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 call-specific 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 stand-alone 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.

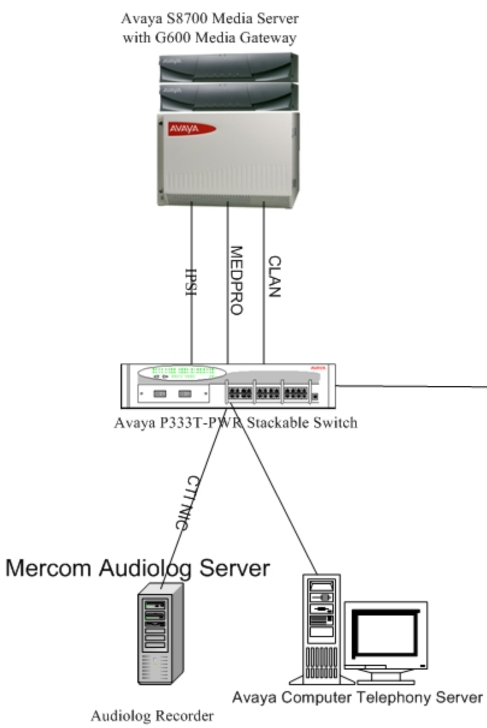


Figure 1: Avaya Developer*Connection* Compliance Test Configuration

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.	From the SAT terminal, type "display system-parameters customer-options" and go to page 3 to verify the options below are set to "**y**". Contact Avaya if these options 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-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? y
          ARS/AAR Dialing without FAC? n       Co-Res DEFINITY LAN Gateway? y
          ASAI Link Core Capabilities? y  Cvg Of Calls Redirected Off-net? n
          ASAI Link Plus Capabilities? y                       DCS (Basic)? y
       Async. Transfer Mode (ATM) PNC? n                 DCS Call Coverage? y
  Async. Transfer Mode (ATM) Trunking? n                DCS with Rerouting? y
              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.)
```

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`
	`CTI LINK`
	`CTI Link: 6`
	`Extension: 23999`
	`Type: ADJ-IP`
	`COR: 1`
	`Name: CT Server AIC LAB`

2.	Type "**change node-names ip**". Enter the name of the Avaya Computer Telephony server and its IP Address.
	`change node-names 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.	Type "**change ip-services**". Go to page 3. Enter the CTI link number created in step 4.2.1 for CTI Link, "**y**" for Enabled, enter Avaya Computer Telephony node name created in step 4.2.2 for Client Name, and enter **1** for Client Link.
	`change ip-services                                 Page   3 of   3`
	`DLG Administration`
	`CTI Link   Enabled   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:

Step	Description
1.	From the SAT terminal, type "**change COR 1**". Set Can be Service Observed and Can Be A Service Observer to "**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: none       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 Be Picked Up By Directed Call Pickup? n
                                    Can Use Directed Call Pickup? n
                                    Group Controlled Restriction: inactive
```

2.	From the SAT terminal, type "**add DS1 port ####**". Enter the name for this DS1 board, **hdb3** for Line Coding, **2.048** for Bit Rate, **CAS** for Signaling Mode, **pbx** for Interconnect, **mulaw** for Interface Companding. Make sure that the jumper setting on the TN464 board is set to 32 channels.

```
change ds1 1b04                                             Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01B04                           Name: MERCOM
            Bit Rate: 2.048                    Line Coding: hdb3

      Signaling Mode: CAS

        Interconnect: pbx                 Country Protocol: 1

Interface Companding: mulaw                            CRC? n
           Idle Code: 11111111
      Slip Detection? n                 Near-end CSU Type: other
```

3.	From the SAT terminal, type "**add station #####**". Enter **DS1DF** for Type, enter the port number and name.

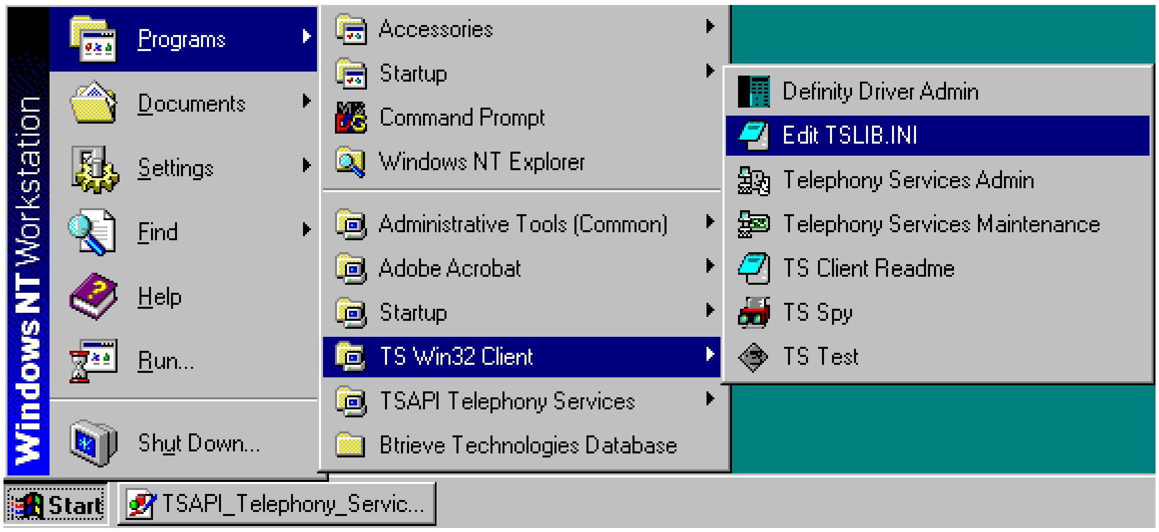
```
add station 27001                                        Page   1 of   3
                                     STATION

Extension: 27001                          Lock Messages? n          BCC: 0
     Type: DS1FD                           Security Code:            TN: 1
     Port: 01B0401                       Coverage Path 1:           COR: 1
     Name: MERCOM                        Coverage Path 2:           COS: 1
                                         Hunt-to Station:         Tests? y
STATION OPTIONS
               Loss Group: 4
    Off Premises Station? y
       R Balance Network? n
```

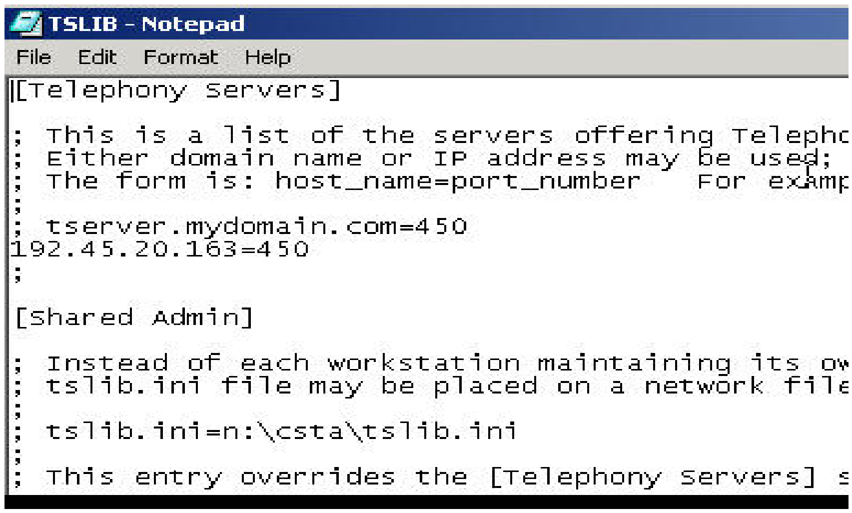
Step	Description

	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
	         Multimedia Mode: basic
	    MWI Served User Type:
	              AUDIX Name:
	                                             Coverage After Forwarding? s
	 Emergency Location Ext: 27001
	``

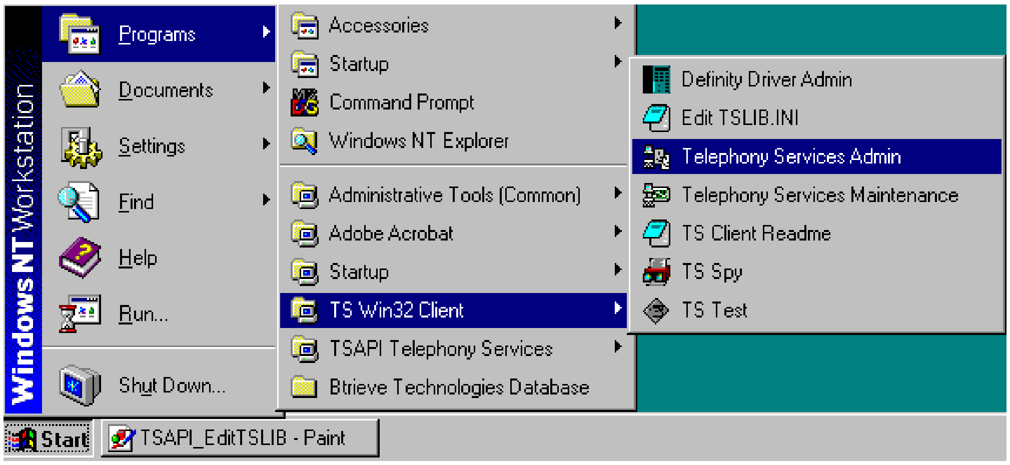
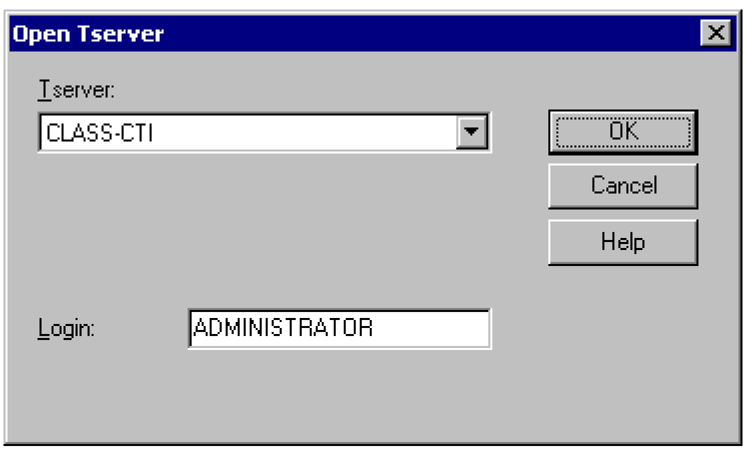
5. Configure the Avaya Computer Telephony Server

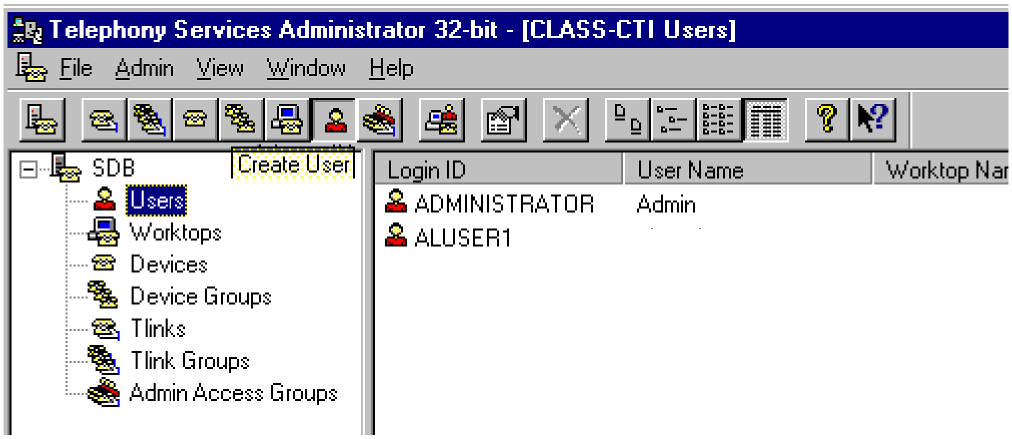
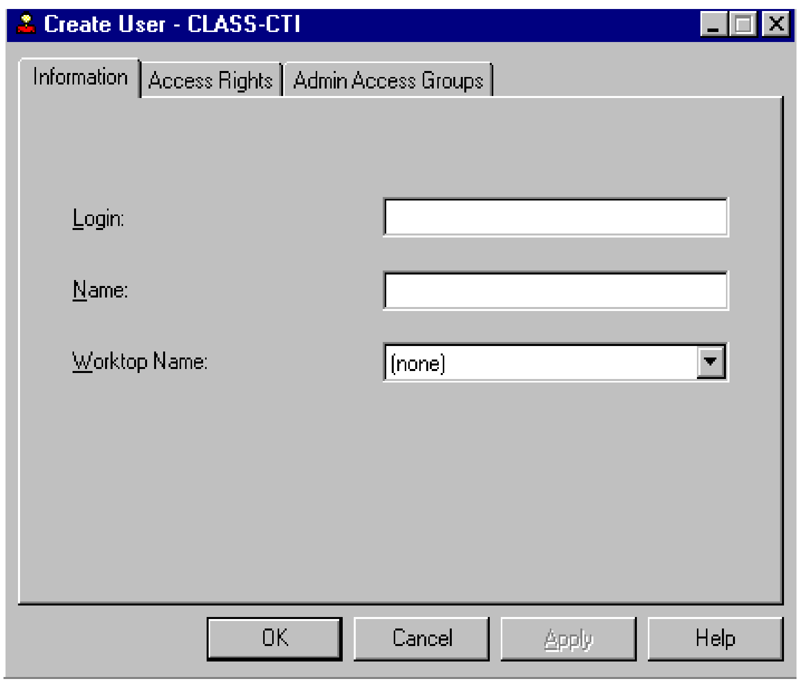
5.1. Modify the TSLIB.INI File

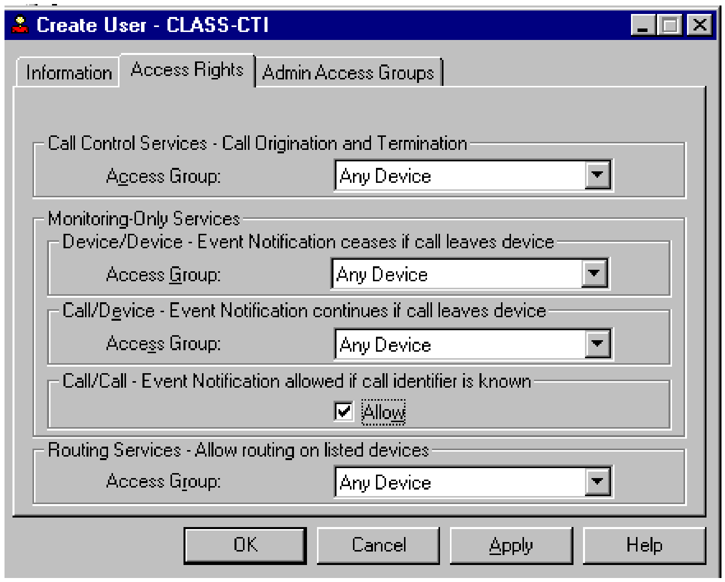
Step	Description
1.	On the computer running Avaya Computer Telephony Server, open the TSLIB.INI file by going to **Start → Programs → TS Win32 Client → Edit TSLIB.INI**. 

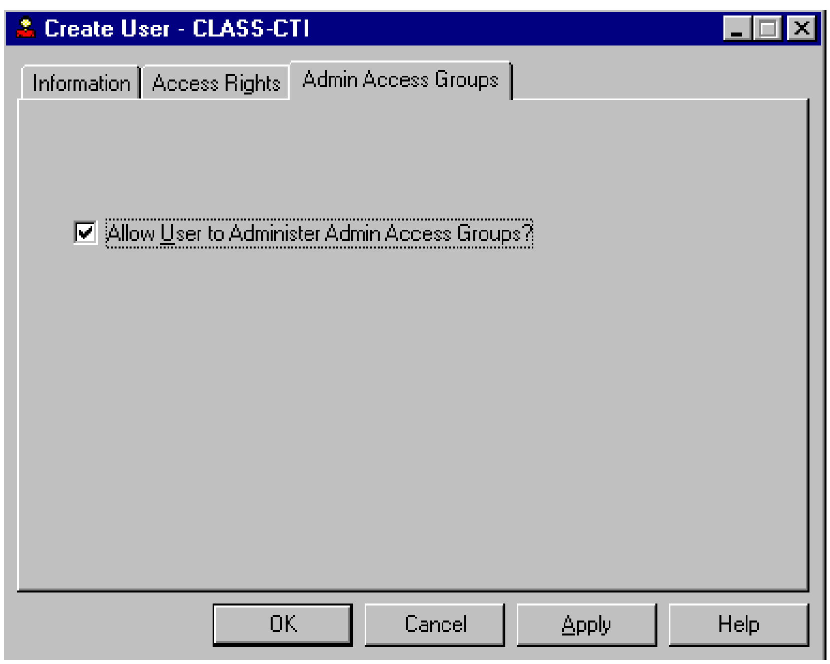
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>Change the IP address shown in the [Telephony Servers] section of the TSLIB.ini file to match the IP address (or Computer Name) of the Avaya Computer Telephony Server to which the Audiolog server will connect as a client. Audiolog 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.

</td></tr>
<tr><td>3.</td><td>Save and Exit the file.</td></tr>
<tr><td>4.</td><td>Copy the TSLIB.ini file to the C:\Winnt directory on the Audiolog.</td></tr>
</table>

5.2. Telephony Services Administration

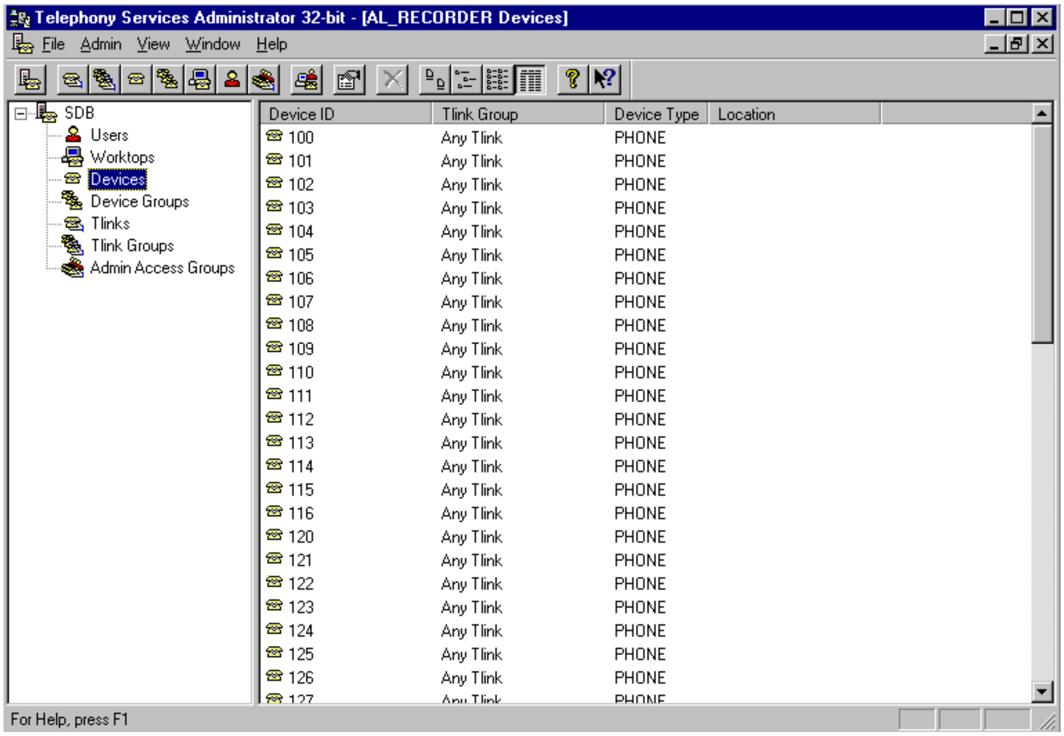
Step	Description
1.	On the Avaya Computer Telephony Server, open the **Telephony Services Admin** by going to **Start → Programs→ TS Win32 Client→ Telephony Services Admin**. 
2.	Select the computer name of the Avaya Computer Telephony Server (TSAPI Server) from the **Tserver** drop-down list.  NOTE**:** 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. 
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). 
6.	On the **Access Rights** tab, select "Any Device" for all sections.

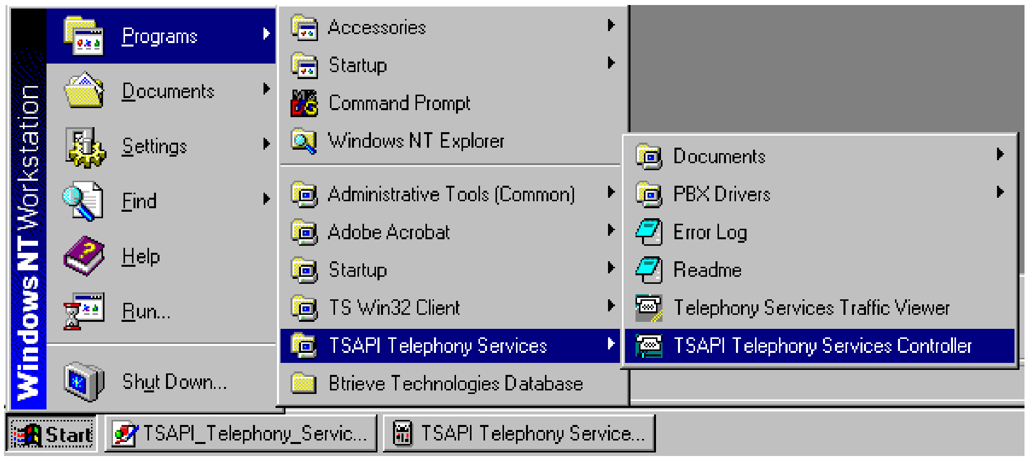
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>Check the Allow box under Call/Call section.


</td></tr>
<tr><td>8.</td><td>On the Admin Access Groups tab, check the Allow User to Administer Admin Access Groups checkbox.


</td></tr>
</table>

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. 

5.3. Telephony Services Controller

Step	Description
1.	Verify that the Telephony Service is running by going to **Start → Programs→ TSAPI Telephony Services → TSAPI Telephony Services Controller**. 
2.	When changes are made, click on the **Refresh** button to update. 

6. Configure the Audiolog Recorder

6.1. Requirements

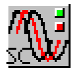
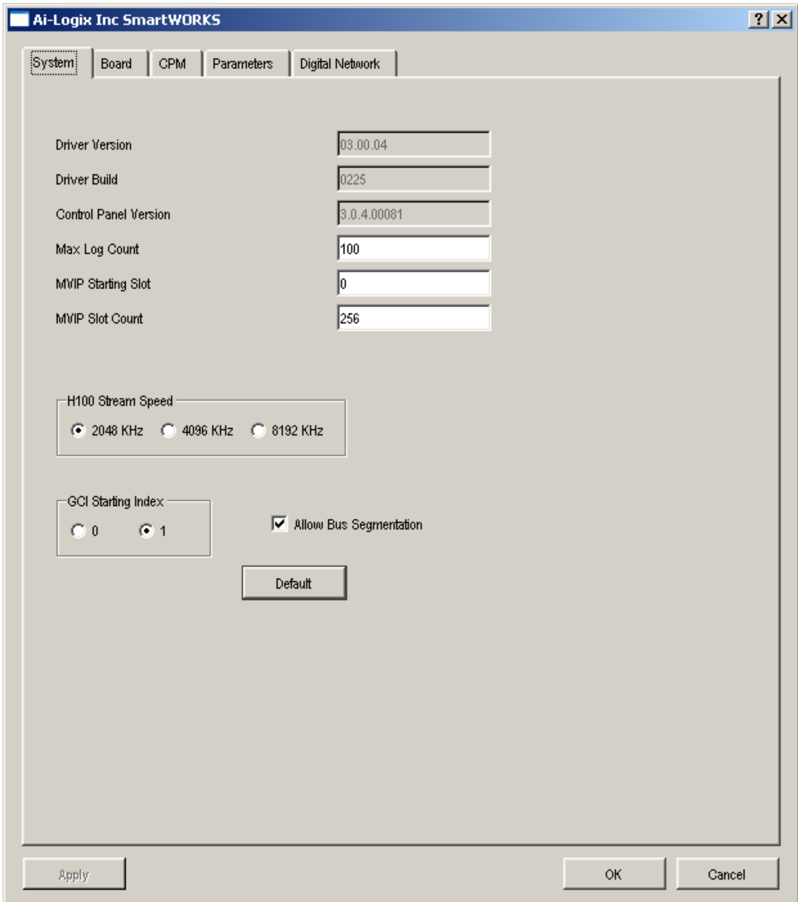
- Audiolog Version 2.70 or later
- Single or Dual Network Cards - Depending on the LAN Environment
- Audiolog CTILink Module option
- Audiolog TSAPI Avaya Definity G3 Integration Enabler option
- Audiolog Free-Seating option is required if Free Seating (Agent Logon/Logoff detection) is to be used
- Audiolog 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

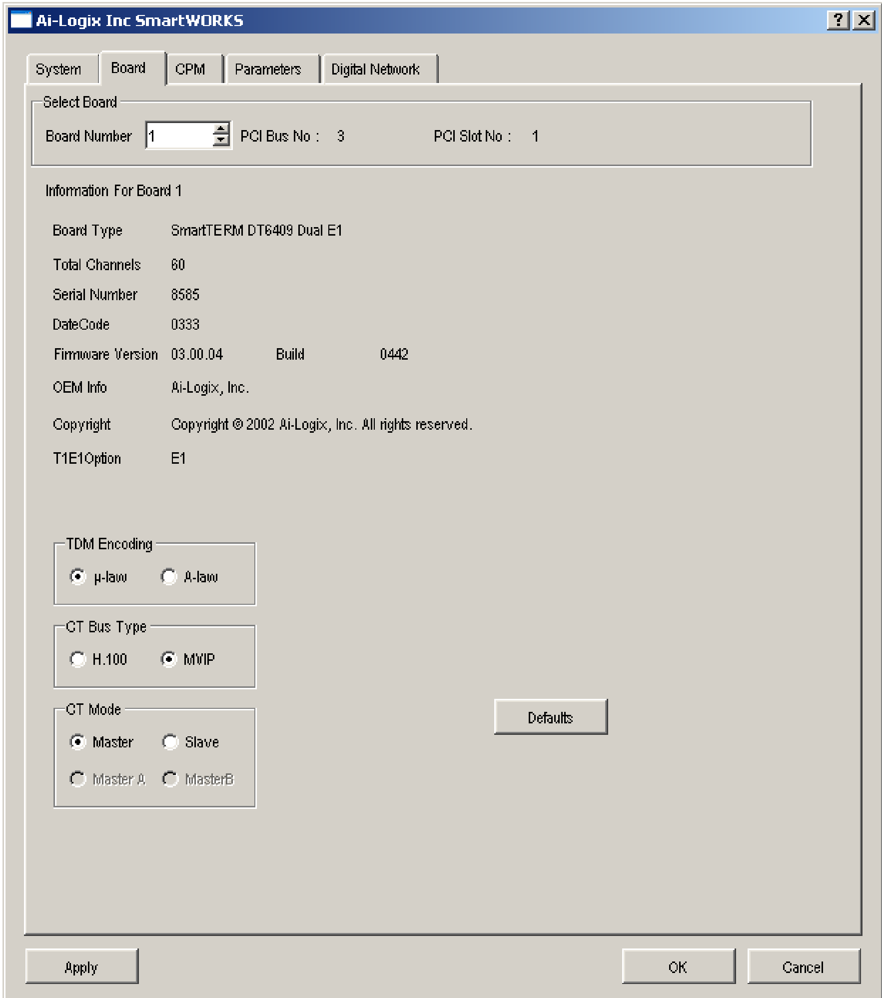
Connect the Audiolog recorder to the same LAN segment as the Avaya Computer Telephony Server.

6.3. Audiolog E1 Configuration

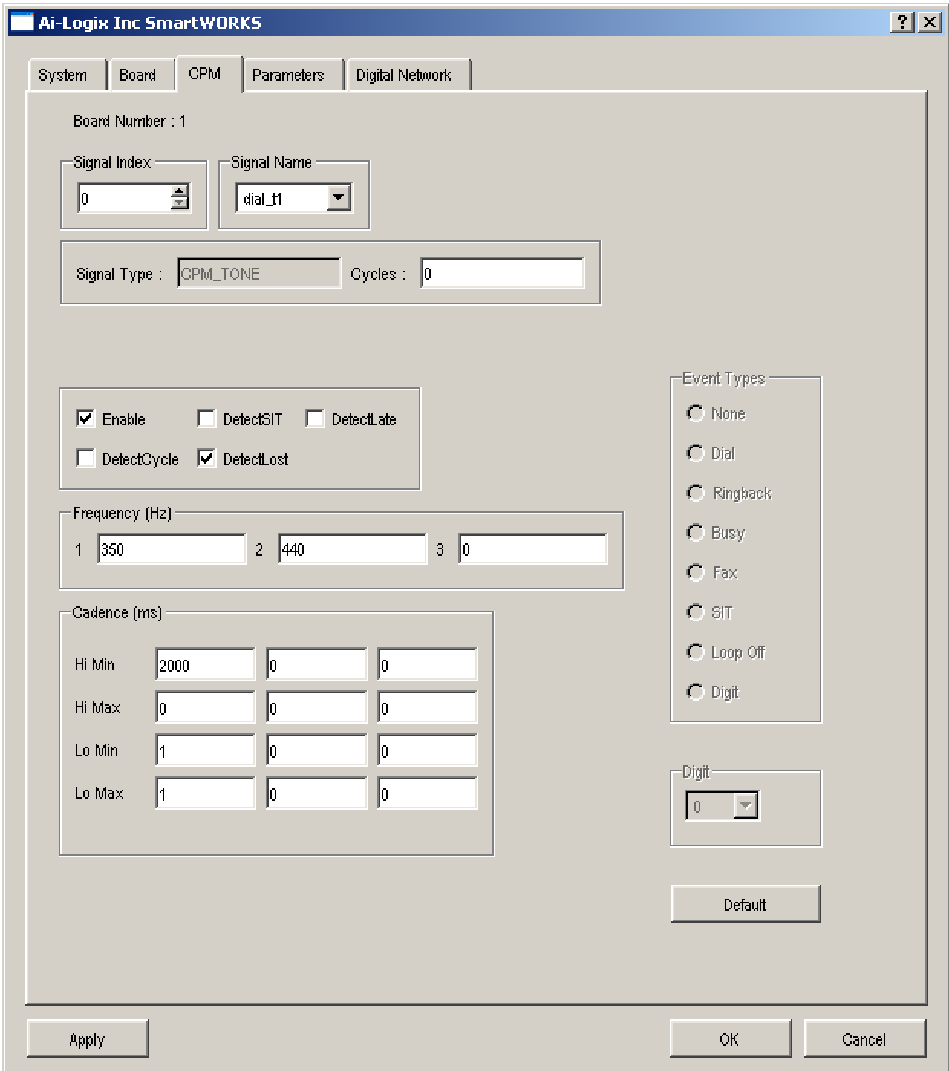
Step	Description
1.	Verify the settings on the PCI telephony interface cards. If there is no **Smart Control** icon on the desktop, from the **Start** button, click **Control Panel**.

Step	Description
2.	 Click the **SmartControl** SmartControl icon. The Ai-Logix SmartWORKS dialog box opens. 
3.	On the System tab, verify the **GCI Starting Index** is set to 1.
4.	Check the **Allow Bus Segmentation** box.

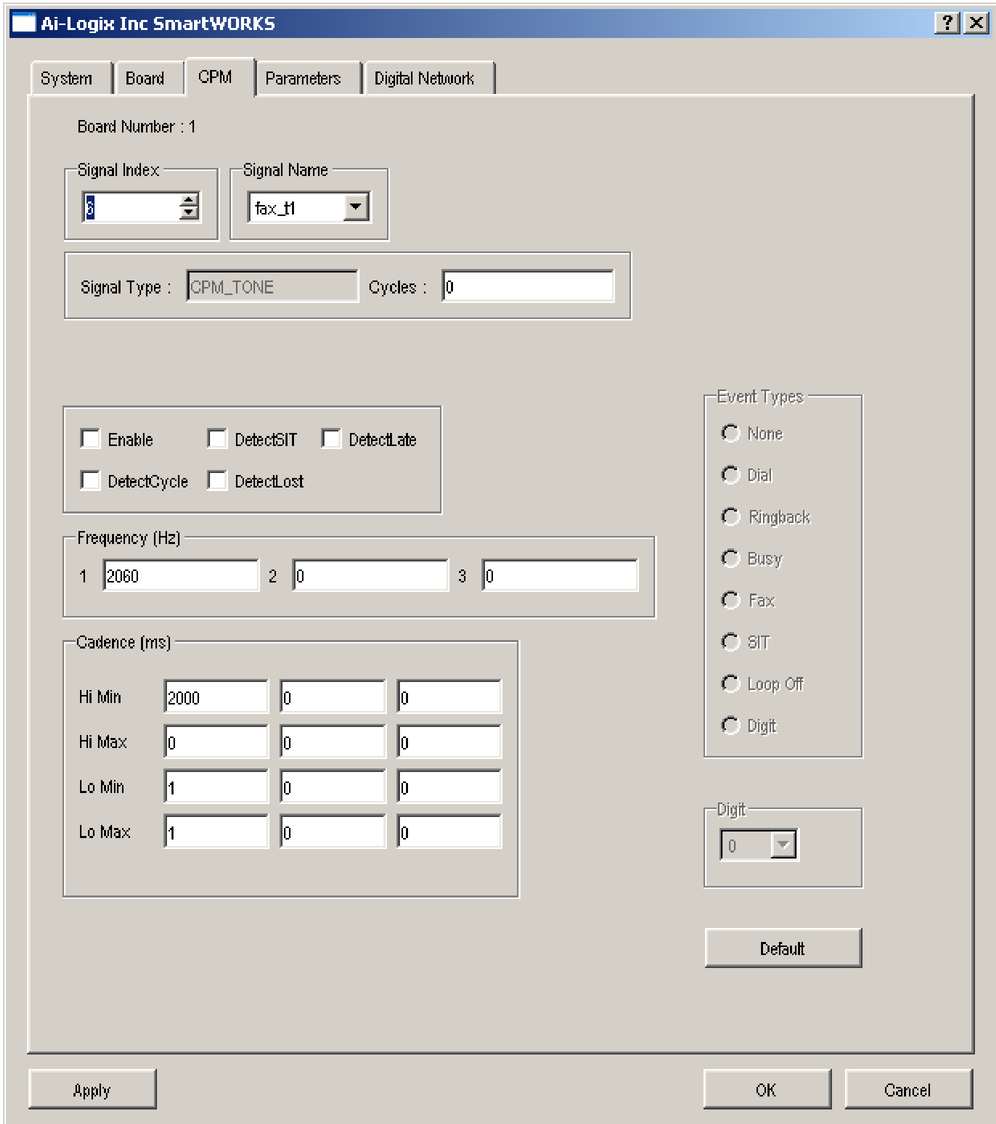
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>Click the Board tab.



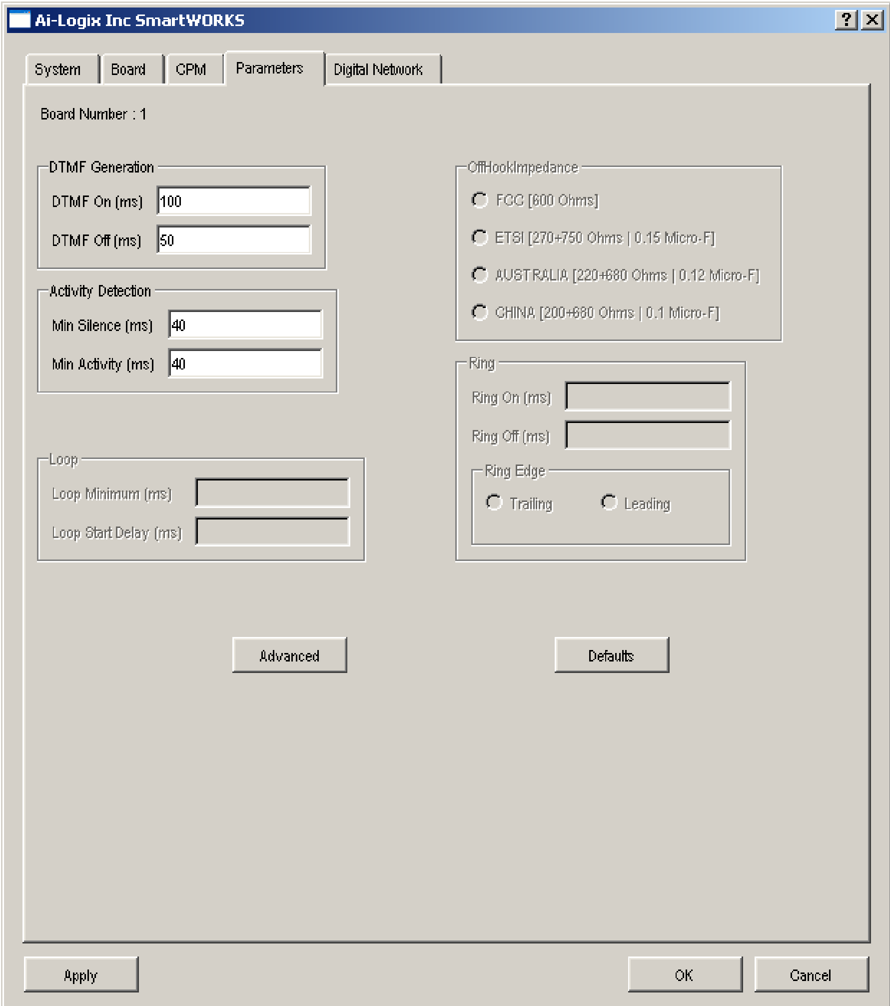
</td></tr>
<tr><td>6.</td><td>Select the Board Number of the T1/E1 card from the Board Number drop-down list.</td></tr>
<tr><td>7.</td><td>Verify the TDM Encoding is set to µ-law.</td></tr>
<tr><td>8.</td><td>Verify the Avaya Computer Telephony Bus Type is set to MVIP.</td></tr>
<tr><td>9.</td><td>Verify the Avaya Computer Telephony Mode is set to Master.</td></tr>
<tr><td>10.</td><td>Click the Apply button if you made any changes.</td></tr>
<tr><td>11.</td><td>Click the CPM tab.</td></tr>
</table>

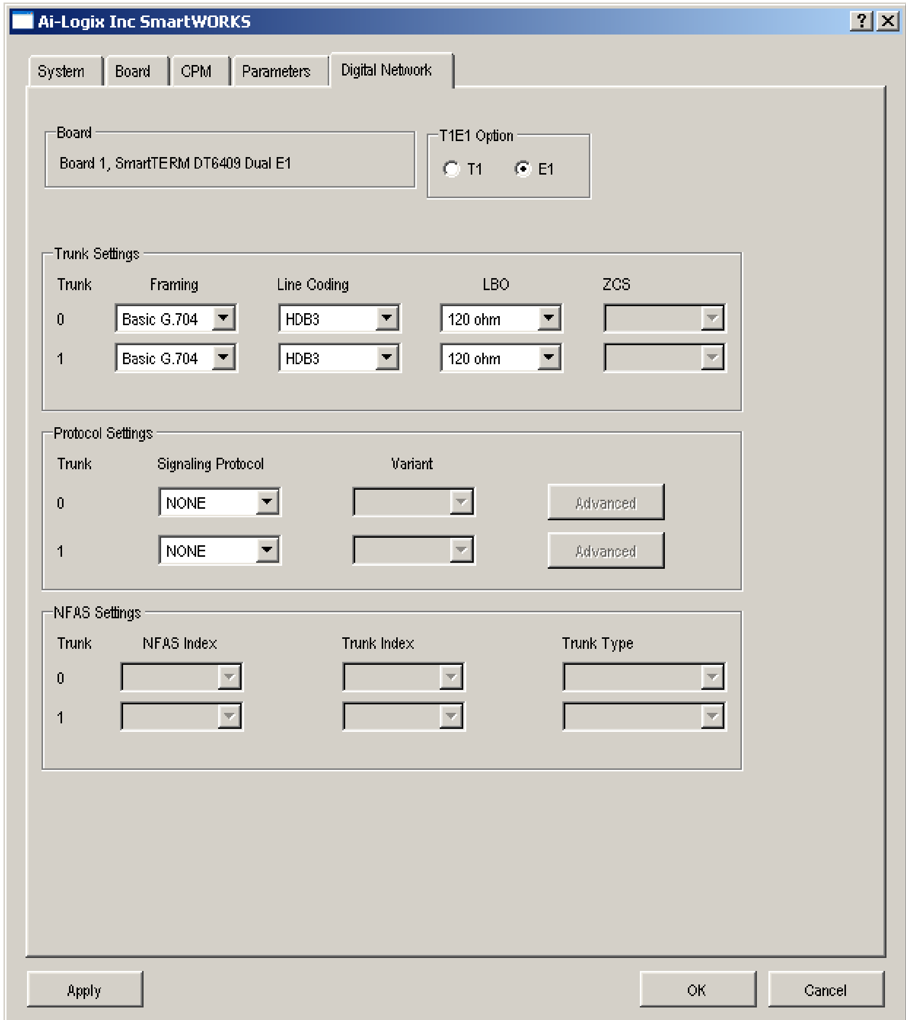
Step	Description
12.	Verify the settings shown in the following illustration. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>13.</td><td>Increment the Signal Index box until fax_t1 is displayed in the Signal Name box.



</td></tr>
<tr><td>14.</td><td>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.</td></tr>
</table>

Step	Description
15.	Click the **CPM** tab and verify the settings shown in the following illustration. 

Step	Description
16.	Click the **Digital Network** tab. 
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 **120OHM** 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

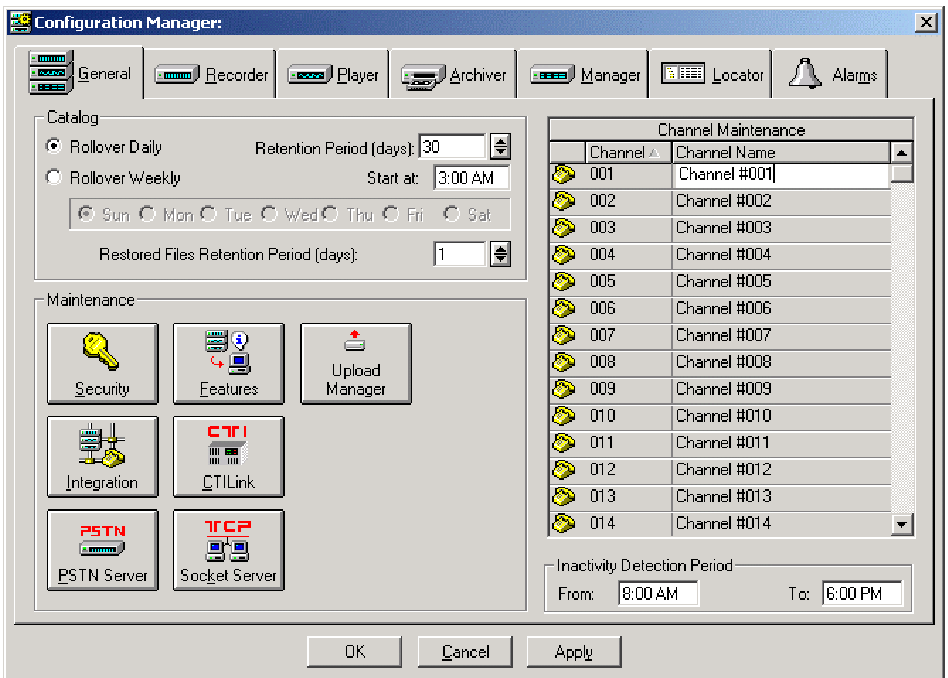
Caution: If you use Registry Editor incorrectly, you may cause serious problems that may require you to reinstall Audiolog and/or your operating system. Mercom cannot guarantee that you can solve problems that result from using Registry Editor incorrectly. Use the Registry Editor at your own risk.

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

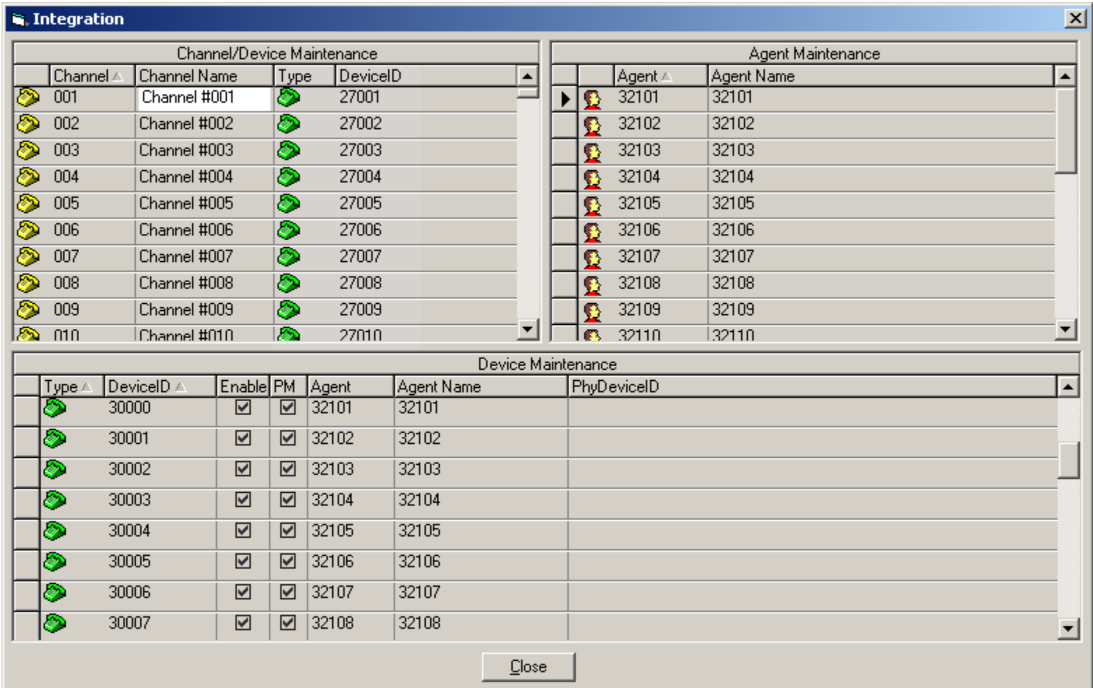
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Click the center button on the Manager module to access Configuration Manager.

</td></tr>
<tr><td>2.</td><td>On the General tab of Configuration Manager, click the Integration button.



</td></tr>
</table>

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.

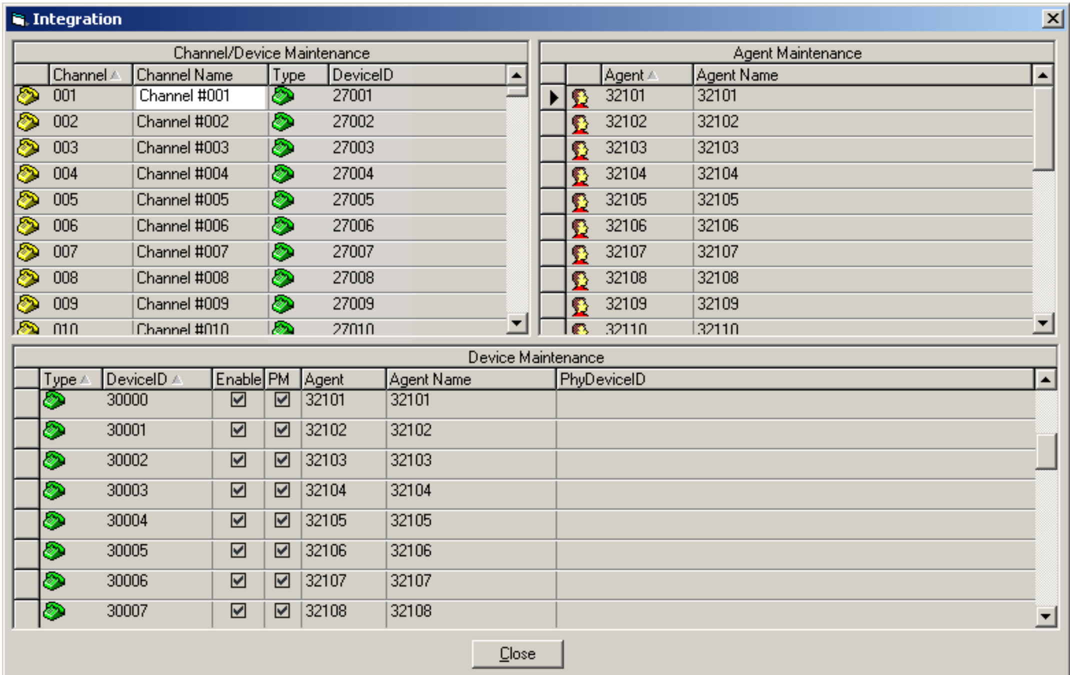


4. In the Device Maintenance table, click in the Type field until the (telephone) icon appears. This table is used to tell Audiolog which devices (stations, trunks, hunt groups, etc.) are to be monitored and/or recorded. The Device Maintenance Table is also used to designate the ACD Agent currently "seated" at a Station. CTI Integration also permits “free seating” where information in the CTI data stream allows agents to be dynamically associated with a device (telephone).

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.

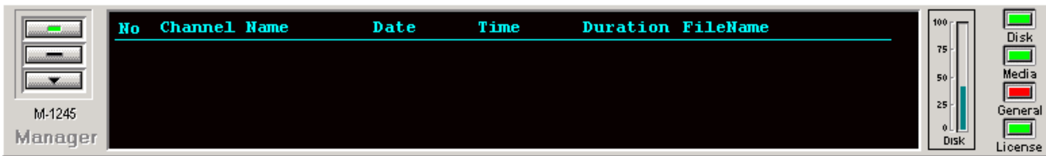
<table>
<tr><td>6.</td><td>Check both the Enable and PM checkboxes for each extension.


</td></tr>
<tr><td>7.</td><td>Click the Close button when finished.</td></tr>
</table>

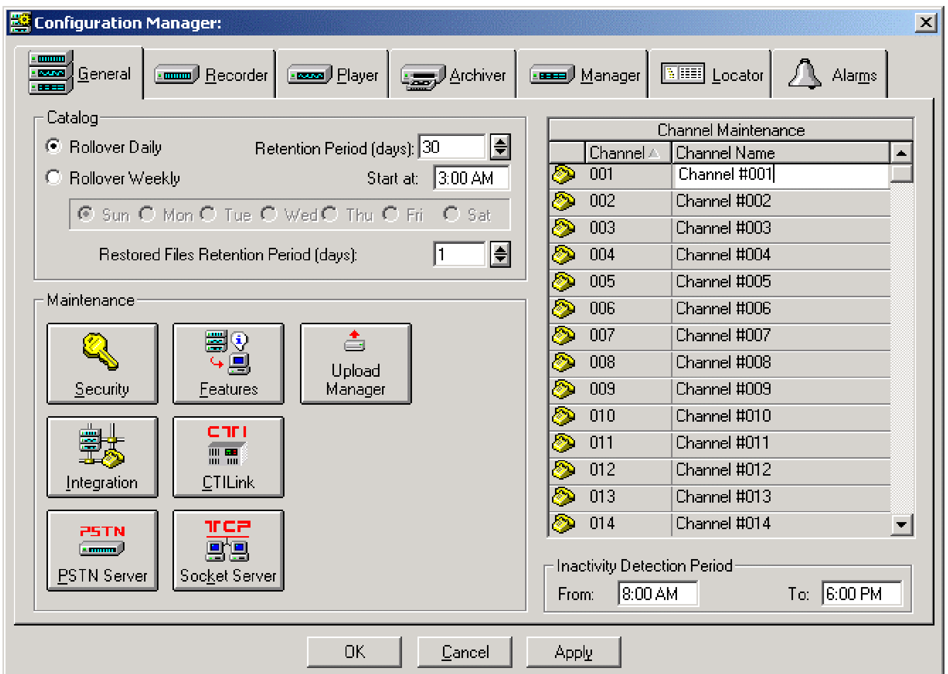
6.6. CTILink Configuration

6.6.1. General Link Tab

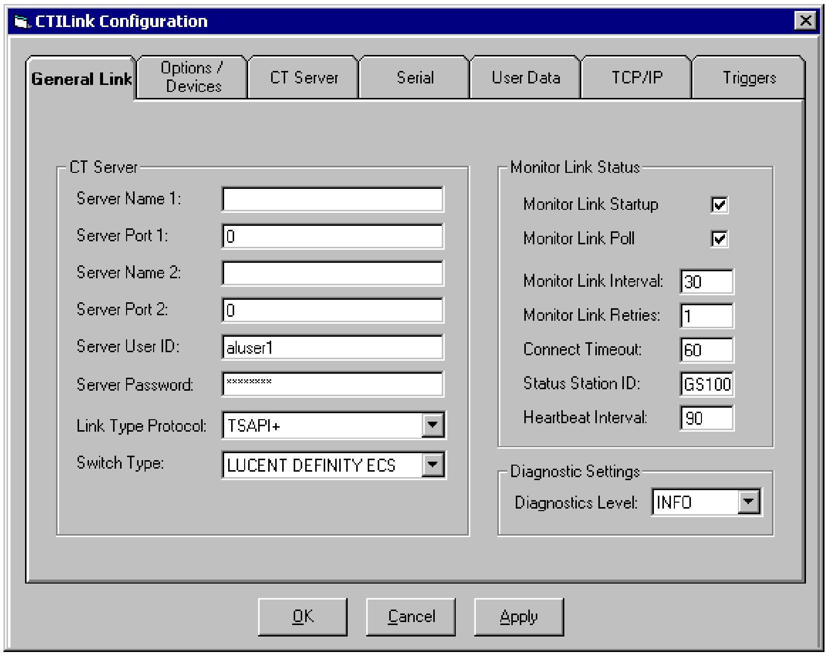
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>If Configuration Manager is not already open, click the center button on the Manager module to access Configuration Manager.



</td></tr>
<tr><td>2.</td><td>On the General tab of Configuration Manager, click the CTILink button.

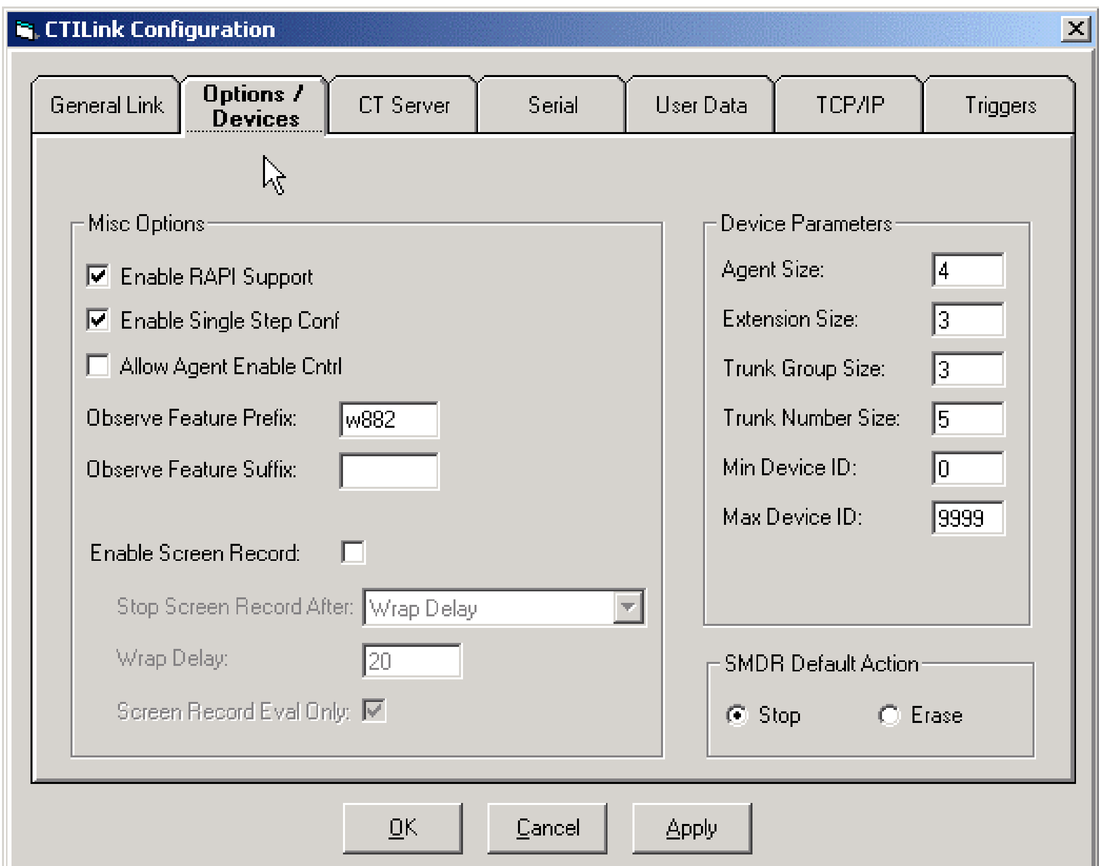


</td></tr>
</table>

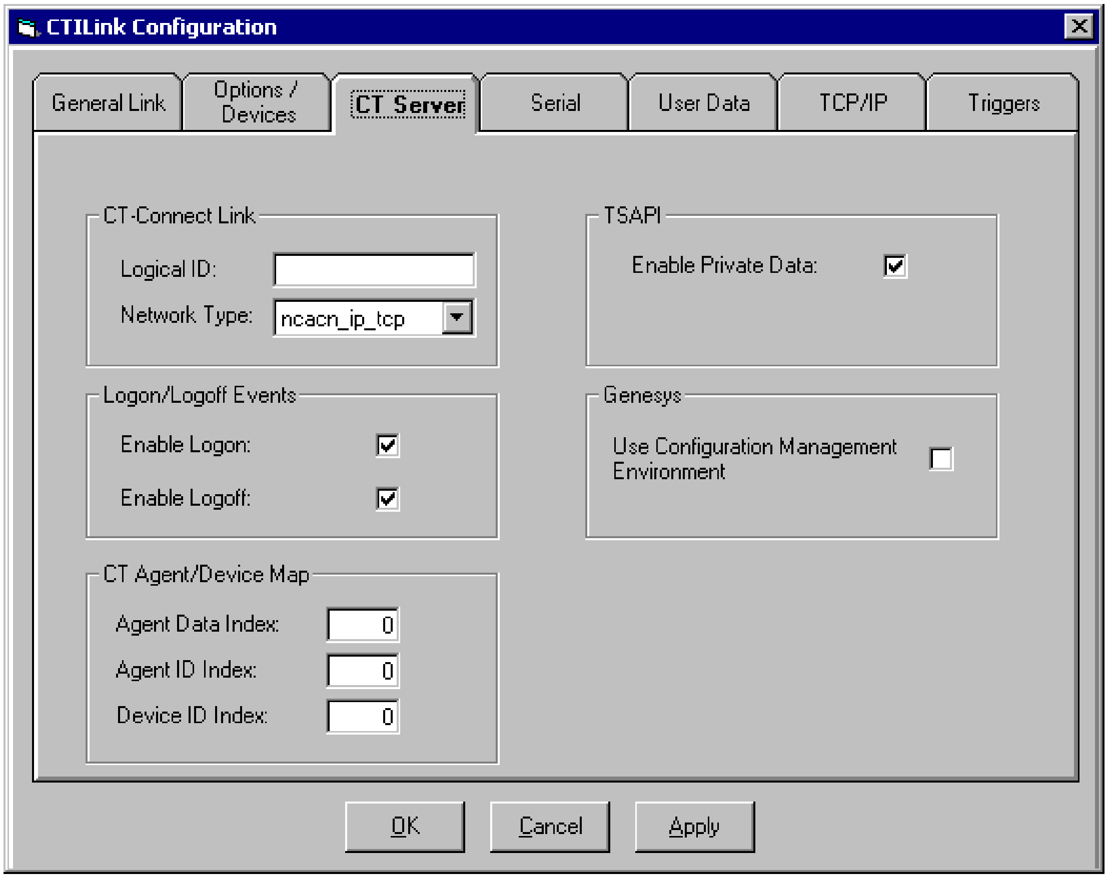
Step	Description
3.	Click the **General Link** tab. 
4.	Leave the Server Name 1 field blank.
5.	Leave the Server Port 1 field blank.
6.	Set the Server User ID to the user ID created for Audiolog in Avaya Computer Telephony Server (usually aluser1).
7.	Set the Server Password to the password created for 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

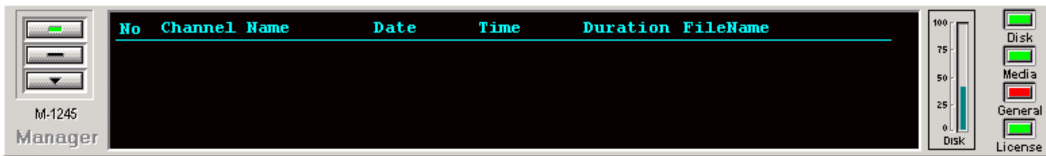
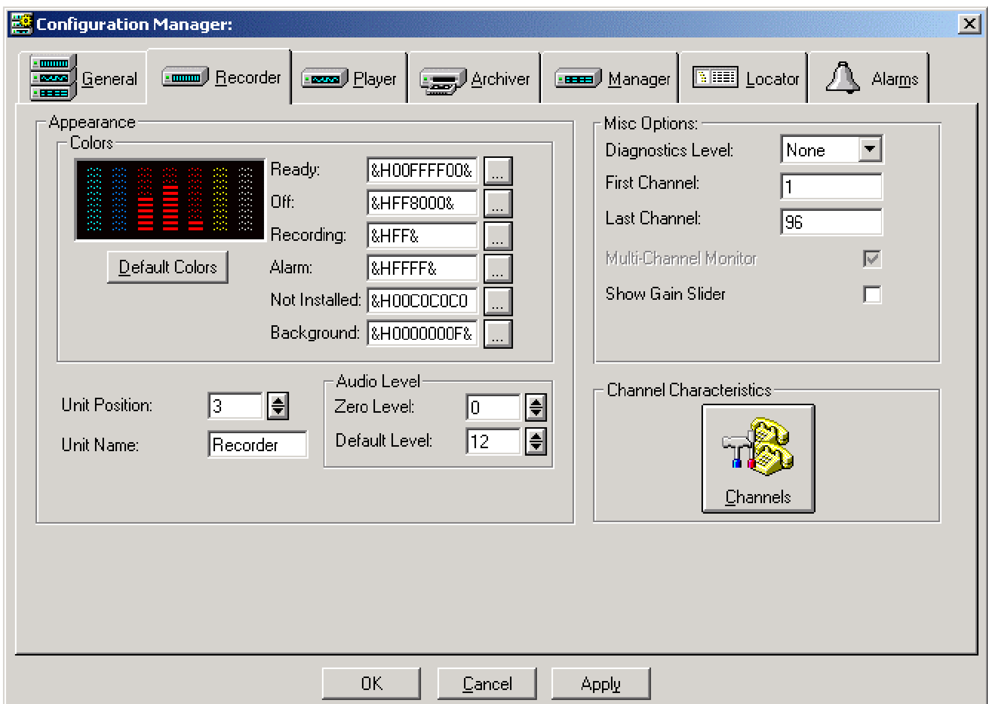
Step	Description
1.	On the **Options/Devices** tab, check both the **Enable RAPI Support** and **Enable Single Step Conf** boxes. 
2.	Click the **Apply** button.

6.6.3. Avaya Computer Telephony Server Tab

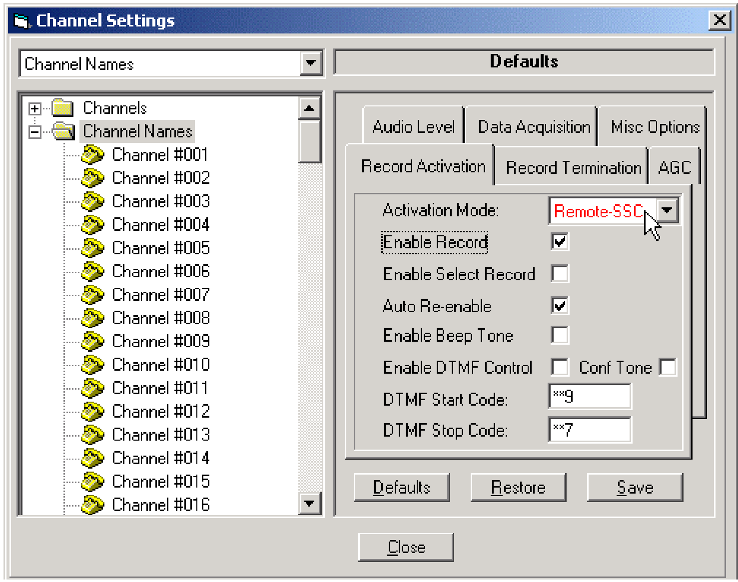
Step	Description
1.	On the Avaya Computer Telephony Server tab, check both the **Enable Logon** and **Enable Logoff** boxes. 
2.	Check the **Enable Private Data** checkbox.
3.	Click the **Apply** button.
4.	Click the **OK** button to close CTILink configuration.

6.7. Channel Settings

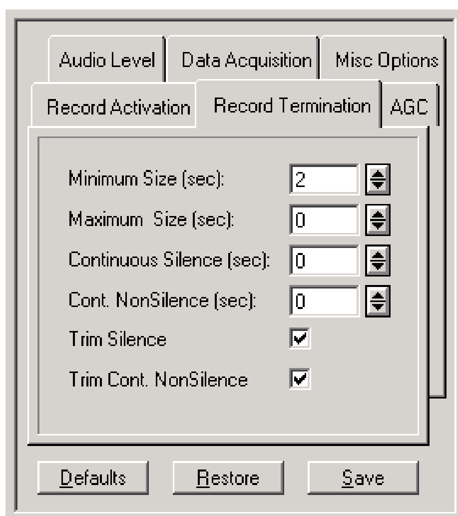
Perform the following steps for each channel:

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>If Configuration Manager is not already open, click the center button on the Manager module to access Configuration Manager.

</td></tr>
<tr><td>2.</td><td>Click the Recorder tab of Configuration Manager.

</td></tr>
<tr><td>3.</td><td>Click the Channels button.</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>On the Record Activation tab, set the Activation Mode to Remote-SSC.



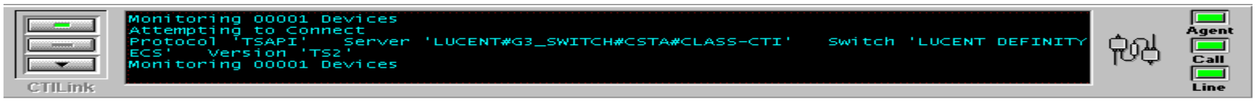
</td></tr>
<tr><td>5.</td><td>Click the Save button.</td></tr>
<tr><td>6.</td><td>On the Record Termination tab, set Continuous Silence to 0.



</td></tr>
<tr><td>7.</td><td>Click the Save button.</td></tr>
<tr><td>8.</td><td>Click the Close button when all channels have been configured.</td></tr>
</table>

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

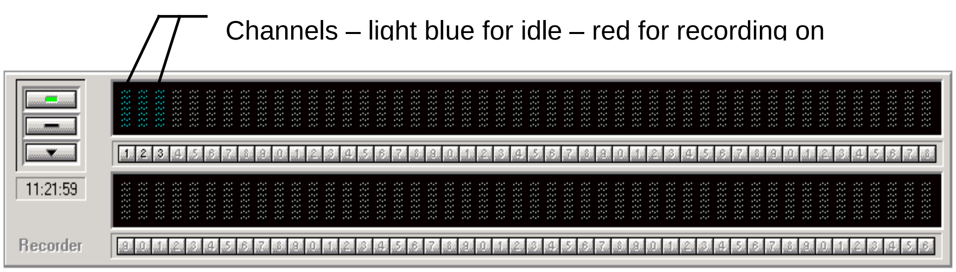
Since Audiolog is based on both hardware and software components, there are several steps involved to verify that both have been installed correctly:

- 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.



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 ⊞ icon adjacent to the Catalog folder. The⊞ 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 *recording* 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.

- www.mercom.com
- Mercom Pre-Sales Department (presales@mercom.com)

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.