Avaya Solution & Interoperability Test Lab

Application Notes for Cetis 3rd Generation 9602IP Cordless SIP 2-Line Telephone Release 3.0.0-043 with Avaya Aura® Session Manager Release 8.0 - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate Cetis 3rd Generation 9602IP Cordless SIP 2-Line Telephone with Avaya Aura® Session Manager. The Cetis 3rd Generation 9602IP SIP 2-Line Telephone with LCD Display is cordless telephone that was 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 3rd Generation 9602IP Cordless SIP 2-Line telephone (hereafter referred as Cetis 9602IP) with Avaya Aura® Session Manager. The Cetis 9602IP Telephone is designed for the hospitality industry. In the compliance test, Cetis 9602IP SIP telephone registered with Avaya Aura® Session Manager as a third-party SIP user and used telephony limited features from Commutation Manager, established calls with other Avaya SIP and H.323 telephones, and executed telephony and hospitality features. As a third-party SIP phone, the range of features available would not be the same as the Avaya first-party SIP phones.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually placed from/to the Cetis 9602IP extension, with call controls such as hold/resume, unattended, attended transfer and conference performed from the caller.

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 the Cetis 9602IP telephone do not utilize TLS and secure media SRTP encryption features as requested by Cetis.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Cetis 9602IP telephone with Session Manager.

- Calls between Cetis 9602IP telephone and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetis 9602IP telephone and the PSTN.
- G.711 and G.729 codec support.
- Transport protocol UDP.
- Proper recognition of DTMF tones, Voice Mail and Messaging Waiting Indicator.
- Basic telephony features, including hospitality feature, inbound/outbound, hold, mute, transfer, forward and conference.
- Use of programmable buttons on the Cetis 9602IP telephone.
- Proper system recovery after a restart of the Cetis 9602IP telephone and loss of IP connectivity.

The serviceability testing focused on verifying that the Cetis 9602IP telephone come back into service after re-connecting the Ethernet connect or rebooting the phone.

2.2. Test Results

All test cases executed and passed with the following observations.

- Cetis 9602IP does not support the attended transfer at this release.
- The call park was failed because the Cetis 9602IP could not transfer an ongoing call to the feature access code of the call park to park the call.
- The blind transfer is only completed by using the telephony base and not using the cordless handset.

2.3. Support

For technical support on the Cetis 9602IP telephone, contact Cetis Support via phone, email, or website.

- **Phone:** +1 (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 9602IP telephone with Session Manager. The Cetis 9602IP telephone registered with Session Manager via SIP.

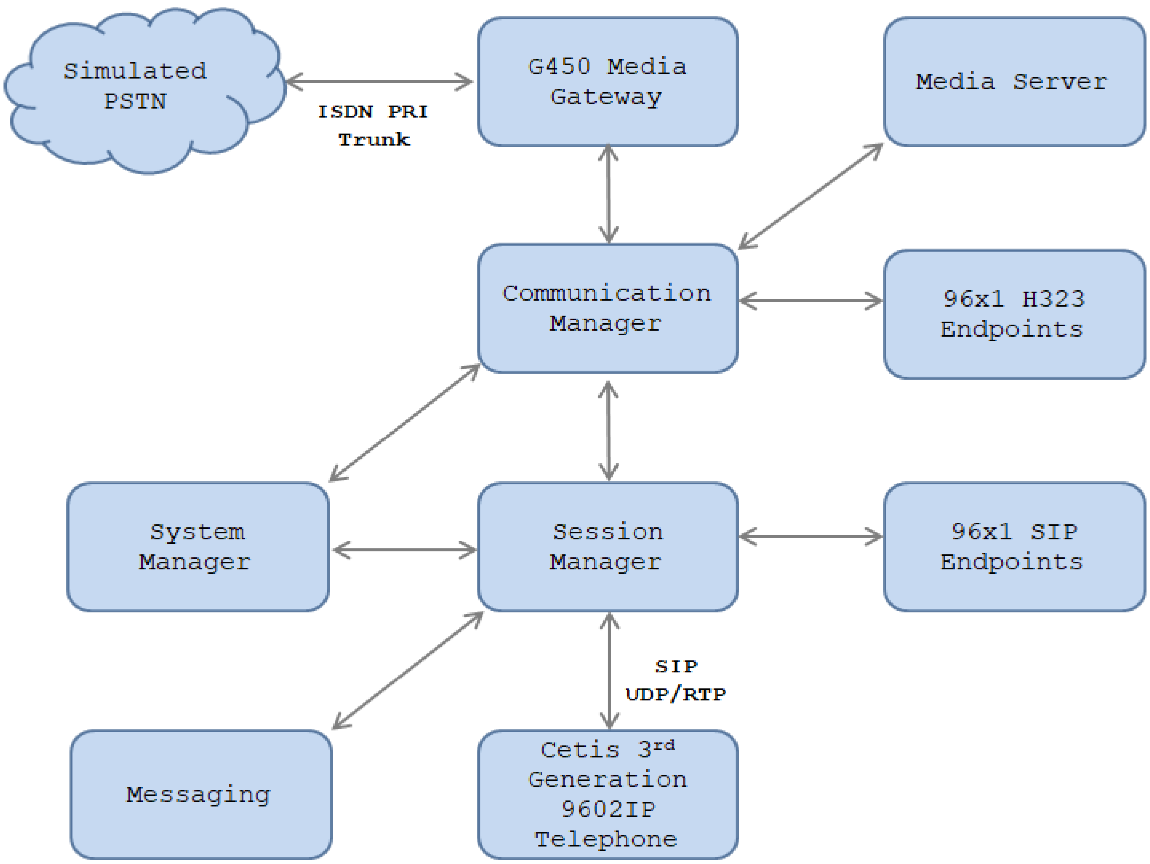


Figure 1: Test Configuration Diagram with Avaya Aura® Session Manager

The following table indicates the IP addresses that were assigned to the systems in the test configuration diagram:

Description	**IP Address**
System Manager	10.33.1.10
Session Manager	10.33.1.11
Communication Manager	10.33.1.6
Messaging	10.33.1.5
Media Server	10.33.1.30
G450 Media Gateway	10.33.1.29
H.323 Endpoints	10.33.5.10-11
SIP Endpoints	10.33.5.12-14
Cetis 9602IP SIP Telephone	192.168.199.15

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: C32-3.0.0-040.bin is the firmware for Cetis Cordless 2-line models including 3300IP and 9600IP.

Common Firmware on Cetis, Inc. SIP Phones

Cetis' current SIP firmware follows a naming convention based on and mated to the phone model name. The newest Cetis SIP phones all share the same base chipset and firmware, meaning that models using the same number firmware version share the same traits and compatibility. Server registration, SIP messaging, and call control are all the same. The different prefixes are to accommodate variances in single vs. 2-line capability, corded vs. cordless radio handsets and LCD display screen sizes.

Example: CC1-3.0.0-040.bin is a firmware file for the models associated with that CC1 prefix. Firmware number 3.0.0-040 could have any of the below prefixes tying it to the associated models

Prefix	Model	Features
CC1	**M100IP, ND2100IP, E100IP** : 1-line, corded	
CC2	**M200IP, ND2200IP, E200IP** : 2-line, corded	
CD1	**9600IP, M103IP, NDC2100IP, E103IP** : No LCD display, 1-line, cordless	
CD2	**9602IP, M203IP, NDC2200IP, E203IP** : 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, M100IP-TRM** : 1-line, corded, Trimline form	
CT2	**3302IP-TRM, M100IP-TRM** : 2-line, corded, Trimline form	
CM1	**E100IP-TRM** : 1-line, corded, Trimline form	
CM2	**E200IP-TRM** : 2-line, corded, Trimline form	

CC = Cetis Corded I CD = Cetis DECT/Cordless I CT/CM = Cetis Trimline I C3 = Cetis 3300 series

The current SIP phone firmware (3.x) is NOT compatible with the SIP phones using (1.x) firmware or (2.x) firmware. Each of these SIP endpoints are distinct and separate hardware technologies, although they will have the same physical form factor and physical aesthetic characteristics in many cases.

Notable additional features in the newest phones are:

Support of LLDP-MED protocols in network deployment I Support of macaddress named configuration files in network deployment. More sophisticated provisioning methods and re-direction server for cloud-based deployment is also supported.

4. Equipment and Software Validated

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

Equipment/Software	Version/Release
Avaya Aura® System Manager running on Virtualized Environment	8.0.1 Software Update Revision No: 8.0.1.0.038826
Avaya Aura® Session Manager running on Virtualized Environment	8.0.1 Build 8.0.1.0.801007
Avaya Aura® Communication Manager running on Virtualized Environment	8.0.1 Build 8.0.0.1.2.822 Patch 24826
Avaya Aura® Messaging running on Virtualized Environment	7.1 SP1
Avaya Aura® Server Media running on Virtualized Environment	8.0.1 Build 8.0.0.117
Avaya G450 Media Gateway	40 .25 .0
Avaya 96x1 IP Deskphones	7.1.4.0.11 (SIP) 6.714 (H323)
Avaya 1416 Digital Deskphone	Fw 1
Cetis 3rd Generation 9602IP Cordless SIP 2-Line Telephone	Firmware Version CD2-3.0.0-043

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: C32-3.0.0-040.bin is the firmware for Cetis Cordless 2-line models including 3300IP and 9600IP.

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
- Configure Dialing Plan

<u>Note</u>: 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**.

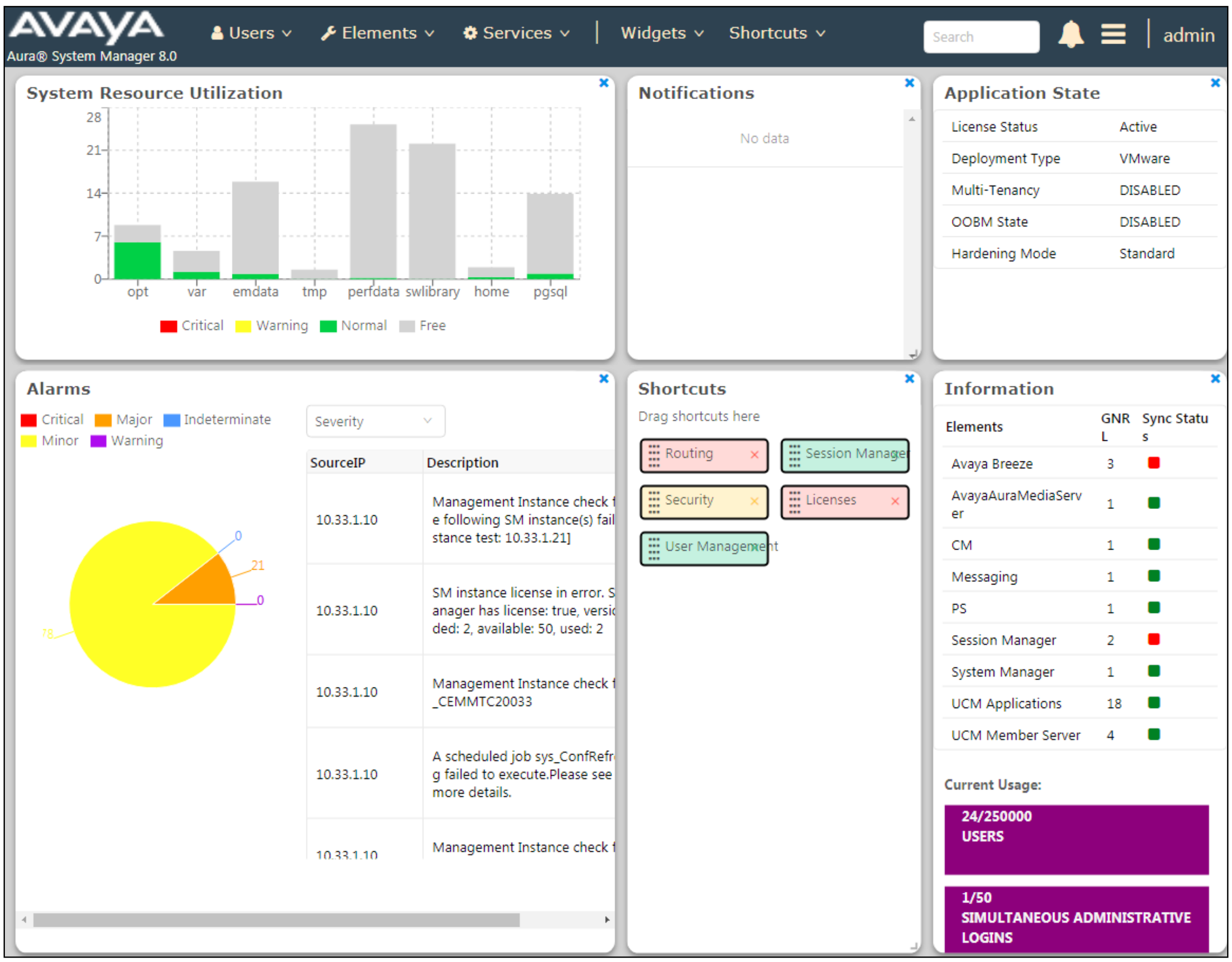
```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 3

    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

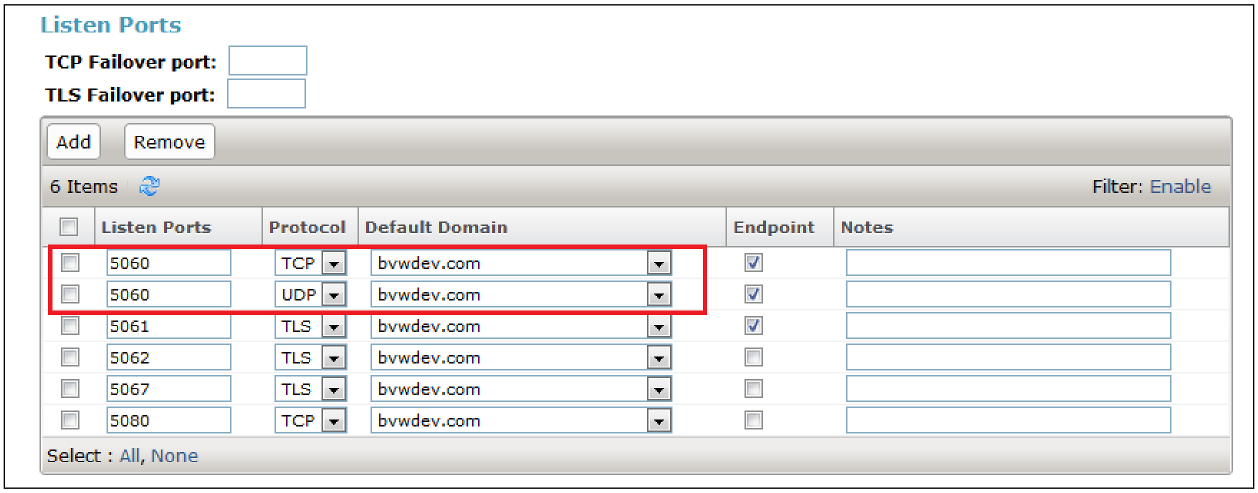
This section describes aspects of the Session Manager configuration required for interoperating with 9602IP SIP Telephone. It is assumed that the Domains, Locations, SIP Entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication Manager, Session Manager and Aura® Messaging.

Session Manager is managed via System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and click the **Log On** button.



6.1. Check Session Manager Ports

Each Session Manager Entity must be configured so that the Cetis 9602IP telephone 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. In the **Listen Ports** section, make sure that **TCP** and **UDP** entries are present with respective sip domain selected and **Endpoint** checked. The TCP and UDP entries are highlighted below however only **UDP** protocol was tested during the compliance test.



Listen Ports

TCP Failover port:

TLS Failover port:

Add Remove

6 Items Filter: Enable

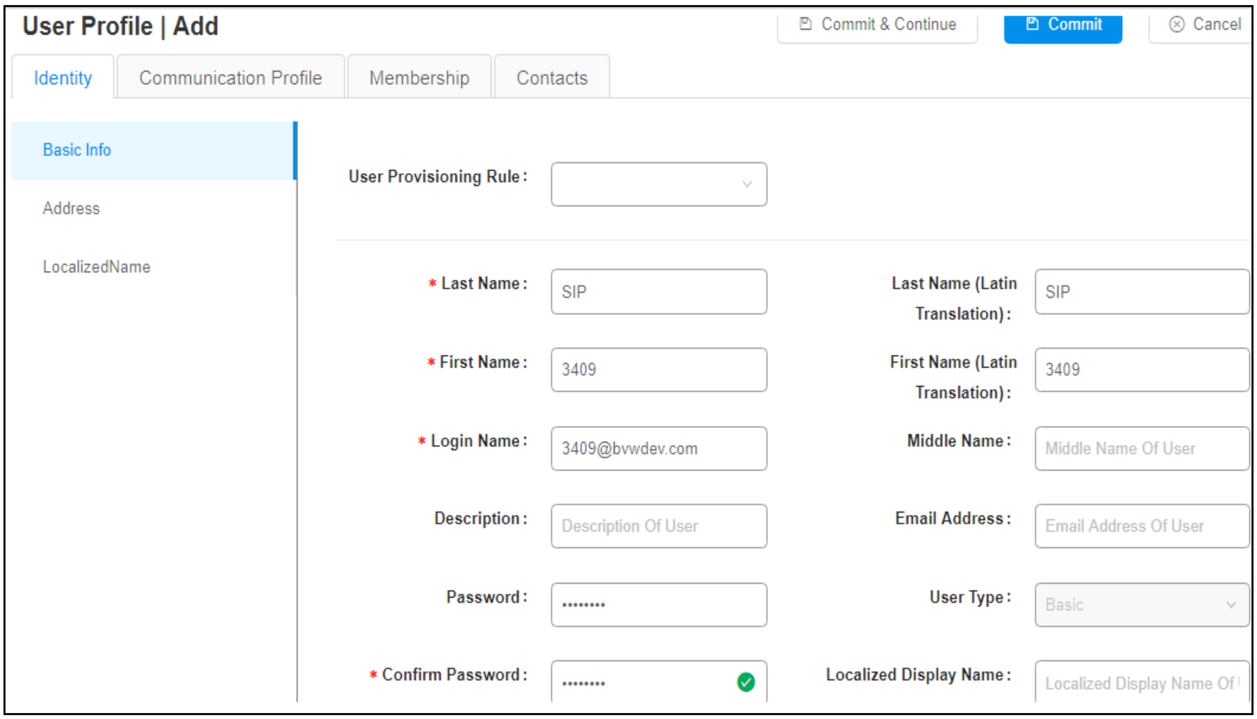
	Listen Ports	Protocol	Default Domain	Endpoint	Notes
☐	5060	TCP	bvwdev.com	☑	
☐	5060	UDP	bvwdev.com	☑	
☐	5061	TLS	bvwdev.com	☑	
☐	5062	TLS	bvwdev.com	☐	
☐	5067	TLS	bvwdev.com	☐	
☐	5080	TCP	bvwdev.com	☐	

Select : All, None

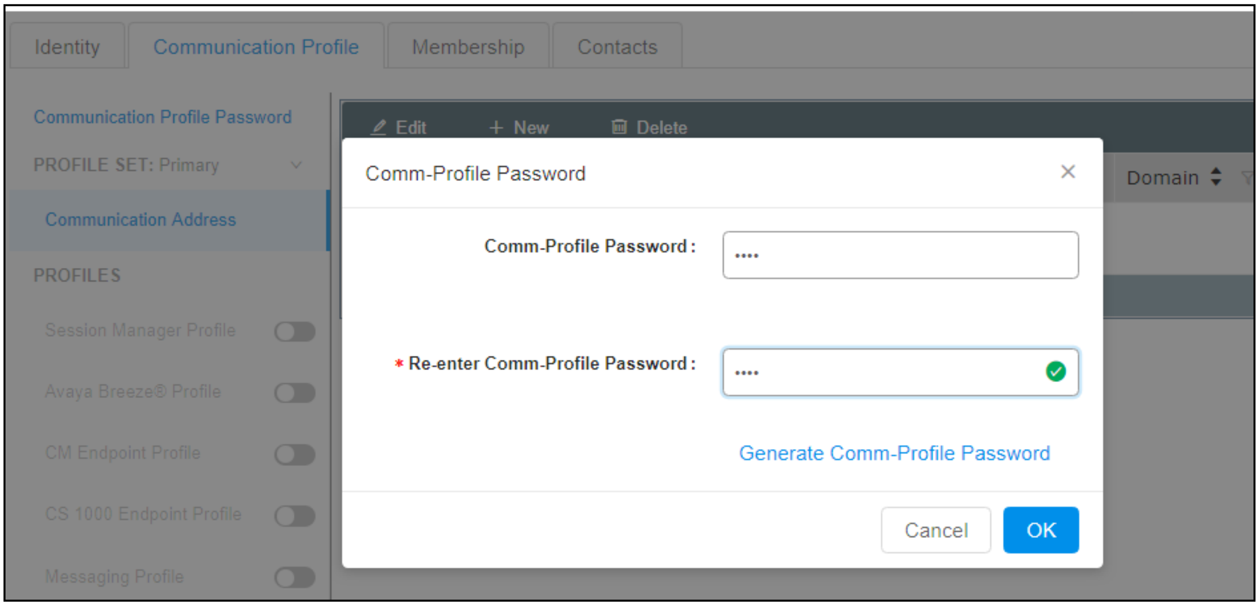
6.2. Add a SIP User

A SIP user must be created for Cetis 9602IP telephone to register to Session Manager. From the top menu of SMGR, navigate to **User Management → Manage Users → New** (not shown) and configure the following in the **Identity** tab.

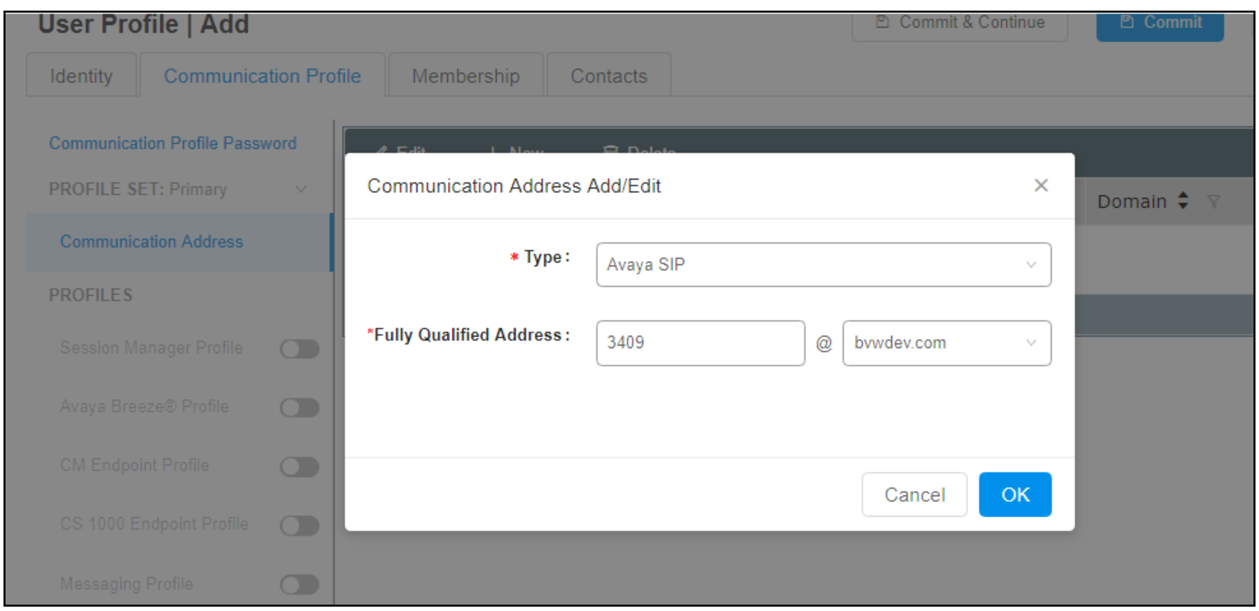
- **First Name** and **Last Name** Enter an identifying name
- **Login Name** Enter the extension number followed by the domain, in this case **3409@bvwdev.com**
- **User Type** Select **Basic** from the drop down list
- **Password** and **Confirm Password** Enter and confirm a password



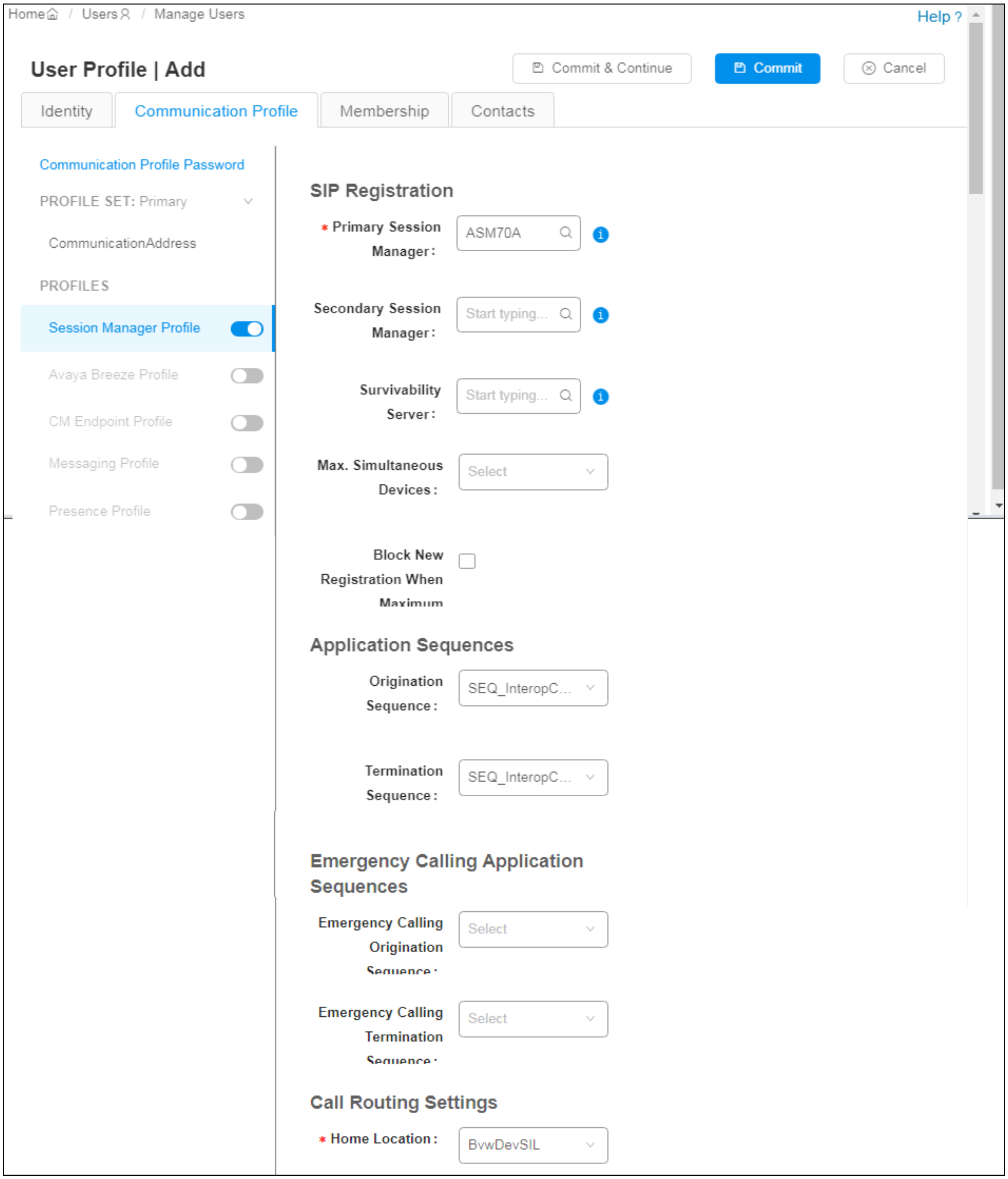
Click the **Communication Profile** tab and in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields, enter a numeric password. This password will be used to register the Cetis 9602IP telephone later in **Section 7**.



In the **Communication Address** section, for **Type** select **Avaya SIP** from the drop down list. In the **Fully Qualified Address** field enter the extension number as required and select the appropriate Domain from the drop down list. Click **OK** when done.



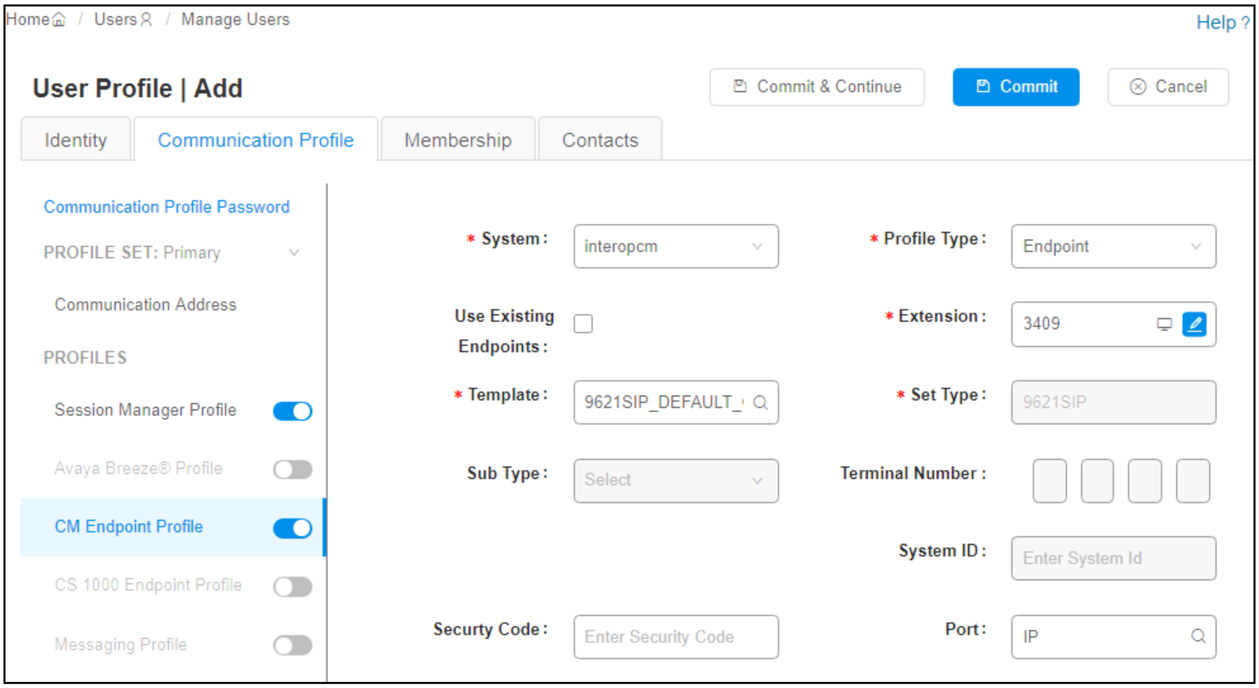
Click on the **Session Manager Profile** and configure the **Primary Session Manager, Origination Application Sequence, Termination Application Sequence** and **Home Location,** from the respective drop down lists. The Primary Session Manager used was **ASM70A**.



Place a tick in the **CM Endpoint Profile** check box and configure as follows:

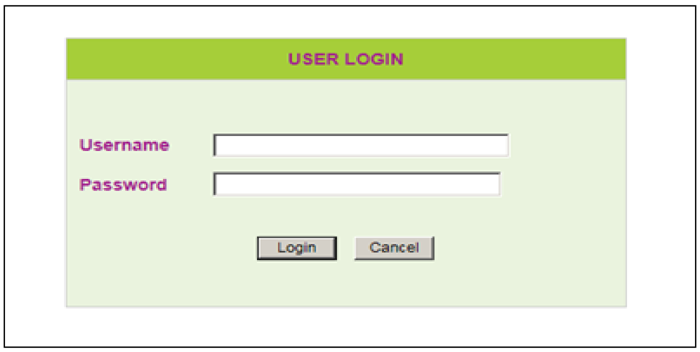
- **System** Select the relevant Communication Manager SIP Entity from the drop down list
- **Profile Type** Select **Endpoint** from the drop down list
- **Extension** Enter the required extension number, in this case **3409**
- **Template** Select **9621SIP_DEFAULT_CM_8_0** from the drop down list
- **Port** The "IP" is auto filled out by the system

Click on **Commit** button to save.



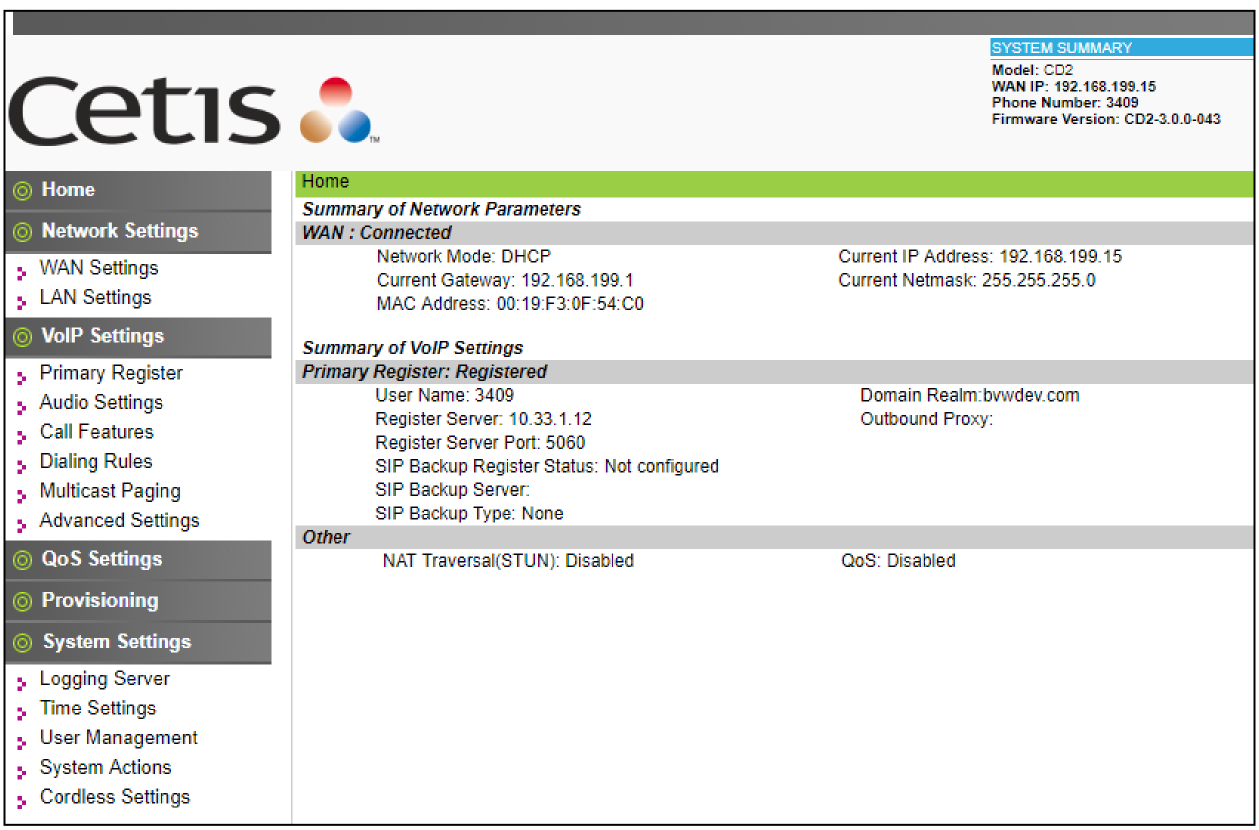
7. Configure Cetis 9602IP SIP Telephone

Access the Cetis 9602IP telephone web interface using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the Cetis 9602IP telephone. By default, DHCP is enabled on the Cetis telephones. To determine the IP address assigned to the Cetis telephone, enter **47# on the telephone to hear the IP address. Default **Username/Password** is admin/admin.



7.1. Network Settings

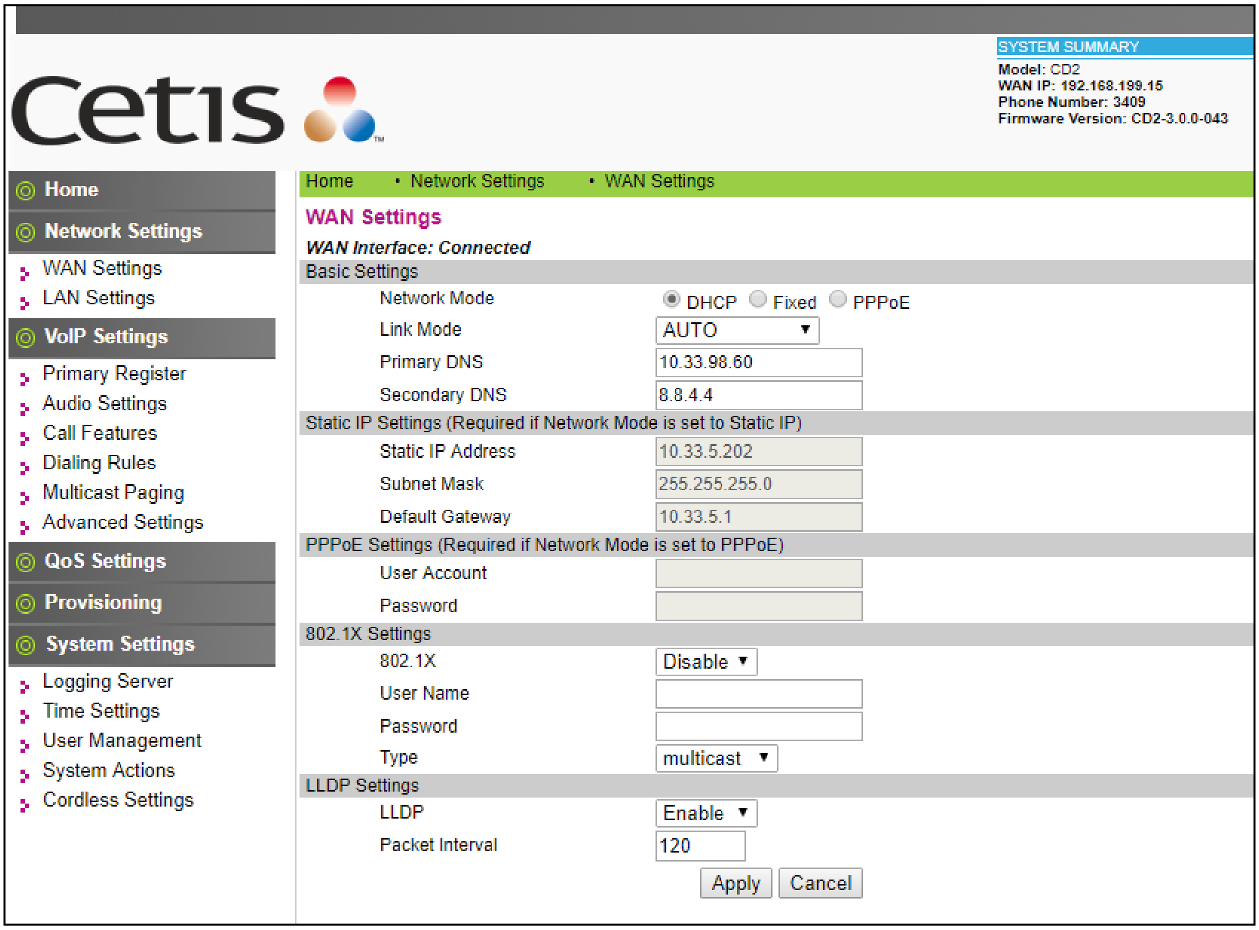
To view the network configuration, select the **WAN Settings** under the **Network Settings** section.



In the **WAN Settings** page, provide the following information in **Basic Settings** section.

- Select the **DHCP** option in the **Network Mode**.
- Enter DNS servers in the **Primary DNS** and **Secondary DNS**.

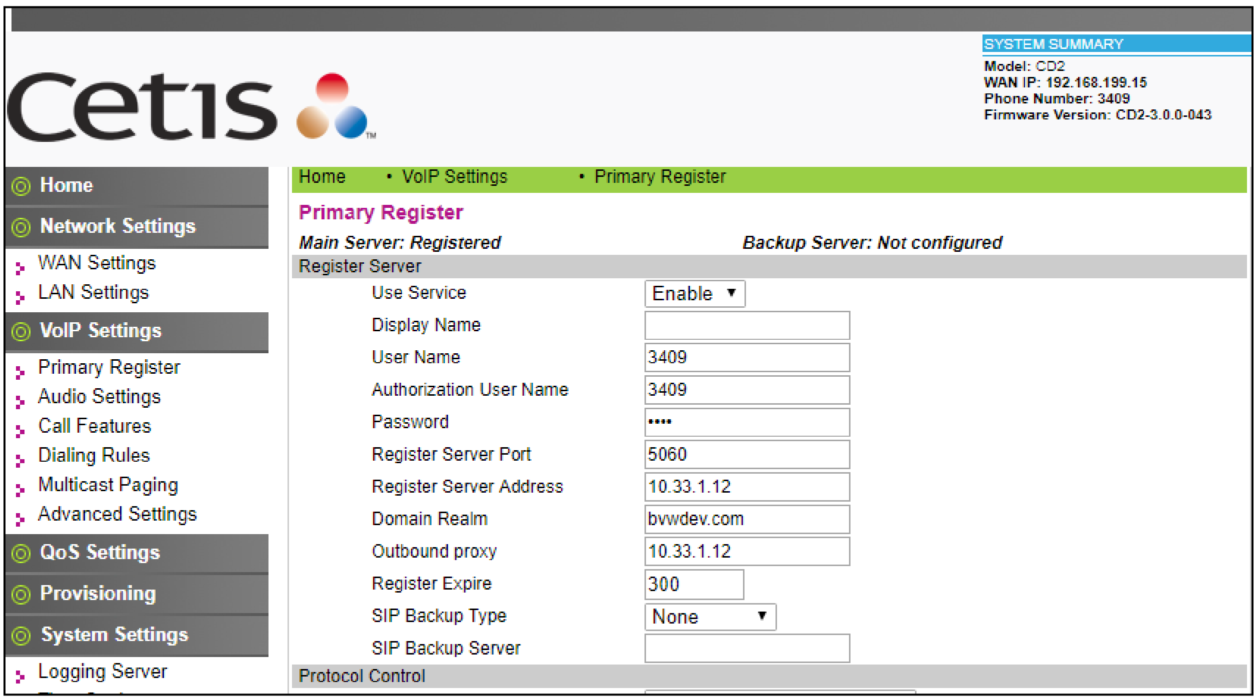
Leave other sections at the default values. The following screen show what was configured and used.



7.2. VoIP Settings

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

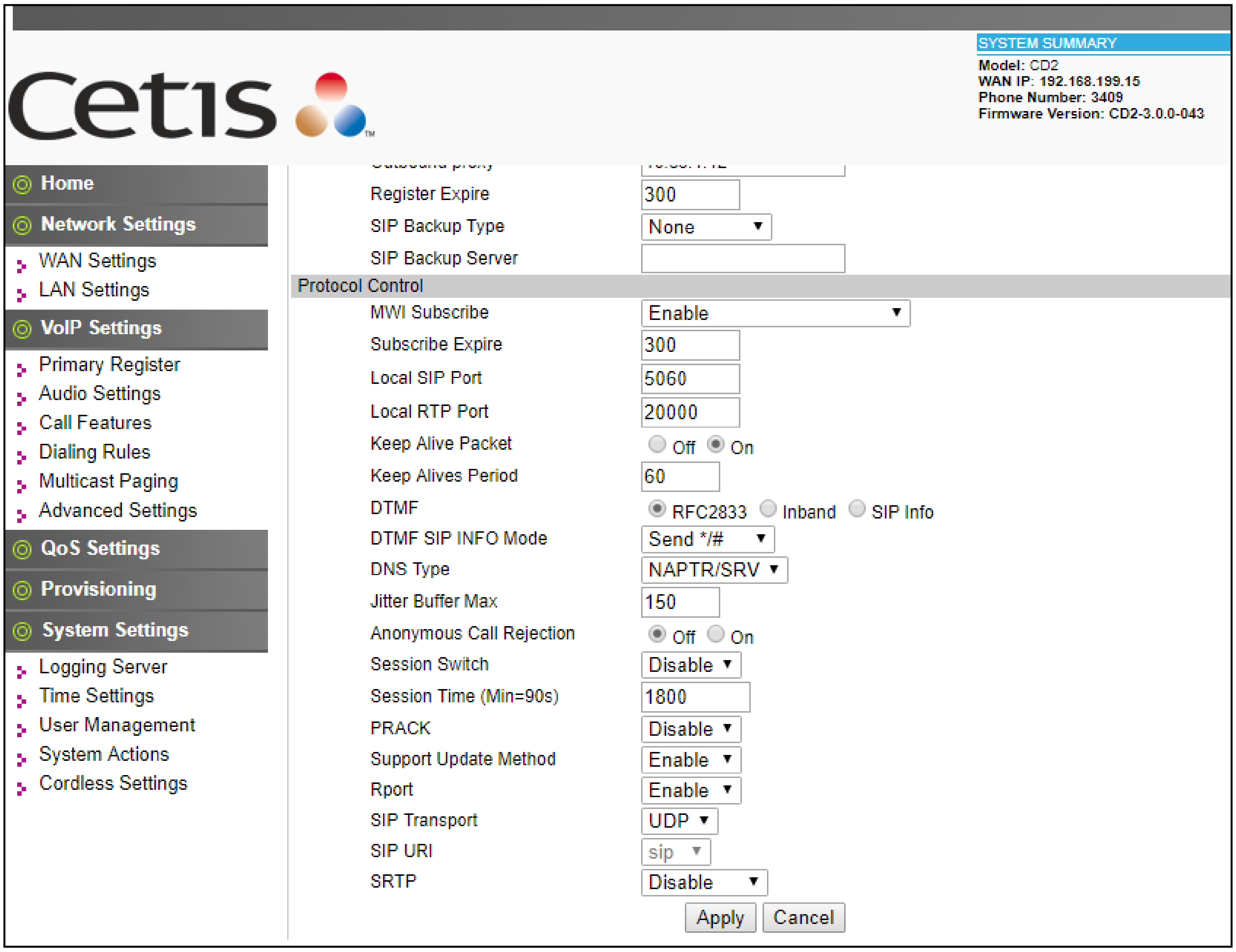
- **Use Service** – Select **Enable**.
- **Display Name** – Enter a descriptive name.
- **Register Server Address** – Enter the signaling IP address of Session Manager.
- **Register Server Port** – Enter **5060** for UDP.
- **User Name** - Enter the user name created in **Section 6.2**.
- **Authorization User Name** - Enter the user name as configured in **Section 6.2**.
- **Password** - Enter the password created in **Section 6.2**.
- **Domain Realm** – Enter sip domain **bvwdev** as configured in **Section 6.1**.
- Leave other fields at default value.



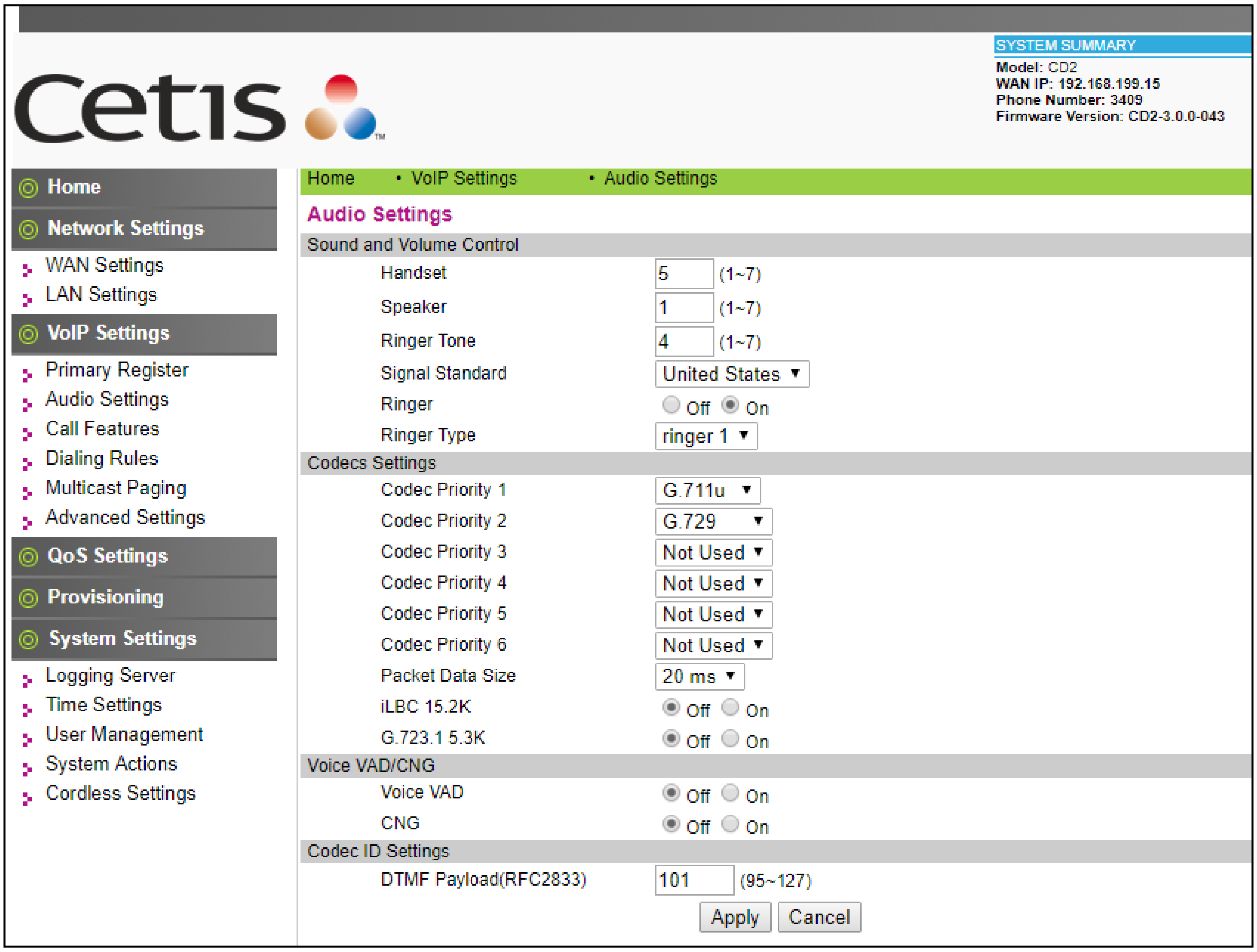
In the **Protocol Control** section, provide the folllowing values.

- **MWI Subscribe** – Select **Enable** from the dropdown menu.
- **DTMF** – Select the RFC2833 option.
- **SIP Tranport** – Select **UDP** from the dropdown menu.
- **Support Update Method** – Select **Enable** from the dropdown menu.
- Leave other fields at default value.

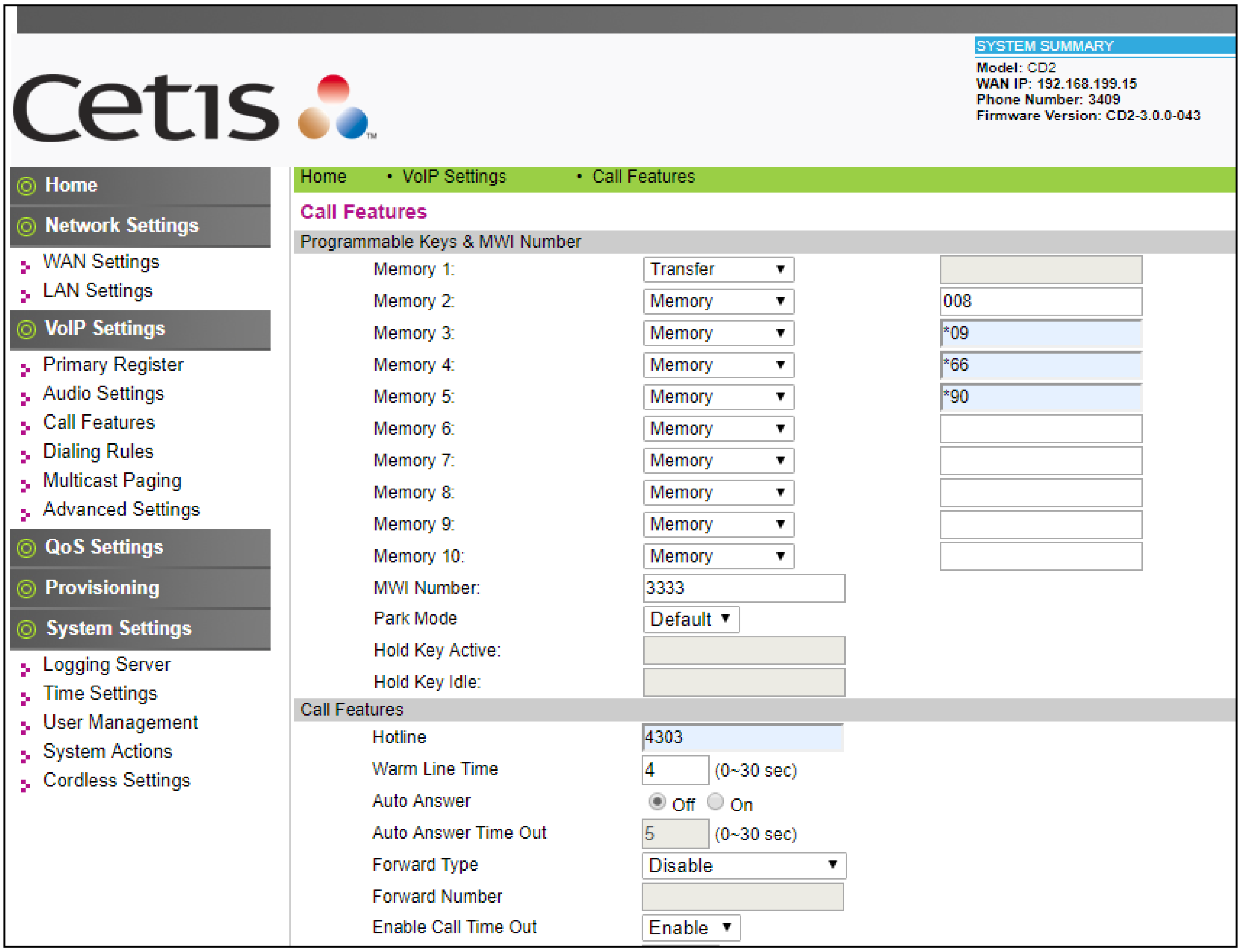
Click **Apply** button to save the changes.



Select **Audio Settings** under the **VoIP Settings** section. In this page, a customer can prioritize codec settings. The picture below shows the codecs **G.711u** and **G.729** used during the compliance test.

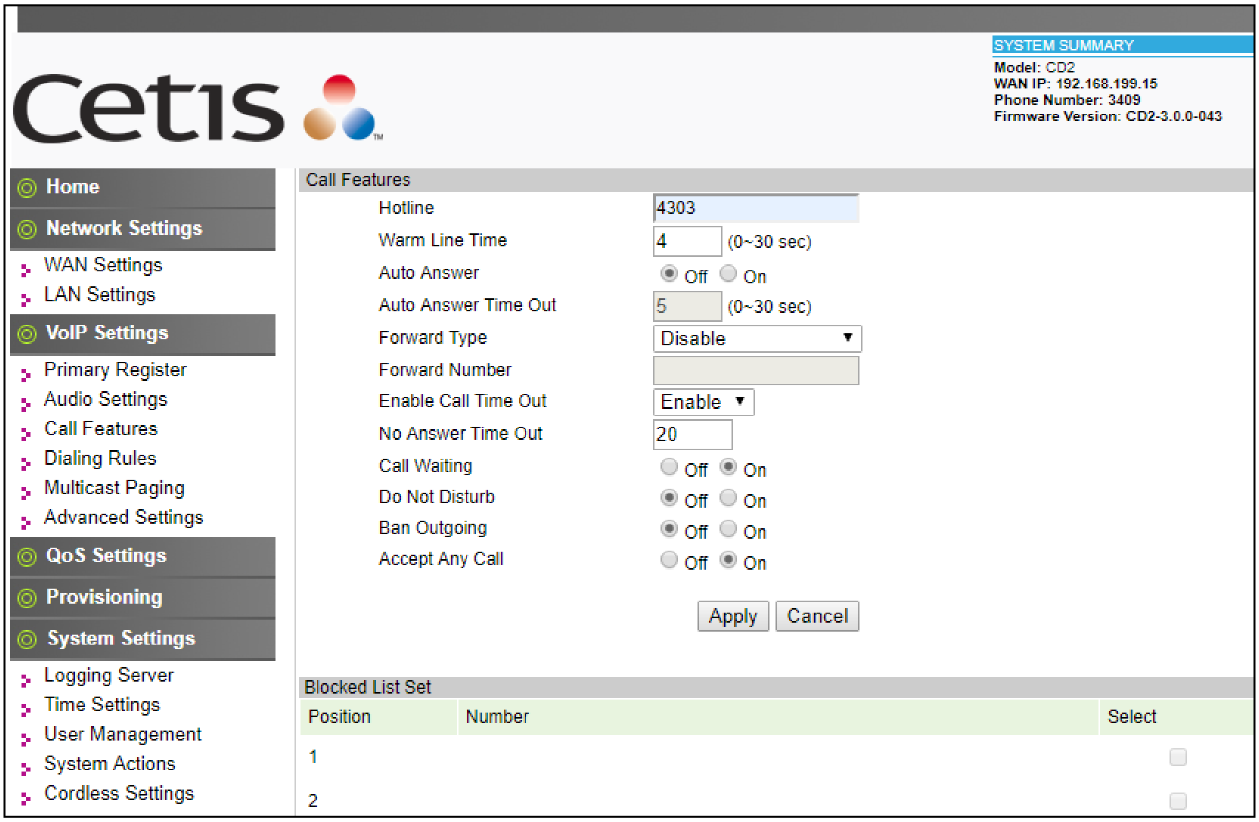


Select **Call Features** under the **VoIP Settings** section. In this page, a user can program the memory buttons, e.g. program the **Transfer** button in the **Memory 1** and other feature access codes in the other **Memory** buttons, the Cetis 9602IP telephone comes with 10 memory buttons. Enter the voicemail number of Aura® Messaging in the **MWI Number** box, this setting allows user to access to the voicemail system by press **Message** button on the 9602IP telephone.



Under the **Call Features** section in the right pane, three features (**Auto Answer**, **Do Not Disturb** and Call Forward) were tested. The configuration below shows these features at their default values.

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

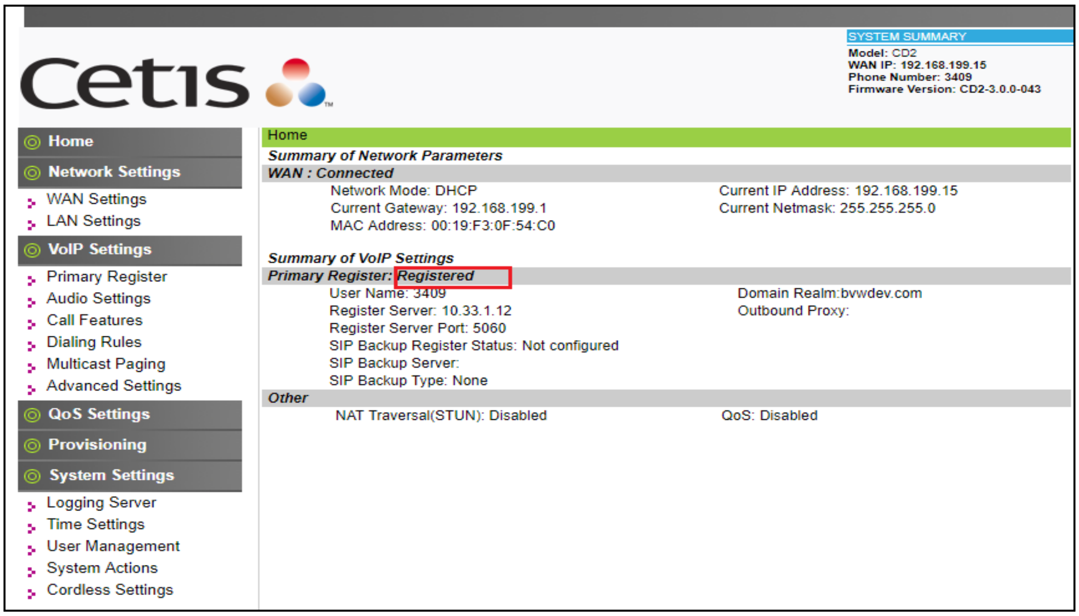


8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and the Cetis 9602IP telephone.

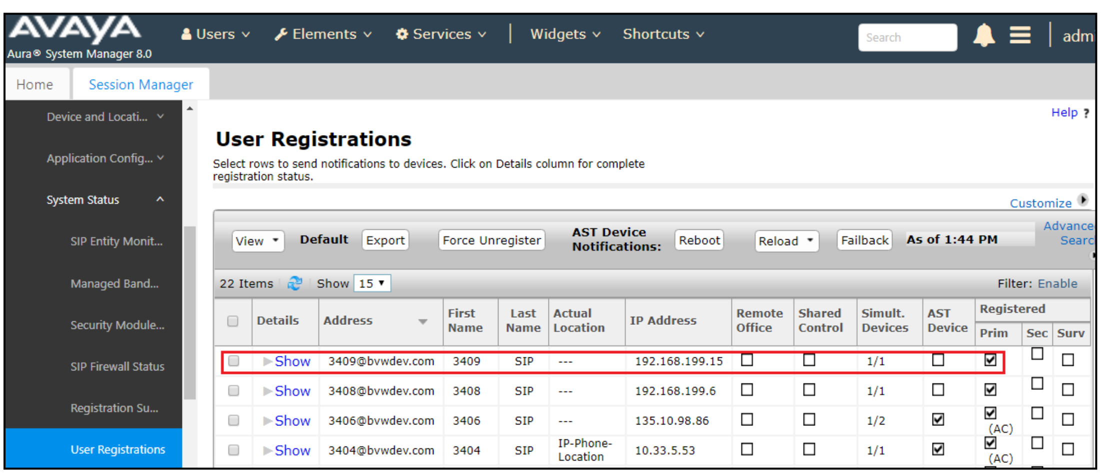
8.1. Verify Cetis SIP Telephones

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



8.2. Verify Avaya Aura® Session Manager

From web access to System Manager with appropriate credentials and navigate to **Home → Elements → Session Manager → System Status → User Registration**. Verify the Cetis 9602IP telephone is registered to Session Manager.



9. Conclusion

These Application Notes have described the administration steps required to integrate Cetis 3rd Generation 9602IP Cordless SIP 2-Line telephone release 3.0.0-043 with Avaya Aura® Session Manager Release 8.0. All of the executed test cases passed and met the objectives outlined in **Section 2.1**, any issues and observations are outlined in **Section 2.2**.

10. References

Product documentation for the Avaya Aura may be found at:
https://support.avaya.com/css/Products/

Avaya Aura Documents:

[1] Administering Avaya Aura® Communication Manager, Release 8.0, August 2018, Document Number 03-300509, Issue 1.

[2] Avaya Aura® Communication Manager Feature Description and Implementation, Release 8.0, August 2018, Document Number 555-245-205, Issue 1.

[3] Administering Avaya Aura® Session Manager, Release 8.0, Issue 1 August 2018.

[4] Administering Avaya Aura® System Manager, Release 8.0, Issue 1, August 2018.

Product documentation for the Cetis 9602IP telephone may be found at:
https://www.cetisgroup.com/support-documents.html

[5] Cetis 9602IP *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.