



Avaya Solution & Interoperability Test Lab

Application Notes for Integrated Research PROGNOSIS VoIP Monitor with Avaya Aura™ Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Integrated Research PROGNOSIS VoIP Monitor to interoperate with Avaya Aura™ Communication Manager.

PROGNOSIS VoIP Monitor is a purpose-built solution for the monitoring of voice quality in Avaya IP telephony environments. PROGNOSIS VoIP Monitor provides best-in-class monitoring of voice quality from a telephony perspective as well as diagnostics for troubleshooting and service level analysis.

PROGNOSIS integrates directly to Communication Manager using Secure Shell (SSH). At the same time, it processes Real-time Transport Control Protocol (RTCP) information from Communication Manager.

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 compliance tested configuration used to validate Integrated Research PROGNOSIS VoIP Monitor with Avaya Aura™ Communication Manager.

PROGNOSIS VoIP Monitor is designed to provide a comprehensive monitoring platform for Avaya IP telephony networks. It does this by collecting data, filtering it as required and then presenting it in a user-friendly format, all in real-time. An additional function allows for data to be used to generate email alerts when pre-defined conditions are exceeded.

In order to collect and present data, the VoIP Monitor product must be installed on a dedicated server which is monitoring the Avaya PBX. The VoIP Monitor product includes a Web Interface component which is used to serve data to operators through a Web Browser connection.

The PROGNOSIS VoIP Monitor product uses the following methods to monitor a Communication Manager system.

- **System Access Terminal (SAT)** - The PROGNOSIS VoIP Monitor uses a pool of SSH connections to the SAT using the IP address of the Avaya Server. By default, the solution establishes two concurrent SAT connections to the Communication Manager system and uses the connections to execute SAT commands.
- **Real Time Transport Control Protocol (RTCP) Collection** - The PROGNOSIS VoIP Monitor collects RTCP information sent by the Avaya IP Media Processor (MEDPRO) boards, media gateways, IP Telephones and IP Softphones.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing evaluated the ability of the PROGNOSIS VoIP Monitor to correctly retrieve the configuration, performance, alarms and errors from Communication Manager. In addition, the ability of PROGNOSIS VoIP Monitor to receive and process RTCP information from Communication Manager was also validated.

The serviceability testing introduced failure scenarios to see if PROGNOSIS VoIP Monitor is able to resume service after failure recovery and an Avaya Server interchange.

1.2. Support

For technical support on PROGNOSIS VoIP Monitor, contact the Integrated Research Support Team at:

- Phone: +61 (2) 9966 1066
- Fax: +61 (2) 9921-1042
- Email: support@prognosis.com

2. Reference Configuration

Figure 1 illustrates the test configuration used to verify Integrated Research PROGNOSIS VoIP Monitor interoperability with Communication Manager. It consists of a Communication Manager system running on a pair of Avaya S8720 Servers with two Avaya G650 Media Gateways, an Avaya G450 Media Gateway, an Avaya G430 Media Gateway with EM200 Expansion Module, an Avaya G700 Media Gateway with Avaya S8300 Server as a Local Survivability Processor (LSP) and an Avaya G250-BRI Media Gateway. An Enterprise Survivable Server (ESS) running on Avaya S8500C Server was also configured for failover testing. The system has Avaya IP, digital and analog telephones, and Avaya One-X[®] Communicator users configured for making and receiving calls. Integrated Research PROGNOSIS VoIP Monitor was installed on a server running Microsoft Windows Server 2003 Standard Edition with Service Pack 2. All the systems and telephones are connected using an Avaya C364T-PWR Converged Stackable Switch for network connectivity.

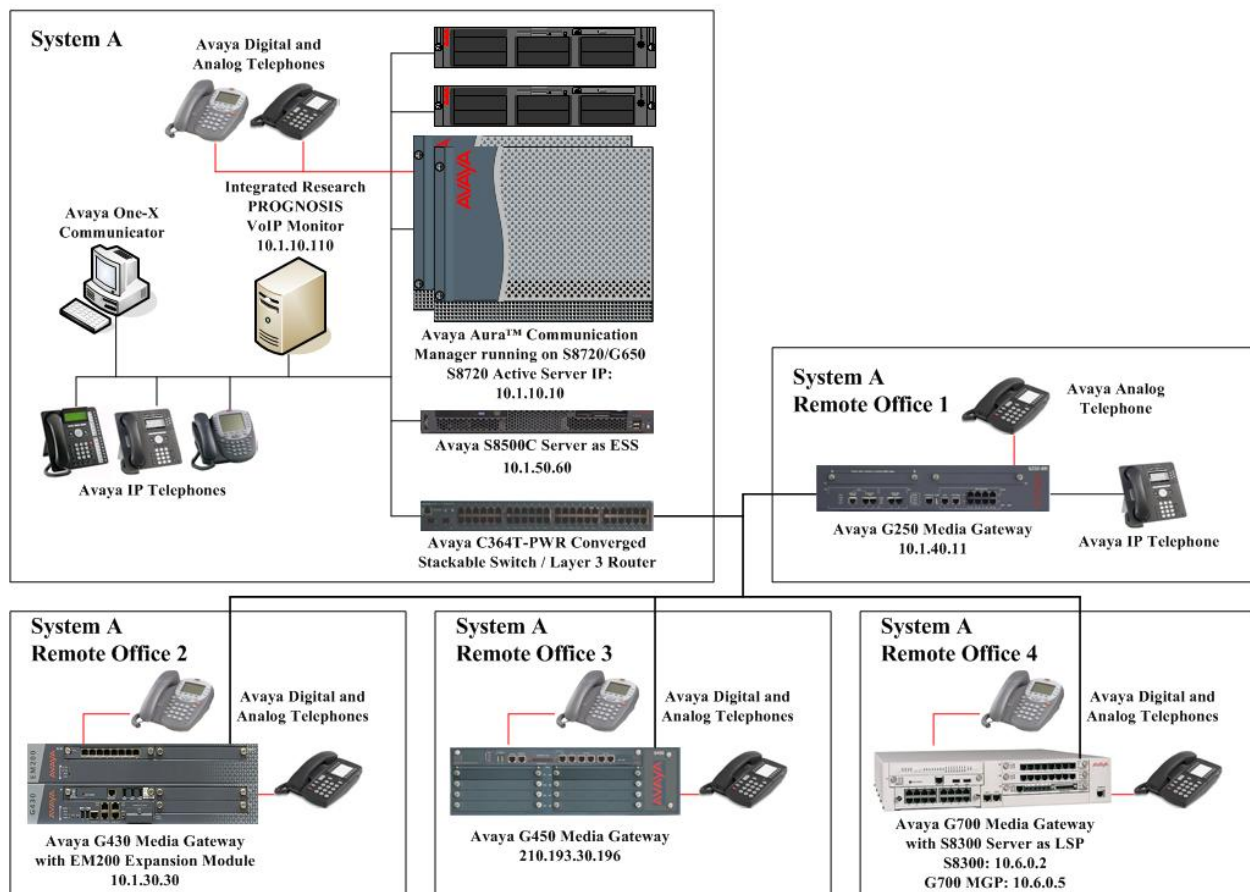


Figure 1: Test Configuration

3. Equipment and Software Validated

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

Equipment	Software
Avaya S8720 Servers	Avaya Aura™ Communication Manager 5.2 (Service Pack 02.0.947.3-17294)
Avaya G650 Media Gateway <ul style="list-style-type: none">- TN2312BP IP Server Interface- TN799DP C-LAN Interface- TN2302AP IP Media Processor- TN2602AP IP Media Processor- TN2214CP Digital Line- TN793CP Analog Line- TN2464BP DS1 Interface- TN2464CP DS1 Interface	- HW07 FW046 and HW15 FW042 HW01 FW032 and HW01 FW026 HW20 FW120 and HW20 FW117 HW02 FW048 and HW02 FW041 HW08 FW015 HW09 FW010 HW05 FW024 HW02 FW024
Avaya G250-BRI Media Gateway	29.23.0
Avaya G450 Media Gateway	29.23.0
Avaya G430 Media Gateway	29.23.0
Avaya S8300 Server as LSP	5.2 (Service Pack 02.0.947.3-17294)
Avaya G700 Media Gateway	29.23.0
Avaya S8500C Server as ESS	5.2 (Service Pack 02.0.947.3-17294)
Avaya 9600 Series IP telephones <ul style="list-style-type: none">- 9630, 9640, 9650	3.002 (H.323) Service Pack 1
Avaya 9670 IP telephones	2.0 (H.323)
Avaya 1608 IP telephones	1.100 (H.323)
Avaya 6221 analog telephones	-
Avaya 2420 digital telephones	-
Avaya One-X® Communicator	R1.010-SP1-15895
Avaya C364T-PWR Converged Stackable Switch	4.5.18
Integrated Research PROGNOSIS VoIP Monitor	9.5.3

4. Configure Communication Manager

This section describes the steps needed to configure Communication Manager to interoperate with Integrated Research PROGNOSIS VoIP Monitor. This includes creating a login account and a SAT User Profile for PROGNOSIS VoIP Monitor to access Communication Manager and enabling RTCP reporting. The steps are repeated for each Communication Manager system, ESS and LSP Servers.

4.1. Configure SAT User Profile

A SAT User Profile specifies which SAT screens may be accessed by the user assigned the profile and the type of access to each screen. As PROGNOSIS VoIP Monitor does not modify any system configuration, create a SAT User Profile with limited permissions to assign to the PROGNOSIS VoIP Monitor login account.

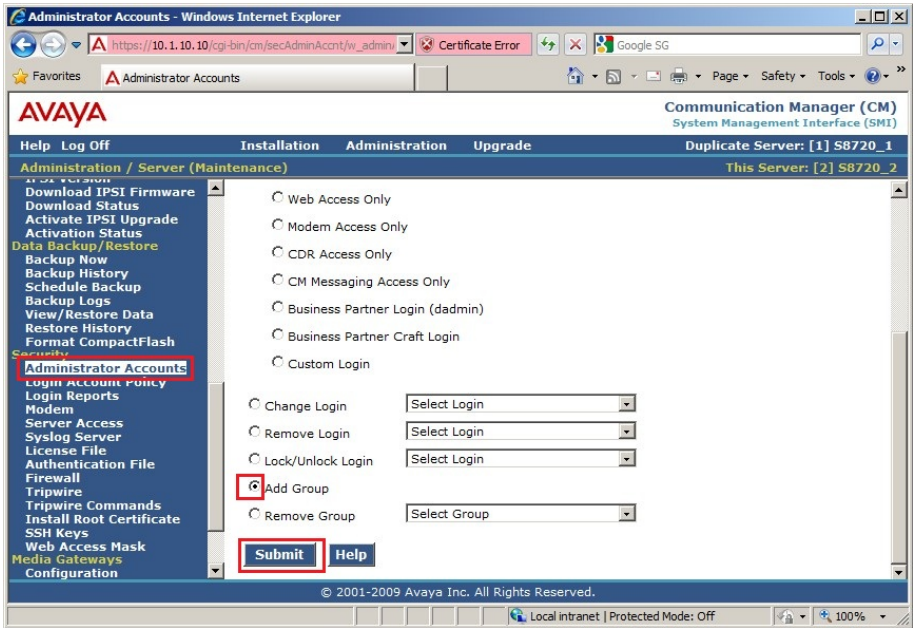
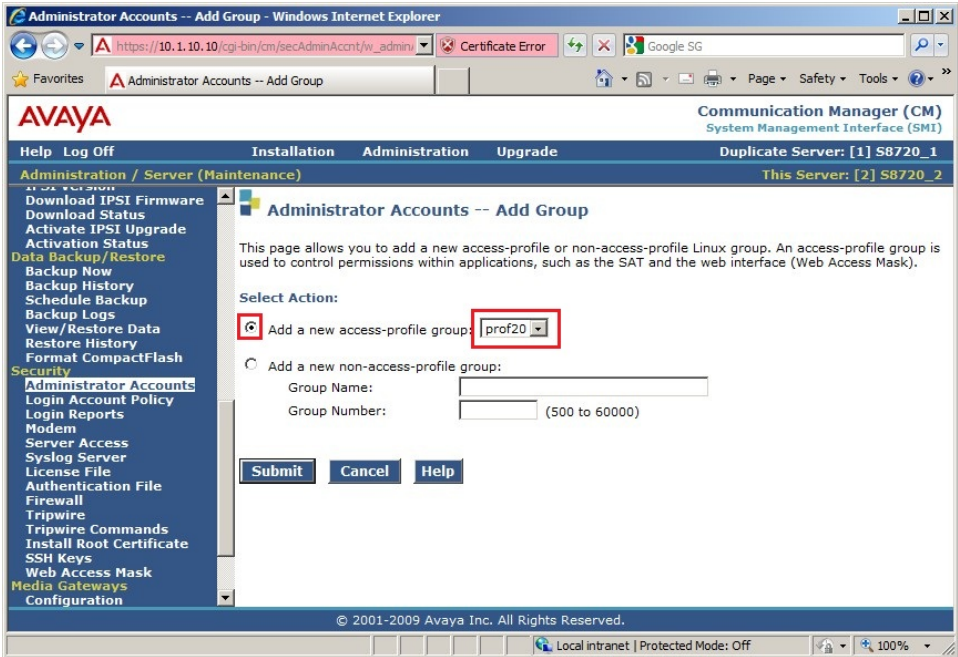
Step	Description
1.	<p>Enter the add user-profile <i>n</i> command, where <i>n</i> is the next unused profile number. Enter a descriptive name for User Profile Name and enable all categories by setting the Enbl field to y. In this configuration, the user profile 20 is created.</p> <pre>add user-profile 20 Page 1 of 41 USER PROFILE 20 User Profile Name: Prognosis This Profile is Disabled? n Shell Access? n Facility Test Call Notification? n Acknowledgement Required? n Grant Un-owned Permissions? n Extended Profile? n Name Cat Enbl Name Cat Enbl Adjuncts A [y] Routing and Dial Plan J [y] Call Center B [y] Security K [y] Features C [y] Servers L [y] Hardware D [y] Stations M [y] Hospitality E [y] System Parameters N [y] IP F [y] Translations O [y] Maintenance G [y] Trunking P [y] Measurements and Performance H [y] Usage Q [y] Remote Access I [y] User Access R [y]</pre>

Step	Description																																													
2.	<p>On Pages 2 to 41 of the USER PROFILE forms, set the permissions of all objects to rm (read and maintenance). This can be accomplished by typing rm into the field Set All Permissions To. Submit the form to create the user profile.</p>																																													
	<div>add user-profile 20<div>Page2 of 41</div></div> <div>USER PROFILE 20</div> <div>Set Permissions For Category:To: Set All Permissions To:rm</div> <div>'-'=no access 'r'=list,display,status 'w'=add,change,remove+r 'm'=maintenance</div> <table><thead><tr><th>Name</th><th>Cat</th><th>Perm</th></tr></thead><tbody><tr><td>aar analysis</td><td>J</td><td>rm</td></tr><tr><td>aar digit-conversion</td><td>J</td><td>rm</td></tr><tr><td>aar route-chosen</td><td>J</td><td>rm</td></tr><tr><td>abbreviated-dialing 7103-buttons</td><td>C</td><td>rm</td></tr><tr><td>abbreviated-dialing enhanced</td><td>C</td><td>rm</td></tr><tr><td>abbreviated-dialing group</td><td>C</td><td>rm</td></tr><tr><td>abbreviated-dialing personal</td><td>C</td><td>rm</td></tr><tr><td>abbreviated-dialing system</td><td>C</td><td>rm</td></tr><tr><td>aca-parameters</td><td>P</td><td>rm</td></tr><tr><td>access-endpoints</td><td>P</td><td>rm</td></tr><tr><td>adjunct-names</td><td>A</td><td>rm</td></tr><tr><td>administered-connections</td><td>C</td><td>rm</td></tr><tr><td>aesvcs cti-link</td><td>A</td><td>rm</td></tr><tr><td>aesvcs interface</td><td>A</td><td>rm</td></tr></tbody></table>	Name	Cat	Perm	aar analysis	J	rm	aar digit-conversion	J	rm	aar route-chosen	J	rm	abbreviated-dialing 7103-buttons	C	rm	abbreviated-dialing enhanced	C	rm	abbreviated-dialing group	C	rm	abbreviated-dialing personal	C	rm	abbreviated-dialing system	C	rm	aca-parameters	P	rm	access-endpoints	P	rm	adjunct-names	A	rm	administered-connections	C	rm	aesvcs cti-link	A	rm	aesvcs interface	A	rm
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4.2. Configure Login Group

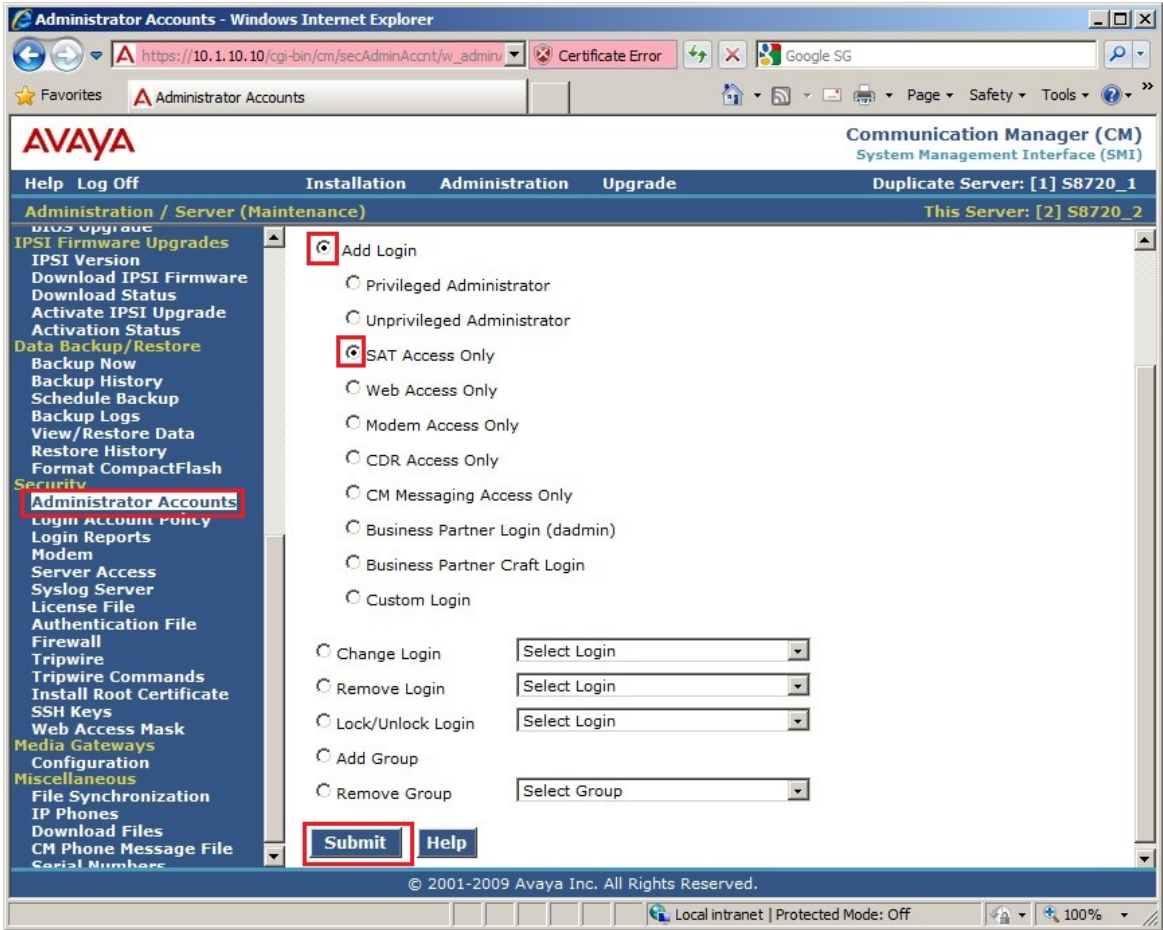
Create an Access-Profile Group to correspond to the SAT User Profile created in **Section 4.1**.

Step	Description
1.	<p>Using a web browser, enter https://<IP address of Avaya Server> to connect to the Avaya Server being configured and log in using appropriate credentials.</p>
2.	<p>Click Administration > Server (Maintenance). This will open up the Server Administration Interface that will allow the user to complete the configuration process.</p>

Step	Description
3.	<p>From the navigation panel on the left side, click Administrator Accounts. Select Add Group and click Submit.</p> 
4.	<p>Select Add a new access-profile group and select prof20 from the drop-down box to correspond to the user-profile created in Section 4.1 Step 1. Click Submit. This completes the creation of the login group.</p> 

4.3. Configure Login

Create a login account for PROGNOSIS VoIP Monitor to access the Communication Manager SAT.

Step	Description
1.	<p>From the navigation panel on the left side, click Administrator Accounts. Select Add Login and SAT Access Only to create a new login account with SAT access privileges only. Click Submit.</p> 

Step	Description
2.	<p>For the field Login name, enter the login. In this configuration, the login prognosis is created. Configure the other parameters for the login as follows:</p> <ul style="list-style-type: none"> • Primary group: users [Limits the permissions of the login] • Additional groups (profile): prof20 [Select the login group created in Section 4.2.] • Select type of authentication: Password [Uses a password for authentication.] • Enter password or key / Re-enter password or key [Define the password] <p>Click Submit to continue. This completes the configuration of the login.</p>

Administrator Accounts -- Add Login: SAT Access Only - Windows Internet Explorer

https://10.1.10.10/cgi-bin/cm/secAdminAcct/w_adminAcct

Communication Manager (CM)
System Management Interface (SMI)

Help Log Off Installation Administration Upgrade

Administration / Server (Maintenance)

Duplicate Server: [1] S8720_1
This Server: [2] S8720_2

Administrator Accounts -- Add Login: SAT Access Only

This page allows you to create a login that is intended to have access only to the Communication Manager System Administration Terminal (SAT) interface.

Login name: prognosis

Primary group: ☐ susers ☒ users

Additional groups (profile): prof20

Linux shell: /opt/ecs/bin/autosat

Home directory: /var/home/prognosis

Lock this account: ☐

Date after which account is disabled-blank to ignore (YYYY-MM-DD):

Select type of authentication: ☒ Password ☐ ASG: enter key ☐ ASG: Auto-generate key

Enter password or key:

Re-enter password or key:

Force password/key change on next login: ☐ Yes ☒ No

Submit Cancel Help

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Local intranet | Protected Mode: Off

4.4. Configure RTCP Monitoring

To allow PROGNOSIS VoIP Monitor to monitor the quality of IP calls, configure Communication Manager to send RTCP reporting to the IP address of the PROGNOSIS VoIP Monitor server.

Step	Description
1.	<p>Enter the change system-parameters ip-options command. In the RTCP MONITOR SERVER section, set Default Server IP Address to the IP address of the PROGNOSIS VoIP Monitor server. Set Default Server Port to 5005 and Default RTCP Report Period (secs) to 5.</p> <pre>change system-parameters ip-options Page 1 of 3 IP-OPTIONS SYSTEM PARAMETERS IP MEDIA PACKET PERFORMANCE THRESHOLDS Roundtrip Propagation Delay (ms) High: 800 Low: 400 Packet Loss (%) High: 40 Low: 15 Ping Test Interval (sec): 20 Number of Pings Per Measurement Interval: 10 RTCP MONITOR SERVER Default Server IP Address: 10 .1 .10 .110 Default Server Port: 5005 Default RTCP Report Period(secs): 5 AUTOMATIC TRACE ROUTE ON Link Failure? y H.248 MEDIA GATEWAY H.323 IP ENDPOINT Link Loss Delay Timer (min): 5 Link Loss Delay Timer (min): 5 Primary Search Time (sec): 75 Periodic Registration Timer (min): 20</pre>

Step	Description
2.	<p>Enter the change ip-network-region <i>n</i> command, where <i>n</i> is IP network region number to be monitored. Set RTCP Reporting Enabled to y and Use Default Server Parameters to y.</p> <p>Note: Only one RTCP MONITOR SERVER can be configured per IP network region.</p> <pre> change ip-network-region 1 Page 1 of 19 IP NETWORK REGION Region: 1 Location: 1 Authoritative Domain: Name: Local 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? y UDP Port Max: 65535 DIFFSERV/TOS PARAMETERS RTCP Reporting Enabled? <input checked="" type="checkbox"/> Call Control PHB Value: 46 RTCP MONITOR SERVER PARAMETERS Audio PHB Value: 46 Use Default Server Parameters? <input checked="" type="checkbox"/> 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 </pre>
3.	Repeat Step 2 for all IP network regions that are required to be monitored.

5. Configure Integrated Research PROGNOSIS VoIP Monitor

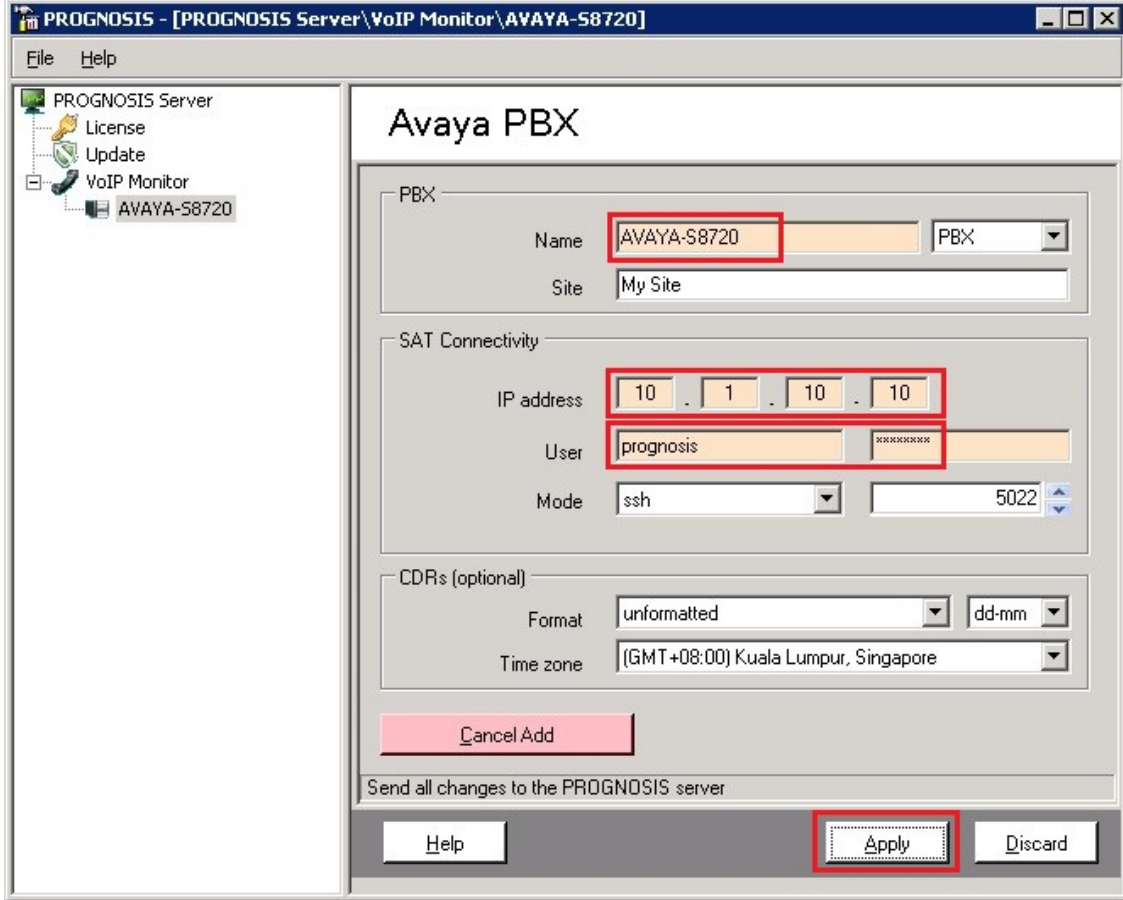
This section describes the configuration of Integrated Research PROGNOSIS VoIP Monitor required to interoperate with Communication Manager.

Step	Description
1.	On the Integrated Research PROGNOSIS VoIP Monitor server, click Start > All Programs > PROGNOSIS VoIP Monitor > Configure to start the configuration application.
2.	To configure the Communication Manager system to be monitored, click VoIP Monitor on the left pane and click Add a new PBX .

Metric	Good	Fair	Poor
MOS (1-5)	4.00	3.60	3.00
Latency (0-500)	50 ms	100 ms	150 ms
Packet Loss (0-20)	1.00 %	5.00 %	10.00 %
Jitter (0-50)	20 ms	30 ms	40 ms

CDR Ports	Value
PBX	50000
Legacy	50002
DEFINITY	50003

RTCP	Value
Port	5005

Step	Description
3.	<p>Specify a Name for the PBX and set IP address to that of the Avaya S8720 Servers, which in this configuration is 10.1.10.10. Enter the login account created in Section 4.3 for User and password. The remaining fields may be left at their defaults. Click Apply to save the changes.</p>
	 <p>The screenshot shows the 'Avaya PBX' configuration window. The 'PBX' section has 'Name' set to 'AVAYA-S8720' and 'Site' set to 'My Site'. The 'SAT Connectivity' section has 'IP address' set to '10.1.10.10', 'User' set to 'prognosis', and 'Mode' set to 'ssh'. The 'CDRs (optional)' section has 'Format' set to 'unformatted' and 'Time zone' set to '(GMT+08:00) Kuala Lumpur, Singapore'. The 'Apply' button is highlighted with a red box.</p>

6. General Test Approach and Test Results

The general test approach was to use PROGNOSIS GUI to display the configurations of the Communication Manager systems and verify against what is displayed on the SAT interface. The SAT interface is accessed by using either telnet or Secure Shell (SSH) to the Avaya S8720 Servers. Calls were placed between various Avaya endpoints and PROGNOSIS VoIP Monitor was used to display the RTCP information collected.

For feature testing, PROGNOSIS GUI was used to view the configurations of Communication Manager such as port networks, cabinets, media gateways, ESS, LSP, trunk groups, route patterns, CLAN, MEDPRO and DS1 boards, IP network regions, stations, processor occupancy, alarm and error information. Various conditions such as media gateway, port network, trunk group, trunk member and endpoint failures were created to see if PROGNOSIS VoIP Monitor was able to detect the outage. For the collection of RTCP information, the endpoints included Avaya IP, digital and analog telephones, and Avaya One-X[®] Communicator users.

For serviceability testing, reboots were applied to the PROGNOSIS VoIP Monitor Server and Avaya Servers to simulate system unavailability. Interchanging of the Avaya S8720 Servers and failover to ESS and LSP were also performed during testing.

All test cases passed successfully. The following observation was made during testing:

- The RTCP information sent out by the TN2602AP IP MEDPRO board did not include the correct IP address of the board. This caused PROGNOSIS VoIP Monitor to classify voice streams from conference calls incorrectly when using this board.

7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager and PROGNOSIS VoIP Monitor.

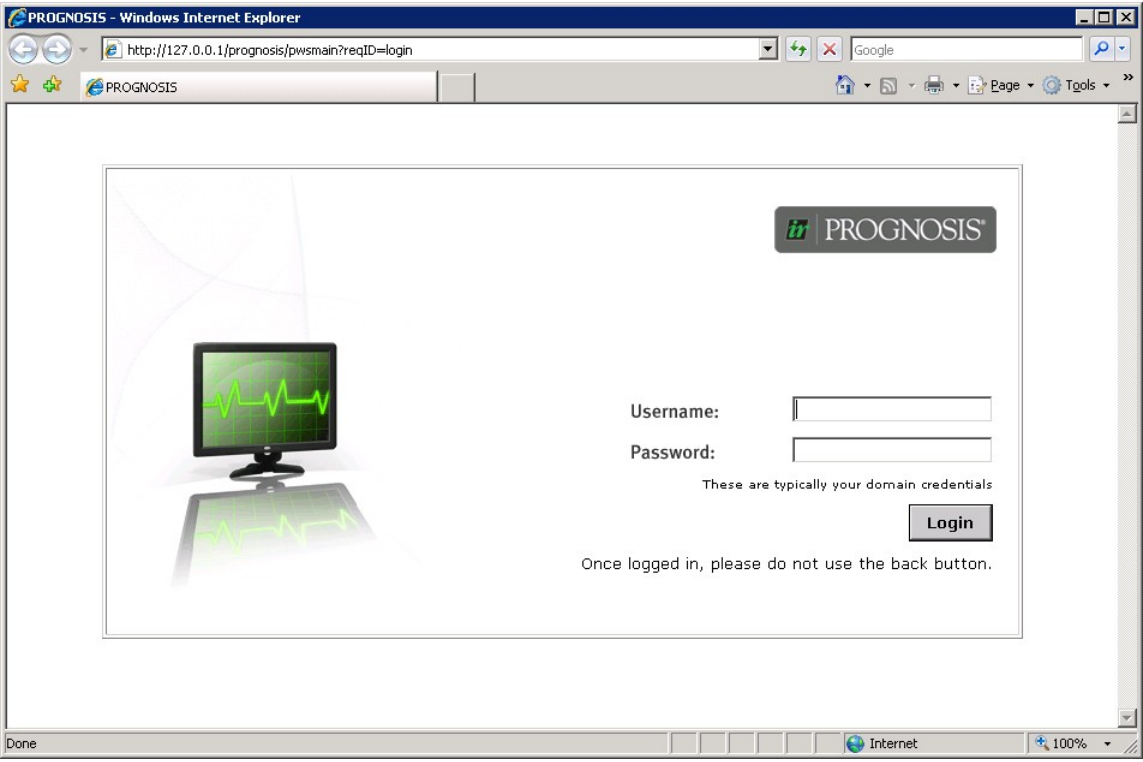
7.1. Verify Communication Manager

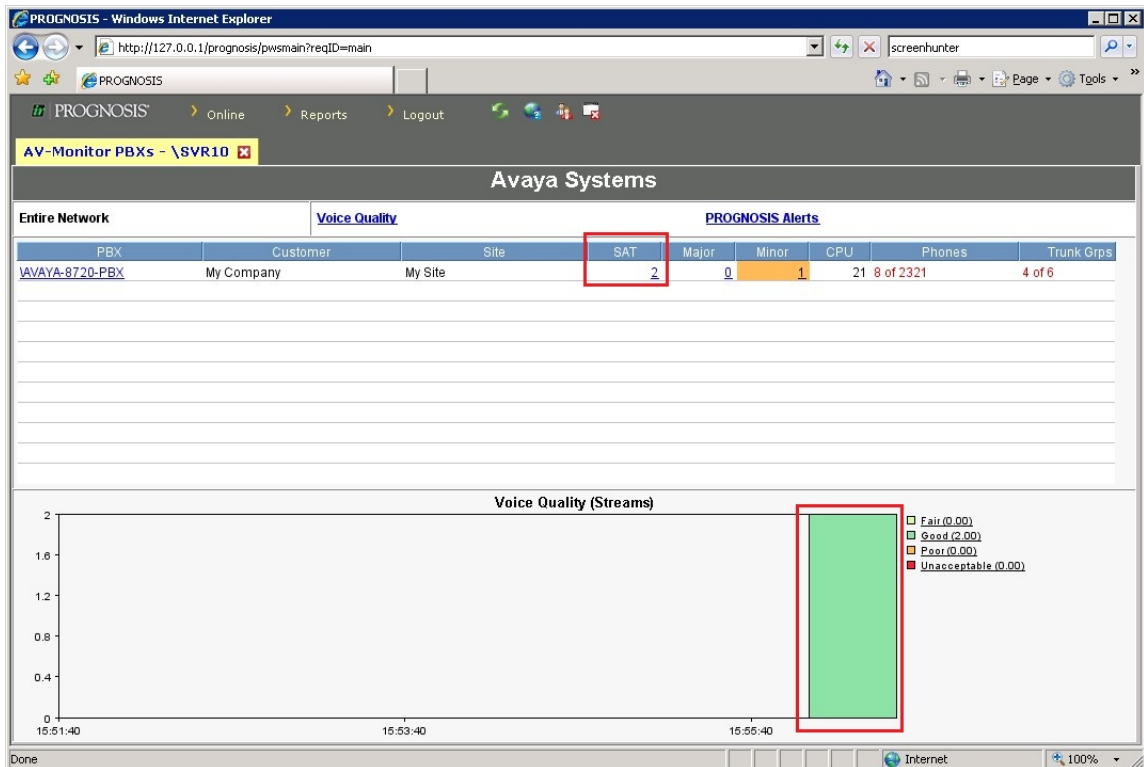
Verify that PROGNOSIS VoIP Monitor has established two concurrent SSH connections to the SAT by using the **status logins** command.

status logins				
COMMUNICATION MANAGER LOGIN INFORMATION				
Login	Profile	User's Address	Active Command	Session
*dadmin	2	10.1.10.152	stat logins	1
prognosi	20	10.1.10.110	list measurements summary	3
prognosi	20	10.1.10.110	list registered-ip-stations	4

7.2. Verify Integrated Research PROGNOSIS VoIP Monitor

The following steps are done using the PROGNOSIS VoIP Monitor web interface.

Step	Description
1.	<p>Using Microsoft Internet Explorer, browse to http://<IP address of PROGNOSIS VoIP Monitor> and login using a valid Windows user account.</p> 

Step	Description
2.	<p>In the Avaya Systems page, verify that the SAT field for the configured Communication Manager shows 2 connections. Make a call between two Avaya IP telephones that belong to an IP Network Region that is being configured to send RTCP information to the PROGNOSIS VoIP Monitor server. Verify that the Voice Quality (Streams) section shows two active voice streams reflecting the quality of the call.</p>  <p>The screenshot shows the PROGNOSIS VoIP Monitor interface in a Windows Internet Explorer browser. The address bar shows the URL http://127.0.0.1/prognosis/pwsmain?reqID=main. The page title is "PROGNOSIS". The navigation bar includes links for "Online", "Reports", and "Logout". The main content area is titled "Avaya Systems" and contains a table with the following columns: PBX, Customer, Site, SAT, Major, Minor, CPU, Phones, and Trunk Grps. The first row of the table shows "AVAYA-8720-PBX" for the PBX, "My Company" for the Customer, "My Site" for the Site, and "2" for the SAT field. The "Voice Quality (Streams)" section below the table shows a graph with a green bar representing the quality of the call. The legend indicates that green represents "Good (2.00)".</p>

8. Conclusion

These Application Notes describe the procedures for configuring the Integrated Research PROGNOSIS VoIP Monitor to interoperate with Avaya Aura™ Communication Manager. In the configuration described in these Application Notes, the PROGNOSIS VoIP Monitor established SSH connections to the SAT to view the configurations of Communication Manager and to monitor for failures. PROGNOSIS VoIP Monitor also processed the RTCP information to monitor the quality of IP calls. During compliance testing, all test cases were completed successfully.

9. Additional References

The following document can be found at <http://support.avaya.com>:

[1] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Release 5.2, Issue 7, May 2009, Document Number 555-245-205.

[2] *Administering Avaya Aura™ Communication Manager*, Release 5.2, Issue 5.0, May 2009, Document Number 03-300509.

The following documentations are provided by Integrated Research.

[3] *PROGNOSIS VoIP Monitor Installation and Configuration Guide*, March 2009

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