



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Biamp Tesira SVC-2 and Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> SIP Enablement Services – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Biamp Tesira SVC-2 which was compliance tested with Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> SIP Enablement Services.

The overall objective of the interoperability compliance testing is to verify Biamp Tesira SVC-2 functionalities in an environment comprised of Avaya Aura<sup>®</sup> Communication Manager, Avaya Aura<sup>®</sup> SIP Enablement Services, various Avaya H.323 and SIP IP Telephones.

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 procedures for configuring Biamp Tesira SVC-2 which was compliance tested with Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> SIP Enablement Services.

The Tesira SVC-2 enables conferencing over VoIP directly from Tesira SERVER-IO, with two channels of VoIP interface per card. Tesira SVC-2 allows Tesira SERVER-IO to connect directly to IP-based phone systems and eliminate the need for VoIP adapters. Used in conjunction with SEC-4 4-Channel Wideband Acoustic Echo Cancellation Input Cards and STC-2 Dual-Channel Telephone Interface Cards, the Tesira SVC-2 makes Tesira SERVER-IO a powerful, flexible, and affordable telephone conferencing product available. Combined with the STC-2 Card, the Tesira SVC-2 makes it possible to create redundancies within a conferencing system for multi-point conferences and/or back-up to VoIP lines. Up to 6 Tesira SVC-2 can be installed into a single Tesira SERVER-IO unit.

These Application Notes assume that Communication Manager and SIP Enablement Services are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. In these Application Notes, the following topics will be described:

- Communication Manager – SIP trunk configuration
- SIP Enablement Services – User configuration
- Biamp Tesira SVC-2 – User registration

For further details on Biamp Tesira SVC-2 configuration steps not covered in this document, consult [3].

## 2. General Test Approach and Test Results

All test cases were performed manually. The general approach was to place various types of calls to and from Biamp Tesira SVC-2. Biamp Tesira SVC-2 operations such as inbound calls, outbound calls, hold, and Biamp Tesira SVC-2 interactions with SIP Enablement Services, Communication Manager, and Avaya SIP and H.323 telephones were verified. For serviceability testing, failures such as cable pulls and resets were applied. All test cases passed.

### 2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing evaluated the interoperability between Biamp Tesira SVC-2, SIP Enablement Services, and Communication Manager. The serviceability testing introduced failure scenarios to see if Biamp Tesira SVC-2 could resume after failure.

### 2.2. Test Results

All test cases passed.

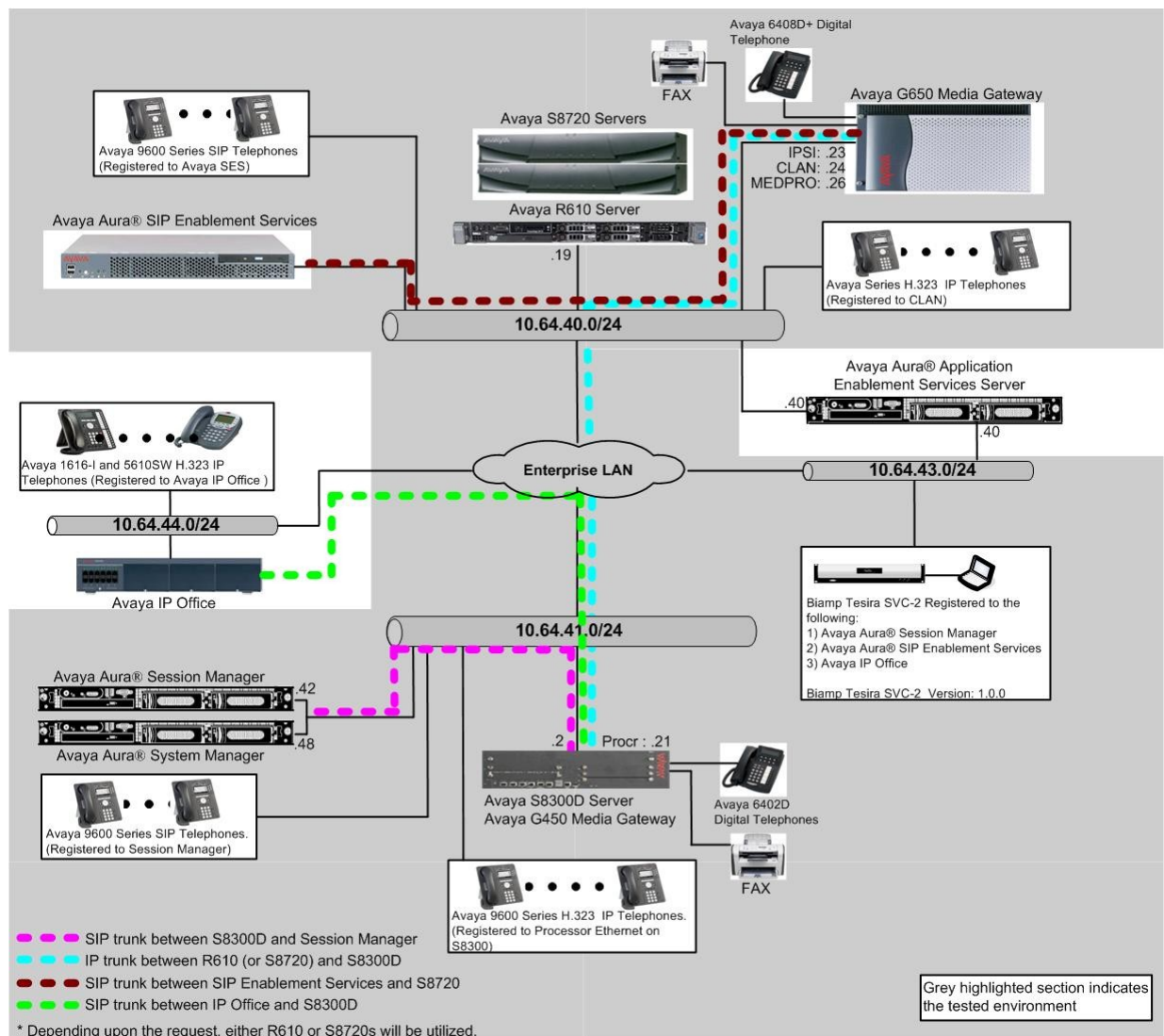
### 2.3. Support

Technical support for Biamp Tesira SVC-2 solution can be obtained by contacting Biamp at:

- <http://www.biamp.com/support/index.aspx>
- (800)-826-1457

### 3. Reference Configuration

**Figure 1** illustrates a sample configuration consisting of an Avaya S8720 Servers, an Avaya G650 Media Gateway, a SIP Enablement Services server, and Biamp Tesira SVC-2. Avaya S8300D Server with an Avaya G450 Media Gateway was included in the test to provide an inter-switch scenario. For completeness, Avaya 9600 Series H.323 IP Telephones, Avaya 9600 Series SIP IP Telephones, and Avaya 6400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between the Biamp Tesira SVC-2 and Avaya SIP, H.323, and digital telephones.



**Figure 1: Test Configuration of Biamp Tesira SVC-2**

## 4. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment		Software/Firmware
Avaya S8720 Servers with Avaya G650 Media Gateway		Avaya Aura® Communication Manager 5.2.1 (R015x.02.1.016.4)
Avaya Aura® SIP Enablement Services		SES-02.1.016.4-SP3b
Avaya S8300D Server with Avaya G450 Media Gateway ( <i>used for inter-switch test scenarios</i> )		Avaya Aura® Communication Manager 6.0.1(R016x.00.1.510.1) w/ patch 00.1.510.1-19303
Avaya Aura® System Manager		6.1.5.0
Avaya Aura® Session Manager		6.1.5.0
Avaya 9600 Series IP Telephones		
	9620 (H.323)	3.1
	9630 (H.323)	3.1
	9650 (H.323)	3.1
Avaya 9600 Series SIP Telephones		
	9620 (H.323)	2.64
	9630 (H.323)	2.64
	9650 (H.323)	2.64
Avaya 6408D+ Digital Telephone		-
Biamp Tesira SVC-2		1.0.0
Biamp Tesira		1.0.0
Linux		2.6.32.28-BIAMP

## 5. Configure Avaya Aura® Communication Manager

This section describes the procedure for setting up a SIP trunk between Communication Manager and SIP Enablement Services. The steps include setting up an IP codec set, an IP network region, IP node name, a signaling group, a trunk group, and a SIP station. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface. Biamp Tesira SVC-2 and other SIP telephones are configured as off-PBX telephones in Communication Manager.

## 5.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses

display system-parameters customer-options		Page 1 of 11
OPTIONAL FEATURES		
G3 Version: V15	Software Package: Standard	
Location: 1	RFA System ID (SID): 1	
Platform: 6	RFA Module ID (MID): 1	
		USED
Platform Maximum Ports:	44000	254
Maximum Stations:	36000	118
Maximum XMOBILE Stations:	0	0
Maximum Off-PBX Telephones - EC500:	50	1
Maximum Off-PBX Telephones - OPS:	100	7
Maximum Off-PBX Telephones - PBFMC:	0	0
Maximum Off-PBX Telephones - PVFMC:	0	0
Maximum Off-PBX Telephones - SCCAN:	0	0

On **Page 2** of the form, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

display system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	100	39
Maximum Concurrently Registered IP Stations:	18000	3
Maximum Administered Remote Office Trunks:	0	0
Maximum Concurrently Registered Remote Office Stations:	0	0
Maximum Concurrently Registered IP eCons:	0	0
Max Concur Registered Unauthenticated H.323 Stations:	5	0
Maximum Video Capable H.323 Stations:	5	0
Maximum Video Capable IP Softphones:	5	0
Maximum Administered SIP Trunks:	100	40
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0
Maximum Number of DS1 Boards with Echo Cancellation:	0	0
Maximum TN2501 VAL Boards:	10	1
Maximum Media Gateway VAL Sources:	0	0
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	1
Maximum Number of Expanded Meet-me Conference Ports:	0	0

## 5.2. IP Codec Set

This section describes the steps for administering a codec set in Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and SIP Enablement Services. Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 5.3** when configuring an IP network region to specify which codec sets may be used within and between network regions.

```

change ip-codec-set 1
Page 1 of 2

IP Codec Set

Codec Set: 1

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt    Size (ms)
1: G.711MU      n          2          20
2:

```

### 5.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and SIP Enablement Services. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- **Authoritative Domain** – Enter the appropriate name for the Authoritative Domain. Set to the appropriate domain. During the compliance test, the authoritative domain is set to **avaya.com**. This should match the SIP Domain value on SIP Enablement Services, in **Section 6.1**.
- **Intra-region IP-IP Direct Audio** – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SIP Enablement Services in the same IP network region. The default value for this field is **yes**.
- **Codec Set** – Set the codec set number as provisioned in **Section 5.2**.
- **Inter-region IP-IP Direct Audio** – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SIP Enablement Services in different IP network regions. The default value for this field is **yes**.

```

change ip-network-region 1
Page 1 of 19

IP NETWORK REGION

Region: 1
Location: Authoritative Domain: avaya.com
Name:
MEDIA PARAMETERS
Codec Set: 1
UDP Port Min: 2048
UDP Port Max: 3329
Intra-region IP-IP Direct Audio: yes
Inter-region IP-IP Direct Audio: yes
IP Audio Hairpinning? n
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
RTCP Reporting Enabled? y
RTCP MONITOR SERVER PARAMETERS
Use Default Server Parameters? y
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5
AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
RSVP Enabled? n

```

## 5.4. Configure IP Node Name

This section describes the steps for setting IP node name for SIP Enablement Services in Communication Manager. Enter the **change node-names ip** command, and add a node name for SIP Enablement Services along with its IP address.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
CLAN	10.64.40.24	
CLAN-AES	10.64.40.25	
G450	10.64.41.21	
MEDPRO	10.64.40.26	
SES	10.64.40.41	
VAL	10.64.40.45	
default	0.0.0.0	

## 5.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for signaling between Communication Manager and SIP Enablement Services. Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tls**
- **Near-end Node Name** - Set to **CLAN** as displayed in **Section 5.4**.
- **Far-end Node Name** - Set to the SIP Enablement Services name configured in **Section 5.4**.
- **Far-end Network Region** - Set to the region configured in **Section 5.3**.
- **Far-end Domain** - Set to **avaya.com**. This should match the SIP Domain value in **Section 6.1**.

add signaling-group 201		Page 1 of 1
Group Number: 201	Group Type: sip	
	Transport Method: tls	
IMS Enabled? n		
Near-end Node Name: CLAN	Far-end Node Name: SES	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain: avaya.com		
	Bypass If IP Threshold Exceeded? n	
D@F over IP: rtp-payload	Direct IP-IP Audio Connections? y	
Session Establishment Timer(min): 3	IP Audio Hairpinning? n	
Enable Layer 3 Test? n	Direct IP-IP Early Media? n	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

## 5.6. Configure SIP Trunk

This section describes the steps for administering a trunk group in Communication Manager for trunking between Communication Manager and SIP Enablement Services. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group and configure the following:

- **Group Type** – Set the Group Type field to **sip**.
- **Group Name** – Enter a descriptive name.
- **TAC (Trunk Access Code)** – Set to any available trunk access code.
- **Signaling Group** – Set to the Group Number field value configured in **Section 5.5**.
- **Number of Members** – Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

**Note:** Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.

```
add trunk-group 201                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 201                                     Group Type: sip          CDR Reports: y
Group Name: to SES                                     COR: 1                TN: 1                TAC: 116
Direction: two-way                                     Outgoing Display? y    Night Service:
Dial Access? n
Queue Length: 0
Service Type: tie                                     Auth Code? n
                                                Signaling Group: 201
                                                Number of Members: 10
```



## 5.7. Configure SIP Endpoint

This section describes the steps for administering OPS stations in Communication Manager and associating the OPS station extensions with the telephone numbers of Biamp Tesira SVC-2.

Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. The following fields were configured for the compliance test.

- **Type** – Set to **9630SIP**.
- **Name** – Enter a descriptive name

Repeat this step as necessary to configure additional SIP endpoint extensions for Biamp Tesira SVC-2.

add station 28003		Page 1 of 6
STATION		
Extension: 28003	Lock Messages? n	BCC: 0
Type: 9630SIP	Security Code:	TN: 1
Port: IP	Coverage Path 1:	COR: 1
Name: SIP-28003	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 28003	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Expansion Module? n	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
Customizable Labels? y		

Enter the **add off-pbx-telephone station-mapping** command and configure the following:

- **Station Extension** – Set the extension of the OPS station as configured above.
- **Application** – Set to **OPS**.
- **Phone Number** – Enter the number that Biamp Tesira SVC-2 will use for registration and call termination. In the example below, the Phone Number is the same as the Station Extension, but is not required to be the same.
- **Trunk Selection** – Set to the trunk group number configured in **Section 5.6**.
- **Config Set** – Set to **1**

Repeat this step as necessary to configure additional off-pbx-telephone station-mapping.

add off-pbx-telephone station-mapping							Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station	Application	Dial	CC	Phone Number	Trunk	Config	
Extension		Prefix			Selection	Set	
28003	OPS	-		28003	201	1	

## 6. Configure Avaya Aura® SIP Enablement Services

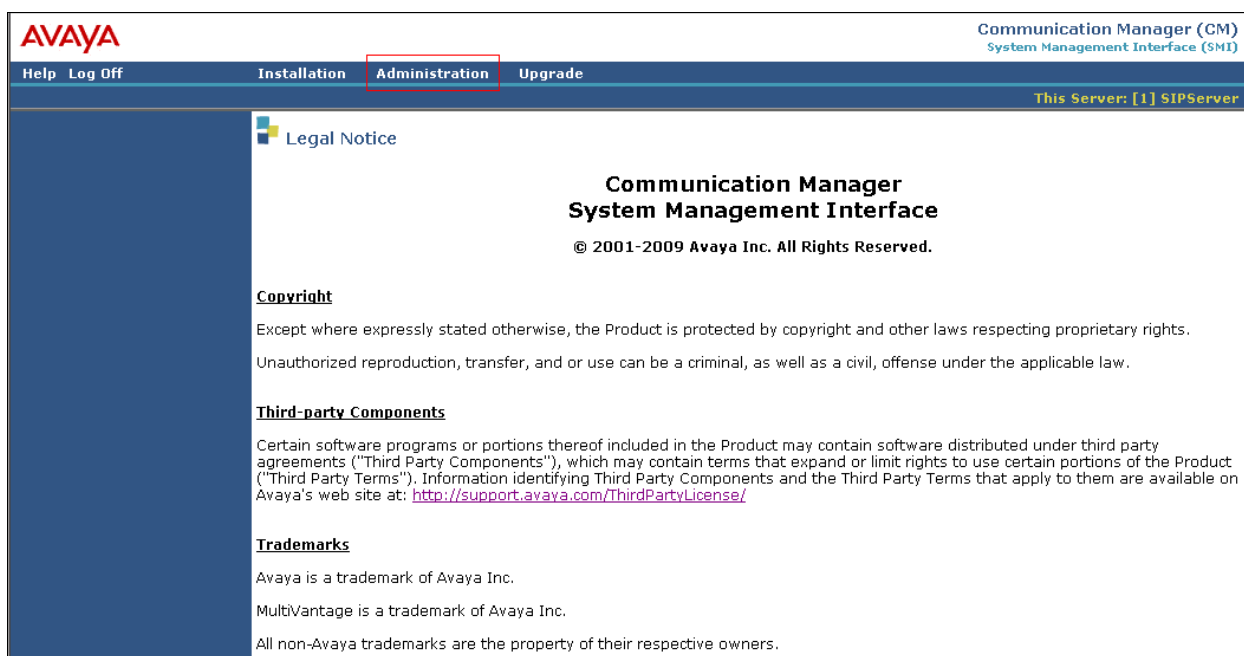
This section describes the steps for creating SIP trunks between SIP Enablement Services and Communication Manager. SIP user accounts are configured in SIP Enablement Services and associated with Communication Manager OPS station extensions. During the compliance test, Biamp Tesira SVC-2 is treated as a SIP endpoint. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

During the compliance test, the TLS transport protocol was tested between the 3<sup>rd</sup> party endpoint (Biamp Tesira SVC-2) and SIP Enablement Services.

### 6.1. Configure SIP Enablement Services Server Properties

Launch a web browser, enter <https://<IP address of SIP Enablement Services server>/admin> in the URL, and log in with the appropriate credentials. Click on the **Launch SIP Enablement Services Administration Interface** link upon successful login.

Navigate to **Administration** → **SIP Enablement Services**.



In the **Integrated Management SIP Server Management** page, select the **Server Configuration → System properties** link from the left pane of the screen. Verify the **SIP Domain** matches the Far-end Domain field value configured for the signaling group on Communication Manager in **Section 5.5**. Click on the **Update** button, after the completion.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The left sidebar contains a navigation menu with the following items: Top, Users, Address Map Priorities, Adjunct Systems, Aggregator, Certificate Management, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts, IM logs, Communication Manager Servers, Communication Manager Extensions, Server Configuration (highlighted), Admin Setup, IM Log Settings, License, SNMP Configuration, System Properties, SIP Phone Settings, Survivable Call Processors, System Status, Trace Logger, and Trusted Hosts. The main content area is titled 'View System Properties' and contains the following configuration details:

SES Version	SES-5.2.1.0-016.4
System Configuration	Simplex
Host Type	SES combined home-edge
SIP Domain*	avaya.com

Note that the DNS domain is avaya.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

SIP License Host*	10.64.40.41
-------------------	-------------

**DiffServ/TOS Parameters**

Call Control PHB Value*	46
-------------------------	----

**802.1 Parameters**

Priority Value*	6
Management System Access Login	
Management System Access Password	
DB Log Level	disabled

An 'Update' button is located at the bottom left of the configuration area.

## 6.2. Configure Communication Manager Servers

This section provides steps to add SIP-enabled media servers to the SIP domain. In the **Integrated Management SIP Server Management** page, select the **Communication Manager Servers** → **Add** link from the left pane of the screen. The following screen shows the Add Media Server Interface page. The highlighted fields were configured for the compliance test:

- **Communication Manager Server Interface Name** – Enter a descriptive name for the communication manager server interface.
- **SIP Trunk IP Address** – Enter the IP address for the CLAN IP interface that terminates the SIP link from SIP Enablement Services.

Click **Add** when finished.

**AVAYA** Integrated Management SIP Server Management  
This Server: [1] SIPServer

Help Exit

**Top**

- Users
- Address Map Priorities
- Adjunct Systems
- Aggregator
- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
- Hosts
- IM logs
- Communication Manager Servers
  - Add**
  - List
- Communication Manager Extensions
- Server Configuration
  - Admin Setup
  - IM Log Settings
  - License
  - SNMP Configuration
  - System Properties
- SIP Phone Settings
- Survivable Call Processors
- System Status
- Trace Logger
- Trusted Hosts

**Add Communication Manager Server Interface**

Communication Manager Server Interface Name\* S8720

Host 10.64.40.41

**SIP Trunk**

SIP Trunk Link Type ☐ TCP ☒ TLS

SIP Trunk IP Address\* 10.64.40.24

**Communication Manager Server**

Communication Manager Server Admin Address\* (see Help)

Communication Manager Server Admin Port\* 5022

Communication Manager Server Admin Login\*

Communication Manager Server Admin Password\*

Communication Manager Server Admin Password Confirm\*

**SMS Connection Type** ☒ SSH ☐ Telnet ☐ Not Available

Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and changing connection type to Telnet will change admin port to 5023 when Add or Update is clicked.

Fields marked \* are required.

**Add**

### 6.3. Configure Users

This section provides steps to add users to be administered in the SIP Enablement Services database. In the Integrated Management SIP Server Management page, select the **Users → Add** link from the left pane of the screen. The highlighted fields were configured for the compliance test:

- **Primary Handle** – Enter the phone number of Biamp Tesira SVC-2. This number was configured in **Section 5.7**.
- **Password / Confirm Password** – Enter a password; both field entries must match exactly.
- **First Name** – Enter the first name of the user in alphanumeric characters.
- **Last Name** – Enter the last name of the user in alphanumeric characters.
- **Add Media Server Extension** - Select this field if you want to associate a new extension number with this user in the database now. If so, the Add MS Extension screen will be displayed next, after this user profile has been added. If not, in the future you may choose to associate extensions with the user.

Click **Add** when finished.

Top

Users

Add

Default Profile

Delete

Edit

List

Password

Search

Manage All Registered

Users

Search Registered

Devices

Search Registered

Users

Address Map Priorities

Adjunct Systems

Aggregator

Certificate Management

Conferences

Emergency Contacts

Export/Import to ProVision

Hosts

IM logs

Communication Manager  
Servers

Communication Manager

Add User

Primary Handle\* 28003

User ID 28003

Password\* .....

Confirm Password\* .....

Host\* 10.32.8.41

First Name\* SIP

Last Name\* 28003

Address 1

Address 2

Office

City

State

Country

Zip

Survivable Call Processor none

Add Communication  
Manager Extension ☒

Fields marked \* are required.

Add

From the next screen, enter the numeric telephone extension you want to create in the database. Select the extension's Communication Manager Server from the drop-down list. Click on the **Add** button.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top header features the Avaya logo on the left and the text 'Integrated Management SIP Server Management' on the right, with a sub-header 'This Server: [1] SIPServer'. A navigation menu on the left includes 'Top' and 'Users' (expanded) with sub-items: 'Add', 'Default Profile', 'Delete', 'Edit', 'List', 'Password', 'Search', and 'Manage All Registered Users'. The main content area is titled 'Add Communication Manager Extension' and contains the instruction 'Add Communication Manager extension for user 28003.' Below this, there are two input fields: 'Extension' with the value '28003' and 'Communication Manager Server' with a dropdown menu showing 'S8720'. A note states 'Fields marked \* are required.' At the bottom of the form is an 'Add' button. Red rectangular boxes highlight the 'Extension' and 'Communication Manager Server' fields, and the 'Add' button.

## 6.4. Configure TLS Transport Protocol for Biamp Tesira SVC-2

The following describes steps to configure a 3<sup>rd</sup> party endpoint (Biamp Tesira SVC-2) to utilize TLS protocol.

- RootCA (PEM format) from Biamp Tesira SVC-2 – Biamp sends Root certificates to SIP Enablement Services.

```
-----BEGIN CERTIFICATE-----
MIICqDCCAhGgAwIBAgIJAI1AR5p845/8MA0GCSqGSIb3DQEBBQUAMG0xETAPBgNV
BAMTCHdlc3R5YWtlMQ8wDQYDVQQIEWZPcmVnb24xCzAJBgNVBAYTAlVTMSIwIAIJ
KoZIHvcNAQkBFhN3ZXN0bGFrZUBiaWFtcGMuY29tMRYwFAYDVQQKEWlCaWFtcCBT
eXN0ZWlzMBA4XDTEyMDgxmMjIxMjgxoVoxDTEyMDgxmMTIxMjgxoVowbTERMA8GA1UE
AxMIc2VzdGxha2UxZDZANBgNVBAgTBk9yZWdvdjELMAkGA1UEBhMCVVMxIjAgBgkq
hkiG9w0BCQWE3dlc3R5YWtlQGJpYW1wYy5jb20xZjAUBGNVBAoTDUJpYW1wIFN5
c3RlbXMwgZ8wDQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBALlggAdFrDOckgQSq/VF
SPTccuRp/LQNB2J9OmU8tBroDSbb048iH49RjkoESGivvQZH9JMUKWZLCHdixpPO
utp02oeS/+9s/jSThqBsOI9xVFaxKdUvNhphmMD6CsTKyNo6T5npUNp4ddYeP2Ey
jFzjhqarQHsL/DqFF2orfoH1AgMBAAGjUDBOMAwGA1UdEwQFMAMBAf8wHgYDVR0R
BBcwFYETd2VzdGxha2VAYmlhbXBjLmNvbTAeBgNVHRIEFzAVgRN3ZXN0bGFrZUBi
aWFtcGMuY29tMA0GCSqGSIb3DQEBBQUAA4GBAIF6GN0ZyWxksWGTATez3gGYKiF
J99i1V+m4WY58+fLEMkgKZtA8jMayG0RfDsPXIVIn9NzIOjgBi5nEUXHXBPTO0wp
VWC6vrcECytAkzw7yiyAT200ZhoSFuQ4JKutws39jZZLmOf2V1o3jzOTuwtS0f1O
i08Sjew01eJE1n2B
-----END CERTIFICATE-----
```

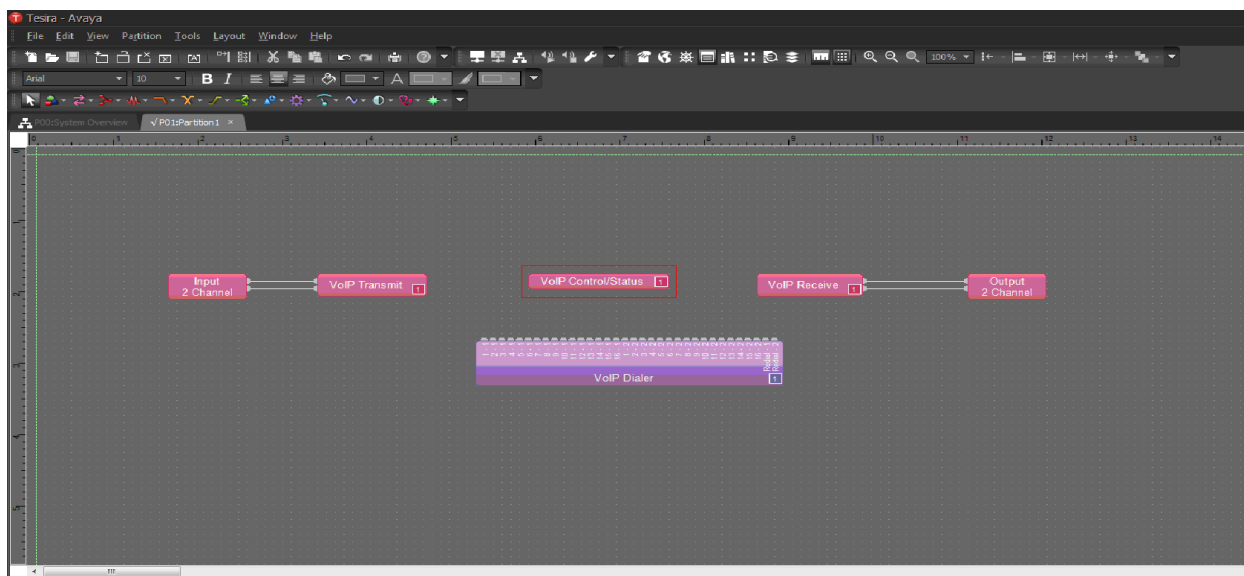
- In SIP Enablement Services, place Root certificates in */var/home/ftp/pub* (the default directory for the Download Files maintenance web page or anonymous ftp, if enabled) before running the command.
- Run **tlscertmanage -l** to list all CA certificates on the system.
- Run **tlscertmanage -i file** to install a Root certificate.



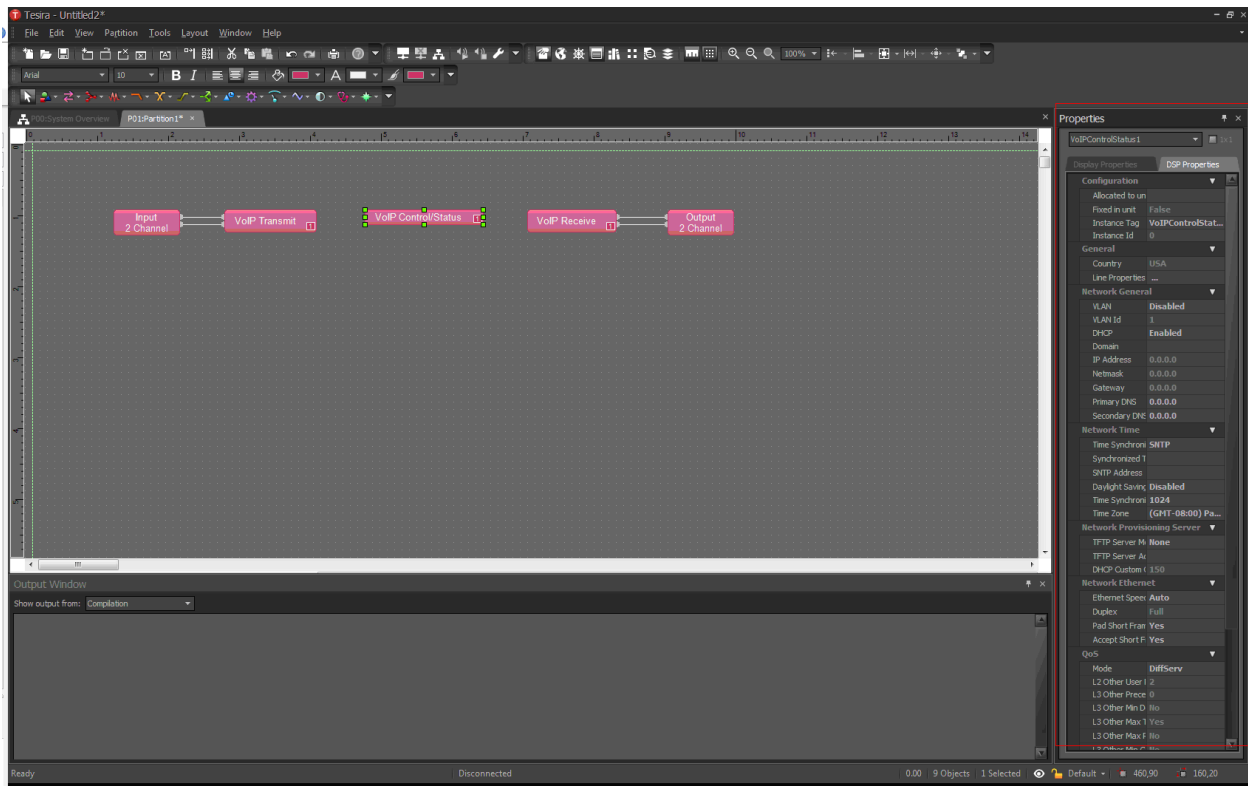
## 7. Configure Biamp Tesira SVC-2

Biamp installs, configures, and customizes the Tesira SVC-2 application for their end customers. This section only provides steps to configure Biamp Tesira SVC-2 to interface with SIP Enablement Services. Select the Tesira icon from Desktop to start Tesira software and design a VoIP system. How to configure a Tesira system is out of the scope.

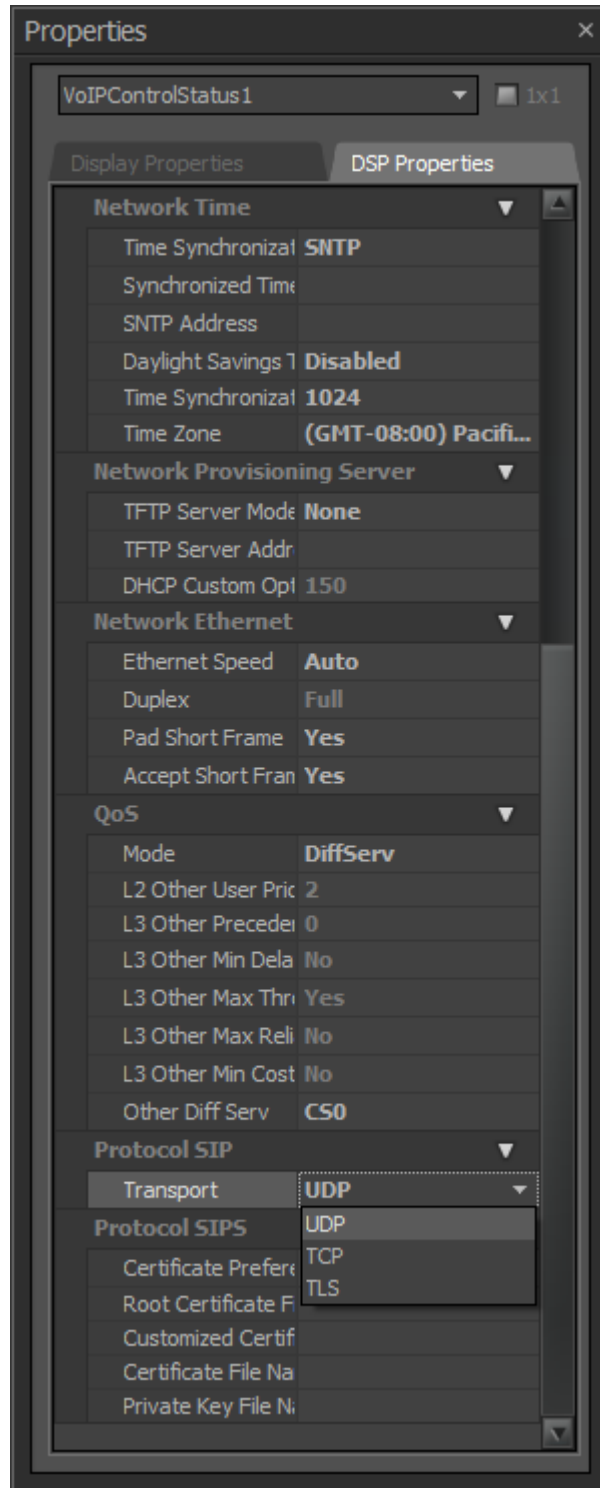
- Highlight the **VoIPControl/Status** block, as shown below.



- Click right mouse button and select **Properties**, and the Properties menu will display on the right



- Navigate the Protocol SIP->Transport to configure transport to be used. The default is UDP. When TLS is selected, please refer to Tesira Operational Manual for additional configuration.



- Select **Line Properties** under the General section

The screenshot shows a 'Properties' window for 'VoIPControlStatus1'. The 'Display Properties' tab is active. The 'General' section is expanded, and the 'Line Properties' option is highlighted with a red rectangle. Other sections like 'Configuration', 'Network General', 'Network Time', 'Network Provisioning Server', and 'Network Ethernet' are also visible.

Configuration	
Allocated to unit	1
Fixed in unit	False
Instance Tag	VoIPControlStatus1
Instance Id	16777227

General	
Country	USA
Line Properties	...

Network General	
VLAN	Disabled
VLAN Id	1
DHCP	Enabled
Domain	Disabled
IP Address	Enabled
Netmask	255.255.255.0
Gateway	10.64.43.1
Primary DNS	205.171.3.65
Secondary DNS	205.171.2.65

Network Time	
Time Synchronizat	SNTP
Synchronized Time	
SNTP Address	
Daylight Savings T	Disabled
Time Synchronizat	1024
Time Zone	(GMT-08:00) Pacifi...

Network Provisioning Server	
TFTP Server Mode	None
TFTP Server Addr	172.16.8.128
DHCP Custom Opt	150

Network Ethernet	
Ethernet Speed	Auto

- From the Line Properties page, click the **Protocol** tab.

The image shows a 'VoIP Line Properties' dialog box. At the top, there are tabs for 'Line 1' and 'Line 2'. Below these are four sub-tabs: 'General', 'Protocol' (which is highlighted with a red box), 'Quality of Service', and 'NAT'. The 'Protocol' tab contains two sections: 'Tones' and 'Call Features'. The 'Tones' section has a 'DTMF Transmit Level' set to '-6.0'. The 'Call Features' section has three options: 'Auto Answer' (Enable), 'Caller Id' (Enabled, highlighted with a red box), and 'Use One Audio Format' (Enable). At the bottom right of the dialog are 'OK' and 'Cancel' buttons.

VoIP Line Properties

Line 1 Line 2 Display Name

General Protocol Quality of Service NAT

Tones

DTMF Transmit Level -6.0

Call Features

Auto Answer Enable

Caller Id Enabled

Use One Audio Format Enable

OK Cancel

- From the Protocol page, provide the following information:
  - **SIP User Name** – Enter a user created in **Section 6.3**.
  - **Authentication User Name** – Enter a user created in **Section 6.3**.
  - **Authentication Password** – Enter the password created in **Section 6.3**
  - **Proxy Vendor** – Select **Avaya SES**
  - **Proxy Address** – Enter the IP address of SIP Enablement Services.
  - **Proxy Port** – Enter either 5060 or 5061.
    - TLS – 5061
    - UDP or TCP – 5060
  - Click on the **OK** button. Default values may be used for all other fields.

Note: *Biamp Tesira SVC-2 can provide two inbound extensions (L1 and L2).*

VoIP Line Properties

Line 1 Line 2 Display Name

General Protocol Quality of Service NAT

SIP

SIP User Name	28003	Registration Expiration	3600 seconds
SIP Display Name	28003,SES	Signaling Port	5060
SIP Domain Name		T1 Timer	500 ms
Authentication User Name	28003	Retransmit Timeout	32000 ms
Authentication Password	*****	Session Timer	Enabled
Proxy Vendor	Avaya SES	Session Refresher	Auto
Proxy Address	10.64.40.41	Session Expiration	1800 seconds
Proxy Port	5060	Minimum Session Expiration	90 seconds
Outbound Proxy Address		Prack	None
Outbound Proxy Port	5060		
Local Dial Plan	[2-9]11 0T 011xxx.T [[0-1][2-9]xxxxxxxx [2-9]xxxxxxxx [2-9]xxxT		

RTP/SRTP

Port Start	10000
Port End	14999
Static RTP Port	Enable
SRTP	
G.723 Encoding Rate	5.3 kbps

SIPS

Keyword	
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OK Cancel

## 8. Verification Steps

The following steps may be used to verify the configuration:

- Verify that Biamp Tesira SVC-2 successfully registers with the SIP Enablement Services server by following the **Users -> Search Registered Users** link on the SIP Enablement Services Administration Web Interface.
- Place calls to and from Biamp Tesira SVC-2 and verify that the calls are successfully established with two-way talk path.

## 9. Conclusion

Biamp Tesira SVC-2 was compliance tested with Communication Manager and SIP Enablement Services. Biamp Tesira SVC-2 functioned properly for feature and serviceability. During compliance testing, Biamp Tesira SVC-2 successfully registered with SIP Enablement Services, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like source transfer.

## 10. Additional References

The following Avaya product documentation can be found at <http://support.avaya.com>

[1] *Administering Avaya Aura™ Communication Manager*, Release 6.0, June 2010, Issue 6.0, Document Number 03-300509

[2] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Issue 9, May 2009, Document Number 555-245-206.

The following document was provided by Biamp.

[3] *Tesira Operation Manual*, Document.



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