



Avaya Solution & Interoperability Test Lab

Application Notes for Cetus 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 Cetus e-Series E203IP SIP cordless Telephones with Avaya Aura® Session Manager. The Cetus 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 Cetus E-Series E203IP SIP cordless Telephones (hereafter referred to as Cetus E203IP SIP Telephones) with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The Cetus E203IP SIP Telephones were designed for the hospitality industry. In the compliance test, Cetus 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 Cetus 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 Cetus 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 Cetus E203IP SIP Telephones with Session Manager.
- Calls between Cetus telephones and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetus 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 Cetus telephones.
- Proper system recovery after a restart of the Cetus 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.

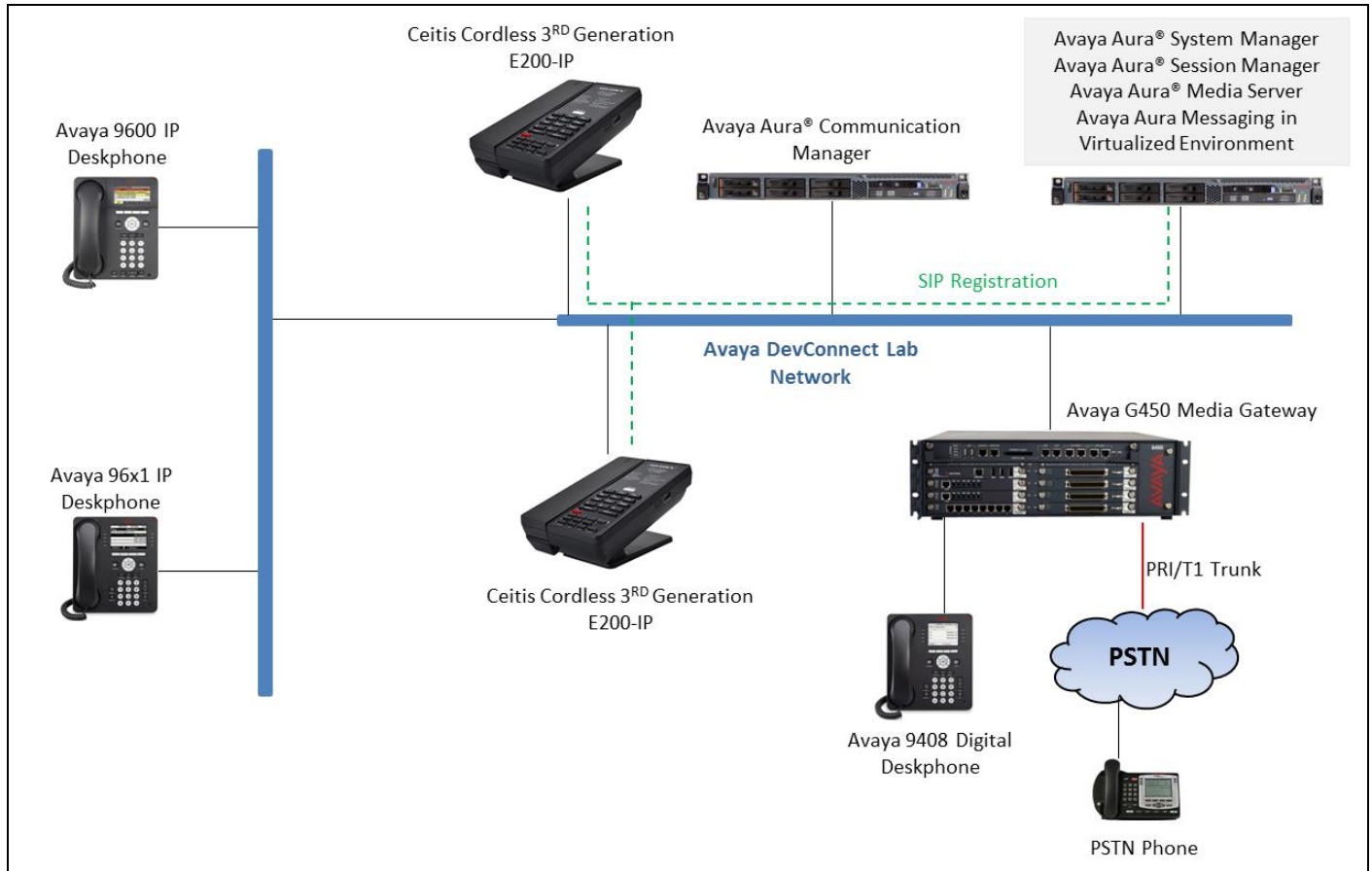


Figure 1: Cetis E203IP Telephones with Session Manager

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)	3.25
	9621G (H.323)	6.6.41
Avaya 96x0 and 96x1 Series SIP Deskphones		
	9611G	7.0.2
	9650	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 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 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 Cetus 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.

Listen Ports

TCP Failover port:

TLS Failover port:

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. Configure User

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

- Identity section
 - **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.

The screenshot shows the 'New User Profile' form in the Avaya Aura System Manager 7.0 interface. The form is titled 'New User Profile' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is selected. The form contains a 'User Provisioning Rule' dropdown menu. Below this, the 'Identity' section includes the following fields:

- * Last Name: Cetus
- Last Name (Latin Translation): Cetus
- * First Name: CC2
- First Name (Latin Translation): CC2
- Middle Name:
- Description:
- * Login Name: 3409@bvwddev.com
- User Type: Basic
- Password:
- Confirm Password:
- Localized Display Name:
- Endpoint Display Name:

The form also has a 'Commit & Continue' button and a 'Cancel' button. The top of the interface shows the Avaya logo and the text 'Aura System Manager 7.0'. The top right corner shows 'Last Logged on at January 30, 2017 9:33 AM' and a 'Log off' button.

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

Identity * Communication Profile Membership Contacts

Communication Profile

Communication Profile Password: [password field]

Confirm Password: [password field]

+ New - Delete Done Cancel

Name
Primary

Select : None

* Name: Primary

Default : ☒

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

Communication Address

+ New Edit - Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

* Fully Qualified Address: 3409 @ bwvdev.com

Add Cancel

- Session Manager Profile section
 - **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.

☒ **Session Manager Profile**

SIP Registration

* Primary Session Manager

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

Block New Registration When Maximum Registrations Active? ☐

Primary	Secondary	Maximum
13	0	13

Application Sequences

Origination Sequence

Termination Sequence

Call Routing Settings

* Home Location

Conference Factory Set

Call History Settings

Enable Centralized Call History? ☐

- 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.

☒ **CM Endpoint Profile**

* System

* Profile Type

Use Existing Endpoints ☐

* Extension [Display Extension Ranges](#)

* Template

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle

Calculate Route Pattern ☐

Sip Trunk

Enhanced Callr-Info display for 1-line phones ☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User. ☒

Override Endpoint Name and Localized Name ☒

Allow H.323 and SIP Endpoint Dual Registration ☐

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


AVAYA
Aura® System Manager 7.0

Last Logged on at January 30, 2017 9:33 AM
GO... [Log off admin](#)


Home **User Management** x

▼ User Management
Manage Users
Public Contacts
Shared Addresses
System Presence
ACLs
Communication
Profile Password
Policy

Home / Users / User Management / Manage Users






Search 


[Help ?](#)

 [Status](#)

User Management

Users

 View  Edit  New  Duplicate  Delete [More Actions](#) [Advanced Search](#)

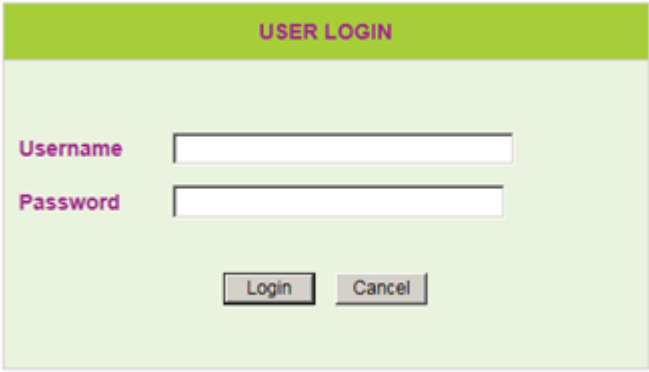
13 Items  Show All [Filter: Enable](#)

<input type="checkbox"/>	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
<input type="checkbox"/>	Cetis	CC2	Cetis, CC2	3407@bvwdev.com	3407	
<input type="checkbox"/>	Cetis	CC2	Cetis, CC2	3409@bvwdev.com	3409	
<input type="checkbox"/>	Cetis	CD2	Cetis, CD2	3408@bvdwv.com	3408	
<input type="checkbox"/>	Cetis	CD2	Cetis, CD2	3410@bvwdev.com	3410	
<input type="checkbox"/>	CS1K	1220	CS1K, 1220	1220@bvwdev.com	1220	
<input type="checkbox"/>	admin	admin	Default Administrator	admin		January 31, 2017 4:22:25 AM -05:00
<input type="checkbox"/>	SIP	3309	SIP, 3309	3309@bvwdev.com	3309	
<input type="checkbox"/>	SIP	3400	SIP, 3400	3400@bvwdev.com	3400	

7. Configure Cetus 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 Cetus E203 SIP Telephone>> in the URL field of your browser. Log in with the appropriate credentials for accessing the Cetus E203IP settings page.

Access the Cetus 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 Cetus telephone. By default, DHCP is enabled on the Cetus telephones. For this compliance test, a dynamic IP address from DHCP was assigned to the Cetus telephone. To determine the IP address assigned to the Cetus telephone, enter **47# on the telephone to hear the IP address



The screenshot shows a web interface titled "USER LOGIN" in a green header bar. Below the header, there are two input fields: "Username" and "Password", both with empty text boxes. At the bottom of the form, there are two buttons: "Login" and "Cancel". The entire form is set against a light green background.

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

Cetis

Home

Network Settings

VoIP Settings

QoS Settings

Provisioning

System Settings

SYSTEM SUMMARY

Model: CD2

WAN IP: 172.16.99.3

Phone Number: 3408

Firmware Version: CD2-3.0.0-029

Home

Summary of Network Parameters

WAN : Connected

Network Mode: DHCP

Current Gateway: 172.16.99.1

MAC Address: 00:19:F3:0F:42:A0

Current IP Address: 172.16.99.3

Current Netmask: 255.255.255.0

Summary of VoIP Settings

Primary Register: Registered

User Name: 3408

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:bwdev.com

Outbound Proxy:

Other

NAT Traversal(STUN): Disabled

QoS: Disabled

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 Cetis **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.

The screenshot displays the Cetis WAN Settings interface. On the left is a navigation menu with options: Home, Network Settings, WAN Settings, LAN Settings, VoIP Settings, QoS Settings, Provisioning, and System Settings. The main content area is titled 'WAN Settings' and shows 'WAN Interface: Connected'. It is divided into several sections: Basic Settings (Network Mode: DHCP, Link Mode: AUTO, Primary DNS: 10.10.98.60, Secondary DNS: empty), Static IP Settings (Static IP Address: 192.168.1.100, Subnet Mask: 255.255.255.0, Default Gateway: 192.168.1.1), PPPoE Settings (User Account: empty, Password: empty), 802.1X Settings (802.1X: Disable, User Name: empty, Password: empty, Type: multicast), and LLDP Settings (LLDP: Enable, Packet Interval: 120). At the bottom right are 'Apply' and 'Cancel' buttons. A 'SYSTEM SUMMARY' box in the top right corner lists: Model: CD2, WAN IP: 172.16.99.3, Phone Number: 3408, and Firmware Version: CD2-3.0.0-029.

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

WAN Settings

WAN Interface: Connected

Basic Settings

Network Mode	<input checked="" type="radio"/> DHCP <input type="radio"/> Fixed <input type="radio"/> PPPoE
Link Mode	AUTO
Primary DNS	10.10.98.60
Secondary DNS	

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

Static IP Address	192.168.1.100
Subnet Mask	255.255.255.0
Default Gateway	192.168.1.1

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

User Account	
Password	

802.1X Settings

802.1X	Disable
User Name	
Password	
Type	multicast

LLDP Settings

LLDP	Enable
Packet Interval	120

Apply Cancel

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 Transport** – 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 Subscribe** enabled and **DTMF** selected as RFC2833.

Click **Apply**.

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

Display Name

User Name

Authorization User Name

Password

Register Server Port

Register Server Address

Domain Realm

Outbound proxy

Register Expire

SIP Backup Type

SIP Backup Server

Protocol Control

MWI Subscribe

Local SIP Port

Local RTP Port

Keep Alive Packet ☐ Off ☒ On

Keep Alives Period

DTMF ☒ RFC2833 ☐ Inband ☐ SIP Info

DTMF SIP INFO Mode

DNS Type

Jitter Buffer Max

Anonymous Call Rejection ☒ Off ☐ On

Session Switch

Session Time (Min=90s)

PRACK

Support Update Method

Rport

SIP Transport


SIP URI

SRTP

Apply

Cancel

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



SYSTEM SUMMARY

Model: CD2
 WAN IP: 172.16.99.3
 Phone Number: 3408
 Firmware Version: CD2-3.0.0-029

Home

VoIP Settings

Audio Settings

Audio Settings

Sound and Volume Control

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

Codecs Settings

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


Voice VAD/CNG

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

Codec ID Settings

DTMF Payload(RFC2833)	101	(95~127)
-----------------------	-----	----------

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.



SYSTEM SUMMARY

Model: CD2
 WAN IP: 172.16.99.3
 Phone Number: 3408
 Firmware Version: CD2-3.0.0-029

- Home
- Network Settings
- VoIP Settings
 - Primary Register
 - Audio Settings
 - Call Features
 - Dialing Rules
 - Multicast Paging
 - Advanced Settings
- QoS Settings
- Provisioning
- System Settings

Home • VoIP Settings • Call Features

Call Features

Programmable Keys & MWI Touchlite

Memory 1:	Transfer	
Memory 2:	DND	
Memory 3:	Memory	
Memory 4:	Memory	
Memory 5:	Memory	
Memory 6:	Memory	
Memory 7:	Memory	
Memory 8:	Memory	
Memory 9:	Memory	
Memory 10:	Memory	
MWI Touchlite:	3333	
Park Mode	Default	
Hold Key Active:		
Hold Key Idle:		

Call Features

Hotline	
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	3304

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**.

The screenshot displays the Cetis web interface. On the left is a navigation menu with options: Home, Network Settings, VoIP Settings (expanded), Primary Register, Audio Settings, Call Features, Dialing Rules, Multicast Paging, Advanced Settings, QoS Settings, Provisioning, and System Settings. The main content area is titled 'Call Features' and contains the following settings:

Setting	Value
Hotline	[Empty text box]
Warm Line Time	4 (0~30 sec)
Auto Answer	<input type="radio"/> Off <input checked="" type="radio"/> On
Auto Answer Time Out	5 (0~30 sec)
Forward Type	Disable
Forward Number	3304
Enable Call Time Out	Enable
No Answer Time Out	20
Call Waiting	<input type="radio"/> Off <input checked="" type="radio"/> On
Do Not Disturb	<input type="radio"/> Off <input checked="" type="radio"/> On
Ban Outgoing	<input checked="" type="radio"/> Off <input type="radio"/> On
Accept Any Call	<input type="radio"/> Off <input checked="" type="radio"/> On

At the bottom right of the settings area are 'Apply' and 'Cancel' buttons. In the top right corner, a 'SYSTEM SUMMARY' box lists: Model: CD2, WAN IP: 172.16.99.3, Phone Number: 3408, and Firmware Version: CD2-3.0.0-029.

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*.

Cetis

SYSTEM SUMMARY
Model: CD2
WAN IP: 172.16.99.3
Phone Number: 3408
Firmware Version: CD2-3.0.0-029

Home • VoIP Settings

VoIP Summary

Primary Register: Registered

User Name: 3408
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: bwdev.com
Outbound Proxy:

Other

NAT Traversal(STUN): Disabled
STUN Server Address:

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.

AVAYA
Aura® System Manager 7.0

Last Logged on at March 20, 2017 8:58 AM
GO... Log off admin

Home Session Manager

Home / Elements / Session Manager / System Status / User Registrations

User Registrations

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

View: Default Force Unregister AST Device Notifications: Reboot Reload Failback As of 8:21 AM Advanced Search

12 Items Show All Filter: Enable

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
											Prim	Sec	Surv
<input type="checkbox"/>	Show	---	3400	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	3401	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	1220	CS1K	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3408@bwdev.com	CD2	Cetis	BvwDevSIL	172.16.99.3	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3309@bwdev.com	3309	SIP	BvwDevSIL	10.33.10.115	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input checked="" type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3404@bwdev.com	3404	SIP	BvwDevSIL	10.33.10.124	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3410@bwdev.com	CD2	Cetis	BvwDevSIL	172.16.99.11	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3409@bwdev.com	CC2	Cetis	BvwDevSIL	172.16.99.5	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	3403	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3406@bwdev.com	3406	SIP	BvwDevSIL	172.16.99.9	<input type="checkbox"/>	<input type="checkbox"/>	1/2	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3402@bwdev.com	3402	SIP	BvwDevSIL	10.33.10.112	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input checked="" type="checkbox"/>	<input type="checkbox"/>

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*.

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