



Avaya Solution & Interoperability Test Lab

Application Notes for Cetus 3rd Generation 3302IP Coded 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 3302IP Coded SIP 2-Line Telephone with Avaya Aura® Session Manager. The Cetus 3rd Generation 3302IP SIP 2-Line Telephone with LCD Display is corded 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 3302IP Corded SIP 2-Line telephone (hereafter referred as Cetus 3302IP) with Avaya Aura® Session Manager. The Cetus 3302IP Telephone is designed for the hospitality industry. In the compliance test, Cetus 3302IP 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 3302IP 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 3302IP 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 3302IP telephone with Session Manager.

- Calls between Cetus 3302IP telephone and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetus 3302IP telephone and the PSTN.
- G.711 and G.729 codec support.
- Transport protocol UDP.
- Proper recognition of DTMF tones, Voice Mail and Message Waiting Indicator.
- Basic telephony features, including hospitality feature, inbound/outbound, hold, mute, transfer, forward and conference.
- Use of programmable buttons on the Cetus 3302IP telephone.
- Proper system recovery after a restart of the Cetus 3302IP telephone and loss of IP connectivity.

The serviceability testing focused on verifying that the Cetus 3302IP 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.

2.3. Support

For technical support on the Cetus 3302IP telephone, contact Cetus 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 3302IP telephone with Session Manager. The Cetus 3302IP telephone registered with Session Manager via SIP.

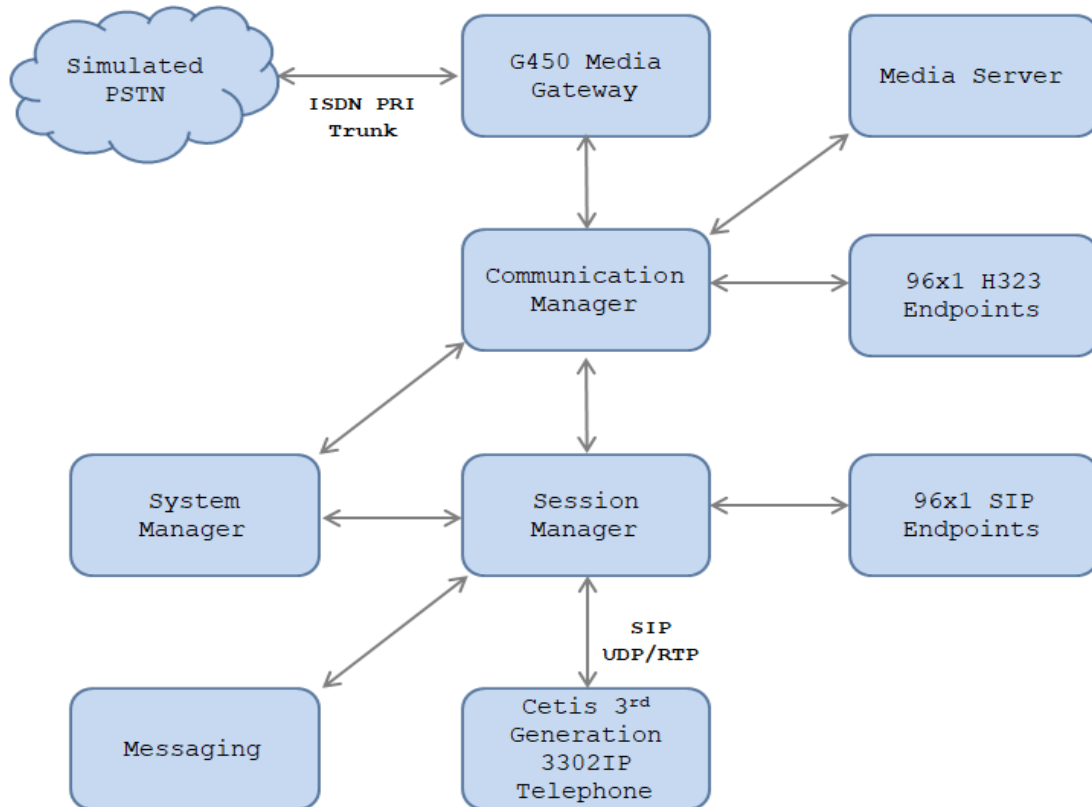


Figure 1: Test Configuration Diagram with Avaya Aura® Session Manager

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 3302IP SIP Telephone	192.168.199.6

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® Server Media running on Virtualized Environment	8.0.1 Build 8.0.0.117
Avaya Aura® Messaging running on Virtualized Environment	7.1 SP1
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 3302IP Corded SIP 2-Line Telephone	Firmware Version C32-3.0.0-043

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, corded vs. cordless radio handsets and LCD display screen sizes. Example: C32-3.0.0-040.bin is the firmware for Cetis Corded 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

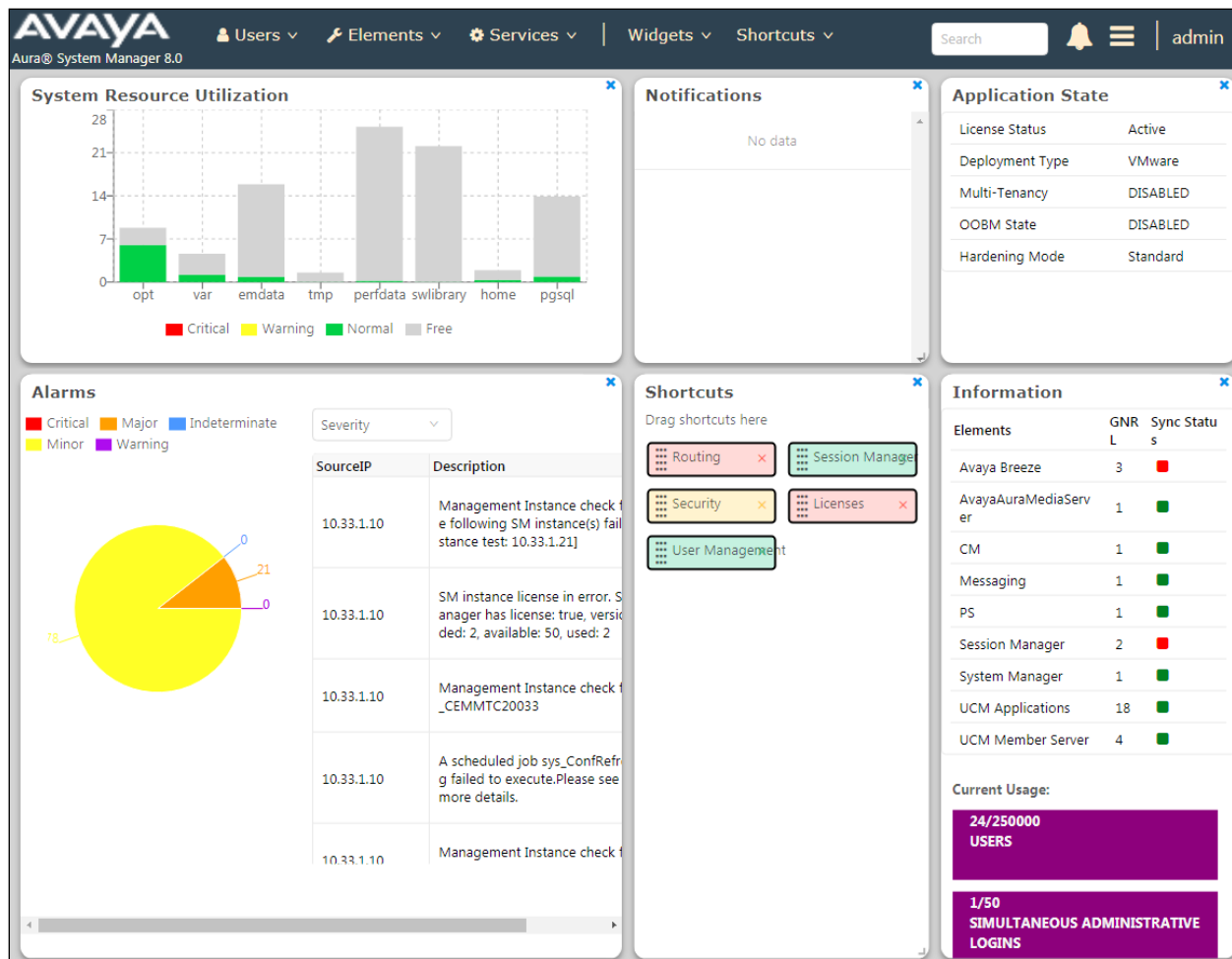
Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are **4** digits long and begin with **3**.

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 Cetus 3302IP 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 3302IP 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:

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 3302IP 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 **3408@bvwddev.com**
- **User Type** Select **Basic** from the drop down list
- **Password and Confirm Password** Enter and confirm a password

Home / Users / Manage Users Help ?

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):

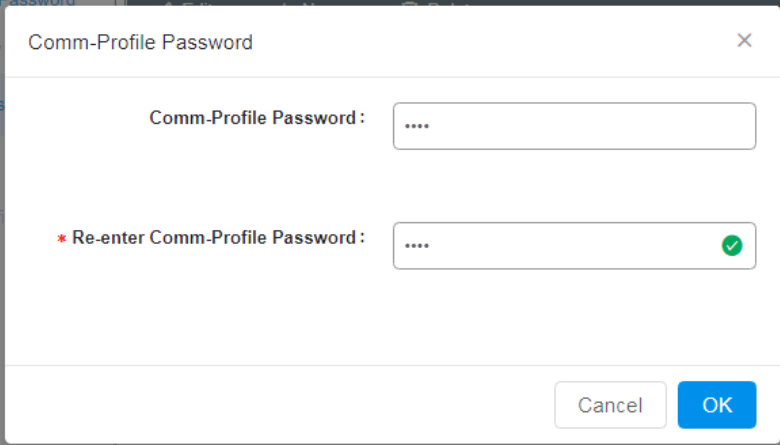
* Login Name: Middle Name:

Description: Email Address:

Password: User Type:

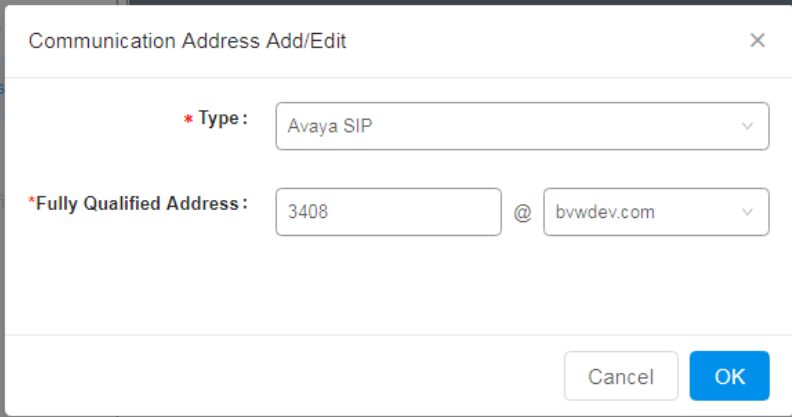
* Confirm Password: Localized Display Name:

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 3302IP telephone later in **Section 7**.



The screenshot shows a web application interface with tabs: Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active. 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 second field has a green checkmark icon to its right. At the bottom of the dialog are 'Cancel' and 'OK' buttons. The background shows a sidebar with 'PROFILES' and a list of profile types including 'Session Manager Profile', 'Avaya Breeze Profile', 'CM Endpoint Profile', 'Messaging Profile', and 'Presence Profile'.

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 same web application interface. A modal dialog titled 'Communication Address Add/Edit' is open. It contains two main fields: '* Type' with a dropdown menu showing 'Avaya SIP', and '* Fully Qualified Address' with a text input containing '3408' and a domain dropdown menu showing 'bvwdev.com'. At the bottom of the dialog are 'Cancel' and 'OK' buttons. The background shows the same sidebar and profile list as the previous screenshot.

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 **3408**
- **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

CommunicationAddress

PROFILES

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

*** System :** interopcm *** Profile Type :** Endpoint

Use Existing Endpoints : ☐ *** Extension :** 3408

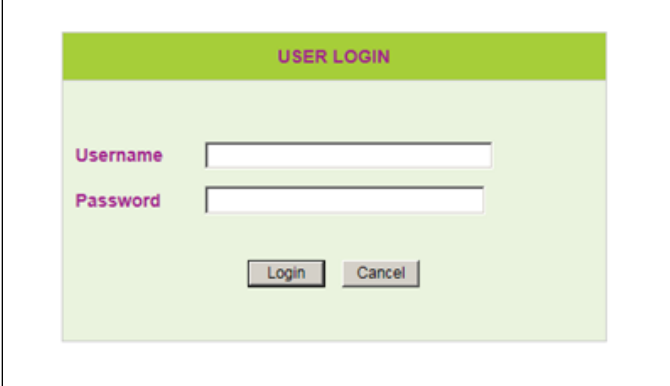
*** Template :** 9621_DEFAULT_CM_8_0 *** Set Type :** 9621

Security Code : Enter Security Code **Port :** IP

Voice Mail Number ... **Preferred Handle :** Select

7. Configure Cetis 3302IP SIP Telephone

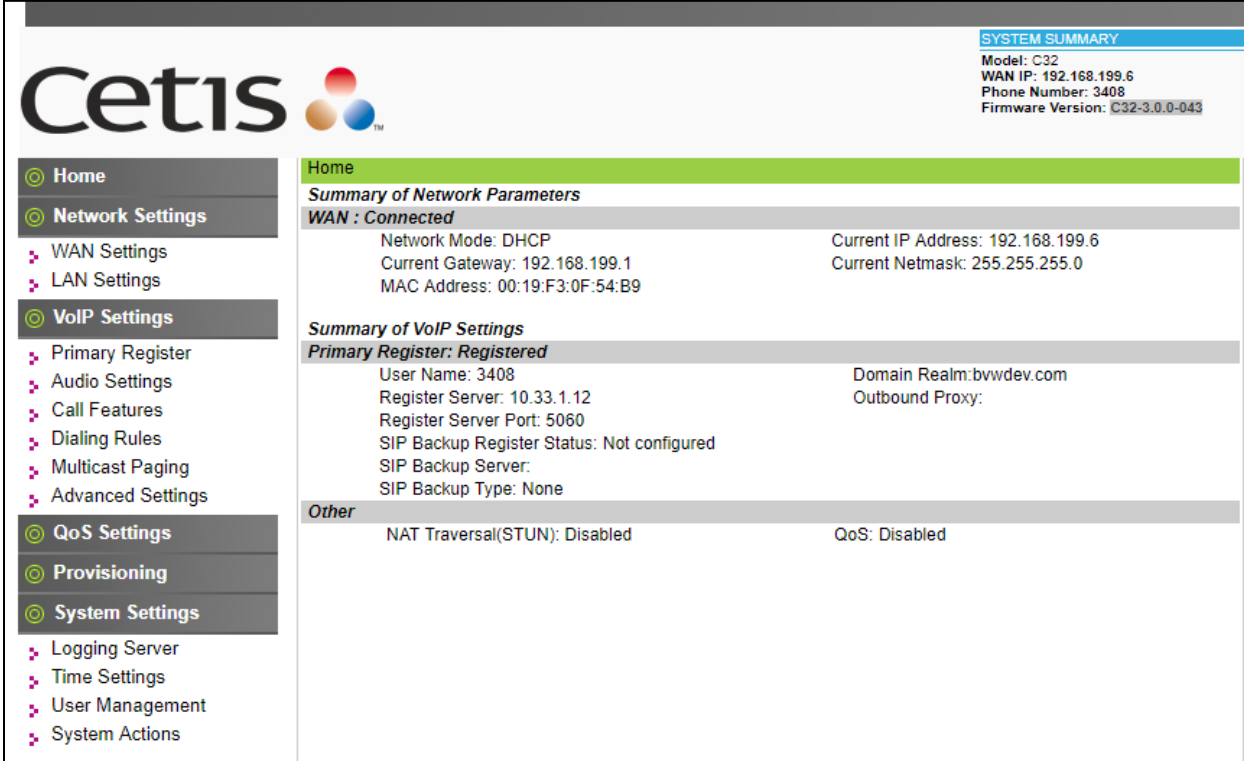
Access the Cetis 3302IP telephone web interface using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the Cetis 3302IP telephone. By default, DHCP is enabled on the Cetis telephones. To determine the IP address assigned to the Cetis telephone, enter **47# on the telephone to hear the IP address. Default **Username/Password** is admin/admin.



The image shows a web form titled "USER LOGIN". It has a light green header with the title in red. Below the header, there are two input fields: "Username" and "Password", both with red labels. Below the fields are two buttons: "Login" and "Cancel".

7.1. Network Settings

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



The screenshot shows the Cetis 3302IP SIP Telephone web interface. The top left features the Cetis logo. The top right has a "SYSTEM SUMMARY" box with the following information: Model: C32, WAN IP: 192.168.199.6, Phone Number: 3408, and Firmware Version: C32-3.0.0-043. The left sidebar contains a navigation menu with the following items: Home, Network Settings (selected), VoIP Settings, QoS Settings, Provisioning, and System Settings. Under Network Settings, there are sub-items: WAN Settings, LAN Settings, Primary Register, Audio Settings, Call Features, Dialing Rules, Multicast Paging, and Advanced Settings. The main content area displays the "Summary of Network Parameters" section, which is titled "WAN : Connected". It shows the following information: Network Mode: DHCP, Current Gateway: 192.168.199.1, MAC Address: 00:19:F3:0F:54:B9, Current IP Address: 192.168.199.6, and Current Netmask: 255.255.255.0. Below this is the "Summary of VoIP Settings" section, which is titled "Primary Register: Registered". It shows the following information: 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: bvwdev.com, and Outbound Proxy. At the bottom, there is an "Other" section with the following information: NAT Traversal(STUN): Disabled and 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 web interface for WAN Settings. The left sidebar contains navigation links: Home, Network Settings (selected), VoIP Settings, QoS Settings, Provisioning, and System Settings. Under Network Settings, WAN Settings is selected. The main content area shows the WAN Interface as Connected. The Basic Settings section has Network Mode set to DHCP, Link Mode set to AUTO, Primary DNS set to 10.10.98.60, and Secondary DNS set to 8.8.4.4. The Static IP Settings section is collapsed. The PPPoE Settings section is also collapsed. The 802.1X Settings section has 802.1X set to Disable, User Name and Password fields are empty, and Type is set to multicast. The LLDP Settings section has LLDP set to Enable and Packet Interval set to 120. At the bottom right are Apply and Cancel buttons. A SYSTEM SUMMARY box in the top right corner lists: Model: C32, WAN IP: 192.168.199.6, Phone Number: 3408, and Firmware Version: C32-3.0.0-043.

SYSTEM SUMMARY	
Model:	C32
WAN IP:	192.168.199.6
Phone Number:	3408
Firmware Version:	C32-3.0.0-043

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	8.8.4.4

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

Static IP Address	10.33.5.200
Subnet Mask	255.255.255.0
Default Gateway	10.33.5.1

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

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 for VoIP settings. The top right corner shows a 'SYSTEM SUMMARY' box with the following details: Model: C32, WAN IP: 192.168.199.6, Phone Number: 3408, and Firmware Version: C32-3.0.0-043. The left sidebar contains a navigation menu with categories: Home, Network Settings (WAN, LAN), VoIP Settings (Primary Register, Audio, Call Features, Dialing Rules, Multicast Paging, Advanced Settings), QoS Settings, Provisioning, and System Settings (Logging Server, Time Settings, User Management, System Actions). The main content area is titled 'Primary Register' and includes a breadcrumb trail: Home > VoIP Settings > Primary Register. Below the title, it indicates 'Main Server: Registered' and 'Backup Server: Not configured'. The 'Register Server' section contains the following fields: Use Service (dropdown set to 'Enable'), Display Name (empty), User Name (3408), Authorization User Name (3408), Password (masked with '****'), 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 (dropdown set to 'None'), and SIP Backup Server (empty). The 'Protocol Control' section includes: MWI Subscribe (dropdown set to 'Enable'), Subscribe Expire (300), Local SIP Port (5060), Local RTP Port (20000), Keep Alive Packet (radio buttons for 'Off' and 'On', with 'On' selected), Keep Alives Period (60), and DTMF (radio buttons for 'RFC2833', 'Inband', and 'SIP Info', with 'RFC2833' selected).

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 C32 web interface. The top right corner features a 'SYSTEM SUMMARY' box with the following information: Model: C32, WAN IP: 192.168.199.6, Phone Number: 3408, and Firmware Version: C32-3.0.0-043. The left sidebar contains a navigation menu with the following items: Home, Network Settings, VoIP Settings (selected), QoS Settings, Provisioning, and System Settings. Under VoIP Settings, the following sub-items are listed: Primary Register, Audio Settings, Call Features, Dialing Rules, Multicast Paging, Advanced Settings, Logging Server, Time Settings, User Management, and System Actions. The main content area is titled 'Protocol Control' and contains the following settings:

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

At the bottom of the settings area, there are 'Apply' and 'Cancel' buttons.

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: C32
 WAN IP: 192.168.199.6
 Phone Number: 3408
 Firmware Version: C32-3.0.0-043

Home

Network Settings

VoIP Settings

QoS Settings

Provisioning

System Settings

Home • VoIP Settings • Audio Settings

Audio Settings

Sound and Volume Control

Handset	7	(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.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)
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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 3302IP 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 3302IP telephone.



SYSTEM SUMMARY
 Model: C32
 WAN IP: 192.168.199.6
 Phone Number: 3408
 Firmware Version: C32-3.0.0-043

- 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

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 ▼	*91
Memory 7:	Memory ▼	3390#*70
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)

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**.

The screenshot displays the Cetis C32 web interface. The left sidebar contains a navigation menu with the following items: Home, Network Settings (with sub-items WAN Settings and LAN Settings), VoIP Settings (with sub-items Primary Register, Audio Settings, Call Features, Dialing Rules, Multicast Paging, and Advanced Settings), QoS Settings, Provisioning, and System Settings (with sub-items Logging Server, Time Settings, User Management, and System Actions). The main content area is titled 'Call Features' and includes the following settings:

- Hotline: 4303
- Warm Line Time: 4 (0~30 sec)
- Auto Answer: ☒ Off ☐ On
- Auto Answer Time Out: 5 (0~30 sec)
- Forward Type: Disable (dropdown menu)
- Forward Number: 3401
- Enable Call Time Out: Enable (dropdown menu)
- No Answer Time Out: 20
- Call Waiting: ☐ Off ☒ On
- Do Not Disturb: ☒ Off ☐ On
- Ban Outgoing: ☒ Off ☐ On
- Accept Any Call: ☐ Off ☒ On

Below the Call Features section is the 'Display Settings' section, which includes:

- LCD Display: ☒ Enable ☐ Disable
- Greeting Message: (empty text field)

At the bottom of the settings area are 'Apply' and 'Cancel' buttons. Below this is a 'Blocked List Set' section with a table header:

Position	Number	Select
----------	--------	--------

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and the Cetis 3302IP 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*.

Cetis

SYSTEM SUMMARY
Model: C32
WAN IP: 192.168.199.6
Phone Number: 3408
Firmware Version: C32-3.0.0-043

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

Home
Summary of Network Parameters
WAN : Connected
Network Mode: DHCP
Current Gateway: 192.168.199.1
MAC Address: 00:19:F3:0F:54:B9
Current IP Address: 192.168.199.6
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: bvwddev.com
Outbound Proxy:

Other
NAT Traversal(STUN): Disabled
QoS: 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 3302IP telephone is registered to Session Manager.

AVAYA
Aura® System Manager 8.0

Users Elements Services Widgets Shortcuts Search admin

Home **Session Manager**

Device and Locati...
Application Config...
System Status
SIP Entity Monit...
Managed Band...
Security Module...
SIP Firewall Status
Registration Su...
User Registrations

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

View: Default Export Force Unregister AST Device Notifications: Reboot Reload Fallback As of 1:55 PM Customize

22 Items Show 15 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	3409@bvwddev.com	3409	SIP	---	192.168.199.15	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/>	Show	3408@bvwddev.com	3408	SIP	---	192.168.199.6	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3406@bvwddev.com	3406	SIP	---	135.10.98.86	<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	3404@bvwddev.com	3404	SIP	IP-Phone-Location	10.33.5.53	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>

9. Conclusion

These Application Notes have described the administration steps required to integrate Cetus 3rd Generation 3302IP Corded 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 3302IP telephone may be found at:

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

- [5] Cetus 3300IP *VoIP Phone User's Manual*.

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