

Avaya Solution & Interoperability Test Lab

Application Notes for Mercom Audiolog Recording Server with Avaya Communication Manager and Avaya Computer Telephony - Issue 1.0

Abstract

MERCOM Audiolog Recording Server monitors real-time call events at selected extensions and skill sets, analyzes the event data, makes recording control decisions, and collects callspecific data for a call center. MERCOM Audiolog Recording Server integrated with the Avaya Computer Telephony Server to provide highly accurate and flexible automatic call recording control. 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

MERCOM Audiolog is an advanced call recording system that is designed for use as a standalone recorder/playback unit and as a specialized resource server within a networked environment.

Audiolog performs the following recording functions:

- Recording audio via direct connection (stations or trunks), Service Observe, or VoIP
- Monitoring audio channels during recording
- Storing/Archiving recordings for playback
- Creation/maintenance of Catalog Database of recordings
- Search, Selection, and Playback of selected recordings
- Prevention of unauthorized access, modification, monitoring and playback

Audiolog software utilizes several co-resident modules to perform all computer telephony integration (CTI), recording, Database, and playback functions:

- Recorder Module provides telephony interface, signaling, compression, and recording functions.
- CTI Link Module provides the direct interface to the Avaya Computer Telephony Server for recording control and call-associated data collection.
- Call Manager Module provides automated update of Audiolog's onboard SQL Catalog database as each call is recorded.
- Player Module provides user-friendly GUI for call search, selection, and playback

MERCOM Audiolog Recording Server can be easily integrated with the Avaya Computer Telephony Server to provide highly accurate and flexible automatic call recording control. MERCOM Audiolog Recording Server monitors real-time call events at selected extensions and skill sets, analyzes the event data, makes recording control decisions, and collects call-specific data for ease of future recording retrieval.

Test configuration of MERCOM Audiolog Recording Server utilized Avaya Computer Telephony Server to provide events associated with each call to be recorded. Call related information, provided through the CTI link, is stored in MERCOM's SQL Catalog Database. Recorded voice is stored on the hard drive(s) in the MERCOM server. By using standard Mercom search and replay applications in conjunction with the Catalog database call records, the voice recording is played back at the Mercom server or at networked user workstation(s).

2. Test Configuration

Figure 1 illustrated a configuration used during the compliance test process.

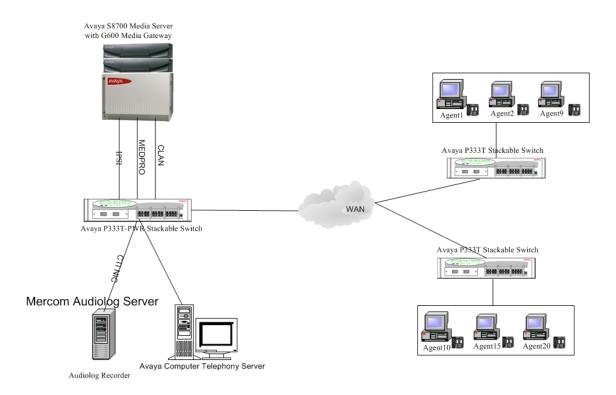


Figure 1: Avaya DeveloperConnection Compliance Test Configuration

3. Equipment and Software Validated

The following equipment and software were used for the tested configuration:

Equipment	Software
Avaya S8700 Media Server with G600 Media Gateway	Avaya Communication Manager 2.1
Avaya Computer Telephony Server	Release 1.3
MERCOM Audiolog Recording Server	v3.20ML RTP
Avaya TN464F DS1 INTFC	V10

4. Configure the Avaya Communication Manager

4.1. Verify Communication Manager Customer Option

The following step verifies customer options on the Avaya Communication Manager.

Step	Description				
1.	1. From the SAT terminal, type "display system-parameters customer-options" and to page 3 to verify the options below are set to " y ". Contact Avaya if these option are not set to " y ".				
 Verify the following customer options are set to "y": Computer Telephony Adjunct Links 					
	Co-Res DEFINITY LAN Gateway				
	display system-parameters customer-option	s 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?	У		
	ARS/AAR Dialing without FAC? n	Co-Res DEFINITY LAN Gateway?	У		
	ASAI Link Core Capabilities? y	Cvg Of Calls Redirected Off-net?	n		
	ASAI Link Plus Capabilities? y	DCS (Basic)?	У		
	Async. Transfer Mode (ATM) PNC? n	DCS Call Coverage?	У		
	Async. Transfer Mode (ATM) Trunking? n	DCS with Rerouting?	У		
	ATM WAN Spare Processor? n				
	ATMS? n	Digital Loss Plan Modification?	n		
	Attendant Vectoring? n	DS1 MSP?			
		DS1 Echo Cancellation?	n		
	(NOTE: You must logoff & login to	effect the permission changes.)			

4.2. Configure CTI Link on Avaya Communication Manager

The following steps configure the CTI link on Avaya Communication Manager.

Step	Description						
1.	From the SAT terminal, type " add cti #", where # is the next available CTI link number. Enter an extension number in the extension field, enter ADJ-IP for Type and a name for this CTI Link.						
	add cti-link 6 Page 1 of 2						
		CT	I LINK				
	CTI Link: 6						
	Extension: 2399	9					
	Type: ADJ-	,IP					
				COR: 1			
	Name: CT S	Server AIC LAB					
<u> </u>							
2.	vaya Computer Telephony						
	change node-nam	les ip		Page 1 of 1			
	IP NODE NAMES						
	Name	IP Address	Name	IP Address			
	CTserver	192.45 .20 .163					
	clan	192.45 .100.66					
	default	0.0.0.0					
	medpro	192.45 .100.69					
	procr						
3.	4.2.1 for CTI Li		ter Avaya Compu	I link number created in step ater Telephony node name ent Link.			
	change ip-servi	ces		Page 3 of 3			
			inistration	2			
	CTI Link F	Inabled Client Name	Client Link	Client Status			
	6	y CTserver	1	in use			

4.3. Configure DS1 Stations for Mercom Recorder

The following steps configure DS1 station used for Single Step Conference:

tep	Description			
1.	From the SAT terminal, type Can Be A Service Observer to	'change COR 1 ''. Set Can be Service Observed and o " y ".		
	change cor 1	Page 1 of 4		
		CLASS OF RESTRICTION		
	COR Number: 1			
	COR Description:			
	FRL: 0	APLT? Y		
	Can Be Service Observed? y	Calling Party Restriction: none		
	Can Be A Service Observer? y	Called Party Restriction: none		
	Partitioned Group Number: 1	Forced Entry of Account Codes? n		
	Priority Queuing? n	Direct Agent Calling? y		
	Restriction Override: no	ne Facility Access Trunk Test? n		
	Restricted Call List? n	Can Change Coverage? n		
	Access to MCT? y	Fully Restricted Service? n		
	Group II Category For MFC: 7	Hear VDN of Origin Annc.? n		
	Send ANI for MFE? n	Add/Remove Agent Skills? n		
	MF ANI Prefix:	Automatic Charge Display? n		
	Hear System Music on Hold? y	PASTE (Display PBX Data on Phone)? n		
	Can B	e Picked Up By Directed Call Pickup? n		
		Can Use Directed Call Pickup? n		
		Group Controlled Restriction: inactive		

2.	From the SAT terminal, type " add board, hdb3 for Line Coding, 2.04 Interconnect, mulaw for Interface the TN464 board is set to 32 channel	18 for Bit Rate, CAS for Signal Companding. Make sure that t	ing Mode, pbx for
	change dsl 1b04		Page 1 of 1
	D	S1 CIRCUIT PACK	
	Location: 01B04	Name: MERCO	ОМ
	Bit Rate: 2.048	Line Coding: hdb3	
	Signaling Mode: CAS		
	Interconnect: pbx	Country Protocol: 1	
	Interface Companding: mulaw	CRC? n	
	Idle Code: 1111111		
	Slip Detection? n	Near-end CSU Type: other	
3.	From the SAT terminal, type "add port number and name. add station 27001	Pag	
		STATION	
	Extension: 27001	Lock Messages? n	BCC: 0
	Type: DS1FD Port: 01B0401 Name: MERCOM STATION OPTIONS	Security Code: Coverage Path 1: Coverage Path 2: Hunt-to Station:	TN: 1 COR: 1 COS: 1 Tests? y
	Loss Group: 4 Off Premises Station? y R Balance Network? n		

Step	Description				
	Go to page 2 of the station form, set Auto Answer to " all " and Adjunt Supervision to " y '. Repeat this step for the remaining 31 channels.				
	add station 27001	Page 2 of 3			
		STATION			
	FEATURE OPTIONS				
	LWC Reception: spe				
	LWC Activation? y	Coverage Msg Retrieval? y			
	LWC Log External Calls? n	Auto Answer: all			
	CDR Privacy? n	Data Restriction? n			
	Redirect Notification? y	Call Waiting Indication? y			
	Per Button Ring Control? n	Att. Call Waiting Indication? y			
		Distinctive Audible Alert? y			
	Switchhook Flash? y	Adjunct Supervision? y			
	Ignore Rotary Digits? n				
	H.320 Conversion? n	Per Station CPN - Send Calling Number?			
	Service Link Mode: as-needed	3			
	Multimedia Mode: basic				
	MWI Served User Type:				
	AUDIX Name:				
		Coverage After Forwarding? s			
	Emergency Location Ext: 27001				

5. Configure the Avaya Computer Telephony Server

5.1. Modify the TSLIB.INI File

Step	Description					
1.	On the computer running Avaya Computer Telephony Server, open the TSLIB.INI file by going to Start → Programs → TS Win32 Client → Edit TSLIB.INI.					
	Image: Decomposition of the sector of the secto	 Accessories Startup Startup Command Prompt Commond Prompt Windows NT Explorer Administrative Tools (Common) Telephony Services Admin Adobe Acrobat Adobe Acrobat TS Client Readme TS Spy TS Vin32 Client Btrieve Technologies Database 				
	Start Start					

Step	Description					
2.	Change the IP address shown in the [Telephony Servers] section of the TSLIB.ini file to match the IP address (or Computer Name) of the Avaya Computer Telephony Server to which the Audiolog server will connect as a client. Audiolog communicates with the Avaya Computer Telephony Server using port 450. The syntax for this line is IPAddress (or Computer Name)=450, for example 192.45.20.163=450 or CTserver=450.					
	Sile Edit Format Help					
	[[Telephony Servers]					
	; This is a list of the servers offering Telephc ; Either domain name or IP address may be used; ; The form is: host_name=port_number For examp ; ; tserver.mydomain.com=450 192.45.20.163=450 ;					
	[Shared Admin]					
	; Instead of each workstation maintaining its ow ; tslib.ini file may be placed on a network file					
	; tslib.ini=n:\csta\tslib.ini					
	; This entry overrides the [Telephony Servers] s					
3.	Save and Exit the file.					
4.	Copy the TSLIB.ini file to the C:\Winnt directory on the Audiolog.					

5.2. Telephony Services Administration

Step	Description				
1.	On the Avaya Computer Telephony Server, open the Telephony Services Admin by going to Start → Programs → TS Win32 Client → Telephony Services Admin .				
	Programs Image: Accessories Image: Documents Image: Startup Image: Documents Image: Startup Image: Documents Image: Command Prompt Image: Documents				
2.	Select the computer name of the Avaya Computer Telephony Server (TSAPI Server) from the Tserver drop-down list.				
	NOTE: In order to access Telephony Services, users must have a Windows NT/2000 login ID and password. In addition, they must have the "Log on as a Service" user right assigned to their account on the Windows NT/2000 machine that is running Telephony Services. Windows NT logins are administered through the Windows NT User Manager.				

Step	Description				
3.	Click the Create User button.				
	Image: Subscription of the second				
4.	On the Information tab, enter ALUSER1 in the Login field.				
5.	Enter ALUSER1 in the Name field. (ALUSER1 must also be added as an Administrator in the User Manager on the TSAPI Machine).				
	Name: Worktop Name: (none) Image: Concel OK Cancel Apply Help				
6.	On the Access Rights tab, select "Any Device" for all sections.				

Step	Description					
7.	Check the Allow box under Call/Call section.					
	La Create User - CLASS-CTI					
	Information Access Rights Admin Access Groups					
	Call Control Services - Call Origination and Termination Access Group: Any Device Monitoring-Only Services					
	Device/Device - Event Notification ceases if call leaves device Access <u>G</u> roup: Any Device ▼					
	Call/Device - Event Notification continues if call leaves device Access Group: Any Device					
	Call/Call - Event Notification allowed if call identifier is known					
	Routing Services - Allow routing on listed devices					
	Access Group: Any Device					
	OK Cancel Apply Help					
8.	On the Admin Access Groups tab, check the Allow User to Administer Admin Access Groups checkbox.					
	Create User - CLASS-CTI					
	Information Access Rights Admin Access Groups					
	Allow User to Administer Admin Access Groups?					
	OK Cancel Apply Help					

Step	Description				
9.	Click Apply and then OK.				
10.	Verify that all devices to be monitored are listed under the Devices node. This includes all extensions and hunt groups, if applicable.				
	Telephony Services Adminis		_RECORDER Devices]		
	Eile Admin ⊻iew Window	<u>H</u> elp			<u>_8×</u>
	🛓 🕿 🧏 🕾 🚇 🖴	🔌 🛎 🖻	× ••••	? №?	
	🖃 📲 SDB	Device ID	Tlink Group	Device Type Location	
	Sers	🕿 100	Any Tlink	PHONE	
	Worktops	🕿 101	Any Tlink	PHONE	
	Devices	🕿 102	Any Tlink	PHONE	
	Bevice Groups	🕿 103	Any Tlink	PHONE	
	Tlinks	🕾 104	Any Tlink	PHONE	
	Tlink Groups	🕾 105	Any Tlink	PHONE	
	Admin Access droups	🕿 106	Any Tlink	PHONE	
		🕿 107	Any Tlink	PHONE	
		🕿 108	Any Tlink	PHONE	
		🕾 109	Any Tlink	PHONE	
		☎ 110	Any Tlink	PHONE	
		☎ 111	Any Tlink	PHONE	
		☎ 112	Any Tlink	PHONE	
		☎ 113	Any Tlink	PHONE	
		☎ 114	Any Tlink	PHONE	
		☎ 115	Any Tlink	PHONE	
		월 116 월 120	Any Tlink Any Tlink	PHONE	
		₩ 120 ₩ 121	Any Tlink Any Tlink	PHONE PHONE	
		₩ 121 ₩ 122	Any Tlink Any Tlink	PHONE	
		₩ 122 ₩ 123	Any Tlink Any Tlink	PHONE	
		23 123 28 124	Any Tlink Any Tlink	PHONE	
		124 125 125	Any Tlink	PHONE	
		₩ 125 ₩ 126	Any Tlink	PHONE	
	<u> </u>	₩ 120 ₩ 127	Any Tlink Any Tlink	PHONE	•
	For Help, press F1				

5.3. Telephony Services Controller

Step	Description
1.	Verify that the Telephony Service is running by going to Start \rightarrow Programs \rightarrow TSAPI Telephony Services \rightarrow TSAPI Telephony Services Controller.
	Programs Image: Accessories Image: Documents Image: Documents<
	Start SAPI_Telephony_Servic
2.	When changes are made, click on the Refresh button to update.
	Status Telephony Services State: Plant Statup Automatically Start Telephony Services On Server Boot Recovery Automatically Restart Telephony Services Automatically Restart Telephony Services

6. Configure the Audiolog Recorder

6.1. Requirements

- Audiolog Version 2.70 or later
- Single or Dual Network Cards Depending on the LAN Environment
- Audiolog CTILink Module option
- Audiolog TSAPI Avaya Definity G3 Integration Enabler option
- Audiolog Free-Seating option is required if Free Seating (Agent Logon/Logoff detection) is to be used
- Audiolog Service Observe option (licenses) required for Single Step Conference
- The Audiolog recorder requires terminate-type telephony interface cards, such as the ANA-TERM or T1/E1 TERM cards depending upon the ECS Station ports selected for use with Single Step Conference
- Workstations running Audiolog client software must be able to access the Audiolog server.

6.2. LAN Connections

Connect the Audiolog recorder to the same LAN segment as the Avaya Computer Telephony Server.

6.3. Audiolog E1 Configuration

Step	Description
1.	Verify the settings on the PCI telephony interface cards. If there is no Smart Control icon on the desktop, from the Start button, click Control Panel.

Step	Description
2.	Click the SmartControl icon. The Ai-Logix SmartWORKS dialog box opens.
	Ai-Logix Inc SmartWORKS
	Syntem Board GPM Parameters Digital Network Driver Version 90.00.4 9225 Centrol Panel Version 90.4.00081 Max Log Count 90.0 MiNP Stating Stot 0 MiNP Stating Stot 0 MiNP Stating Stot 0 O Cold Stating Weix 256 O Cold Stating Weix 0 Diver Juit If Allow Burs Segmentation Diver Juit If Allow Burs Segmentation Detsuit Detsuit
3.	On the System tab, verify the GCI Starting Index is set to 1.
4.	Check the Allow Bus Segmentation box.

Step	Description
5.	Click the Board tab.
	Ai-Logix Inc SmartWORKS
	System Board CPM Parameters Digital Network
	Select Board Board Number PCI Bus No: 3 PCI Slot No: 1
	Information For Board 1
	Board Type SmartTERM DT6409 Dual E1
	Total Channels 60 Serial Number 8585
	DateCode 0333
	Firmware Version 03.00.04 Build 0442 OEM Info Ai-Logix, Inc.
	Copyright Copyright © 2002 Ai-Logix, Inc. All rights reserved.
	T1E1Option E1
	TDM Encoding
	wei-H wei-H
	C H.100 • MVIP
	CT Mode Defaults
	C Master A. C MasterB
	Apply OK Cancel
6.	Select the Board Number of the T1/E1 card from the Board Number drop-down list.
7.	Verify the TDM Encoding is set to µ-law .
8.	Verify the Avaya Computer Telephony Bus Type is set to MVIP .
9.	Verify the Avaya Computer Telephony Mode is set to Master.
10.	Click the Apply button if you made any changes.
11.	Click the CPM tab.

Step	Description	
12.	Verify the settings shown in the following illustration.	
	Ai-Logix Inc SmartWORKS	
	System Board CPM Parameters Digital Network	
	Board Number : 1 Signal Index Signal Name	
	Signal Type : CPM_TONE Cycles : 0	
	Event Types	
	Image: Enable Image: DetectSIT DetectLate Image: DetectLate Image: DetectCycle Image: DetectLate Image: DetectLate	
	Frequency (Hz)	
	1 350 2 440 3 0 C Busy	
	Hi Min 2000 0 0 0 C Loop Off	
	Hi Max 0 0 0 Digit	
	Lo Min 1 0 0 0	
	Default	
	Apply OK Cancel	

Step	Description
13.	Increment the Signal Index box until fax_t1 is displayed in the Signal Name box.
	System Board CPM Parameters Digital Network Board Number : 1
	Signal IndexSignal Name
	Signal Type : CPM_TONE Cycles : 0
	Enable DetectSIT DetectLate DetectCycle DetectLost C Dial C Rindback C Rindback
	Frequency (H2) C Busy 1 2060 2 0 3 0 C Fax Cadence (ms) C SIT 0 0 0 0 0
	Hi Min 2000 0 0 0 C Loop Off Hi Max 0 0 0 0
	Lo Min 1 0 0 Lo Max 1 0 0 0 0 0 0 0 0 0
	Default
	Apply OK Cancel
14.	Uncheck the Enable box and click Apply . NOTE: If Fax events are disabled on the Ai-Logix board, Audiolog will detect beep tones as Fax events.

Step	Description
15.	Click the CPM tab and verify the settings shown in the following illustration.
	Ai-Logix Inc SmartWORKS
	System Board CPM Parameters Digital Network
	Board Number : 1
	Dodart Name: - 1 DTMF On (ms) TOMF On (ms) TOMF Of (ms) Activity Detection Min Silence (ms) 40 Min Activity (ms) Coop Loop Minimum (ms) Loop Start Delay (ms) Advanced Advanced
	Apply OK Cancel

Step	Description
Step	Click the Digital Network tab.
	Apply OK Cancel
17.	Set the T1E1 Option for E1.
18.	Set the Framing to Basic G.704 for each digital trunk on the board.
19.	Set the Line Coding to HDB3 for each digital trunk on the board.
20.	Set the LBO (Line Build Out) to 1200HM for each digital trunk on the board
21.	Set the Signaling to None for each digital trunk on the board

Step	Description
22.	Click the Apply button if any changes were made.
23.	Repeat steps 6 to 22 for each T1/E1 board in the system.
24.	Click the OK button.
25.	After all PCI Telephony interface changes have been made, reboot the Audiolog recording server.

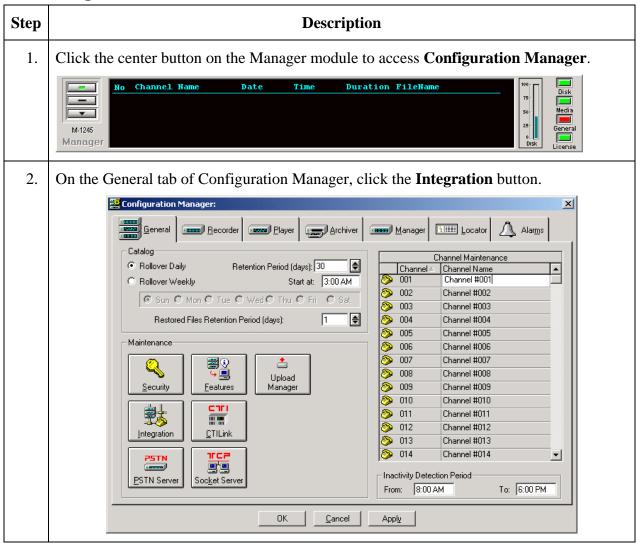
6.4. Registry

Caution: If you use Registry Editor incorrectly, you may cause serious problems that may require you to reinstall Audiolog and/or your operating system. Mercom cannot guarantee that you can solve problems that result from using Registry Editor incorrectly. Use the Registry Editor at your own risk.

Verify the following system registry settings. All settings are located under the HKEY_LOCAL_MACHINE\Software\ Mercom\Audiolog\CurrentVersion\ key.

Кеу	Value	Data
CTILink\Communications\TSAPI	Enumerate Servers	Yes
CTILink\Communications\TSAPI	Enable System Status	Yes
CTILink\Communications\TSAPI	Private Data Version	6
CTILink\Communications\TSAPI\ EventMap\Triggers	Outbound Start Record	59
CTILink\Misc Options	Enable CTI Housekeeping	Yes
CTILink\Misc Options	Enable Enhanced Call Tracking	Yes
CTILink\Misc Options	Enable Free Seating	Yes
CTILink\Misc Options	Enable Service Observe	Yes
Recorder\Misc Options	SSC Feature Code	{enter the code used in the Communication Manager to deactivate the "Send All Calls" feature}

6.5. Integration Tables



3. In the Channel/Device Maintenance table, enter the extension numbers of the PBX station ports the Audiolog is connected to in order to perform SSC. Select the (telephone) icon for the Type.

	ntegration										
				ice M	aintenance					Agent Maintenance	
	Channel 🔺	Channel Na		Тур		ID			Agent 🔺	Agent Name	
2	001	Channel #		٨	27001			▶ 9	•	32101	
۵	002	Channel #0		٨	27002				~	32102	
8	003	Channel #0)03	٨	27003			\$		32103	
8	004	Channel #0)04	٨	27004				32104	32104	
8	005	Channel #0)05	٨	27005				32105	32105	
8	006	Channel #0)06	٨	27006				32106	32106	
8	007	Channel #0)07	8	27007			<u> </u>	32107	32107	
8	008	Channel #0)08	8	27008			<u> </u>	32108	32108	
8	009	Channel #0)09	8	27009				32109	32109	
	010	Channel #0	110		27010				S 32110	32110	
							vice Ma	_			
		vicelD 🔺	Enable	PM	Agent	Agent Name		Ph	yDeviceID		
	×	000			32101	32101					
	300 📎	001			32102	32102					
	300 🃎	002			32103	32103					
	300 🧼	003			32104	32104					
Н	300 🍥	004			32105	32105					
Н	- 300	005			32106	32106					
		006			32107	32107					
	· · · · ·	007			32108	32108					
	🌮 300	107			32100	32100					
							<u>C</u> lose	1			

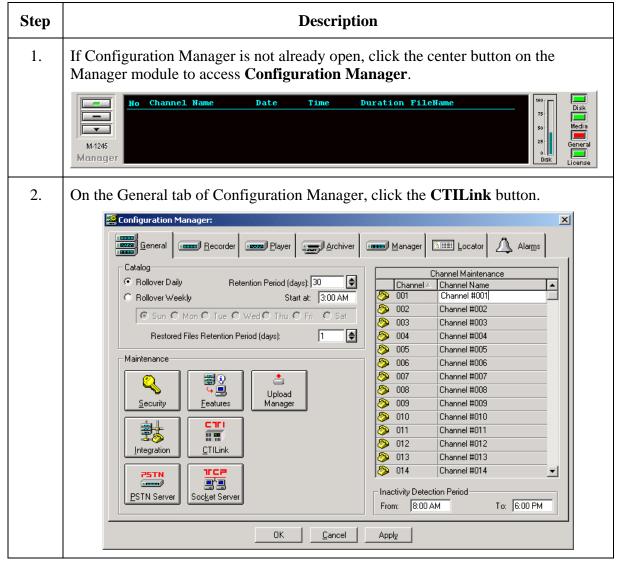
5. Program the Device Maintenance table with the extension numbers to be recorded.

NOTE: When using Expert Agent Selection, the extension number of the "Skill Set Hunt Group" must also be entered in the Device Maintenance table using the 🏈 (telephone) icon. However, leave the **Enable** and **PM** boxes unchecked.

🖼 Integration								
		laintenance			_	<u> </u>	Agent Maintenance	
Channel A Channel Na O01 Channel #0				≜ŀ.		Agent △ 32101	Agent Name 32101	A
> 001 Channel #0				ΞĽ			32102	
OO2 Channel #0							32102	
> 003 Channel #0	··· 🖓						32104	
> 004 Channel #0				H			32105	
> 005 Channel #0	-				2 0		32105	
> 000 Channel #0	-						32107	
> 007 Channel #0							32108	
O00 Channel #0	-				-1 **		32109	
A 010 Channel #0				-I-			32103	
		,,,,,,,	Devic	e Maint				
Type A DeviceID A	Enable PM	Agent	Agent Name	C Main		DevicelD		
30000		32101	32101					
30001		32102	32102					
30002		32103	32103					
30003			32104					
			32104					
30005			32106					
30006		32107	32107					
30007		32108	32108					•
	1	'		lose				

6.6. CTILink Configuration

6.6.1. General Link Tab



Step	Description	
3.	Click the General Link tab.	
	🖌 CTILink Configuration	
	General Link Options / CT Server Serial User Data TCP/IP Triggers	
	CT Server Server Name 1: Server Port 1: 0 Server Name 2: Server Name 2: 0 Server Port 2: 0 Server ID: aluser1 Server Password: Server Password: Server Password: Server Password: Switch Type: LUCENT DEFINITY ECS Diagnostic Settings Diagnostic Settings	
4.	Leave the Server Name 1 field blank.	
5.	Leave the Server Port 1 field blank.	
6.	Set the Server User ID to the user ID created for Audiolog in Avaya Computer Telephony Server (usually aluser1).	
7.	Set the Server Password to the password created for the user "aluser1".	
8.	Select TSAPI+ from the Link Type Protocol drop-down list.	
9.	Select LUCENT DEFINITY ECS from the Switch Type drop-down list.	
10.	Check the Monitor Link Startup checkbox.	
11.	Check the Monitor Link Poll checkbox.	
12.	Set Monitor Link Interval to 30 .	
13.	Set Monitor Link Retries to 1.	

Step	Description
14.	Set Heartbeat Interval to 90 . (In general, the Heartbeat Interval should be set to 3 times the Monitor Link Interval.)
15.	Click the Apply button.

6.6.2. Options/Devices Tab

Step	Description	
1.	On the Options/Devices tab, check both the Enable Step Conf boxes.	RAPI Support and Enable Single
	🖷 CTILink Configuration	×
	General Link Options / CT Server Serial	User Data TCP/IP Triggers
	Misc Options	Device Parameters
	Enable RAPI Support	Agent Size: 4
	Enable Single Step Conf	Extension Size: 3
	Allow Agent Enable Cntrl	Trunk Group Size: 3
	Observe Feature Prefix: w882	Trunk Number Size: 5
	Observe Feature Suffix:	Min Device ID: 0
		Max Device ID: 9999
	Enable Screen Record:	
	Stop Screen Record After: Wrap Delay	
	Wrap Delay: 20	SMDR Default Action
	Screen Record Eval Only: 🔽	⊙ Stop ⊂ Erase
	<u> </u>	
2.	Click the Apply button.	

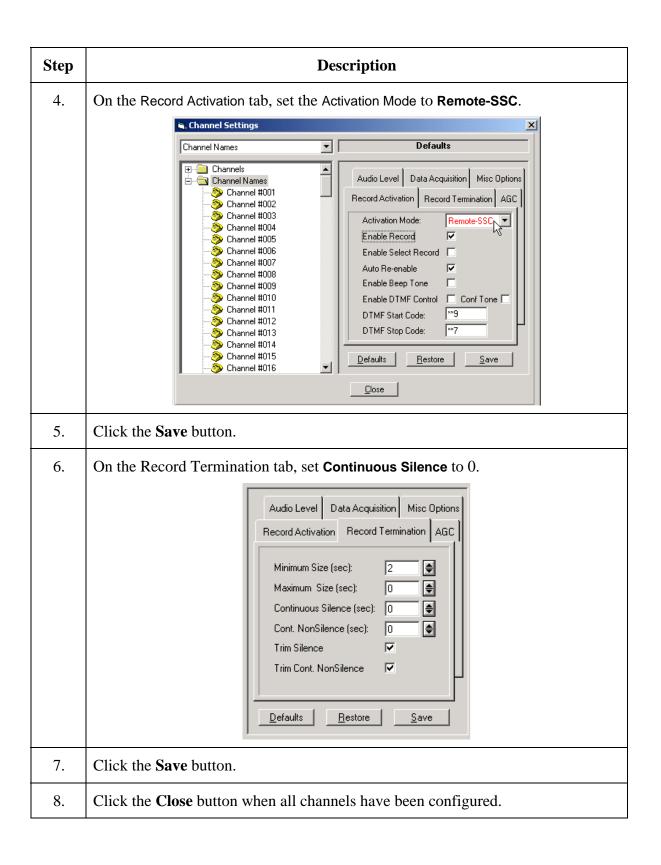
6.6.3. Avaya Computer Telephony Server Tab

Step	Description		
1.	On the Avaya Computer Telephony Server tab, check both the Enable Logon and Enable Logoff boxes.		
	🖹, CTILink Configuration 🔀		
	General Link Options / CT Server Serial User Data TCP/IP Triggers		
	CT-Connect Link TSAPI Logical ID: Enable Private Data: Network Type: ncacn_ip_tcp		
	Logon/Logoff Events Genesys Enable Logon: Image: Configuration Management Environment Enable Logoff: Image: Configuration Management Environment		
	CT Agent/Device Map Agent Data Index: 0 Agent ID Index: 0 Device ID Index: 0		
	<u>QK</u> <u>Cancel</u> <u>Apply</u>		
2.	Check the Enable Private Data checkbox.		
3.	Click the Apply button.		
4.	Click the OK button to close CTILink configuration.		

6.7. Channel Settings

Perform the following steps for each channel:

Step	Description
1.	If Configuration Manager is not already open, click the center button on the Manager module to access Configuration Manager .
	No Channel Name Date Time Duration FileName M1245 Manager 0.5k 0.5k
2.	Click the Recorder tab of Configuration Manager.
3.	Click the Channels button.



6.8. Restart Audiolog

After configuring the system, shutdown Audiolog and restart it. As soon as the CTILink module has started communicating with the Avaya Computer Telephony Server, the CTILink module should appear with the following:



7. Interoperability Compliance Testing

7.1. General Test Approach

The interoperability compliance test verified the ability of Audiolog to record calls. Call sequences included a call answer, transfer, consult transfer, conference, conference transfer, and blind transfer. The compliance test also encompassed a load test where a call generator made calls to the queue and each agent running Avaya Interaction Center along with an automate script tool to simulate agent responded to calls.

7.2. Test Results

MERCOM Audiolog passed all the compliance test cases.

8. Verification Steps

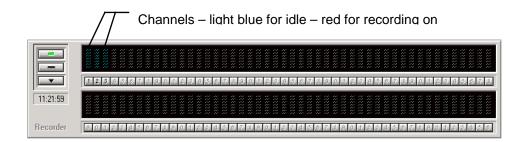
Since Audiolog is based on both hardware and software components, there are several steps involved to verify that both have been installed correctly:

- Ensure that channels can be monitored and recordings can be created
- Ensure that recordings can be accessed and played back from the Catalog
- Ensure that there is network communication between the server and client

8.1. Ensure Channels Can Be Monitored and Recorded

Unless changes are made to the **Appearance** of the **Recorder** module, the idle channels available for recording appear with light blue bars.

Make certain the number of available channels matches the number of channels ordered.



KB; Reviewed: SPOC 4/18/2005

Step	Description
1.	Place or receive a call on the channel. Verify the channel indicator turns red when the channel is active.
2.	Verify the correct channel is being recorded.
3.	Click the channel number below the active channel. You should be able to monitor the call.
4.	Disconnect the call. The channel indicator should return to idle (light blue). The channel should stop recording immediately upon disconnection. When the channel stops recording, a message should appear in the Manager module.

8.2. Ensure that Recordings Can Be Accessed from the Catalog

After created several recordings, verify that they can be found in the catalog and then played back.

Be sure to logged-on with **User Name** and **Password**. If a **Please Logon** screen appears when trying to start the Call Locator, enter **User Name** and **Password**, and click **OK**. Then start the Call Locator again.

To play back the recordings just created,

Step	Description
1.	On the server, to start the Call Locator, simply click on the Find button. This is the middle button at the left side of the Player module (you may also select Find from the Player's top button menu).
	If the Audiolog server has been recording channel audio, then there will be a \textcircled icon adjacent to the Catalog folder. The \textcircled icon indicates that there are entries (recordings) in the Catalog folder.
2.	Click on the icon to view the next level of the search tree.
3.	Find the recordings using the Date/Channel view.
	Verify that all of the appropriate channel names appear under the date. If you have created recordings on all of the recording channels, all of the channel names should appear under the date on which they were created.
4.	Click to select a channel name.

	At least one recording should appear in the right side of the Call Locator.
5.	Right-click the <i>recording</i> and click to select Playback .
	The recording plays back on the server.

9. Support

9.1. Mercom Support

Mercom provides Pre-sales and Technical Support for Audiolog.

• Technical Support: 201-507-8800 (Dial 5)

Tech.support@mercom.com

• Joe Flynn Director Support Operations 201-507-8800 x134 joe.flynn@mercom.com

10. Conclusion

This compliance test verified that Mercom Audiolog Recording server successfully integrated with Avaya Interaction Center.

11. Additional References

11.1. Documentation

For information on MERCOM's Audiolog Recording products refer to the following manuals.

- Audiolog Pro, Max-Pro and Ultra-Pro Installation Manual
- Audiolog User's Guide

Visit Mercom web site or contact Mercom Pre-Sales directly for more information.

- <u>www.mercom.com</u>
- Mercom Pre-Sales Department (presales@mercom.com)

©2005 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by [®] and are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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 Developer*Connection* Program at devconnect@avaya.com.