DevConnect Program

Application Notes for Cetis 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 Cetis E200IP Corded SIP 2-Line Telephone to integrate with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. Cetis 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 Cetis E200IP Corded SIP 2-Line Telephone to integrate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Cetis E200IP Corded SIP 2-Line Telephone is designed for the hospitality industry. In the compliance test, Cetis 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 Cetis 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 Cetis E200IP did not include use of any specific encryption features as requested by Cetis.

2.1. Interoperability Compliance Testing

The following areas were evaluated in the interoperability compliance test:

- Registration of Cetis E200IP to Session Manager.
- Calls between Cetis E200IP and Avaya SIP and H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetis 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 Cetis E200IP Telephone, contact Cetis Support via phone, email, or website.

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

3. Reference Configuration

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

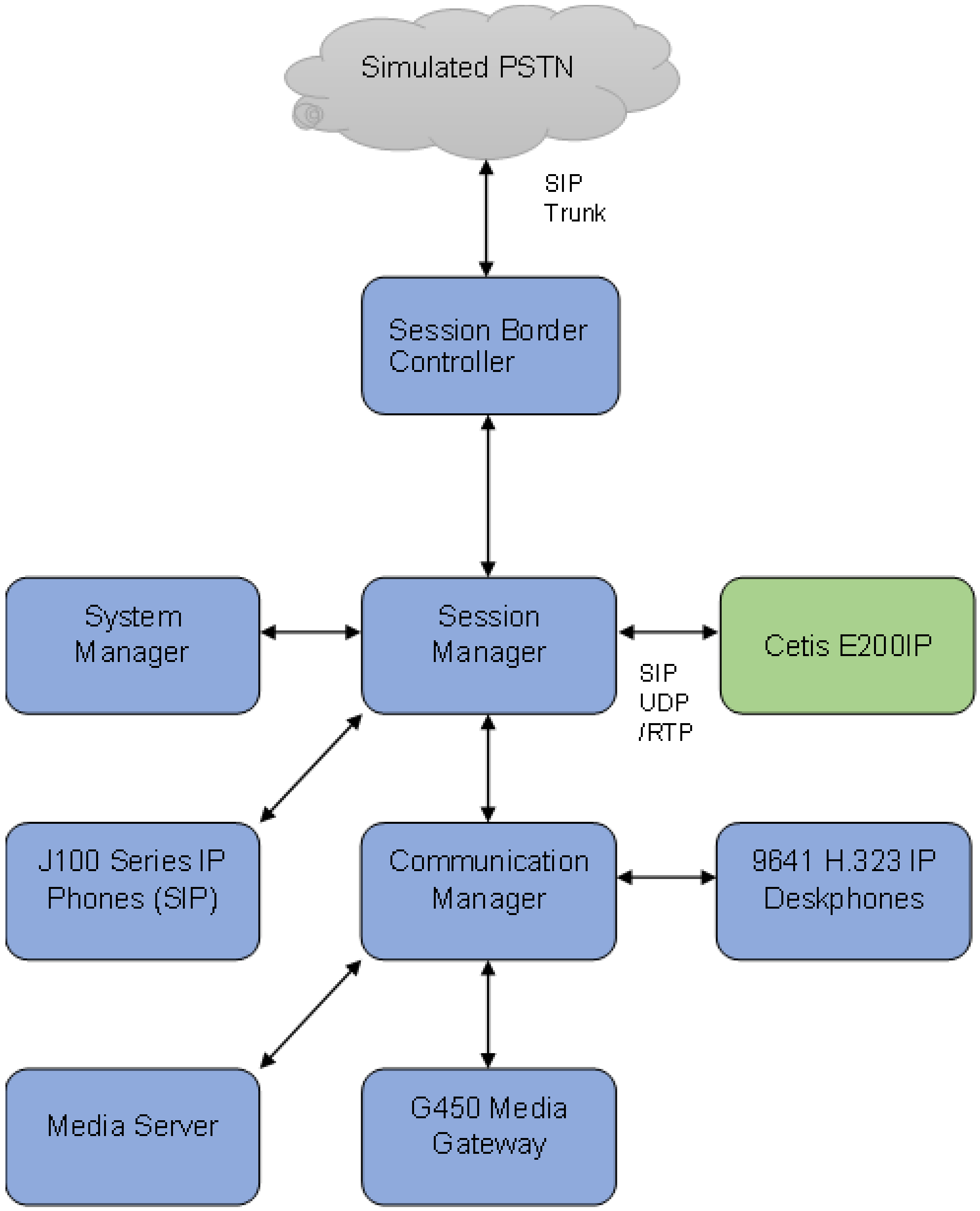


Figure 1: Avaya Test Configuration for Cetis E200IP Phones

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 Cetis 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 Cetis 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. Cetis 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 SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: 10-srtp-aescm256-hmac80
 3: none
 4:
 5:
```

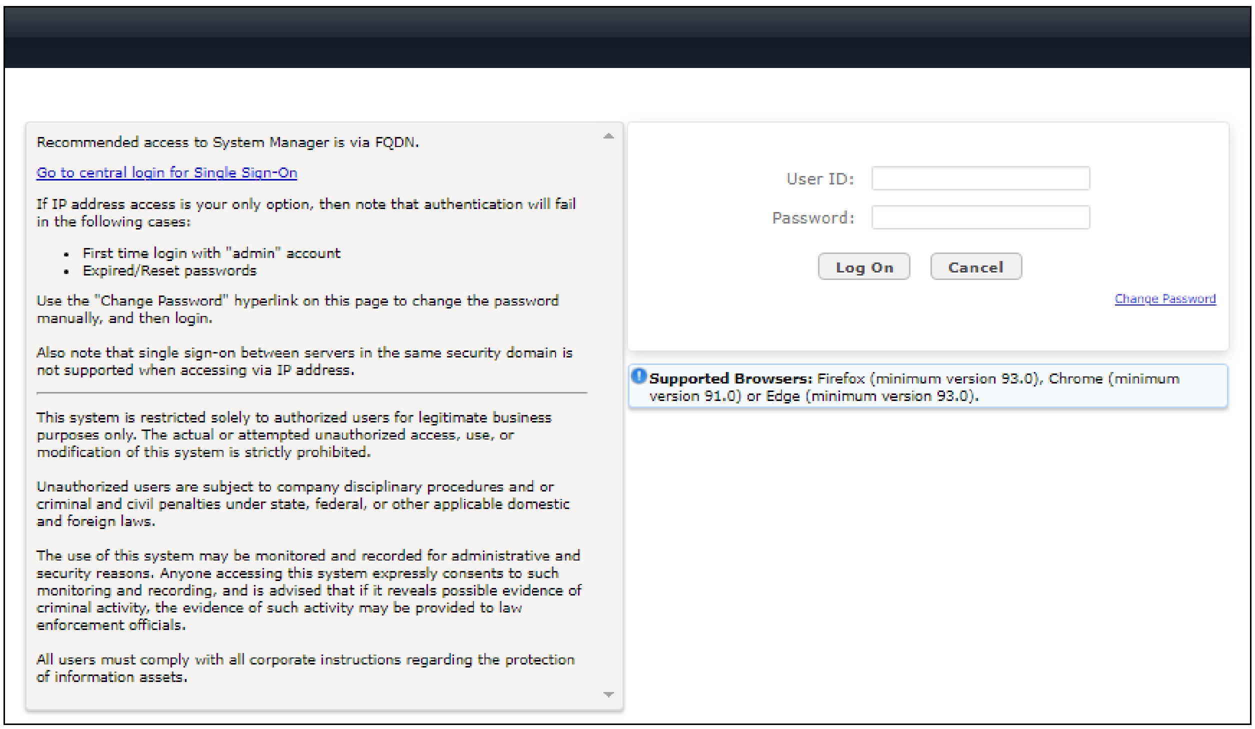
6. Configure Avaya Aura® Session Manager

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

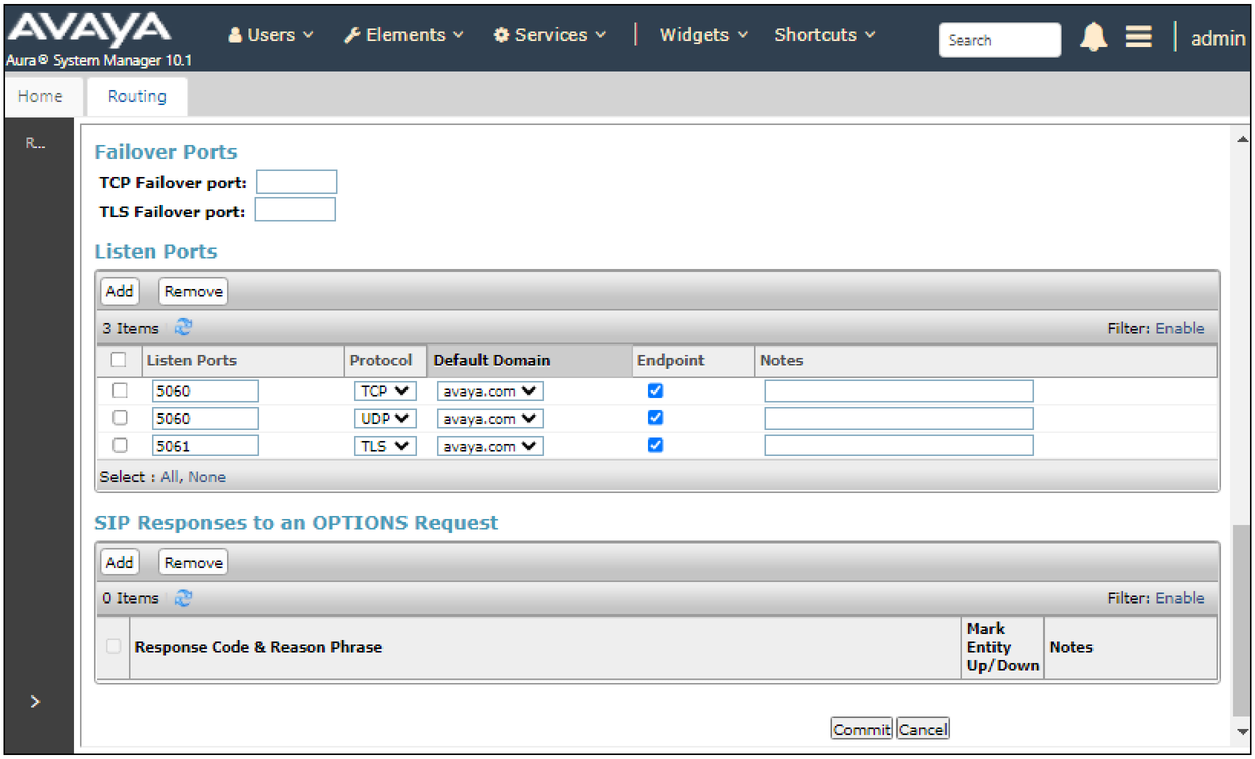
6.1. Launch System Manager

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.



6.2. Verify Session Manager Listening Ports

Each Session Manager must be configured so that Cetis 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 Cetis E200IP was tested during the compliance test.



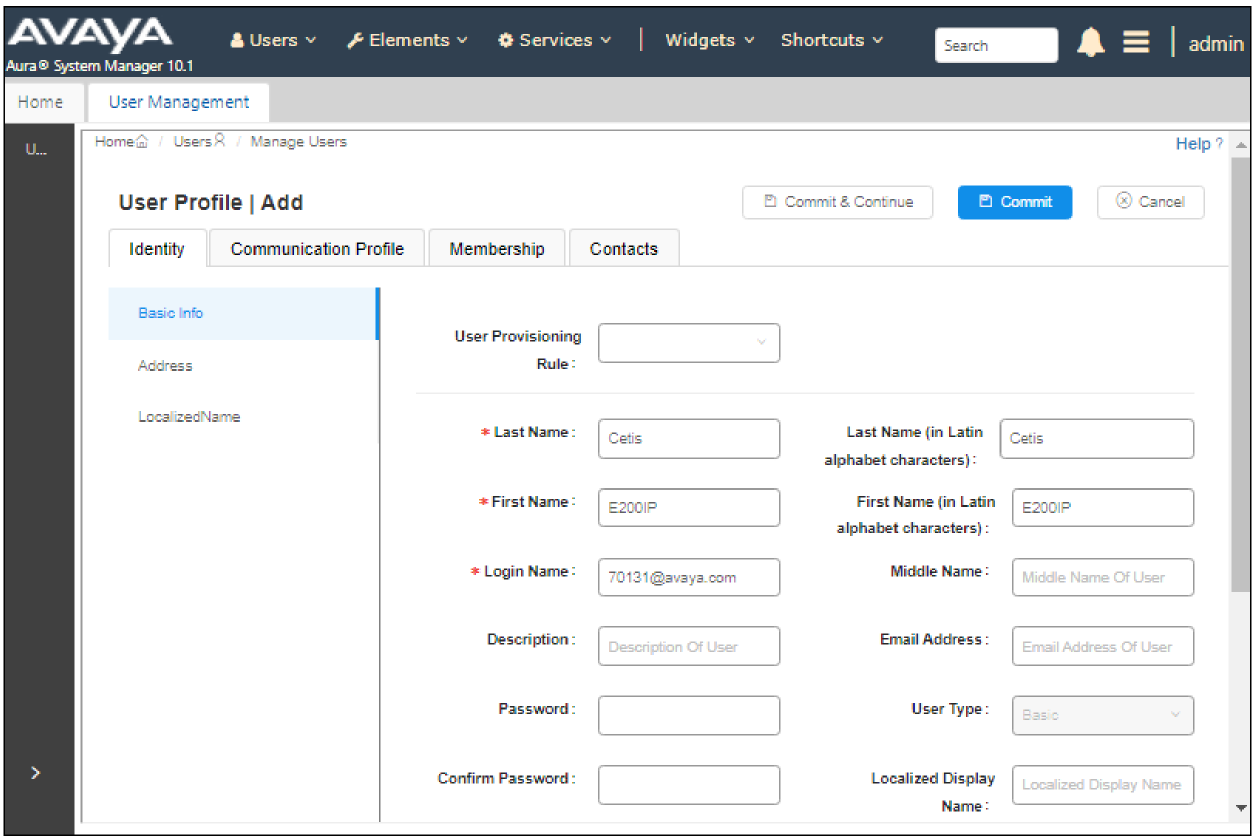
6.3. Administer SIP Users

A SIP user must be created for Cetis 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., *Cetis*)
- **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*)

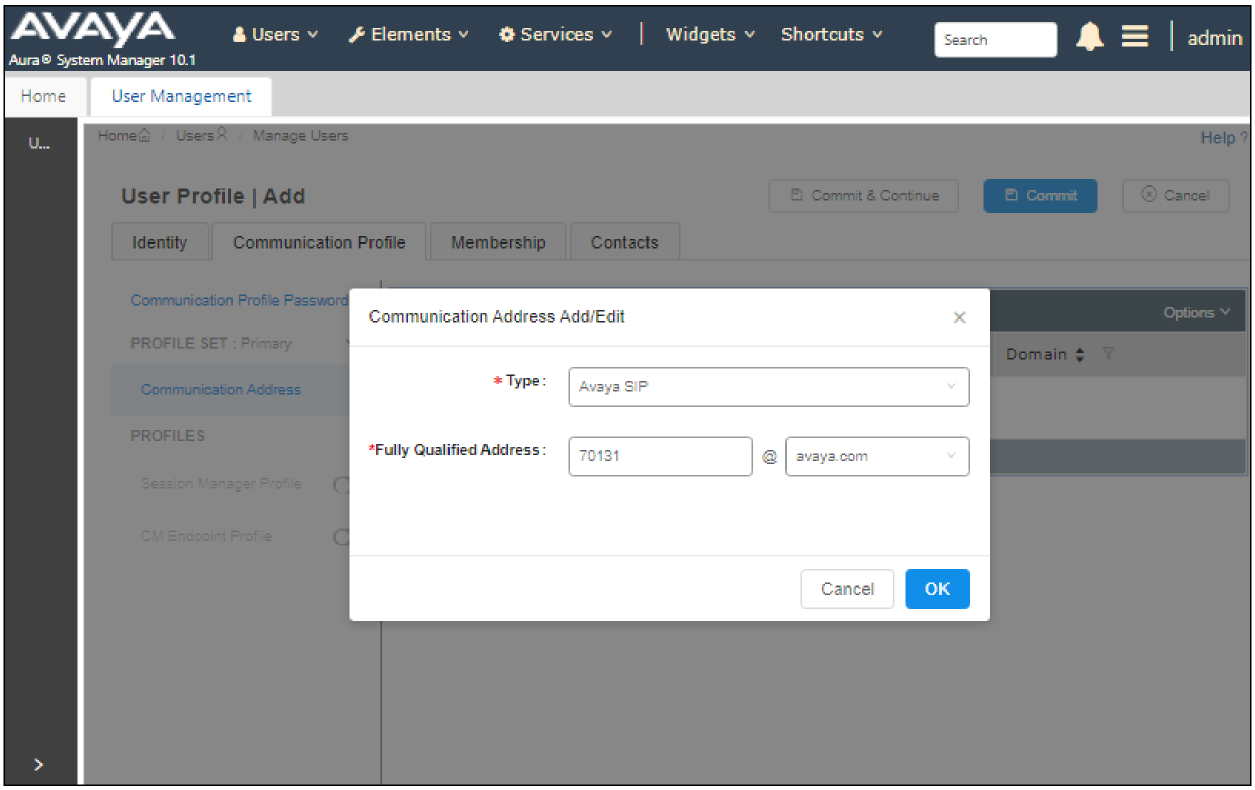


6.3.2. Communication Address

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

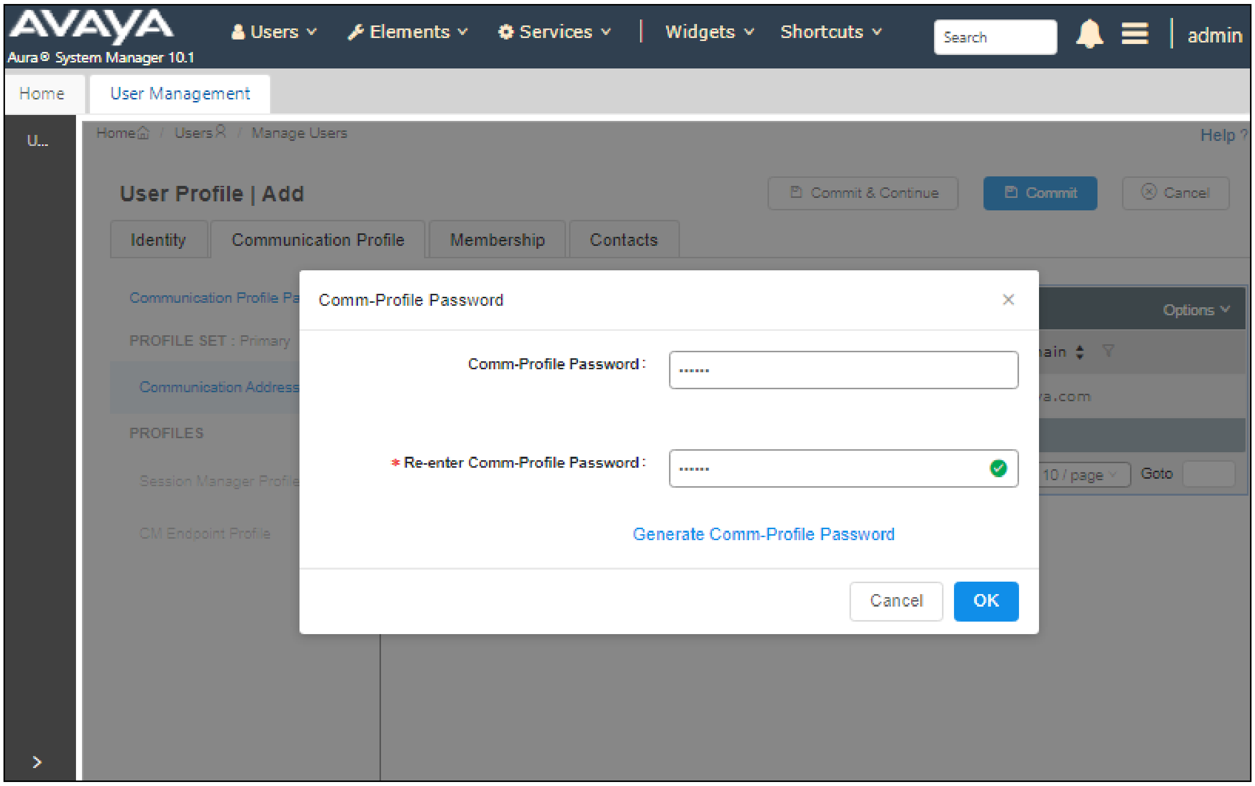
Enter the following attributes for the **Communication Address**:

- **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*)



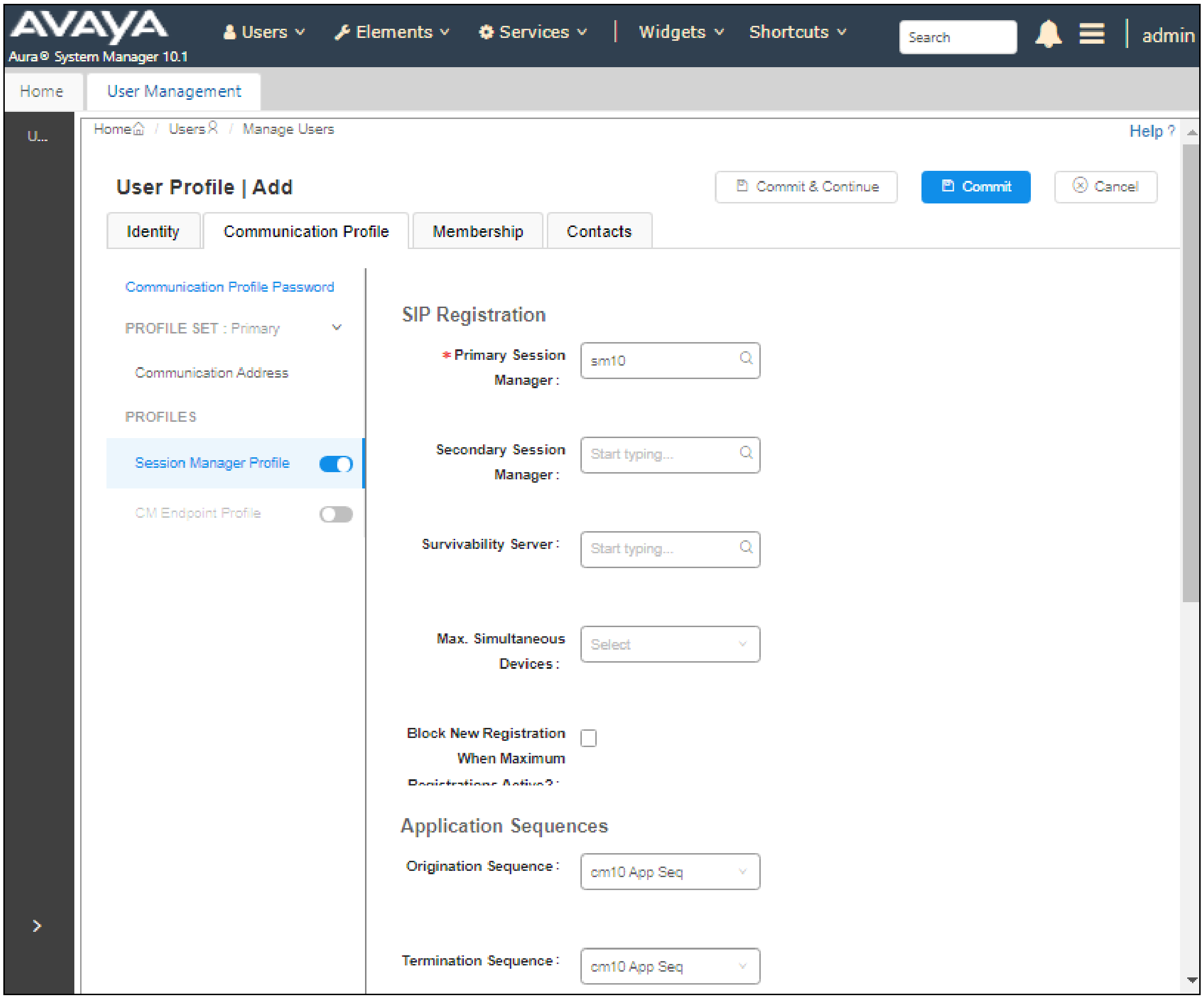
6.3.3. Communication Profile Password

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.

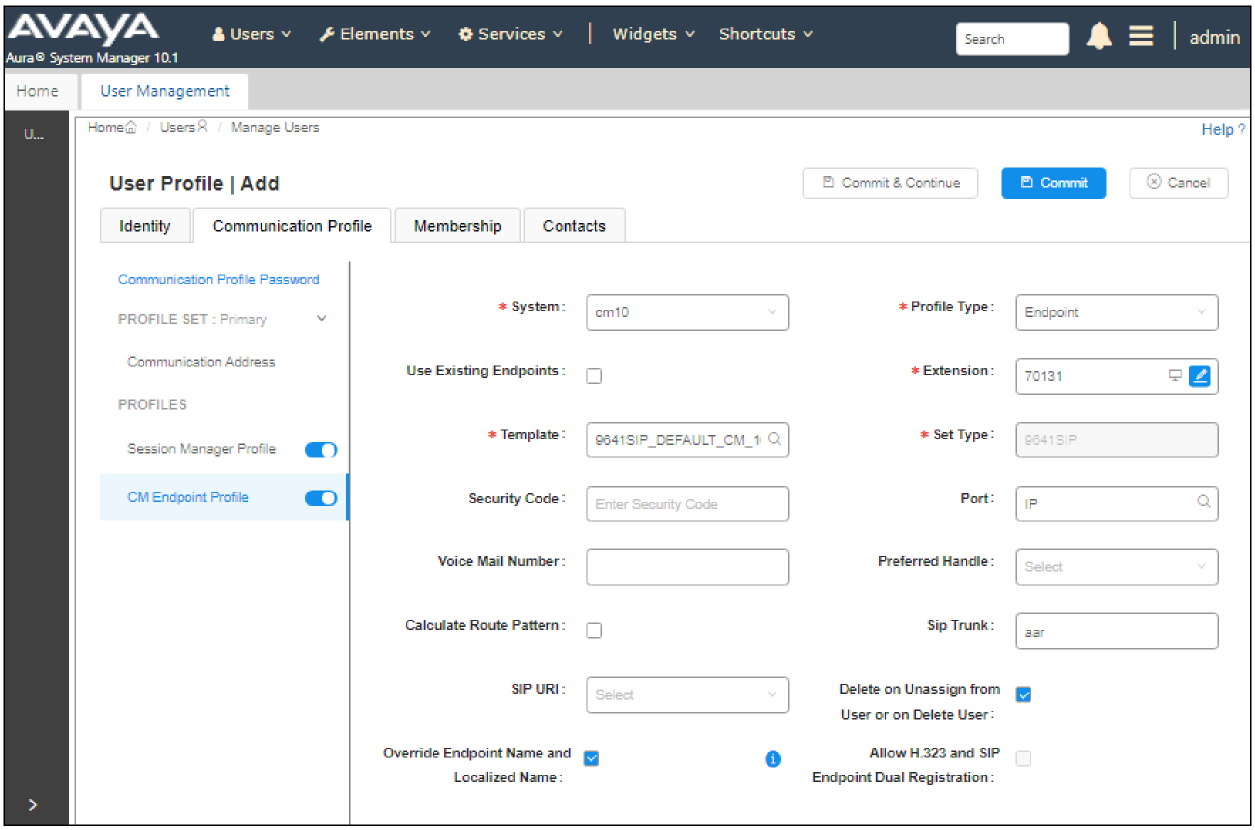


6.3.5. CM Endpoint Profile

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**.



7. Configure Cetis E200IP Corded 2-Line Telephone

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

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

7.1. Configure IP Address

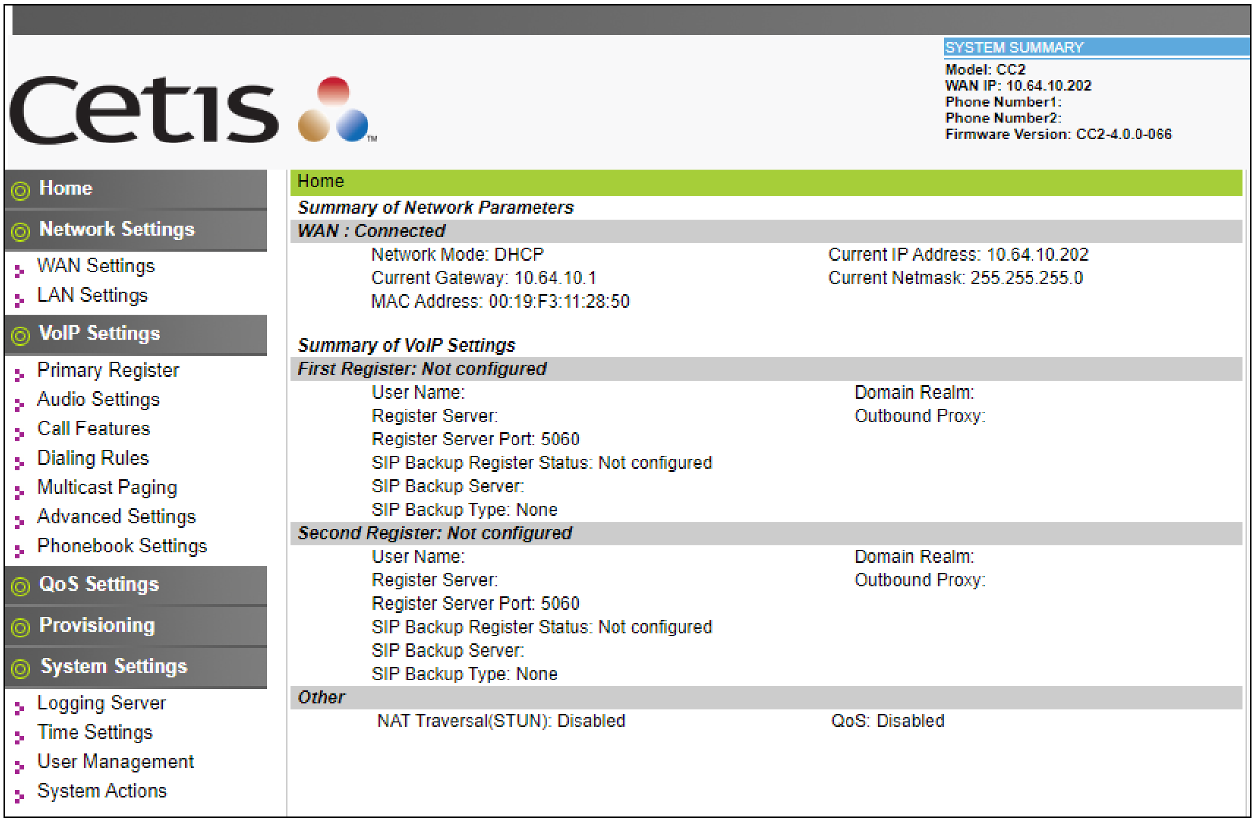
Cetis E200IP is configured for DHCP as a factory default. The following steps provide network connectivity and determine the phone IP address for use in launching the web administration Interface as detailed in **Section 7.2**:

- Connect the WAN port of Cetis 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 Cetis E200IP Web Administration Interface

Access the Cetis 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.**

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

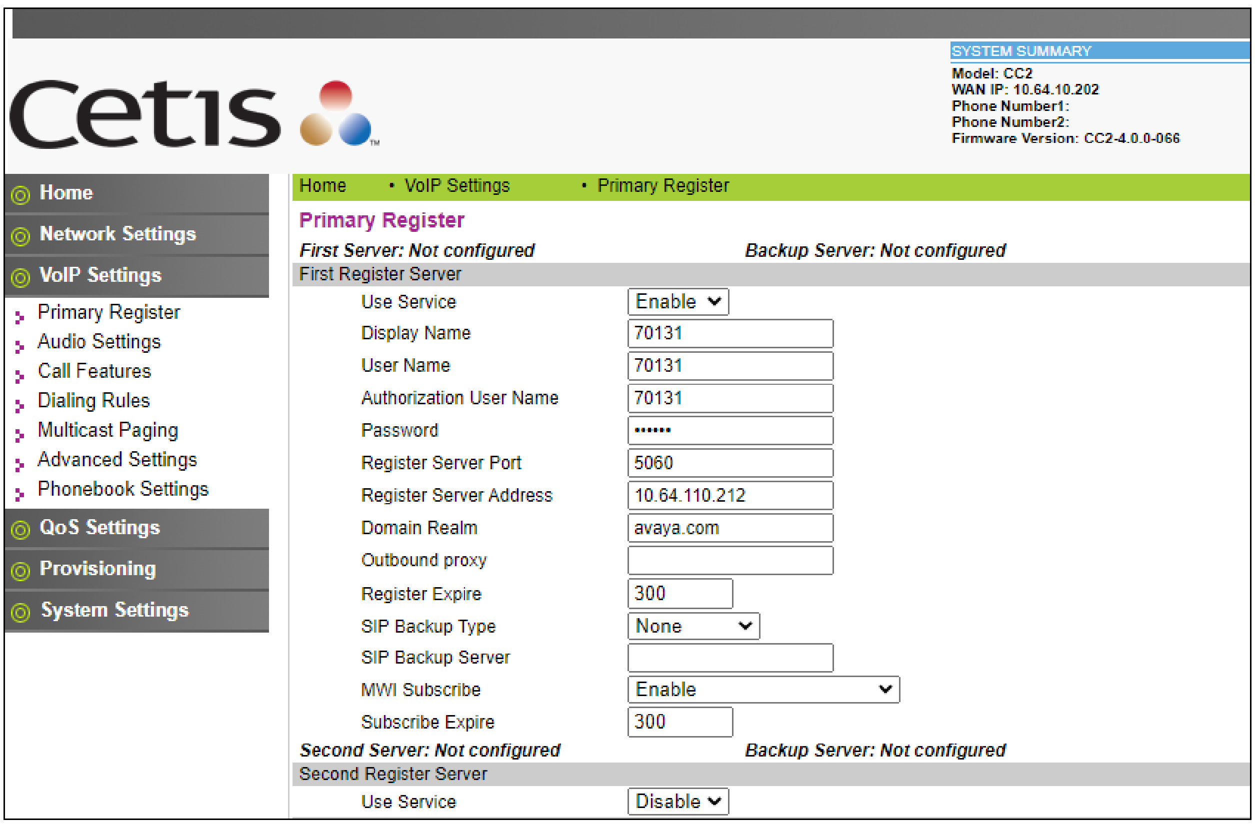


7.3. Configure SIP Account

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

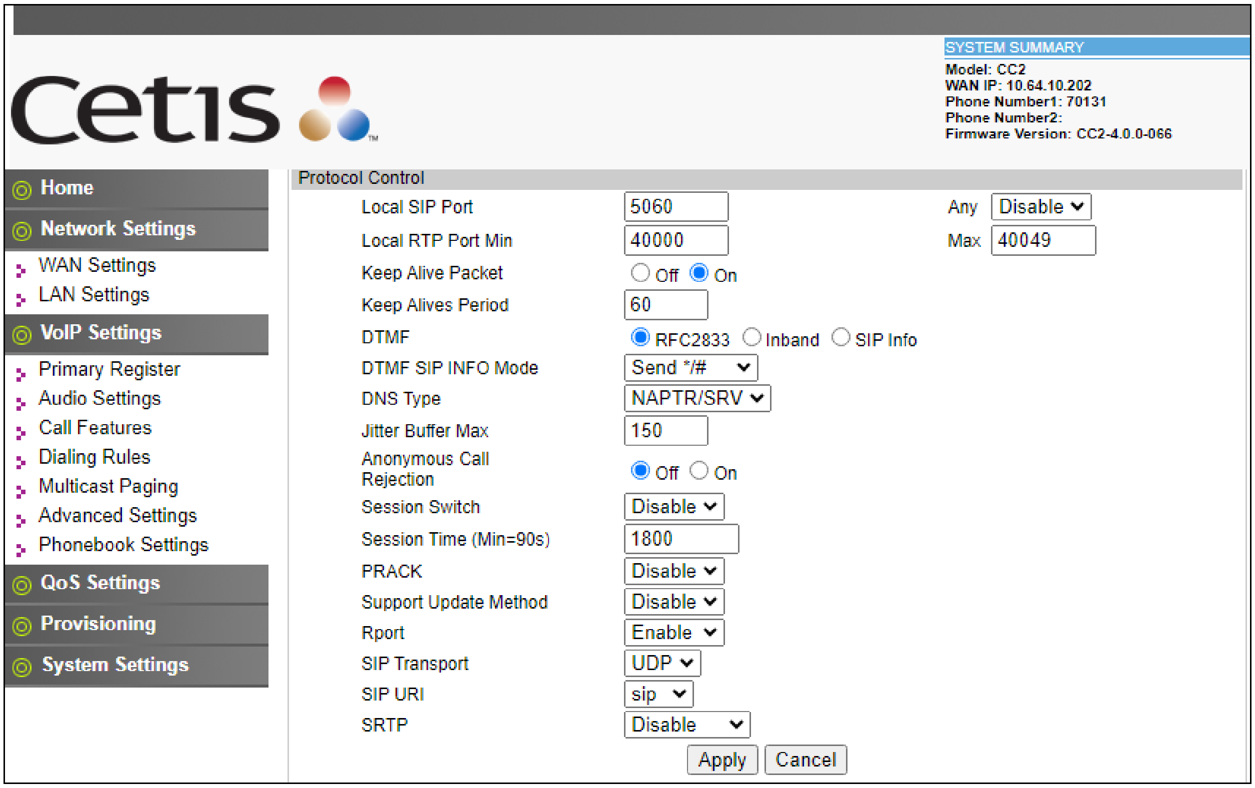
Use default values for the remaining fields.



In the **Protocol Control** section, provide the following values.

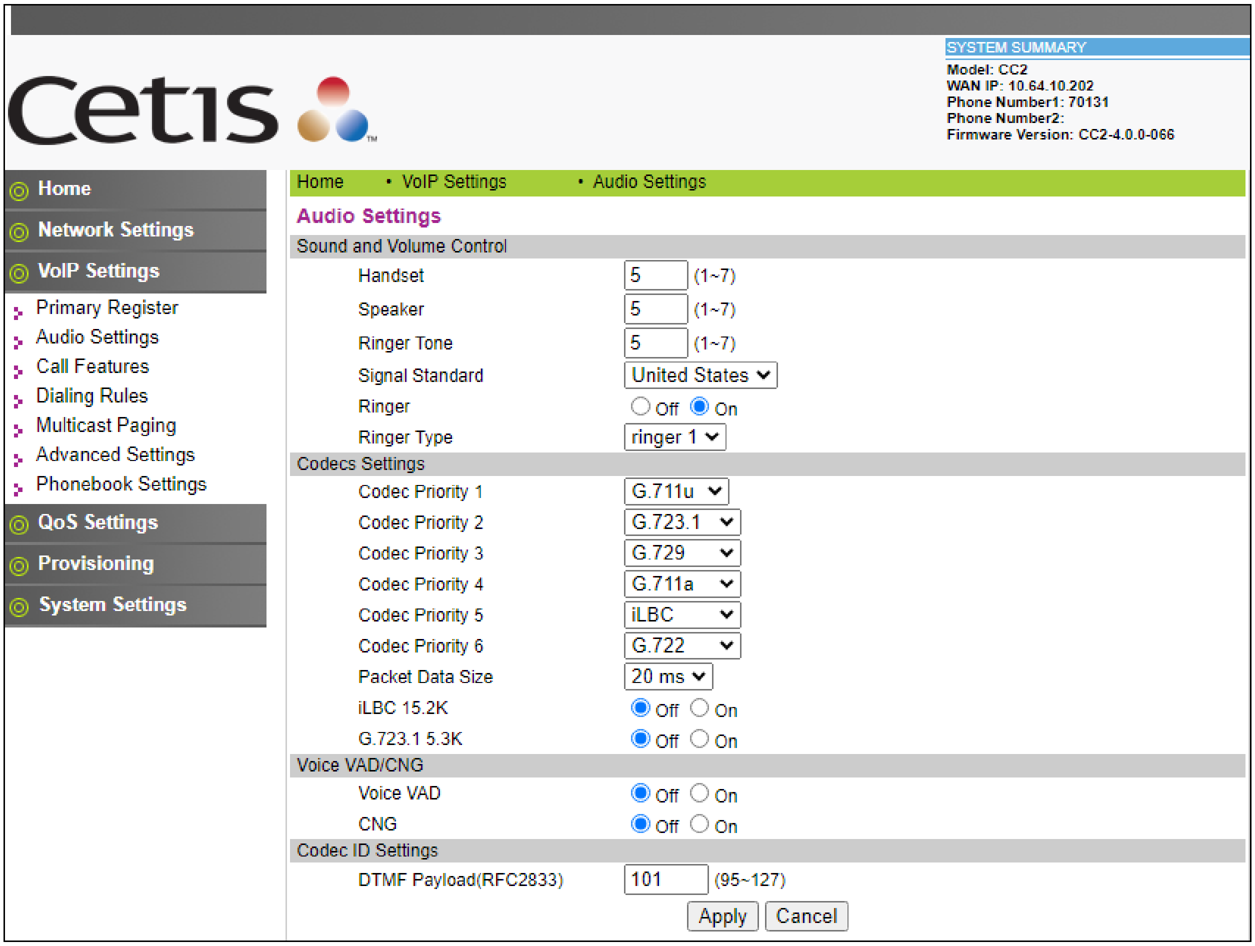
- **Local Port:** Enter *5060*
- **DTMF:** Select the *RFC2833* option
- **SIP Transport:** Select *UDP* from the dropdown menu

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



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.

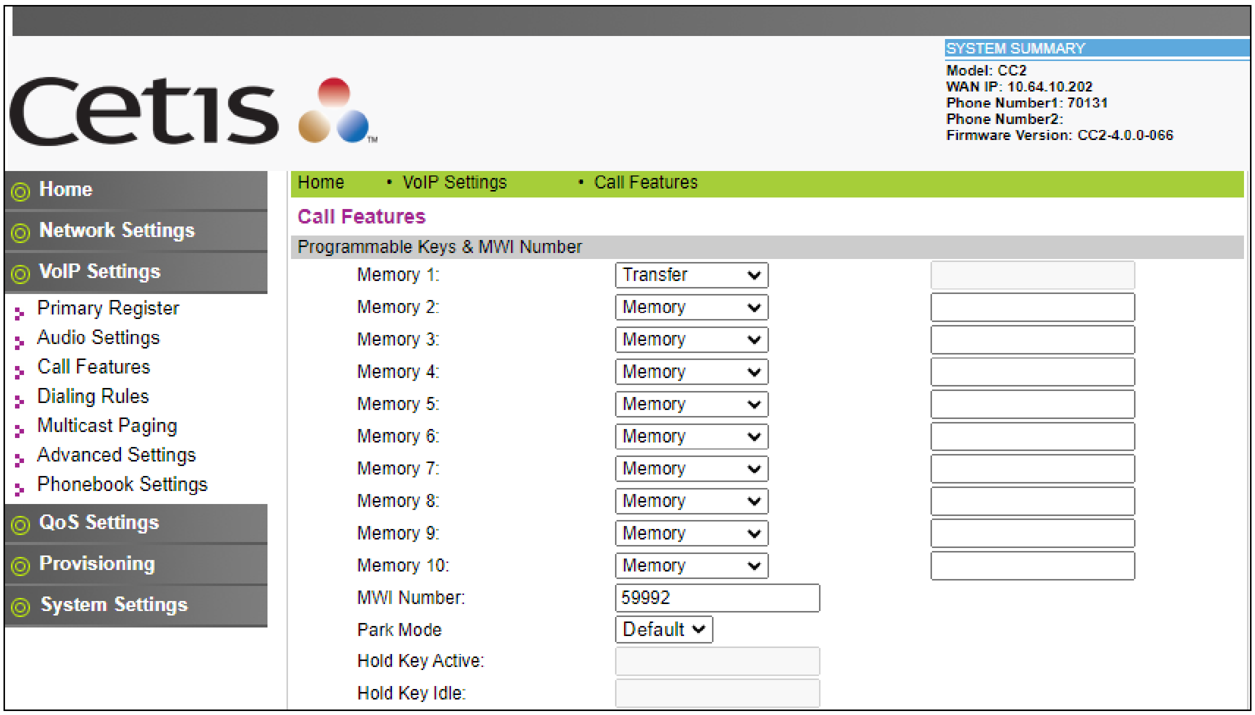


7.5. Assign Feature Buttons

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.

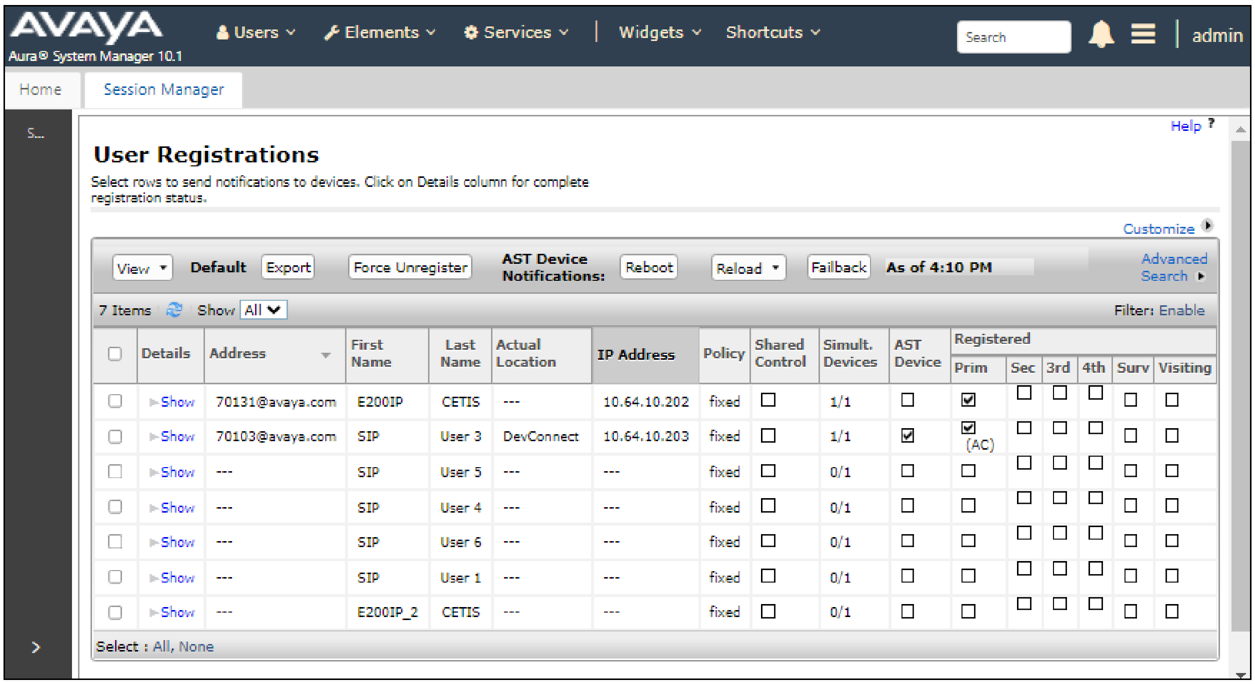


8. Verification Steps

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

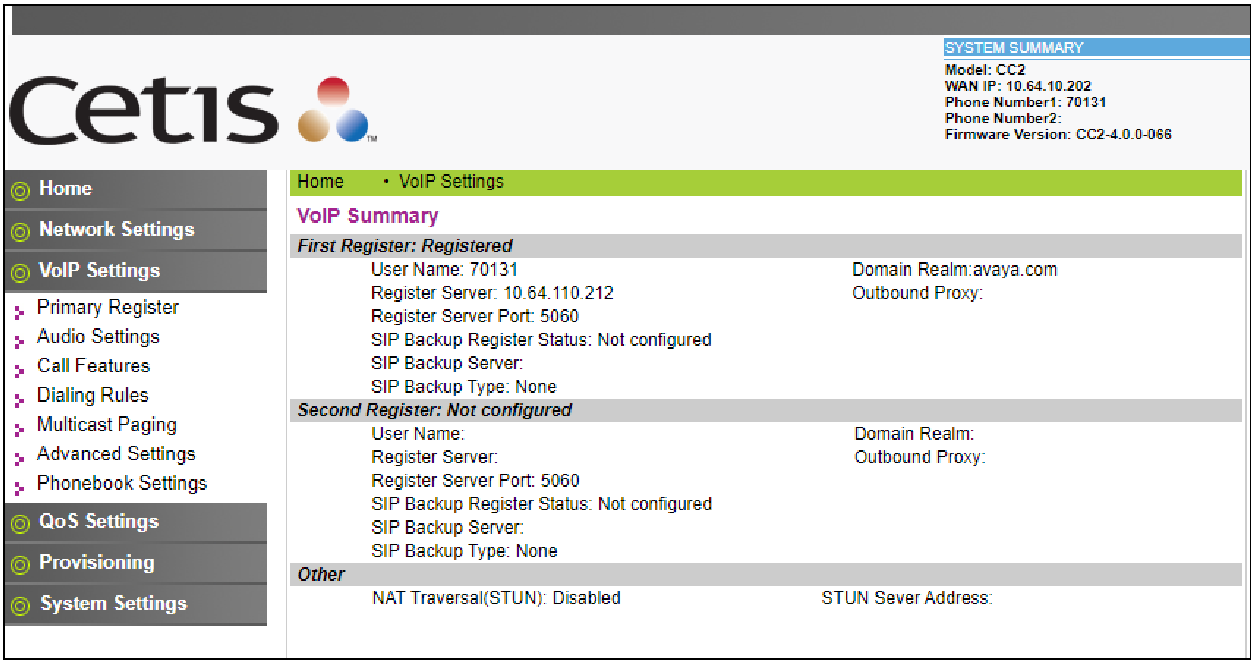
8.1. Verify Session Manager Status

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



8.2. Verify Cetis E200IP Status

Open the Cetis 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*.



8.3. Call Verification

Make incoming and outgoing calls from Cetis 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 Cetis E200IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Cetis 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 Cetis support at https://support.cetis.com for E200IP product documentation.
[4] *Cetis SIP Quick Reference*, February 2023.

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.