Avaya Solution & Interoperability Test Lab

Application Notes for Biamp Tesira SVC-2 with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3 – 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 R6.3 and Avaya Aura® Session Manager R6.3.

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® Session Manager and 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® Session Manager.

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. 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.

For further details on Tesira SVC-2 configuration steps not covered in this document, consult **Section 10 [4]**.

2. General Test Approach and Test Results

All test cases were performed manually. The general test approach was to verify the Biamp Tesira SVC-2 registration and place various types of calls to and from Biamp Tesira SVC-2. Biamp Tesira SVC-2 operations such as inbound calls, outbound calls, hold/resume, DTMF, audio codecs (G.711Mu, G.729, G.722), and Biamp Tesira SVC-2 interactions with Session Manager, Communication Manager, and Avaya SIP and H.323 telephones were verified. For serviceability testing, failures such as cable pulls and resets were applied.

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.

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, Session Manager, 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 S8300D Server, an Avaya G450 Media Gateway, Session Manager and Biamp Tesira SVC-2. The solution described herein is also extensible to other Avaya Servers and Media Gateways. 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.

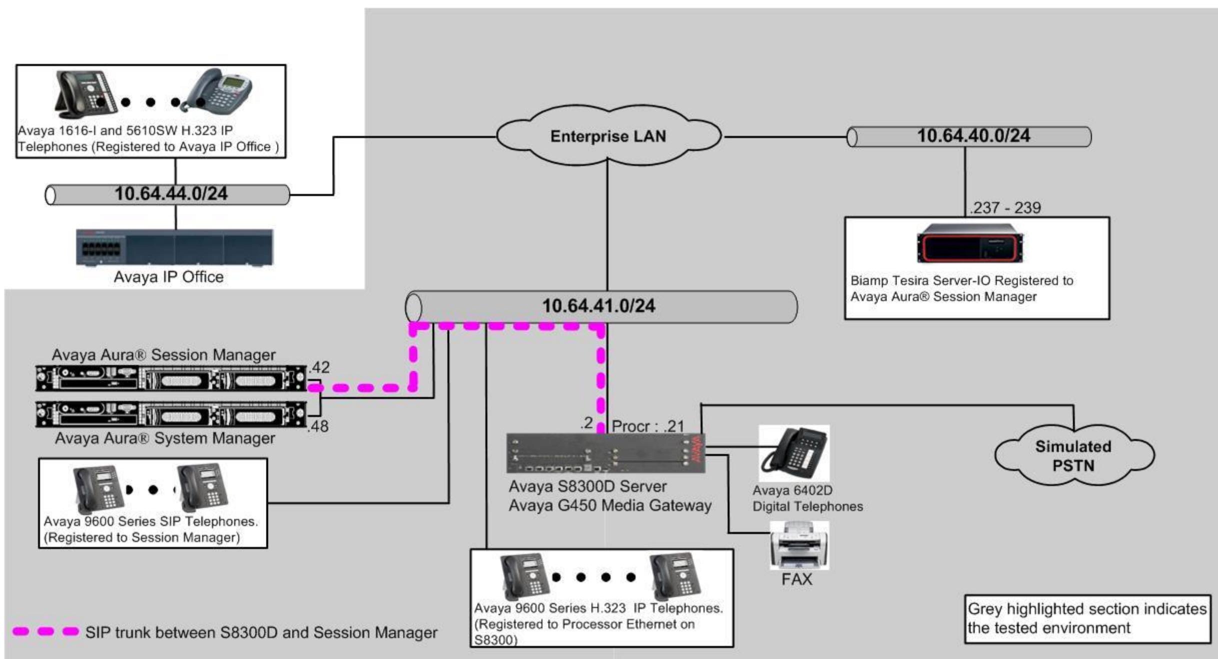


Figure 1: Test Configuration of Biamp Tesira SVC-2

4. Equipment and Software Validated

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

Equipment/Software		Release/Version
Avaya Aura® Communication Manager running on Avaya S8300D Server with Avaya G450 Media Gateway		R016x.03.0.124.0 w/ patch 03.0.124.0-21172
Avaya Aura® System Manager on Avaya S8800 Server		6.3.5.5.2017
Avaya Aura® Session Manager on Dell R610		6.3.0.0.18002
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)	2.6.4 2.6.4 2.6.4
Avaya 6408D+ Digital Telephone		-
Biamp Tesira SVC-2		1.2.1
Biamp Tesira Server-IO		2.0.0
Biamp Linux		3.2.48-BIAMP

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for setting up a SIP trunk between Communication Manager and Session Manager. 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: V16                                Software Package: Enterprise
       Location: 2                                   System ID (SID): 1
       Platform: 28                                  Module ID (MID): 1

                                                               USED
                                Platform Maximum Ports: 6400  254
                                      Maximum Stations: 2400  42
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 9600  1
                    Maximum Off-PBX Telephones -   OPS: 9600  19
                    Maximum Off-PBX Telephones - PBFMC: 9600  0
                    Maximum Off-PBX Telephones - PVFMC: 9600  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   1
```

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: 4000  27
           Maximum Concurrently Registered IP Stations: 2400  2
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 2400  2
                  Maximum Video Capable IP Softphones: 2400  2
                       Maximum Administered SIP Trunks: 4000  65
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 50    1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   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 Session Manager. 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: G.729             n          2        20
 3:
```

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 Session Manager. 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 Session Manager.
- **Intra-region IP-IP Direct Audio** – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager 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 Session Manager in different IP network regions. The default value for this field is **yes**.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: avaya.com
    Name:                           Stub Network Region: n
MEDIA PARAMETERS                    Intra-region IP-IP Direct Audio: yes
      Codec Set: 1                  Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16390                         IP Audio Hairpinning? n
   UDP Port Max: 16999
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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
Keep-Alive Count: 5
```

5.4. Configure IP Node Name

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

```
change node-names ip                                               Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
CLAN                10.64.40.24
IPOffice            10.64.44.21
SES                 10.64.40.41
SM-1                10.64.41.42
SM-2                10.64.21.31
default             0.0.0.0
procr               10.64.41.21
procr6              ::
```

5.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for signaling between Communication Manager and Session Manager. 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 **procr** as displayed in **Section 5.4**.
- **Far-end Node Name** - Set to the Session Manager name configured in **Section 5.4**.
- **Far-end Network Region** - Set to the region configured in **Section 5.3**.

```
change signaling-group 92                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 92              Group Type: sip
  IMS Enabled? n         Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? y        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: SM-1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
                                   Far-end Secondary Node Name:
Far-end Domain:
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? 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? y                     Initial IP-IP Direct 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 Session Manager. 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.

```
add trunk-group 92                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 92                   Group Type: sip         CDR Reports: y
  Group Name: SM_41_42                    COR: 1        TN: 1        TAC: 1092
   Direction: two-way        Outgoing Display? y
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 92
                                                     Number of Members: 10
```

5.7. Configure SIP Endpoint

The following screen displays a SIP extension created from System Manager.

```
display station 72032                                           Page   1 of   6
                                     STATION

Extension: 72032                       Lock Messages? n               BCC: 0
     Type: 9630SIP                     Security Code: *                TN: 1
     Port: S00107                    Coverage Path 1: 99              COR: 1
     Name: Biamp-2                   Coverage Path 2:                 COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
              Loss Group: 19
                                               Message Lamp Ext: 72032

        Display Language: english                Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                        IP SoftPhone? n

                                                       IP Video? n
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. Session Manager is comprised of two functional components: The Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

This section assumes that Session Manager and System Manager have been installed, and network connectivity exists between the two platforms.

The following will be covered in this section for configuring Session Manager.

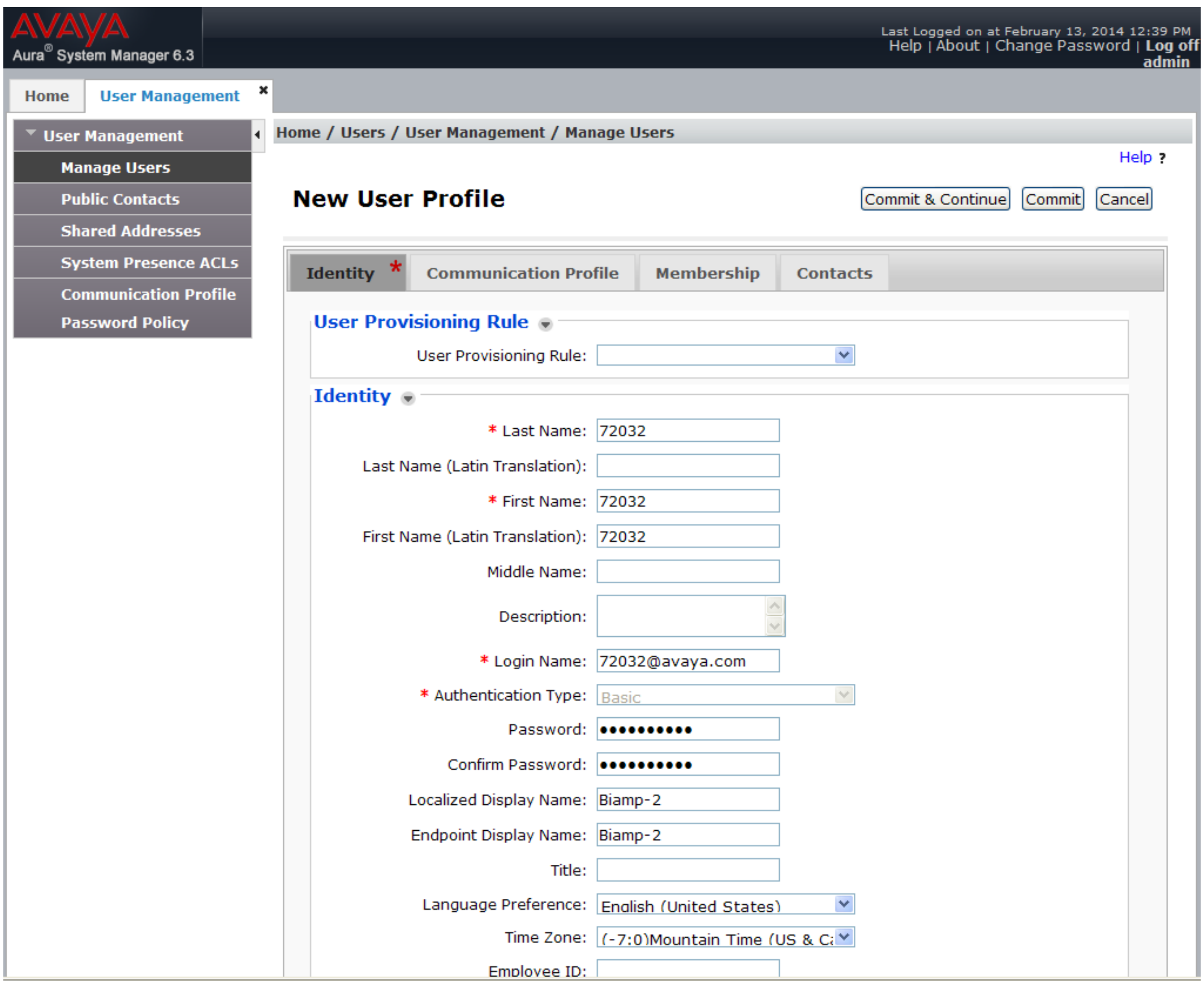
- User Management
- TLS certificate between 3rd party endpoint and Session Manager

6.1. Configure a SIP User

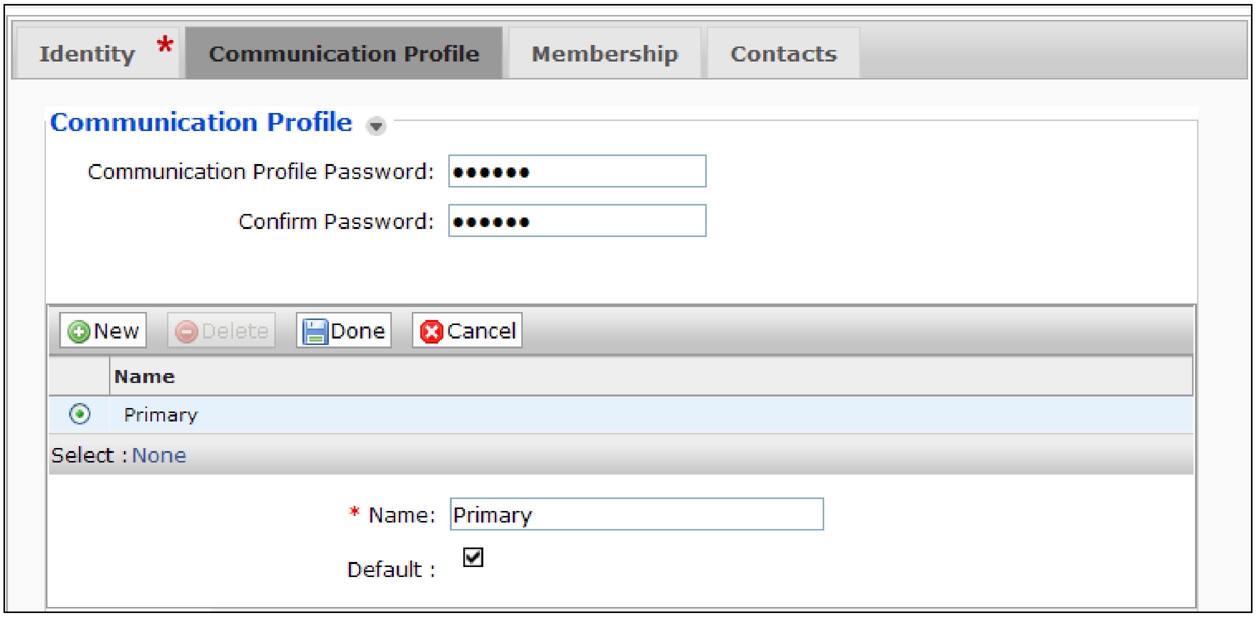
When adding new SIP user, use the option to automatically generate the SIP station in Communication Manager, after adding a new SIP user.

To add new SIP users, Navigate to **Home → Users → User Management → Manage Users**. Click **New** and provide the following information:

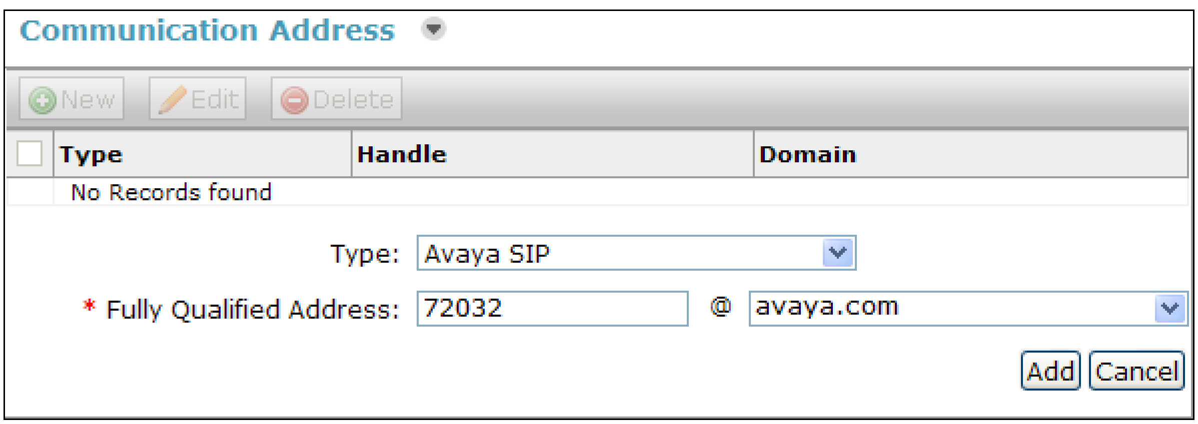
- Identity section
  - **Last Name** – Enter last name of user.
  - **First Name** – Enter first name of user.
  - **Login Name** – Enter extension number@sip domain. The sip domain is defined as Authoritative Domain in **Section 5.3**.
  - **Authentication Type** – Verify **Basic** is selected.
  - **Password** – Enter password to be used to log into System Manager.
  - **Confirm Password** – Repeat value entered above.



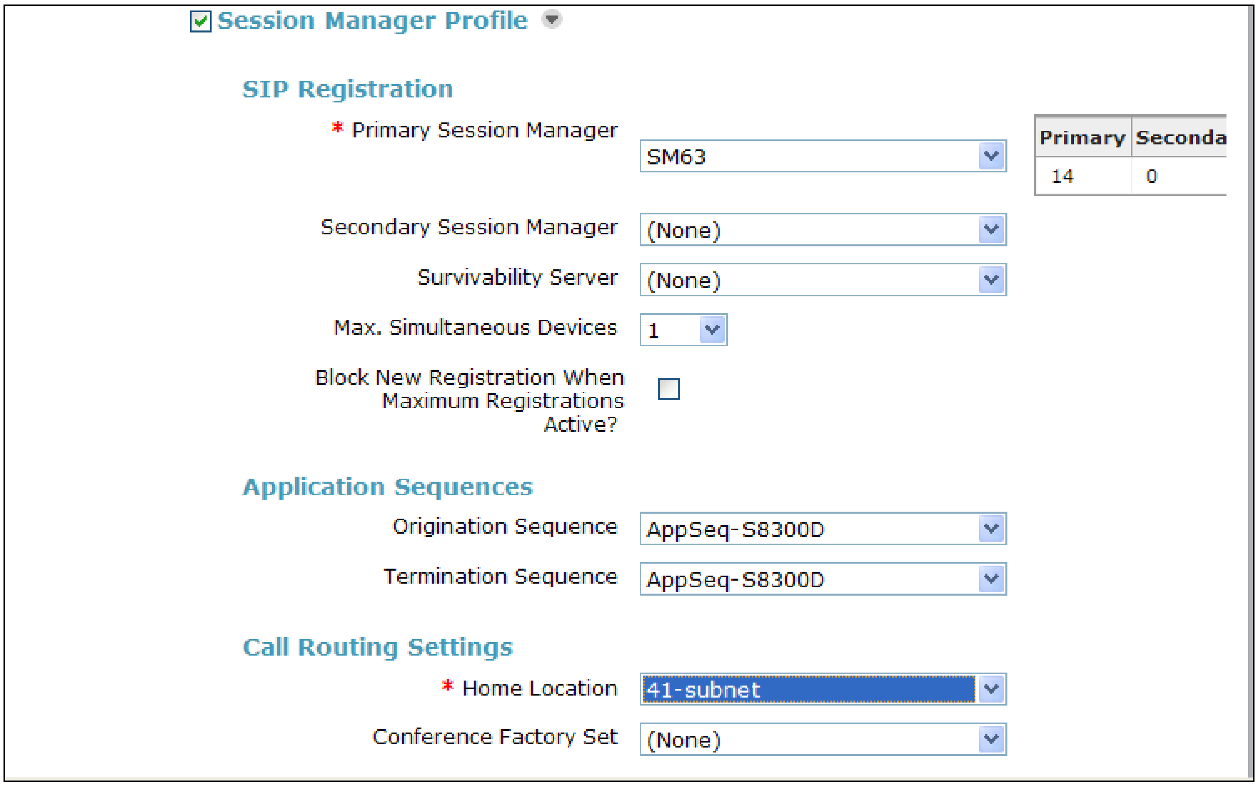
- <u>Communication Profile section</u>
    - **Communication Profile Password** – Type Communication profile password in this field
    - **Confirm Password** – Repeat value entered above.



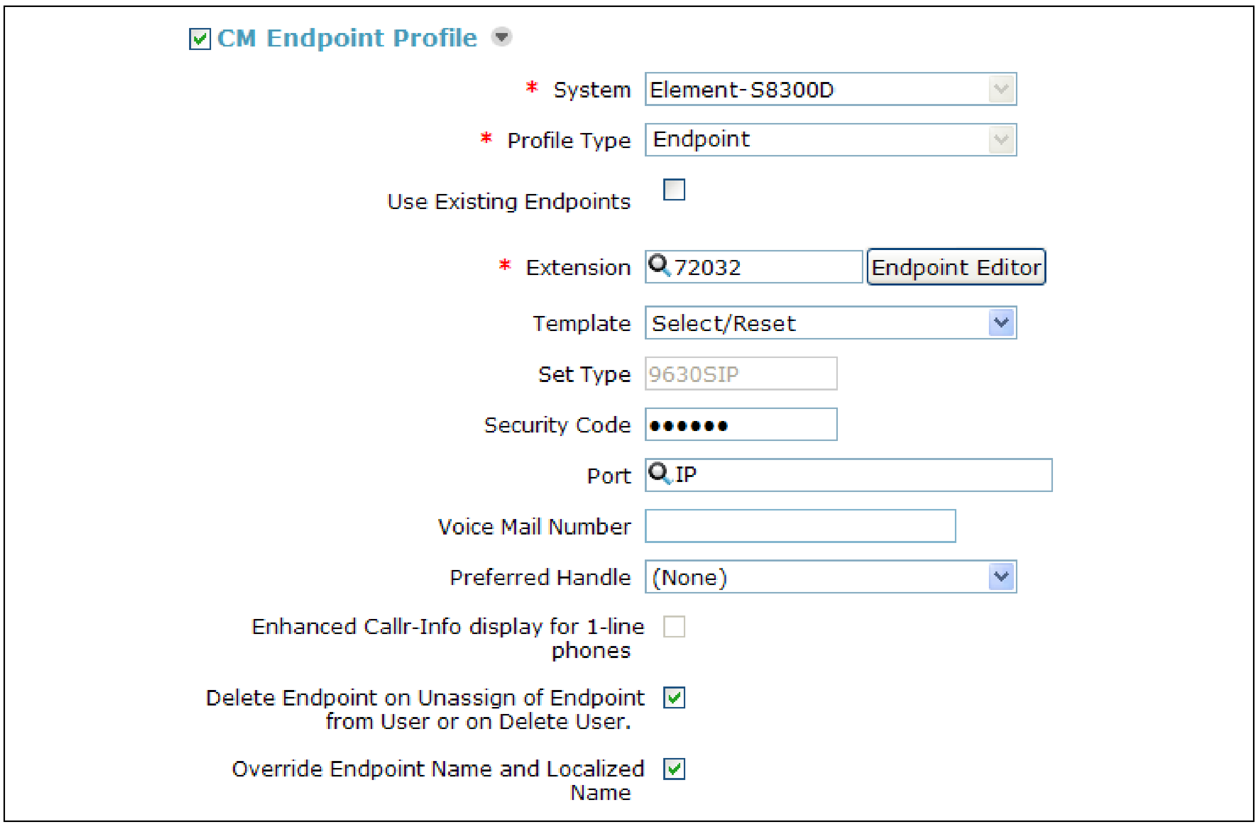
- <u>Communication Address sub-section</u>
    - Select "Avaya SIP" for the **Type** field.
    - **Fully Qualified Address –** Enter the extension of the user and select the relevant domain name.
    - Click **Add** button



- Session Manager Profile section
  - **Primary Session Manager** – Select one of the Session Managers.
  - **Secondary Session Manager** – Select **(None)** from the drop-down menu.
  - **Survivability Server** – Select **(None)** from the drop-down menu.
  - **Origination Sequence –** Select Application Sequence defined for Communication Manager.
  - **Termination Sequence** – Select Application Sequence for Communication Manager.
  - **Home Location** – Select Location.

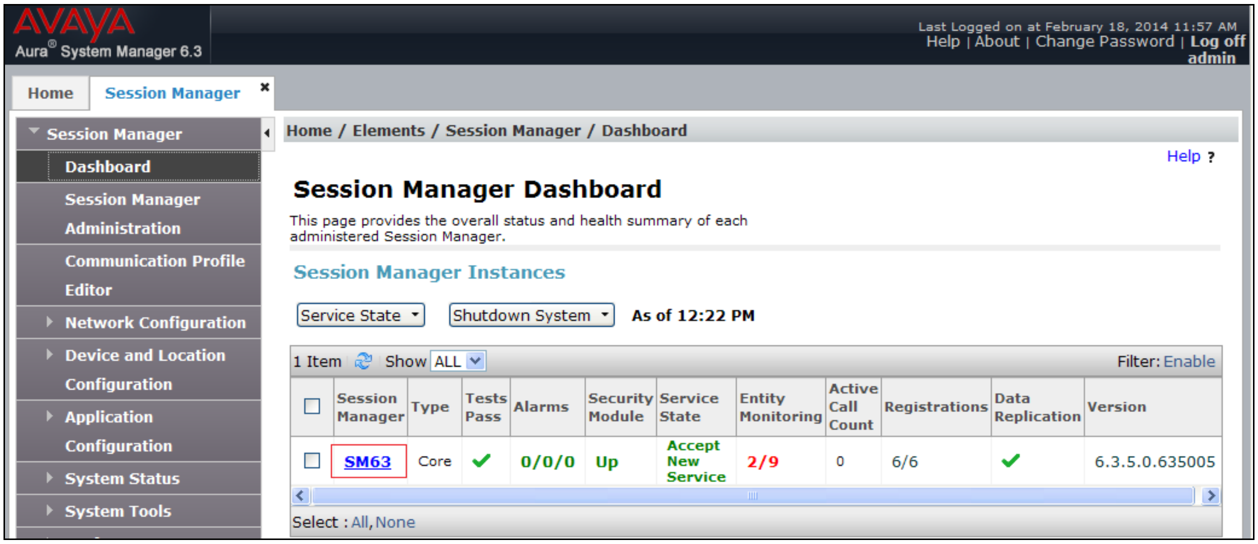


- Endpoint Profile section
  - **System** – Select Managed Element defined in System Manager.
  - **Profile Type** – Select **Endpoint**.
  - **Use Existing Endpoints -** Leave unchecked to automatically create new endpoint when new user is created. Or else, check the box if endpoint is already defined in Communication Manager.
  - **Extension** - Enter same extension number used in this section.
  - **Template** – Select template for type of SIP phone
  - **Security Code** – Enter numeric value used to logon to SIP telephone. (**Note:** this field must match the value entered for the Shared Communication Profile Password field).
  - **Port** – Select **IP** from drop down menu
  - **Delete Station on Unassign of Endpoint** – Check the box to automatically delete station when Endpoint Profile is un-assigned from user.

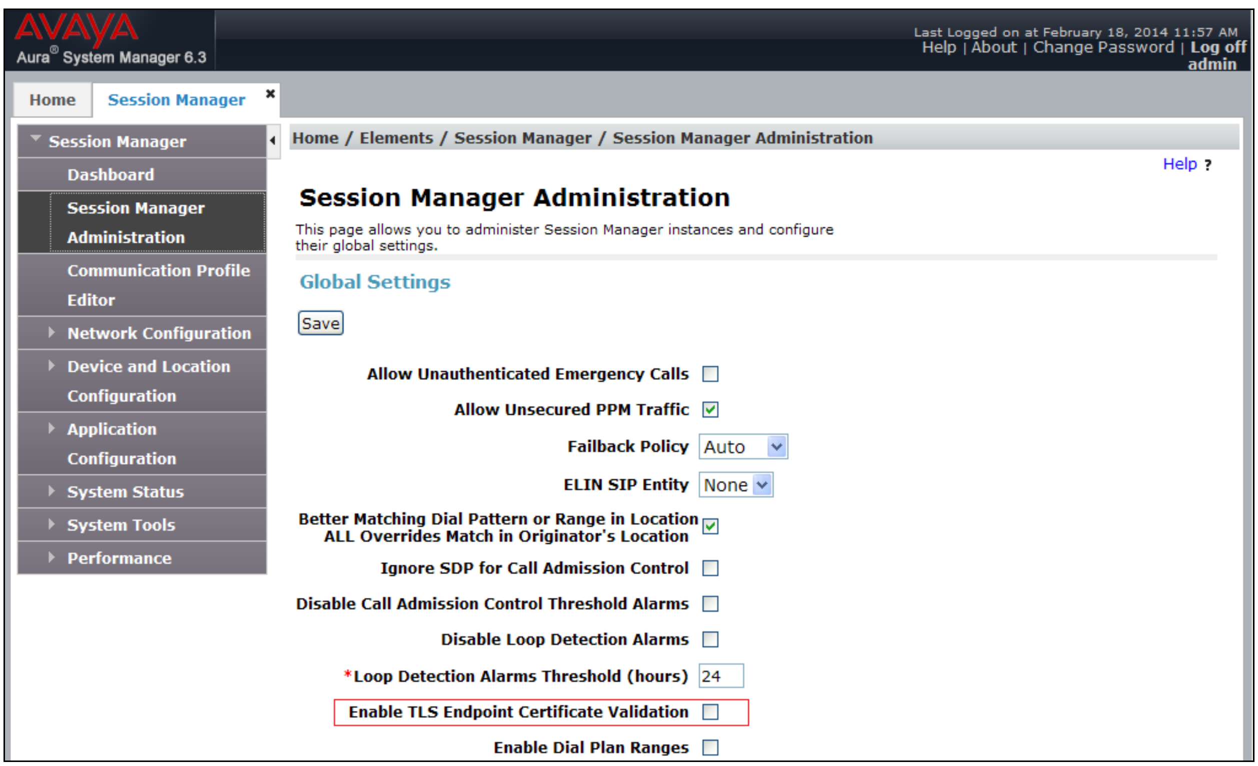


6.2. Configure Biamp Tesira SVC-2 for TLS

In System Manager, Navigate to **Home →Elements → Session Manager → Dashboard**. Select a **Session Manager**.



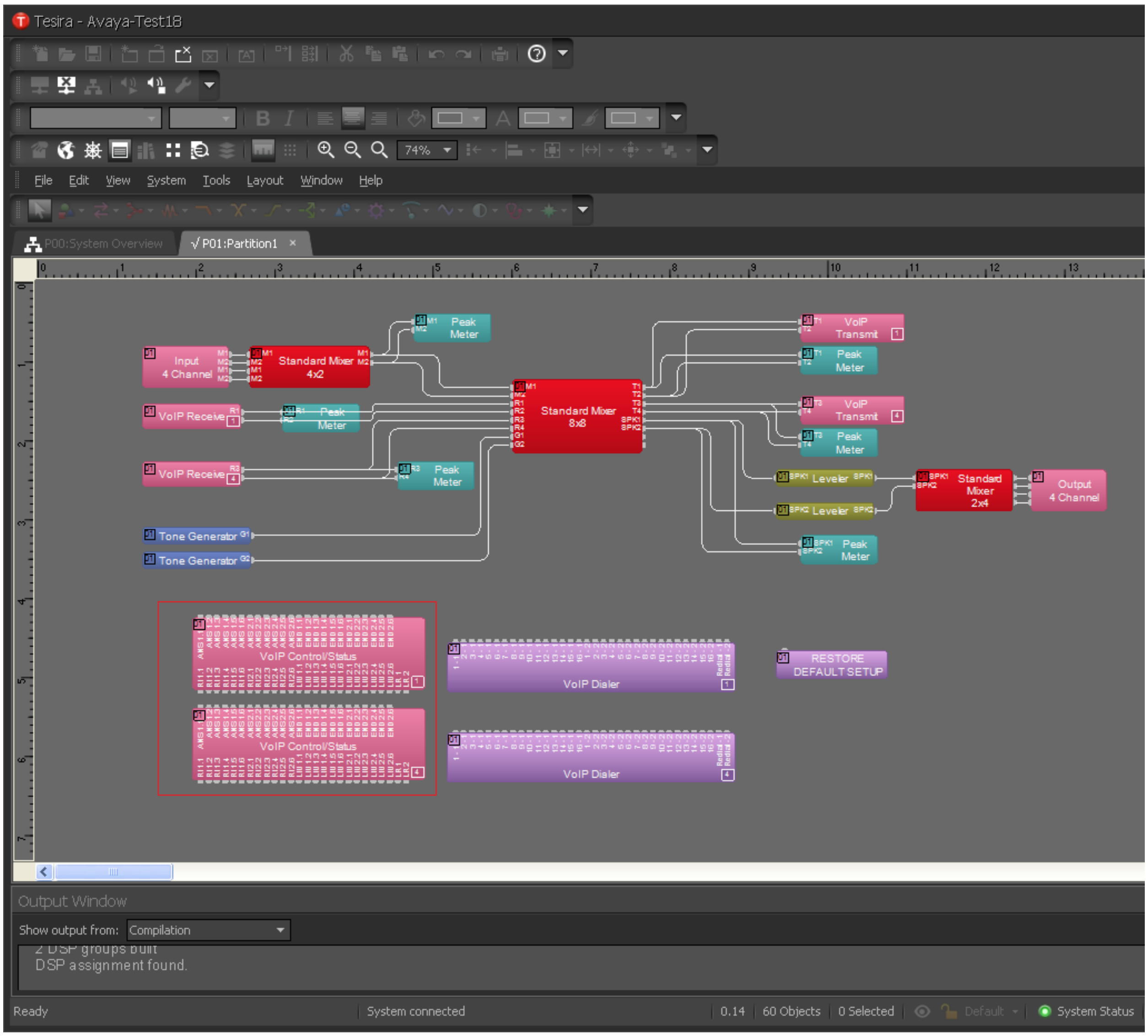
- From the Session Manager Administration page, verify that the Enable TLS Endpoint Certificate Validation field is not checked. By not checking, Session Manager does not request a certificate from the 3rd party endpoint.



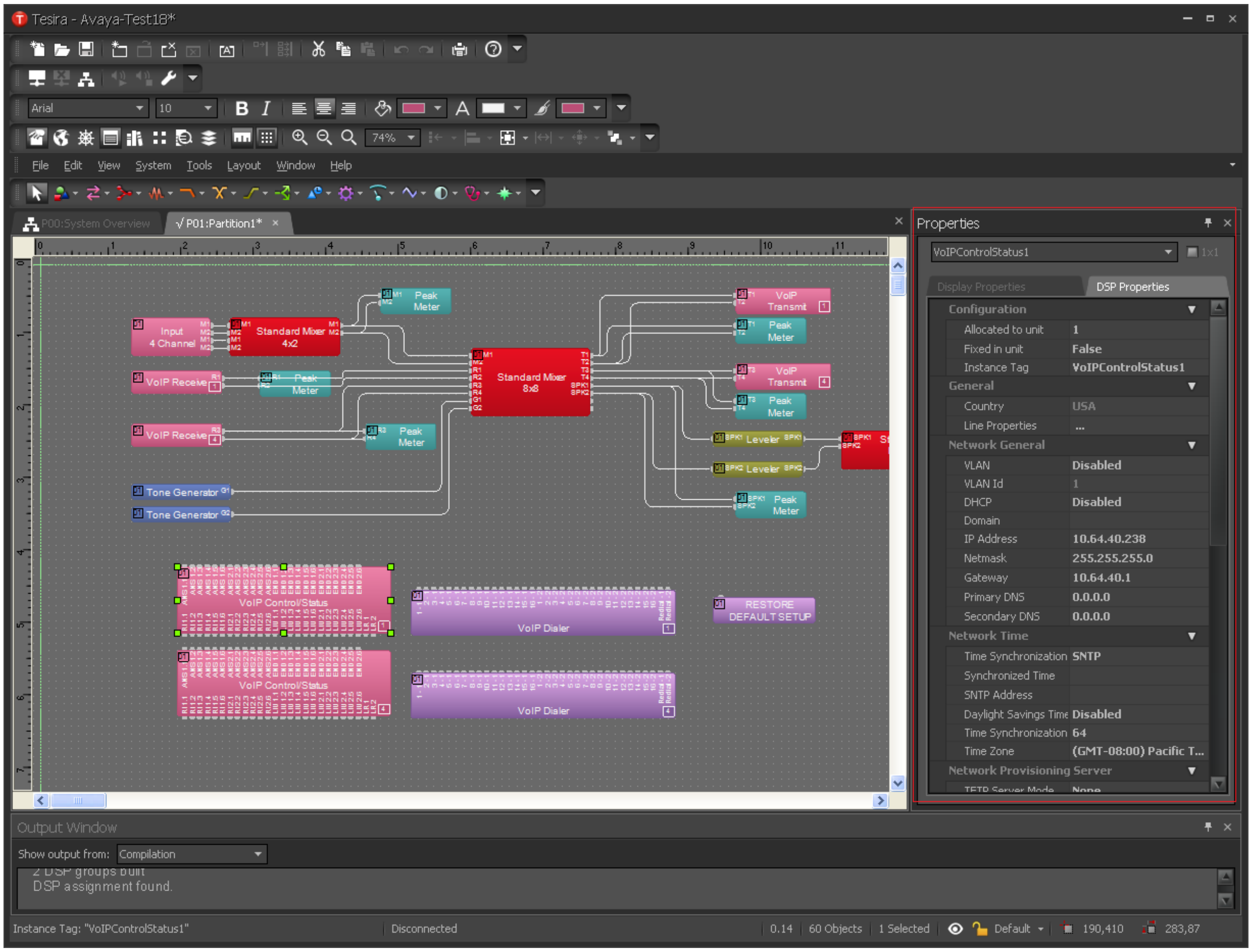
7. Configure Biamp Tesira SVC-2

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 Session Manager. 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 of this application note.

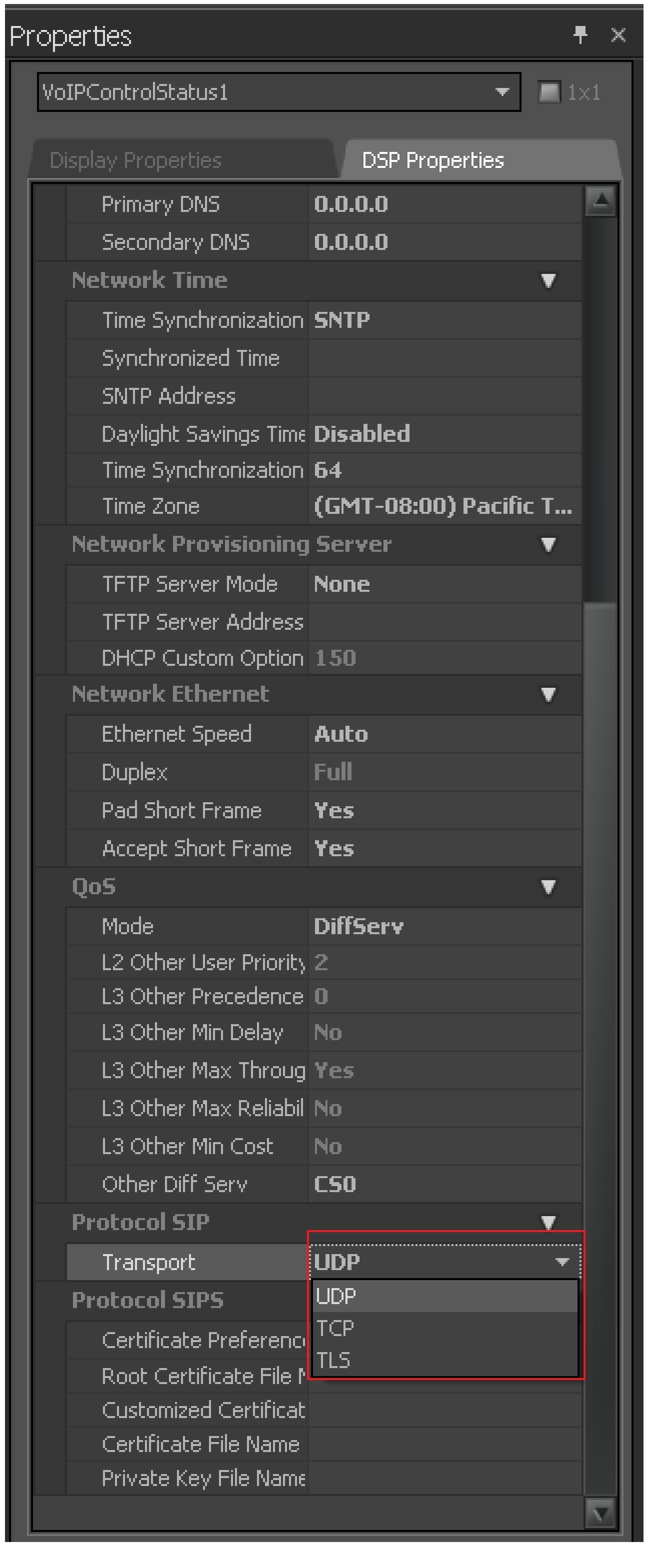
- Highlight the **VoIPControl/Status** block, as shown below.



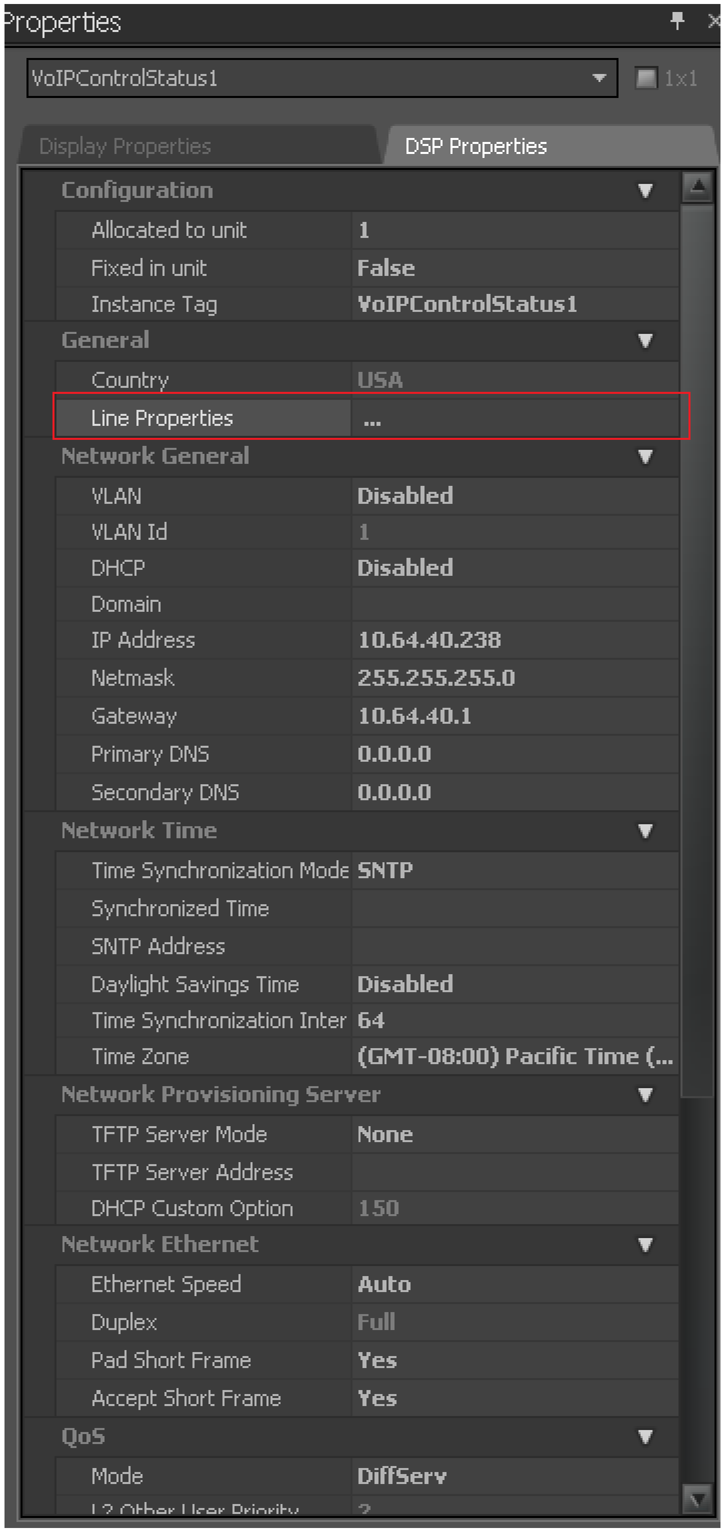
- Click right mouse button and select **Properties**, and the Properties menu will display on the right



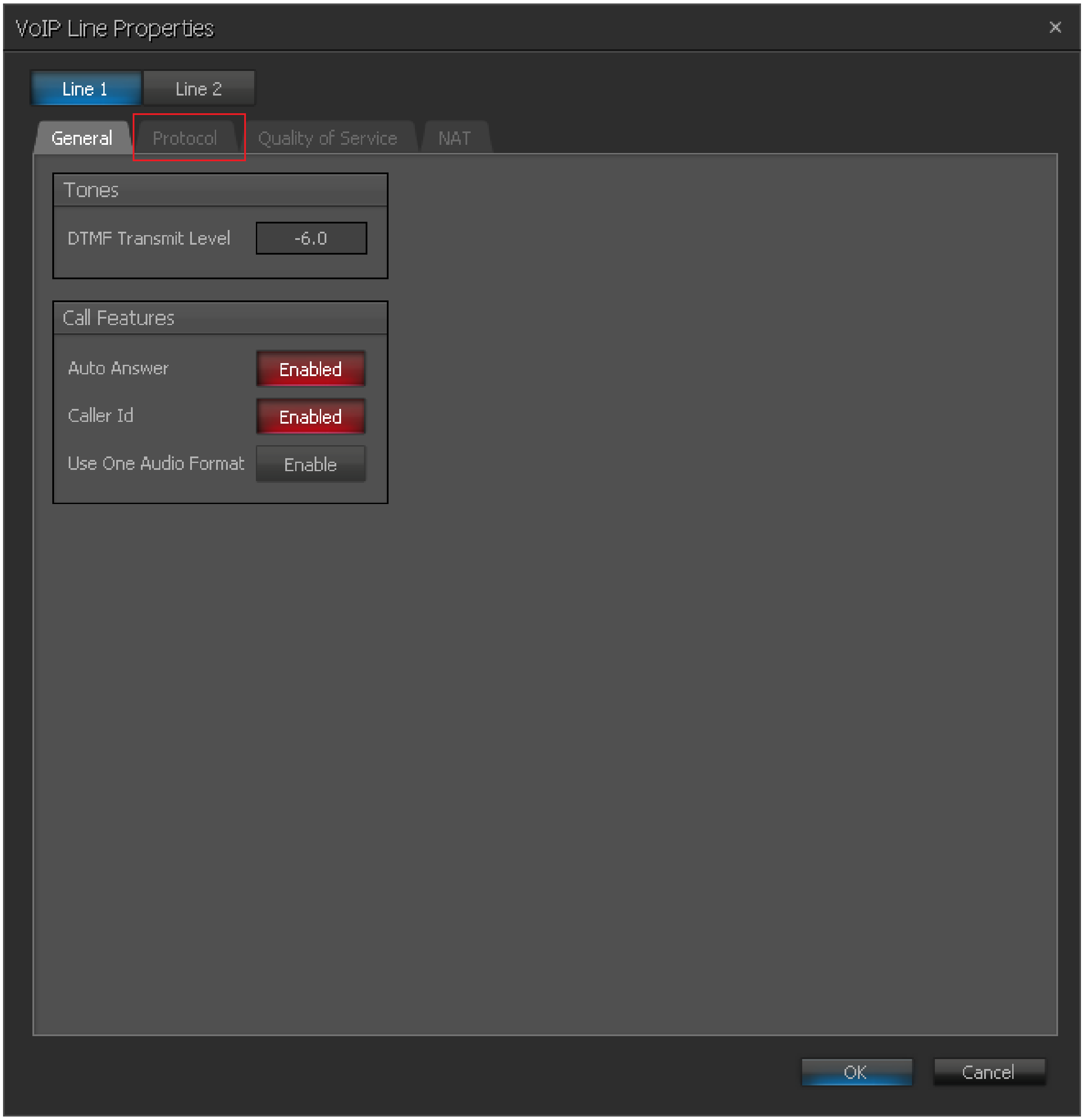
- Navigate the **Protocol SIP→Transport** to configure transport to be used. The default is UDP. During the compliance test, UDP was utilized. When TLS is selected, please refer to Tesira Operational Manual for additional configuration.



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

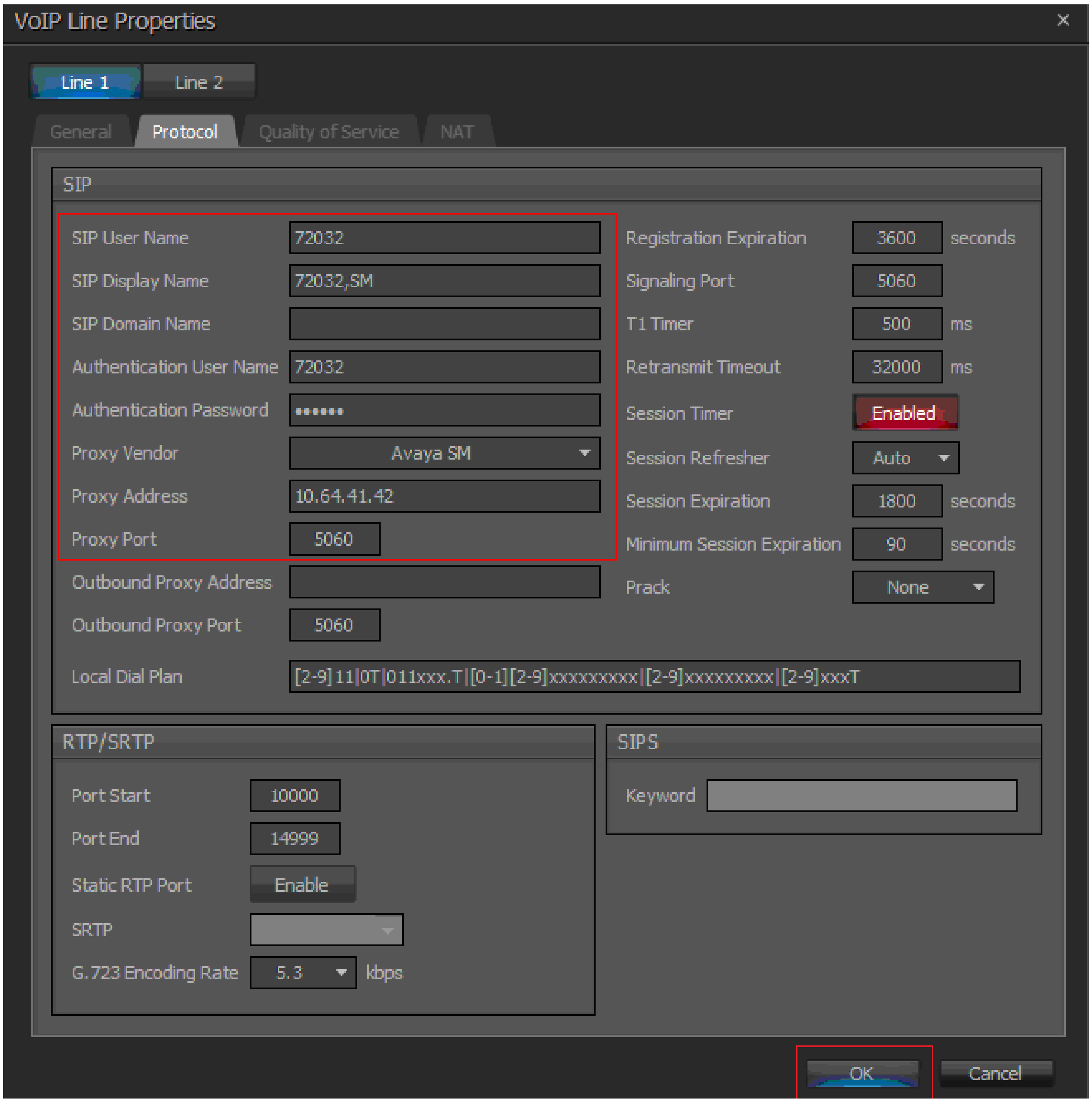


- From the Line Properties page, click the **Protocol** tab.



- From the Protocol page, provide the following information:
  - **SIP User Name** – Enter a user created in **Section 6.1**.
  - **Authentication User Name** – Enter a user created in **Section 6.1**.
  - **Authentication Password** – Enter the password for the user in **Section 6.1**
  - **Proxy Vendor** – Select Avaya SM
  - **Proxy Address** – Enter the IP address of Session Manager.
  - **Proxy Port** – Enter either 5060 or 5061.
  - 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 Session Manager server by navigating to **Home → Elements → Session Manager → System Status → User Registrations** System Manager.
- 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 Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3. Compliance testing between Biamp and Avaya Aura® Session Manager and Avaya Aura® Communication Manager was successful as per the tests outlined in **Section 2**.

10. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com

[1] *Administering Avaya Aura® Communication Manager*, Release 6.3, October 2013, Issue 9,Document Number 03-300509

[2] *Administering Avaya Aura® Session Manager*, Release 6.3, October 2013, Issue 3, Document Number03-603324

[3] *Administering Avaya Aura® System Manager*, Release 6.3, October 2013, Issue 3

The following document was provided by Biamp.

[4] *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.