



Avaya Solution & Interoperability Test Lab

Application Notes for Cetus 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 Cetus 3rd Generation 9602IP Cordless SIP 2-Line Telephone with Avaya Aura® Session Manager. The Cetus 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 Cetus 3rd Generation 9602IP Cordless SIP 2-Line telephone (hereafter referred as Cetus 9602IP) with Avaya Aura® Session Manager. The Cetus 9602IP Telephone is designed for the hospitality industry. In the compliance test, Cetus 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 Cetus 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 Cetus 9602IP telephone do not utilize TLS and secure media SRTP encryption features as requested by Cetus.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Cetus 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 Cetus 9602IP telephone with Session Manager. The Cetus 9602IP telephone registered with Session Manager via SIP.

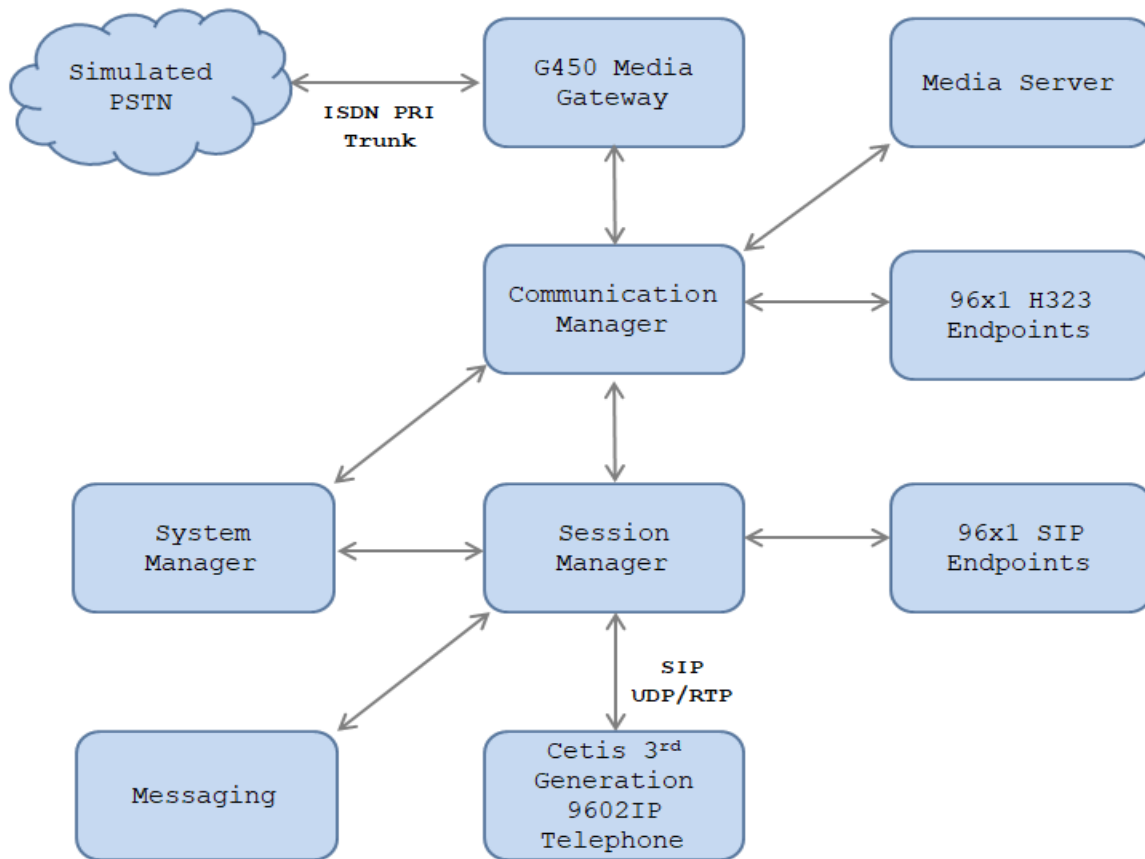


Figure 1: Test Configuration Diagram with Avaya Aura® Session Manager

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
Cetus 9602IP SIP Telephone	192.168.199.15

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

All Cetus IP phones share the same base chipset and firmware, meaning that models using the same number firmware version share the same traits and compatibility. Server registrations, SIP messaging, and call control are all the same. The different model prefixed versions are to accommodate variances in single vs. 2-line capability, cordless vs. cordless radio handsets and LCD display screen sizes. Example: C32-3.0.0-040.bin is the firmware for Cetus Cordless 2-line models including 3300IP and 9600IP.

Common Firmware on Cetus, Inc. SIP Phones

Cetus' current SIP firmware follows a naming convention based on and mated to the phone model name. The newest Cetus 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 = Cetus Corded | CD = Cetus DECT/Cordless | CT/CM = Cetus Trimline | C3 = Cetus 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 | 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

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

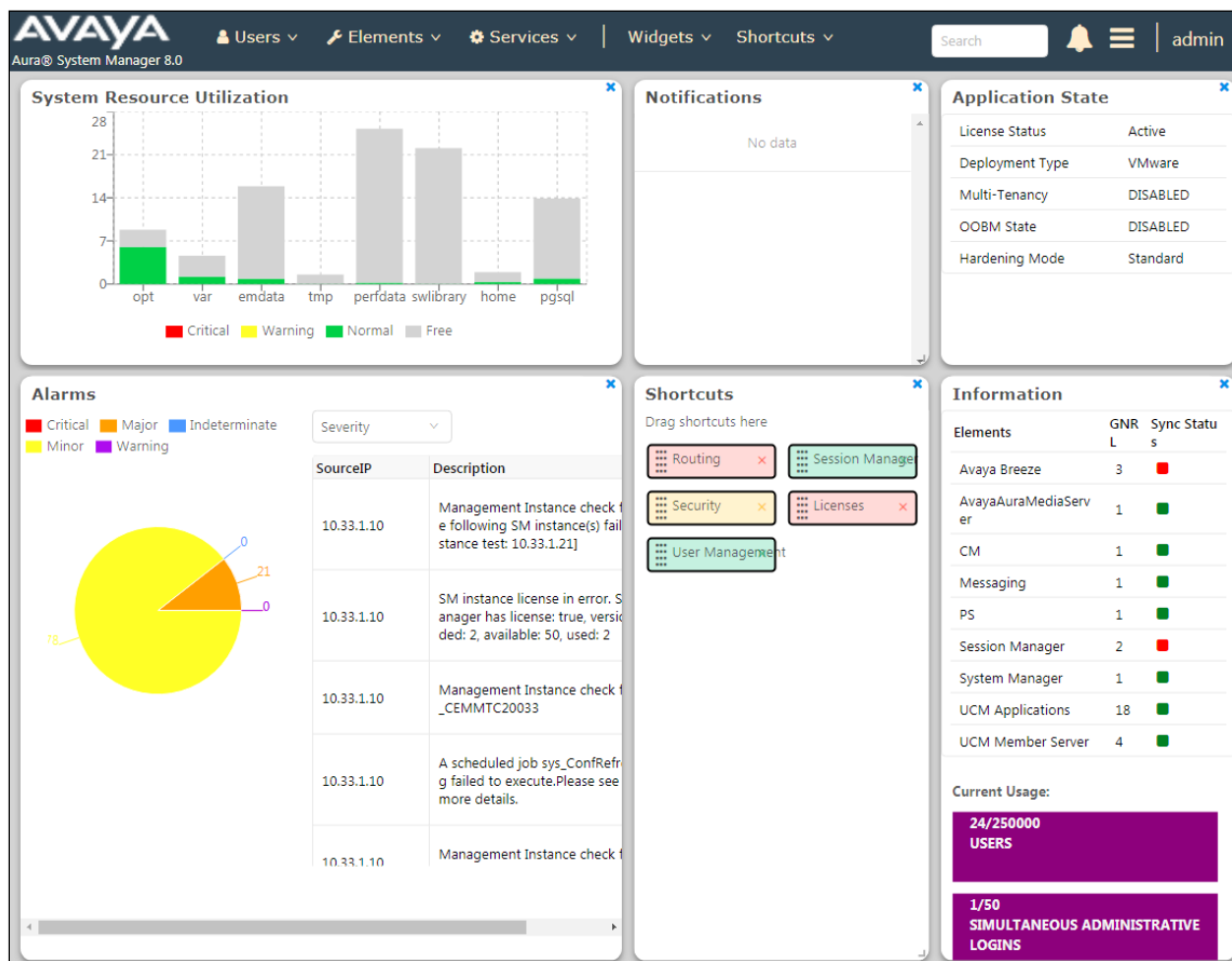
Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are **4** digits long and begin with **3**.

change dialplan analysis							Page 1 of 12		
DIAL PLAN ANALYSIS TABLE									
Location: all							Percent Full: 3		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call 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:

AddRemove

6 Items  Filter: [Enable](#)

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	bvwdev.com	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5060	UDP	bvwdev.com	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5061	TLS	bvwdev.com	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5062	TLS	bvwdev.com	<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5067	TLS	bvwdev.com	<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5080	TCP	bvwdev.com	<input type="checkbox"/>	<input type="text"/>

Select : [All](#), [None](#)

6.2. Add a SIP User

A SIP user must be created for Cetus 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

User Profile | Add Commit & Continue Commit Cancel

Identity | Communication Profile | Membership | Contacts

Basic Info

Address

LocalizedName

User Provisioning Rule:

* Last Name: Last Name (Latin Translation):

* First Name: First Name (Latin Translation):

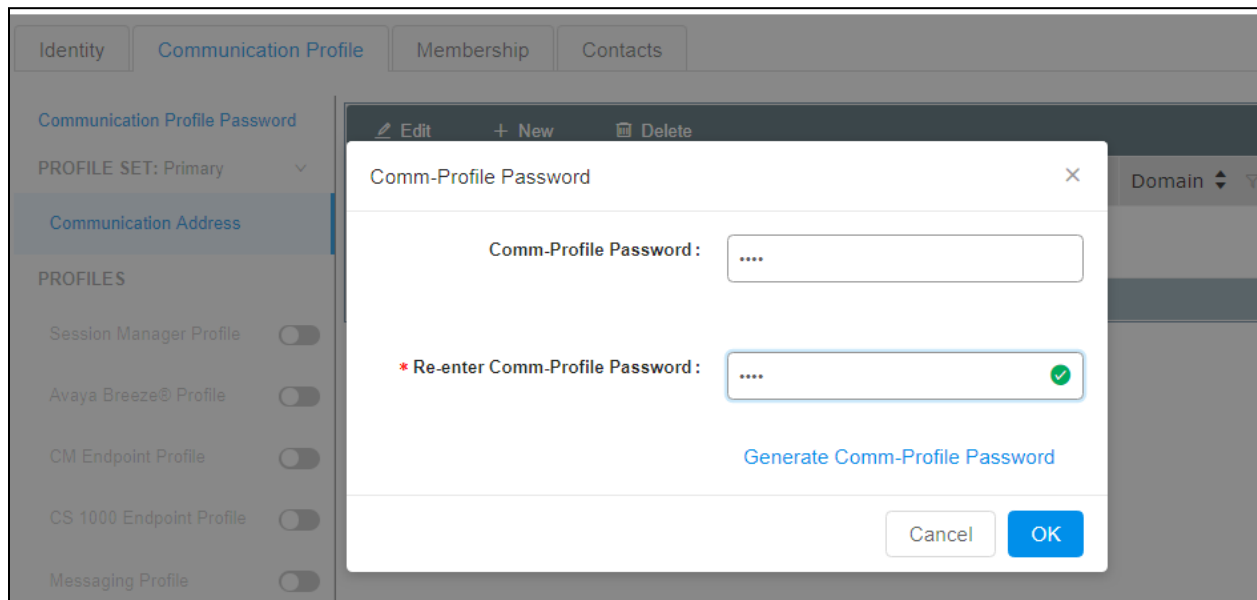
* Login Name: Middle Name:

Description: Email Address:

Password: User Type:

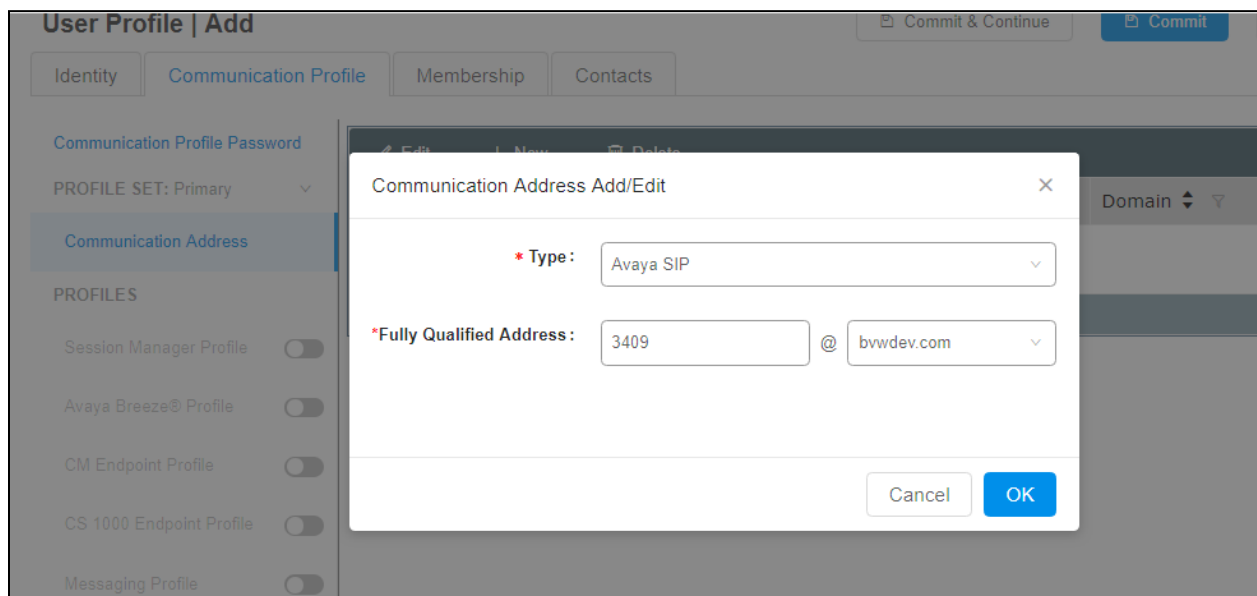
* Confirm Password: Localized Display Name:

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**.



The screenshot shows the 'Communication Profile' tab in a web interface. A modal dialog titled 'Comm-Profile Password' is open. It contains two password input fields: 'Comm-Profile Password' and '* Re-enter Comm-Profile Password'. The re-enter field has a green checkmark icon. Below the fields is a blue link 'Generate Comm-Profile Password'. At the bottom are 'Cancel' and 'OK' buttons. The background shows the 'Communication Address' section with a list of profiles and a domain dropdown.

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.



The screenshot shows the 'Communication Address' section in a web interface. A modal dialog titled 'Communication Address Add/Edit' is open. It contains a dropdown for '* Type' with 'Avaya SIP' selected. Below it is a field for '* Fully Qualified Address' with '3409' entered, followed by an '@' symbol and a dropdown for the domain with 'bvwddev.com' selected. At the bottom are 'Cancel' and 'OK' buttons. The background shows the 'Communication Profile' tab with a list of profiles and a domain dropdown.

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**.

Home / Users / Manage Users Help ?

User Profile | Add

Commit & Continue Commit Cancel

Identity **Communication Profile** Membership Contacts

Communication Profile Password

PROFILE SET: Primary

CommunicationAddress

PROFILES

- Session Manager Profile** ☒
- Avaya Breeze Profile ☐
- CM Endpoint Profile ☐
- Messaging Profile ☐
- Presence Profile ☐

SIP Registration

* Primary Session Manager: ⓘ

Secondary Session Manager: ⓘ

Survivability Server: ⓘ

Max. Simultaneous Devices:

Block New Registration When ☐ Maximum

Application Sequences

Origination Sequence:

Termination Sequence:

Emergency Calling Application Sequences

Emergency Calling Origination Sequence:

Emergency Calling Termination Sequence:

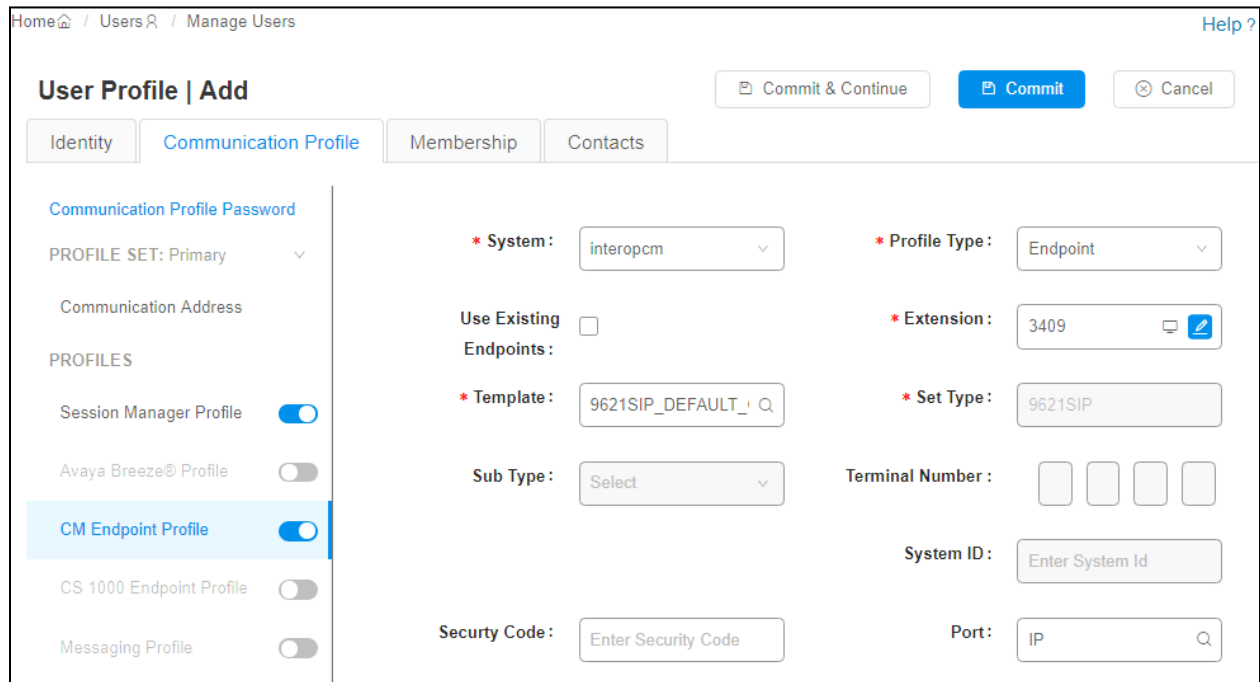
Call Routing Settings

* Home Location:

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.



Home / Users / Manage Users Help ?

User Profile | Add

[Commit & Continue](#) [Commit](#) [Cancel](#)

Identity **Communication Profile** **Membership** **Contacts**

Communication Profile Password

PROFILE SET: Primary

Communication Address

PROFILES

Session Manager Profile ☒

Avaya Breeze® Profile ☐

CM Endpoint Profile ☒

CS 1000 Endpoint Profile ☐

Messaging Profile ☐

*** System :** interopcm

*** Profile Type :** Endpoint

Use Existing Endpoints : ☐

*** Extension :** 3409

*** Template :** 9621SIP_DEFAULT_CM_8_0

*** Set Type :** 9621SIP

Sub Type : Select

Terminal Number :

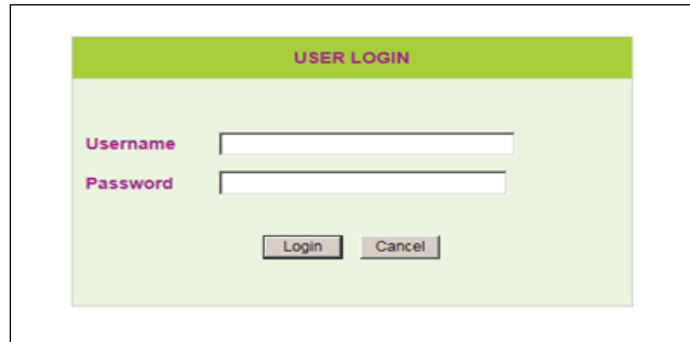
System ID : Enter System Id

Security Code : Enter Security Code

Port : IP

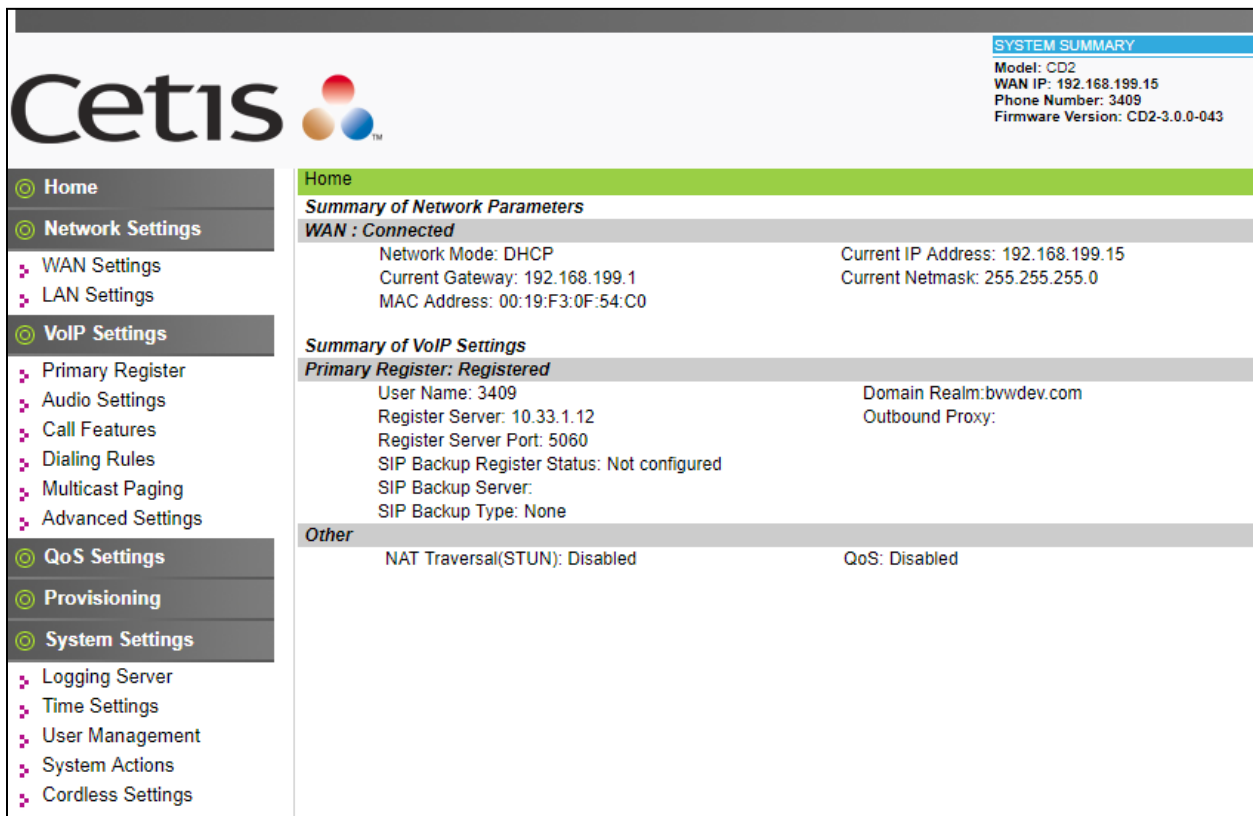
7. Configure Cetus 9602IP SIP Telephone

Access the Cetus 9602IP telephone web interface using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the Cetus 9602IP telephone. By default, DHCP is enabled on the Cetus telephones. To determine the IP address assigned to the Cetus 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.



Cetus

SYSTEM SUMMARY
Model: CD2
WAN IP: 192.168.199.15
Phone Number: 3409
Firmware Version: CD2-3.0.0-043

Home

Summary of Network Parameters

WAN : Connected

Network Mode: DHCP
Current Gateway: 192.168.199.1
MAC Address: 00:19:F3:0F:54:C0
Current IP Address: 192.168.199.15
Current Netmask: 255.255.255.0

Summary of VoIP Settings

Primary Register: Registered

User Name: 3409
Register Server: 10.33.1.12
Register Server Port: 5060
SIP Backup Register Status: Not configured
SIP Backup Server:
SIP Backup Type: None
Domain Realm: bvwdev.com
Outbound Proxy:

Other

NAT Traversal(STUN): Disabled
QoS: Disabled

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.

The screenshot displays the Cetis WAN Settings page. The left sidebar contains navigation links: Home, Network Settings (selected), VoIP Settings, QoS Settings, Provisioning, and System Settings. The main content area shows the WAN Settings page with the WAN Interface status as 'Connected'. The Basic Settings section includes Network Mode (DHCP selected), Link Mode (AUTO), Primary DNS (10.33.98.60), and Secondary DNS (8.8.4.4). The Static IP Settings section shows Static IP Address (10.33.5.202), Subnet Mask (255.255.255.0), and Default Gateway (10.33.5.1). The PPPoE Settings section shows User Account and Password fields. The 802.1X Settings section shows 802.1X (Disable), User Name, Password, and Type (multicast). The LLDP Settings section shows LLDP (Enable) and Packet Interval (120). The bottom right of the form has 'Apply' and 'Cancel' buttons.

Cetis

SYSTEM SUMMARY
Model: CD2
WAN IP: 192.168.199.15
Phone Number: 3409
Firmware Version: CD2-3.0.0-043

Home • Network Settings • WAN Settings

WAN Settings
WAN Interface: Connected

Basic Settings

Network Mode: ☒ DHCP ☐ Fixed ☐ PPPoE
Link Mode:
Primary DNS:
Secondary DNS:

Static IP Settings (Required if Network Mode is set to Static IP)

Static IP Address:
Subnet Mask:
Default Gateway:

PPPoE Settings (Required if Network Mode is set to PPPoE)

User Account:
Password:

802.1X Settings

802.1X:
User Name:
Password:
Type:

LLDP Settings

LLDP:
Packet Interval:

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.

The screenshot displays the Cetis web interface. On the left is a navigation menu with options: Home, Network Settings, WAN Settings, LAN Settings, VoIP Settings (selected), QoS Settings, Provisioning, System Settings, and Logging Server. The VoIP Settings section is expanded, showing sub-options: Primary Register (selected), Audio Settings, Call Features, Dialing Rules, Multicast Paging, and Advanced Settings. The main content area shows the 'Primary Register' configuration. At the top, it indicates 'Main Server: Registered' and 'Backup Server: Not configured'. Below this is the 'Register Server' section with the following fields: 'Use Service' (set to 'Enable'), 'Display Name' (empty), 'User Name' (3409), 'Authorization User Name' (3409), 'Password' (masked with dots), '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' (set to 'None'), and 'SIP Backup Server' (empty). A 'Protocol Control' section is partially visible at the bottom.

Cetis

SYSTEM SUMMARY
Model: CD2
WAN IP: 192.168.199.15
Phone Number: 3409
Firmware Version: CD2-3.0.0-043

Home • VoIP Settings • Primary Register

Primary Register

Main Server: Registered Backup Server: Not configured

Register Server

Use Service	Enable ▾
Display Name	
User Name	3409
Authorization User Name	3409
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

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


- **MWI Subscribe** – Select **Enable** from the dropdown menu.
- **DTMF** – Select the RFC2833 option.
- **SIP Transport** – 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.

The screenshot displays the Cetis web interface. On the left is a navigation menu with options: Home, Network Settings (selected), WAN Settings, LAN Settings, VoIP Settings, QoS Settings, Provisioning, and System Settings. The main content area is titled 'Protocol Control' and contains various configuration fields. At the top right, a 'SYSTEM SUMMARY' box shows: Model: CD2, WAN IP: 192.168.199.15, Phone Number: 3409, and Firmware Version: CD2-3.0.0-043. The 'Protocol Control' section includes fields for Register Expire (300), SIP Backup Type (None), SIP Backup Server, MWI Subscribe (Enable), Subscribe Expire (300), Local SIP Port (5060), Local RTP Port (20000), Keep Alive Packet (On), Keep Alives Period (60), DTMF (RFC2833), DTMF SIP INFO Mode (Send */#), DNS Type (NAPTR/SRV), Jitter Buffer Max (150), Anonymous Call Rejection (Off), Session Switch (Disable), Session Time (1800), PRACK (Disable), Support Update Method (Enable), Rport (Enable), SIP Transport (UDP), SIP URI (sip), and SRTP (Disable). 'Apply' and 'Cancel' buttons are at the bottom right.

Field	Value
Register Expire	300
SIP Backup Type	None
SIP Backup Server	
Protocol Control	
MWI Subscribe	Enable
Subscribe Expire	300
Local SIP Port	5060
Local RTP Port	20000
Keep Alive Packet	On
Keep Alives Period	60
DTMF	RFC2833
DTMF SIP INFO Mode	Send */#
DNS Type	NAPTR/SRV
Jitter Buffer Max	150
Anonymous Call Rejection	Off
Session Switch	Disable
Session Time (Min=90s)	1800
PRACK	Disable
Support Update Method	Enable
Rport	Enable
SIP Transport	UDP
SIP URI	sip
SRTP	Disable

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.



SYSTEM SUMMARY
 Model: CD2
 WAN IP: 192.168.199.15
 Phone Number: 3409
 Firmware Version: CD2-3.0.0-043

- 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
 - Logging Server
 - Time Settings
 - User Management
 - System Actions
 - Cordless Settings

Home • VoIP Settings • Audio Settings

Audio Settings

Sound and Volume Control

Handset	5	(1~7)
Speaker	1	(1~7)
Ringer Tone	4	(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.729 ▾	
Codec Priority 3	Not Used ▾	
Codec Priority 4	Not Used ▾	
Codec Priority 5	Not Used ▾	
Codec Priority 6	Not Used ▾	
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)	101	(95~127)
-----------------------	-----	----------

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.



SYSTEM SUMMARY

Model: CD2
 WAN IP: 192.168.199.15
 Phone Number: 3409
 Firmware Version: CD2-3.0.0-043

Home

Network Settings

VoIP Settings

QoS Settings

Provisioning

System Settings

Home • VoIP Settings • Call Features

Call Features

Programmable Keys & MWI Number

Memory 1:	Transfer ▼	
Memory 2:	Memory ▼	008
Memory 3:	Memory ▼	*09
Memory 4:	Memory ▼	*66
Memory 5:	Memory ▼	*90
Memory 6:	Memory ▼	
Memory 7:	Memory ▼	
Memory 8:	Memory ▼	
Memory 9:	Memory ▼	
Memory 10:	Memory ▼	
MWI Number:	3333	
Park Mode	Default ▼	
Hold Key Active:		
Hold Key Idle:		

Call Features

Hotline	4303
Warm Line Time	4 (0~30 sec)
Auto Answer	<input checked="" type="radio"/> Off <input type="radio"/> On
Auto Answer Time Out	5 (0~30 sec)
Forward Type	Disable ▼
Forward Number	
Enable Call Time Out	Enable ▼

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**.

Cetis

Home

Network Settings

VoIP Settings

QoS Settings

Provisioning

System Settings

WAN Settings

LAN Settings

Primary Register

Audio Settings

Call Features

Dialing Rules

Multicast Paging

Advanced Settings

Logging Server

Time Settings

User Management

System Actions

Cordless Settings

SYSTEM SUMMARY

Model: CD2
WAN IP: 192.168.199.15
Phone Number: 3409
Firmware Version: CD2-3.0.0-043

Call Features

Hotline

4303

Warm Line Time

4

(0~30 sec)

Auto Answer

☒ Off ☐ On

Auto Answer Time Out

5

(0~30 sec)

Forward Type

Disable

Forward Number

Enable Call Time Out

Enable

No Answer Time Out

20

Call Waiting

☐ Off ☒ On

Do Not Disturb

☒ Off ☐ On

Ban Outgoing

☒ Off ☐ On

Accept Any Call

☐ Off ☒ On

Apply

Cancel

Blocked List Set

Position	Number	Select
1		<input type="checkbox"/>
2		<input type="checkbox"/>

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*.

The screenshot shows the Cetis web interface. On the left is a navigation menu with options: Home, Network Settings (WAN, LAN), VoIP Settings (Primary Register, Audio, Call Features, Dialing Rules, Multicast Paging, Advanced), QoS Settings, Provisioning, and System Settings (Logging Server, Time, User Management, System Actions, Cordless). The main content area is titled 'Home' and contains a 'Summary of Network Parameters' section showing WAN as Connected with DHCP mode, gateway 192.168.199.1, netmask 255.255.255.0, and IP 192.168.199.15. Below this is a 'Summary of VoIP Settings' section where 'Primary Register' is 'Registered' (highlighted with a red box). Other details include User Name: 3409, Register Server: 10.33.1.12, and Port: 5060. A final 'Other' section shows NAT Traversal(STUN) and QoS are disabled.

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.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The left sidebar has a menu with 'User Registrations' selected. The main area is titled 'User Registrations' and includes a table of 22 items. The first row of the table is highlighted with a red box. The table columns are: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, Registered (Prim, Sec, Surv).

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered	Prim	Sec	Surv
Show	3409@bvwddev.com	3409	SIP	---	192.168.199.15			1/1			<input checked="" type="checkbox"/>		
Show	3408@bvwddev.com	3408	SIP	---	192.168.199.6			1/1			<input checked="" type="checkbox"/>		
Show	3406@bvwddev.com	3406	SIP	---	135.10.98.86			1/2	<input checked="" type="checkbox"/>	(AC)			
Show	3404@bvwddev.com	3404	SIP	IP-Phone-Location	10.33.5.53			1/1	<input checked="" type="checkbox"/>	(AC)			

9. Conclusion

These Application Notes have described the administration steps required to integrate Cetus 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 Cetus 9602IP telephone may be found at:

<https://www.cetisgroup.com/support-documents.html>

- [5] Cetus 9602IP *VoIP Phone User's Manual*.

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