



Avaya Solution & Interoperability Test Lab

Application Notes for Codima autoVoIP with Avaya Communication Manager, Avaya SIP Enablement Services and Avaya 4600 Series IP Telephones - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Codima autoVoIP to interoperate with Avaya Communication Manager, Avaya SIP Enablement Services and Avaya 4600 Series IP Telephones. autoVoIP discovers Avaya 4600 Series SIP IP Telephones deployed over an enterprise network by monitoring the traffic. autoVoIP reports on quality of service parameters including: delay, lost packets, jitter, state of the network and the Avaya SIP Enablement Services (SES) performance.

Information in these Application Notes has been obtained through Developer*Connection* compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

autoVoIP discovers Avaya 4600 Series IP Telephones (SIP) deployed over an enterprise network by monitoring and analyzing the traffic on the network. Port mirroring or a hub can be used to passively sniff the traffic. autoVoIP obtains information about quality of service parameters including: delay, lost packets, jitter, state of the network and the Avaya SIP Enablement Services (SES) performance. autoVoIP does this by analysing the information held in the following protocols: SIP, Session Description Protocol (SDP), RTP and RTP Control Protocol (RTCP). autoVoIP correlates and filters this information and presents it through a grid style overview. Detailed reports on the telephones, problem area's and service assurance reports can also be created and viewed. Avaya SES server performance is tracked continuously, checking response time and identifying error patterns. In the compliance tested configuration described in these Application Notes, autoVoIP passively sniffs the traffic via the hub that the Avaya 4600 Series IP Telephones (SIP) are plugged into shown in **Figure 1**.

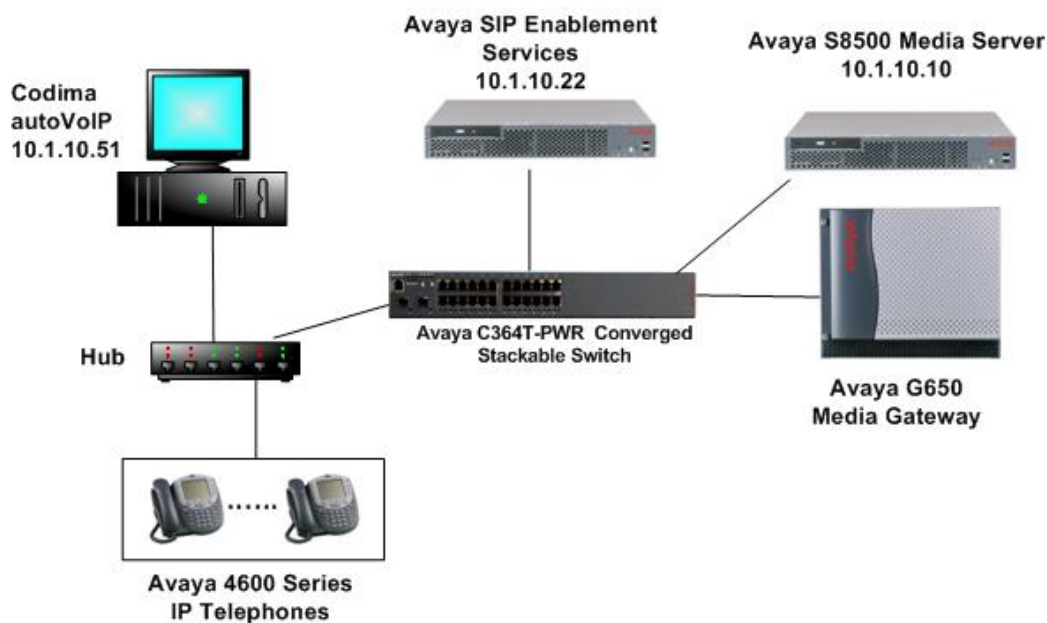


Figure 1: Ayaya SES and Avaya 4600 Series IP Telephones (SIP) with Codima autoVoIP

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya S8500 Media Server running Avaya Communication Manager	4.0
Avaya G650 Media Gateway	N/A
Avaya SIP Enablement Services	3.1.2
Avaya 4600 Series IP Telephones (SIP)	2.2.2
Avaya C364T-PWR Converged Stackable Switch	4.3.12
3COM OfficeConnect Dual Speed 16 port Hub	N/A
Codima autoVoIP Consultancy Kit running on Dell Workstation 370 Pentium 4 CPU 2.80GHz RAM 1.00GB Disk Space >4GB	3.10 0023 Windows XP Professional SP2

3. Configure Avaya Communication Manager

Avaya Communication Manager has been configured for connectivity to Avaya SES and Off-PBX Stations (OPS). Refer to SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series, and S8700 series Media Server document in the references section 10, which contains information for performing this type of configuration.

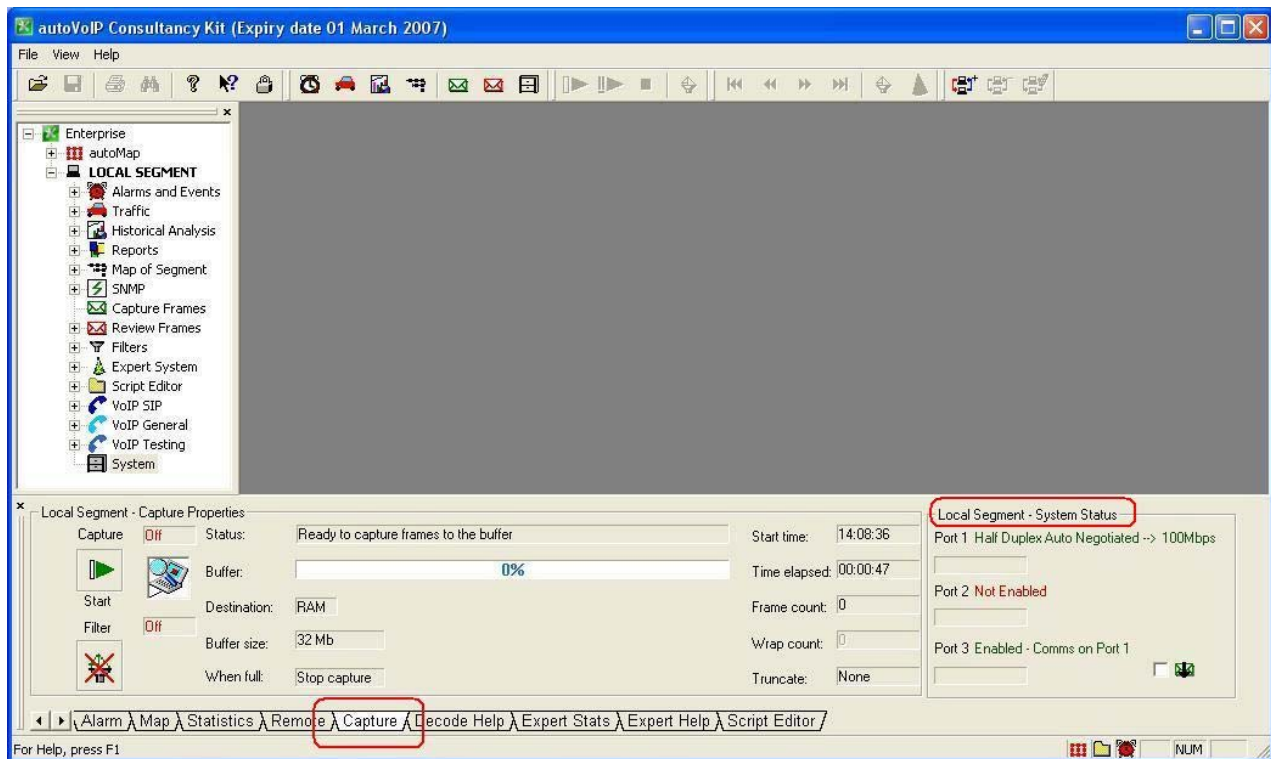
4. Configure Avaya SIP Enablement Services

It is assumed that Avaya 4600 Series IP Telephones on Avaya SES has been properly configured and is operational. Refer to Installing and Administering SIP Enablement Services document in the references section 10, which contains information for performing this type of configuration.

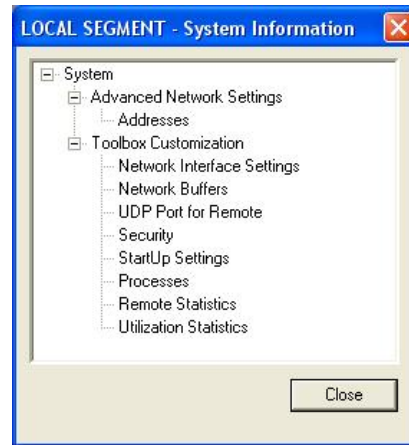
5. Configure autoVoIP

5.1. Verify the Network Interface Settings

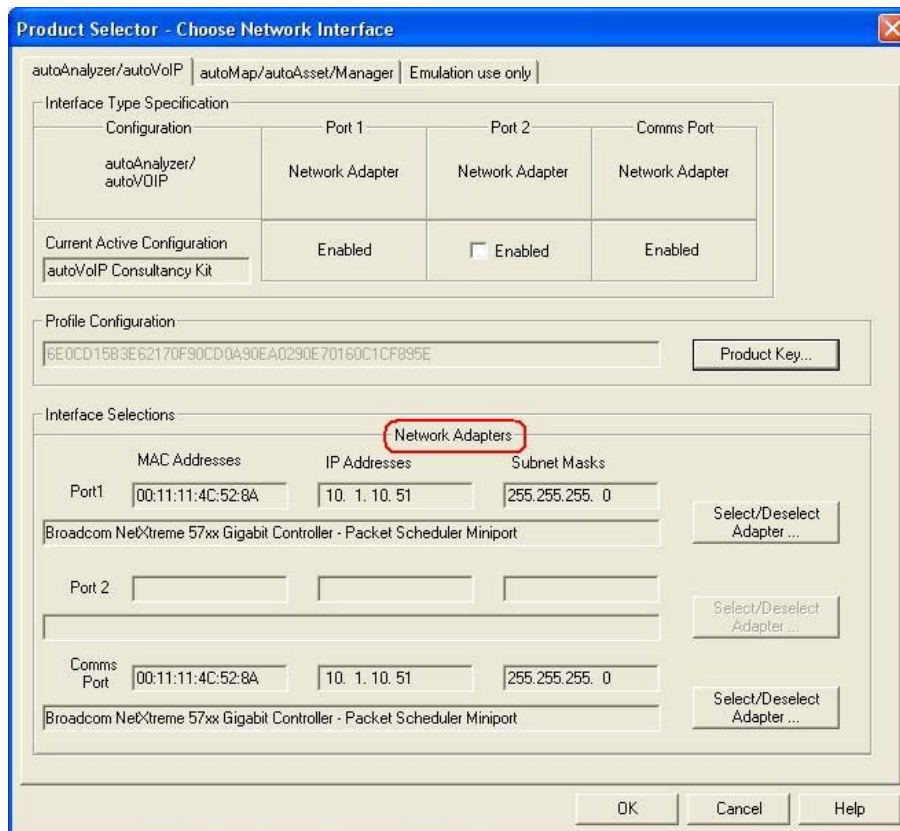
From the workstation that has autoVoIP installed, launch the Codima Toolbox console by clicking **Start → Programs → Codima Technologies → Codima Toolbox** and log in with the appropriate password. Click on the **Capture** tab. **Port 1** in the Local Segment-System Status section should be automatically configured to the workstations Network Interface Card (NIC). To verify the IP address and NIC of the workstation, click on **Enterprise → LOCAL SEGMENT → System** in the tree menu.



In the **LOCAL SEGMENT** dialog box expand **System** → **Toolbox Customization** → **Network Interface Settings**.

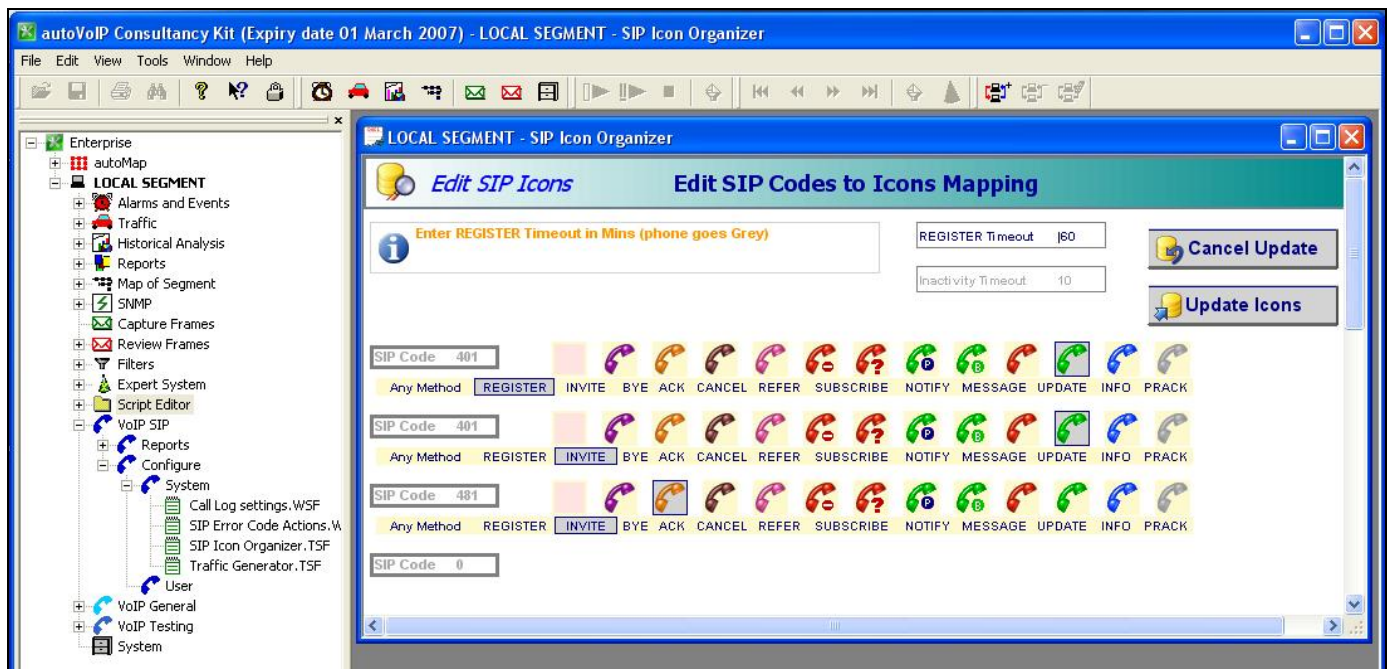


In the Network Adapters section, verify the IP address and the selected NIC. The NIC configured on Port 1 is used for passive analysis of the IP packets. Port 3 NIC card is used for communication such as pings etc. When using a switch, two NICs are needed for Ports 1 and 3. When using a hub, one NIC can be used as Port 3 is assigned to use the same NIC as Port 1.



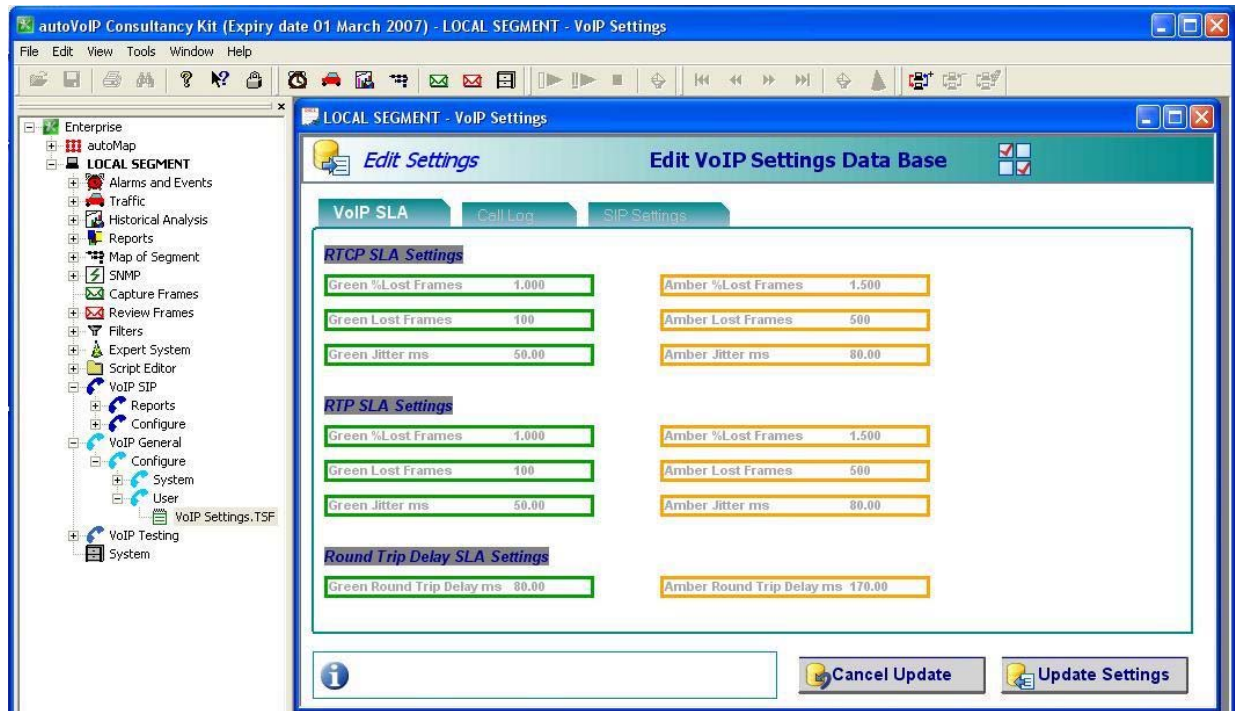
5.2. Configure Timeout Values

There are two timeout values. The first value is the **REGISTER Timeout** which is the length of time the SIP telephones stay registered for before changing status and requiring a register request. The second timeout is the **Inactivity Timeout** which is the length of time the SIP telephone can be idle before changing to the inactive state. Expand the **Enterprise** tree menu by clicking on **LOCAL SEGMENT → VoIP SIP → Configure → System → SIP Icon Organizer.TSF**. In the Edit SIP Icons screen, click on the **REGISTER Timeout** box and enter a value. The value “60” was used for the **REGISTER Timeout** during testing. Click on the **Inactivity Timeout** and enter a value. The value “10” was used for the **Inactivity Timeout** during testing. Click on **Update Icons** to save the changes.



5.3. Configure Service Level Agreement Settings

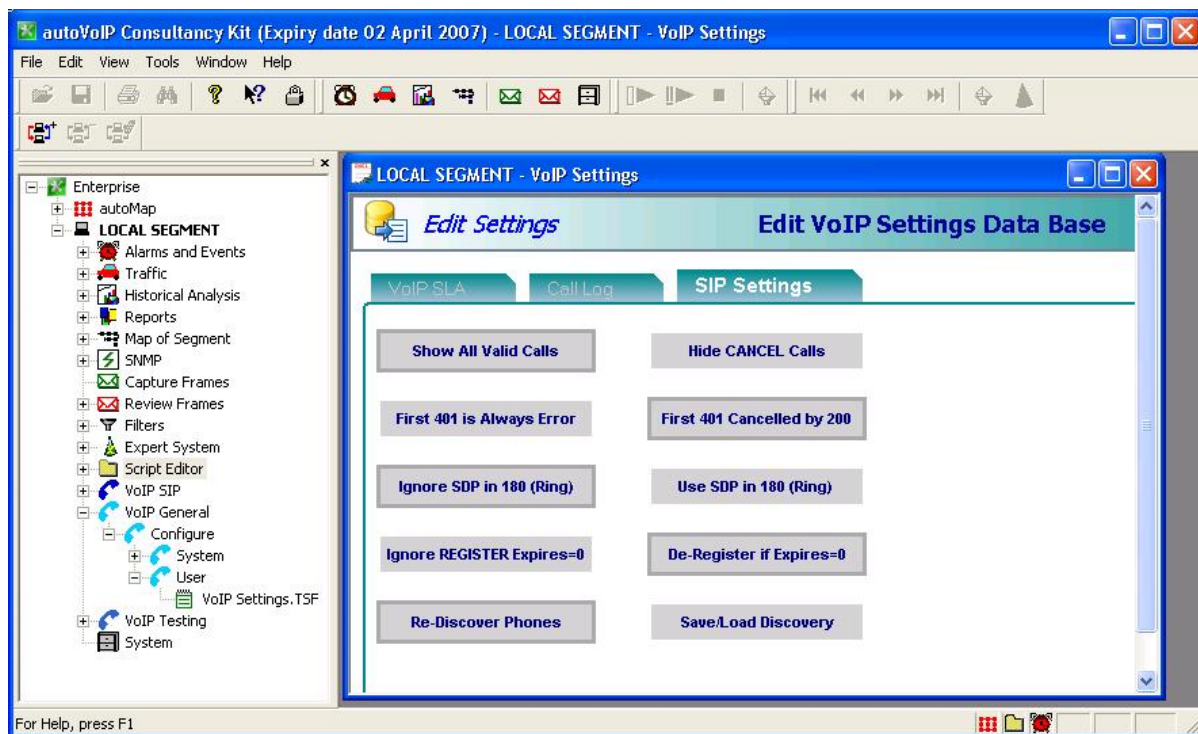
It is possible to set three Service Level Agreement (SLA) settings: RTCP, RTP and Round Trip Delay. To set the SLA values expand the **Enterprise** tree menu by clicking on **LOCAL SEGMENT** → **VoIP General** → **Configure** → **User** → **VoIP Settings.TSF**. Click on the **VoIP SLA** tab. Default values were used during testing.



5.4. Configure SIP Settings

To configure specific SIP responses expand the **Enterprise** tree menu by clicking on **LOCAL SEGMENT** → **VoIP General** → **Configure** → **User** → **VoIP Settings.TSF**. Click on the **SIP Settings** tab. Click on the **First 401 Cancelled by 200** button and **De-Register if Expires=0**.

- **First 401 Cancelled by 200** – Applies to the treatment of a 401 SIP response following an INVITE. In this case it is not treated as an error if cancelled by a 200 SIP response.
- **De-Register if Expires = 0** – is to track the register REGISTER Expires field set by the phone contact parameter. If REGISTER Expires is set to zero, this will classify the phone as being de-registered.



6. Verification Steps

6.1. Avaya 4600 Series IP Telephones registered on Avaya SES

Expand the **Users** option and select **Registered Users**, leave the screen with the default options and click **Search**.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The left sidebar contains a navigation menu with options like Top, Users, Conferences, Media Server Extensions, Emergency Contacts, Hosts, Media Servers, Adjunct Systems, Services, and Server Configuration. The main content area is titled "Registered Users on 10.1.10.22". It includes a search form with fields for Handle, First Name, Last Name, and Address. There are checkboxes for "Include Registered Users" (checked) and "Include Provisioned Users". A "Search" button is present. Below the search form, there is a section for "Apply to all registered users with compatible devices on this Home." with a "Task" dropdown set to "Reload-complete" and a "Submit" button.

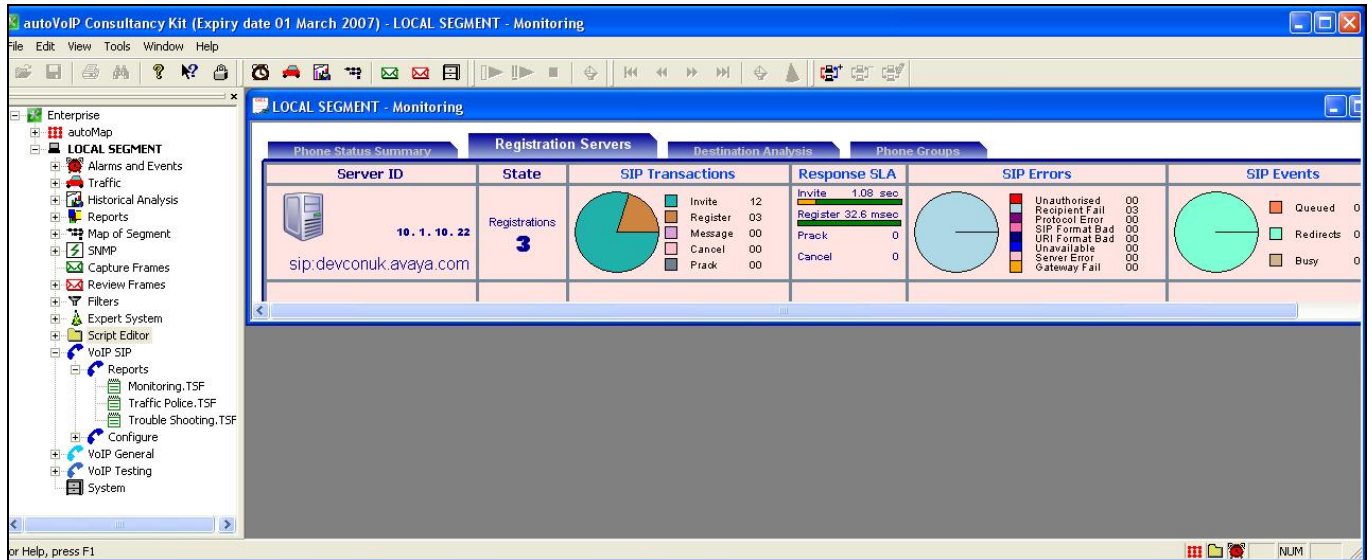
The screen below displays the three registered SIP telephones that were used during the testing.

The screenshot shows the Avaya Integrated Management SIP Server Management interface, displaying the "Registered Users on 10.1.10.22" page. The page shows a list of three registered contacts. The table has columns for "Handle and Name" and "Address". The contacts are listed with their handles and names, and their addresses. Below the table, there are checkboxes for "Apply to all registered users with compatible devices on this Home." and "Apply to all registered users with compatible devices on this page." A "Task" dropdown is set to "Reload-complete" and a "Submit" button is present.

Handle and Name	Address
<input type="checkbox"/> 10051@devconuk.avaya.com 10051, 10051	sip:10051@10.1.10.64:5060
<input type="checkbox"/> 10052@devconuk.avaya.com SIP 10052, SIP 10052	sip:10052@10.1.10.62:5060
<input type="checkbox"/> 10053@devconuk.avaya.com SIP 10053, SIP 10053	sip:10053@10.1.10.71:5060

6.2. Monitoring and Troubleshooting on autoVoIP

The following screens shows monitoring the SIP registrations, SIP telephones and the bandwidth. Expand the **Enterprise** tree menu by clicking on **LOCAL SEGMENT → VoIP SIP → Reports → Monitoring.TSF**. Click on the **Registration Servers** tab. The screen below displays there has been 3 registrations with Server ID IP address “10.1.10.22” and SIP domain name “devconuk.avaya.com”.



Under the same menu as above, click on **Trouble Shooting.TSF** and then click on the **All Phones** tab. The screen below displays three SIP telephones that have been registered and a call between two SIP telephones. The two SIP telephone icons change color to green and the **SIP Current State / Last SIP** box displays “Call Established”. It is possible to get real time call detail analyses as well as information about previous calls that a SIP telephone has made. Click on **Call Analyzer** and **Call Log** to display the information.

The screenshot shows the 'LOCAL SEGMENT - Trouble Shooting' window. The 'All Phones' tab is selected, showing three SIP phones: SIP:10051@DEVCONUK.AVAYA.COM, SIP:10052@DEVCONUK.AVAYA.COM, and SIP:10053@DEVCONUK.AVAYA.COM. The 'SIP Current State / Last SIP' box for SIP:10051@DEVCONUK.AVAYA.COM shows 'Call Established'. The 'Call Log' and 'Call Analyzer' buttons are highlighted. The 'Call Analyzer' window is open, showing a detailed call log for the selected SIP phone.

Frame No	OSI Layer	Event	Delta Time	Absolute Time
0	SIP	Continue INVITE Request TO: 10052 <sip:10052@devconuk.avaya.com> - FROM: 10051 <sip:10051@devconuk.avaya.com>	0	0
1	SIP	INVITE Response 100 Trying 100 Trying	2.53 msec	2.53 msec
2	SIP	INVITE Response 407 Proxy Authentication Required 7 Proxy Authentication Required	1.37 msec	3.90 msec
3	SIP	ACK Request ACK Request to 10052 <sip:10052@devconuk.avaya.com>	105 msec	109 msec
4	SIP	Continue INVITE Request TO: 10052 <sip:10052@devconuk.avaya.com> - FROM: 10051 <sip:10051@devconuk.avaya.com>	8.93 msec	118 msec
5	SIP	INVITE Response 100 Trying 100 Trying	2.51 msec	121 msec
6	RTCP	Sent Bytes=320, Packets=2, RTP Stamp=40554.4140 Sender Report	138 msec	258 msec
6	RTCP	Source Description	0	258 msec

The 'Selected SIP Phone ID' is sip:10051@devconuk.avaya.com. The 'SIP Current State / Last SIP' is Call Established. The 'Calls' section shows 3 calls, 3 good, 0 error, and 0 server error. The 'RTCP' section shows 31 sent and 0 received. The 'Jitter' section shows 1 local and 0 remote. The 'Lost Frames' section shows 0 local and 0 remote. The 'Delay' section shows 0 one way and 0 round trip. The 'QoS' section shows 93% RTT and 4.4 MOS.

The Call Log screen shown below displays the history of calls related to a SIP telephone.

The screenshot shows the 'LOCAL SEGMENT - Call Log' window. The 'Last Phone Selected' tab is selected, showing a call log table with columns: Time/Date, Dir, From / To, Result, Duration, ResponseTime, Jitter, Lost %, QoS, and Link.

Time/Date	Dir	From / To	Result	Duration	ResponseTime	Jitter	Lost %	QoS	Link
15:12:23	Out	sip:10051@devconuk.avaya.com	BYE Response 200 OK	3.15 sec	Ring 411 msec				
07 Feb 27	In	sip:10052@devconuk.avaya.com	Pi=34008 RRI=0 Po=2360 RR=0 ID= 4c3@10.1.10.64		Answer 1.89 sec			8:PCMA/8000 127.1	
15:12:49	In	sip:10053@devconuk.avaya.com:5061	BYE Response 407 Proxy Authentication Required	3.58 sec	Ring 102 msec				
07 Feb 27	Out	sip:10051@devconuk.avaya.com	Pi=34008 RRI=0 Po=2372 RR=0 ID= d583451574409		Answer 3.34 sec			8:PCMA/8000 18:07	
15:25:05	Out	sip:10051@devconuk.avaya.com	BYE Response 200 OK	2.61 sec	Ring 337 msec				
07 Feb 27	In	sip:10052@devconuk.avaya.com	Pi=34008 RRI=0 Po=2056 RR=0 ID= 321@10.1.10.64		Answer 2.12 sec			8:PCMA/8000 127.1	
15:33:20	Out	sip:10051@devconuk.avaya.com	BYE Response 200 OK	319 sec	Ring 340 msec				
07 Feb 27	In	sip:10052@devconuk.avaya.com	Pi=34008 RRI=0 Po=2064 RR=0 ID= 12b@10.1.10.64		Answer 246 sec			8:PCMA/8000 127.1	

The bottom of the window shows filter buttons: All, Good, Any Error, Client Error, Server Error, and Global Error.

7. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing focused on the ability for Codima autoVoIP to accurately monitor the Avaya 4600 Series IP Telephones registration, de-registration, on-hook, off-hook and call established operations.

7.1. General Test Approach

The feature test was performed by registering/de-registering SIP telephones and making/receiving calls between SIP telephones while monitoring the activity on autoVoIP Call Analyser. Calls made and received were verified via the autoVoIP Call Log. Serviceability tests included disconnecting the network cable and also registering/de-registering SIP telephones.

7.2. Test Results

All test cases were executed and passed.

8. Conclusion

These Application Notes describe the required configuration steps for Codima autoVoIP 3.10 to successfully interoperate with Avaya Communication Manager, Avaya SES and Avaya 4600 Series IP Telephones. All test cases were completed successfully and the configuration described in these Application Notes has been successfully compliance tested.

9. Support

For any support related enquiries, contact: tech_support@codimatech.com or
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10. Additional References

This section references the Avaya and Codima product documentation that are relevant to these Application Notes.

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

- SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series, and S8700 series Media Server, March 2007; Doc ID: 555-245-206
- Installing and Administering SIP Enablement Services, March 2007; Doc ID: 03-600768

Company and product information available from Codima can be found at <http://www.codimatech.com>

- <http://www.codimatech.com/products.php>
- http://www.codimatech.com/products_datasheets.php
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