Avaya Solution & Interoperability Test Lab

Application Notes for Biamp Tesira SVC-2 and Avaya Aura® Communication Manager and Avaya Aura® 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® Communication Manager and Avaya Aura® 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® Communication Manager, Avaya Aura® 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® Communication Manager and Avaya Aura® 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.

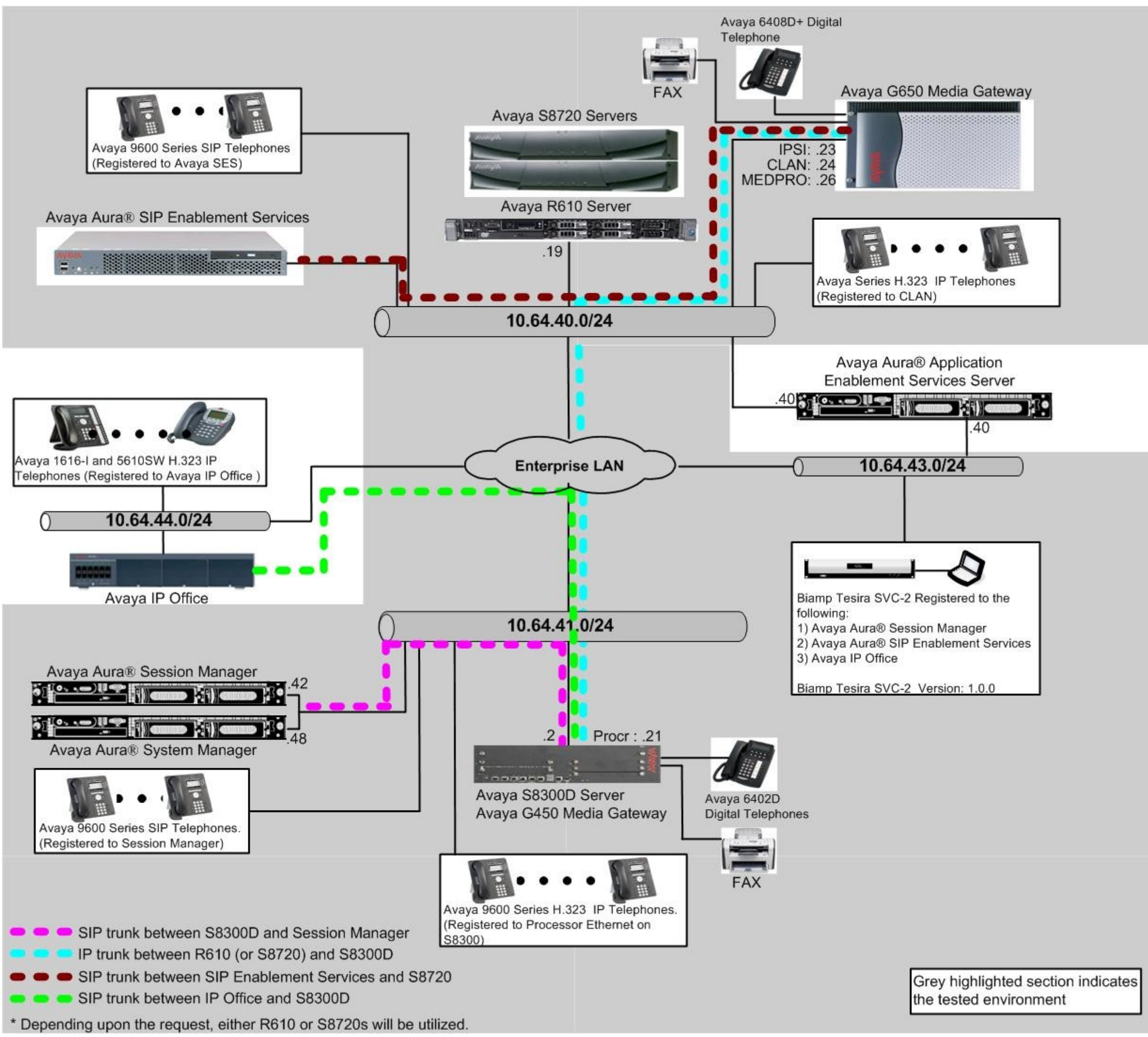


Figure 1: Test Configuration of Biamp Tesira SVC-2

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 (*used for inter-switch test scenarios*)		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) 9630 (H.323) 9650 (H.323)	3.1 3.1 3.1
Avaya 9600 Series SIP Telephones		
	9620 (H.323) 9630 (H.323) 9650 (H.323)	2.64 2.64 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                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                     IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                 RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
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                                  RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

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

         DTMF 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
 Dial Access? n                                   Night Service:
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
                                           Time of Day Lock Table:
              Loss Group: 19         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

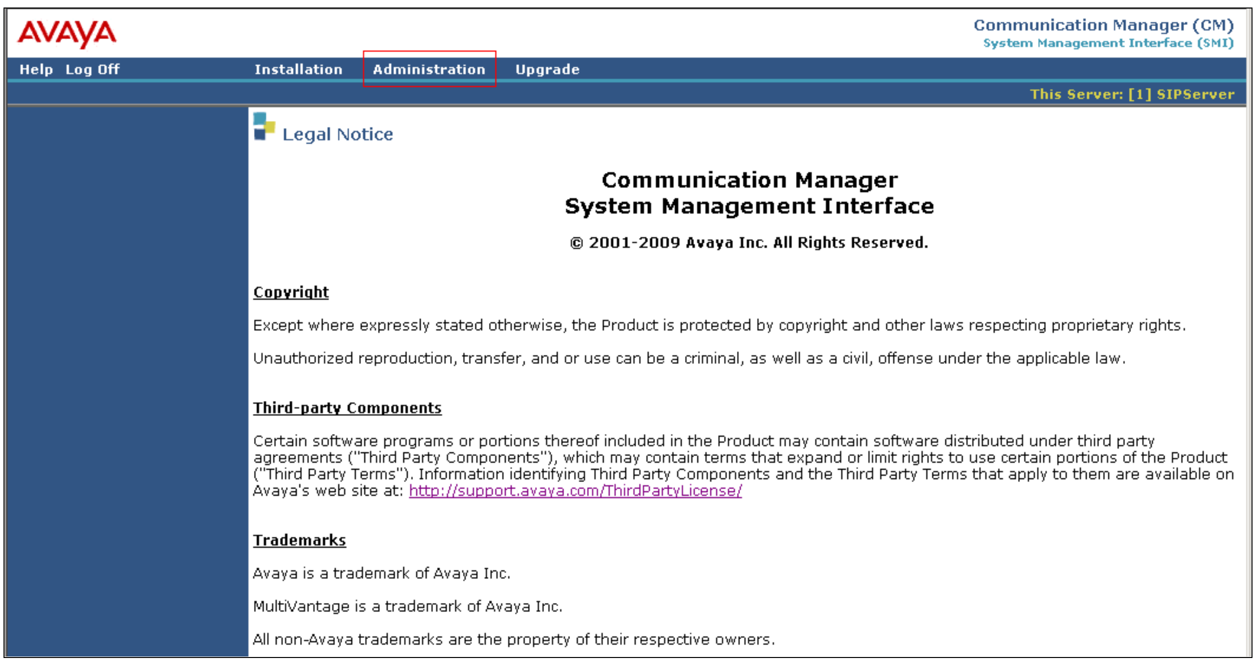
This section describes the steps for creating SIP trunks between SIP Enablement Services and Communication Manager. SIP user accounts are configured in SIP Enablement Services and associated with Communication Manager OPS station extensions. During the compliance test, Biamp Tesira SVC-2 is treated as 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 3rd party endpoint (Biamp Tesira SVC-2) and SIP Enablement Services.

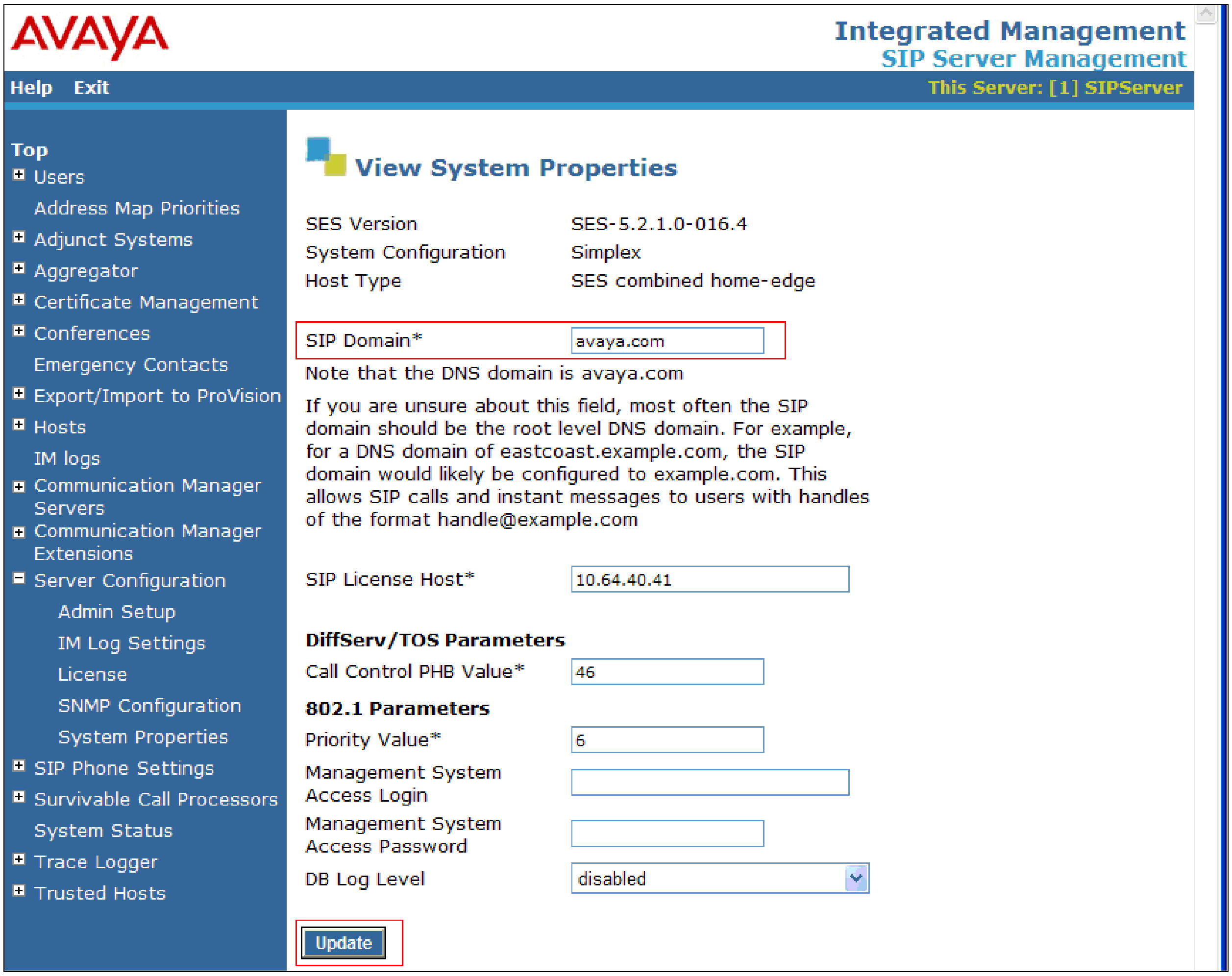
6.1. Configure SIP Enablement Services Server Properties

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.

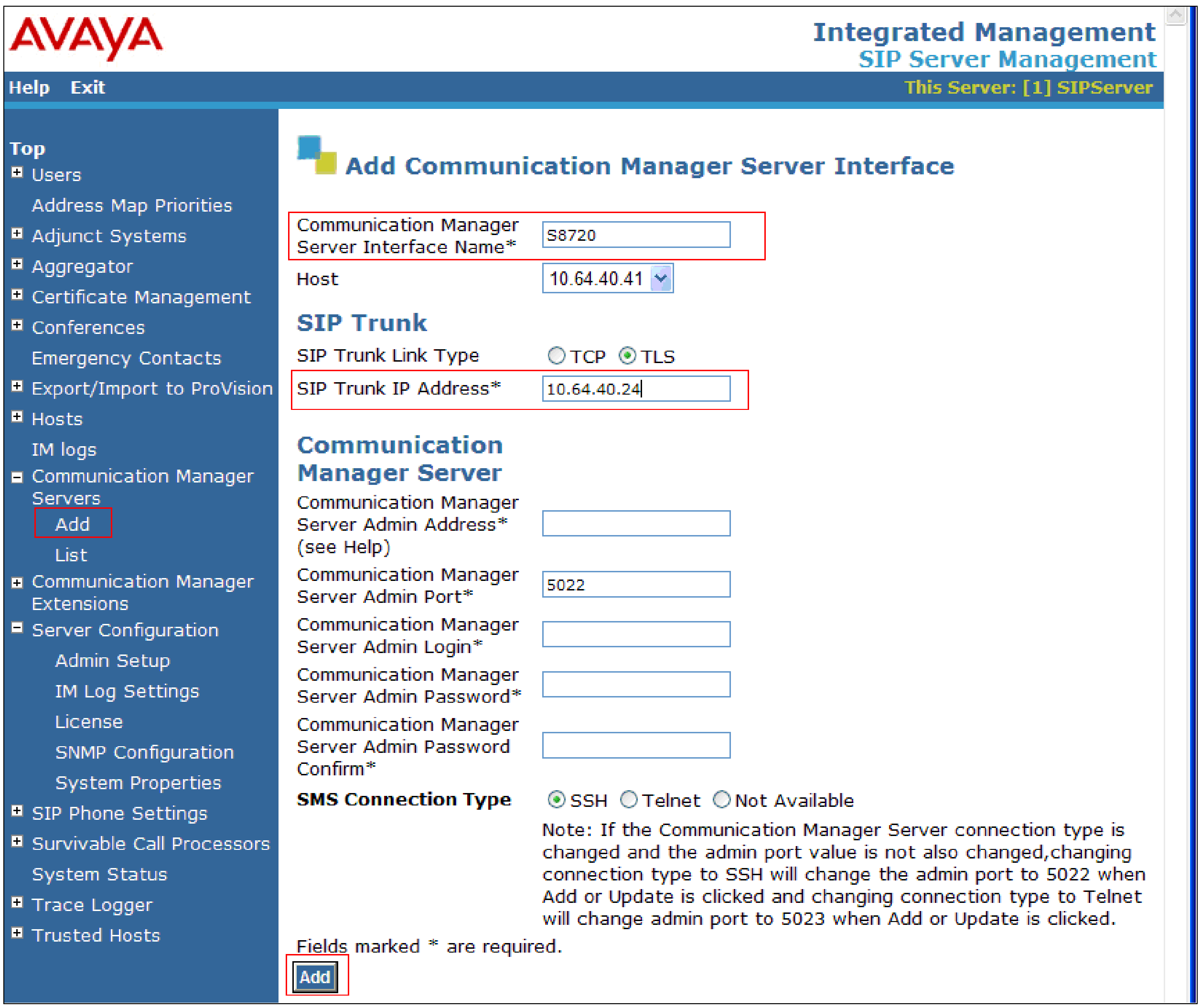


6.2. Configure Communication Manager Servers

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.

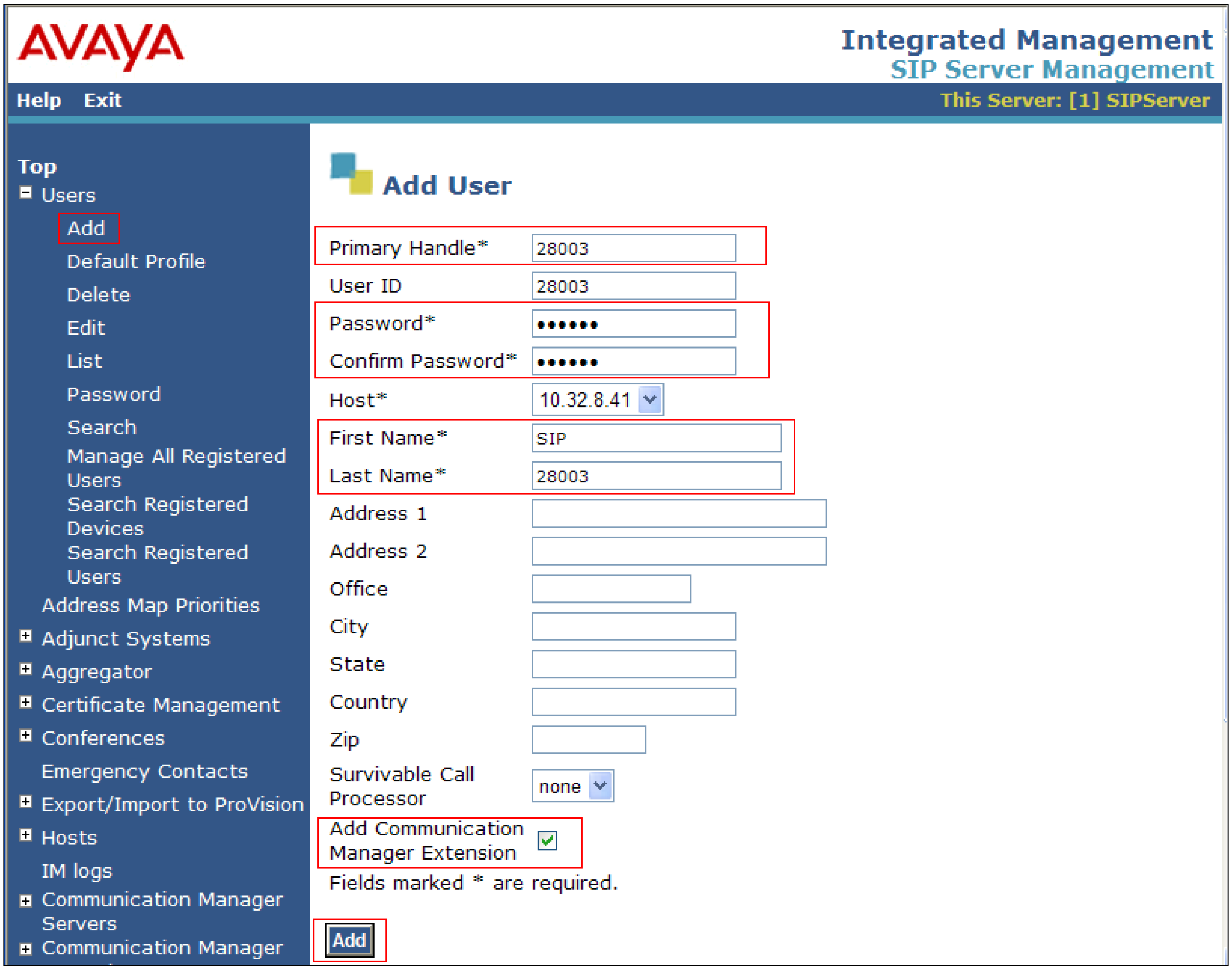


6.3. Configure Users

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.


AVAYA
Integrated Management
SIP Server Management
Help Exit
This Server: [1] SIPServer
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.

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

The following describes steps to configure a 3rd 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.

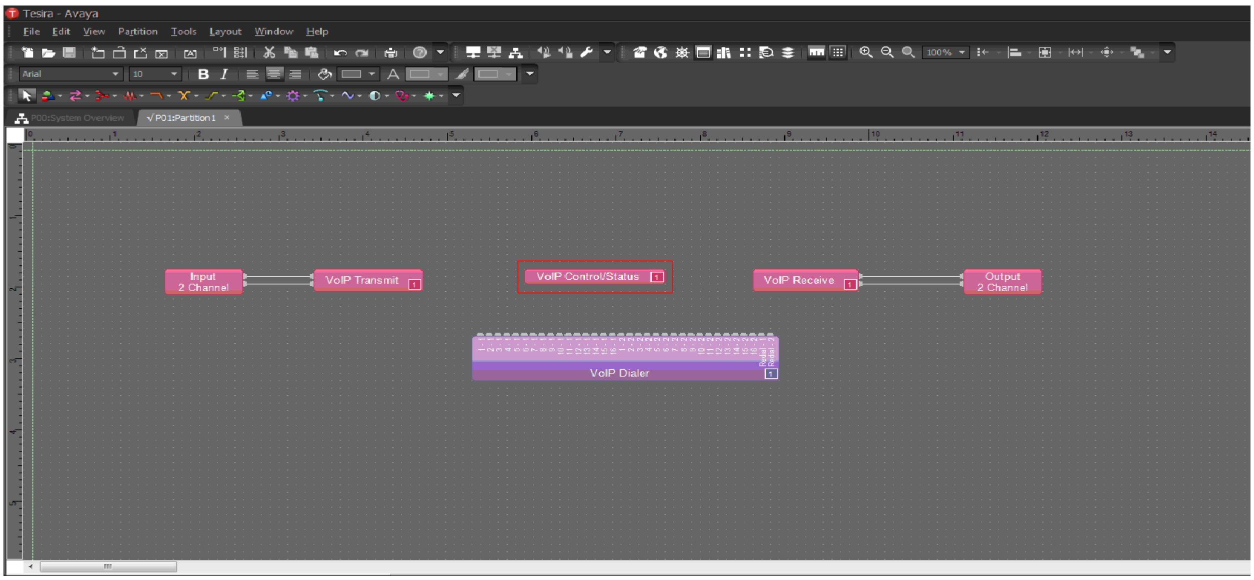

```
-----BEGIN CERTIFICATE-----
MIICqDCCAhGgAwIBAgIJAI1AR5p845/8MA0GCSqGSIb3DQEBBQUAMG0xETAPBgNV
BAMTCHdlc3RsYWtlMQ8wDQYDVQQIEwZPcmVnb24xCzAJBgNVBAYTAlVTMSIwIAYJ
KoZIhvcNAQkBFhN3ZXN0bGFrZUBiaWFtcGMuY29tMRYwFAYDVQQKEw1CaWFtcCBT
eXN0ZW1zMB4XDTExMDgxMjIxMjgxOVoXDTEyMDgxMTIxMjgxOVowbTERMA8GA1UE
AxMId2VzdGxha2UxDzANBgNVBAgTBk9yZWdvbjELMAkGA1UEBhMCVVMxIjAgBgkq
hkiG9w0BCQEWE3dlc3RsYWtlQGJpYW1wYy5jb20xFjAUBgNVBAoTDUJpYW1wIFN5
c3RlbXMwgZ8wDQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBAL1ggAdFrDOckgQSq/VF
SPTccuRp/LQNB2J9OmU8tBroDSbbO48iH49RjkOESGivyQZH9JMUKWZLCHdixpPO
utp02oeS/+9s/jSThqBsOI9xVFaXKdUvNhphmMD6CsTKyNo6T5npUNp4ddYeP2Ey
jfZjhqaRQHsL/DqFF2orfOH1AgMBAAGjUDBOMAwGA1UdEwQFMAMBAf8wHgYDVR0R
BBcwFYETd2VzdGxha2VAYmlhbXBjLmNvbTAeBgNVHRIEFzAVgRN3ZXN0bGFrZUBi
aWFtcGMuY29tMA0GCSqGSIb3DQEBBQUAA4GBAIKF6GN0ZywXksWGTATez3gGYKiF
J99i1V+m4WY58+flEMkgKZtA8jMAyG0RfDsPXIVIn9NzIOjgBi5nEUXHXBPTO0wp
VWC6vrcECytAkzw7yiYAT200ZhoSFuQ4JKutws39jZZLmOf2V1o3jzOTuwtS0flO
i08SjewO1eJE1n2B
-----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 installs a Root certificates.

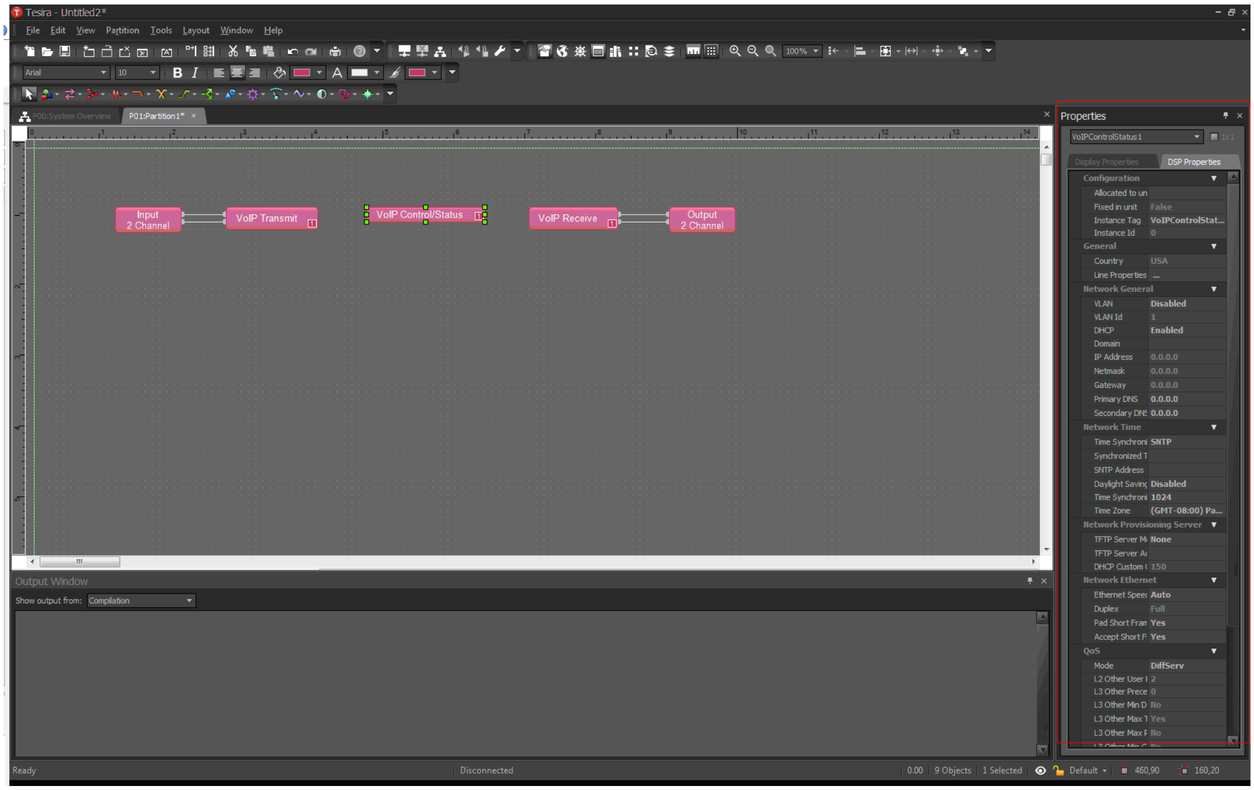
7. Configure Biamp Tesira SVC-2

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.

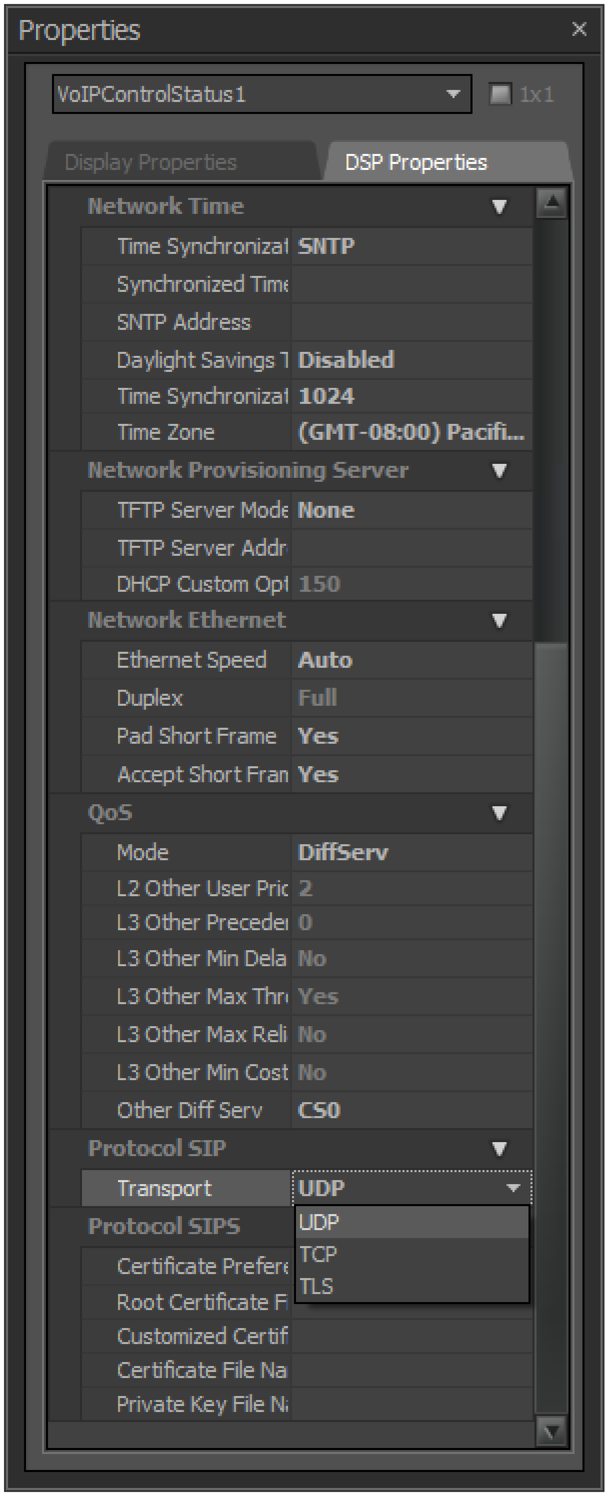
- Highlight the **VoIPControl/Status** block, as shown below.



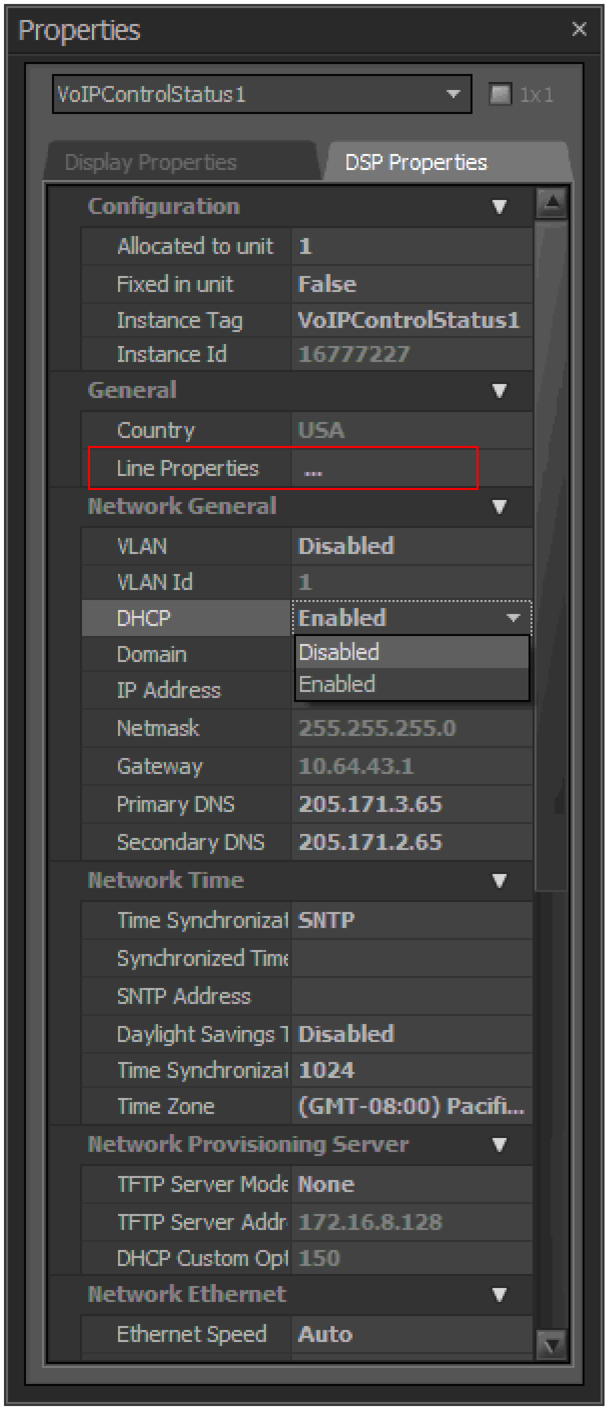
- Click right mouse button and select **Properties**, and the Properties menu will display on the right



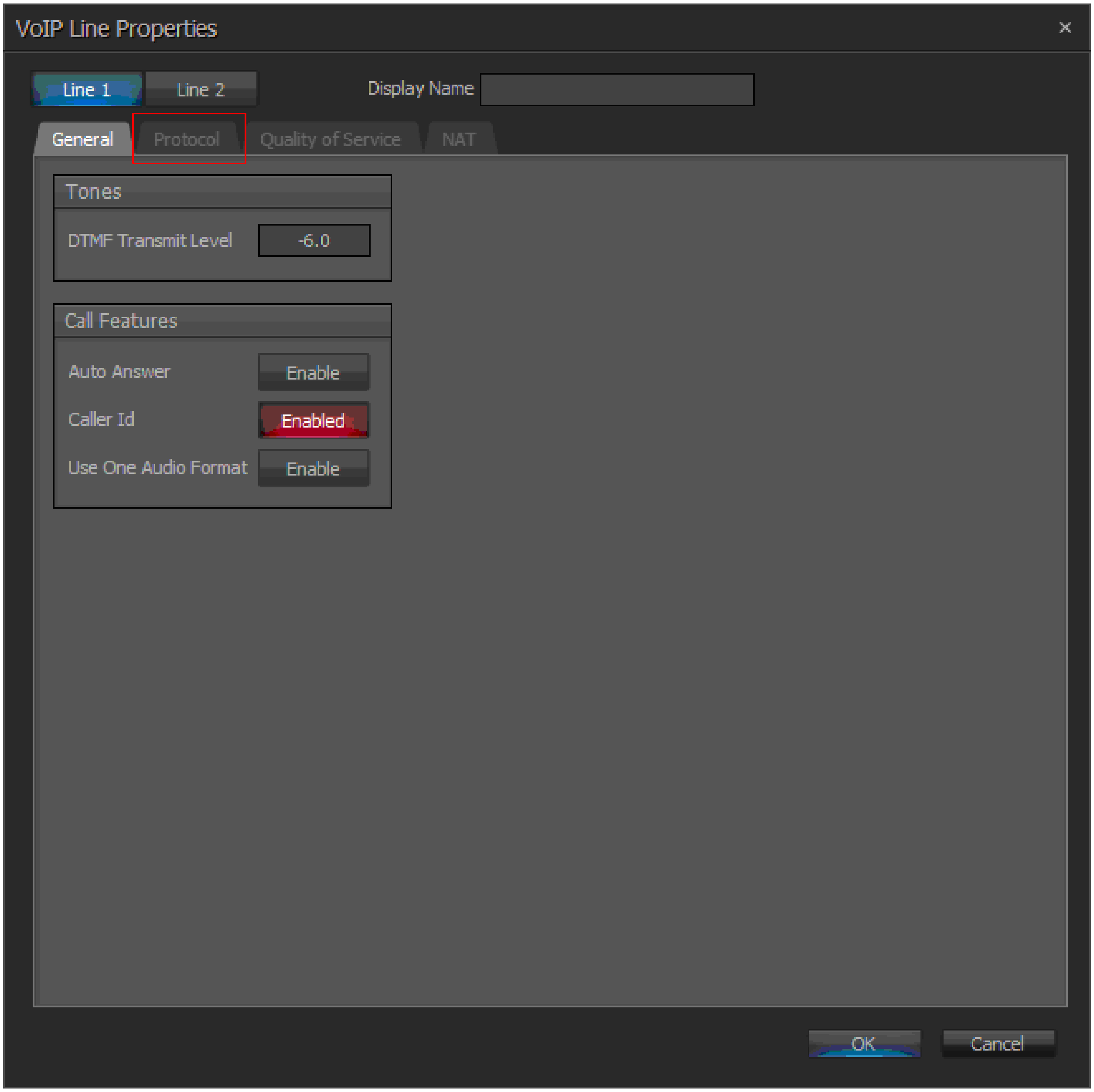
- Navigate the Protocol SIP->Transport to configure transport to be used. The default is UDP. When TLS is selected, please refer to Tesira Operational Manual for additional configuration.



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

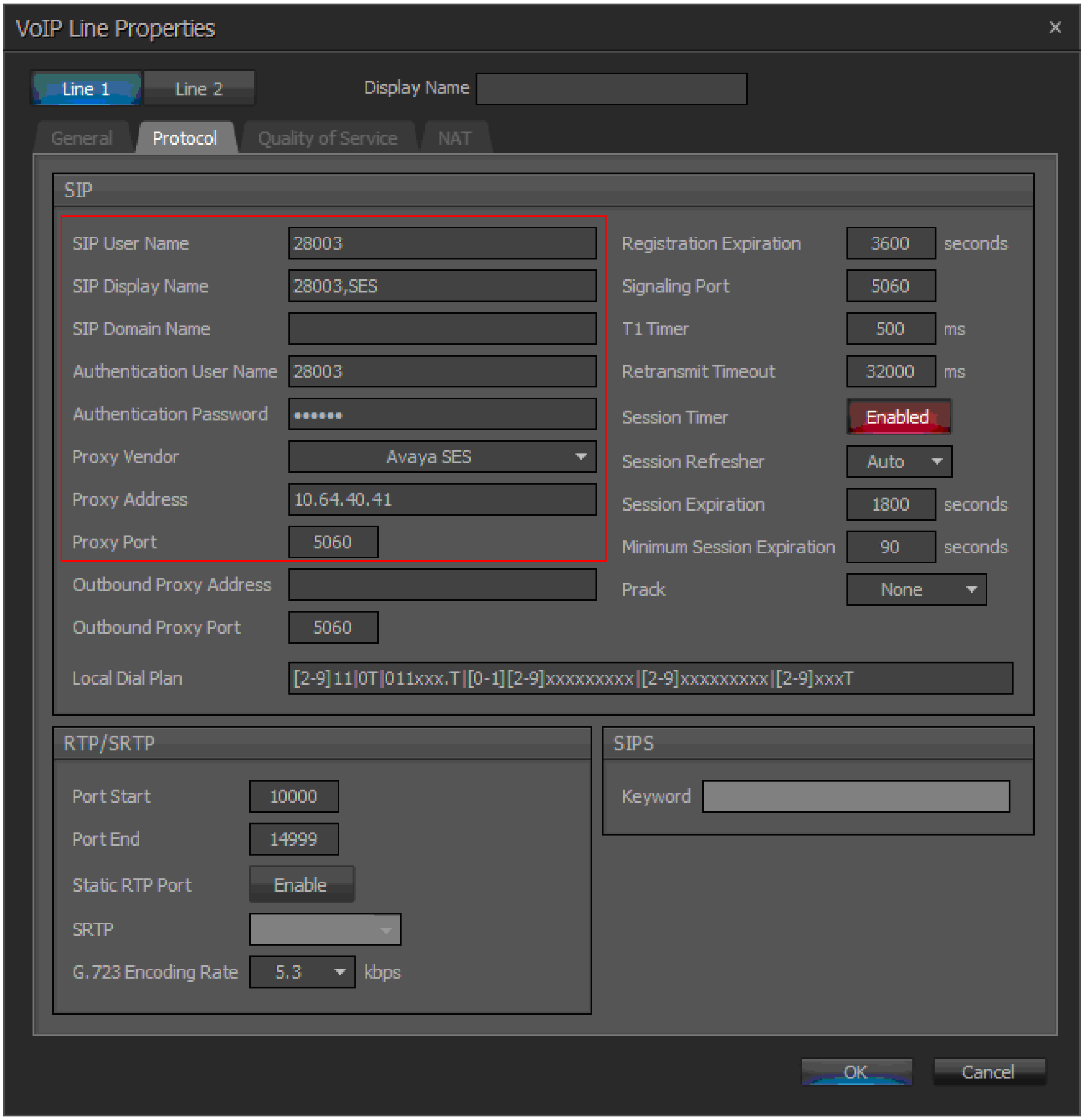


- From the Line Properties page, click the **Protocol** tab.



- 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).*



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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.