Avaya Solution & Interoperability Test Lab

Application Notes for Micro-Tel MicroCall with Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the Micro-Tel MicroCall to successfully interoperate with Avaya Aura® Communication Manager.

Micro-Tel MicroCall is a call accounting software that interoperates with Avaya Aura® Communication Manager over the Avaya Reliable Session Protocol (RSP). Call records can be generated for various types of calls. Micro-Tel MicroCall collects, and processes the call records.

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

1. Introduction

The overall objective of this interoperability compliance testing is to verify that the Micro-Tel MicroCall call accounting software can interoperate with Avaya Aura® Communication Manager 6.2. Micro-Tel MicroCall (herein referred to as MicroCall) connects to Avaya Aura® Communication Manager over the local or wide area network using a CDR link running on RSP. Avaya Aura® Communication Manager is configured to send CDR records to MicroCall using a specific port.

MicroCall provides traditional call collection, rating, and reporting for any size businesses. MicroCall can interface with most telephone systems - in particular, with Avaya Aura® Communication Manager - to collect and interpret the detailed records of inbound, outbound, tandem, and internal telephone calls. MicroCall then calculates the appropriate charge for local, long distance, international & special calls and allocates them to responsible parties.

During the test, SIP endpoints were included. SIP endpoints registered with Avaya Aura® Session Manager. An assumption is made that Avaya Aura® Session Manager and Avaya Aura® System Manager are already installed and basic configuration have been performed.

Only steps relevant to this compliance test will be described in this document. In these Application Notes, the following topics will be described:

- Avaya Aura® Communication Manager – A SIP trunk configuration between Avaya Aura® Communication Manager and Avaya Aura® Session Manager. A CDR link configuration on Avaya Aura® Communication Manager.
- Avaya Aura® Session Manager – SIP trunk configuration between Avaya Aura® Communication Manager and Avaya Aura® Session Manager.
- MicroCall – A CDR link configuration on MicroCall.

2. General Test Approach and Test Results

The general test approach was to manually place intra-switch calls, inbound trunk and outbound trunk calls, transfer, conference, and verify that MicroCall collects the CDR records, and properly classifies and reports the attributes of the call.

For serviceability testing, physical and logical links were disabled/re-enabled, Avaya Servers were reset and MicroCall was restarted.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included features and serviceability tests. The focus of the compliance testing was primarily on verifying the interoperability between MicroCall and Communication Manager.

2.2. Test Results

All executed test cases passed. MicroCall successfully collected the CDR records from Communication Manager via a RSP connection for all types of calls generated including intra-switch calls, inbound/outbound PSTN trunk calls, inbound/outbound inter-switch calls over an H.323 trunk, transferred calls, and conference calls.

For serviceability testing, MicroCall was able to resume collecting CDR records after failure recovery including buffered CDR records for calls that were placed during the outages.

2.3. Support

Technical support for MicroCall can be obtained through the following:

- http://www.microcall.com
- (770) 447-5408

3. Reference Configuration

Figure 1 illustrates a call path consisting of an Avaya S8300D Server, an Avaya G450 Media Gateway and a Session Manager on one side, and Avaya S8720 Servers with an Avaya G650 Media Gateway and Avaya Aura ® SIP Enablement Services on the other side. Here, SIP Enablement Services was utilized only to register SIP endpoints in Location B.

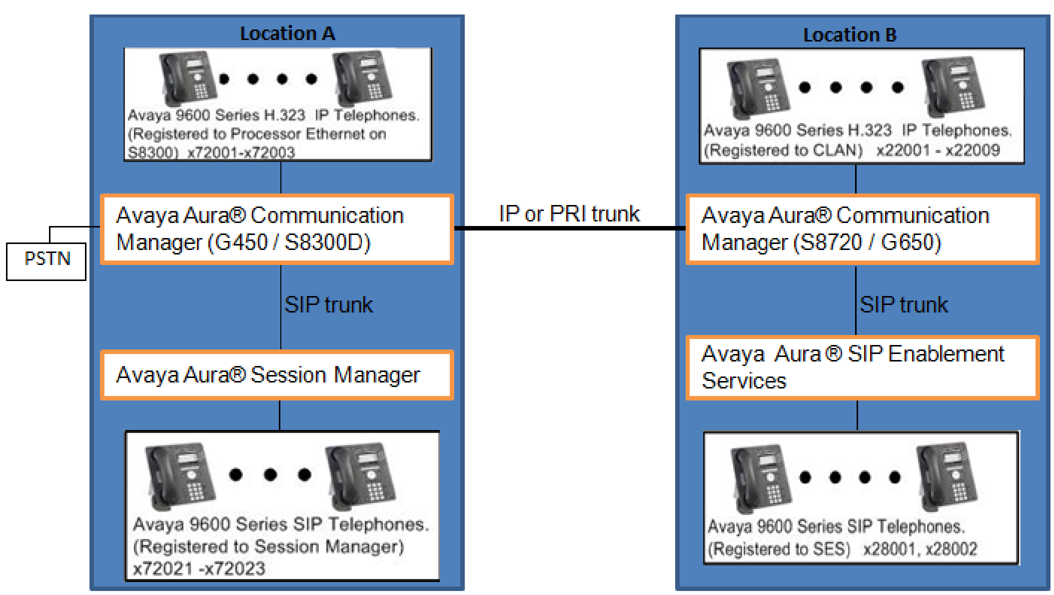


Figure 1. Call Path Configuration between two Locations

Figure 2 illustrates a configuration used during the compliance test. For completeness, Avaya 9600 Series SIP IP Telephones on the Avaya S8300D Server side have been registered to Session Manager. Avaya 9600 Series SIP IP Telephones on the Avaya S8720 Server side have been registered to Avaya Aura® SIP Enablement Services, and are included in Figure 1 to demonstrate calls between the SIP IP telephones that are going through the IP/PRI trunk between two Avaya Communication managers. The solution described herein is also extensible to other Avaya Servers and Media Gateways.

Note1: *SIP Enablement Services is not a part of this compliance test (only the SIP endpoints were utilized). Thus, there will not be any discussion on configuring SIP Enablement Services.*

Note2: *Avaya S8720 Servers with an Avaya G650 Media Gateway was included in the test only to provide an inter-switch scenario. Thus, there will not be any discussion on configuring Avaya S8720 Servers with an Avaya G650 Media Gateway.*

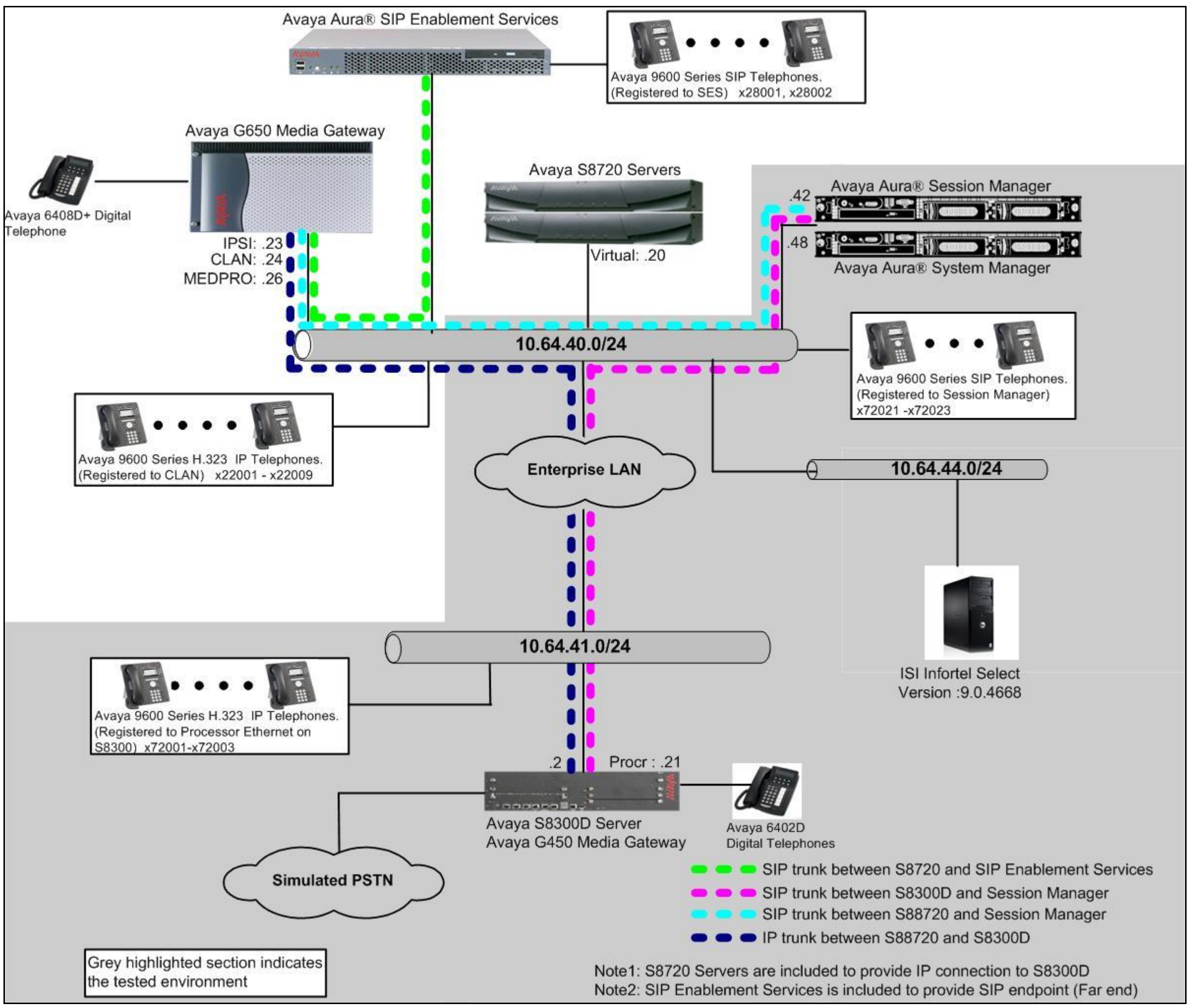


Figure 2. Test configuration of MicroCall with Avaya Aura® Communication Manager

4. Equipment and Software Validated

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

Equipment		Software
Avaya S8300D Server with Avaya G450 Media Gateway		Avaya Aura® Communication Manager 6.3 (R016x.03.0.124.0) with Patch 03.0.124.0-21291
Avaya Aura® System Manager		6.3.5.5.2017
Avaya Aura® Session Manager		6.3.5.0.635005
Avaya 9600 Series SIP IP Telephone		
	9620 9630	2.6.3 2.6.2
Avaya 9600 and 96X1 Series H.323 IP Telephone		
	9620 9621G 9650	3.1 6.22 3.22
MicroCall on Windows 2003 Server		5.40

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring call detail recording (CDR) in Communication Manager. These steps are performed through the System Access Terminal (SAT). These steps describe the procedure used for the Avaya S8300D Server. All steps are the same for the other Avaya Servers.

Communication Manager will be configured to generate CDR records using RSP over TCP/IP to the IP address of the PC running MicroCall. For the Avaya S8300D Media Server, the RSP link originates at the IP address of the local processor (with node-name - “procr”). For the Avaya S8720 Server, the SIP trunk terminates at the IP address of the CLAN board.

5.1. Configure CDR

Use the **change node-names ip** command to create a new node name, for example, **microcall**. This node name is associated with the IP Address of the PC running the MicroCall application. Also, take note of the node name – "procr". It will be used in the next step. The "procr" entry on this form was previously administered.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
microcall             10.64.43.249
default               0.0.0.0
procr                 10.64.41.21
procr6                ::
rdtt-1                10.64.40.14
SM-1                  10.64.41.42
```

Use the **change ip-services** command to define the CDR link to use the RSP over TCP/IP. To define a primary CDR link, provide the following information:

- **Service Type**: "CDR1" [If needed, a secondary link can be defined by setting Service Type to CDR2.]
- **Local Node**: "procr" [For the Avaya S8720 Servers set the Local Node to the node name of the CLAN board.]
- **Local Port**: "0" [The Local Port is fixed to 0 because Communication Manager initiates the CDR link.]
- **Remote Node**: "microcall" [The Remote Node is set to the node name previously defined.]
- **Remote Port**: "9000" [The Remote Port may be set to a value between 5000 and 64500 inclusive, and must match the port configured in MicroCall.]

```
change ip-services                                              Page   1 of   4

                                   IP SERVICES
 Service      Enabled      Local          Local         Remote       Remote
  Type                     Node           Port          Node         Port
AESVCS         y       procr              8765
CDR1                   procr              0         microcall        9000
CDR2                   procr              0         rdtt-1           9001
```

On **Page 3** of the ip-services form, enable the Reliable Session Protocol (RSP) for the CDR link by setting the **Reliable Protocol** field to "y".

```
change ip-services                                              Page   3 of   4

                              SESSION LAYER TIMERS
  Service      Reliable  Packet Resp   Session Connect  SPDU  Connectivity
   Type        Protocol    Timer        Message Cntr    Cntr     Timer

 CDR1             y         30               3            3        60
 CDR2             y         30               3            3        60
```

Enter the **change system-parameters cdr** command from the SAT to set the parameters for the type of calls to track and the format of the CDR data. The example below shows the settings used during the compliance test. Provide the following information:

- **CDR Date Format**: "month/day"
- **Primary Output Format**: "expanded"
- **Primary Output Endpoint**: "CDR1"

The remaining parameters define the type of calls that will be recorded and what data will be included in the record. See reference [2] for a full explanation of each field. The test configuration used some of the more common fields described below.

- **Use Legacy CDR Formats?**: "n" [Allows CDR formats to use 4.x CDR formats. If the field is set to "y", then CDR formats utilize the 3.x CDR formats.]
- **Intra-switch CDR**: "y" [Allows call records for internal calls involving specific stations. Those stations must be specified in the INTRA-SWITCH CDR form.]
- **Record Outgoing Calls Only?**: "n" [Allows incoming trunk calls to appear in the CDR records along with the outgoing trunk calls.]
- **Outg Trk Call Splitting?**: "y" [Allows a separate call record for any portion of an outgoing call that is transferred or conferenced.]
- **Inc Trk Call Splitting?**: "y" [Allows a separate call record for any portion of an incoming call that is transferred or conferenced.]
- **Call Account Code Length:** "6" [The length may be set to a value between 1 and 15. However, during the compliance test, "6" was used.]

```
change system-parameters cdr                                    Page   1 of   1
                            CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID): 1                        CDR Date Format: month/day
      Primary Output Format: expanded       Primary Output Endpoint: CDR1
    Secondary Output Format: unformatted Secondary Output Endpoint: CDR2
           Use ISDN Layouts? n                    Enable CDR Storage on Disk? y
       Use Enhanced Formats? n      Condition Code 'T' For Redirected Calls? n
      Use Legacy CDR Formats? n                   Remove # From Called Number? n
Modified Circuit ID Display? n                                Intra-switch CDR? y
                  Record Outgoing Calls Only? n      Outg Trk Call Splitting? y
  Suppress CDR for Ineffective Call Attempts? n        Outg Attd Call Record? n
      Disconnect Information in Place of FRL? n      Interworking Feat-flag? n
 Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
                                    Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
Record Agent ID on Incoming? y       Record Agent ID on Outgoing? y
     Inc Trk Call Splitting? y                     Inc Attd Call Record? n
  Record Non-Call-Assoc TSC? n            Call Record Handling Option: warning
      Record Call-Assoc TSC? n   Digits to Record for Outgoing Calls: dialed
   Privacy - Digits to Hide: 0               CDR Account Code Length: 6
```

If the **Intra-switch CDR** field is set to "y" on **Page 1** of the **system-parameters cdr** form, then use the **change intra-switch-cdr** command to define the extensions that will be subject to call detail records. In the Assigned Members field, enter the specific extensions whose usage will be tracked.

Note3: *To simplify the process of adding multiple extensions in the Assigned Members field, the* ***Intra-switch CDR by COS (SA8202)*** *feature may be utilized in the SPECIAL APPLICATIONS form under the system-parameters section. To utilize this feature, contact an authorized Avaya account representative to obtain the license.*

```
change intra-switch-cdr                                         Page   1 of   3
                           INTRA-SWITCH CDR

                                Assigned Members:   9    of 1000   administered
   Extension              Extension             Extension            Extension
   72001
   72002
   72003
```

5.2. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and Session Manager. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- **Authoritative Domain** – Enter the appropriate name for the Authoritative Domain. Set to the appropriate domain. During the compliance test, the authoritative domain is set to "avaya.com".
- **Codec Set** – Set the codec set number as provisioned in the **IP Codec Set** form.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:           Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                   RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.3. Configure IP Node Name

This section describes the steps for setting IP node name for Session Manager in Communication Manager. Enter the **change node-names ip** command, and add a node name for **SM-1** (Session Manager) along with its IP address.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
MicroCall               10.64.43.249
default                 0.0.0.0
procr                   10.64.41.21
procr6                  ::
rdtt                    10.64.40.14
SM-1                    10.64.41.42
```

5.4. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for signaling between Communication Manager and Session Manager. Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following:

- **Group Type** – Set to "sip".
- **Transport Method** – Set to "tls".
- **Near-end Node Name** - Set to "procr" as displayed in **Section 5.3**.
- **Far-end Node Name** - Set to the "SM-1" configured in **Section 5.3**.
- **Far-end Network Region** - Set to the region configured in **Section 5.2**.
- **Far-end Domain** - Set to "avaya.com".
- **Direct IP-IP-Audio Connections:** Set to "y"

```
add signaling-group 92                                            Page   1 of   2
                                SIGNALING GROUP

 Group Number: 92             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n          Priority Video? y        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: SM-1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain:avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

5.5. Configure SIP Trunk

This section describes the steps for administering a trunk group in Communication Manager for trunking between Communication Manager and Session Manager. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group and configure the following:

- **Group Type** – Set the Group Type field to "sip".
- **Group Name** – Enter a descriptive name.
- **TAC** (Trunk Access Code) – Set to any available trunk access code.
- **Signaling Group** – Set to the Group Number field value configured in **Section 5.4**.
- **Number of Members –** Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

```
add trunk-group 92                                                  Page   1 of  21
                                TRUNK GROUP

Group Number: 92                  Group Type: sip           CDR Reports: y
  Group Name: SM_41_42                   COR: 1        TN: 1         TAC: 1092
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 92
                                                    Number of Members: 10
```

5.6. Configure Uniform Dial Plan

This section describes the steps for administering a uniform dial plan in Communication Manager. Enter **change uniform-dialplan <u>**, where **u** is the uniform-dialplan number. The following screen shows the Uniform Dial Plan configuration. The 5-digit extension range starting with 2xxxx was used for the Avaya S8720 Servers side IP/SIP telephones, and utilized Automatic Alternate Routing (AAR).

```
change uniform-dialplan 2                                              Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                     Percent Full: 0

  Matching                           Insert                  Node
  Pattern           Len Del          Digits           Net Conv Num
 2                   5   0                            aar   n
```

5.7. Configure Automatic Alternate Routing

Enter **change aar analysis <a>**, where **a** is the AAR number. Automatic Alternate Routing (AAR) was used to route calls to the appropriate route pattern. The 5-digit extension range starting with 2 was used the route pattern 10. 2xxxx extensions are H.323 IP and SIP phones in S8720, and 28xxx extensions are SIP IP phones in S8720/SIP Enablement Services. To call these H.323 IP and SIP phones from S8300D Server, utilizes the route pattern 10 which is an ISDN/PRI or IP trunk.

```
change aar analysis 2                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 3

          Dialed            Total     Route     Call    Node   ANI
          String           Min  Max  Pattern    Type    Num    Reqd
    20004                   5    5     91       unku           n
    2                       5    5     10       aar            n
    33                      5    5     91       unku           n
    415                     10   10    92       aar            n
    50000                   5    5     92       unku           n
    53005                   5    5     91       unku           n
```

5.8. Configure Route Pattern

Enter **change route-pattern <r>**, where **r** is the route-pattern number. The route pattern 10 routes calls to the trunk group 10, which is either the IP or PRI trunk to S8720.

```
change route-pattern 10                                         Page   1 of   3
                    Pattern Number: 10     Pattern Name: To8720
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 10   0                                                               n   user
 2:                                                                      n   user
 3:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n             rest                                         none
 2: y y y y y n  n             rest                                         none
 3: y y y y y n  n             rest                                         none
```

5.9. Configure Off-PBX-Telephone Configuration-Set

SIP endpoints and off-pbx-telephone stations will be automatically created in Communication Manager when users (SIP endpoints) were created in Session Manager.

However, the **off-pbx-telephone configuration-set** form needs to be modified. Enter **change off-pbx-telephone configuration-set 1**. Set the **CDR for Origination** field to “none”, and disable the **CDR for Calls to EC500 Destination?** field by setting it to “n”.

```
change off-pbx-telephone configuration-set 1                     Page   1 of   1



                                     CONFIGURATION SET: 1

                         Configuration Set Description:
                                  Calling Number Style: network
                                   CDR for Origination: none
                     CDR for Calls to EC500 Destination? n
                           Fast Connect on Origination? n
                          Post Connect Dialing Options: dtmf
                         Cellular Voice Mail Detection: timed  (seconds): 4
                                         Barge-in Tone? n
                           Calling Number Verification? y
             Call Appearance Selection for Origination: primary-first
                                      Confirmed Answer? n

 Use Shared Voice Connections for Second Call Answered? n
Use Shared Voice Connections for Second Call Initiated? n
               Provide Forced Local Ringback for EC500? n
                        Apply Ringback upon Receipt of: Call-Proceeding
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

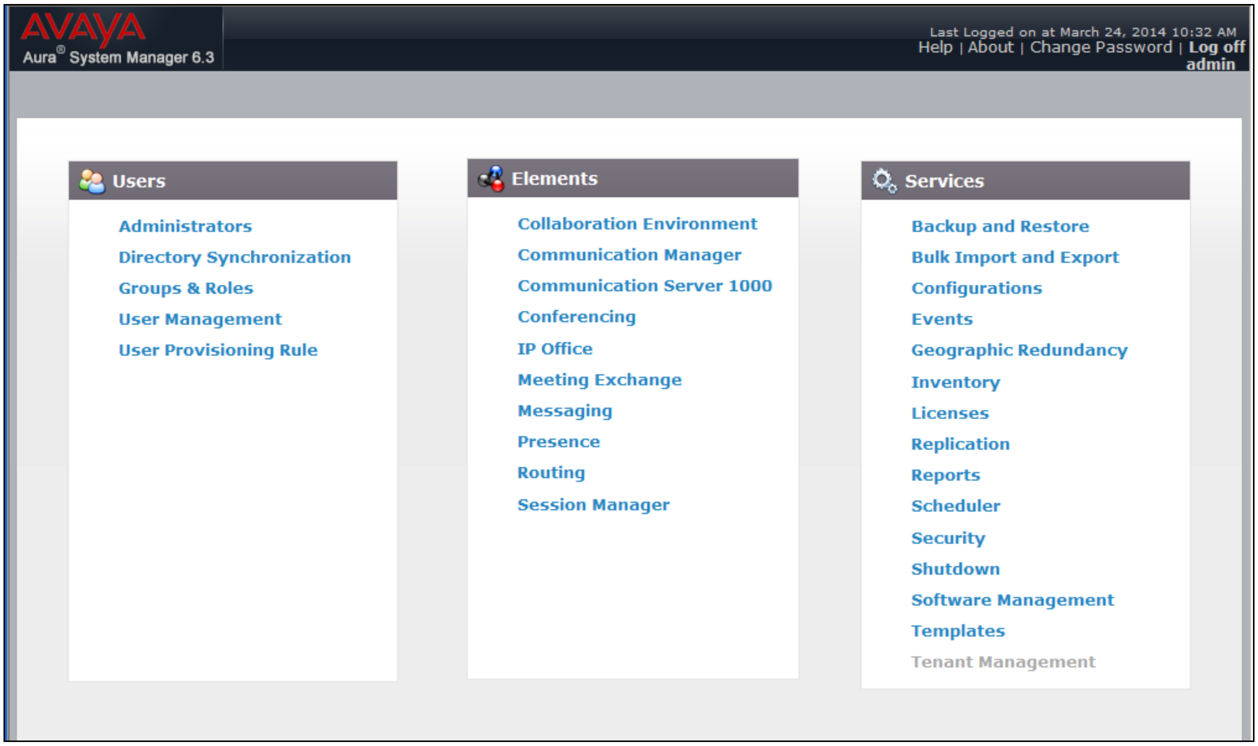
It is assumed that Session Manager and System Manager have been installed, network connectivity exists between the two platforms, and following topics are already configured:

- **SIP Domains**
- **Locations**
- **SIP Entities**
- **Entity Links**
- **Time Ranges**
- **Routing Policy**
- **Dial Patterns**
- **Manage Element**
- **Applications**
- **Application Sequence**

This section only discusses the User Management process to add SIP users that will be used during the compliance test.

6.1. Configure SIP Users

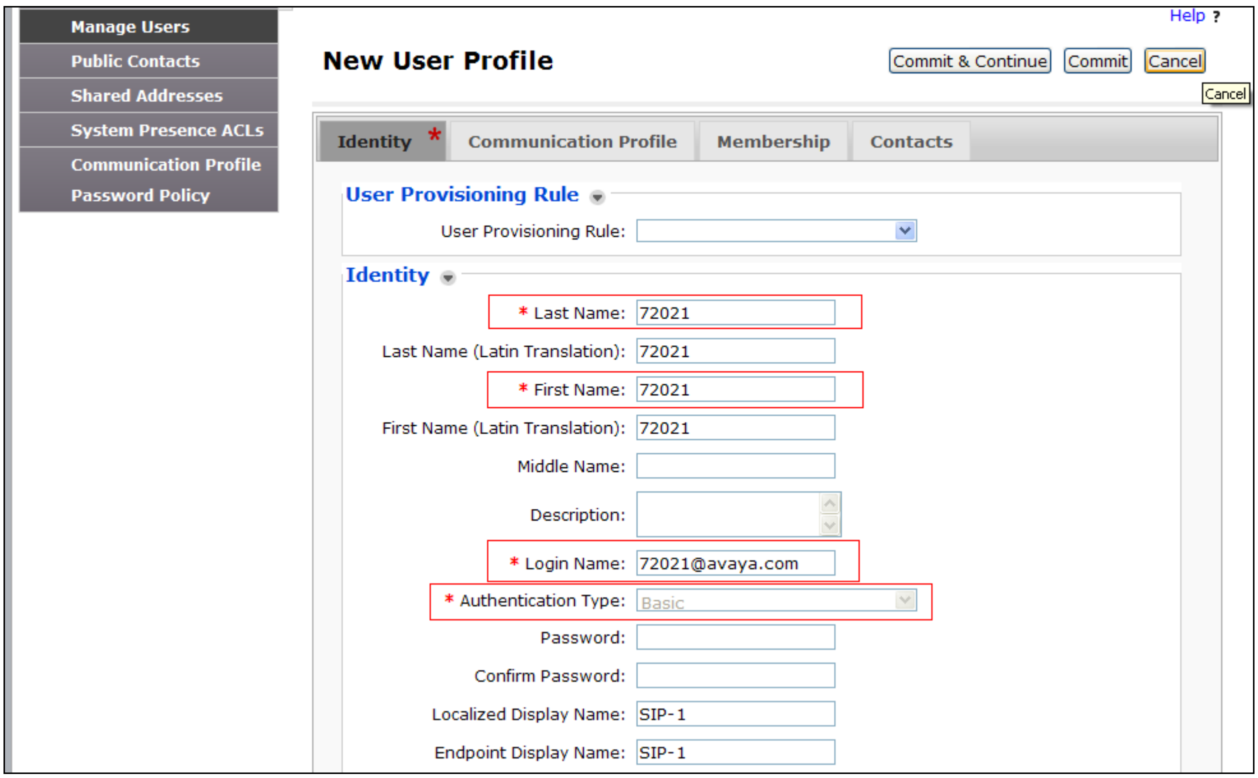
Launch a web browser, enter **http://<IP address of System Manager>** in the URL, and log in with the appropriate credentials (not shown).



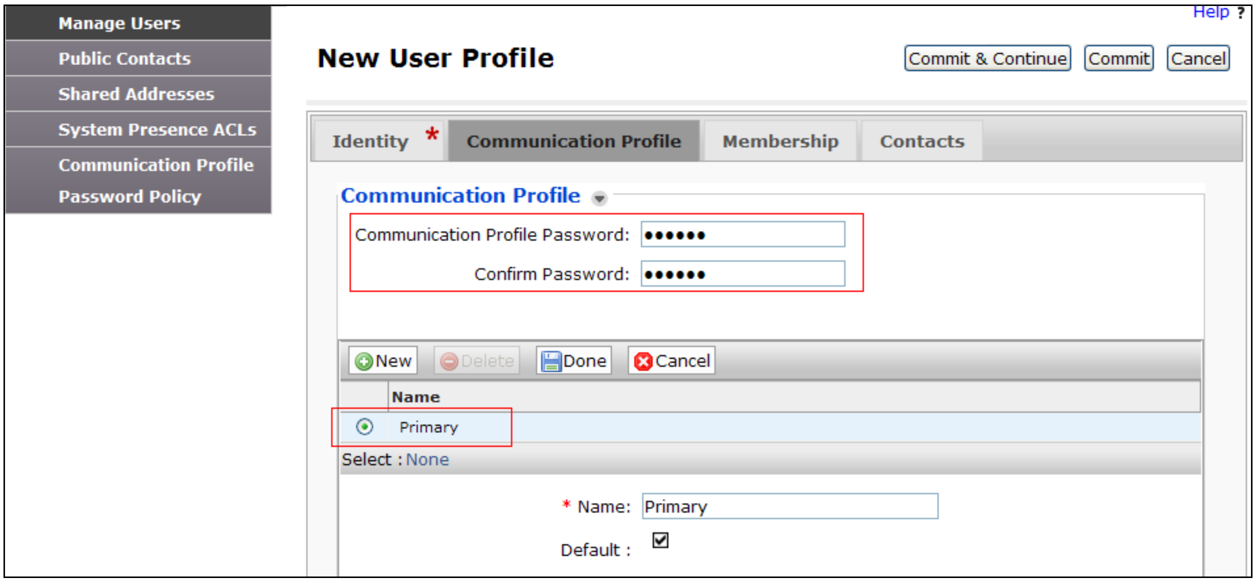
During the compliance test, no special users were created for this solution. All users were created prior to the compliance test. However, steps to configure a user are included. When adding new SIP user, use the option to automatically generate the SIP station in Communication Manager, after adding a new SIP user.

To add new SIP users, Navigate to **Home →Users → User management → Manage Users**. Click **New** (not shown) and provide the following information:

- Identity section
  - **Last Name –** Enter last name of user.
  - **First Name –** Enter first name of user.
  - **Login Name –** Enter extension number@sip domain. The sip domain is defined as Authoritative Domain in **Section 5.2**.
  - **Authentication Type** – Verify **Basic** is selected.

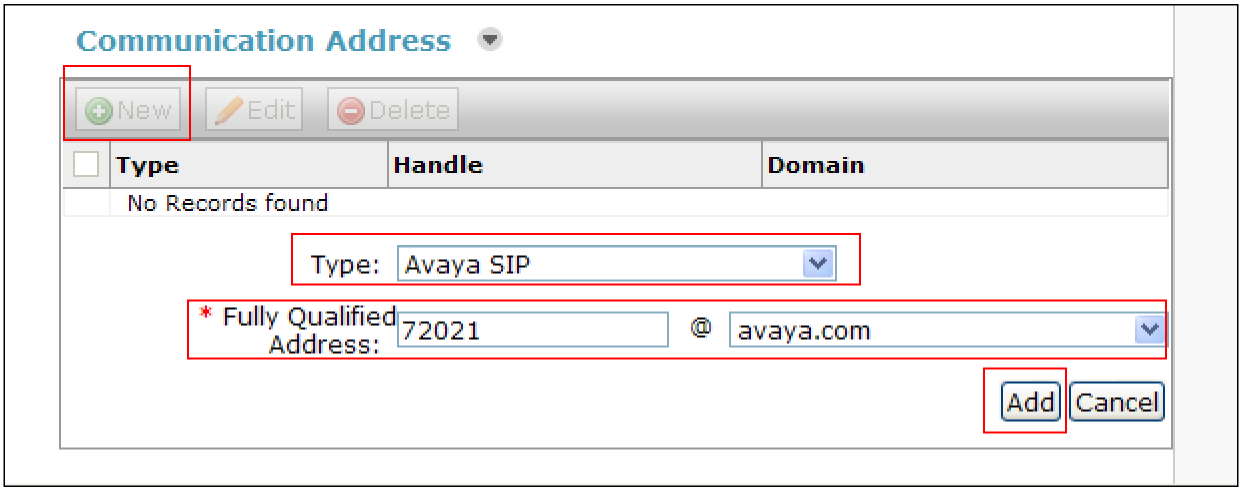


- <u>Communication Profile section</u>
  - **Communication Profile Password** – Enter a numeric value used to logon to SIP telephone.
  - **Confirm Password** – Repeat numeric password
  - Verify there is a default entry identified as the **Primary** profile for the new SIP user. If an entry does not exist, select **New** and enter values for the following required attributes:
    - **Name** – Enter **Primary**.
    - **Default** – Enter ☑

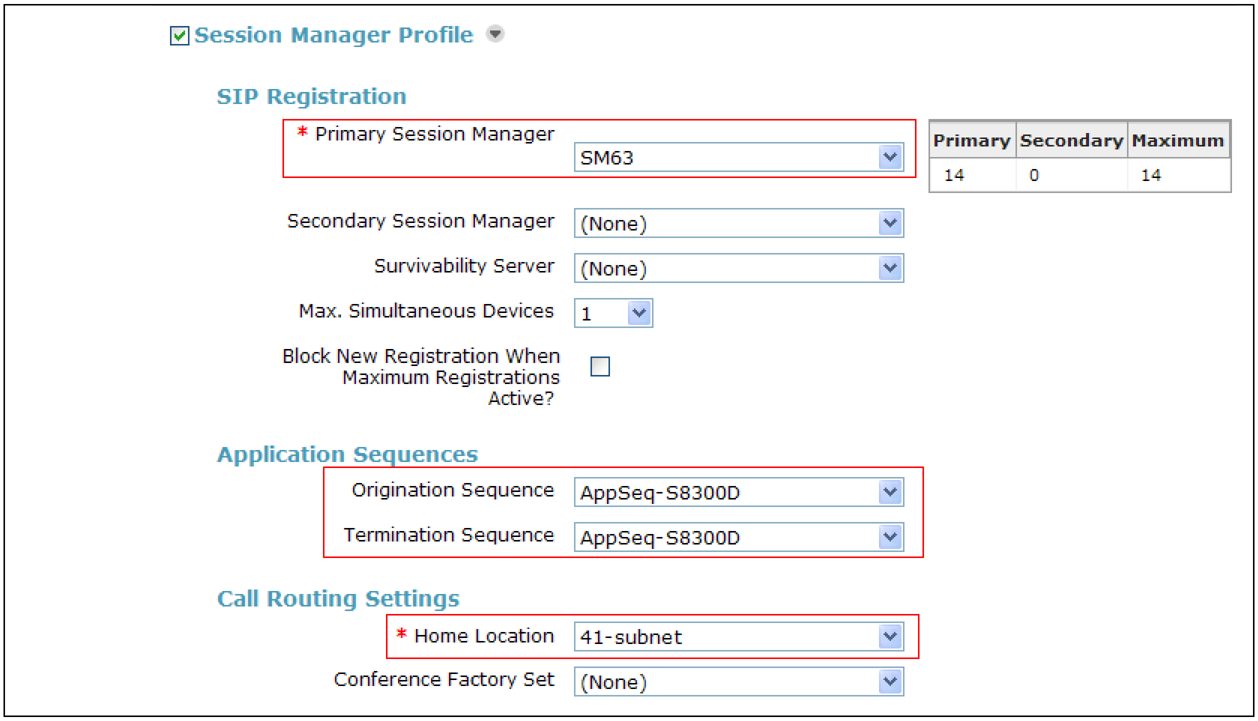


- Communication Address sub-section
  Select **New** to define a **Communication Address** for the new SIP user, and provide the following information.
    - **Type** – Select **Avaya SIP** using drop-down menu.
    - **Fully Qualified Address** – Enter same extension number and domain used for Login Name, created previously.

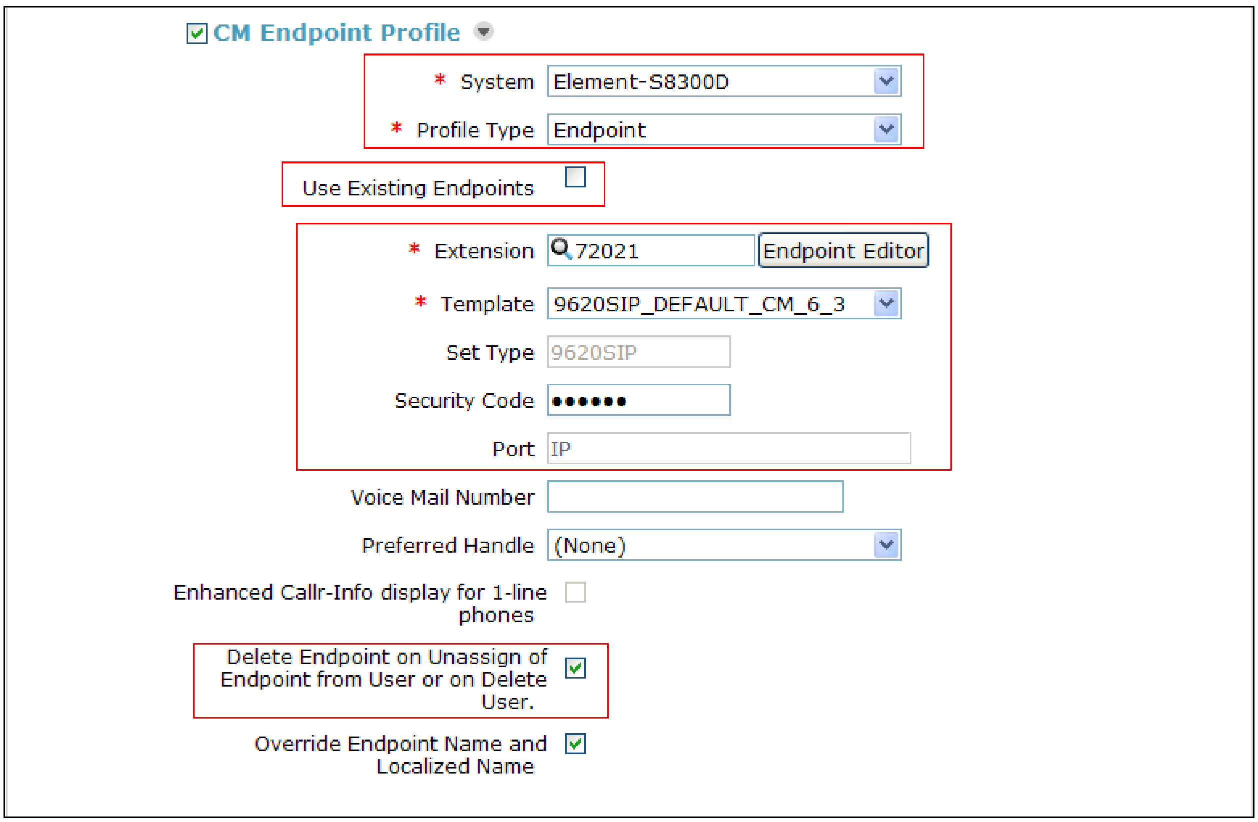
  Click the **Add** button to save the Communication Address for the new SIP user.



- <u>Session Manager Profile section</u>
  - **Primary Session Manager** – Select one of the Session Managers from the drop down list.
  - **Origination Application Sequence –** Select Application Sequence for Communication Manager from the drop down list.
  - **Termination Application Sequence** – Select Application Sequence for Communication Manager from the drop down list.
  - **Home Location** – Select a location already defined in the **Location** form.

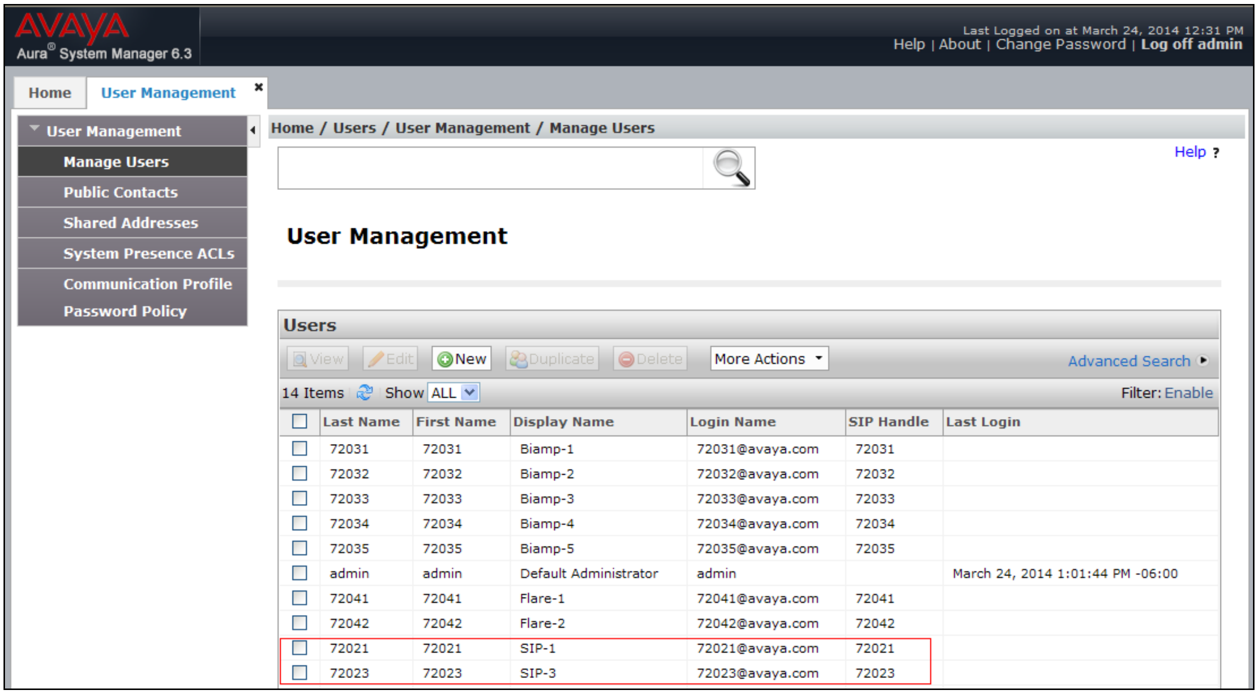


- <u>CM Endpoint Profile section</u>
    - **System** – Select "Managed Element", using the drop down menu.
    - **Profile Type** – Select "Endpoint", using the drop down menu.
    - **Use Existing Endpoints -** Leave unchecked to automatically create new endpoint when new user is created. Or else, check the box if endpoint is already defined in Communication Manager.
    - **Extension** - Enter same extension number used in this section.
    - **Template** – Select template for type of SIP phone, using the drop down menu
    - **Security Code** – Enter numeric value used to logon to SIP telephone.
    - **Port** – Verify "IP" is shown for this field.
    - **Delete Endpoint on Unassign of Endpoint from User or on Delete User** – Check the box to automatically delete station when Endpoint Profile is un-assigned from user.



Click **Commit** (not shown) to save definition of new user.

The following screen shows the created users during the compliance test.



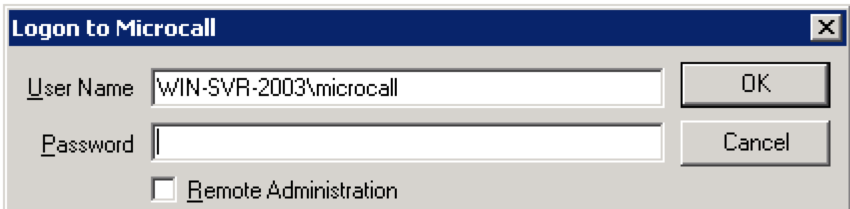
7. Configure MicroCall

This section describes the operation of MicroCall to receive CDR data from Communication Manager. Installation of the MicroCall software was performed by a Micro-Tel engineer prior to the actual compliance test. In this section, the following topics are discussed:

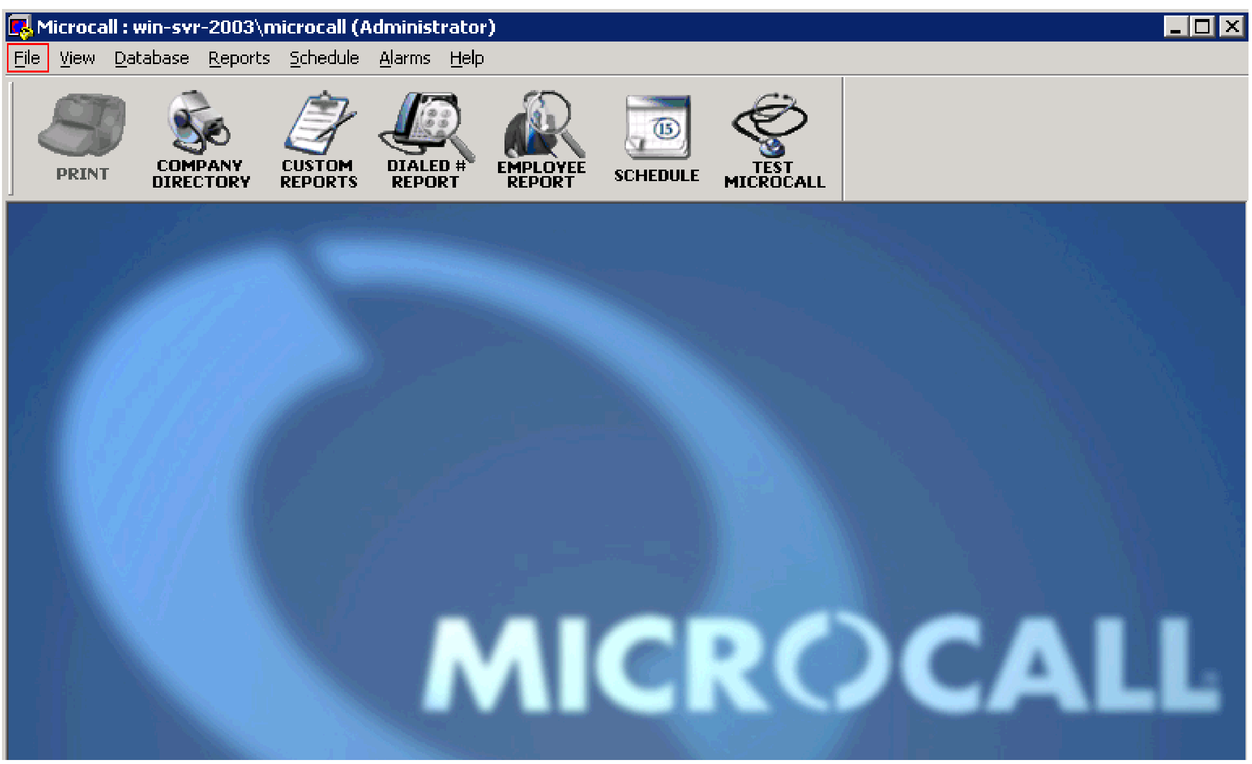
- Configure MicroCall
- View MicroCall CDR report

7.1. Configure MicroCall

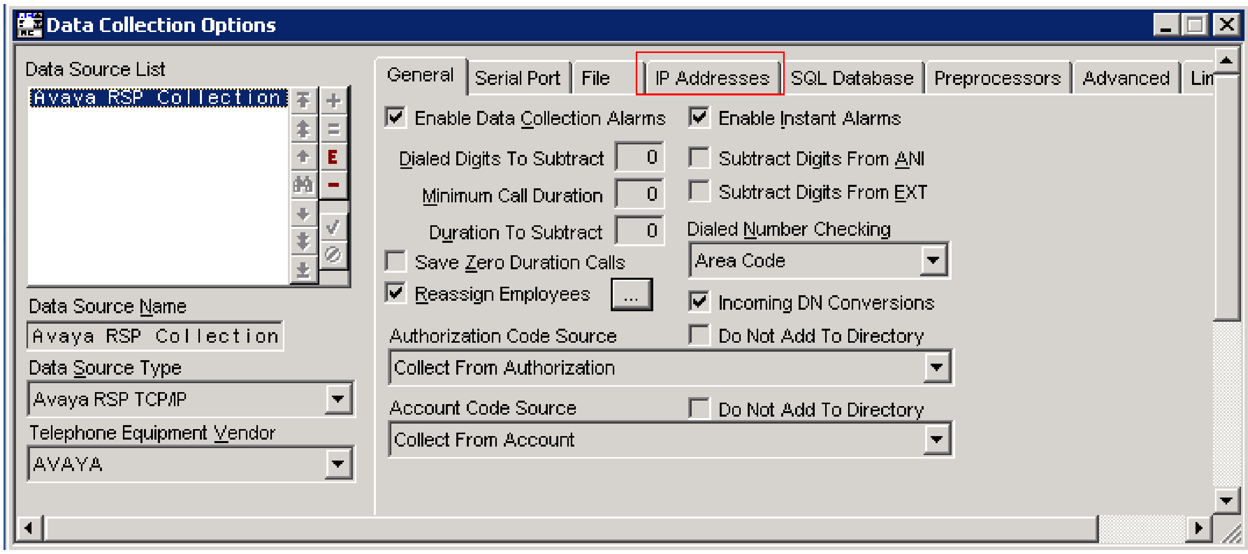
To configure MicroCall to communicate with Communication Manager, navigate to **Start → All Programs→ Microcall→Microcall**, and provide credentials to log into the Control Center page.



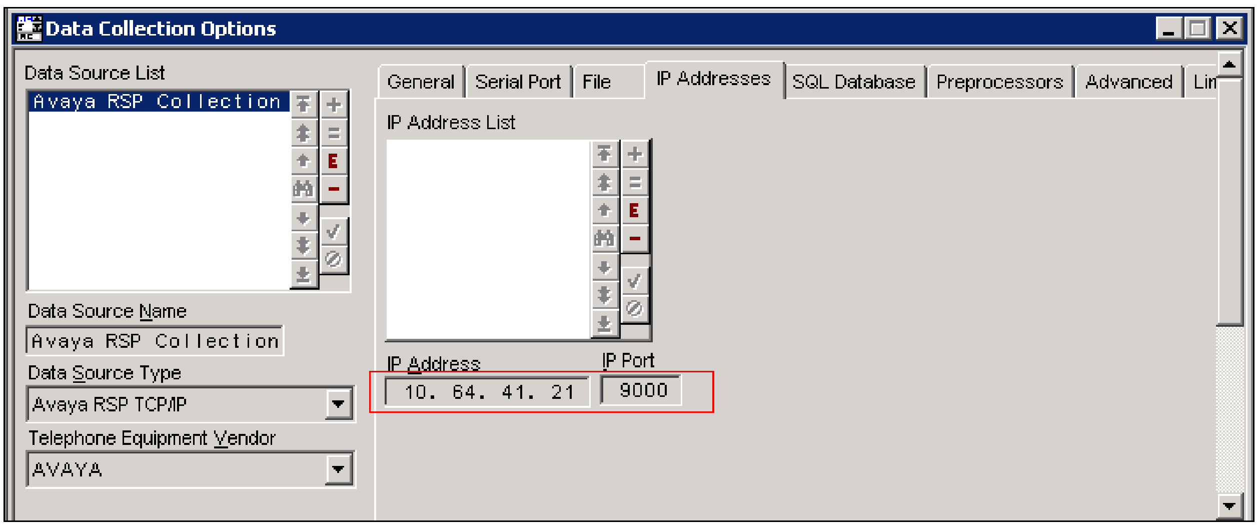
From the Main page, navigate to **File → Data Collection Options → Data Sources** (not shown).



From the **Data Collection Options** page, select **IP Addresses** submenu.



Enter the IP address of Communication Manager, and port that is utilizing for CDR data.

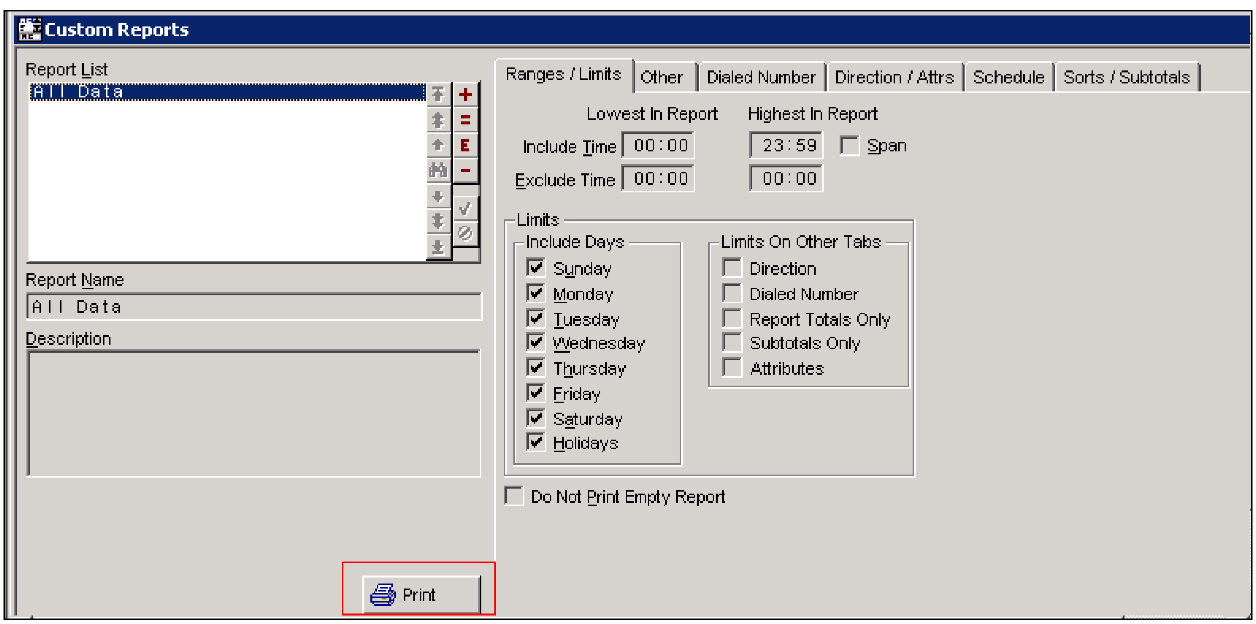


Note4: *A Micro-Tel engineer configured setting up a* ***format type*** *and* ***RSP configuration*** *prior to the actual test. Please contact Micro-Tel for above configuration issues.*

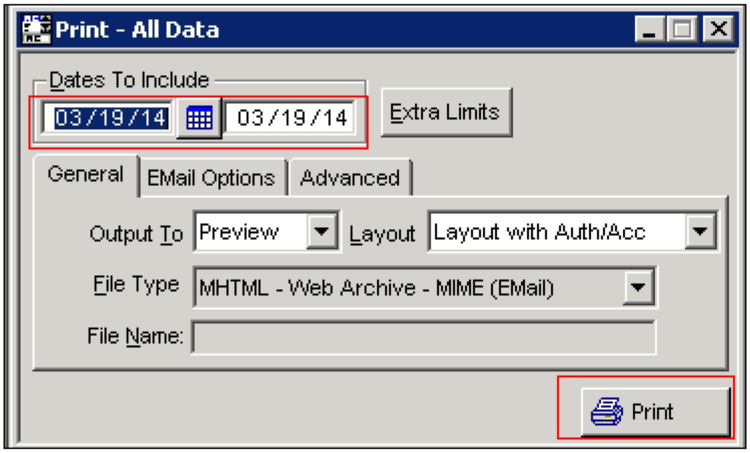
7.2. View MicroCall Report

To view the CDR report, launch the **Custom Report** from the Main menu.

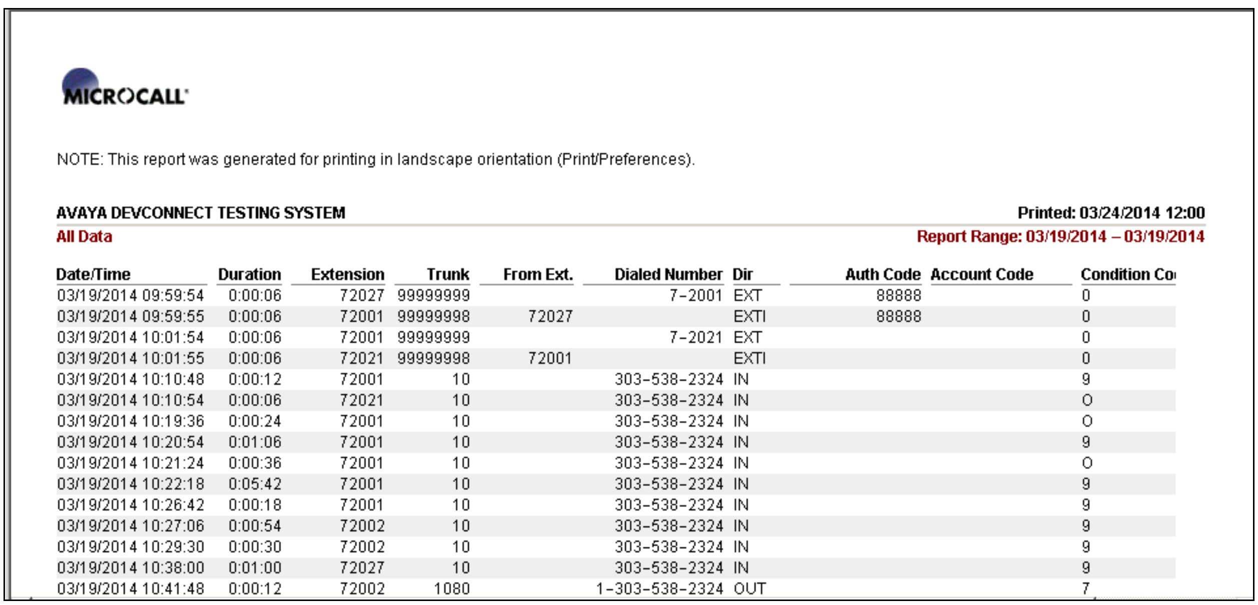
From the Custom Reports page, select the **Print** button at the bottom.



From the Print - All Data page, select **From** and **To** dates to collect CDR data. Click the **Print** button. The following screen shows From date (03/19/14) and To date (03/19/14).



The following shows the sample report collected during the compliance test.



MICROCALL

NOTE: This report was generated for printing in landscape orientation (Print/Preferences).

AVAYA DEVCONNECT TESTING SYSTEM **Printed: 03/24/2014 12:00**

All Data **Report Range: 03/19/2014 – 03/19/2014**

Date/Time	Duration	Extension	Trunk	From Ext.	Dialed Number	Dir	Auth Code	Account Code	Condition Co
03/19/2014 09:59:54	0:00:06	72027	99999999		7-2001	EXT	88888		0
03/19/2014 09:59:55	0:00:06	72001	99999998	72027		EXTI	88888		0
03/19/2014 10:01:54	0:00:06	72001	99999999		7-2021	EXT			0
03/19/2014 10:01:55	0:00:06	72021	99999998	72001		EXTI			0
03/19/2014 10:10:48	0:00:12	72001	10		303-538-2324	IN			9
03/19/2014 10:10:54	0:00:06	72021	10		303-538-2324	IN			O
03/19/2014 10:19:36	0:00:24	72001	10		303-538-2324	IN			O
03/19/2014 10:20:54	0:01:06	72001	10		303-538-2324	IN			9
03/19/2014 10:21:24	0:00:36	72001	10		303-538-2324	IN			O
03/19/2014 10:22:18	0:05:42	72001	10		303-538-2324	IN			9
03/19/2014 10:26:42	0:00:18	72001	10		303-538-2324	IN			9
03/19/2014 10:27:06	0:00:54	72002	10		303-538-2324	IN			9
03/19/2014 10:29:30	0:00:30	72002	10		303-538-2324	IN			9
03/19/2014 10:38:00	0:01:00	72027	10		303-538-2324	IN			9
03/19/2014 10:41:48	0:00:12	72002	1080		1-303-538-2324	OUT			7

8. Verification Steps

The following steps may be used to verify the configuration:

- Check the CDR status, by running the **status cdr** command in Communication Manager, and verify the **Link State** is "up" and **Reason Code** is "OK".

```
status cdr-link
                                CDR LINK STATUS
                   Primary                        Secondary

       Link State: up                             down
Number of Retries:                                999
      Date & Time: 2014/03/19 14:16:45            2014/03/19 16:17:47
  Forward Seq. No: 11                             0
 Backward Seq. No: 0                              0
CDR Buffer % Full:   0.00                           0.07
      Reason Code: OK                             CDR connection is closed
```

- Make several SIP calls between two Communication Managers, and verify that call records were collected from MicroCall.

9. Conclusion

These Application Notes describe the procedures for configuring MicroCall to collect call detail records from Communication Manager. Testing was successful.

10. References

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

[1] *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 9 Release 6.3, October 2013, available at http://support.avaya.com.
[2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document 555-245-205, Release 6.3, October 2013, available at http://support.avaya.com.

The MicroCall Solution and Product information is available from MicroCall. Visit https://www.microcall.com/literature_request.html

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 DevConnect Program at devconnect@avaya.com.