



## DevConnect Program

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# Application Notes for Cetus E200IP Corded SIP 2-Line Telephone CC2-4.0.0-066 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1 – Issue 1.0

## Abstract

These Application Notes describe the configuration steps required for Cetus E200IP Corded SIP 2-Line Telephone to integrate with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. Cetus E200IP is a corded SIP 2-line telephone that is designed for the hospitality industry and registers with Avaya Aura® Session Manager as a SIP endpoint.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 Avaya DevConnect Program.

## 1. Introduction

These Application Notes describe the configuration steps required for Cetus E200IP Corded SIP 2-Line Telephone to integrate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Cetus E200IP Corded SIP 2-Line Telephone is designed for the hospitality industry. In the compliance test, Cetus E200IP registered with Avaya Aura® Session Manager as a SIP endpoint.

## 2. General Test Approach and Test Results

The general test approach was to place calls to and from Cetus E200IP and exercise basic telephone operations.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Cetus E200IP did not include use of any specific encryption features as requested by Cetus.

## 2.1. Interoperability Compliance Testing

The following areas were evaluated in the interoperability compliance test:

- Registration of Cetus E200IP to Session Manager.
- Calls between Cetus E200IP and Avaya SIP and H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetus E200IP and the PSTN.
- G.711 and G.729 codec support, codec negotiation, and session refresh interval.
- UDP transport.
- Proper recognition of DTMF tones, Voice Mail and Message Waiting Indicator.
- Basic telephony features including answer/drop, hold/resume, mute/un-mute, forwarding, blind and attended transfer, and attended conference.
- Automatic Wakeup Call and Housekeeping status hospitality features.
- Serviceability testing to validate recovery from network connectivity loss.

## 2.2. Test Results

All test cases were completed successfully.

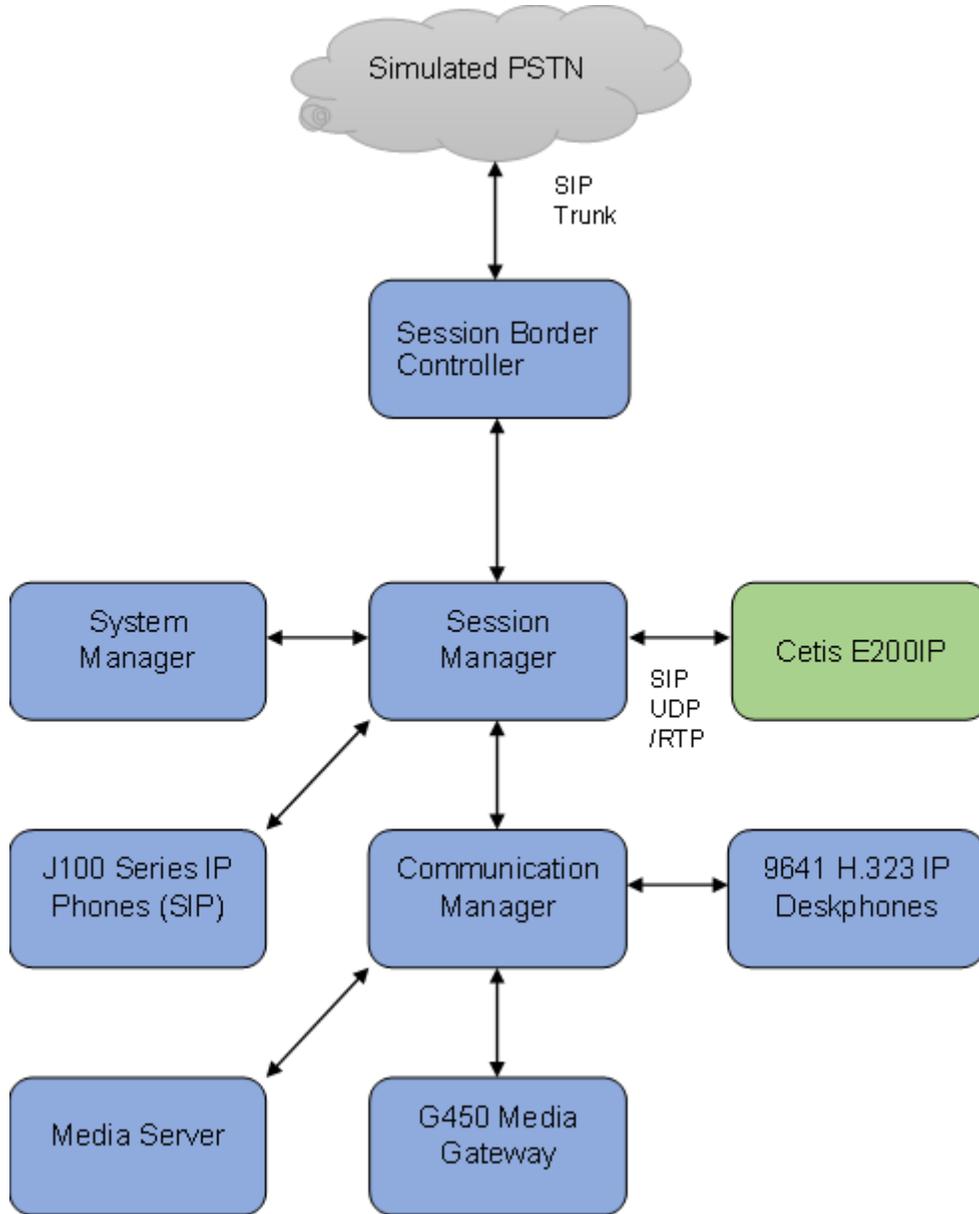
## 2.3. Support

For technical support on the Cetus E200IP Telephone, contact Cetus Support via phone, email, or website.

- **Phone:** +1 (719) 638-8821
- **Email:** [sipsupport@cetisgroup.com](mailto:sipsupport@cetisgroup.com)
- **Web:** <https://support.cetis.com/index.php>

### 3. Reference Configuration

**Figure 1** illustrates the test configuration diagram for Cetus E200IP integrated with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.



**Figure 1: Avaya Test Configuration for Cetus E200IP Phones**

## 4. Equipment and Software Validated

The following equipment and software were used for the compliance test provided:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	10.1.2.0 FP2 10.1.2.0.0.974.27783
Avaya Aura® System Manager	10.1.2.0 Feature Pack 2 10.1.2.0.0715476
Avaya Aura® Session Manager	10.1.2.0 Feature Pack 2 10.1.2.0.1012016
Avaya Session Border Controller for Enterprise	10.1.0.0-32-21432
Avaya Aura® Media Server	10.1.0.125
Avaya G450 Media Gateway	FW 42.18.0
Avaya 9641G IP DeskPhone	6.8.5.4 (H.323)
Avaya J179 IP Phone	4.1.1.0.7 (SIP)
Cetis E200IP Corded 2-Line IP Telephone	CC2-4.0.0-066

## 5. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager OPS Licensed Capacity
- Administer IP Network Region
- Administer IP Codec Set

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials. The configuration steps illustrate field values changed for this reference configuration. Default values were used for all other fields.

**Note:** It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for Cetus E200IP is configured through System Manager in **Section 6.3**.

### 5.1. Verify Communication Manager OPS Licensed Capacity

Using the SAT, verify that the Off-PBX Stations (OPS) and SIP Trunks features 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** allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V20                                     Software Package: Enterprise
Location: 2                                         System ID (SID): 1
Platform: 28                                       Module ID (MID): 1

                                USED
Platform Maximum Ports: 48000 150
Maximum Stations: 150 73
Maximum XMOBILE Stations: 36000 0
Maximum Off-PBX Telephones - EC500: 150 0
Maximum Off-PBX Telephones - OPS: 150 42
Maximum Off-PBX Telephones - PBFMC: 150 0
Maximum Off-PBX Telephones - PVFMC: 150 0
Maximum Off-PBX Telephones - SCCAN: 0 0
Maximum Survivable Processors: 313 0

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

## 5.2. Administer IP Network Region

This IP network region is for the signaling group associated with the SIP trunk between Session Manager and Communication Manager. This form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. Verify the following values:

- **Authoritative Domain:** The applicable domain (e.g., *avaya.com*)
- **Codec Set:** The codec set number from **Section 5.3**

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 G450 Media Gateway or Media Server.

```
change ip-network-region 1                                     Page 1 of 20
                                                              IP NETWORK REGION
  Region: 1
  Location: 1          Authoritative Domain: avaya.com
    Name: Main
  MEDIA PARAMETERS          Intra-region IP-IP Direct Audio: yes
    Codec Set: 1          Inter-region IP-IP Direct Audio: yes
    UDP Port Min: 2048          IP Audio Hairpinning? n
    UDP Port Max: 3329
  DIFFSERV/TOS PARAMETERS
    Call Control PHB Value: 46
    Audio PHB Value: 46
    Video PHB Value: 26
  802.1P/Q PARAMETERS
    Call Control 802.1p Priority: 6
    Audio 802.1p Priority: 6
    Video 802.1p Priority: 5          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
```

### 5.3. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Cetus E200IP. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set *1* is specified in **IP Network Region 1** from **Section 5.2**. The form shows the list of codecs tested. Enter values for the following:

- **Audio Codec:** The audio codecs tested
- **Media Encryption:** Include *none*

**Note:** Media encryption was enabled for Avaya IP endpoints in the test configuration. Cetus E200IP wasn't configured to support SRTP, so the *none* option is included.

```
display ip-codec-set 1                                     Page 1 of 2

                                IP MEDIA PARAMETERS

Codec Set: 1

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

Media Encryption                               Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: 10-srtp-aescm256-hmac80
3: none
4:
5:
```

## 6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The steps include the following areas.

- Launch System Manager
- Verify Session Manager Listening Ports
- Administer SIP Users

### 6.1. Launch System Manager

Access Session Manager Administration web interface by entering **http://<ip-address>/SMGR** in a web browser, where **<ip-address>** is the IP address of System Manager. Log in using the appropriate credentials.

The screenshot shows the Avaya Aura Session Manager Administration web interface. On the left, there is a grey sidebar with the following text:

Recommended access to System Manager is via FQDN.  
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

On the right, there is a login form with the following fields and buttons:

User ID:

Password:

[Change Password](#)

At the bottom right, there is a blue box with the following text:

**Supported Browsers:** Firefox (minimum version 93.0), Chrome (minimum version 91.0) or Edge (minimum version 93.0).

## 6.2. Verify Session Manager Listening Ports

Each Session Manager must be configured so that Cetus E200IP can register to it using UDP/TCP. From the web interface click **Elements** → **Routing** → **SIP Entities** (not shown) and select the Session Manager entity used for registration. In the **Listen Ports** section, make sure that **TCP** and **UDP** entries are present with the respective SIP domain selected and **Endpoint** checked. Only **UDP** protocol for Cetus E200IP was tested during the compliance test.

The screenshot shows the Avaya Aura System Manager 10.1 web interface. The top navigation bar includes the Avaya logo, user information (Users), and various menu items (Elements, Services, Widgets, Shortcuts). The main content area is titled "Routing" and contains three sections:

- Failover Ports:** Includes input fields for "TCP Failover port:" and "TLS Failover port:".
- Listen Ports:** A table with 3 items. The table has columns for "Listen Ports", "Protocol", "Default Domain", "Endpoint", and "Notes". The data rows are:

Listen Ports	Protocol	Default Domain	Endpoint	Notes
5060	TCP	avaya.com	<input checked="" type="checkbox"/>	
5060	UDP	avaya.com	<input checked="" type="checkbox"/>	
5061	TLS	avaya.com	<input checked="" type="checkbox"/>	
- SIP Responses to an OPTIONS Request:** A table with 0 items. The table has columns for "Response Code & Reason Phrase", "Mark Entity Up/Down", and "Notes".

At the bottom of the interface, there are "Commit" and "Cancel" buttons.

### 6.3. Administer SIP Users

A SIP user must be created for Cetus E200IP to register to Session Manager. This configuration is automatically synchronized with Communication Manager. In Session Manager, select **Users** → **User Management** → **Manage Users** to display the **User Management** screen (not shown). Click + **New** to add a user.

#### 6.3.1. Identity

Enter values for the following required attributes for a new SIP user in the **New User Profile** screen:

- **Last Name:** Enter the last name of the user (e.g., *Cetus*)
- **First Name:** Enter the first name of the user (e.g., *E200IP*)
- **Login Name:** Enter <extension>@<sip domain> of the user (e.g., *70131@avaya.com*)

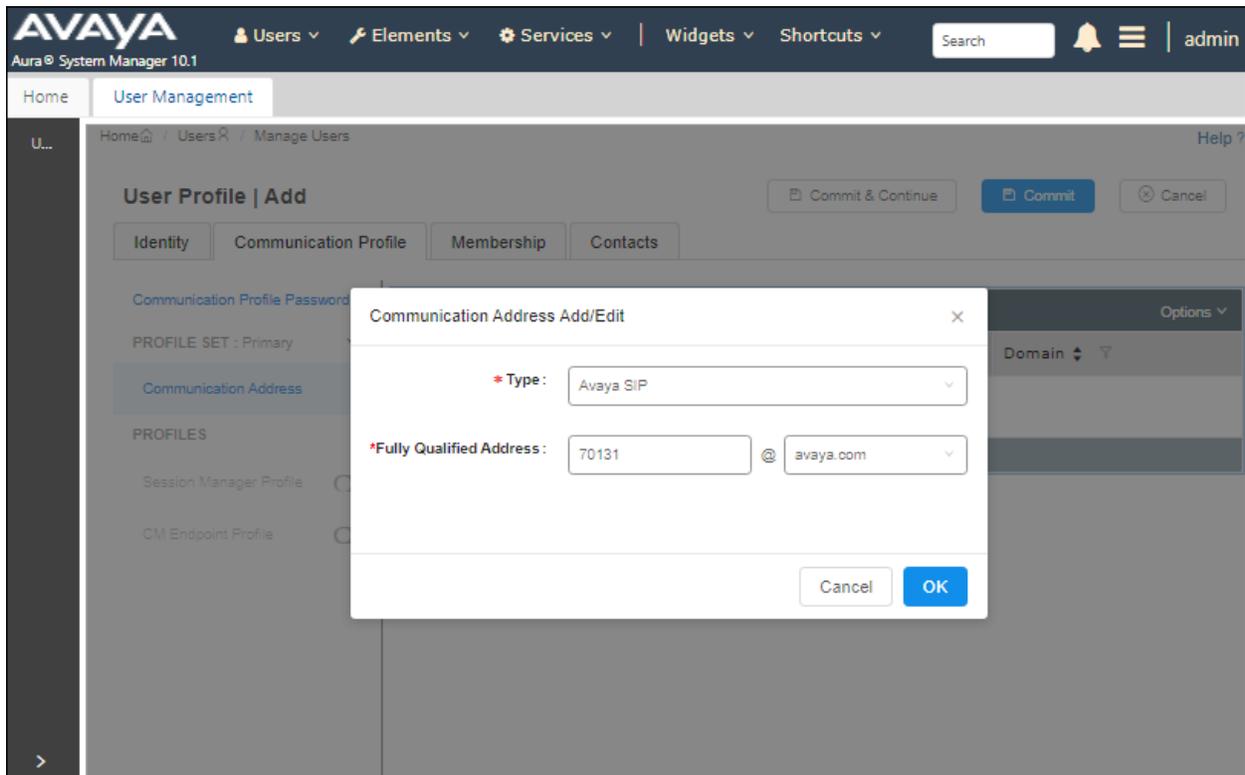
The screenshot displays the 'User Profile | Add' form in the Avaya Aura System Manager 10.1 interface. The form is divided into several sections: 'User Provisioning Rule' (a dropdown menu), 'Basic Info' (a sidebar menu), and a main form area with various input fields. The 'Basic Info' sidebar includes 'Address' and 'LocalizedName'. The main form area contains the following fields: 'Last Name' (Cetus), 'First Name' (E200IP), 'Login Name' (70131@avaya.com), 'Description' (Description Of User), 'Password', 'Confirm Password', 'Last Name (in Latin alphabet characters)' (Cetus), 'First Name (in Latin alphabet characters)' (E200IP), 'Middle Name' (Middle Name Of User), 'Email Address' (Email Address Of User), 'User Type' (Basic), and 'Localized Display Name' (Localized Display Name). At the top right of the form area, there are three buttons: 'Commit & Continue', 'Commit', and 'Cancel'. The top navigation bar includes the Avaya logo, 'Users', 'Elements', 'Services', 'Widgets', 'Shortcuts', a search bar, a notification bell, and the user name 'admin'.

### 6.3.2. Communication Address

Select the **Communication Profile** tab. Select **Communication Address** in the left-hand side list and click + **New** (not shown).

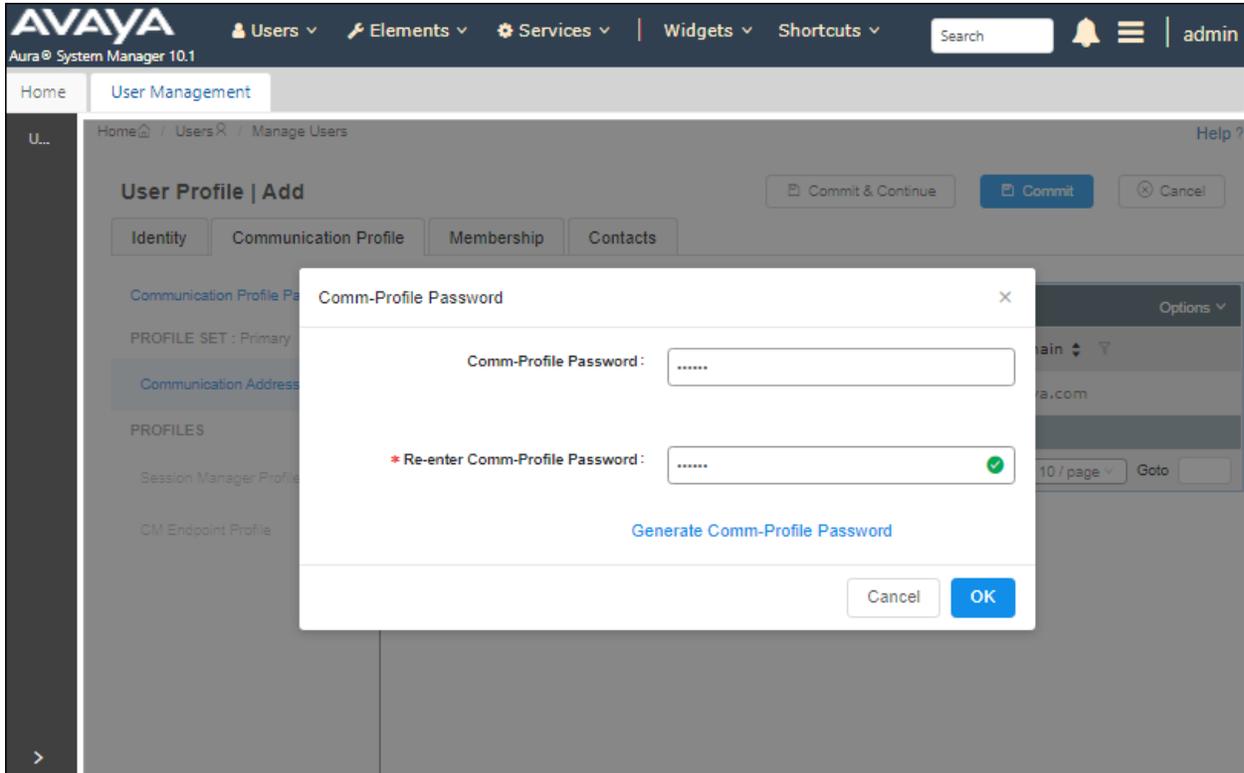
Enter the following attributes for the **Communication Address**:

- **Type:** Select *Avaya SIP* from the drop-down list
- **Fully Qualified Address:** Enter the extension number (e.g., *70131*)
- **Domain:** Enter the domain (e.g., *avaya.com*)



### 6.3.3. Communication Profile Password

Click the **Communication Profile Password** tab and in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields, enter a numeric password. This will be used to register the device. Click **OK**.



### 6.3.4. Session Manager Profile

Click on the **Session Manager Profile** slide button. For **Primary Session Manager**, **Origination Sequence**, **Termination Sequence**, and **Home Location** (not shown), select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

The screenshot displays the Avaya Aura System Manager 10.1 interface for adding a user profile. The top navigation bar includes the Avaya logo, navigation menus for Users, Elements, Services, Widgets, and Shortcuts, a search bar, and a user profile icon labeled 'admin'. The breadcrumb trail shows 'Home > User Management > Home > Users > Manage Users'. The main content area is titled 'User Profile | Add' and features three tabs: 'Identity', 'Communication Profile', and 'Membership'. The 'Communication Profile' tab is selected, and the 'Session Manager Profile' toggle is turned on. The 'SIP Registration' section contains the following fields: 'Primary Session Manager' (value: sm10), 'Secondary Session Manager' (value: Start typing...), 'Survivability Server' (value: Start typing...), and 'Max. Simultaneous Devices' (value: Select). The 'Block New Registration When Maximum Registrations Active?' checkbox is unchecked. The 'Application Sequences' section includes 'Origination Sequence' and 'Termination Sequence', both set to 'cm10 App Seq'. Action buttons 'Commit & Continue', 'Commit', and 'Cancel' are located at the top right of the form.

### 6.3.5. CM Endpoint Profile

Click on the **CM Endpoint Profile** slide button. Fill in the following fields:

- **System:** Select the relevant Communication Manager SIP Entity (e.g., *cm10*)
- **Profile Type:** Select *Endpoint*
- **Template:** Select *9641SIP\_DEFAULT\_CM\_10\_1*
- **Extension:** Enter the extension number (e.g., *70131*)

Click on the **Editor** icon in the **Extension** field to edit Communication Manager settings. Input the appropriate **Coverage Path 1** number (not shown) configured to route unanswered calls to voicemail. Click **Done** to close the Endpoint Editor. Click **Commit**.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, menu items for Users, Elements, Services, Widgets, and Shortcuts, a search bar, and a user profile icon labeled 'admin'. The main content area is titled 'User Profile | Add' and features several tabs: Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active. On the left, there is a sidebar with 'Communication Profile Password', 'PROFILE SET : Primary', 'Communication Address', and 'PROFILES' section containing 'Session Manager Profile' and 'CM Endpoint Profile' (which is selected). The main form area contains the following fields and controls:

- \* System:** A dropdown menu with 'cm10' selected.
- \* Profile Type:** A dropdown menu with 'Endpoint' selected.
- Use Existing Endpoints:** An unchecked checkbox.
- \* Extension:** A text input field containing '70131' with an edit icon.
- \* Template:** A dropdown menu with '9641SIP\_DEFAULT\_CM\_10\_1' selected.
- \* Set Type:** A text input field containing '9641SIP'.
- Security Code:** A text input field with the placeholder 'Enter Security Code'.
- Port:** A dropdown menu with 'IP' selected.
- Voice Mail Number:** An empty text input field.
- Preferred Handle:** A dropdown menu with 'Select' selected.
- Calculate Route Pattern:** An unchecked checkbox.
- Sip Trunk:** A text input field containing 'aar'.
- SIP URI:** A dropdown menu with 'Select' selected.
- Delete on Unassign from User or on Delete User:** A checked checkbox.
- Override Endpoint Name and Localized Name:** A checked checkbox.
- Allow H.323 and SIP Endpoint Dual Registration:** An unchecked checkbox.

At the top right of the form, there are three buttons: 'Commit & Continue', 'Commit', and 'Cancel'.

## 7. Configure Cetus E200IP Corded 2-Line Telephone

The steps to configure Cetus E200IP to integrate with Communication Manager and Session Manager are as follows:

- Configure IP Address
- Launch Cetus E200IP Web Administration Interface
- Configure SIP Account
- Configure Audio Settings
- Assign Feature Buttons

### 7.1. Configure IP Address

Cetus E200IP is configured for DHCP as a factory default. The following steps provide network connectivity and determine the phone IP address for use in launching the web administration Interface as detailed in **Section 7.2**:

- Connect the WAN port of Cetus E200IP to a Power over Ethernet (PoE) switch.
- Determine the assigned IP address. Use the built-in voice response which will read out the IP address. The IP address is echoed by pressing the “Quick Key” command **\*\*47#**. For more information, refer to **[4]** in **Section 10**.

### 7.2. Launch Cetus E200IP Web Administration Interface

Access the Cetus E200IP web administration interface using the URL **http://<ip-address>** in an Internet browser window, where **<ip-address>** is the IP address obtained from **Section 7.1**. The login prompt displays. Enter the appropriate **Username** and **Password**.



The screenshot shows the login page for the Cetus E200IP web administration interface. At the top, the Cetus logo is displayed, consisting of the word "Cetus" in a bold, sans-serif font followed by three colored circles (red, yellow, and blue). Below the logo is a green horizontal bar. Underneath the bar, the text "Please enter your User name and Password below to login" is written in a purple font. There are two input fields: one for "Username" and one for "Password", both with labels in purple font to their left. At the bottom of the form, there are two buttons: "Login" and "Cancel", both in a light blue color.

Once logged in, the Home screen is displayed. The WAN status and VoIP settings (not yet configured) for E200IP are shown.

The screenshot displays the Cetis web interface. At the top left is the Cetis logo. At the top right is a 'SYSTEM SUMMARY' box containing the following information: Model: CC2, WAN IP: 10.64.10.202, Phone Number1, Phone Number2, and Firmware Version: CC2-4.0.0-066. The main content area is divided into a left sidebar and a main panel. The sidebar contains a navigation menu with the following items: Home (selected), Network Settings (with sub-items WAN Settings and LAN Settings), VoIP Settings (with sub-items Primary Register, Audio Settings, Call Features, Dialing Rules, Multicast Paging, Advanced Settings, and Phonebook Settings), QoS Settings, Provisioning, and System Settings (with sub-items Logging Server, Time Settings, User Management, and System Actions). The main panel shows the 'Home' page with a green header. It contains three sections: 'Summary of Network Parameters' with a sub-section 'WAN : Connected' showing Network Mode: DHCP, Current Gateway: 10.64.10.1, MAC Address: 00:19:F3:11:28:50, Current IP Address: 10.64.10.202, and Current Netmask: 255.255.255.0; 'Summary of VoIP Settings' with a sub-section 'First Register: Not configured' showing User Name, Register Server, Register Server Port: 5060, SIP Backup Register Status: Not configured, SIP Backup Server, and SIP Backup Type: None, and a sub-section 'Second Register: Not configured' showing similar fields; and 'Other' settings showing NAT Traversal(STUN): Disabled and QoS: Disabled.

### 7.3. Configure SIP Account

Select **Primary Register** under the **VoIP Settings** section. In the **First Register Server** section, provide the following information:

- **Use Service:** Select *Enable* from the dropdown list
- **Display Name:** Enter a descriptive name (e.g., *70131*)
- **User Name:** Enter the user name created in **Section 6.3** (e.g., *70131*)
- **Authorization User Name:** Enter the user name as configured in **Section 6.3** (e.g., *70131*)
- **Password:** Enter the password created in **Section 6.3**
- **Register Server Port:** Enter *5060*
- **Register Server Address:** Enter the signaling IP address of Session Manager
- **Domain Realm:** Enter the default sip domain from **Section 6.2** (e.g., *avaya.com*)
- **MWI Subscribe:** Select *Enable* from the dropdown list

Use default values for the remaining fields.

**Cetis** 

**SYSTEM SUMMARY**  
 Model: CC2  
 WAN IP: 10.64.10.202  
 Phone Number1:  
 Phone Number2:  
 Firmware Version: CC2-4.0.0-066

Home • VoIP Settings • Primary Register

**Primary Register**

First Server: *Not configured* Backup Server: *Not configured*

First Register Server

Use Service	Enable ▾
Display Name	70131
User Name	70131
Authorization User Name	70131
Password	*****
Register Server Port	5060
Register Server Address	10.64.110.212
Domain Realm	avaya.com
Outbound proxy	
Register Expire	300
SIP Backup Type	None ▾
SIP Backup Server	
MWI Subscribe	Enable ▾
Subscribe Expire	300

Second Server: *Not configured* Backup Server: *Not configured*

Second Register Server

Use Service	Disable ▾
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In the **Protocol Control** section, provide the following values.

- **Local Port:** Enter *5060*
- **DTMF:** Select the *RFC2833* option
- **SIP Transport:** Select *UDP* from the dropdown menu

Use default values for the remaining fields. Click **Apply** button to save the changes.

**Cetis** 

**SYSTEM SUMMARY**  
Model: CC2  
WAN IP: 10.64.10.202  
Phone Number1: 70131  
Phone Number2:  
Firmware Version: CC2-4.0.0-066

**Protocol Control**

Local SIP Port	<input type="text" value="5060"/>	Any	<input type="text" value="Disable"/>
Local RTP Port Min	<input type="text" value="40000"/>	Max	<input type="text" value="40049"/>
Keep Alive Packet	<input type="radio"/> Off <input checked="" type="radio"/> On		
Keep Alives Period	<input type="text" value="60"/>		
DTMF	<input checked="" type="radio"/> RFC2833 <input type="radio"/> Inband <input type="radio"/> SIP Info		
DTMF SIP INFO Mode	<input type="text" value="Send */#"/>		
DNS Type	<input type="text" value="NAPTR/SRV"/>		
Jitter Buffer Max	<input type="text" value="150"/>		
Anonymous Call Rejection	<input checked="" type="radio"/> Off <input type="radio"/> On		
Session Switch	<input type="text" value="Disable"/>		
Session Time (Min=90s)	<input type="text" value="1800"/>		
PRACK	<input type="text" value="Disable"/>		
Support Update Method	<input type="text" value="Disable"/>		
Rport	<input type="text" value="Enable"/>		
SIP Transport	<input type="text" value="UDP"/>		
SIP URI	<input type="text" value="sip"/>		
SRTP	<input type="text" value="Disable"/>		

## 7.4. Configure Audio Settings

Select **Audio Settings** under the **VoIP Settings** section. In this page, a user can select and prioritize codec settings. The picture below shows codec *G.711u* is prioritized over *G.729* in the settings configured for E200IP.

**Cetis** 

**SYSTEM SUMMARY**  
Model: CC2  
WAN IP: 10.64.10.202  
Phone Number1: 70131  
Phone Number2:  
Firmware Version: CC2-4.0.0-066

Home • VoIP Settings • Audio Settings

### Audio Settings

Sound and Volume Control

Handset	<input type="text" value="5"/>	(1~7)
Speaker	<input type="text" value="5"/>	(1~7)
Ringer Tone	<input type="text" value="5"/>	(1~7)
Signal Standard	United States ▾	
Ringer	<input type="radio"/> Off <input checked="" type="radio"/> On	
Ringer Type	ringer 1 ▾	

Codecs Settings

Codec Priority 1	G.711u ▾	
Codec Priority 2	G.723.1 ▾	
Codec Priority 3	G.729 ▾	
Codec Priority 4	G.711a ▾	
Codec Priority 5	iLBC ▾	
Codec Priority 6	G.722 ▾	
Packet Data Size	20 ms ▾	
iLBC 15.2K	<input checked="" type="radio"/> Off <input type="radio"/> On	
G.723.1 5.3K	<input checked="" type="radio"/> Off <input type="radio"/> On	

Voice VAD/CNG

Voice VAD	<input checked="" type="radio"/> Off <input type="radio"/> On	
CNG	<input checked="" type="radio"/> Off <input type="radio"/> On	

Codec ID Settings

DTMF Payload(RFC2833)	<input type="text" value="101"/>	(95~127)
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## 7.5. Assign Feature Buttons

Select **Call Features** under the **VoIP Settings** section to optionally assign feature access to the memory buttons and the voicemail pilot number to the message button on Cetis E200IP. The following assignments were used for compliance testing.

- **Memory 1:** Select *Transfer* in the dropdown list to initiate transfers from the first programmable key
- **MWI number:** Input a voicemail pilot number to assign to the message button on Cetis E200IP (e.g., 59992)

Click **Apply** (not shown) to save changes.

**Cetis** 

**SYSTEM SUMMARY**  
Model: CC2  
WAN IP: 10.64.10.202  
Phone Number1: 70131  
Phone Number2:  
Firmware Version: CC2-4.0.0-066

Home • VoIP Settings • Call Features

### Call Features

Programmable Keys & MWI Number

Memory 1:	Transfer	
Memory 2:	Memory	
Memory 3:	Memory	
Memory 4:	Memory	
Memory 5:	Memory	
Memory 6:	Memory	
Memory 7:	Memory	
Memory 8:	Memory	
Memory 9:	Memory	
Memory 10:	Memory	
MWI Number:	59992	
Park Mode:	Default	
Hold Key Active:		
Hold Key Idle:		

## 8. Verification Steps

The proper configuration of Cetus E200IP with Session Manager and Communication Manager is verified by the following steps.

### 8.1. Verify Session Manager Status

Verify Cetus E200IP has successfully registered with Session Manager. In System Manager, Navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations**. Verify Cetus E200IP (here 70131) is registered with Session Manager by noting that 70131 is listed as a registered user.

**AVAYA** Aura System Manager 10.1

Users | Elements | Services | Widgets | Shortcuts | Search | admin

Home | Session Manager

### User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

View: Default | Export | Force Unregister | AST Device Notifications: Reboot | Reload | Failback | As of 4:10 PM | Advanced Search

7 Items | Show: All | Filter: Enable

	Details	Address	First Name	Last Name	Actual Location	IP Address	Policy	Shared Control	Simult. Devices	AST Device	Registered					
											Prim	Sec	3rd	4th	Surv	Visiting
<input type="checkbox"/>	<a href="#">Show</a>	70131@avaya.com	E200IP	CETIS	---	10.64.10.202	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>				
<input type="checkbox"/>	<a href="#">Show</a>	70103@avaya.com	SIP	User 3	DevConnect	10.64.10.203	fixed	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>				
<input type="checkbox"/>	<a href="#">Show</a>	---	SIP	User 5	---	---	fixed	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	---	SIP	User 4	---	---	fixed	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	---	SIP	User 6	---	---	fixed	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	---	SIP	User 1	---	---	fixed	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	---	E200IP_2	CETIS	---	---	fixed	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Select : All, None

## 8.2. Verify Cetus E200IP Status

Open the Cetus E200IP Web Administration interface. Select **VOIP Settings** in the left pane to display the **VoIP Summary** page. Verify that **First Register** is set to *Registered*.

The screenshot shows the Cetus E200IP Web Administration interface. The top left features the Cetus logo. The top right corner displays a **SYSTEM SUMMARY** box with the following information: Model: CC2, WAN IP: 10.64.10.202, Phone Number1: 70131, Phone Number2: (blank), and Firmware Version: CC2-4.0.0-066. A navigation menu on the left lists various settings categories, with **VoIP Settings** selected. The main content area shows the **VoIP Summary** page, which is divided into three sections: **First Register: Registered**, **Second Register: Not configured**, and **Other**. The **First Register** section lists: User Name: 70131, Register Server: 10.64.110.212, Register Server Port: 5060, SIP Backup Register Status: Not configured, SIP Backup Server: (blank), and SIP Backup Type: None. The **Second Register** section lists: User Name: (blank), Register Server: (blank), Register Server Port: 5060, SIP Backup Register Status: Not configured, SIP Backup Server: (blank), and SIP Backup Type: None. The **Other** section lists: NAT Traversal(STUN): Disabled and STUN Sever Address: (blank).

SYSTEM SUMMARY	
Model:	CC2
WAN IP:	10.64.10.202
Phone Number1:	70131
Phone Number2:	
Firmware Version:	CC2-4.0.0-066

Home • VoIP Settings

### VoIP Summary

**First Register: Registered**

User Name:	70131	Domain Realm:	avaya.com
Register Server:	10.64.110.212	Outbound Proxy:	
Register Server Port:	5060		
SIP Backup Register Status:	Not configured		
SIP Backup Server:			
SIP Backup Type:	None		

**Second Register: Not configured**

User Name:		Domain Realm:	
Register Server:		Outbound Proxy:	
Register Server Port:	5060		
SIP Backup Register Status:	Not configured		
SIP Backup Server:			
SIP Backup Type:	None		

**Other**

NAT Traversal(STUN):	Disabled	STUN Sever Address:	
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## 8.3. Call Verification

Make incoming and outgoing calls from Cetus E200IP to Avaya SIP and H.323 endpoints and verify two-way audio.

## 9. Conclusion

These Application Notes describe the configuration steps required to integrate Cetus E200IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Cetus E200IP registers to Avaya Aura® Session Manager as a third-party SIP endpoint. Calls were established with Avaya H.323 / SIP deskphones and the PSTN. In addition, basic telephony features were verified. All feature and serviceability test cases were completed successfully.

## 10. Additional References

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

Avaya product documentation is available at <https://support.avaya.com>.

[1] *Administering Avaya Aura® Communication Manager*, Release 10.1.x, Issue 5, March 2023.

[2] *Administering Avaya Aura® System Manager*, Release 10.1.x, Issue 8, February 2023.

[3] *Administering Avaya Aura® Session Manager*, Release 10.1.x, Issue 5, February 2023.

Contact Cetus support at <https://support.cetus.com> for E200IP product documentation.

[4] *Cetus SIP Quick Reference*, February 2023.

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