Avaya Solution & Interoperability Test Lab

Application Notes for Cetis M-Series M203IP SIP cordless Telephones with Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate Cetis M-Series M203IP SIP cordless Telephones with Avaya Aura® Session Manager. The Cetis M-Series M203IP SIP cordless Telephones were designed for the hospitality industry and register with Avaya Aura® Session Manager.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 Cetis M-Series M203IP SIP cordless Telephones (hereon refers to as Cetis M203IP SIP Telephones) with Avaya Aura® Session Manager. The Cetis M203IP SIP Telephones were designed for the hospitality industry. In the compliance test, Cetis SIP Telephones registered with Avaya Aura® Session Manager, established calls with other Avaya SIP and H.323 telephones, and executed telephony and hospitality features.

2. General Test Approach and Test Results

This section details the general approach to the testing, what was covered, and results of the testing. If the testing was successfully concluded but it was necessary to implement workarounds or certain non-critical features did not work, it should be noted in **Section 2.2**.

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. The feature testing focused on establishing calls between Cetis M203IP SIP Telephones and Avaya SIP and H.323 telephone and exercising basic telephony features, such as hold, mute, hold, transfer and conference. In addition, hospitality features, such as call forward and Do Not Disturb were covered.

The serviceability testing focused on verifying that the Cetis M203IP SIP Telephones come back into service after re-connecting the Ethernet connect or rebooting the phone.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Cetis M203IP SIP Telephones with Session Manager.
- Calls between Cetis telephones and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetis telephones and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including inbound/outbound, hold, mute.
- Use of programmable buttons on the Cetis telephones.
- Proper system recovery after a restart of the Cetis telephones and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- Cetis M203IP SIP Telephones phone did not support blind transfer
- Cetis M203IP SIP Telephones phone did not support blind conference
- Cetis M203IP SIP Telephones phone did not support forwarding on busy

2.3. Support

For technical support on the Cetis M203IP SIP Telephone, contact Cetis support via phone, email, or website.

- **Phone:** (719) 638-8821
- **Email:** customerservice@cetisgroup.com **or** sipsupport@cetisgroup.com
- **Web:** http://www.cetisgroup.com/support/

3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Cetis M203IP SIP Telephones with Session Manager. The Cetis telephones registered with Session Manager via SIP.

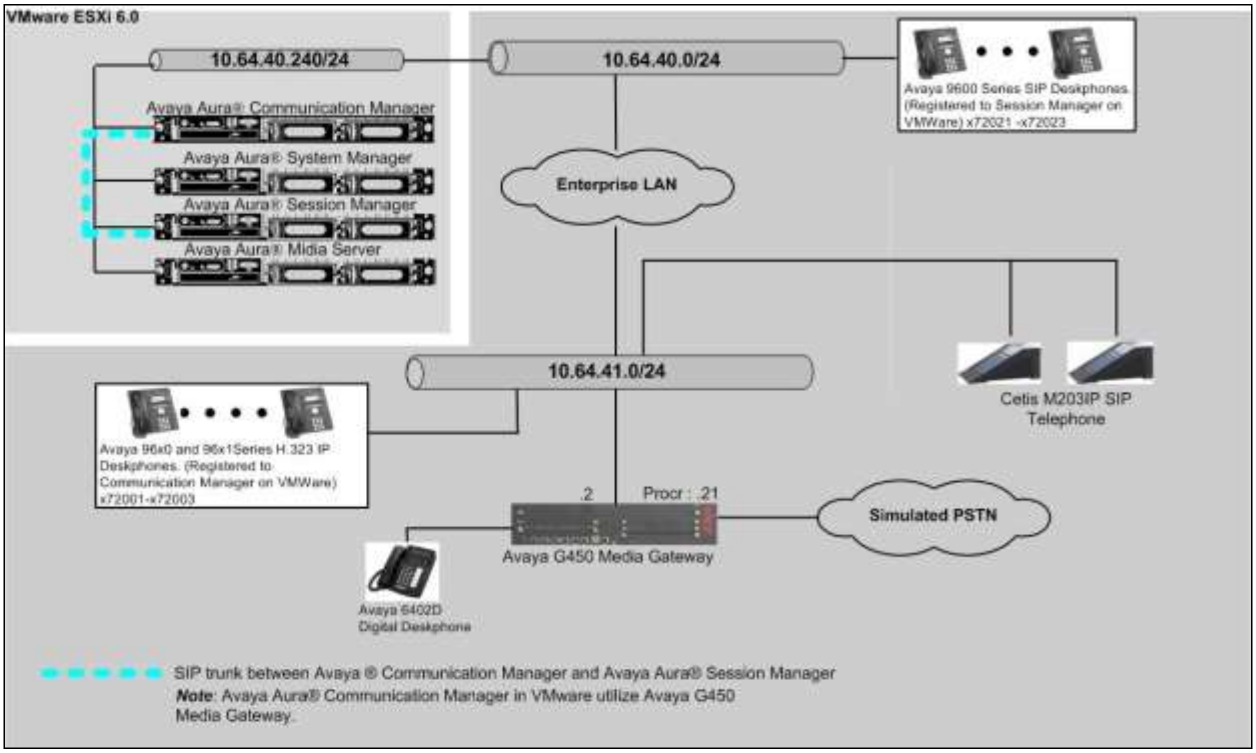


Figure 1: Cetis M203IP Telephones with Avaya Aura® Session Manager

4. Equipment and Software Validated

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

Equipment		Software/Firmware
Avaya Aura® System Manager on VMware		7.0.0.0.3929
Avaya Aura® Session Manager on VMware		7.0.0.0.700007
Avaya Communication Manager on VMware Avaya G450 Media Gateway Avaya Aura® Media Server VMware		7.0 (R017x.00.0.441.0) 37.19.0 7.7.0.226 ESXi 6.0
Avaya 96x0 and 96x1 Series IP Deskphones		
	9620 (H.323) 9621G (H.323)	3.25 6.6
Avaya 96x0 and 96x1 Series SIP Deskphones		
	9611G 9650	7.0.0.39 2.6.14
Cetis M203IP		2.1.0-105

5. Configure Avaya Aura® Session Manager

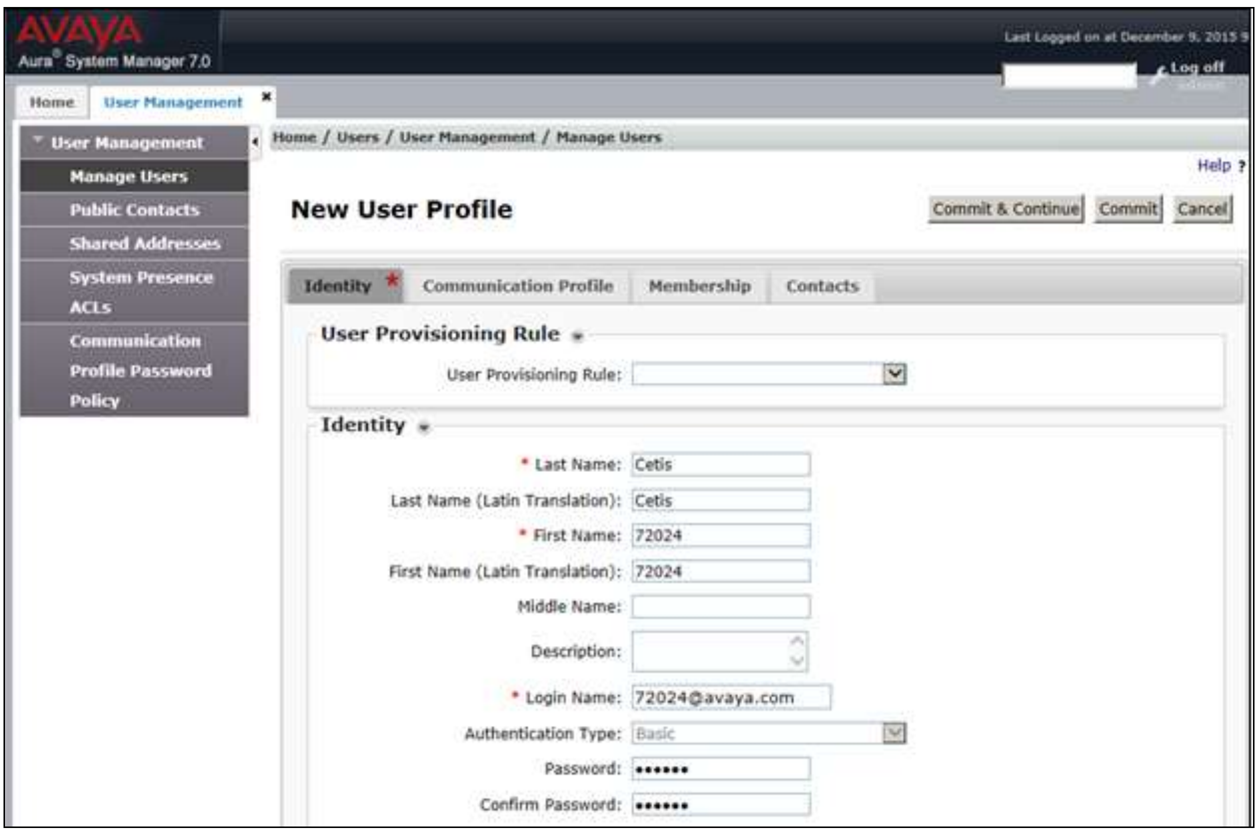
This section provides the procedures for configuring Session. Session Manager is comprised of two functional components: The Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

This section assumes that Session Manager and System Manager have been installed, and network connectivity exists between the two platforms.

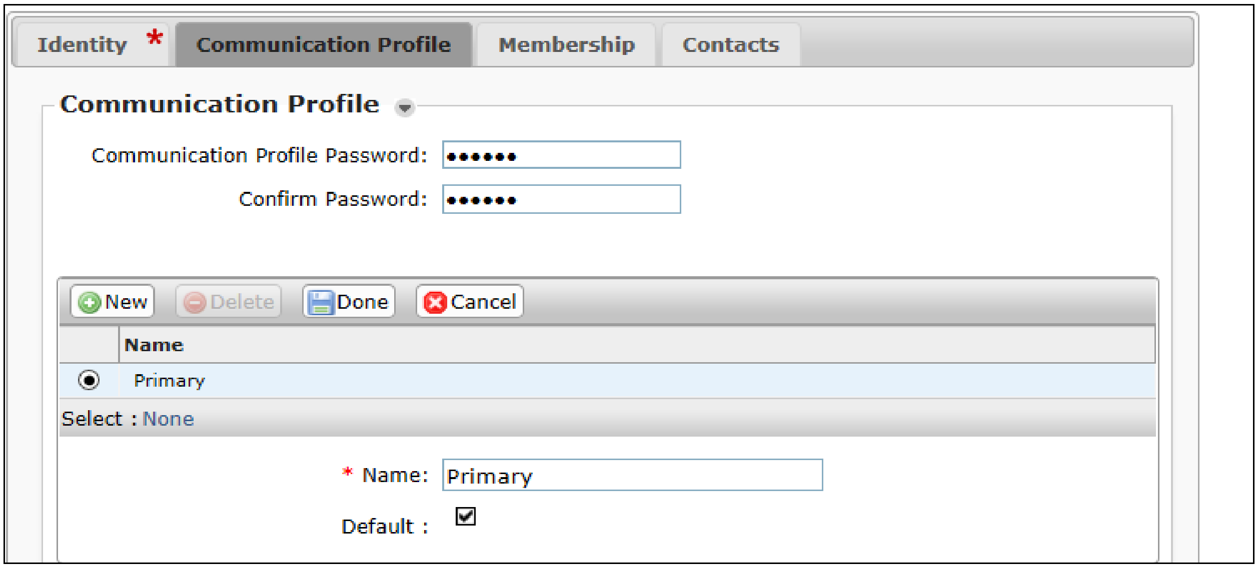
5.1. Configure User

To add new SIP users, Navigate to **Home → Users → User Management → Manage Users**. Click **New** and provide the following information:

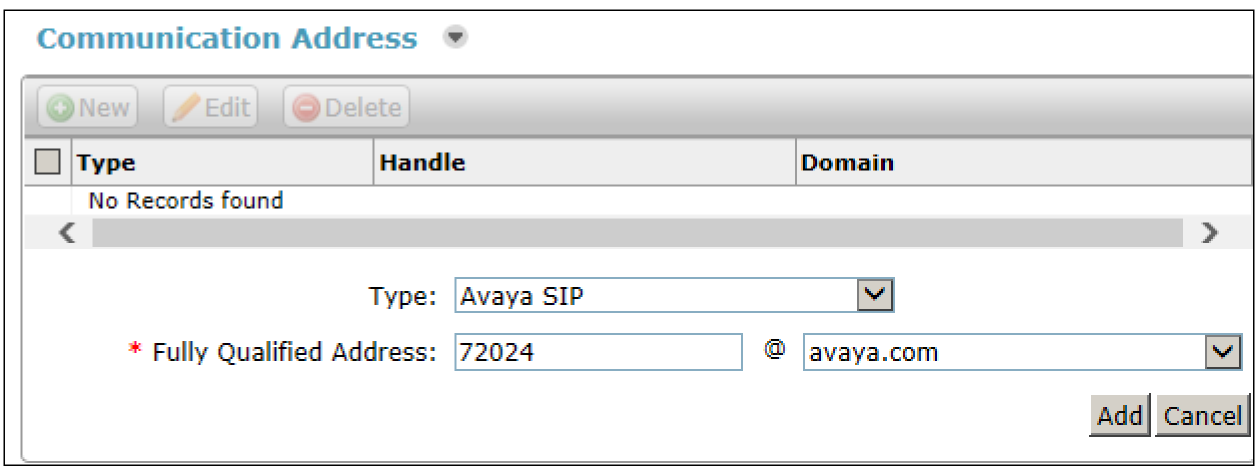
- Identity section
  - **Last Name –** Enter last name of user.
  - **First Name –** Enter first name of user.
  - **Login Name –** Enter extension number@sip domain. The sip domain is defined as Authoritative Domain in Communication Manager.
  - **Password –** Enter password to be used to log into System Manager.
  - **Confirm Password –** Repeat value entered above.



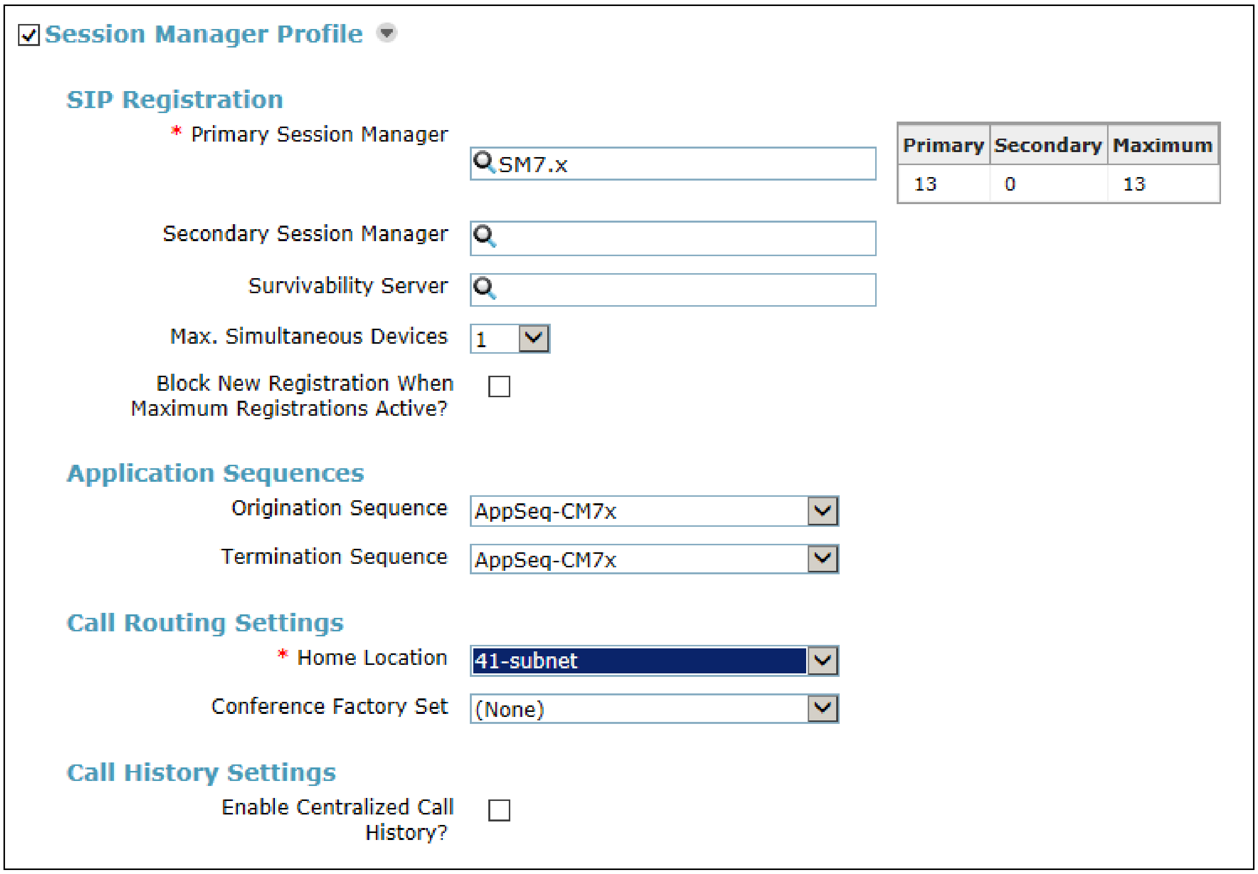
- Communication Profile section
    - **Communication Profile Password –** Type Communication profile password in this field
    - **Confirm Password –** Repeat value entered above.



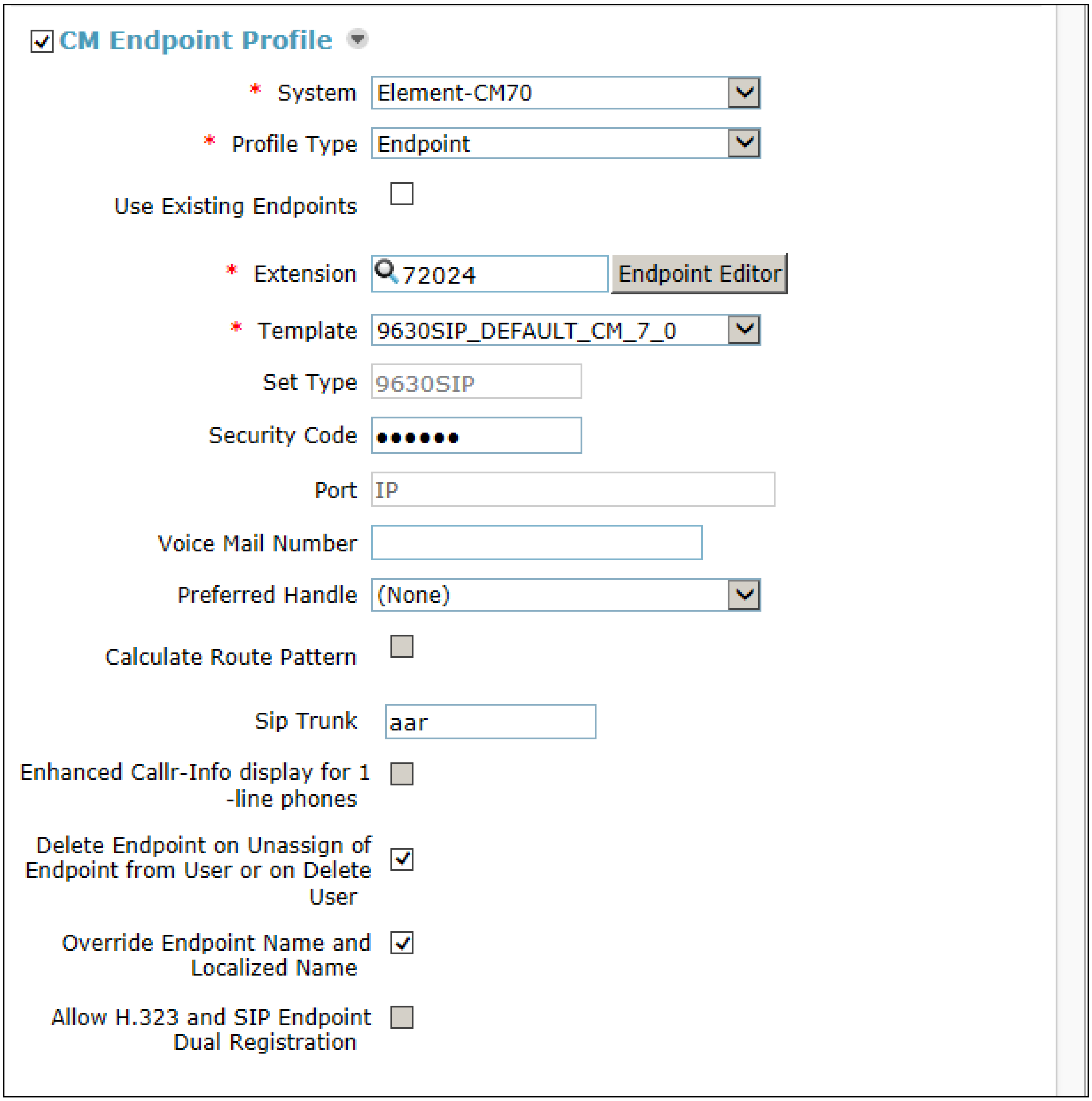
- Communication Address sub-section
    - **Fully Qualified Address –** Enter the extension of the user and select a domain name.
    - Click the **Add** button



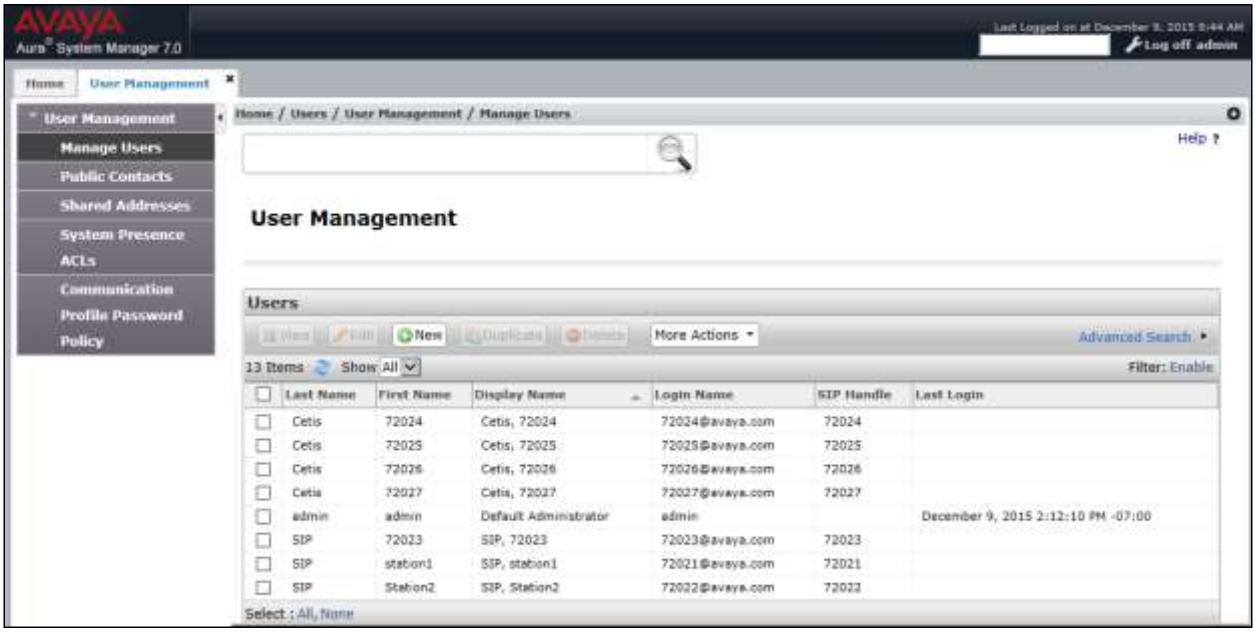
- <u>Session Manager Profile section</u>
  - **Primary Session Manager** – Select one of the Session Managers.
  - **Secondary Session Manager** – Select **(None)** from the drop-down menu.
  - **Survivability Server** – Select **(None)** from the drop-down menu.
  - **Origination Sequence –** Select Application Sequence defined for Communication Manager.
  - **Termination Sequence** – Select Application Sequence for Communication Manager.
  - **Home Location** – Select Location**.**



- Endpoint Profile section
  - **System** – Select Managed Element defined in System Manager.
  - **Profile Type** – Select **Endpoint**.
  - **Extension** - Enter same extension number used in this section.
  - **Template** – Select template for type of SIP phone
  - **Security Code** – Enter numeric value used to logon to SIP telephone. (**Note:** this field must match the value entered for the Shared Communication Profile Password field.
  - Click **Commit** at the bottom of the page.



The following page shows the Cetis M203IP users created during the test.

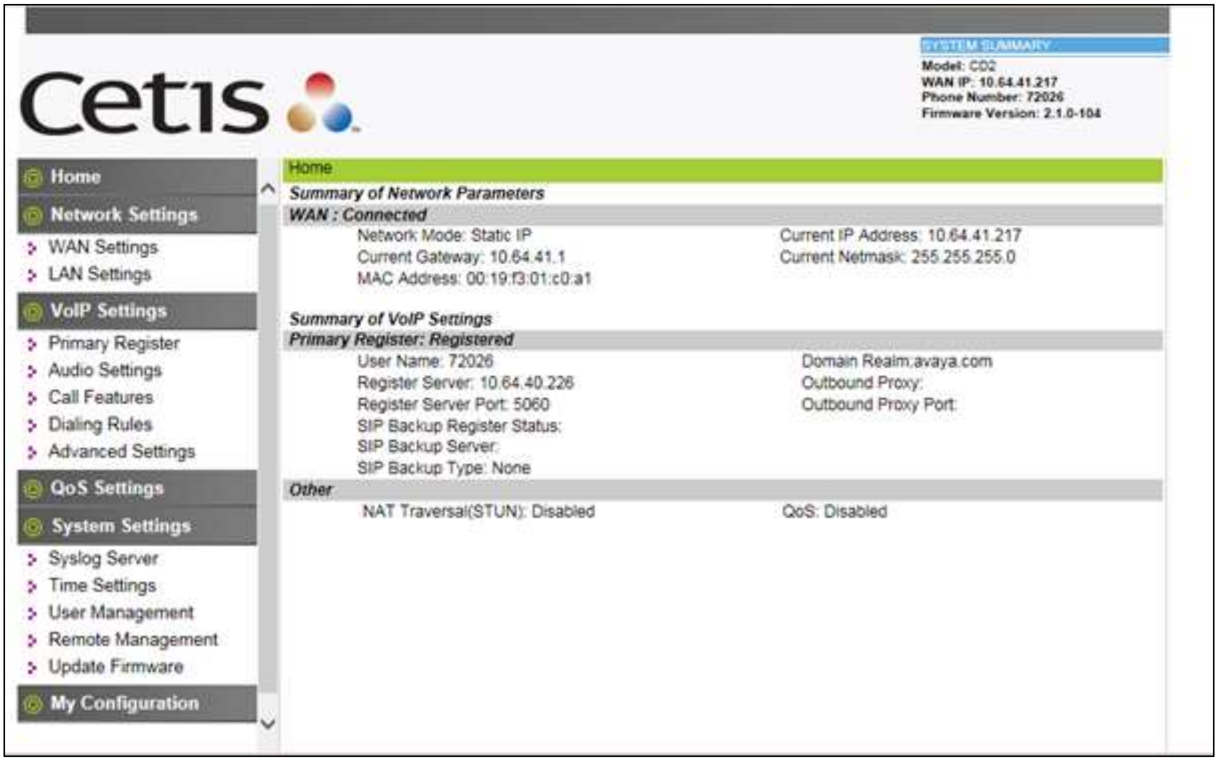


6. Configure Cetis M-Series M203IP SIP cordless Telephones

In this section, an assumption was made that an engineer was able to connect to the phone thru the web interface (i.e, using the default imp address). To configure the phone setting, enter http://<IP address of the Cetis M203 SIP Telephone> in the URL field of your browser. Log in with the appropriate credentials for accessing the Cetis M203IP settings page.

Access the Cetis M203IP SIP Telephones web interface using the URL "http://ip-address" in an Internet browser window, where "ip-address" is the IP address of the Cetis telephone. By default, DHCP is enabled on the Cetis telephones. For this compliance test, a static IP address was assigned to the Cetis telephone. To determine the IP address assigned to the Cetis telephone, enter **47# on the telephone to hear the IP address

To view the network configuration, select the **WAN Settings** under the **Network Settings** section. As it is visible, the firmware version was **2.1.0.104**. However, during the compliance test, Cetis provide an update, which is **2.1.0.105**.



Note**:** Cetis SIP firmware follows a naming convention based on model.

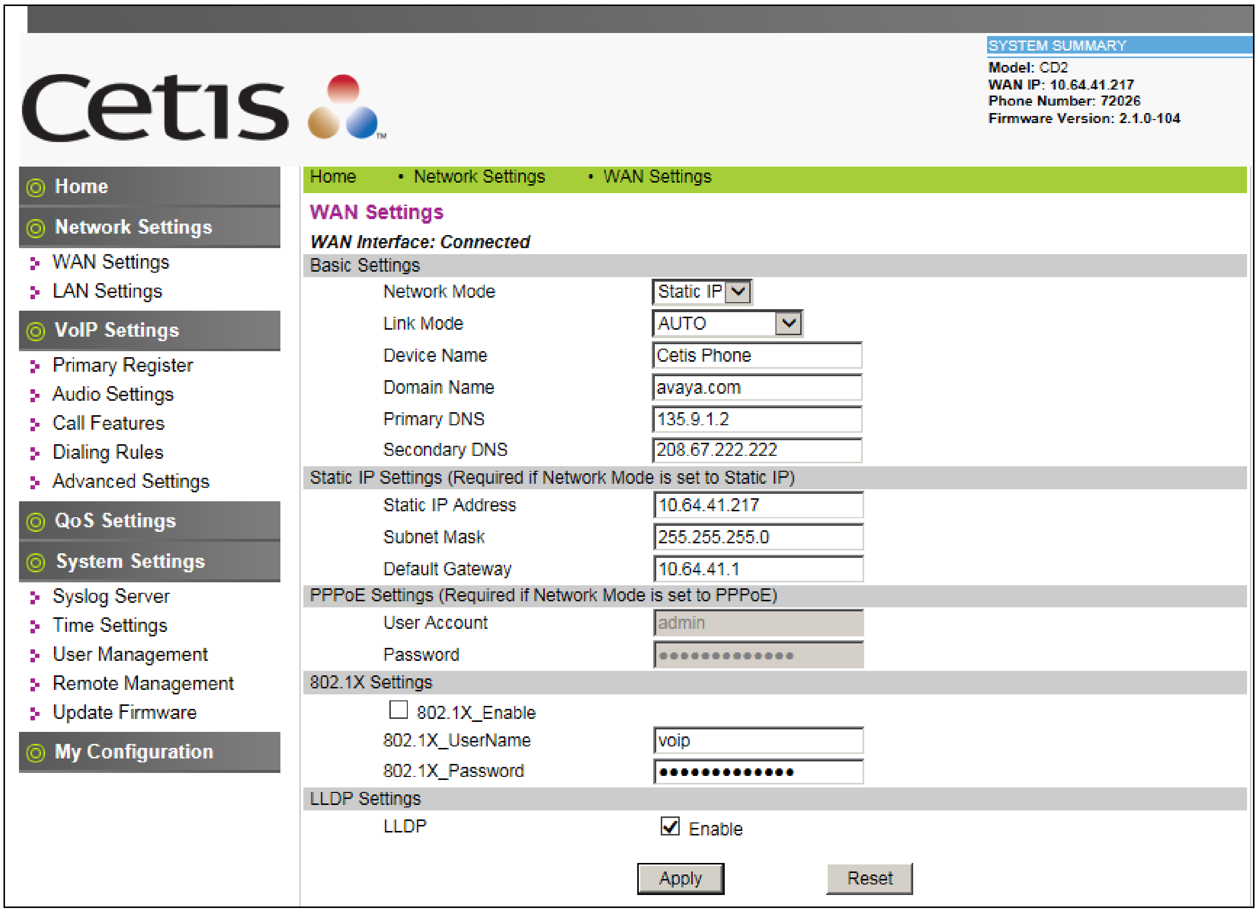
All Cetis IP phones share the same base chipset and firmware, meaning that models using the same number firmware version share the same traits and compatibility. Server registrations, SIP messaging, and call control are all the same. The different model prefixed versions are to accommodate variances in single vs. 2-line capability, corded vs. cordless radio handsets and LCD display screen sizes. Example: CC2-2.1.0-105.bin is the firmware for **C**etis **C**orded **2**-line models including M200IP, E200IP,and ND2200IP

CC1	**E100IP, M100IP, ND2100IP** : 1-line, corded (LCD and non-LCD models)
CC2	**E200IP, M200IP, ND2200IP** : 2-line, corded (LCD and non-LCD models)
CD1	**9600IP, E103IP, M103IP, NDC2100IP** : No LCD display, 1-line, cordless
CD2	**9602IP, E203IP, M203IP, NDC2200IP** : No LCD display, 2-line, cordless
C31	**3300IP** : 2-Line LCD display, 1-line, corded
C32	**3302IP** : 2-Line LCD display, 2-line, corded
CT1	**3300IP-TRM** : 4-Line LCD display, 1-line, corded, different keys, Trimline form
CT2	**3302IP-TRM** : 4-Line LCD display, 2-line, corded, different keys, Trimline form
CM1	**M100IP-TRM** : No LCD Display, 1-line, corded, different keys, Trimline form
CM2	**M200IP-TRM** : No LCD Display, 2-line, corded, different keys, Trimline form

In the **WAN Settings** page, provide the following information:

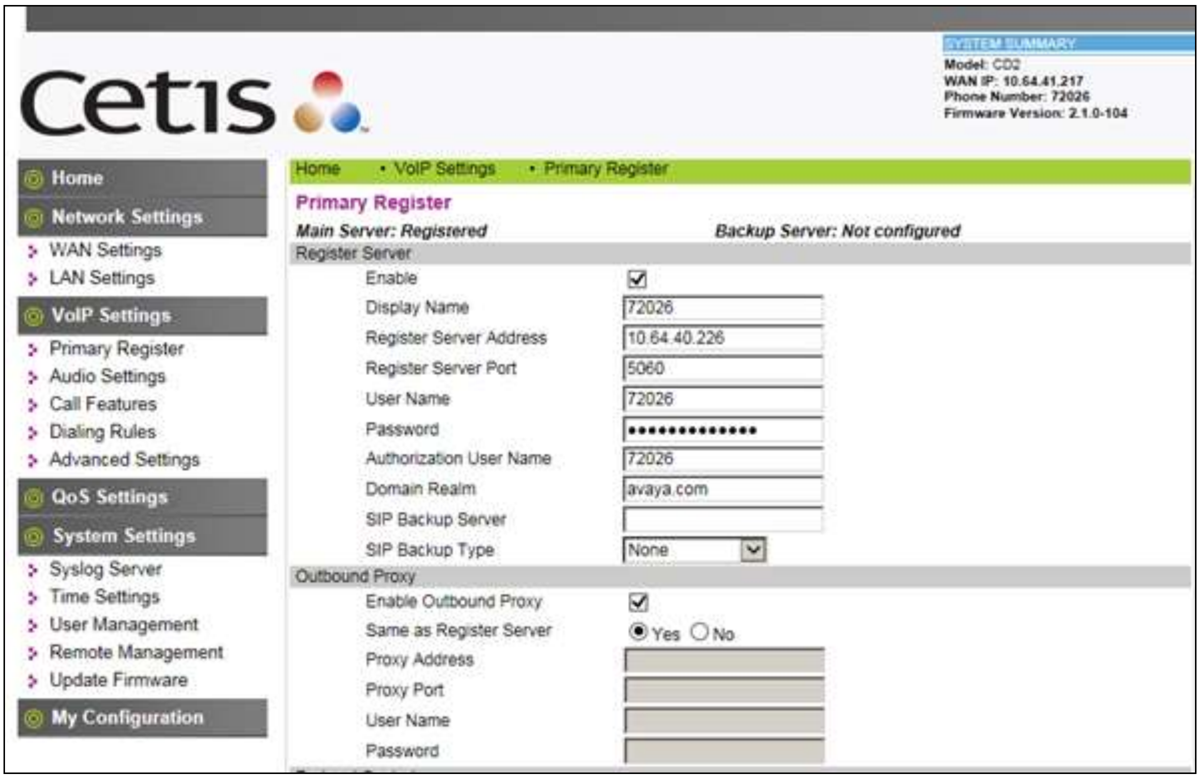
- **Static IP Address**
- **Subnet Mask**
- **Default Gateway**
- Click **Apply**.

During the compliance test, a static IP address was utilized. The following screen show what was configured and used.

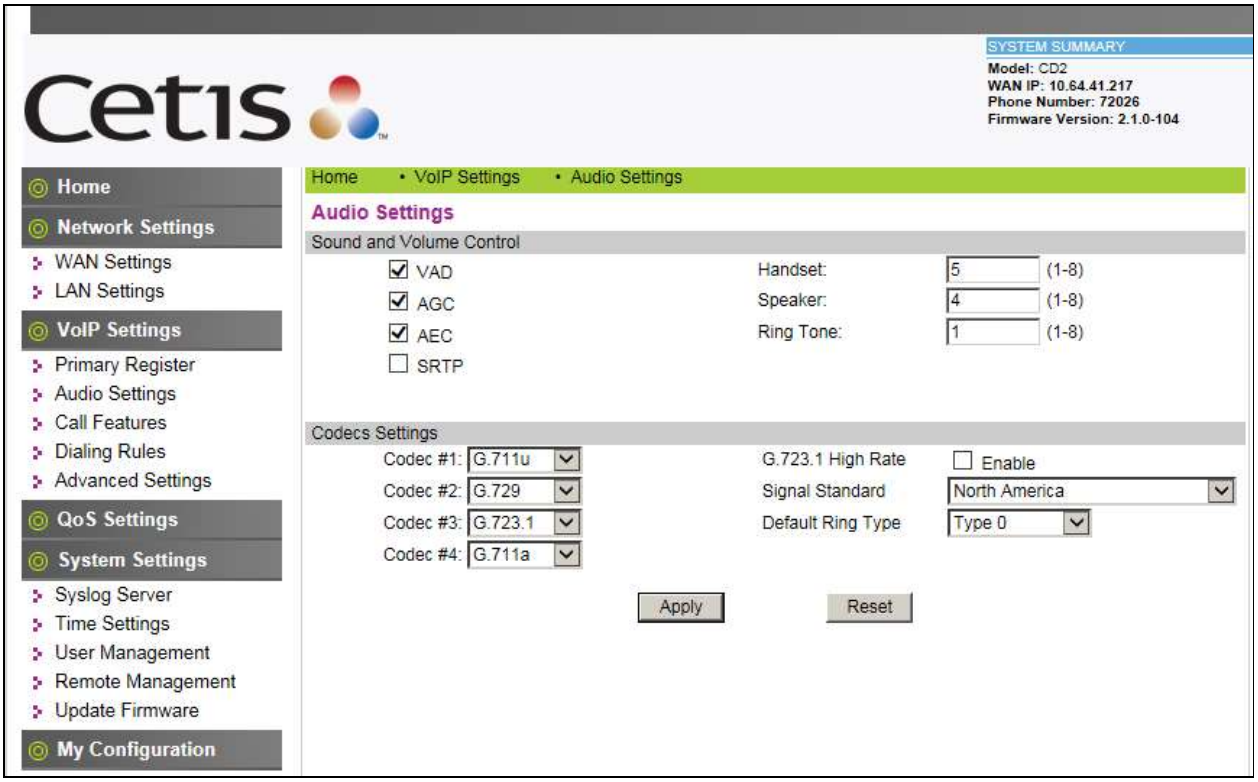


Select **Primary Register** under the **VoIP Settings** section.
Provide the following information:

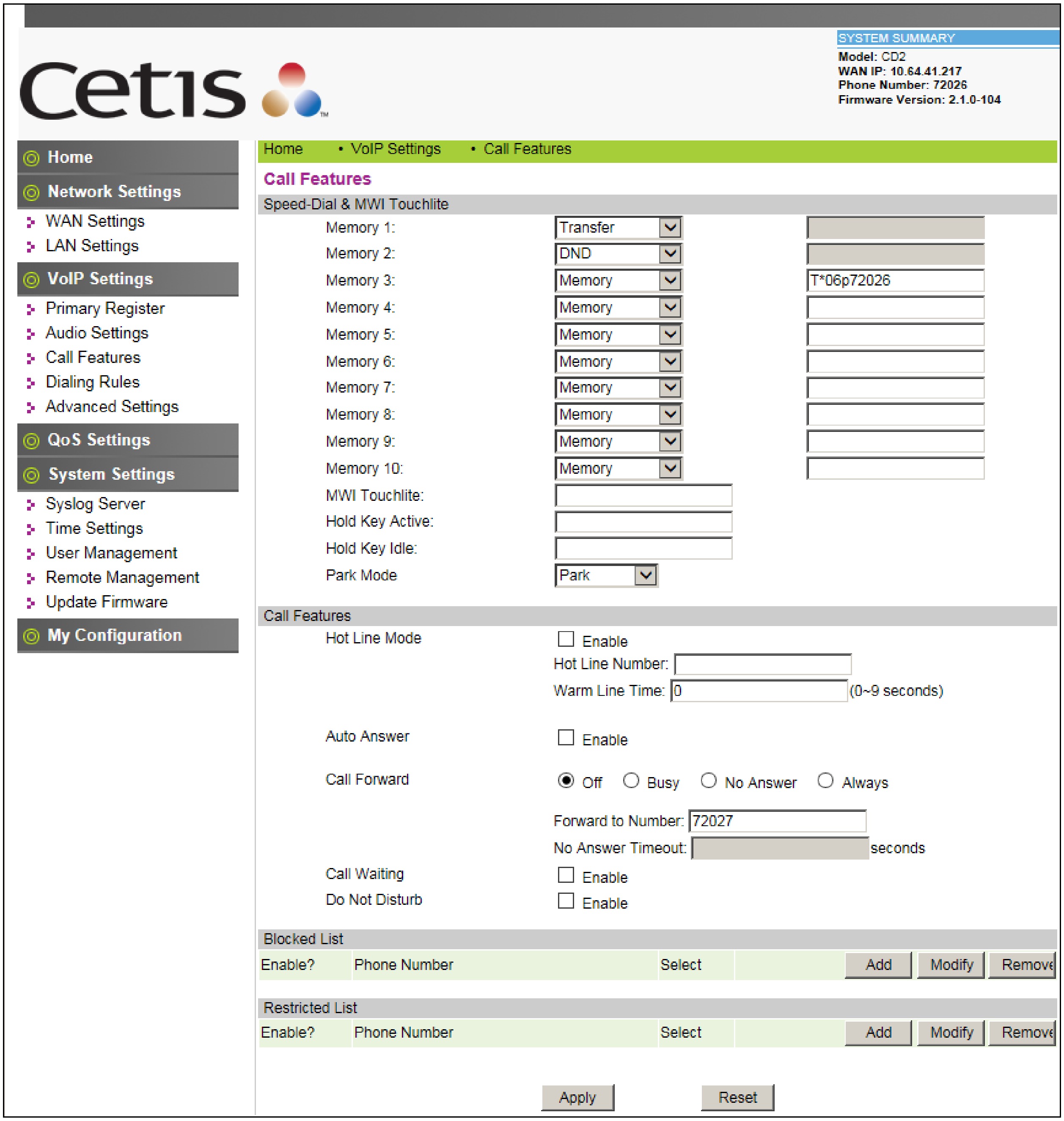
- Click the checkbox on the **Enable** field.
- **Display Name** – Enter a descriptive name.
- **Register Server Address** – Enter the IP address of Session Manager.
- **Register Server Port** – Enter **5060** for UDP.
- **User Name –** Enter the user name created in **Section 5.1**.
- **Password -** Enter the password created in **Section 5.1**.
- **Authorization User Name** – Enter the User Name
- **Domain Realm –** Used **avaya.com** during the test.
- Click **Apply**.



Select **Audio Settings** under the **VoIP Settings** section. In this page, a customer can prioritize codec settings.



Select **Call Features** under the **VoIP Settings** section. In this page, a customer can program the memory buttons. For Cetis M203IP comes with 10 memory buttons. Under the Call Features section in the right pane, two features (Do Not Disturb and Call Forward) are tested. After the configuration is completed, click **Apply**.

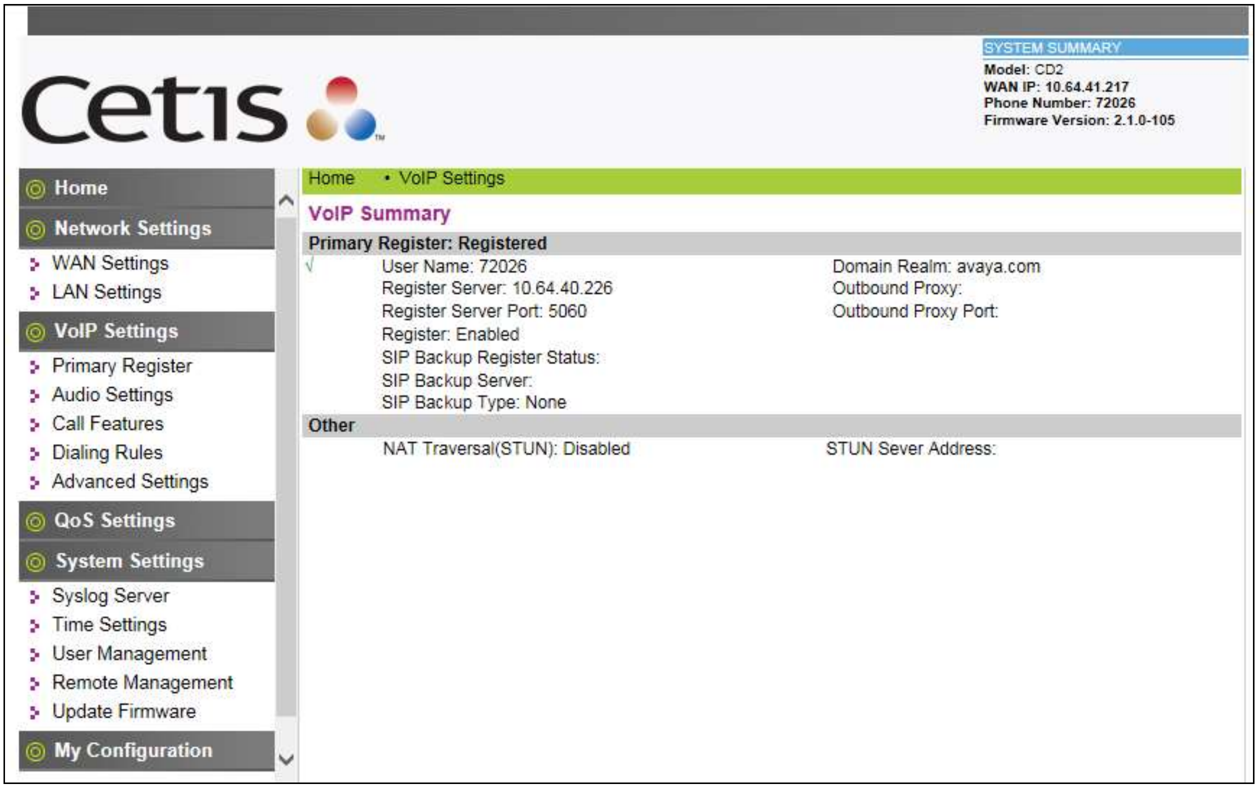


7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and the Cetis M203IP SIP Telephones.

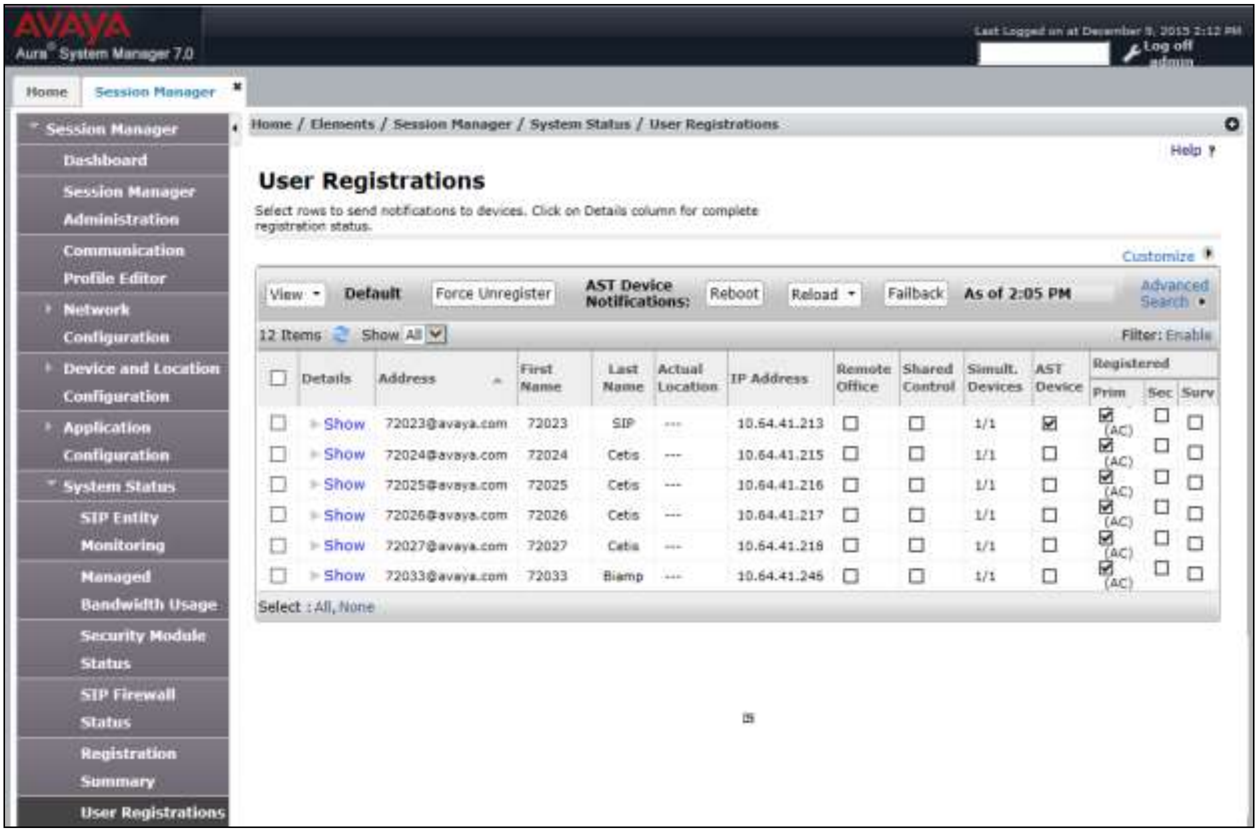
7.1. Cetis M203IP SIP Telephones.

Select **VOIP Settings** in the left pane to display the **VoIP Summary** page. Verify that the **Primary Register** is set to *Registered*.



7.1. Session Manager.

Web access to System Manager with appropriate credentials, and navigate to **Home → Elements → Session Manager → System Status → User Registration**. Verify the Cetis M203IP SIP Telephones are registered to Session Manager.



8. Conclusion

These Application Notes have described the administration steps required to integrate the Cetis M203IP SIP Telephones with Avaya Aura® Session Manager. The Cetis SIP telephones registered successfully with Avaya Aura® Session Manager via SIP. Incoming and outgoing calls were placed to/from the Cetis SIP telephones and basic telephony and hospitality features were exercised. All test cases passed with observations noted in **Section 2.2**.

9. 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*, Release 7.0, August 2015, Issue 1,Document Number 03-300509

[2] *Administering Avaya Aura® Session Manager*, Release 7.0, August2015, Issue 1

[3] *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0, December 2015, Issue 1

[4] Cetis M203IP *VoIP Phone User's Manual*.

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.