



Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the steps required to integrate Cetus M-Series M203IP SIP cordless Telephones with Avaya Aura® Session Manager. The Cetus 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 Cetus M-Series M203IP SIP cordless Telephones (hereon refers to as Cetus M203IP SIP Telephones) with Avaya Aura® Session Manager. The Cetus M203IP SIP Telephones were designed for the hospitality industry. In the compliance test, Cetus 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 Cetus 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 Cetus 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 Cetus M203IP SIP Telephones with Session Manager.
- Calls between Cetus telephones and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetus 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 Cetus telephones.
- Proper system recovery after a restart of the Cetus telephones and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

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

2.3. Support

For technical support on the Cetus M203IP SIP Telephone, contact Cetus 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 Cetus M203IP SIP Telephones with Session Manager. The Cetus telephones registered with Session Manager via SIP.

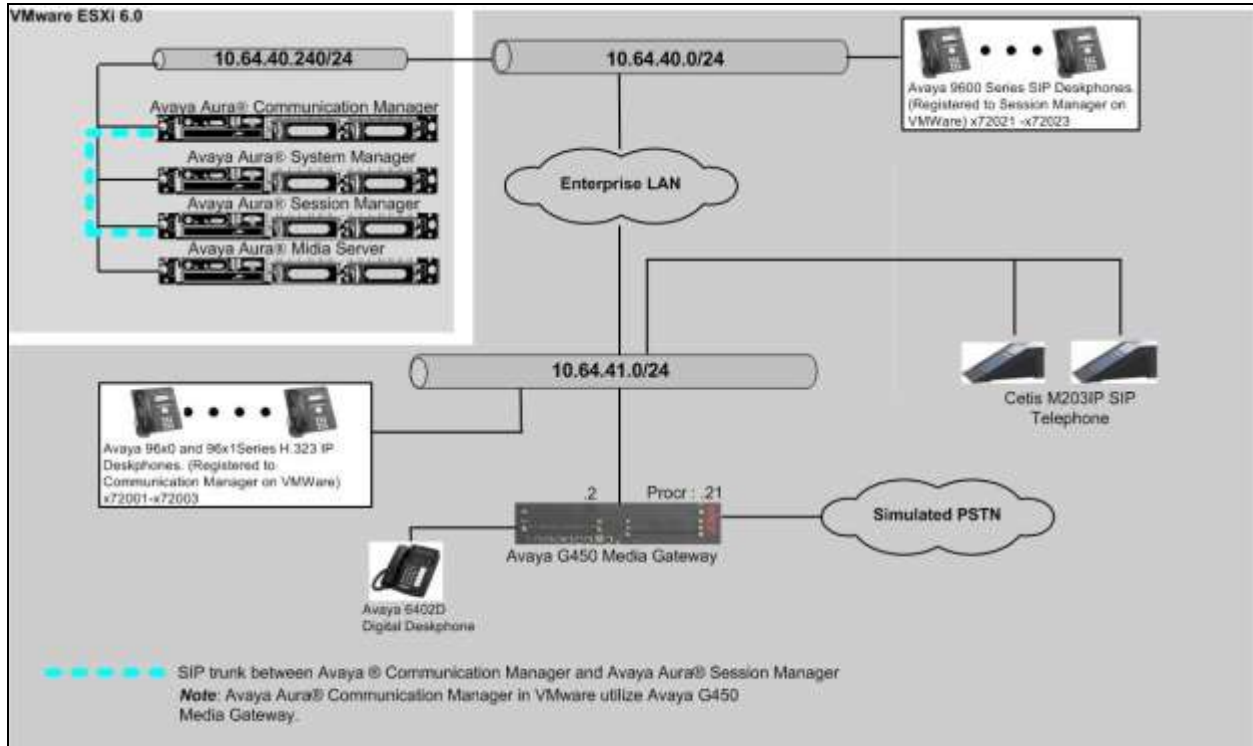


Figure 1: Cetus M203IP Telephones with Avaya Aura® Session Manager

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		7.0 (R017x.00.0.441.0)
Avaya G450 Media Gateway		37.19.0
Avaya Aura® Media Server		7.7.0.226
VMware		ESXi 6.0
Avaya 96x0 and 96x1 Series IP Deskphones		
	9620 (H.323)	3.25
	9621G (H.323)	6.6
Avaya 96x0 and 96x1 Series SIP Deskphones		
	9611G	7.0.0.39
	9650	2.6.14
Cetis M203IP		2.1.0-105

5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. 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.

The screenshot displays the 'New User Profile' form in the Avaya Aura System Manager 7.0 web interface. The form is divided into several sections: 'User Provisioning Rule' (with a dropdown menu), 'Identity' (with fields for Last Name, First Name, Middle Name, and Description), and 'Login Name' (with a dropdown menu for Authentication Type, Password, and Confirm Password). The 'Last Name' field is filled with 'Cetis', 'First Name' with '72024', and 'Login Name' with '72024@avaya.com'. The 'Authentication Type' is set to 'Basic'. The 'Password' and 'Confirm Password' fields are masked with asterisks. The interface also shows a navigation menu on the left with 'Manage Users' selected, and a breadcrumb trail at the top: 'Home / Users / User Management / Manage Users'.

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

Identity * Communication Profile Membership Contacts

Communication Profile

Communication Profile Password: [Masked]

Confirm Password: [Masked]

+ New - Delete Done Cancel

Name
<input checked="" type="radio"/> Primary

Select : None

* Name: Primary

Default :

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

Communication Address

+ New Edit - Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

* Fully Qualified Address: 72024 @ avaya.com

Add Cancel

- Session Manager Profile section
 - **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.

Session Manager Profile ▼

SIP Registration

* Primary Session Manager

Primary	Secondary	Maximum
13	0	13

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices ▼

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▼

Termination Sequence ▼

Call Routing Settings

* Home Location ▼

Conference Factory Set ▼

Call History Settings

Enable Centralized Call History?

- 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.

CM Endpoint Profile ▼

* System ▼

* Profile Type ▼

Use Existing Endpoints

* Extension

* Template ▼

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle ▼

Calculate Route Pattern

Sip Trunk

Enhanced Callr-Info display for 1
-line phones

Delete Endpoint on Unassign of
Endpoint from User or on Delete
User

Override Endpoint Name and
Localized Name

Allow H.323 and SIP Endpoint
Dual Registration

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

The screenshot displays the Avaya Aura System Manager 7.0 User Management interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 7.0', and a user session indicator showing 'Last Logged on at December 9, 2015 2:44 AM' and a 'Log off admin' button. The left sidebar contains a menu with 'User Management' selected, and sub-items: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main content area is titled 'User Management' and features a search bar and a 'Help ?' link. Below the title is a 'Users' section with a toolbar containing 'View', 'Edit', 'New', 'Duplicate', 'Delete', and 'More Actions' buttons, along with an 'Advanced Search' link. A table lists 13 items, with columns for 'Last Name', 'First Name', 'Display Name', 'Login Name', 'SIP Handle', and 'Last Login'. The table contains entries for Cetus users (72024-72027), an admin user, and SIP users (station1, station2). A 'Filter: Enable' dropdown is visible on the right side of the table. At the bottom left of the table area, it says 'Select: All, None'.

<input type="checkbox"/>	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
<input type="checkbox"/>	Cetus	72024	Cetus, 72024	72024@avaya.com	72024	
<input type="checkbox"/>	Cetus	72025	Cetus, 72025	72025@avaya.com	72025	
<input type="checkbox"/>	Cetus	72026	Cetus, 72026	72026@avaya.com	72026	
<input type="checkbox"/>	Cetus	72027	Cetus, 72027	72027@avaya.com	72027	
<input type="checkbox"/>	admin	admin	Default Administrator	admin		December 9, 2015 2:12:10 PM -07:00
<input type="checkbox"/>	SIP	72023	SIP, 72023	72023@avaya.com	72023	
<input type="checkbox"/>	SIP	station1	SIP, station1	72021@avaya.com	72021	
<input type="checkbox"/>	SIP	Station2	SIP, Station2	72022@avaya.com	72022	

6. Configure Cetus M-Series M203IP SIP cordless Telephones

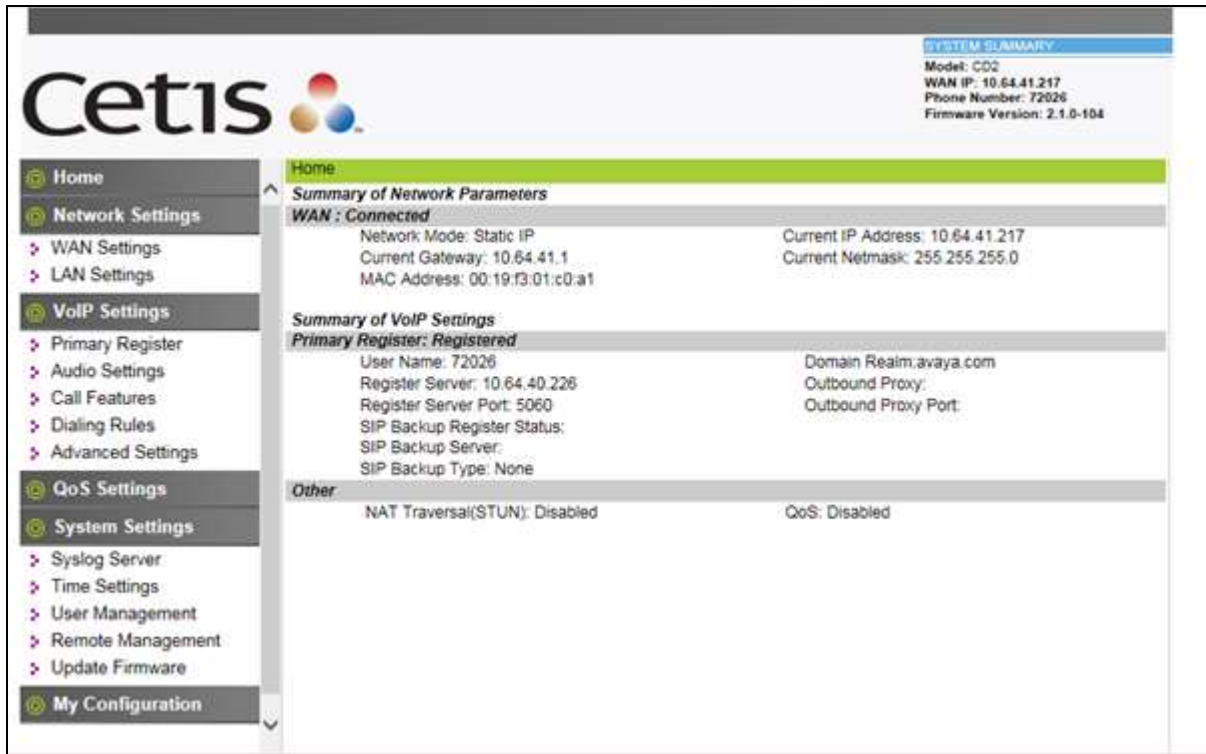
In this section, an assumption was made that an engineer was able to connect to the phone thru the web interface (i.e, using the default imp address). To configure the phone setting, enter <http://<IP address of the Cetus M203 SIP Telephone>> in the URL field of your browser. Log in with the appropriate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

Access the Cetus 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 Cetus telephone. By default, DHCP is enabled on the Cetus telephones. For this compliance test, a static IP address was assigned to the Cetus telephone. To determine the IP address assigned to the Cetus telephone, enter **47# on the telephone to hear the IP address



The image shows a web interface for user login. It features a green header bar with the text "USER LOGIN" in white. Below the header, there are two input fields: "Username" and "Password". The "Username" field is on the left, and the "Password" field is on the right. Below the input fields, there are two buttons: "Login" and "Cancel". The "Login" button is on the left, and the "Cancel" button is on the right. The background of the form is light green.

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, Cetus provide an update, which is **2.1.0.105**.



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

All Cetus 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 Cetus Corded 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.

The screenshot displays the Cetis web interface for WAN Settings. The breadcrumb trail is Home > Network Settings > WAN Settings. The page title is 'WAN Settings' and the status is 'WAN Interface: Connected'. The configuration is organized into several sections:

- Basic Settings:** Network Mode (Static IP), Link Mode (AUTO), Device Name (Cetis Phone), Domain Name (avaya.com), Primary DNS (135.9.1.2), and Secondary DNS (208.67.222.222).
- Static IP Settings (Required if Network Mode is set to Static IP):** Static IP Address (10.64.41.217), Subnet Mask (255.255.255.0), and Default Gateway (10.64.41.1).
- PPPoE Settings (Required if Network Mode is set to PPPoE):** User Account (admin) and Password (masked).
- 802.1X Settings:** 802.1X_Enable (unchecked), 802.1X_UserName (voip), and 802.1X_Password (masked).
- LLDP Settings:** LLDP (checked/Enabled).

Buttons for 'Apply' and 'Reset' are located at the bottom of the configuration area. A 'SYSTEM SUMMARY' box in the top right corner provides device information: Model: CD2, WAN IP: 10.64.41.217, Phone Number: 72026, and Firmware Version: 2.1.0-104.

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**.

Cetis

SYSTEM SUMMARY
Model: CD2
WAN IP: 10.64.41.217
Phone Number: 72026
Firmware Version: 2.1.0-104

Home • VoIP Settings • Primary Register

Primary Register
Main Server: Registered Backup Server: Not configured


Register Server:

Enable	<input checked="" type="checkbox"/>
Display Name	72026
Register Server Address	10.64.40.226
Register Server Port	5060
User Name	72026
Password	*****
Authorization User Name	72026
Domain Realm	avaya.com
SIP Backup Server	
SIP Backup Type	None

Outbound Proxy

Enable Outbound Proxy	<input checked="" type="checkbox"/>
Same as Register Server	<input checked="" type="radio"/> Yes <input type="radio"/> No
Proxy Address	
Proxy Port	
User Name	
Password	

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

Cetis 

SYSTEM SUMMARY
Model: CD2
WAN IP: 10.64.41.217
Phone Number: 72026
Firmware Version: 2.1.0-104

Home • VoIP Settings • Audio Settings

Audio Settings

Sound and Volume Control

<input checked="" type="checkbox"/> VAD	Handset:	<input type="text" value="5"/> (1-8)
<input checked="" type="checkbox"/> AGC	Speaker:	<input type="text" value="4"/> (1-8)
<input checked="" type="checkbox"/> AEC	Ring Tone:	<input type="text" value="1"/> (1-8)
<input type="checkbox"/> SRTP		

Codecs Settings

Codec #1:	<input type="text" value="G.711u"/>	G.723.1 High Rate	<input type="checkbox"/> Enable
Codec #2:	<input type="text" value="G.729"/>	Signal Standard	<input type="text" value="North America"/>
Codec #3:	<input type="text" value="G.723.1"/>	Default Ring Type	<input type="text" value="Type 0"/>
Codec #4:	<input type="text" value="G.711a"/>		

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**.

Cetis

SYSTEM SUMMARY
 Model: CD2
 WAN IP: 10.64.41.217
 Phone Number: 72026
 Firmware Version: 2.1.0-104

Home • VoIP Settings • Call Features

Call Features

Speed-Dial & MWI Touchlite

Memory 1:	Transfer	
Memory 2:	DND	
Memory 3:	Memory	T*06p72026
Memory 4:	Memory	
Memory 5:	Memory	
Memory 6:	Memory	
Memory 7:	Memory	
Memory 8:	Memory	
Memory 9:	Memory	
Memory 10:	Memory	
MWI Touchlite:		
Hold Key Active:		
Hold Key Idle:		
Park Mode	Park	

Call Features

Hot Line Mode Enable
 Hot Line Number:
 Warm Line Time: (0~9 seconds)

Auto Answer Enable

Call Forward Off Busy No Answer Always
 Forward to Number: 72027
 No Answer Timeout: seconds

Call Waiting Enable
 Do Not Disturb Enable

Blocked List

Enable?	Phone Number	Select	Add	Modify	Remove

Restricted List

Enable?	Phone Number	Select	Add	Modify	Remove

Apply Reset

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*.

The screenshot displays the Cetis M203IP SIP Telephone web interface. The top left features the Cetis logo. The top right corner shows a 'SYSTEM SUMMARY' box with the following information: Model: CD2, WAN IP: 10.64.41.217, Phone Number: 72026, and Firmware Version: 2.1.0-105. The left navigation pane is expanded to 'VoIP Settings', which includes sub-items like Primary Register, Audio Settings, Call Features, Dialing Rules, Advanced Settings, QoS Settings, System Settings, and My Configuration. The main content area is titled 'VoIP Summary' and shows the 'Primary Register' status as 'Registered'. Below this, a table lists configuration details: User Name: 72026, Register Server: 10.64.40.226, Register Server Port: 5060, Register: Enabled, SIP Backup Register Status, SIP Backup Server, and SIP Backup Type: None. The 'Other' section shows NAT Traversal(STUN): Disabled and STUN Sever Address:.

Primary Register: Registered	
User Name: 72026	Domain Realm: avaya.com
Register Server: 10.64.40.226	Outbound Proxy:
Register Server Port: 5060	Outbound Proxy Port:
Register: Enabled	
SIP Backup Register Status:	
SIP Backup Server:	
SIP Backup Type: None	

Other	
NAT Traversal(STUN): Disabled	STUN Sever Address:

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.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The main content area is titled "User Registrations" and displays a table of 12 items. The table columns are: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, Surv). The registered status for all items is checked under the "Prim" column.

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered
<input type="checkbox"/> Show	72023@avaya.com	72023	SIP	---	10.64.41.213	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> Show	72024@avaya.com	72024	Cetis	---	10.64.41.215	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> Show	72025@avaya.com	72025	Cetis	---	10.64.41.216	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> Show	72026@avaya.com	72026	Cetis	---	10.64.41.217	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> Show	72027@avaya.com	72027	Cetis	---	10.64.41.218	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/> Show	72033@avaya.com	72033	Blamp	---	10.64.41.246	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)

8. Conclusion

These Application Notes have described the administration steps required to integrate the Cetus M203IP SIP Telephones with Avaya Aura® Session Manager. The Cetus SIP telephones registered successfully with Avaya Aura® Session Manager via SIP. Incoming and outgoing calls were placed to/from the Cetus 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, August 2015, Issue 1
- [3] *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0, December 2015, Issue 1
- [4] *Cetus M203IP VoIP Phone User's Manual*.

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