Avaya Solution & Interoperability Test Lab

Application Notes for Biamp Tesira SVC-2 and Avaya Communication Server 1000 SIP Line Release 7.6 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.6 and Biamp Tesira SVC-2. The overall objective of the interoperability compliance testing is to verify Biamp Tesira SVC-2 functionalities in an environment comprised of Avaya Communication Server 1000 SIP Line with various Avaya Unistim and SIP IP Telephones.

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 procedures for configuring Biamp Tesira SVC-2 which was compliance tested with Avaya Communication Server 1000 (CS 1000) SIP Line Release 7.6.

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 the Tesira SERVER-IO telephone conferencing product powerful, flexible, and affordable. 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.

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 CS 1000 SIP Line and Avaya SIP and Unisitm telephones were verified.

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 focus of this testing was to verify that the Biamp Tesira SVC-2 was able to interoperate with the CS 1000 SIP Line system. The following areas were tested:

- Registration of Biamp Tesira SVC-2 Lines to the CS1000 SIP Line Gateway.
- Call establishment of Biamp Tesira SCV-2 with CS1000 SIP and Unistim telephones.
- Telephony features: basic calls, conference, DTMF (Dual-Tone Multi-Frequency), Messaging, Call Busy, Hold Call and Call waiting.
- PSTN calls over SIP trunk.
- Codec negotiation – G.711 and G.729.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. The following observations were made during the compliance test:

- UDP transport protocols was used for SIP Line User on Tesira.
- Since Message Waiting Indicator (MWI) is not supported on Tesira, the compliance test only verified that user was able to send and receive messages.
- Local conference made by Biamp Tesira SVC-2 has a choppy and noisy audio if different parties used different codec. All parties need to use the same codec to ensure audio quality.

2.3. Support

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

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

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance test between the Avaya CS1000 SIP Line Release 7.6 and the Biamp Tesira SVC-2.

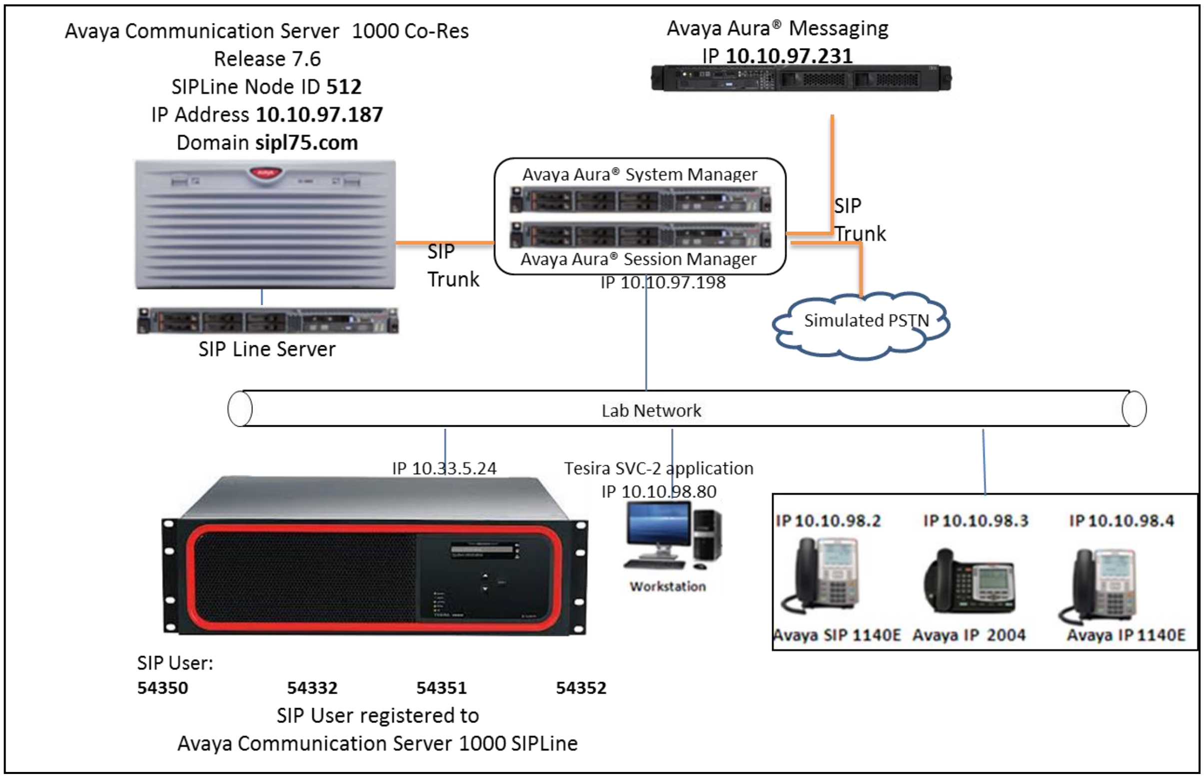


Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

The following equipment and software was used during the lab testing:

Equipment / Software	Release / Version
Avaya CS1000E	Call Server (CPPM): 7.65P+ Signaling Server (CPPM): 7.65
Avaya CS 1000 SIP Line	7.65P+
Avaya Aura® Messaging	6.3
Avaya IP Phone 1140E Avaya IP Phone 2004P2	0625C6O 0692D93
Avaya SIP 1140	02.02.21.00
Biamp Tesira SVC-2	1.2.2
Biamp Tesira	2.1

5. Configure SIP Phones for Tesira on Avaya Communication Server 1000

This document assumes that the CS1000 and SIP Line server has:

- Been installed with CS1000 Release 7.6 Linux Base.
- Joined CS1000 Release 7.6 Security Domain.
- Been deployed with SIP Line Application.

The SIP Line Node has been set up and in operational state. The IP address of SIP Line used in this test is **10.10.97.187** and its domain is **sipl75.com**, as displayed in **Figure 1.**

For more detail on how to configure SIP Line Node, D-Channel, SIP Trunk, Application Module Link, Value Added Server and Zone, see the document ***Application Notes for Biamp Tesira SVC-2 and Avaya Communication Server 1000 SIP Line Release 7.5 – Issue 1.0*** available on support.avaya.com.

The following is a summary of tasks for configuring the CS1000 SIP User for Tesira SVC-2.

To create a SIP Line phone on the Call Server, log in as administrator and use overlay command **LD 20** as shown below.

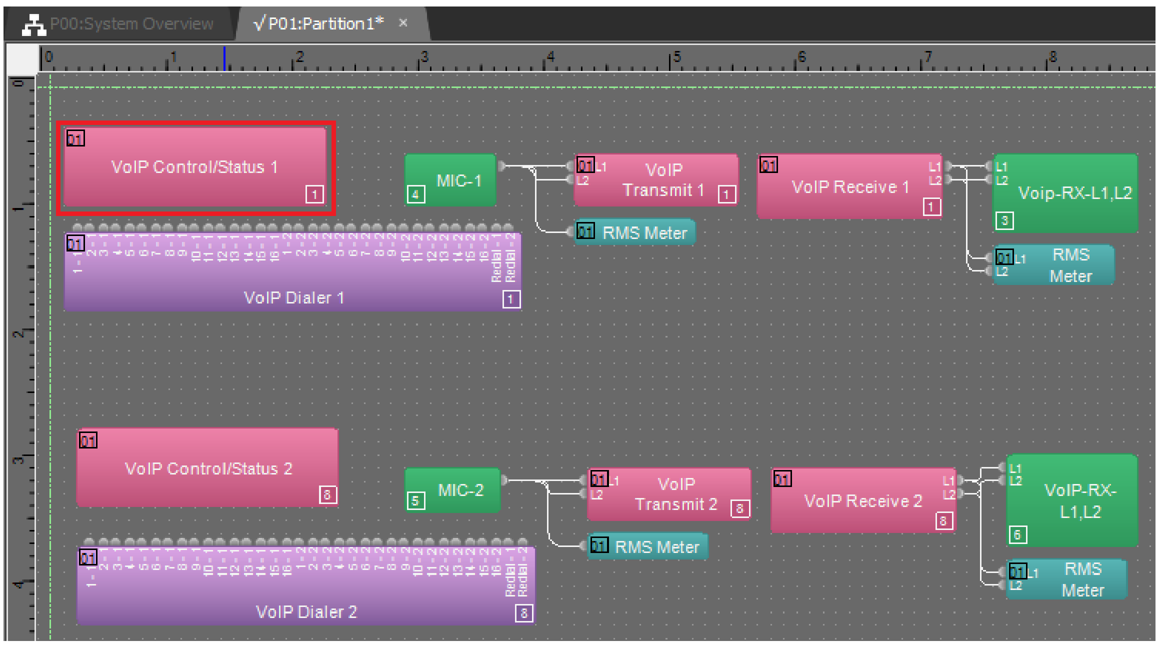
The bold fields must be properly inputted as they are configured on the Call server, for other fields press <Enter> to accept default values.

```
>LD 20
REQ: NEW
TYPE: UEXT  - Universal extension type for SIP Line phone
TN   104 0 0 14
DES  Tesira1
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL – Universal extension type is SIP Line
MCCL YES
SIPN 0 – For SIP phone third party, enter 0 in this field
SIP3 1 – For SIP phone third party, enter 1 in this field
FMCL 0
TLSV 0
SIPU 54350 – SIP phone user name
NDID 512 – Node ID of SIP Line
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
SCPW 1234 – Password of SIP Line user. Need to register Tesira to SIP Line Node
CLS dnda cnda swa - Class of Service SWA for Call Waititng
VSIT NO
KEY  00 SCR 54350 0     MARP
        CPND new
          CPND_LANG
            NAME Tesira 1
            XPLN 13
            DISPLAY_FMT FIRST,LAST
KEY  01 HOT U 2654350 MARP 0 - Key 1 hot U with prefix + DN
KEY  02 CWT – Key 2 for Call Waiting
```

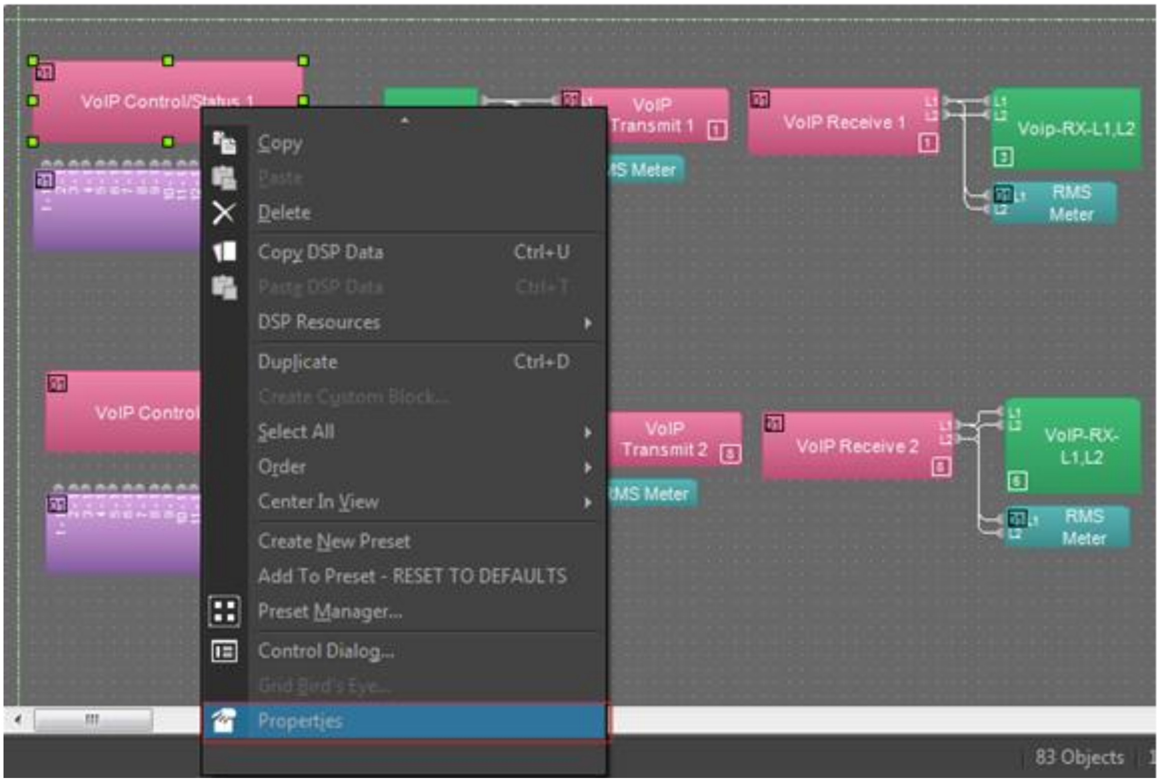
6. Configure Biamp Tesira SVC-2

Biamp installs, configures, and customizes the Tesira SVC-2 application for the end customers. How to configure a Tesira system is beyond the scope of these Application Notes. This section only provides steps to configure Biamp Tesira SVC-2 to interface with Avaya CS 1000 SIP Line server. For more information on how to administer Biamp Tesira SVC-2, please refer to the Tesira SVC-2 documents from Biamp.

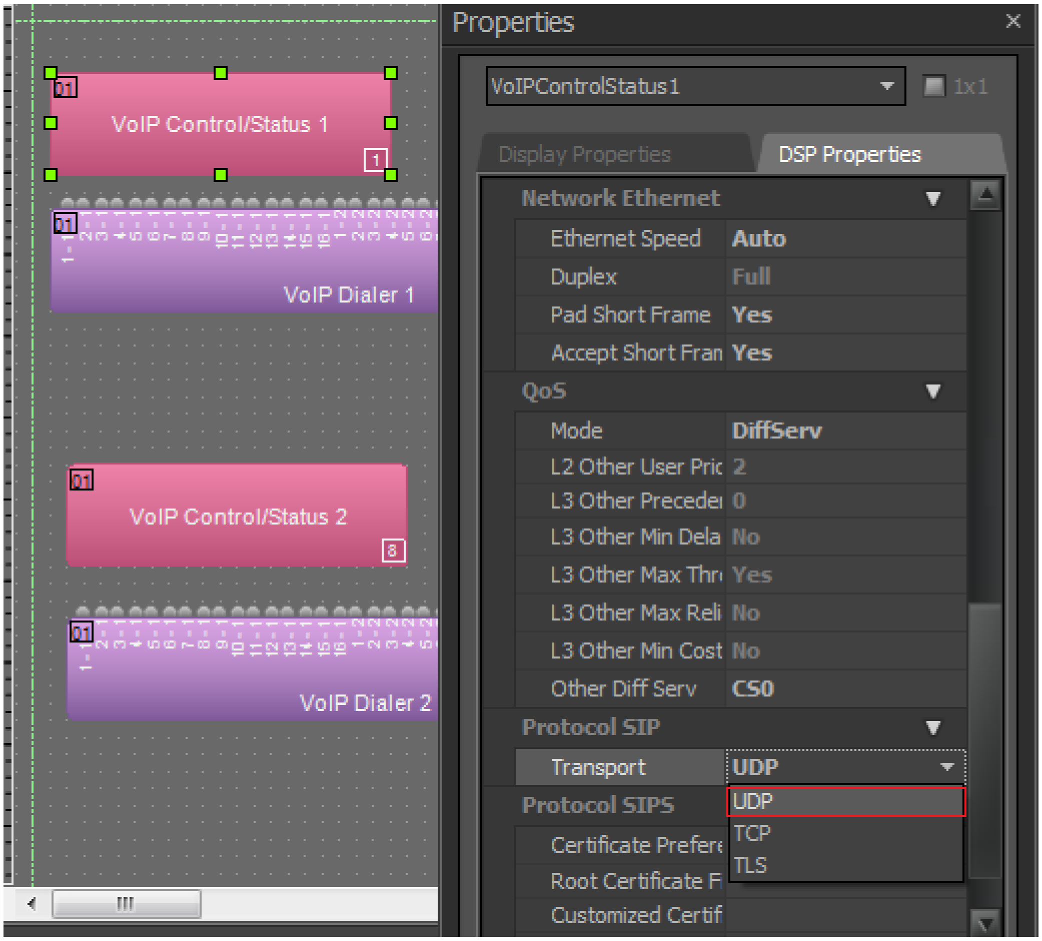
Select the Tesira icon from the workstation where Tesira software was installed to start Tesira software. Highlight the **VoIPControl/Status** block, as shown below.



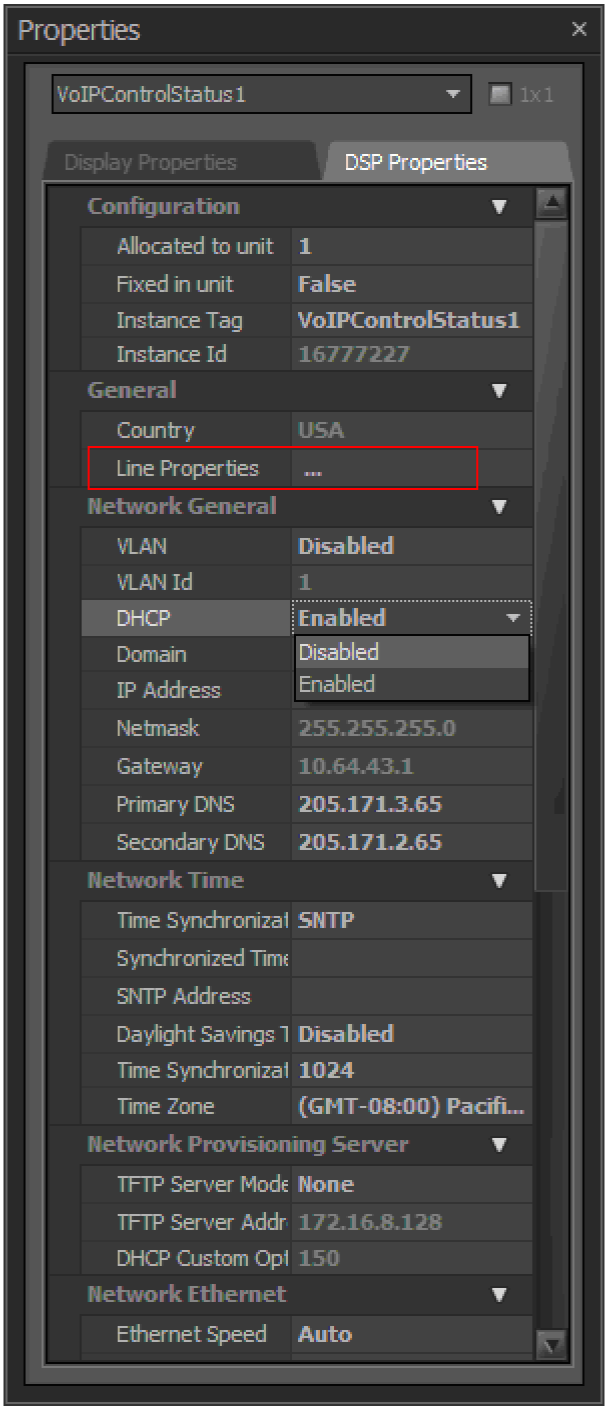
Righ-click mouse button and select **Properties**.



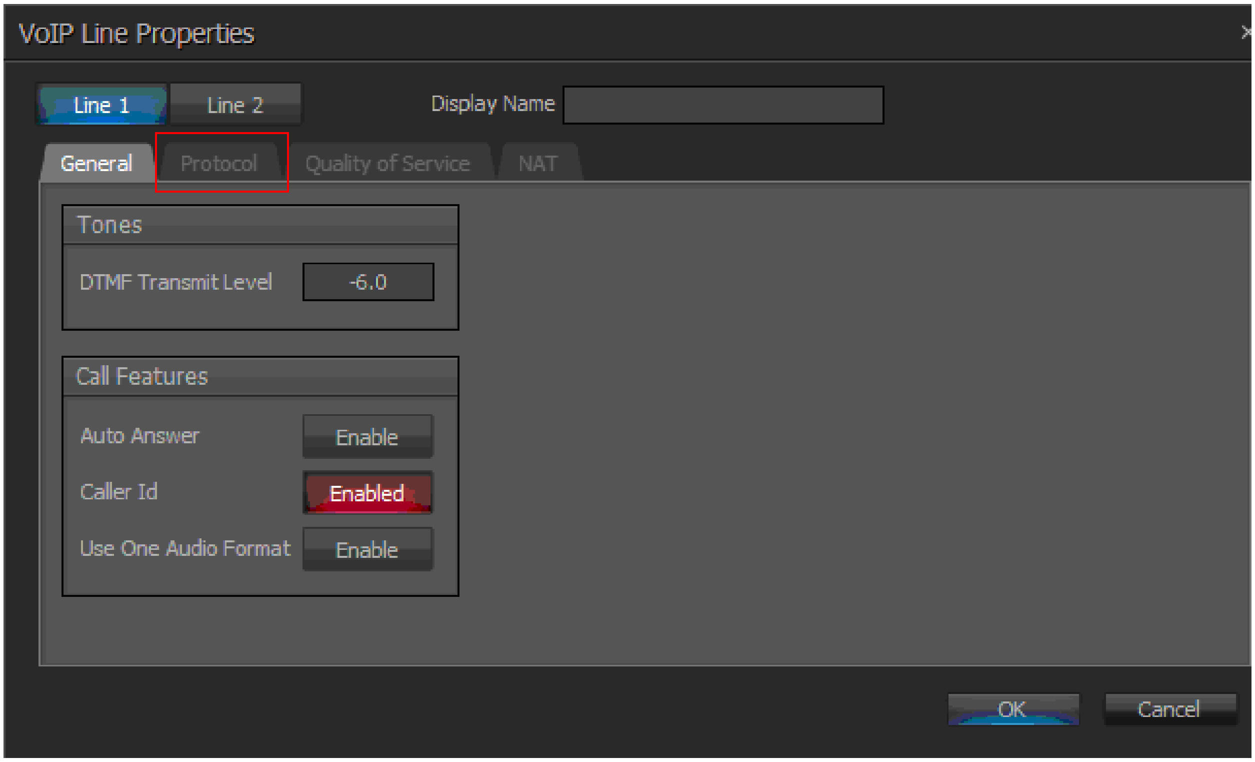
The **Properties** menu will display on the right. Form the **Properties** pane, select the **DSP Properties** tab and navigate to **Protocol SIP→Transport** to configure transport to be used. Select UDP transport (default setting).



In the **DSP Properties** tab Sselect **Line Properties** in the **General** section.



The **VoIP Line Properties** window is displayed. Select the **Protocol** tab.

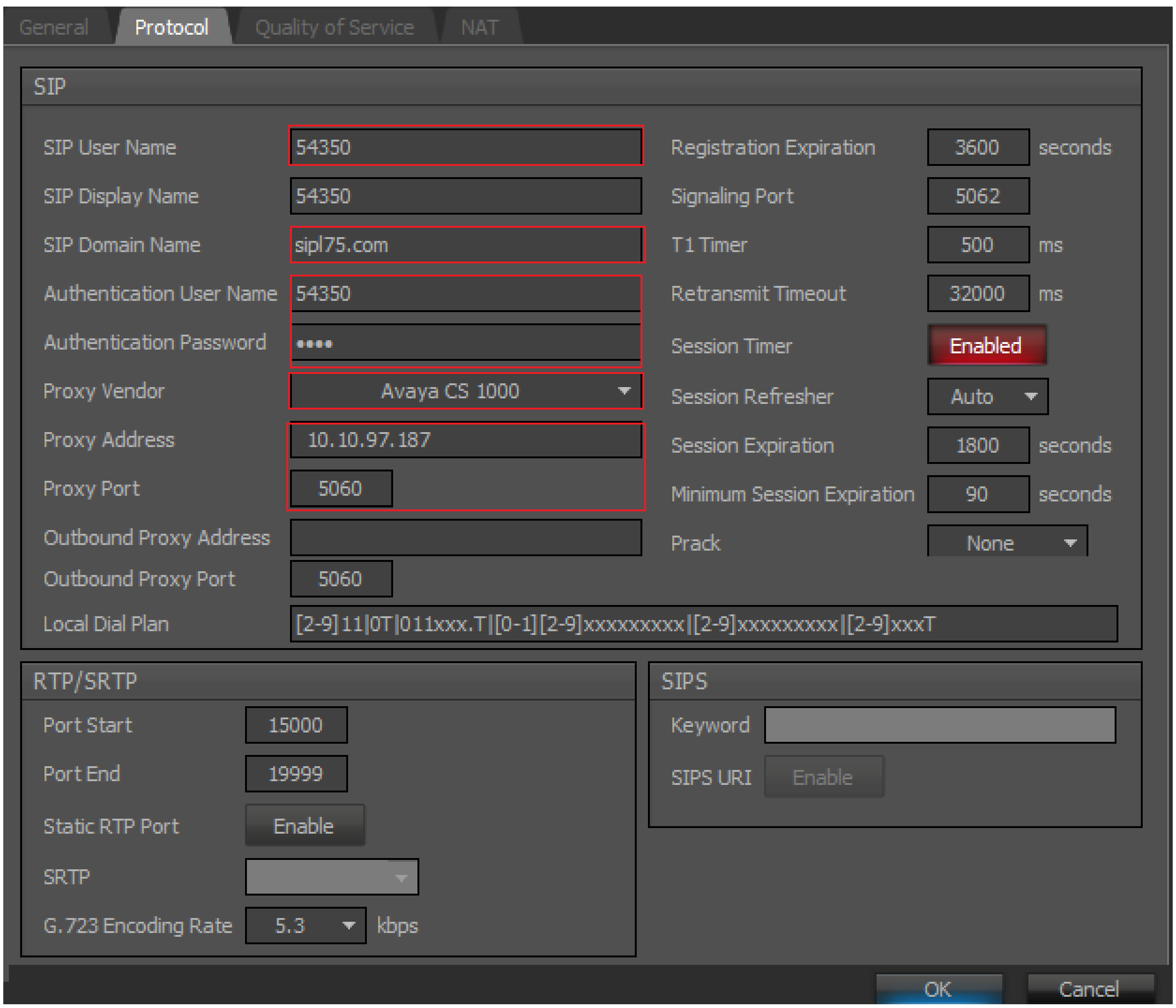


In the **Protocol** tab, provide the following information:

- **SIP User Name:** Enter a user created in **Section 0**.
- **Authentication User Name** – Enter a user created in **Section 0**.
- **SIP Domain Name** – Enter the SIP domain name of CS1000 system, example ***sipl75.com***
- **Authentication Password** – Enter the password for the user created **Section 0**
- **Proxy Vendor** – Select Avaya CS 1000.
- **Proxy Address** – Enter the IP address of CS 1000 SIP Line server.
- **Proxy Port** – Enter 5060 for UDP
- Default values may be used for all other fields.

Click on the **OK** button to save changes.

Note: *Biamp Tesira SVC-2 can provide two inbound extensions (Line 1 and Line 2).*



7. Verification Steps

This section includes some steps that can be followed to verify the configuration.

Verify that the Biamp Tesira SVC-2 registers successfully with the CS 1000 SIP Line Gateway server. Log in to the SIP Line server as an administrator by. Issue command **slgSetShowByUID *userID*** where ***userID*** is the SIP Line user's ID being checked.

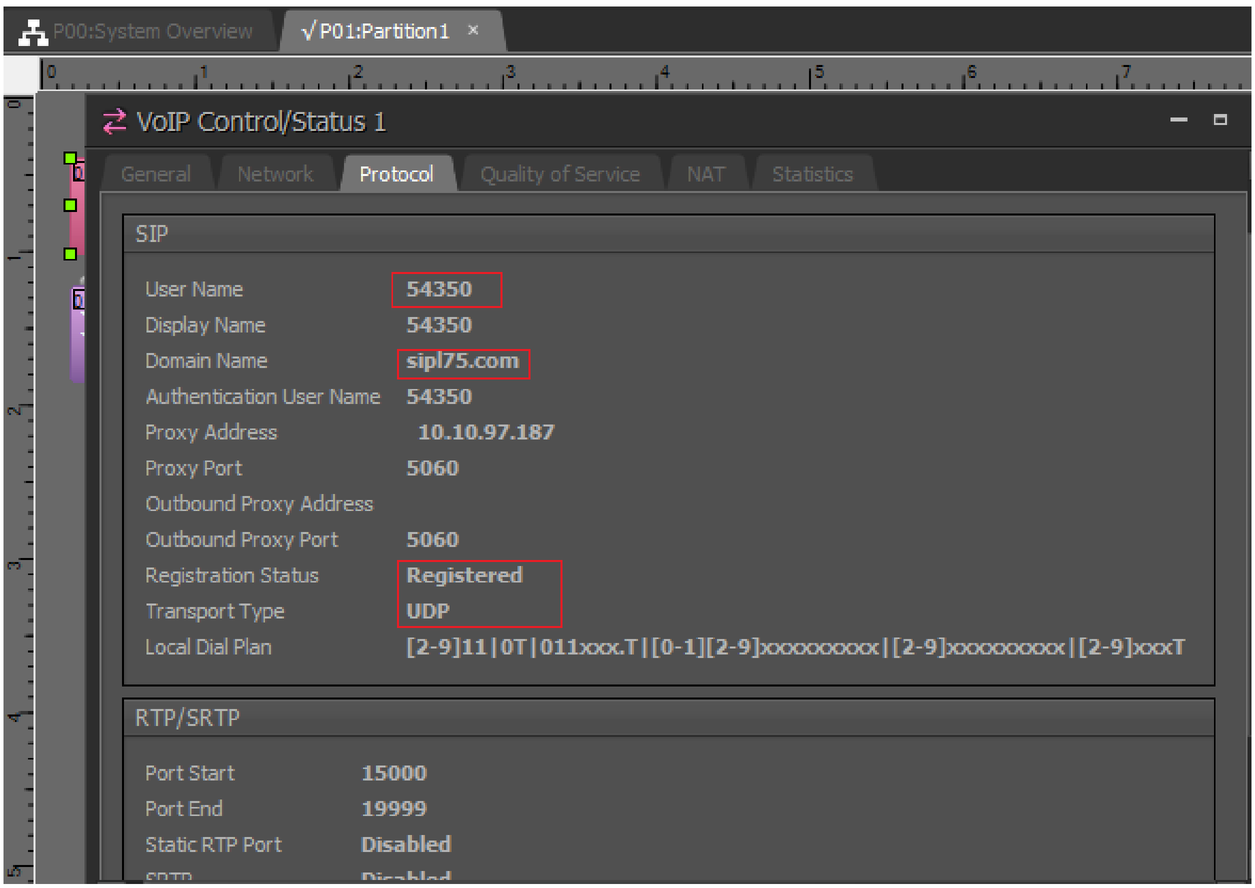
```
[admin2@sipl75 ~]$ slgSetShowByUID 54350
=== VTRK ===
UserID           AuthId      TN               Clients  Calls  SetHandle  Pos ID
SIPL Type
---------------  ----------  ---------------  -------  -----  ---------  -------
          54350       54350     104-00-00-10        1      0  0x8d35ee0
SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = -1
        Current Client = 0, Total Clients = 1
         == Client 0 ==
         IPv4:Port:Trans = 10.33.5.73:5062:udp
         Type          = Unknown
         UserAgent     = Tesira/1.2.2.12
         x-nt-guid     = da9a2b408582f5773fd645e4ff59b139
         RegDescrip    =
         RegStatus     = 1
         PbxReason     = OK
         SipCode       = 200
         hTransc       = (nil)
         Expire        = 3600
         Nonce         = d7f107851a9b4825c30f5bebd42e9d99
         NonceCount    = 2
         hTimer        = 0x8ccb000
         TimeRemain    = 2844
         Stale         = 0
         Outbound      = 0
         ClientGUID    = 0
         MSec CLS      = MSNV (MSEC-Never)
         Contact       = sip:54350@10.33.5.73:5062;transport=udp;rinstance=39003
02786
         KeyNum        = 255
         AutoAnswer    = NO

        Key  Func  Lamp  Label
        0    3     0     54350
        1    126   0     2654350
        3    29    0
        4    9     0
        17   16    0
        18   18    0
        19   27    0
         == Subscription Info ==
         Subscription Event  = None
         Subscription Handle = (nil)
         SubscribeFlag = 0
```

Verify that the Biamp Tesira SVC-2 registers successfully using CS1000 overlay LD 32. Log in to the call server using the admin account. Load **LD 32** and then issue command **stat *TN*** where ***TN*** is the SIP Line user's TN being checked

```
>ld 32
NPR000
.stat 104 0 0 10
IDLE REGISTERED 00
```

In the Tesira SVC-2 application, verify that the Biamp Tesira SVC-2 registers successfully to CS1000 by double-click on **VoIP Control/Status 1 → Protocol**, verify that SIP user ***54350*** status is ***Registered*** with **UDP** Transport Type.



Place a call from and to a Biamp Tesira SVC-2 SIP user and verify that the call is established with 2-way speech path.

During the call, use a sniffer tool (ethereal/wireshark) at the SIP Line Gateway and clients to make sure that all SIP request/response messages are correct.

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 2.1**, with some exceptions outlined in **Section 2.2**. The Biamp Tesira SVC-2 is considered to be in compliance with Avaya CS 1000 SIP Line System Release 7.6.

9. Additional References

Product documentation for the Avaya CS 1000 products may be found at:
https://support.avaya.com/css/Products/

Avaya CS1000 Documents:

[1] Avaya Communication Server 1000E Installation and Commissioning.
[2] Avaya Communication Server 1000 SIP Line Fundamental, Release 7.6.
[3] Avaya Communication Server 1000 Element Manager System Reference – Administration.
[4] Avaya Communication Sever 1000 Co-resident Call Server and Signaling Server Fundamentals.
[5] Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals.
[6] Avaya Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning.
[7] Application Notes for Biamp Tesira SVC-2 and Avaya Communication Server 1000 SIP Line Release 7.5 – Issue 1.0

Product documentation for the Biamp Tesira SVC-2 products may be found at:
http://www.biamp.com

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.