



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

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# **Application Notes for Biamp Tesira SVC-2 and Avaya IP Office R9.0 – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Biamp Tesira SVC-2 which were compliance tested with Avaya IP Office R9.0.

The overall objective of the interoperability compliance testing is to verify Biamp Tesira SVC-2 functionalities in an environment comprised of Avaya IP Office and various Avaya H.323, SIP IP Telephones, and DCP 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 IP Office.

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.

For further details on Tesira SVC-2 configuration steps not covered in this document, consult **Section 9 [2]**.

These Application Notes assume that Avaya IP Office is already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult **Section 9 [1]**.

## 2. General Test Approach and Test Results

The general test approach was to place calls to and from Biamp Tesira SVC-2 and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Codecs (G.711MU,G.729)
- Inbound calls
- Outbound calls
- Hold/Resume
- Call termination (origination/destination)
- Serviceability

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

All test cases were performed manually. The general test 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/resume, and Biamp Tesira SVC-2 interactions with Avaya IP Office. Avaya SIP, H.323, and DCP telephones were also verified during testing. For serviceability testing, failures such as cable pulls and resets were applied.

## 2.2. Test Results

All test cases passed.

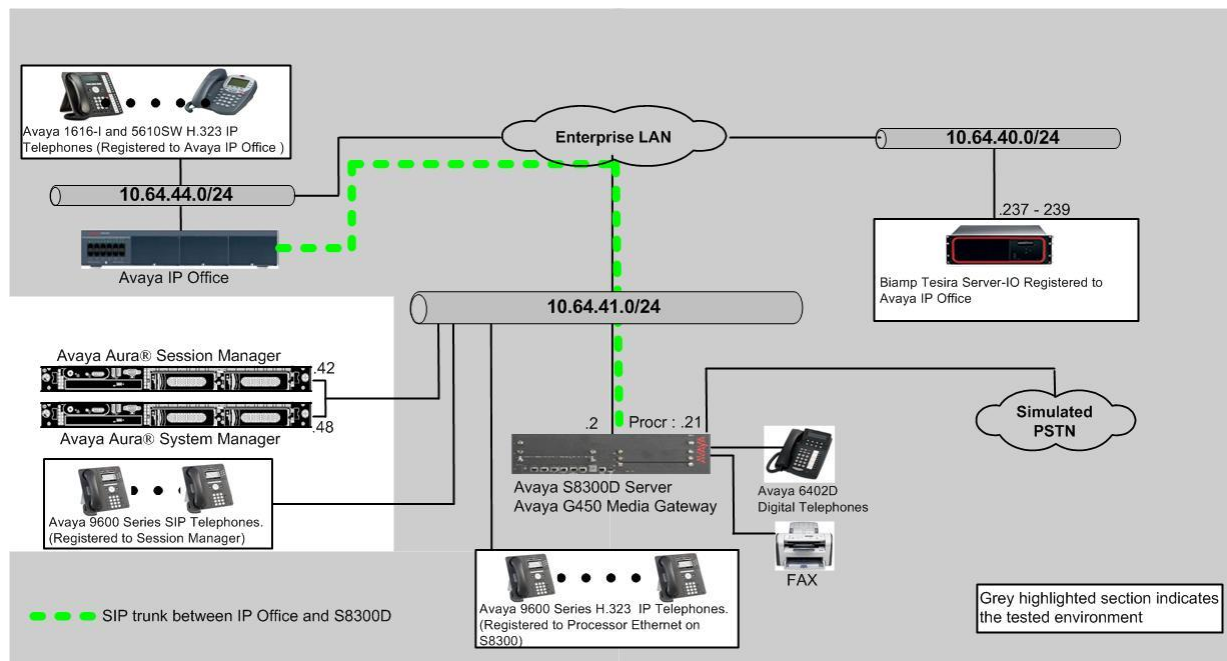
## 2.3. Support

Technical support for the 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 IP Office and Biamp Tesira SVC-2. Avaya S8300D Server with an Avaya G450 gateway was included in the test to provide an inter-switch test scenario. For completeness, Avaya 5610 and 1616-I H.323 IP Telephones, Avaya 9600 Series SIP IP Telephones, Avaya 9600 Series H.323 IP Telephones, Avaya 6400 and 1416 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between Biamp Tesira SVC-2 and Avaya SIP, H.323, and digital telephones.



**Figure 1: Test Configuration of Biamp Tesira SVC-2 with Avaya IP Office**

## 4. Equipment and Software Validated

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

Equipment/Software		Release/Version
Avaya IP Office 500 V2		9.0 (829)
Avaya IP Office Manager		9.0 (829)
Avaya Aura® Communication Manager running on Avaya S8300D Media Server with Avaya G450 Media Gateway		6.3 (03.0.124.0-21172)
Avaya H.323 IP Telephones on IP Office		
	5610 (H.323)	2.9.1
	1616-I (H.323)	1.22
Avaya 1416 Digital Telephone on IP Office		-
Avaya 9600 Series H.323 Telephones on S8300D server		
	9620 (H.323)	3.1
	9630 (H.323)	3.1
	9650 (H.323)	3.1
Avaya 9600 Series SIP Telephones		
	9620 (H.323)	2.6.4
	9630 (H.323)	2.6.4
Avaya 6408D+ Digital Telephone		-
Biamp Tesira SVC-2		
Biamp Tesira		2.0.0
Biamp Linux		3.2.48-BIAMP

## 5. Configure Avaya IP Office

This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

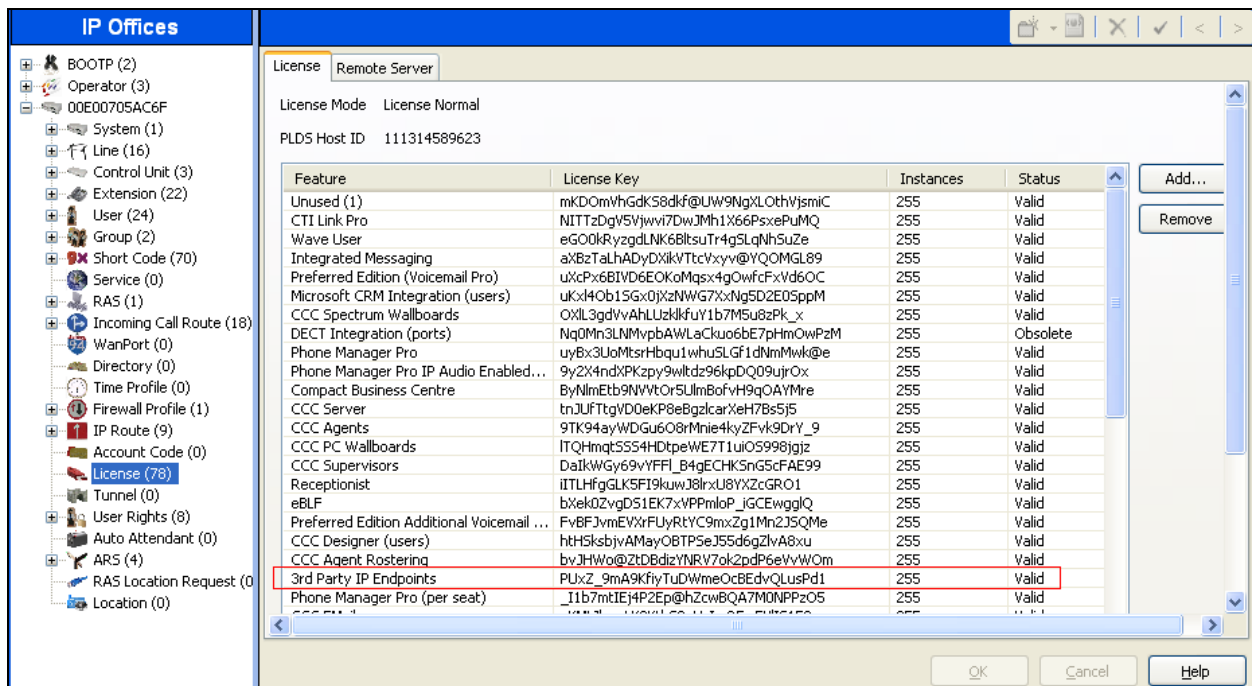
- Verify IP Office license
- Obtain LAN IP address
- Administer SIP registrar
- Administer SIP extensions
- Administer SIP users

These steps are performed from the Avaya IP Office Manager.

### 5.1. Verify IP Office License

From a PC running the Avaya IP Office Manager application, select **Start → All Programs → IP Office → Manager** to launch the Manager application. Select the proper IP Office system if there are more than one IP Office system, and log in with the appropriate credentials.

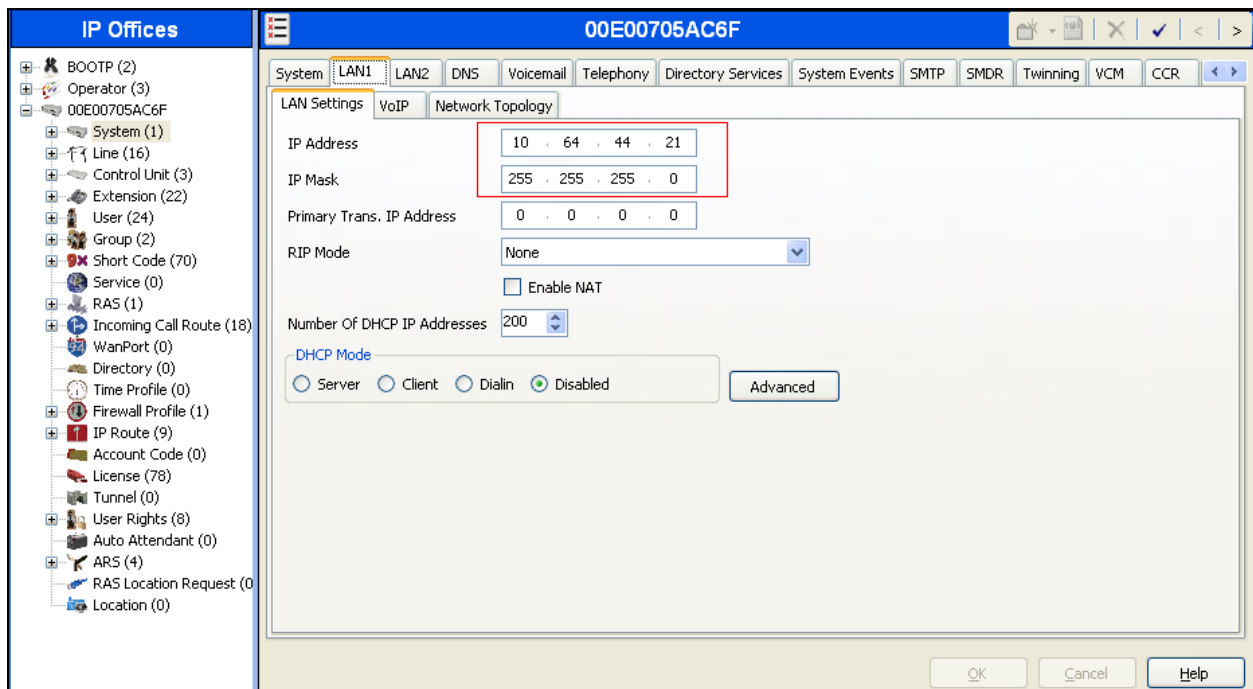
The Avaya IP Office Manager screen is displayed. From the configuration tree in the left pane, select **Licence → 3<sup>rd</sup> Party IP Endpoints** to display the Avaya IP endpoints screen in the right pane. Verify that the License Status field is set to **Valid**.



## 5.2. Obtain LAN IP Address

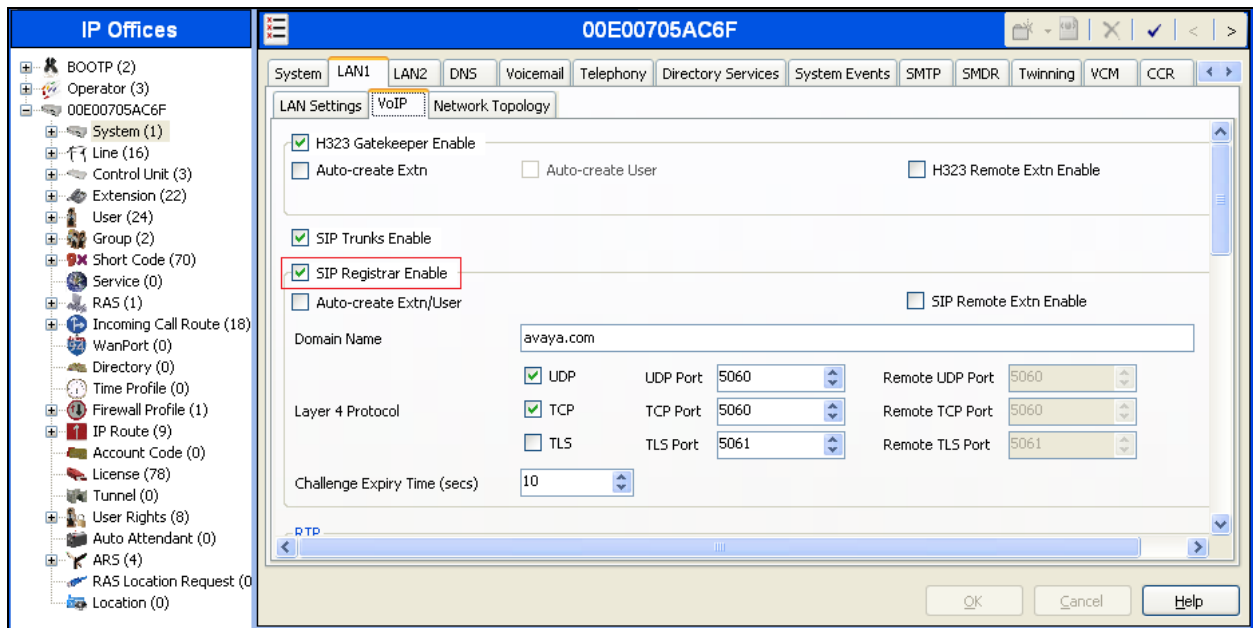
From the configuration tree in the left pane, select **System** to display the System screen in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure Biamp Tesira SVC-2 in **Section 6**.

**Note:** During the initial configuration of Avaya IP Office, the LAN1 was configured on the private network side and LAN2 was configured on the public network side. Avaya IP Office can support SIP extensions on the LAN1 and/or LAN2 interfaces, but the compliance test used the LAN1 interface. Thus, only the LAN1 configuration will be discussed in these Application Notes.



### 5.3. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** is checked, as shown below.



## 5.4. Administer SIP Extensions

From the configuration tree in the left pane, right-click on **Extension** and select **New → SIP Extension** from the pop-up list to add a new SIP extension (not shown). Enter the desired digits for the **Base Extension** field.

The screenshot shows the 'SIP Extension: 8003 77018' configuration window. The left pane shows the 'IP Offices' tree with 'Extension (22)' selected. The main pane has tabs for 'Extn', 'VoIP', and 'T38 Fax'. The 'Extn' tab is active, showing fields for 'Extension Id' (8003), 'Base Extension' (77018), 'Caller Display Type' (On), 'Reset Volume After Calls' (unchecked), 'Device Type' (Unknown SIP device), 'Location' (Automatic), 'Module' (0), 'Port' (0), and 'Force Authorization' (unchecked). Buttons for 'OK', 'Cancel', and 'Help' are at the bottom right.

Select the **VoIP** tab, and select codecs as shown below.

Repeat this section to add a new SIP extension for each Biamp Tesira SVC-2. During the compliance test, extensions 77018 and 77019 were created for Biamp Tesira SVC-2.

The screenshot shows the 'SIP Extension: 8003 77018' configuration window with the 'VoIP' tab selected. The 'IP Address' field is set to '0 . 0 . 0 . 0'. The 'Codec Selection' dropdown is set to 'System Default'. Below this, there are two lists: 'Unused' (G.722 64K) and 'Selected' (G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP, G.723.1 6K3 MP-MLQ). The 'Selected' list is highlighted with a red box. Other fields include 'Reserve License' (None), 'Fax Transport Support' (None), 'TDM->IP Gain' (Default), and 'IP->TDM Gain' (Default). Checkboxes for 'VoIP Silence Suppression', 'Local Hold Music', 'Allow Direct Media Path', 'Re-invite Supported', and 'Codec Lockdown' are on the right. Buttons for 'OK', 'Cancel', and 'Help' are at the bottom right.



## 5.5. Administer SIP Users

From the left pane, right-click on **User**, and select **New** from the pop-up list (not shown). Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension created in **Section 5.4**.

The screenshot shows the 'Ext218: 77018' configuration window. The left pane shows a tree view of 'IP Offices' with 'User (24)' expanded. The main pane has tabs for 'User', 'Voicemail', 'DND', 'Short Codes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', 'Voice Recording', 'Button Programming', and 'Menu'. The 'User' tab is active, showing fields for Name (Ext218), Password (\*\*\*\*\*), Confirm Password (\*\*\*\*\*), Account Status (Enabled), Full Name (Biamp1), Extension (77018), Email Address, Locale, Priority (5), System Phone Rights (None), and Profile (Basic User). Below these are checkboxes for Receptionist, Enable Softphone, Enable one-X Portal Services, Enable one-X Telecommuter, Enable Remote Worker, Enable Flare, Enable Mobile VoIP Client, Send Mobility Email, and Ex Directory. The 'OK', 'Cancel', and 'Help' buttons are at the bottom right.

Select the **Telephony** tab, followed by the **Call Settings** sub-tab. Check the **Call Waiting On** field, as shown below.

The screenshot shows the 'Ext218: 77018' configuration window with the 'Telephony' tab selected. The 'Call Settings' sub-tab is active, showing fields for Outside Call Sequence (Default Ring), Inside Call Sequence (Default Ring), Ringback Sequence (Default Ring), No Answer Time (secs) (System Default (15)), Wrap-up Time (secs) (2), Transfer Return Time (secs) (Off), and Call Cost Mark-Up (100). The 'Call Waiting On' checkbox is checked, and the 'Answer Call Waiting On Hold' checkbox is also checked. The 'Busy On Hold' and 'Offhook Station' checkboxes are unchecked. The 'OK', 'Cancel', and 'Help' buttons are at the bottom right.

Select the **Supervisor Settings** sub-tab, and enter a desired **Login Code**.

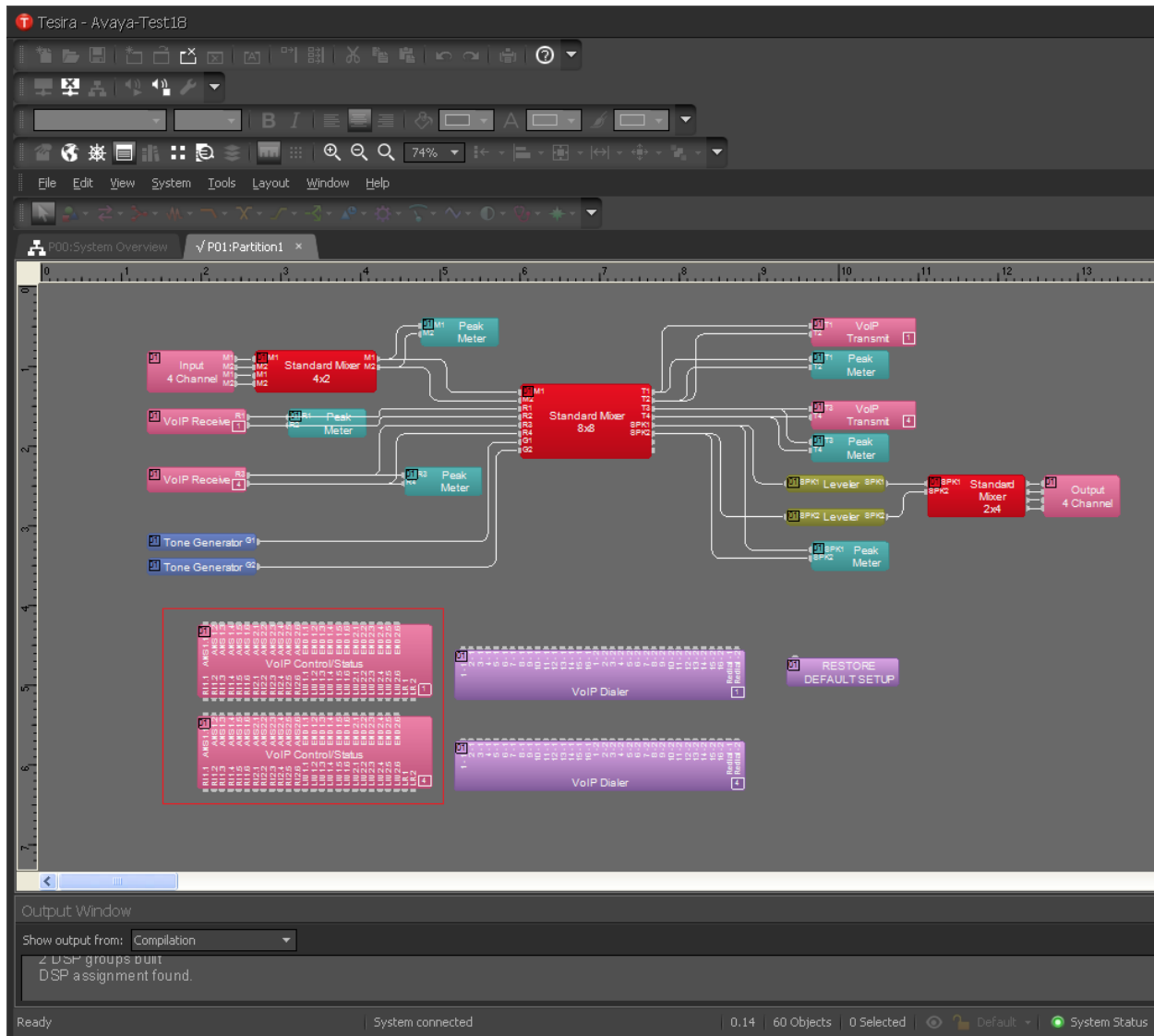
Repeat this section for each SIP extension from **Section 5.4**.

The screenshot displays the Avaya IP Office configuration window for extension 77018. The left sidebar shows a tree of IP Offices and users, with 77018 Extn218 selected. The main window has tabs for User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming, and Menu. The 'Supervisor Settings' sub-tab is active. The 'Login Code' field is highlighted with a red box and contains '\*\*\*\*\*'. Other fields include 'Login Idle Period (secs)', 'Monitor Group' (set to '<None>'), 'Coverage Group' (set to '<None>'), and 'Status on No-Answer' (set to 'Logged On (No change)'). There are checkboxes for 'Force Login', 'Force Account Code', 'Incoming Call Bar', 'Outgoing Call Bar', 'Inhibit Off-Switch Forward/Transfer', 'Can Intrude', 'Cannot be Intruded' (checked), 'Can Trace Calls', 'CCR Agent', 'Automatic After Call Work', and 'Deny Auto Intercom Calls'. A 'Reset Longest Idle Time' section has radio buttons for 'All Calls' (selected) and 'External Incoming'. The 'After Call Work Time (secs)' is set to 'System Default (10)'. The bottom of the window has 'OK', 'Cancel', and 'Help' buttons.

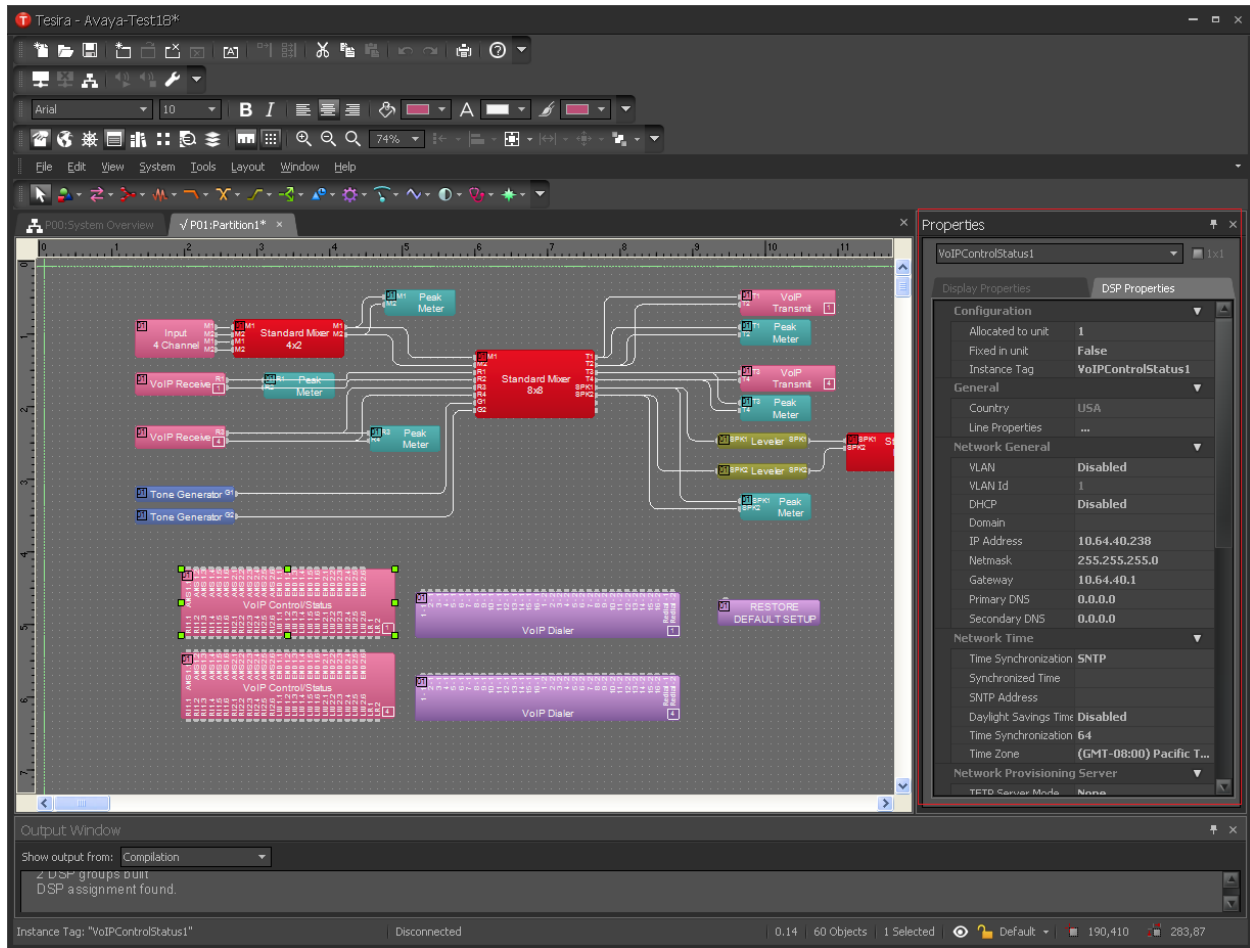
## 6. Configure Biamp Tesira SVC-2

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 Avaya IP Office. Select the Tesira icon from the 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.

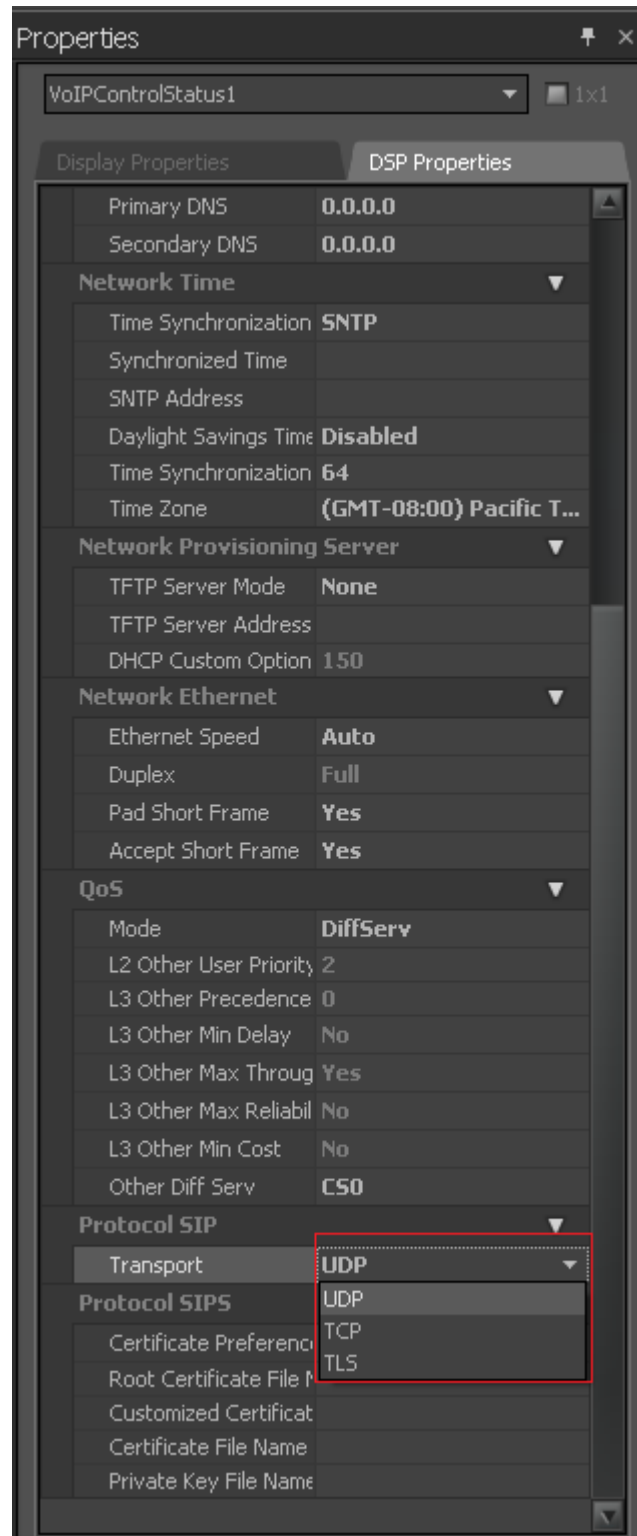
- Highlight the **VoIPControl/Status** block, as shown below.



- Click on the right mouse button and select **Properties**, the Properties menu will display on the right as shown below.



- Navigate the **Protocol SIP→Transport** to configure transport to be used. The default is UDP. During the compliance test, the default value was utilized.



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

The screenshot shows a 'Properties' window for 'VoIPControlStatus1'. The 'General' section is expanded, and the 'Line Properties' option is highlighted with a red rectangle. The 'Line Properties' option is currently set to '...'.

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

General	
Country	USA
Line Properties	...

Network General	
VLAN	Disabled
VLAN Id	1
DHCP	Disabled
Domain	
IP Address	10.64.40.238
Netmask	255.255.255.0
Gateway	10.64.40.1
Primary DNS	0.0.0.0
Secondary DNS	0.0.0.0

Network Time	
Time Synchronization Mode	SNTP
Synchronized Time	
SNTP Address	
Daylight Savings Time	Disabled
Time Synchronization Inter	64
Time Zone	(GMT-08:00) Pacific Time (...)

Network Provisioning Server	
TFTP Server Mode	None
TFTP Server Address	
DHCP Custom Option	150

Network Ethernet	
Ethernet Speed	Auto
Duplex	Full
Pad Short Frame	Yes
Accept Short Frame	Yes

QoS	
Mode	DiffServ
12 Other User Priority	?

- 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 border), '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 border), 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 5.4**.
  - **Authentication User Name** – Enter a user created in Avaya IP Office.
  - **Authentication Password** – Enter the password for the user in Avaya IP Office.
  - **Proxy Vendor** – Select Avaya IP Office
  - **Proxy Address** – Enter the IP address of IP Office.
  - **Proxy Port** – Enter either 5060.
    - 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

General Protocol Quality of Service NAT

**SIP**

SIP User Name: 770 18

SIP Display Name: 770 18,IP

SIP Domain Name:

Authentication User Name: 770 18

Authentication Password: .....

Proxy Vendor: Avaya IP Office

Proxy Address: 10.64.44.21

Proxy Port: 5060

Registration Expiration: 3600 seconds

Signaling Port: 5060

T1 Timer: 500 ms

Retransmit Timeout: 32000 ms

Session Timer: Enabled

Session Refresher: Auto

Session Expiration: 1800 seconds

Minimum Session Expiration: 90 seconds

Outbound Proxy Address:

Outbound Proxy Port: 5060

Pack: None

Local Dial Plan: [2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxx|[2-9]xxxxxxxx|[2-9]xxxT

**RTP/RTCP**

Port Start: 10000

Port End: 14999

Static RTP Port: Enable

SRTP:

G.723 Encoding Rate: 5.3 kbps

**SIPS**

Keyword:

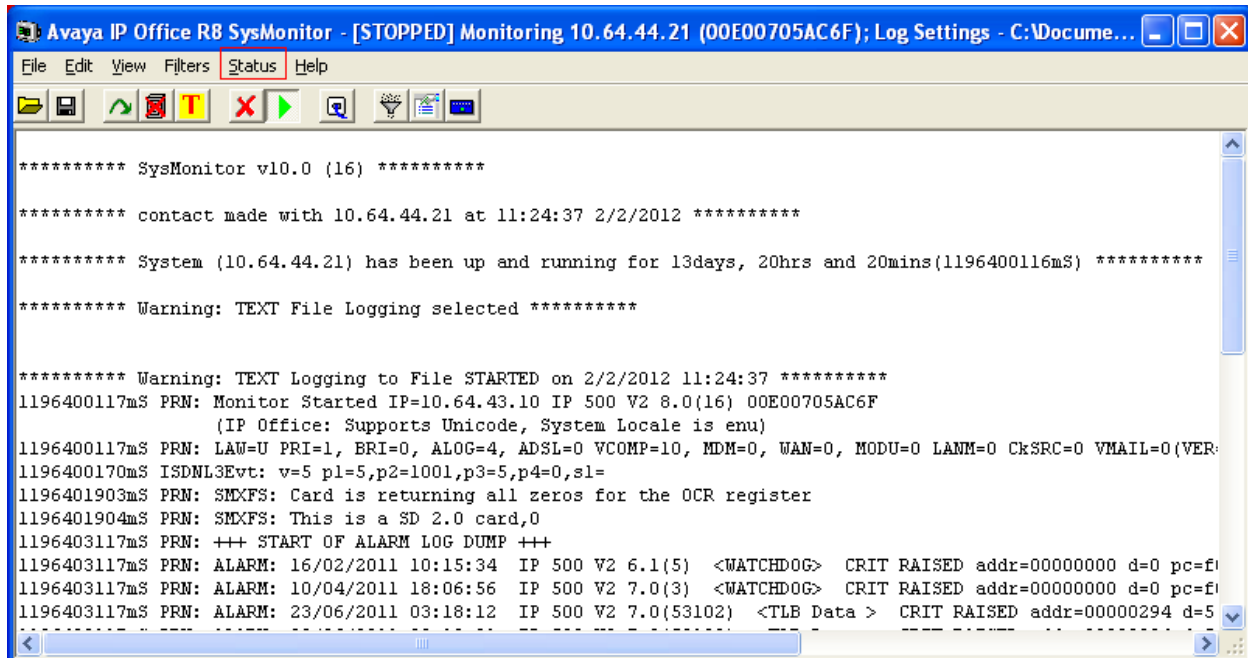
OK Cancel



## 7. Verification Steps

The following steps may be used to verify the configuration:

- From a PC running the Avaya IP Office Monitor application, select **Start → All Programs → IP Office → Monitor** to launch the application. Select **Status → SIP Phone Status** from the top menu.



- Verify that there is an entry for each Biamp Tesira SVC-2 extensions from **Section 5.4**, and the Status is **SIP: Registered on the SIPPhoneStatus page**.
- Place calls to and from Biamp Tesira SVC-2 Wireless telephones and verify that the calls are successfully established with two-way talk path.

## 8. Conclusion

Biamp Tesira SVC-2 was compliance tested with Avaya IP Office R9.0. Compliance testing between Biamp and Avaya IP Office was successful as per the tests outlined in **Section 2**.

## 9. Additional References

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

[1] *IP Office Manager*, January 2014, Release 9.0, Issue 9.02.0, Document Number 15-601011.

The following document was provided by Biamp

[2] *Tesira Operation Manual Document*

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