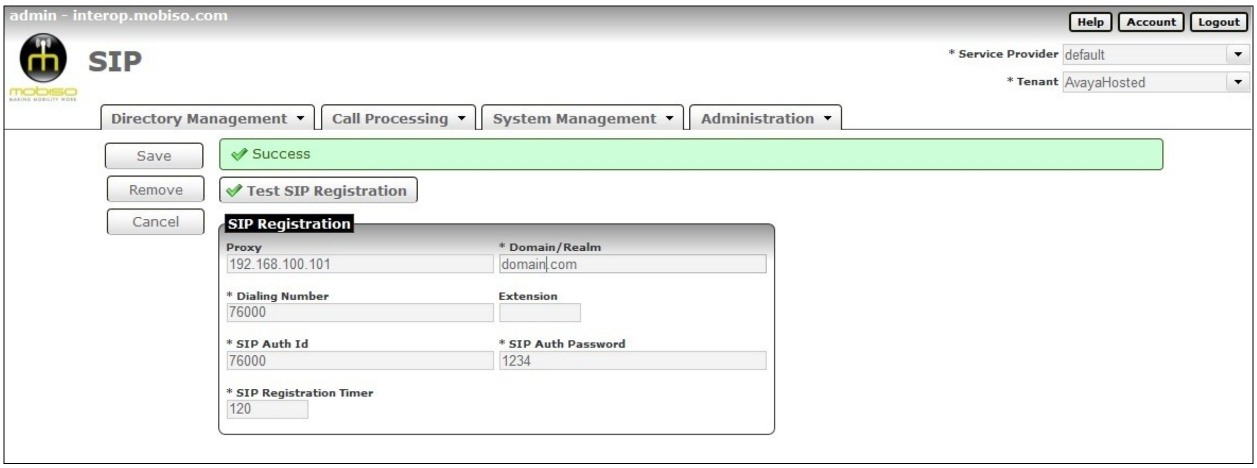

Figure 17: Successful Registration

2. Place a call to Mobiso at number 76000. Caller will hear a prompt asking for name of the person to be reached. The caller will say one of the names added to the People or Department pages and will be transferred to the phone number assigned to that person or department with two ways audio path established with the caller.

7. Conclusion

These Application Notes have described the administration steps required to integrate the Mobiso as a SIP user endpoint to the Avaya Aura® SIP network system. Call is transferred by Mobiso and established with 2 way speech paths. All test cases passed with observations noted in **Section 2.2**.

8. References

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

[1] *Administering Avaya Aura® Communication Manager*, August 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.

[3] Lyrix Mobiso documentation can be found at http://www.mobiso.com/administration.htm

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.