



Avaya Solution & Interoperability Test Lab

Application Notes for the Voice Print Activ! Voice Call Logger with Avaya Communication Manager using Avaya Communication Manager Application Programming Interface – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring the Voice Print Activ! Voice Call Logger to monitor and record calls placed to and from stations, softphones, and agents on an Avaya Communication Manager system. In the configuration described in these Application Notes, Activ! Voice employs Avaya Communication Manager Application Programming Interface (API) “virtual” stations as recording ports. During compliance testing, Activ! Voice Call Logger successfully recorded calls placed to and from Avaya IP and Digital Telephones, analog telephones, Avaya IP Softphones, and agents, as well as calls placed to a Vector Directory Number (VDN) and then queued to an agent hunt/skill group. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a compliance-tested configuration comprised of Avaya Communication Manager, Avaya Computer Telephony Integration (CTI) related interfaces, specifically the CallVisor Adjunct Switch Application Interface (ASAI) and the Avaya Communication Manager Application Programming Interface (API), and the Voice Print Activ! Voice Call Logger. Activ! Voice monitors, records, stores, and plays back phone calls for verification and quality assurance.

Activ! Voice interacts with an Intel NetMerge Call Processing Server, which in turn interacts with Avaya Communication Manager via an ASAI CTI link, to receive event reports and call information concerning particular stations, agents, and agent hunt/skill groups, and can use those event reports as recording triggers. Activ! Voice also interacts with an Avaya Communication Manager API server to register Communication Manager API “virtual” stations with Avaya Communication Manager. The Communication Manager API stations essentially appear as IP softphones to Avaya Communication Manager. Activ! Voice records a call by issuing a Single Step Conference (SSC) request to Avaya Communication Manager (via the CTI link) to bridge a Communication Manager API station onto the active call. Since the IP address of the Communication Manager API station is that of the Activ! Voice server, the audio portion of the call is directed to the Activ! Voice server and can thus be recorded.

Figure 1 illustrates a sample configuration consisting of an Avaya S8500 Media Server, an Avaya G650 Media Gateway, an Avaya Communication Manager API server, Avaya IP and Digital Telephones, analog telephones, an Avaya IP Softphone, and a Voice Print Activ! Voice Server. Avaya Communication Manager runs on the S8500 Media Server, though the solution described herein is also extensible to other Avaya Media Servers and Media Gateways. The Intel NetMerge Call Processing Server is installed on the same server as the Activ! Voice server, but may be installed on a separate server in other configurations. The Avaya C364T-PWR Layer 2/3 Switch supports the illustration and verification of the Avaya / Voice Print solution. The data network configuration is not the focus of these Application Notes and is thus not described here.

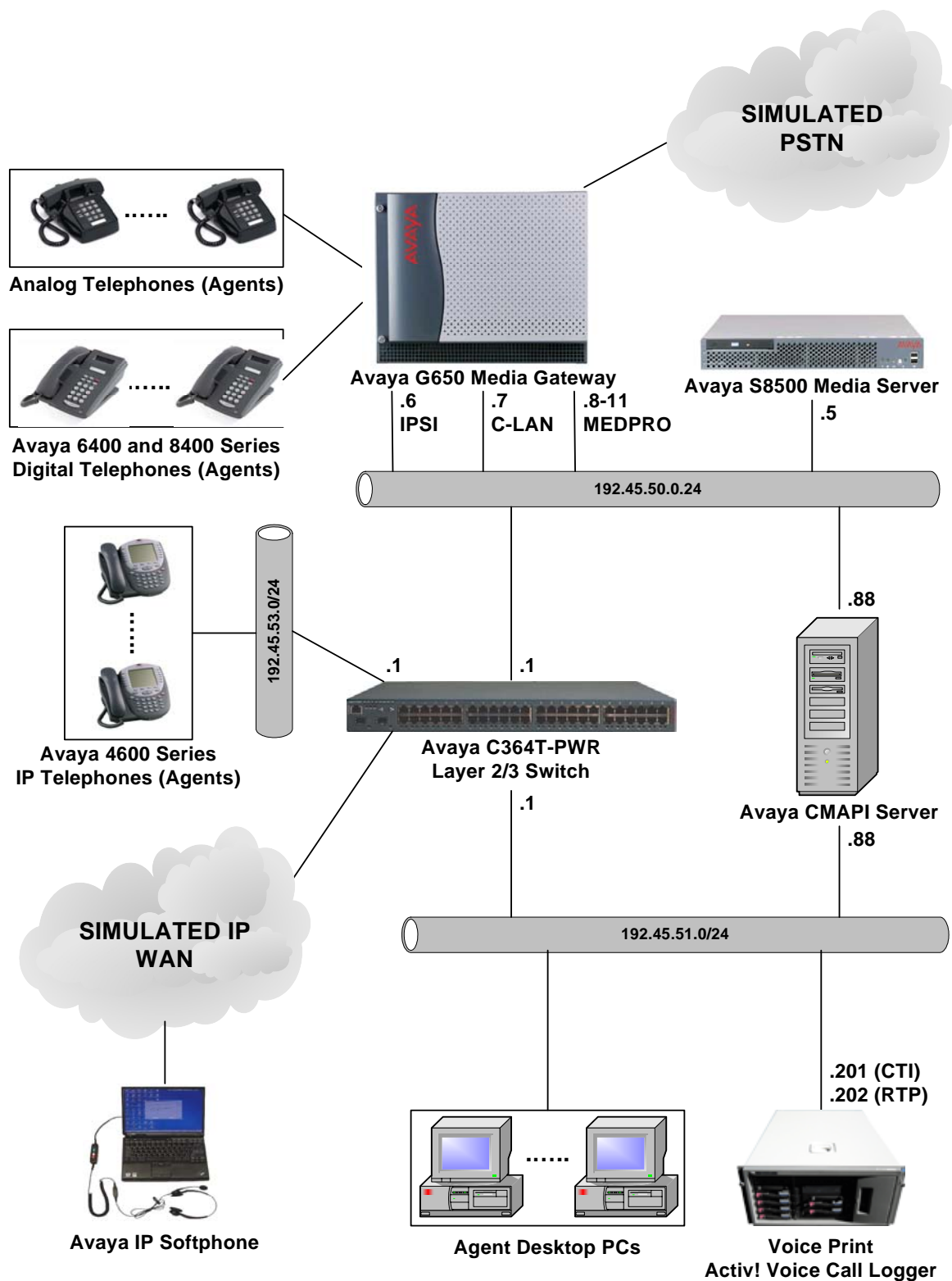


Figure 1: Sample Configuration.

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8500 Media Server	2.2 (R012x.02.0.111.4)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	13
TN799DP C-LAN Interface	14
TN2302AP IP Media Processor	102
Avaya 4600 Series IP Telephones	1.8.3 (4606) 1.8.3 (4612) 1.8.3 (4624) 1.8.2 (4602SW) 2.2 (4610SW) 2.2 (4620SW) 2.0.2 (4630SW)
Avaya IP Softphone	5.2
Avaya 6400 Series Digital Telephones	-
Avaya 8400 Series Digital Telephones	-
Analog Telephones	-
Avaya Communication Manager API Server	2.1.25
Avaya C364T-PWR Layer 2/3 Switch	4.3.12
Voice Print Activ! Voice Call Logger	2.8.1.4
Voice Print Server Configuration Program	1.1.0.1
Intel NetMerge Call Processing Server	6.0

3. Configure Avaya Communication Manager

This section describes the steps for configuring CTI links, hunt/skill groups, vectors, Vector Directory Numbers (VDNs), agents, agent login/logoff codes, recording ports, and codecs on Avaya Communication Manager. The steps are performed through the System Access Terminal (SAT) interface.

3.1. CTI Link

The Intel NetMerge Call Processing Server forwards CTI requests, responses, and events between the Voice Print Activ! Voice server and Avaya Communication Manager. The NetMerge Call Processing Server communicates with Avaya Communication Manager over a CTI link. The following steps demonstrate the configuration of the Avaya Communication Manager side of the CTI link. See Section 5 for details on configuring the NetMerge Call Processing Server side of the CTI link.

Step	Description
1.	<p>Enter the display system-parameters customer-options command and verify that ASAI Link Core Capabilities and Co-Res DEFINITY LAN Gateway are set to “y”. If not, contact an authorized Avaya account representative to obtain these licenses.</p> <pre> display system-parameters customer-options Page 3 of 11 OPTIONAL FEATURES Abbreviated Dialing Enhanced List? n Audible Message Waiting? n Access Security Gateway (ASG)? n Authorization Codes? n Analog Trunk Incoming Call ID? n Backup Cluster Automatic Takeover? n A/D Grp/Sys List Dialing Start at 01? n CAS Branch? n Answer Supervision by Call Classifier? n CAS Main? n ARS? y Change COR by FAC? n ARS/AAR Partitioning? y Computer Telephony Adjunct Links? n ARS/AAR Dialing without FAC? y Co-Res DEFINITY LAN Gateway? y ASAI Link Core Capabilities? y Cvg Of Calls Redirected Off-net? n ASAI Link Plus Capabilities? n DCS (Basic)? n Async. Transfer Mode (ATM) PNC? n DCS Call Coverage? n Async. Transfer Mode (ATM) Trunking? n DCS with Rerouting? n ATM WAN Spare Processor? n ATMS? n Digital Loss Plan Modification? n Attendant Vectoring? n DS1 MSP? n DS1 Echo Cancellation? n (NOTE: You must logoff & login to effect the permission changes.) </pre>

Step	Description
2.	<p>Enter the add cti-link m command, where m is a number between 1 and 16, inclusive. Enter an Extension valid under the provisioned dial plan in Avaya Communication Manager, set Type to “ASAI-IP”, and assign a descriptive Name to the CTI link.</p> <p>Note: For simplicity, since the NetMerge Call Processing Server is installed and running on the same server as the Activ! Voice server in the sample configuration, the NetMerge Call Processing Server is referred to as the Activ! Server in this and subsequent steps.</p> <pre> add cti-link 4 CTI LINK CTI Link: 4 Extension: 29004 Type: ASAI-IP Name: ASAI Link to Activ! Server COR: 1 </pre>
3.	<p>Enter the change node-names ip command. Note the node name and IP address for the C-LAN board and specify a node name for the Activ! Voice server, and enter their respective IP addresses.</p> <pre> change node-names ip IP NODE NAMES Name IP Address Name IP Address CLAN-1A02 192.45 .50 .7 . . . Activ!-Server 192.45 .51 .201 . . . MEDPRO-1A03 192.45 .50 .8 . . . MEDPRO-1A13 192.45 .50 .9 . . . MEDPRO-1B03 192.45 .50 .10 . . . MEDPRO-1B13 192.45 .50 .11 . . . default 0 .0 .0 .0 . . . procr </pre>

Step	Description																		
4.	Enter the change ip-services command. On page 1 of the ip-services form, configure and enable a “ DLG ” Service Type and specify the node name configured in Step 3 above for the C-LAN board as the Local Node . The Local Port should be fixed at 5678 .																		
	<div>change ip-services<div>Page1 of 3</div></div> <table><thead><tr><th colspan="6">IP SERVICES</th></tr><tr><th>Service Type</th><th>Enabled</th><th>Local Node</th><th>Local Port</th><th>Remote Node</th><th>Remote Port</th></tr></thead><tbody><tr><td>DLG</td><td>y</td><td>CLAN-1A02</td><td>5678</td><td></td><td></td></tr></tbody></table>	IP SERVICES						Service Type	Enabled	Local Node	Local Port	Remote Node	Remote Port	DLG	y	CLAN-1A02	5678		
	IP SERVICES																		
	Service Type	Enabled	Local Node	Local Port	Remote Node	Remote Port													
DLG	y	CLAN-1A02	5678																
On page 3 of the ip-services form, configure and enable a CTI Link with the same link number configured in Step 2. For Client Name , enter the node name configured in Step 3 for the Activ! Voice server. For Client Link , enter the link number to be configured on the Activ! Voice server side of the CTI link (see Step 4 of Section 5).																			
<div>change ip-services<div>Page3 of 3</div></div> <div>DLG Administration</div> <table><thead><tr><th>CTI Link</th><th>Enabled</th><th>Client Name</th><th>Client Link</th><th>Client Status</th></tr></thead><tbody><tr><td>4</td><td>y</td><td>Activ!-Server</td><td>1</td><td></td></tr></tbody></table>	CTI Link	Enabled	Client Name	Client Link	Client Status	4	y	Activ!-Server	1										
CTI Link	Enabled	Client Name	Client Link	Client Status															
4	y	Activ!-Server	1																

3.2. Agent Hunt/Skill Groups, Agent Logins, and Call Vectoring

The following steps describe the configuration of hunt/skill groups, agent logins, and call vectoring in Avaya Communication Manager.

Step	Description
1.	<p>Enter the display system-parameters customer-options command and verify that ACD and Vectoring (Basic) are set to “y”. If not, contact an authorized Avaya account representative to obtain these licenses. Expert Agent Selection was enabled for the testing, but the feature is not required.</p> <pre> display system-parameters customer-options Page 6 of 11 CALL CENTER OPTIONAL FEATURES Call Center Release: 12.0 ACD? y PASTE (Display PBX Data on Phone)? n BCMS (Basic)? y Reason Codes? n BCMS/VuStats Service Level? n Service Level Maximizer? n BSR Local Treatment for IP & ISDN? n Service Observing (Basic)? y Business Advocate? n Service Observing (Remote/By FAC)? n Call Work Codes? n Service Observing (VDNs)? n DTMF Feedback Signals For VRU? n Timed ACW? n Dynamic Advocate? n Vectoring (Basic)? y Expert Agent Selection (EAS)? y Vectoring (Prompting)? n EAS-PHD? n Vectoring (G3V4 Enhanced)? n Forced ACD Calls? n Vectoring (ANI/II-Digits Routing)? n Least Occupied Agent? n Vectoring (G3V4 Advanced Routing)? n Lookahead Interflow (LAI)? n Vectoring (CINFO)? n Multiple Call Handling (On Request)? n Vectoring (Best Service Routing)? n Multiple Call Handling (Forced)? n Vectoring (Holidays)? n Vectoring (Variables)? n (NOTE: You must logoff & login to effect the permission changes.) </pre>

Step	Description
2.	<p>Enter the add hunt-group n command, where n is an unused hunt group number. On page 1 of the hunt group form, assign a Group Name and Group Extension valid under the provisioned dial plan and set ACD, Queue, and Vector to “y”. When ACD is enabled, hunt group members serve as ACD agents and must log in to receive ACD split/skill calls. When Queue is enabled, calls to the hunt group will be served by a queue. When Vector is enabled, the hunt group will be vector controlled.</p>
	<pre> add hunt-group 1 HUNT GROUP Page 1 of 61 Group Number: 1 ACD? y Group Name: Agent pool Queue? y Group Extension: 73000 Vector? y Group Type: ucd-mia TN: 1 COR: 1 MM Early Answer? n Security Code: ISDN Caller Display: Calls Warning Threshold: Port: Time Warning Threshold: Port: </pre>
	<p>On page 2, set Skill to “y”, which means that agent membership in the hunt group is based on skills, rather than pre-programmed assignment to the hunt group.</p>
	<pre> add hunt-group 1 HUNT GROUP Page 2 of 3 Skill? y AAS? n Measured: internal Supervisor Extension: Controlling Adjunct: none Redirect on No Answer (rings): 5 Redirect to VDN: Forced Entry of Stroke Counts or Call Work Codes? n </pre>

Step	Description
3.	<p>Enter the add agent-loginID p command, where p is an extension valid under the provisioned dial plan. On page 1 of the agent-loginID form, enter a descriptive Name and Password.</p> <pre> add agent-loginID 75001 Page 1 of 2 AGENT LOGINID Login ID: 75001 AAS? n Name: Agent-75001 AUDIX? n TN: 1 LWC Reception: spe COR: 1 LWC Log External Calls? n Coverage Path: AUDIX Name for Messaging: Security Code: LoginID for ISDN Display? n Password: 12345 Password (enter again): 12345 Auto Answer: station WARNING: Agent must log in again before skill changes take effect </pre> <p>On page 2, set the Skill Number (SN) to the hunt group number assigned in Step 2. The Skill Level (SL) may be set according to customer requirements.</p> <pre> add agent-loginID 75001 Page 2 of 2 AGENT LOGINID Direct Agent Skill: Call Handling Preference: skill-level SN SL SN SL 1: 1 1 16: 2: 17: 3: 18: 4: 19: 5: 20: 6: 7: 8: 9: 10: 11: 12: 13: 14: 15: </pre> <p>Repeat this step as necessary to configure additional agent extensions.</p>

Step	Description
4.	<p>Enter the change vector q command, where q is an unused vector number. Enter a descriptive Name, and program the vector to deliver calls to the hunt/skill group number defined in Step 2. Agents that are logged into the hunt/skill group will be able to answer calls queued to the hunt/skill group.</p> <pre> change vector 1 CALL VECTOR Page 1 of 3 Number: 1 Name: Queue to skill1 Meet-me Conf? n Lock? n Basic? y EAS? y G3V4 Enhanced? n ANI/II-Digits? n ASAI Routing? y Prompting? n LAI? n G3V4 Adv Route? n CINFO? n BSR? n Holidays? n Variables? n 01 wait-time 2 secs hearing ringback 02 queue-to skill 1 pri m 03 </pre>
5.	<p>Enter the add vdn r command, where r is an extension valid under the provisioned dial plan. Specify a descriptive Name for the VDN and the Vector Number configured in Step 4. In the example below, incoming calls to the extension 72000 will be routed to VDN 72000, which in turn will invoke the actions specified in vector 1.</p> <pre> add vdn 72000 VECTOR DIRECTORY NUMBER Page 1 of 2 Extension: 72000 Name: VDN-72000 Vector Number: 1 Meet-me Conferencing? n Allow VDN Override? n COR: 1 TN: 1 Measured: internal 1st Skill: 2nd Skill: 3rd Skill: </pre>

Step	Description
6.	Enter the change feature-access-codes command. Define the Auto-In Access Code , Login Access Code , and Logout Access Code .
	<pre> change feature-access-codes Page 5 of 6 FEATURE ACCESS CODE (FAC) Automatic Call Distribution Features After Call Work Access Code: Assist Access Code: Auto-In Access Code: #66 Aux Work Access Code: Login Access Code: #65 Logout Access Code: *65 Manual-in Access Code: Add Agent Skill Access Code: Remove Agent Skill Access Code: Remote Logout of Agent Access Code: </pre>

3.3. Recording Ports

The recording ports in this configuration are Communication Manager API stations that essentially appear as IP softphones to Avaya Communication Manager. Each Communication Manager API station requires an “IP_API_A” license. Note that this is separate and independent of Avaya IP Softphone licenses, which are required for Avaya IP Softphones but not required for Communication Manager API stations. Enter the **display system-parameters customer-options** command and verify that there are sufficient **IP_API_A** licenses. If not, contact an authorized Avaya account representative to obtain these licenses.

display system-parameters customer-options			Page 10 of 11
MAXIMUM IP REGISTRATIONS BY PRODUCT ID			
Product ID	Rel. Limit	Used	
IP_API_A	: 1000	60	
IP_API_B	: 0	0	
IP_API_C	: 0	0	
IP_Agent	: 1000	0	
IP_IR_A	: 1000	0	
IP_Phone	: 1000	9	
IP_ROMax	: 5	0	
IP_Soft	: 1000	0	
IP_eCons	: 0	0	
	: 0	0	

Enter the **add station s** command, where s is an extension valid under the provisioned dial plan. On Page 1 of the **station** form, set **Type** to an IP or Digital telephone set type, set **Port** to **IP**, enter a descriptive **Name**, specify the **Security Code**, and set **IP Softphone** to “y.”

Note: Activ! Voice requires a block of consecutive extension numbers for the Communication Manager API softphones that it uses for recording.

add station 60001		Page 1 of 3	
STATION			
Extension: 60001	Lock Messages? n	BCC: 0	
Type: 4610	Security Code: 12345	TN: 1	
Port: IP	Coverage Path 1:	COR: 1	
Name: CMAPI Recording Line 1	Coverage Path 2:	COS: 1	
	Hunt-to Station:		
STATION OPTIONS			
Loss Group: 19	Personalized Ringing Pattern: 1		
	Message Lamp Ext: 60001		
Speakerphone: 2-way	Mute Button Enabled? y		
Display Language: english			
Survivable GK Node Name:	Media Complex Ext:		
	IP SoftPhone? y		

3.4. Codec Configuration

Enter the **change ip-codec-set t** command, where t will be the ip-codec-set used for communication to the Voice Print Activ! Voice Server. In the first row, enter “**G.711MU**” for **Audio Codec**. The codec configured on the Voice Print Activ! Voice Server in Section 6, Step 4, must match this value. The rest of the row may be left at the defaults.

change ip-codec-set 1		Page 1 of 2	
IP Codec Set			
Codec Set: 1			
Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1: G.711MU	n	2	20
2:			
3:			
4:			
5:			
6:			
7:			

Enter the **change ip-network-region u** command, where u is a number between 1 and 250, inclusive. Set **Codec Set** to the ip-codec-set number configured above. In the compliance-tested

configuration, all devices were in network region 1, including the Communication Manager API softphones used by the Activ! Voice Server for recording.

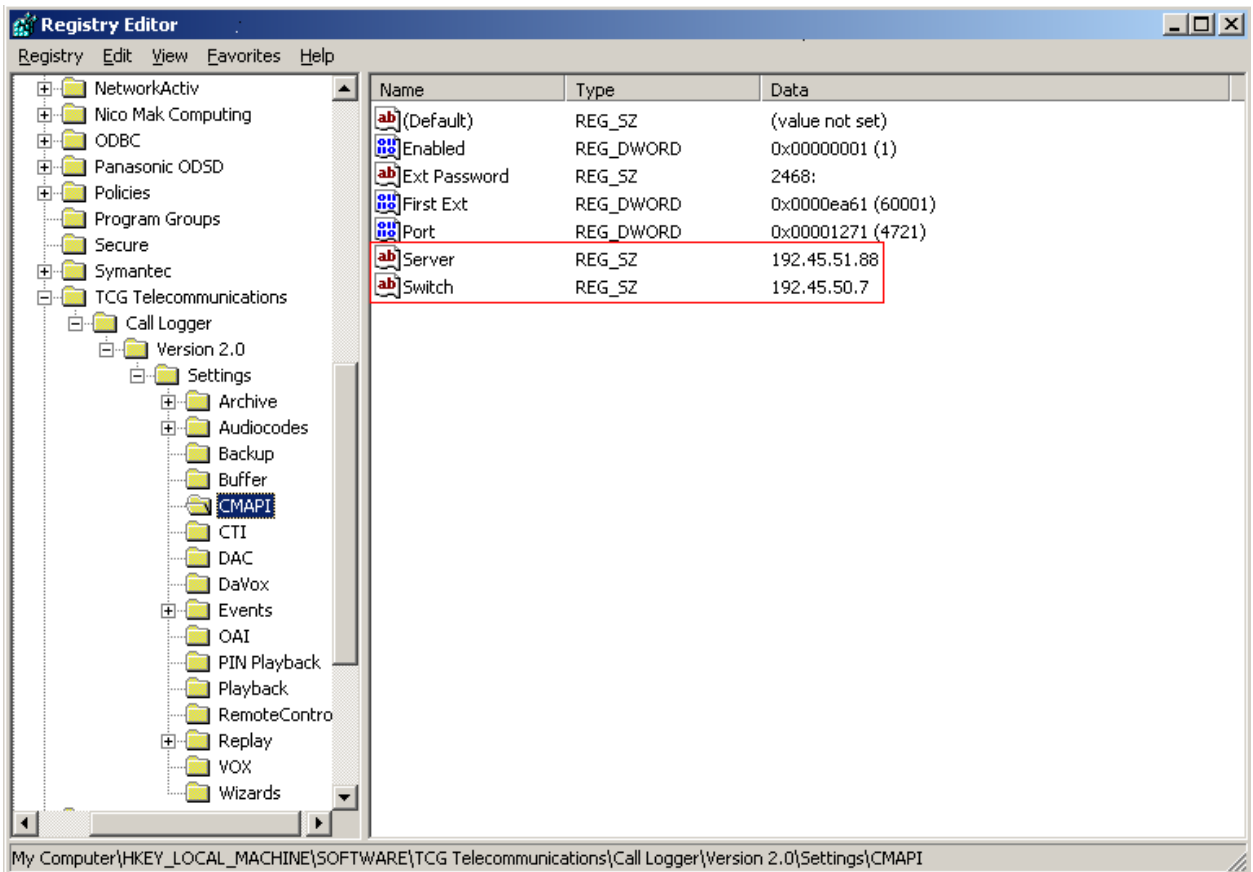
change ip-network-region 1	Page 1 of 19
IP NETWORK REGION	
Region: 1	
Location:	Home Domain:
Name:	
AUDIO PARAMETERS	Intra-region IP-IP Direct Audio: yes
Codec Set: 1	Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048	IP Audio Hairpinning? y
UDP Port Max: 3029	RTCP Reporting Enabled? y
	RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS	Use Default Server Parameters? y
Call Control PHB Value: 34	
Audio PHB Value: 46	
802.1P/Q PARAMETERS	
Call Control 802.1p Priority: 7	
Audio 802.1p Priority: 6	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	

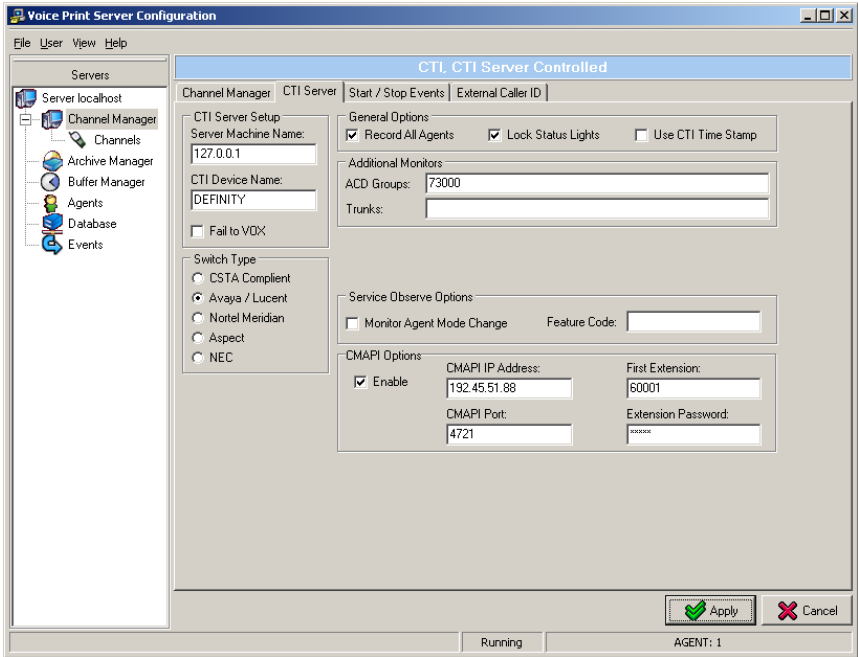
3.5. Recorded Stations

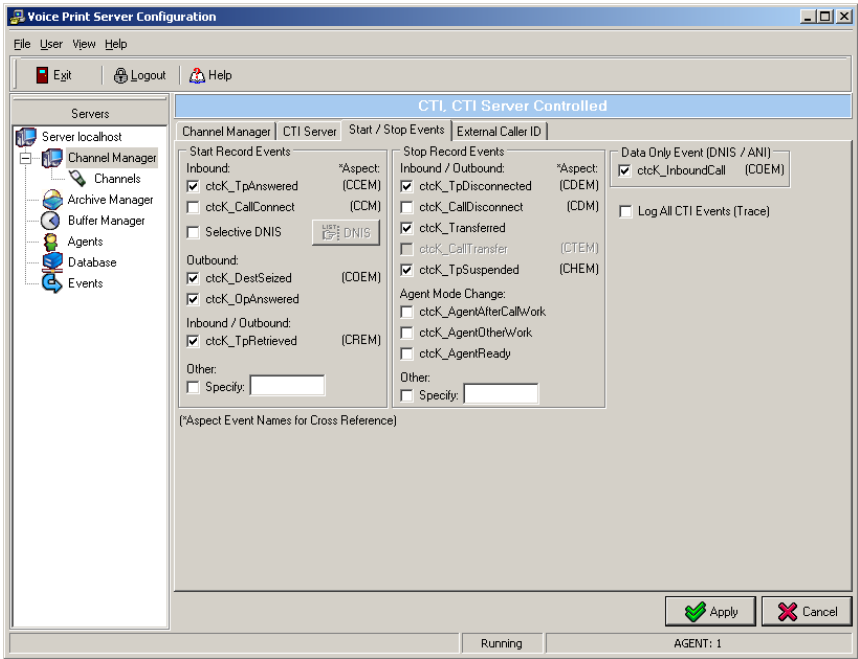
The stations that were recorded during the compliance testing include analog, digital, and IP telephones, and Avaya IP Softphone in both Road Warrior mode and Telecommuter mode. The extensions used were in the ranges 50001 – 50016 and 50101 – 50180.

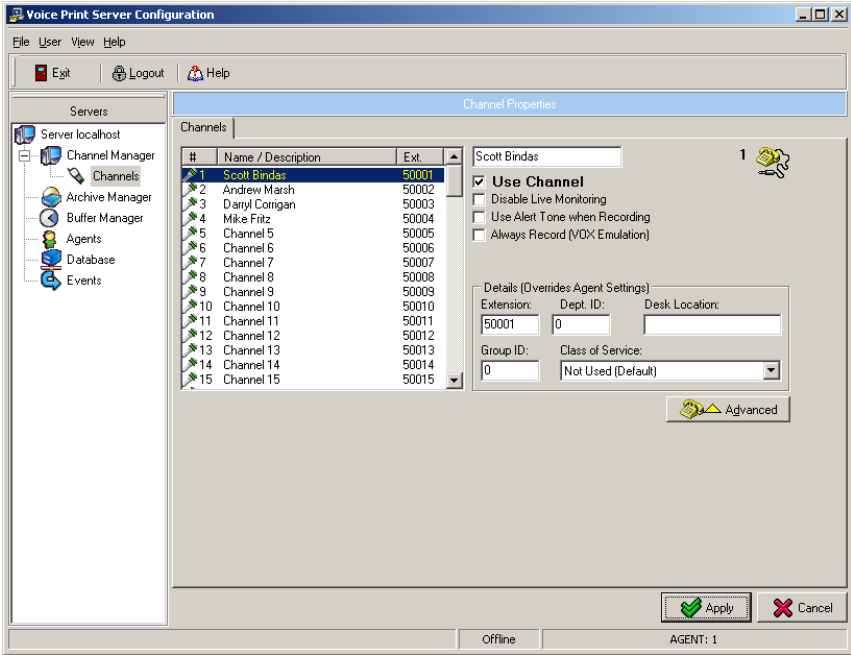
4. Configure Voice Print Activ! Voice

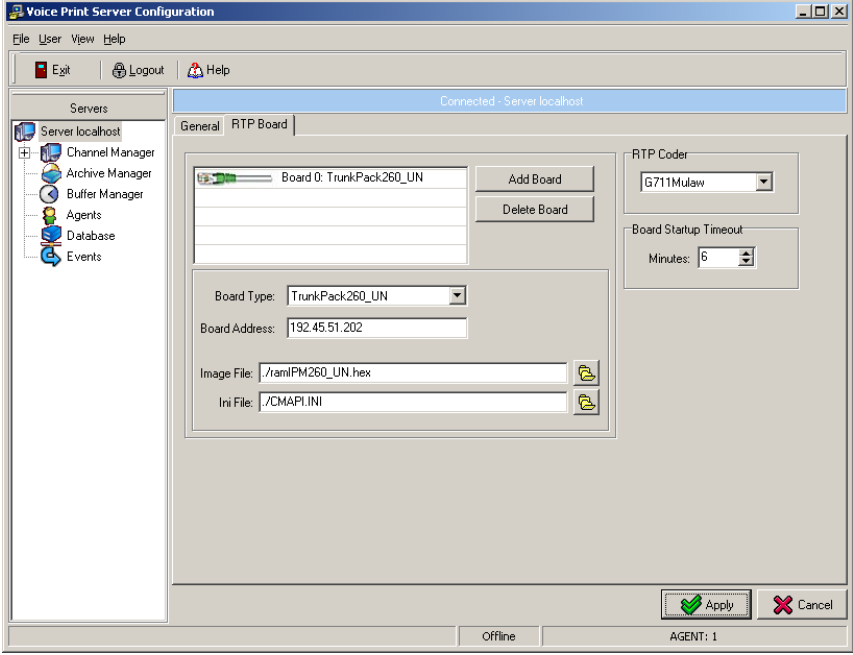
The steps in this section describe the configuration of CTI and Communication Manager API settings, stations to be recorded, and Communication Manager API recording stations on the Activ! Voice server.

Step	Description
1.	<p>On the Activ! Voice server, edit the Windows registry as shown below. Set Server to the IP address of the Communication Manager API server and Switch to the IP address of the C-LAN (S8500 and S8700 Media Servers) or Processor Ethernet (S8300 Media Server).</p> 
2.	<p>Launch the Voice Print Server Configuration program and log in with the appropriate credentials.</p>

Step	Description
3.	<p>Select “Channel Manager” in the left panel, and then the “CTI Server” tab. Set Server Machine Name to the hostname or IP address of the NetMerge Call Processing Server (127.0.0.1 in the example below because the NetMerge Call Processing Server is installed on the same server as the Activ! Voice server), enter a descriptive CTI Device Name, set Switch Type to “Avaya/Lucent”, and enter the extensions of any skill/hunt groups to be monitored as a comma-separated list in the ACD Groups field.</p> <p>In the CMAPI Options area, check the Enable checkbox and specify the following:</p> <ul style="list-style-type: none"> • CMAPI IP Address: IP address of the Communication Manager API server • CMAPI Port: Communication Manager API listen port configured on the Communication Manager API server (default is 4721). • First Extension and Extension Password: Extension and password of the first Communication Manager API station to be used for recording (see Section 3.3). Note that for Activ! Voice, the passwords of all Communication Manager API stations used for recording must be the same. <p>Click on “Apply”.</p> <p>Note: The Activ! Voice server automatically sets the number of Communication Manager API recording stations to the number of channels configured on the installed hardware on the Activ! Voice server.</p> 

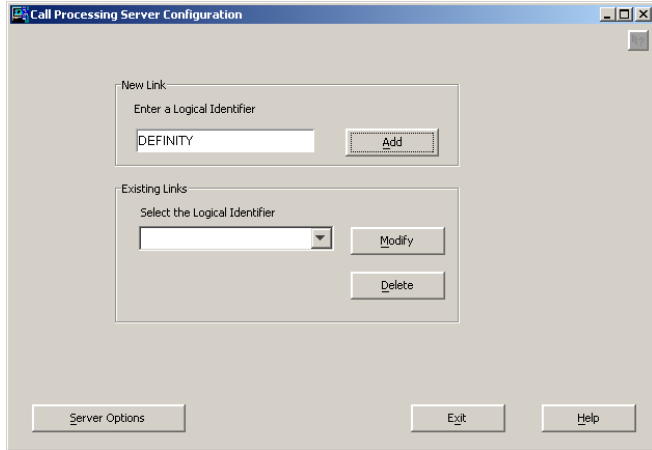
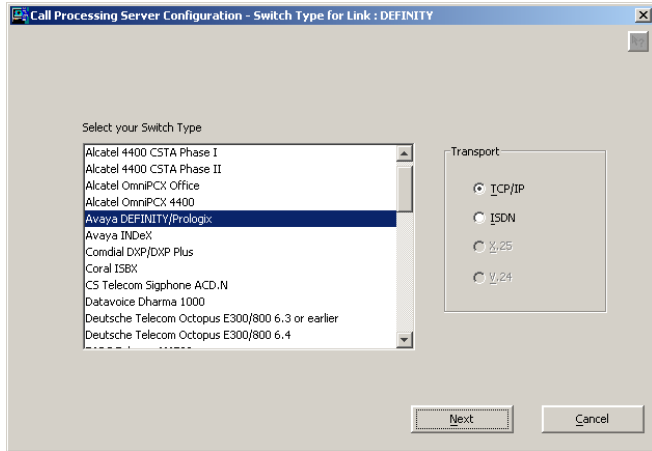
Step	Description
4.	<p>Click on the “Start / Stop Events” tab. Check the following checkboxes:</p> <ul style="list-style-type: none"> • ctcK_TpAnswered • ctcK_DestSeized • ctcK_OpAnswered • ctcK_TpRetrieved • ctcK_TpDisconnected • ctcK_Transferred • ctcK_TpSuspended • ctcK_InboundCall <p>Click on “Apply”.</p> 

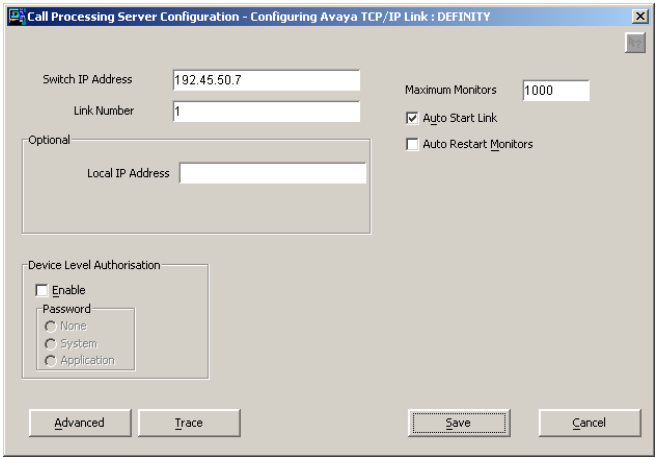
Step	Description
5.	<p>Select “Channels” in the left panel. For each station to be recorded by Activ! Voice (see Section 3.5), select a channel, enter the Extension of the station and a meaningful Name / Description, and check the Use Channel checkbox. Click on “Apply” when all stations to be recorded have been specified.</p>  <p>The screenshot shows the 'Voice Print Server Configuration' window. On the left, a tree view under 'Servers' has 'Channels' selected. The main area displays a table of channels with columns for ID, Name/Description, and Extension. Channel 1 is selected, showing 'Scott Bindas' and extension '50001'. To the right, the 'Channel Properties' panel for Scott Bindas is visible, with the 'Use Channel' checkbox checked. Below this, a 'Details' section contains fields for Extension (50001), Dept. ID (0), Desk Location, Group ID (0), and Class of Service (Not Used (Default)). At the bottom right are 'Apply' and 'Cancel' buttons. The status bar at the very bottom indicates 'Offline' and 'AGENT: 1'.</p>

Step	Description
6.	<p>Select “Server localhost” in the left panel, and then the “RTP Board” tab. Select a board, configure an IP address for Board Address, and select “G711Mulaw” for RTP Codec. Note that RTP Codec must match the codec configured on Avaya Communication Manager in Section 3.4. Click on “Apply”.</p>  <p>The screenshot shows the 'Voice Print Server Configuration' window. On the left, under 'Servers', 'Server localhost' is selected. The main window has two tabs: 'General' and 'RTP Board'. The 'RTP Board' tab is active, showing a list of boards with 'Board 0: TrunkPack260_UN' selected. Below this, the 'Board Type' is 'TrunkPack260_UN', 'Board Address' is '192.45.51.202', 'Image File' is './ramlPM260_UN.hex', and 'Ini File' is './CMAPI.INI'. On the right, the 'RTP Codec' is set to 'G711Mulaw' and the 'Board Startup Timeout' is '6' minutes. At the bottom right, there are 'Apply' and 'Cancel' buttons. The status bar at the bottom shows 'Offline' and 'AGENT: 1'.</p>

5. Configure Intel NetMerge Call Processing Server

The steps in this section describe the CTI link configuration on the Intel NetMerge Call Processing Server.

Step	Description
1.	Launch the Intel NetMerge Call Processing Server Configuration Program.
2.	<p>Enter a descriptive name for Enter a Logical Identifier, and click on “Add”.</p> 
3.	<p>Select “Avaya DEFINITY/Prologix” from the Select your Switch Type list, and “TCP/IP” for Transport. Click on “Next”.</p> 

Step	Description
4.	<p>For Switch IP Address, enter the IP address of the C-LAN (S8500 and S8700 Media Servers) or Processor Ethernet (S8300 Media Server). For Link Number, enter the client link number configured in Step 4 of section 3.1. Click on “Save”.</p> 

6. Interoperability Compliance Testing

The interoperability compliance testing included feature, serviceability, and performance testing. The feature testing evaluated the ability of Activ! Voice to monitor and record calls placed to and from stations, agents, and VDNs. The serviceability testing introduced failure scenarios to see if Activ! Voice can resume recording after failure recovery. The performance testing stressed the Activ! Voice server by continuously placing calls to a VDN over extended periods of time.

6.1. General Test Approach

The general approach was to place various types of calls to and from stations, IP Softphones, agents, and VDNs, monitor and record the calls using Activ! Voice, and verify the recordings. For feature testing, the types of calls included internal calls, inbound trunk calls, outbound trunk calls, transferred calls, conference calls, Redirection On No Answer (RONA) calls, and Switch-Classified calls. For performance testing, a call generator continuously placed calls to a VDN that queues the calls in a hunt/skill group, which in turn delivers the calls to agents logged into the hunt/skill group. For serviceability testing, failures such as cable pulls, CTI link busyouts/releases, and resets were applied.

6.2. Test Results

Activ! Voice successfully monitored, recorded, stored, and played back the various types of calls discussed in Section 6.1. For serviceability testing, Activ! Voice was able to resume recording calls after restoration of connectivity to the S8500 Media Server, after busyout/release of the CTI link, and after resets of the Activ! Voice server, S8500 Media Server, and Communication

Manager API server. For performance testing, Activ! Voice successfully recorded calls under a moderate call volume using 60 Communication Manager API virtual stations as the recording ports for over 16 consecutive hours.

7. Verification Steps

The following steps may be used to verify the configuration:

- From the Voice Print Activ! Voice server, ping the agent desktop computers and Avaya IP telephones and verify connectivity.
- From the Voice Print Activ! Voice server, ping the Avaya G650 Media Gateway C-LAN and Media Processor boards and verify connectivity.
- Verify the CTI link between Avaya Communication Manager and the Intel NetMerge Call Processing Server is up (use the **status dlgt-cti-link** command on the SAT).
- Verify that the Voice Print Activ! Voice server recording ports are registered as Communication Manager API stations in Avaya Communication Manager (use the **list registered-ip-stations** command on the SAT).
- Verify that calls may be successfully completed between the Avaya IP and Digital telephones, analog telephones, and Avaya IP Softphones. Verify that the call recordings are accurate and complete.
- Log agents into a hunt/skill group and verify that calls may be successfully completed to and from the agents. Verify that the call recordings are accurate and complete.

8. Support

For technical support on Voice Print products, contact Voice Print at:

- Phone: (805) 389-5201
- Email: support@voiceprintonline.com

9. Conclusion

These Application Notes illustrate the procedures for configuring the Voice Print Activ! Voice Call Logger to monitor and record calls placed to and from stations, softphones, and agents on an Avaya Communication Manager system. In the configuration described in these Application Notes, Activ! Voice employs Communication Manager API virtual stations as recording ports. During compliance testing, Activ! Voice successfully monitored and recorded calls placed to and from Avaya IP and Digital Telephones, analog telephones, Avaya IP Softphones, and agents, as well as calls placed to a VDN and then queued to an agent hunt/skill group. Activ! Voice was also able to record calls under continuous call volumes over extended periods of time.

10. Additional References

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

Product information for Voice Print products may be found at <http://www.voiceprintonline.com/call-recorders.asp>.

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