Avaya Solution & Interoperability Test Lab

Application Notes for Cetis e-Series E203IP SIP cordless Telephones with Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Cetis e-Series E203IP SIP cordless Telephones with Avaya Aura® Session Manager. The Cetis e-Series E203IP 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 E-Series E203IP SIP cordless Telephones (hereafter referred to as Cetis E203IP SIP Telephones) with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The Cetis E203IP SIP Telephones were designed for the hospitality industry. In the compliance test, Cetis SIP telephones registered with Avaya Aura® Session Manager and used telephony features from Commutation 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 E203IP 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 E203IP 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 E203IP 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.
- Transport protocol TCP and UDP.
- 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:

- When the Cetis E203IP SIP Telephone registers to Session Manager using TCP transport having shuffling/direct media enabled, an incoming call from an Avaya H.323 endpoint was dropped after 30 seconds. This was due to Communication Manager not receiving an ACK response from the Cetis SIP Telephone for the Re-INVITE message Communication Manager sent to establish shuffling/direct media. The issue does not happen with an Avaya SIP endpoint. If the Cetis E203IP SIP Telephone is using UDP to register to Session Manager the problem does not occur. Also if shuffling/direct media is disabled the problem does not occur.

2.3. Support

For technical support on the Cetis E203IP 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/sipsupport/

3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Cetis E203IP SIP Telephones with Avaya Aura® Session Manager. The Cetis E203 SIP telephones registered with Avaya Aura® Session Manager via SIP.

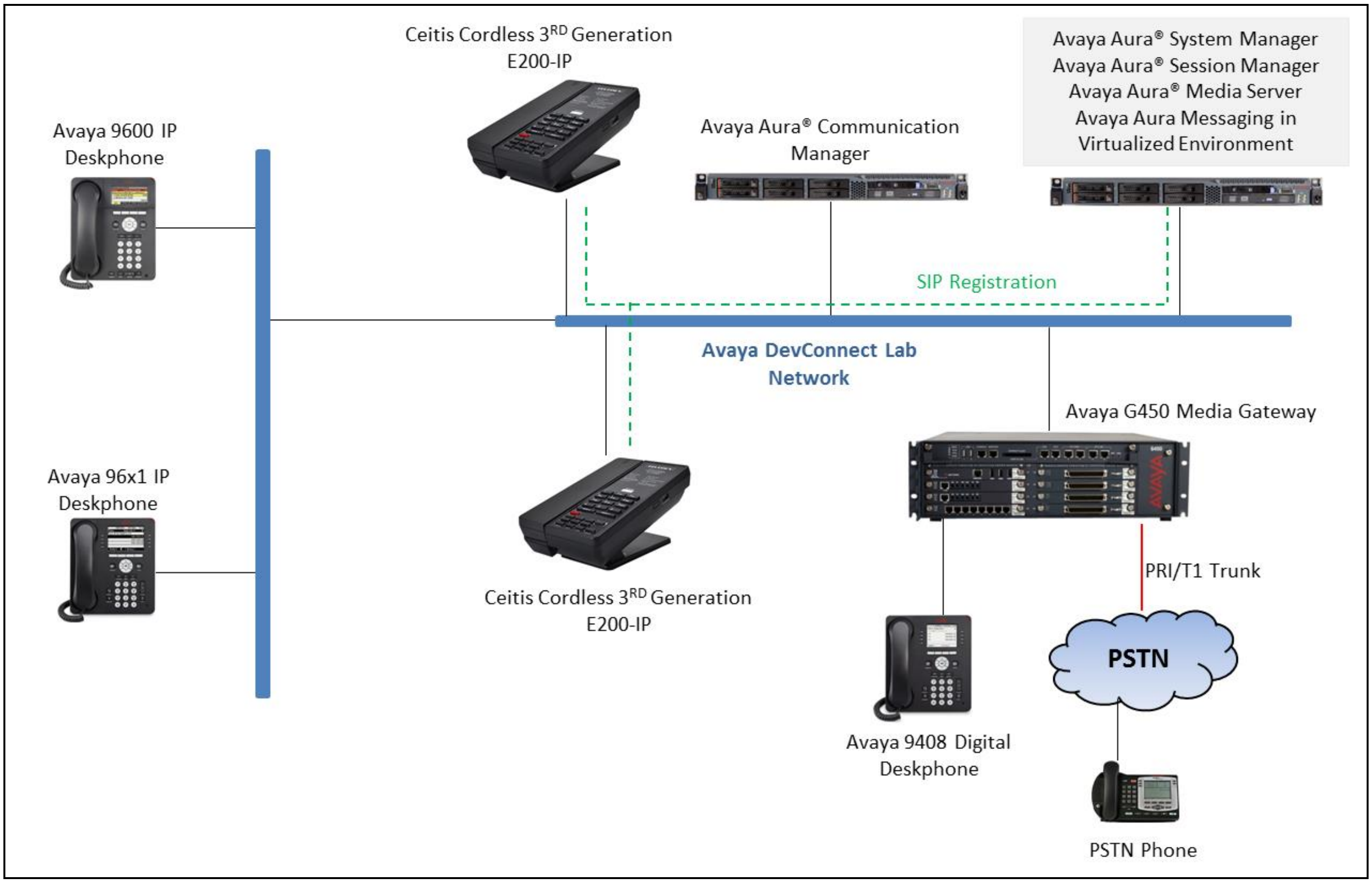


Figure 1: Cetis E203IP Telephones with Session Manager

4. Equipment and Software Validated

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

Equipment/Software		Release/Version
Avaya Aura® Session Manager on Virtual Environment		7.0.1.2.701230
Avaya Aura® System Manager on Virtual Environment		7.0.1.2.701230
Avaya Aura® Communication Manager on Virtual Environment		7.0 (R017x.00.0.441.0)
Avaya Aura Messaging on Virtual Environment		6.3
Avaya G450 Media Gateway		37.39.0
Avaya 96x0 and 96x1 Series IP Deskphones		
	9620 (H.323) 9621G (H.323)	3.25 6.6.41
Avaya 96x0 and 96x1 Series SIP Deskphones		
	9611G 9650	7.0.2 2.6.9
Cetis E203IP		CD2-3.0.0-029

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of Communication Manager for this solution. It is implied a working system is already in place, including SIP trunks to a Session Manager. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration described in this section can be summarized as follows:

- Verify System Capacity
- Define the Dial Plan

Note: Any settings not in **Bold** in the following screen shots may be left as default

5.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On Page 1, verify that the **Maximum Off PBX Telephones** allowed in the system is sufficient. One OPS station is required per SIP device.

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

     G3 Version: V17                                Software Package: Enterprise
       Location: 2                                   System ID (SID): 1
       Platform: 28                                  Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 48000 118
                                      Maximum Stations: 36000 24
                              Maximum XMOBILE Stations: 36000 0
                    Maximum Off-PBX Telephones - EC500: 41000 1
                    Maximum Off-PBX Telephones -   OPS: 41000 11
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   1
```

On Page 2 of the **System Parameters Customer Options** form, verify that the number of Maximum Administered SIP Trunks supported by the system is sufficient.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 36000 0
                   Maximum Video Capable IP Softphones: 18000 7
                       Maximum Administered SIP Trunks: 12000 48
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Configure Dialing Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are **4** digits long and begin with **3**.

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 3

    Dialed    Total  Call      Dialed    Total  Call      Dialed    Total  Call
    String    Length Type      String    Length Type      String    Length Type
   3            4    ext      56           5   udp
   13           5    aar      8            1   fac
   14           5    aar      9            1   fac
   20           4    aar      *            3   dac
   23           5    aar      #            3   dac
```

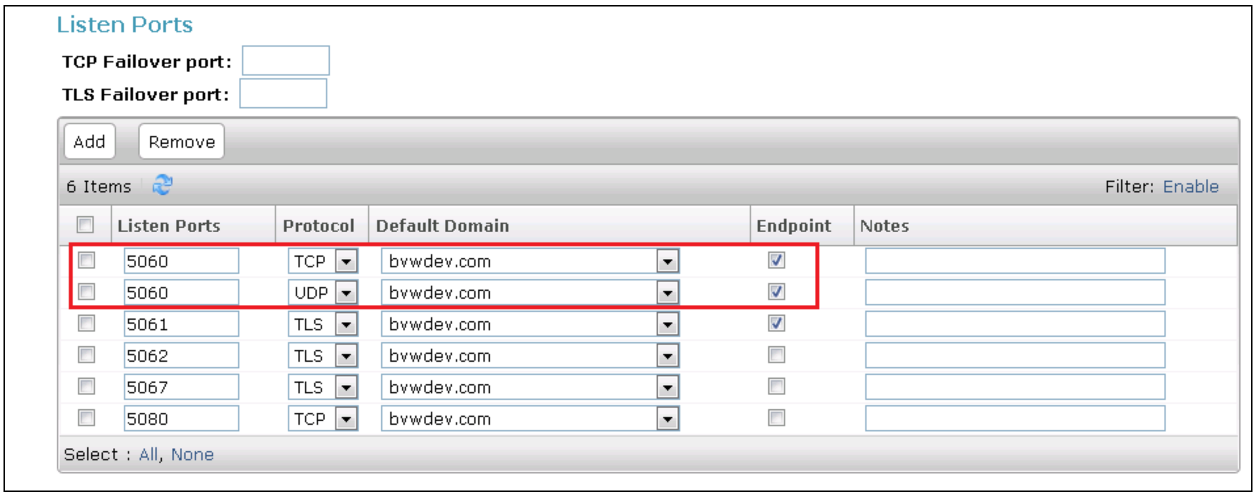
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. 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.

6.1. Configure Listen Port for SIP endpoint

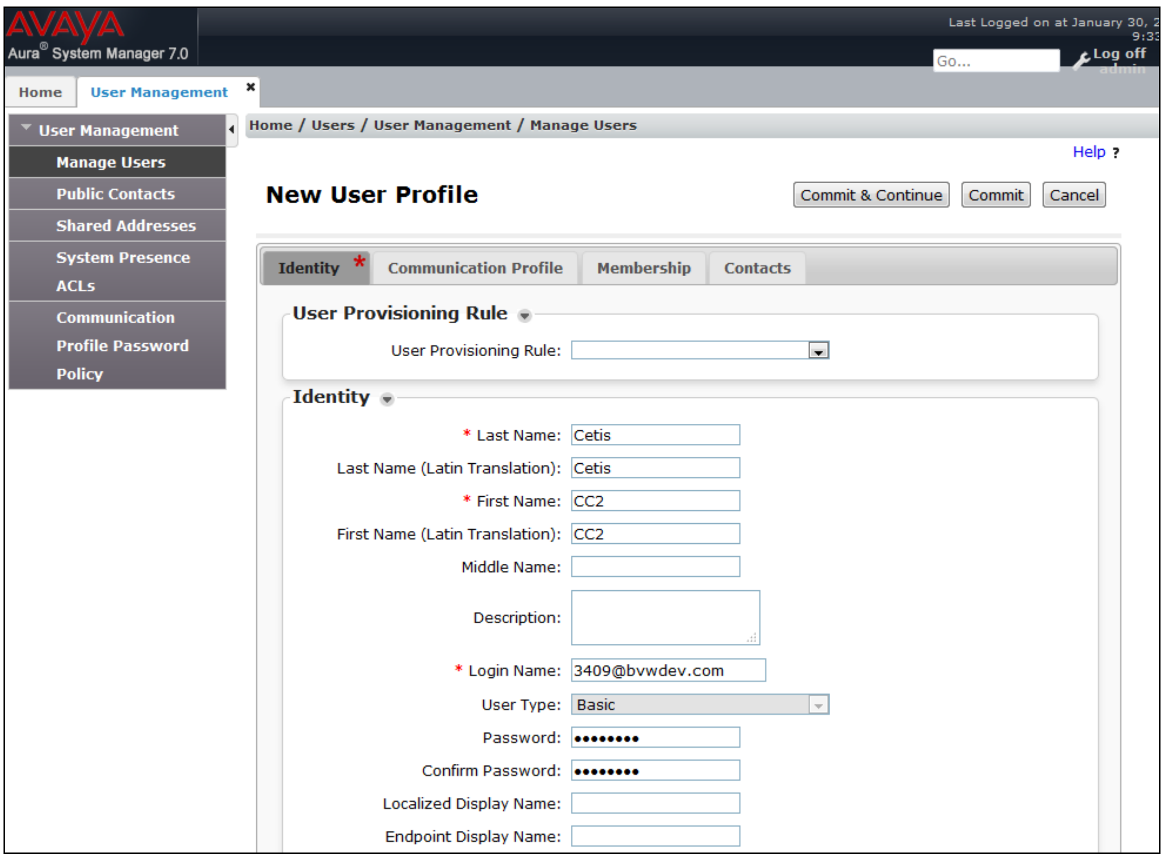
Each Session Manager Entity must be configured with the listen ports so that the Cetis SIP telephones can register to it using UDP/TCP. From the web interface click **Routing → SIP Entities** (not shown) and select the Session Manager entity used for registration. Make sure that TCP and UDP entries are present. The TCP and UDP entries are highlighted below.



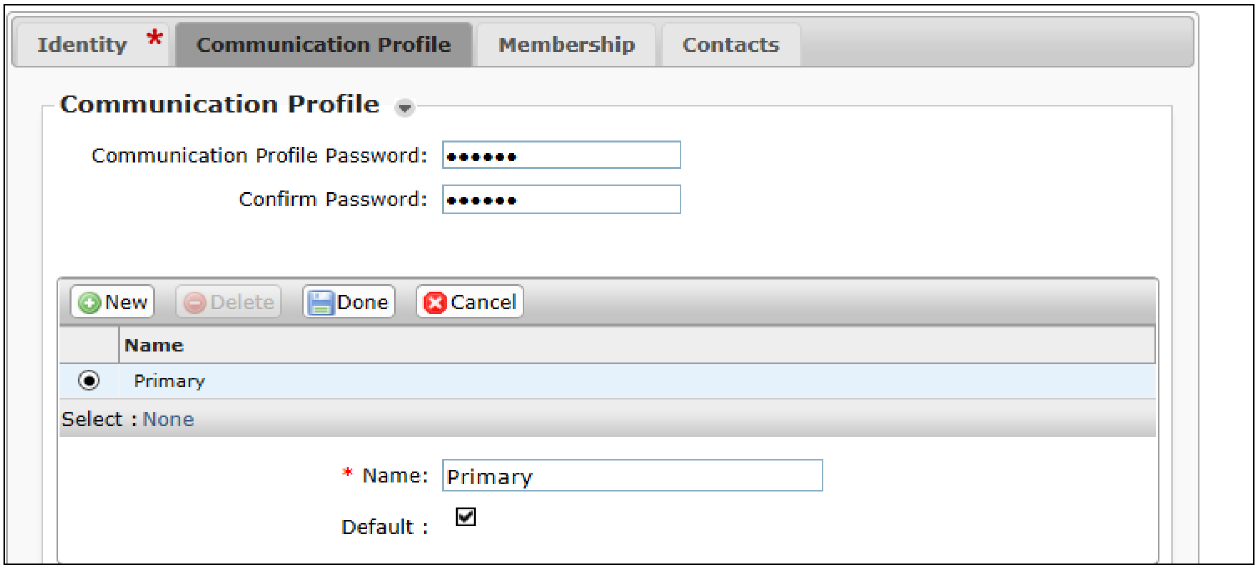
6.2. Configure User

To add new SIP users, Navigate to **Home → Users → User Management → Manage Users**. Click **New(not shown)** and provide the following information:

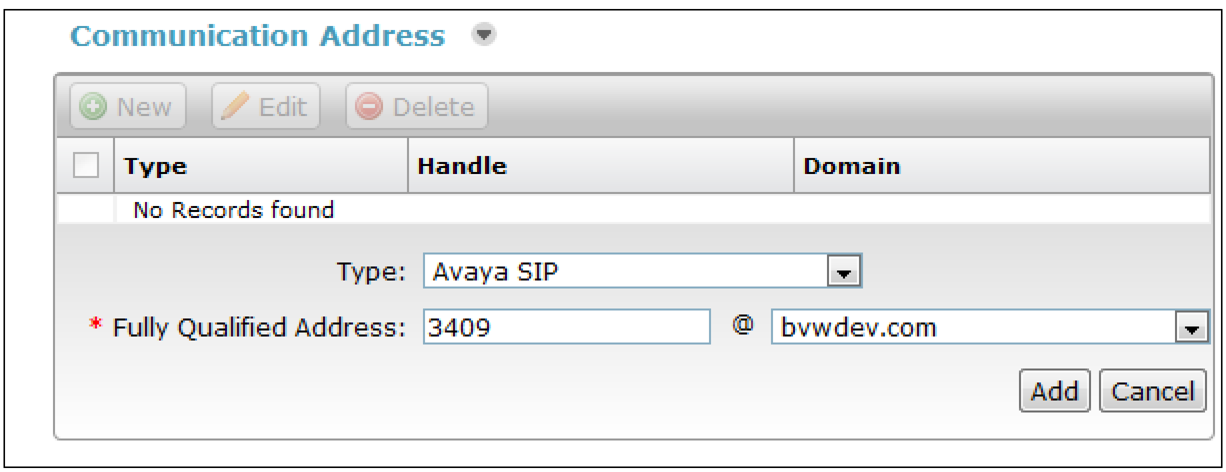
- <u>Identity section</u>
  - **Last Name** – Enter last name of user.
  - **First Name** – Enter first name of user.
  - **Login Name** – Enter extension number@sip domain. The sip domain is defined as Authoritative Domain in Communication Manager.
  - **Password** – Enter password to be used to log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



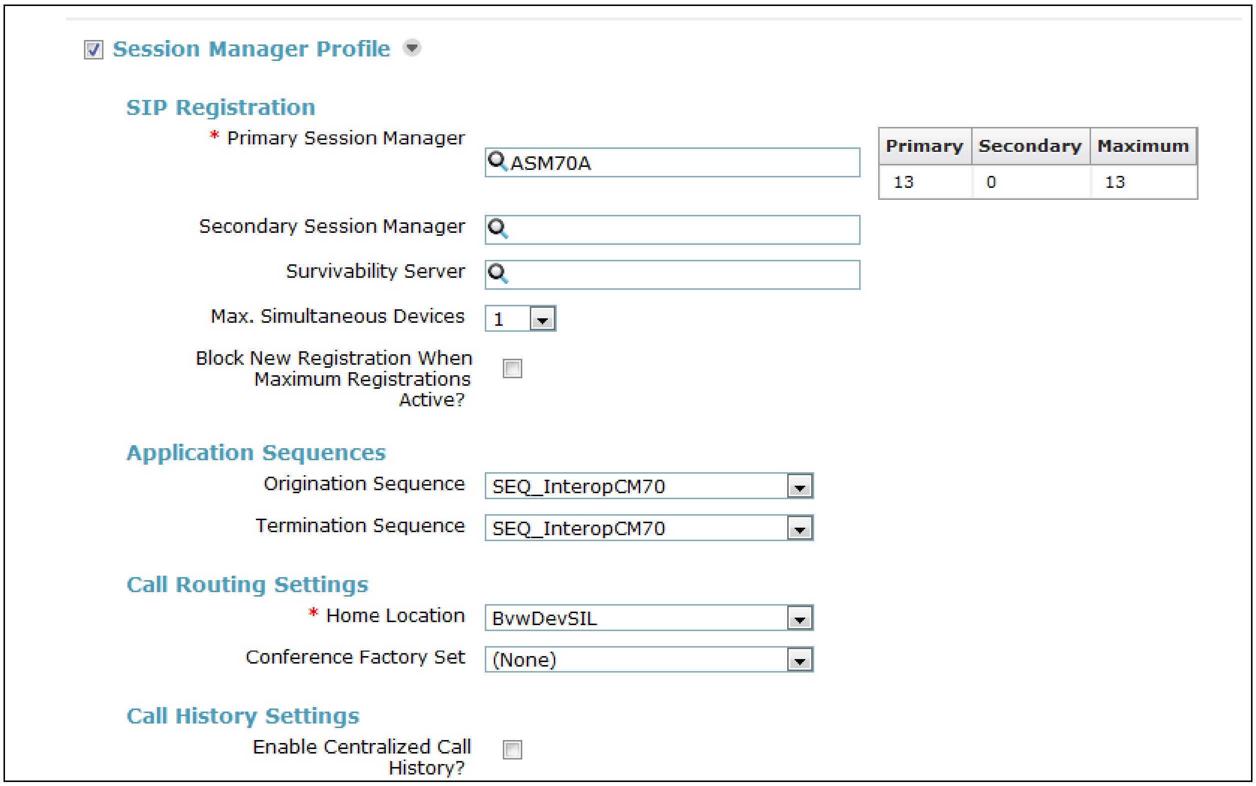
- <u>Communication Profile section</u>
    - **Communication Profile Password –** Type Communication profile password in this field
    - **Confirm Password** – Repeat value entered above.



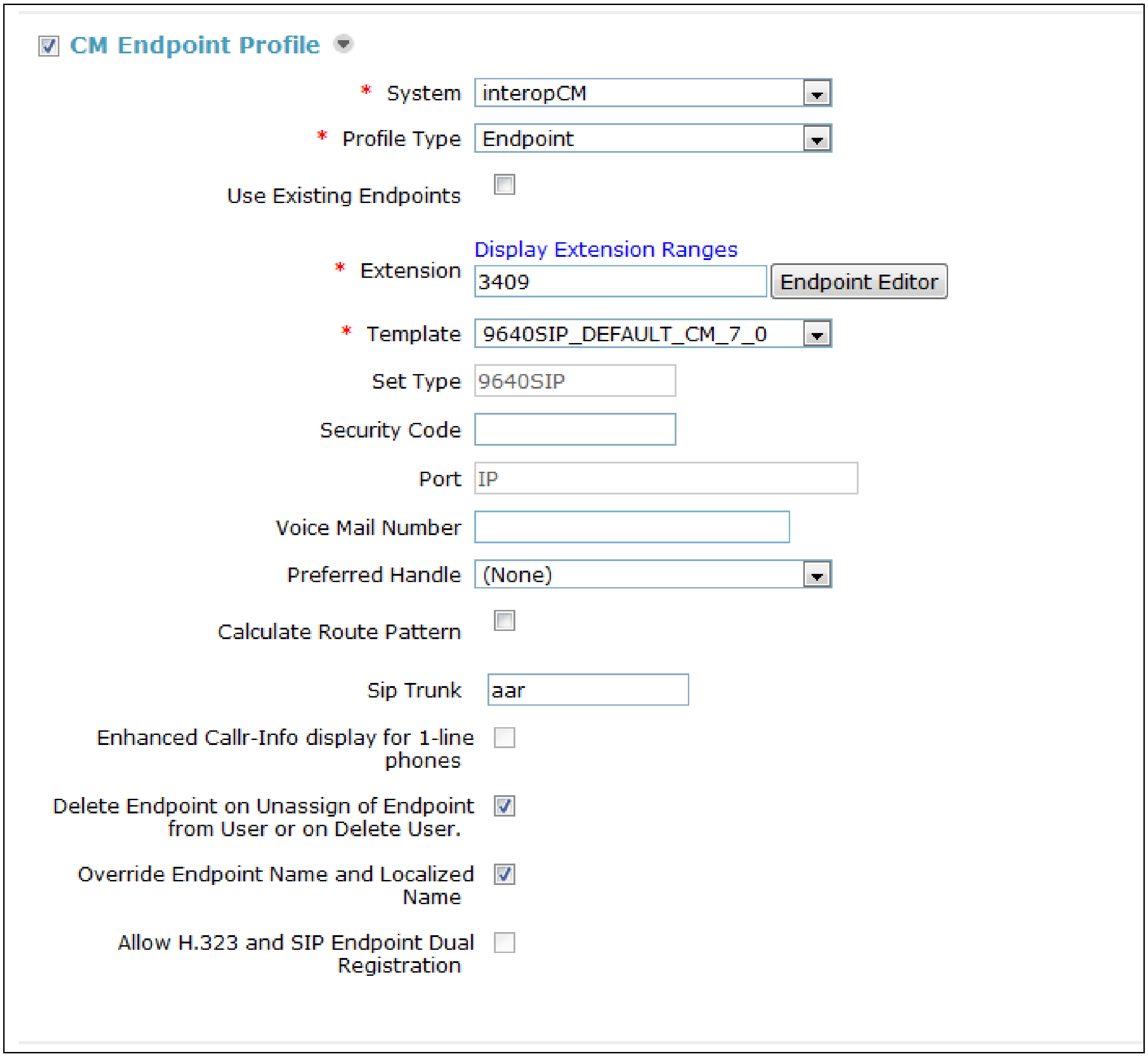
- <u>Communication Address sub-section</u>
    - **Fully Qualified Address –** Enter the extension of the user and select a domain name.
    - Click the **Add** button



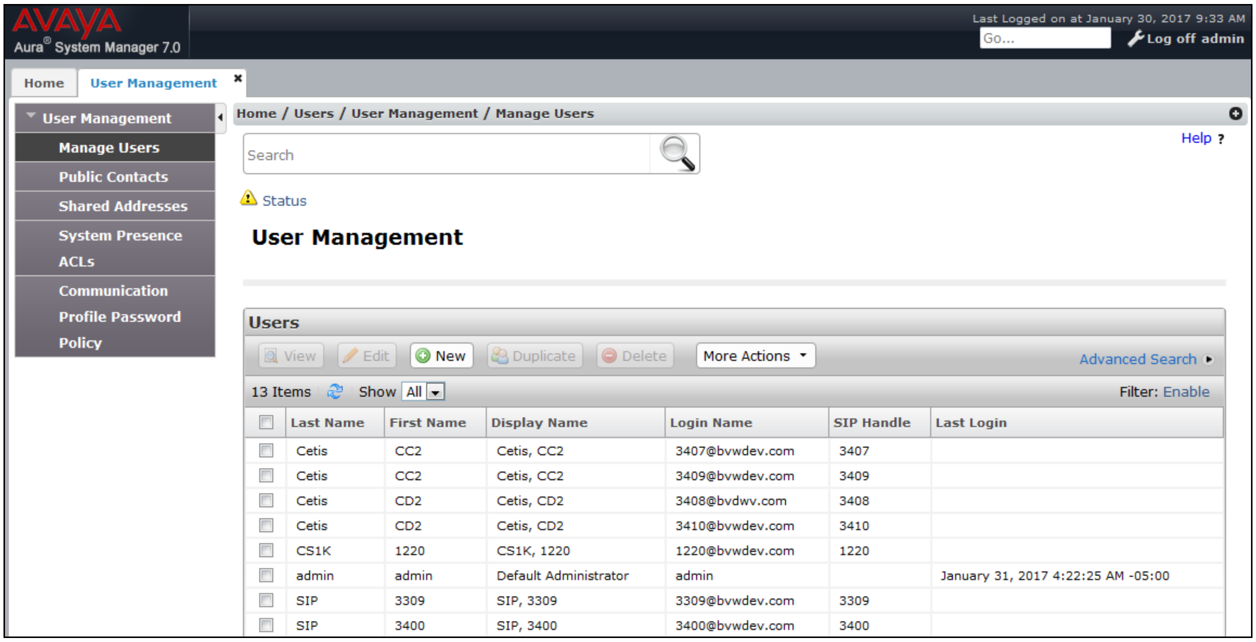
- <u>Session Manager Profile section</u>
    - **Primary Session Manager** – Select one of the Session Managers.
    - **Secondary Session Manager** – Leave this field as blank at default.
    - **Survivability Server** – Select **(None)** from the drop-down menu.
    - **Origination Sequence –** Select 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.



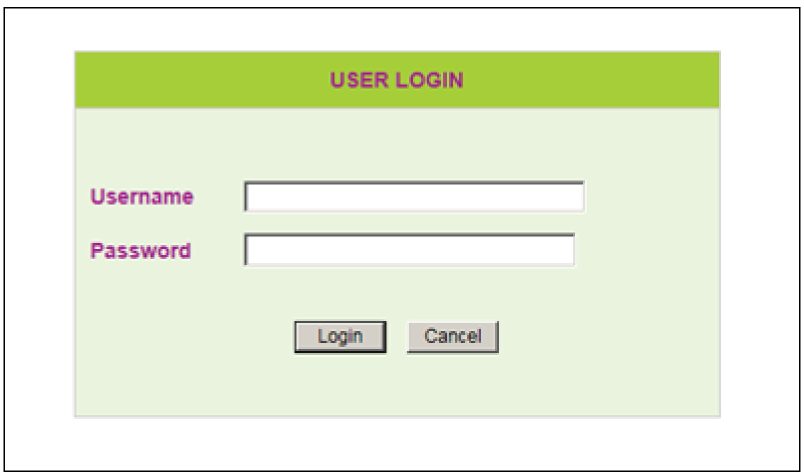
The following page shows the Cetis E203IP users created during the test.



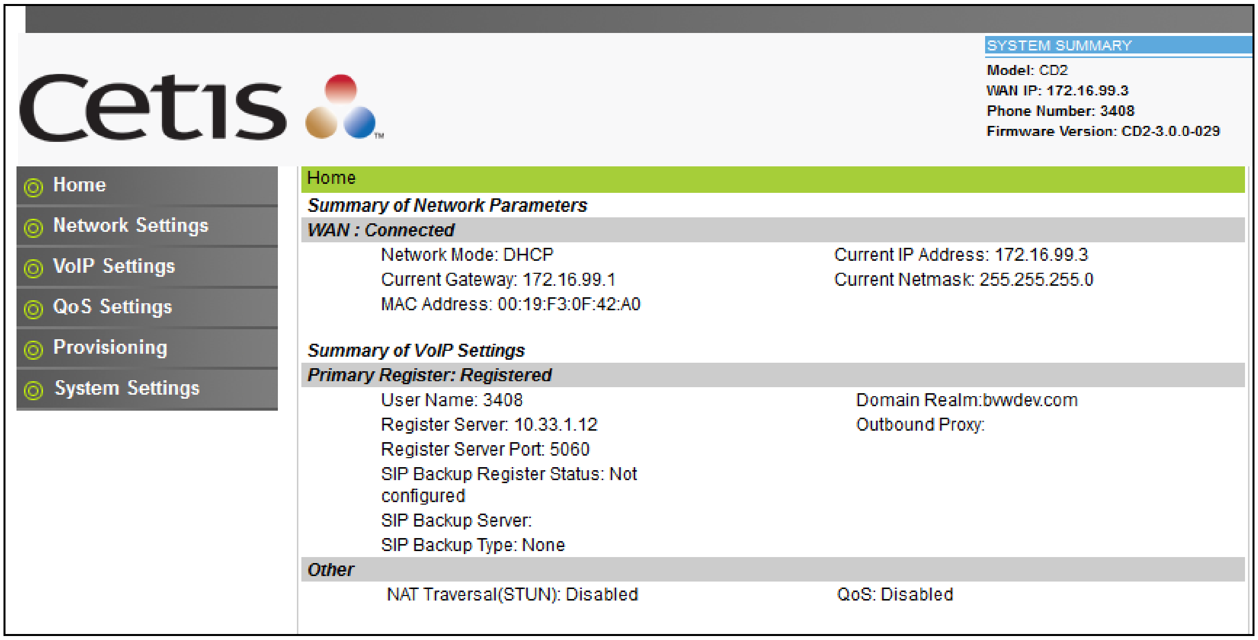
7. Configure Cetis e-Series E203IP SIP Telephones

In this section, an assumption was made that an engineer was able to connect to the phone through the web interface (i.e., using the default IP address). To configure the phone setting, enter http://<ip address of the Cetis E203 SIP Telephone> in the URL field of your browser. Log in with the appropriate credentials for accessing the Cetis E203IP settings page.

Access the Cetis E203IP 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 dynamic IP address from DHCP 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.



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

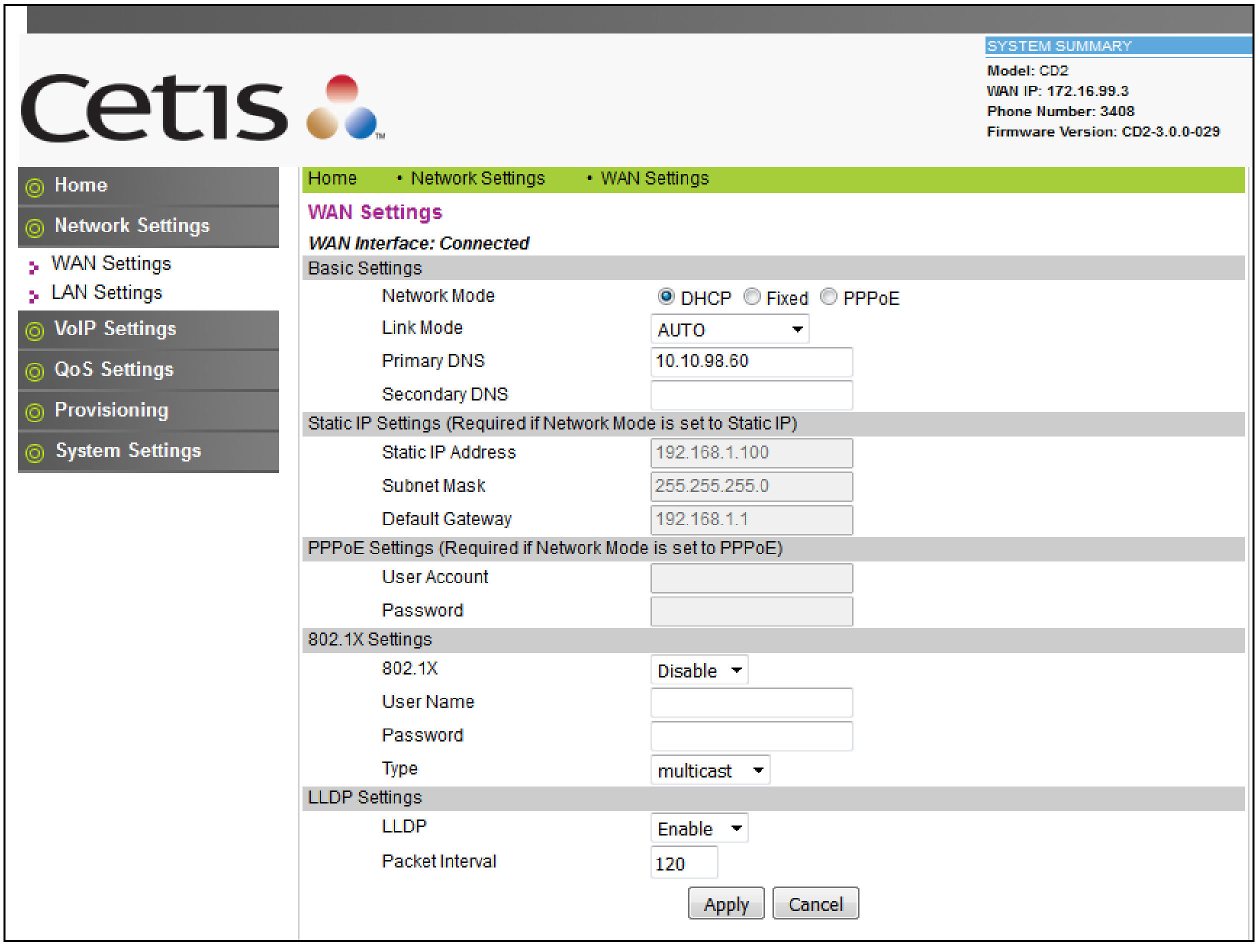
All Cetis IP phones share the same base chipset and firmware, meaning that models using the same number firmware version share the same traits and compatibility. Server registrations, SIP messaging, and call control are all the same. The different model prefixed versions are to accommodate variances in single vs. 2-line capability, cordless vs. cordless radio handsets and LCD display screen sizes. Example: CC2-3.0.0-029.bin is the firmware for **C**etis **Cordless 2**-line models including E203IP, M200IP, 9602IP, and NDC2200IP

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:

- **Basic Settings**
- **802.1X Settings**
- **LLDP Settings**

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

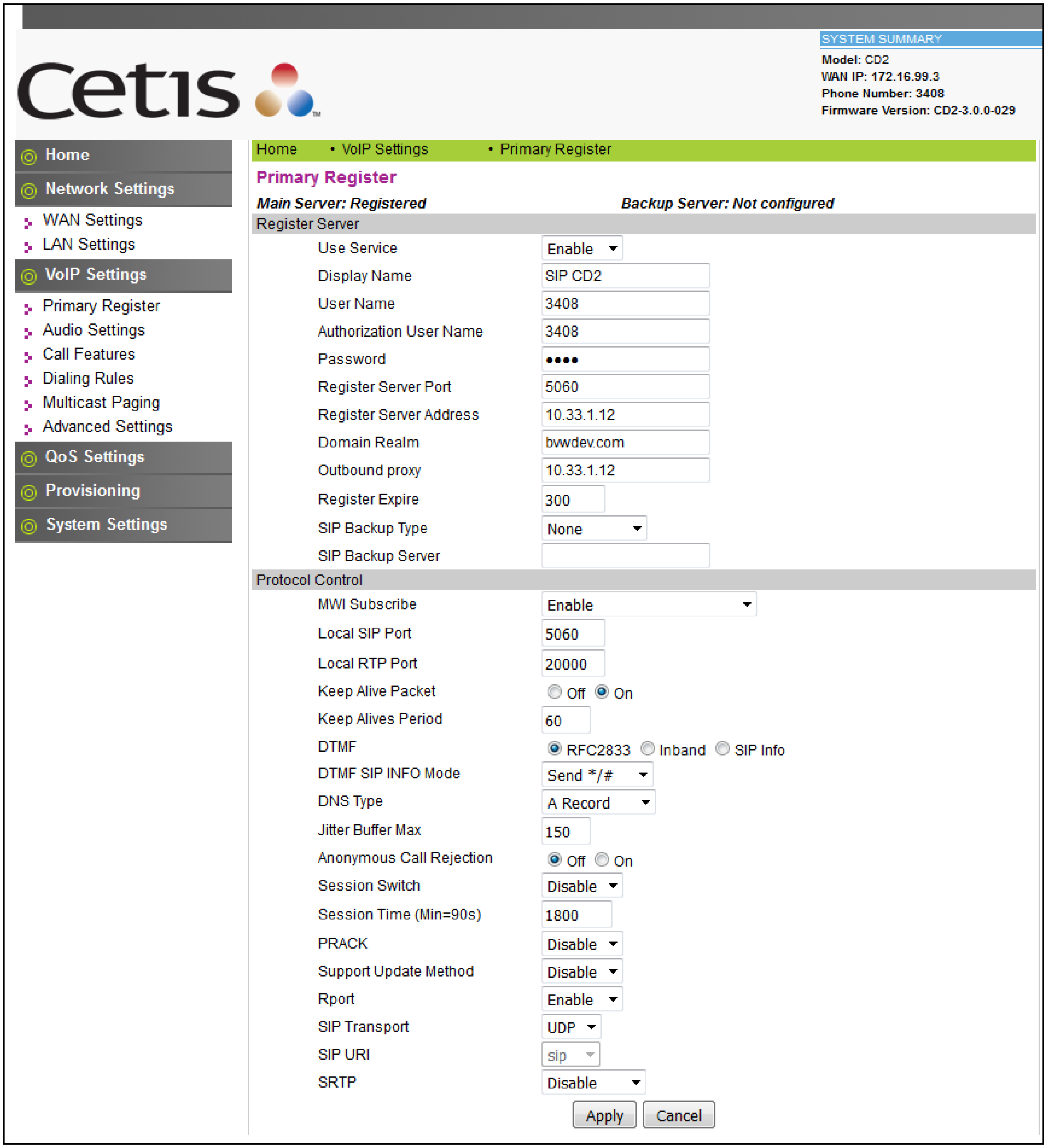


Select **Primary Register** under the **VoIP Settings** section.
Provide the following information:

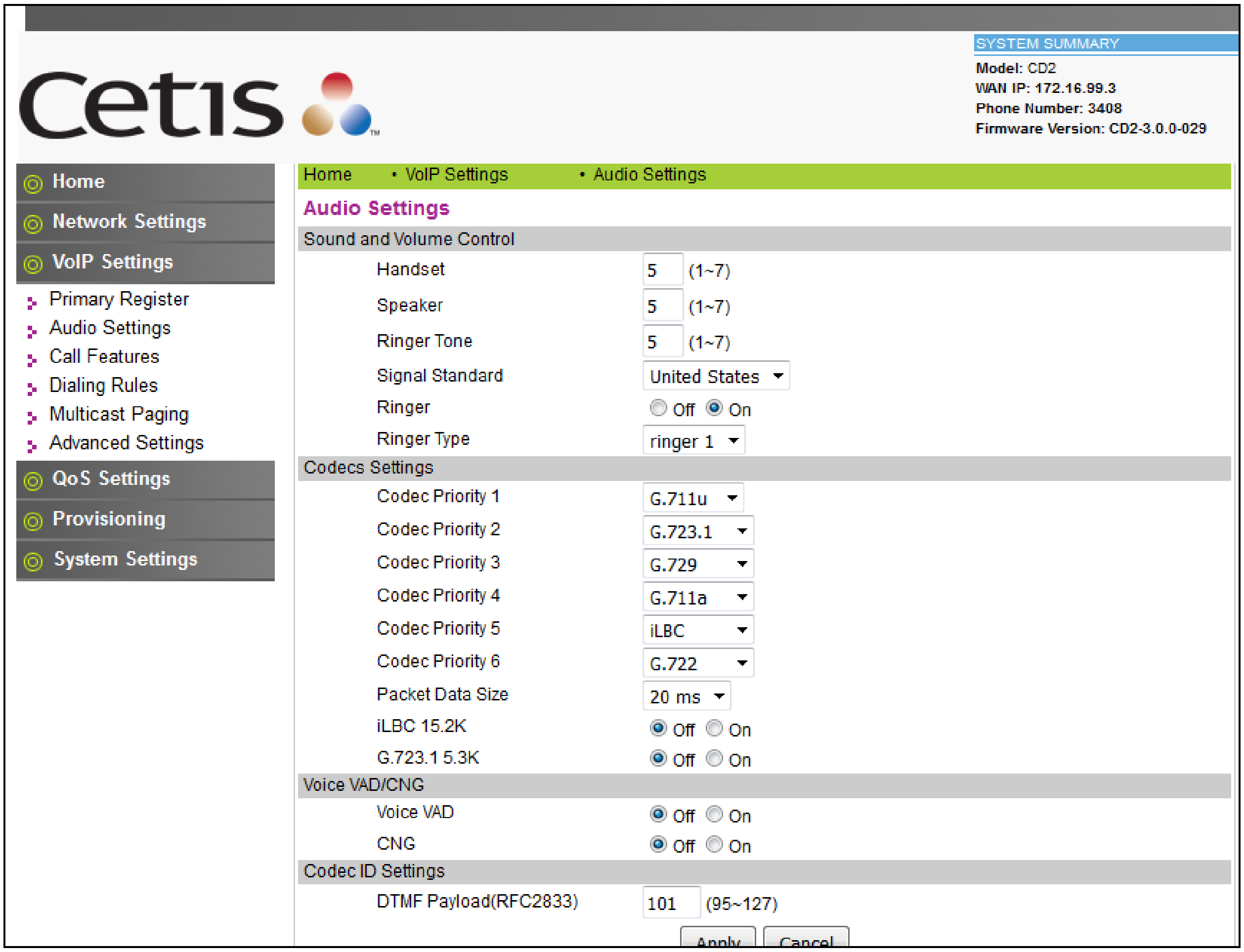
- **Use Service** – Select **Enable**.
- **Display Name** – Enter a descriptive name.
- **Register Server Address** – Enter the IP address of Session Manager.
- **Register Server Port** – Enter **5060** for UDP.
- **User Name** - Enter the user name created in **Section 6.2.**
- **Password -** Enter the Communication Profile password created in **Section 6.2**
- **Authorization User Name** - Enter the user name as configured in **Section 6.2**.
- **Domain Realm** – Used **bvwdev.com** during the test.
- **Outbound proxy -** Enter the IP address of Session Manager.
- **SIP Tranport –** Select **UDP** from the dropdown menu. Note that UDP transport is recommended to avoid the issue as listed in **Section 2.2**.

In the **Protocol Control** section leaves all value at default which has **MWI Subcribe** enabled and **DTMF** selected as RFC2833.

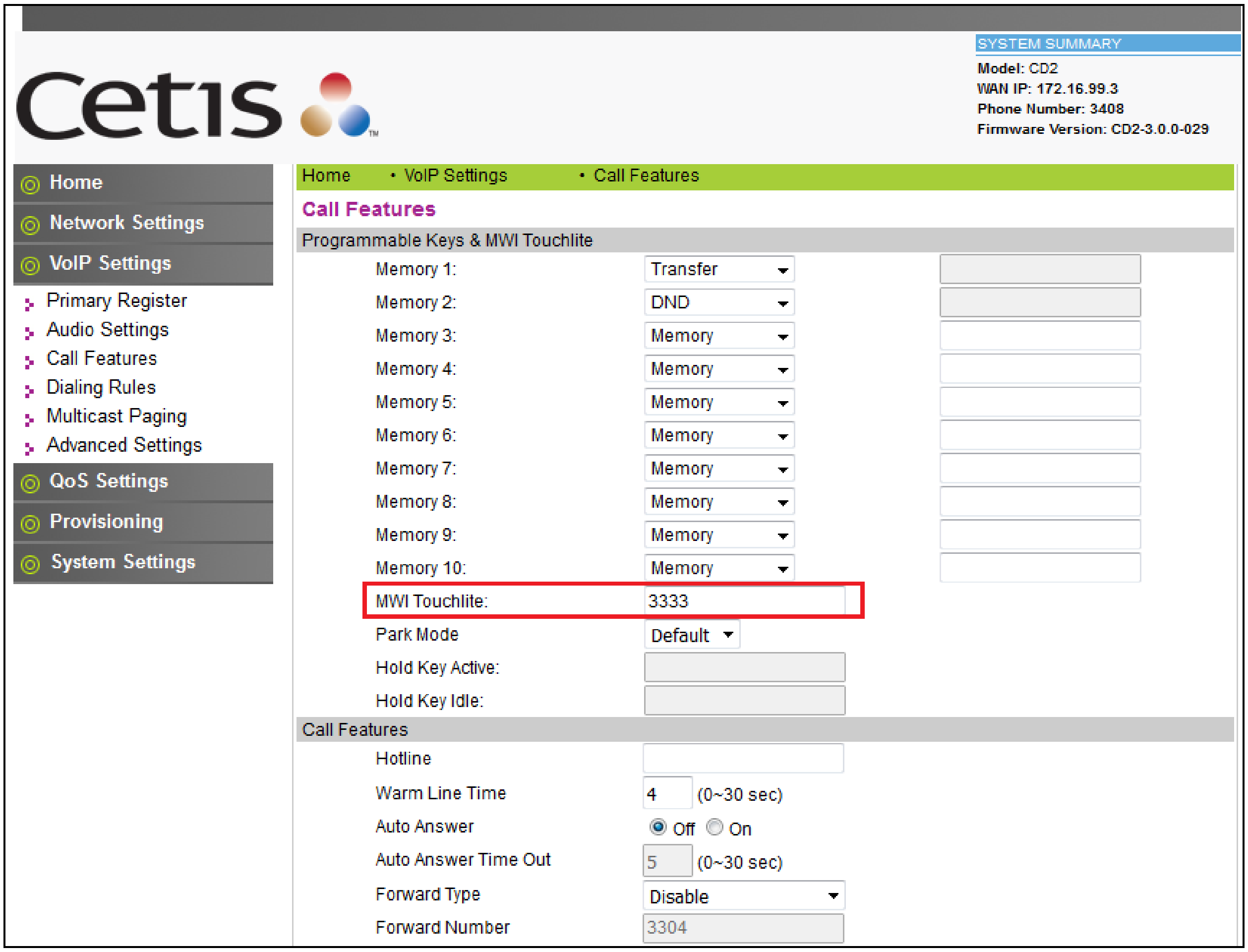
Click **Apply**.


Cetis
SYSTEM SUMMARY
Model: CD2
WAN IP: 172.16.99.3
Phone Number: 3408
Firmware Version: CD2-3.0.0-029
Home
Network Settings
WAN Settings
LAN Settings
VoIP Settings
Primary Register
Audio Settings
Call Features
Dialing Rules
Multicast Paging
Advanced Settings
QoS Settings
Provisioning
System Settings
Home • VoIP Settings • Primary Register
Primary Register
Main Server: Registered
Backup Server: Not configured
Register Server
Use Service Enable
Display Name SIP CD2
User Name 3408
Authorization User Name 3408
Password ••••
Register Server Port 5060
Register Server Address 10.33.1.12
Domain Realm bvwdev.com
Outbound proxy 10.33.1.12
Register Expire 300
SIP Backup Type None
SIP Backup Server
Protocol Control
MWI Subscribe Enable
Local SIP Port 5060
Local RTP Port 20000
Keep Alive Packet Off On
Keep Alives Period 60
DTMF RFC2833 Inband SIP Info
DTMF SIP INFO Mode Send */#
DNS Type A Record
Jitter Buffer Max 150
Anonymous Call Rejection Off On
Session Switch Disable
Session Time (Min=90s) 1800
PRACK Disable
Support Update Method Disable
Rport Enable
SIP Transport UDP
SIP URI sip
SRTP Disable
Apply
Cancel

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

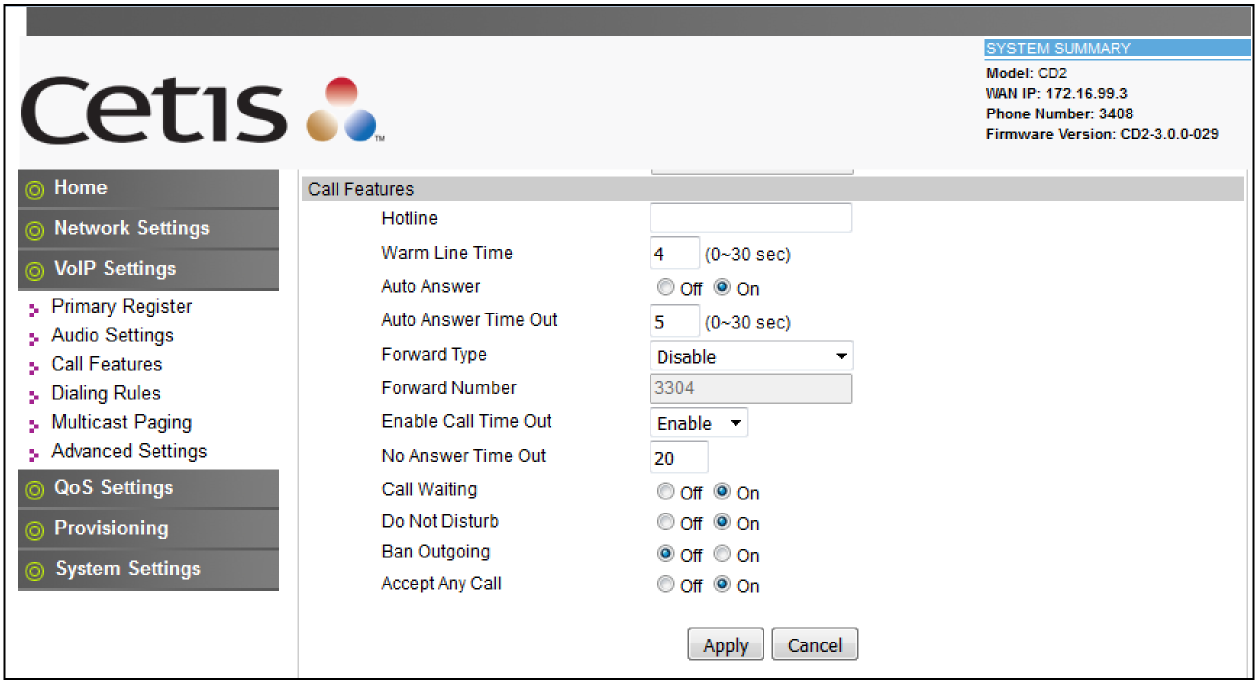


Select **Call Features** under the **VoIP Settings** section. In this page, a customer can program the memory buttons. The Cetis E203IP comes with 10 memory buttons. Enter the voicemail number of Aura Messaging in the **MWI Touchlite** box this setting allows user to access to the voicemail system by pressing the Message button on the phone.



Under the Call Features section in the right pane, two features (Auto Answer, Do Not Disturb and Call Forward) are tested.

After the configuration is completed, click **Apply**.

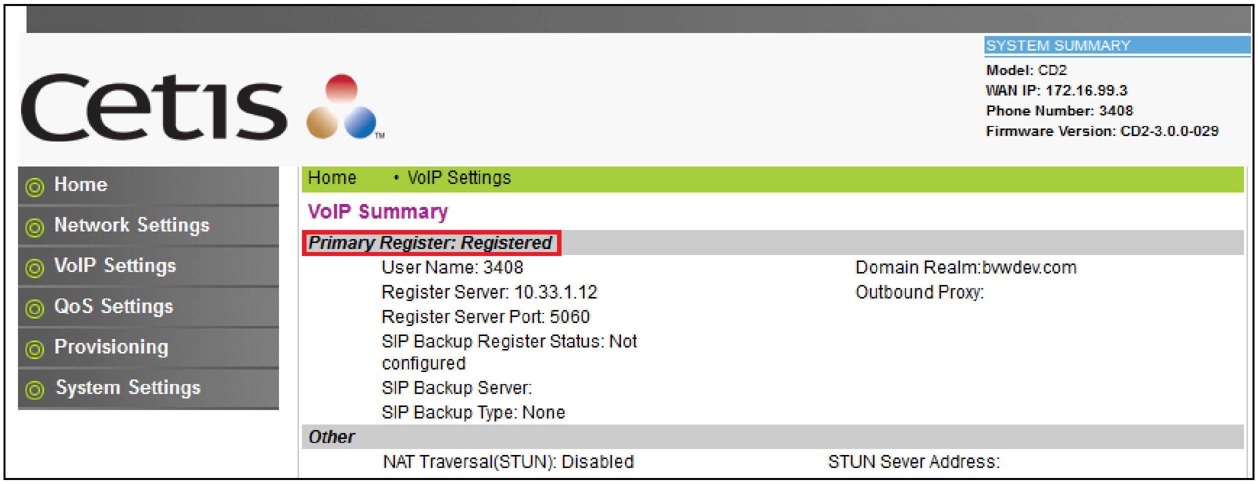


8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and the Cetis E203IP SIP Telephones.

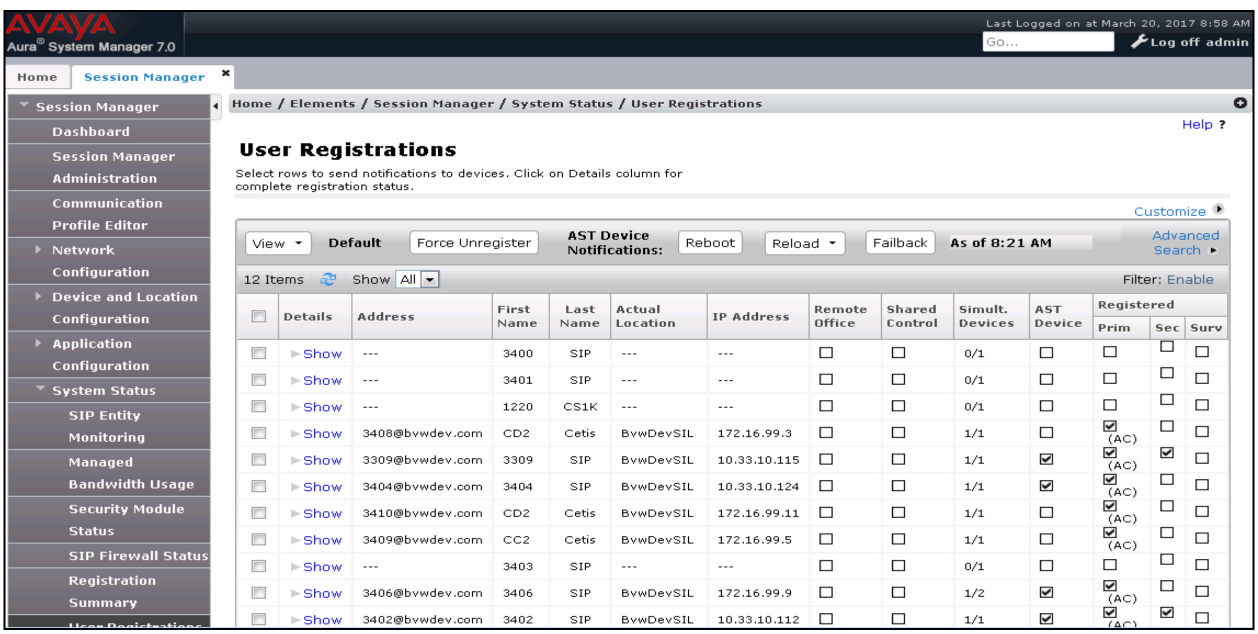
8.1. Cetis E203IP SIP Telephones.

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



8.1. Session Manager.

Web access to System Manager with appropriate credentials, and navigate to **Home → Elements → Session Manager → System Status → User Registration**. Verify the Cetis E203IP SIP Telephones are registered to Session Manager.



9. Conclusion

These Application Notes have described the administration steps required to integrate the Cetis E203IP 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**.

10. 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, May 2016, Issue 2, Document Number 03-300509

[2] *Administering Avaya Aura® Session Manager*, Release 7.0, May 2016, Issue 2

[3] *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0, May 2016, Issue 2

[4] Cetis E203IP *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.