Avaya Solution & Interoperability Test Lab

Application Notes for ASC telecom MARATHON EVOLUTION Voice Recorder with Avaya Communication Manager Using Single Step Conferencing – Issue 1.1

Abstract

These Application Notes describe the conformance testing of the ASC telecom MARATHON EVOLUTION voice recorder with Avaya Communication Manager using the Single Step Conferencing feature. These Application Notes contain an extensive description of the configurations for both MARATHON EVOLUTION and Avaya Communication Manager which were used for testing. The testing which was performed tested the major functions of the MARATHON EVOLUTION product.

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

Table of Contents

1. Introduction

These Application Notes describe the configuration used to enable the ASC telecom MARATHON EVOLUTION voice recording server to interoperate with Avaya Communication Manager and Avaya Application Enablement Services (AES). The MARATHON EVOLUTION voice recorder offers various methods of voice recording. For the purpose of the tests described by these Application Notes, the Avaya Communication Manager Single Step Conference facility was used.

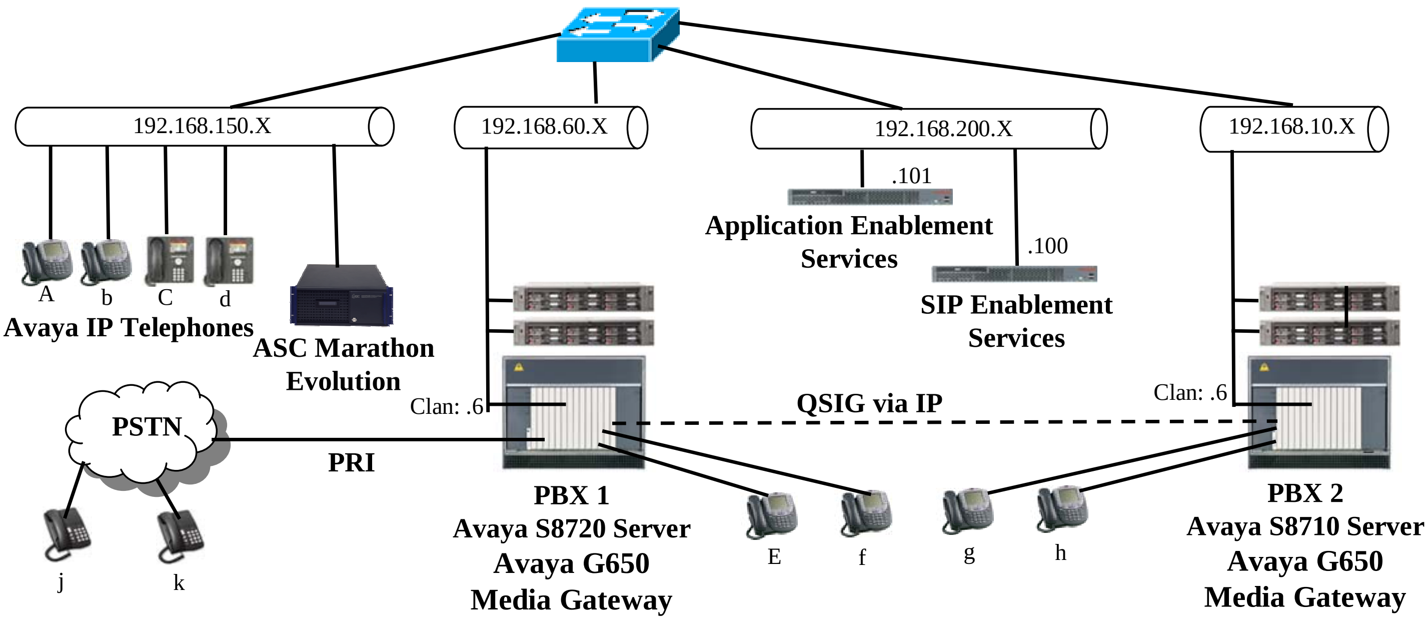

Figure 1: MARATHON EVOLUTION Test Configuration

In the above diagram, the ASC MARATHON EVOLUTION and ASC RIA*active* components act in concert to record voice conversations from telephones attached to PBX 1. The RIA*active* receives events from the Avaya AES server when the state of calls associated with PBX 1 change, and informs the MARATHON EVOLUTION of these transitions. The DMCC service provided by Avaya AES is used to monitor call activity associated with PBX 1. The ASC MARATHON EVOLUTION voice recorder is attached to PBX 1 via the local area network. PBX 2 is included in the configuration solely to test the ability to monitor conversations which traverse a trunk to a networked PBX.

When a call is to be recorded, the MARATHON EVOLUTION voice recorder initiates a single-step conference with the station being monitored using one of its Virtual CTI Stations, and thus includes itself in the call which it wishes to record. The voice stream for such calls is received via the LAN interface to PBX 1.

The PBX 2 system is attached to PBX 1 via an IP/QSIG interface, and is used as a networked PBX system. This allows remote networked telephones (g, h) to be included in the test.

The telephones depicted in these Application Notes are designated by an upper case letter if configured to be monitored by the MARATHON EVOLUTION voice recorder. A lower case letter designates those terminals which have been configured to not be monitored or are possibly unable to be monitored.

The following table contains additional information about each of the telephones shown in Figure 1. A "*" in the "Monitored" column indicated that the telephone is monitored by the MARATHON EVOLUTION voice recorder. Note that one Virtual CTI Station is required for each conversation to be monitored, as these are used by MARATHON EVOLUTION to create single step conferences with the stations to be monitored. Since a Virtual CTI Station can be used to monitor only one call at a time, the number of Virtual CTI Stations must be equal to the maximum number of simultaneous monitored calls. Note that calls between parties which are both monitored and conferences among monitored participants require additional Virtual CTI Stations.

Phone	Monitored	Model	Extension
A	*	Avaya 4610SW IP	60113
b		Avaya 4610SW IP	60114
C	*	Avaya 9620 (SIP)	60171
d		Avaya 9620 (SIP)	60172
E	*	Avaya 2410	60007
f		Avaya 2410	60008
g		Avaya 2420	1000013
h		Avaya 2420	1000014
j		N/A	069 7505 6176
k		N/A	069 7505 6630
L		Hunt Group (A & C)	
x		Virtual CTI Station	61401
y		Virtual CTI Station	61402
z		Virtual CTI Station	61403

Table 1: Device Monitor Configuration

2. Equipment and Software Validated

Software Component	Version
Avaya Communication Manager	R015x.00.0.825.4
Avaya TN2312BP IP Server Interface	HW15/FW042
Avaya TN799DP Control LAN	HW01/FW026
Avaya TN2302AP Media Processor	HW20/FW033
Avaya TN2464CP DS1 Interface	HW01/FW19
Avaya Application Enablement Services	r4-1-0-31-2-0
Avaya SIP Enablement Services	SES-5.0.0.0-825.31
Avaya 4610SW IP Telephone (H.323)	2.887
Avaya 9620 IP Telephone (SIP)	2.0.3.0
ASC MARATHON EVOLUTION SW	8.0
ASC MARATHON EVOLUTION platform OS	SuSE Linux
ASC RIAactive	5.0
ASC RIAactive platform OS	MS Windows XP 2003

Table 2: Hardware/Software Component Versions

3. Configuration

The configuration information in this section covers only PBX 1 – the system to which the MARATHON EVOLUTION voice recorder is attached.

3.1. Configure Avaya Communication Manager

The configuration and verification operations illustrated in this section were all performed using the Avaya Communication Manager System Administration Terminal (SAT) via SSH port 5022.

The information provided in this section describes the configuration of Avaya Communication Manager for this solution. For all other provisioning information such as installation and configuration, please refer to the product documentation in references **[1]** and **[2]**.

3.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer options** command to verify that Avaya Communication Manager is configured to meet the minimum requirements to run MARATHON EVOLOUTION. Those items shown in **bold** indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Parameter	Usage
Maximum Concurrently Registered IP Stations (p.2)	This must be sufficient to support the total number of IP stations used.
Computer Telephony Adjunct Links? (p.3)	This parameter must be set to “y”.
IP Stations? (p.4)	This parameter must be set to “y”.
IP_API_A (p.10)	This parameter must be set the number of Virtual CTI Stations.
IP Phone (p.10)	This parameter must be set the number of IP stations plus the number of Virtual CTI Stations.

Table 3: System-Parameters Customer-Options Parameters

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   60
           Maximum Concurrently Registered IP Stations: 12000 4
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 10    0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 1000  255
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 10    0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 2: System-Parameters Customer-Options Screen (p.2)

```
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? y
        Analog Trunk Incoming Call ID? n                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? n                         CAS Main? n
Answer Supervision by Call Classifier? n                Change COR by FAC? n
                                 ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? n
          ARS/AAR Dialing without FAC? n                      DCS (Basic)? n
          ASAI Link Core Capabilities? y                DCS Call Coverage? n
          ASAI Link Plus Capabilities? y               DCS with Rerouting? n
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? n
              ATM WAN Spare Processor? n                          DS1 MSP? n
                                ATMS? n             DS1 Echo Cancellation? y
                  Attendant Vectoring? n
```

Figure 3: System-Parameters Customer-Options Screen (p.3)

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                                IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                         ISDN Feature Plus? n
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                           ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                  ISDN-PRI? y
              ESS Administration? n               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                    Malicious Call Trace? n
     External Device Alarm Admin? n                 Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                  Multifrequency Signaling? y
      Global Call Classification? n        Multimedia Call Handling (Basic)? n
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? n
 Hospitality (G3V3 Enhancements)? n               Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? y
```

Figure 4: System-Parameters Customer-Options Screen (p.4)

```
display system-parameters customer-options                      Page  10 of  11
                     MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID  Rel. Limit          Used
IP_API_A       : 1000           0
IP_API_B       : 1000           0
IP_API_C       : 1000           0
IP_Agent       : 1000           0
IP_IR_A        : 1000           0
IP_Phone       : 12000          4
IP_ROMax       : 12000          0
IP_Soft        : 1000           0
IP_eCons       : 128            0
oneX_Comm      : 12000          0
```

Figure 5: System-Parameters Customer-Options Screen (p.10)

3.1.2. Configure Dial Plan

Use the **change dialplan analysis** command to specify that dialed strings which begin with "1", or "6" are extensions. Include the strings "*01", "*83", and "*9" which are Trunk Access Codes.

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all           Percent Full:    0

       Dialed    Total  Call     Dialed    Total  Call      Dialed    Total  Call
       String    Length Type     String    Length Type      String    Length Type
     0             1    fac
     1             7    ext
     6             6    ext
     *01           3    dac
     *83           3    dac
     *9            2    dac
```

Figure 6: Dialplan Analysis Screen

3.1.3. Configure IP Network Interface

Use the **change node-names ip** command to configure IP address, as shown in the following table.

Parameter	Usage
clan	Enter the IP address of the CLAN interface of PBX1.
ses	Enter the IP address of the SES server.

Table 4: Node-Names IP Parameters

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
clan                192.168.60.6
default             0.0.0.0
ses                 192.168.200.100
```

Figure 7: Node-Names IP Screen

Use the **change ip-network-region <x>** command to designate a network region to be used for the IP telephone communications using the parameters shown in the following table, where <x> is the network region assigned to the clan IP interface. In this case "1" is used, as the procr IP interface is assigned to default network region of "1".

Parameter	Usage
Location	Enter "1".
Authoritative Domain	Enter the domain name to be used for SIP communications.
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in **Figure 9**.

Table 5: IP-Network-Region Parameters

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: ffm.com
    Name: FFM
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                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

Figure 8: IP-Network-Region Form, p.2

Use the **change ip-codec-set** command to designate a codec set to be used. Testing was done with the G.711A codec.

Parameter	Usage
Audio Codec	Enter "G.711A".

Table 6: IP-Codec-Set Parameters

```
change change ip-codec-set 1                                          Page   1 of   2

                          IP Codec Set
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711A            n          2        20
```

Figure 9: IP-Codec-Set Form

3.1.4. Configure Interface to AES

Use the **change ip-services** command to configure the interface to the AES server, as shown in the following table.

Parameter	Usage
Service Type (p.1)	Enter "AESVCS".
Enabled (p.1)	Enter "y" to enable the service.
Local Node (p.1)	Enter the IP node name for the CLAN interface.
Local Port (p.1)	Enter "8765".
AE Services Server (p.4)	Enter the name that was assigned to the AES server when it was installed.
Password (p.4)	Enter the password that was assigned to the switch connection, as shown in **Figure 27**.
Enabled (p.4)	Enter "y" to enable the connection.

Table 7: IP Services Parameters

```
change ip-services                                             Page   1 of   4

                                    IP SERVICES
 Service     Enabled     Local         Local       Remote      Remote
  Type                   Node          Port        Node        Port
AESVCS        y      clan              8765
```

Figure 10: IP Services Screen, p. 1

```
change ip-services                                             Page   4 of   4
                            AE Services Administration

   Server ID    AE Services       Password          Enabled    Status
                   Server
      1:      aes_server_1      XXXXXXXXXXXXXXXX      y        in use
```

Figure 11: IP Services Screen, p. 4

Use the **add cti-link** command to add a CTI link for use by TSAPI. The link number can be any value between 1 and 64 which is not currently assigned to another link. The link number specified must be the same value that is used in the “Switch CTI Link Number” field shown in **Figure 30**. Use an unused extension as the value for the “Extension” parameter. The value chosen for the “Name” parameter is a matter of personal preference.

```
change cti-link 4                                               Page   1 of   3
                                    CTI LINK
 CTI Link: 4
Extension: 69996
     Type: ADJ-IP
                                                                        COR: 1
     Name: AES-devcon223-tsapi
```

Figure 12: CTI-link Screen

3.1.5. Configure Stations

3.1.5.1 Configure H.323 IP Stations

Use the **add station** command to create an IP station for extensions A and b in **Table 1**, using the values shown in the following table.

Parameter	Usage
Extension	Use an unused extension which is compatible with the dial plan.
Type	Use a type value which corresponds to the physical station to be used.
Name	Any alphanumeric string can be assigned as an extension name, which is used for identification purposes.

Table 8: Configuration IP Stations

```
change station 60113                                            Page   1 of   5
                                     STATION

Extension: 60113                         Lock Messages? n              BCC: 0
     Type: 4610                          Security Code: 31106           TN: 1
     Port: S00101                      Coverage Path 1: 1              COR: 1
     Name: extn 60113                  Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 60113
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n



                                              Customizable Labels? y
```

Figure 13: IP Station Screen

3.1.5.2 Configure SIP Stations

Use the **add station** command to create SIP IP station for extension 60171, using the values shown below. Repeat this section for extension 60172.

Parameter	Usage
Extension	Use an unused extension which is compatible with the dial plan.
Type	Use a type value which corresponds to the physical station to be used.
Name	Any alphanumeric string can be assigned as an extension name.

Table 9: Configuration SIP IP Stations

```
change station 60171                                            Page   1 of   6
                                     STATION

Extension: 60171                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code:                 TN: 1
     Port: S00126                      Coverage Path 1:                COR: 1
     Name: extn 60171                  Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 60171
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n




                                              Customizable Labels? y
```

Figure 14: SIP IP Stations Screen

Use the **add off-pbx-telephone station-mapping** command for each of the SIP stations added in the previous step.

Parameter	Usage
Extension (p.1)	Enter the extension of the SIP telephone from **Figure 15**.
Application (p.1)	Enter "OPS".
Phone Number (p.1)	Enter the extension of the SIP telephone from **Figure 15**.
Trunk Selection (p.1)	Enter the number of the SIP trunk which was allocated in **Figure 22**.
Call Limit (p.2)	Enter a value which is sufficient for the user to participate in transfer operations or conference calls. A value of "3" was used for these tests.

Table 10: Configuration Off-Pbx-Telephone Station-Mapping

```
add off-pbx-telephone station-mapping                           Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station         Application Dial   CC  Phone Number      Trunk       Config
 Extension                   Prefix                       Selection   Set
 60171              OPS           -     60171              83         1
                                  -
```

Figure 15: SIP Off-Pbx-Telephone Station-Mapping Screen, p. 1

```
add off-pbx-telephone station-mapping                           Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Call       Mapping      Calls       Bridged        Location
 Extension      Limit      Mode         Allowed     Calls
 60171           3       both           all         none
```

Figure 16: SIP Off-Pbx-Telephone Station-Mapping Screen, p. 2

3.1.5.3 Configure Virtual CTI Stations

Use the **add station** command to create a station for each of the Virtual CTI Stations listed in **Table 1**. Sufficient Virtual CTI Stations must be created to monitor the maximum number of simultaneous monitored conversations. These stations are subsequently assigned by the ASC DataManager for monitoring in **Figure 52**. Note that the station numbers must be sequential.

Parameter	Usage
Type	Enter “4620”.
Name	Any alphanumeric string can be assigned as an extension name.
Security Code	Enter a security code which identical to the extension.
IP Softphone	Enter “y”.

Table 11: Virtual CTI Station Parameters

```
add station 61401                                            Page   1 of   5
                                     STATION

Extension: 61401                         Lock Messages? n             BCC: 0
     Type: 4620                          Security Code: 61401          TN: 1
     Port: S00104                      Coverage Path 1:               COR: 1
     Name: CTI 61401                   Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                             Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                   Message Lamp Ext: 61401
            Speakerphone: 2-way                 Mute Button Enabled? y
        Display Language: english                  Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal                Media Complex Ext:
   Survivable Trunk Dest? y                            IP SoftPhone? y

                                                  IP Video Softphone? n


                                                 Customizable Labels? y
```

Figure 17: Virtual CTI Station Screen

3.1.6. Configure Hunt Group

Use the **add hunt-group** command to create a hunt group which is used to test the ability of MARATHON EVOLOUTION to monitor hunt groups. Assign an unused extension to the hunt group. Add extensions for telephones "A" and "C" to the hunt group, which are assigned to IP phones which are monitored by MARATHON EVOLOUTION.

Parameter	Usage
Group Name	Any alphanumeric string can be used as a Group Name.
Group Extension	Use an unused extension which is compatible with the dial plan.
MEMBER ASSIGNMENTS	Add the extensions which are to be assigned to this hunt group to this list. For this test, extensions 60113 and 60171 are used.

Table 12: Configuration IP Stations

```
add hunt-group 4                                                Page   1 of  60
                                  HUNT GROUP

            Group Number: 4                                ACD? n
              Group Name: asc                            Queue? n
         Group Extension: 61304                         Vector? n
              Group Type: ucd-mia                Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display:
```

Figure 18: Hunt Group Screen, p. 1

```
change hunt-group 4                                             Page   3 of  60
                                  HUNT GROUP
         Group Number: 4    Group Extension: 61304         Group Type: ucd-mia
  Member Range Allowed: 1 - 1500       Administered Members (min/max): 1   /2
                                           Total Administered Members: 2
GROUP MEMBER ASSIGNMENTS
      Ext           Name(19 characters)       Ext           Name(19 characters)
   1: 60113         extn 60113            14:
   2: 60171         extn 60171            15:
   3:                                     16:
   4:                                     17:
   5:                                     18:
   6:                                     19:
   7:                                     20:
   8:                                     21:
   9:                                     22:
  10:                                     23:
  11:                                     24:
  12:                                     25:
  13:                                     26:

  At End of Member List
```

Figure 19: Hunt Group Screen, p. 3

3.1.7. Configure Interface to SIP Enablement Services

Use the **add signaling-group** command to allocate a signaling group for interface to SES using the following parameters:

Parameter	Usage
Group Type	Enter “sip”.
Near-end Node Name	Enter “clan” (defined in **Figure 7**) to designate the Control LAN as the near end node name.
Far-end Node Name	Enter “ses” to assign the SES server as the far end node name.
DTMF over IP	Enter “rtp-payload”. This value used to have Avaya Communication Manager send DTMF transmissions using RFC 2833 ([7]).
Direct IP-IP Audio Connections	Enter “y” to allow direct IP-IP endpoint connections (shuffling).

Table 13: Signaling-Group Parameters

```
add signaling-group 83                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: tls








   Near-end Node Name: clan                  Far-end Node Name: ses
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain:

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3
```

Figure 20: Signaling-Group Form

Use the **add trunk-group <*n*>** command, were **<*n*>** is an unused trunk number, to allocate a trunk group to be used as an interface to the SIP Enablement Services server. Use the parameters shown in the following table.

Parameter	Usage
Group Type (p.1)	Enter "sip".
Group Name (p.1)	Assign a name for identification purposes.
TAC (p.1)	Enter the Trunk Access Code allocated in **Figure 6**.
Service Type (p.1)	Enter "tie".
Signaling Group (p.1)	Enter the number of the signaling group allocated in **Figure 20**.
Number of Members (p.1)	Enter a number large enough to support the maximum number of anticipated simultaneous calls to be made via the SIP trunk.

Table 14: Trunk-Group Parameters

```
add trunk-group 83                                             Page   1 of  21
                                TRUNK GROUP

Group Number: 83                   Group Type: sip           CDR Reports: y
  Group Name: SIP                         COR: 1        TN: 1        TAC: *83
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 83
                                                     Number of Members: 255
```

Figure 21: Trunk-Group Form, p.1

Use the **add off-pbx-telephone station-mapping <x>** command for each of the SIP stations shown in **Table 1**.

Parameter	Usage
Station Extension (p.1)	The extension of the SIP telephone. This extension should have been allocated as described in **Figure 14**.
Application (p.1)	Enter "OPS".
Phone Number (p.1)	Enter the extension.
Trunk Selection (p.1)	Enter the number of the SIP trunk which allocated in **Figure 21**.
Call Limit (p.2)	Enter "3" to allow transfer/conference operations.

Table 15: off-pbx-telephone station-mapping Parameters

```
add off-pbx-telephone station-mapping 60171                    Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 60171             OPS          -    60171                83         1
```

Figure 22: off-pbx-telephone station-mapping Screen, p.1

```
change off-pbx-telephone station-mapping 60171                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged        Location
 Extension     Limit      Mode         Allowed     Calls
 60171          3       both           all         none
```

Figure 23: off-pbx-telephone station-mapping Screen, p.2

3.2. Configure Avaya AES

The AES server is configured via a web browser by accessing the following URL:

https://<AES server address>:8443/MVAP/

Once the login screen appears, enter either the appropriate login ID/password for performing administrative activities or user management.

Figure 24: AES Welcome Screen

After logging in, select “CTI OAM Admin” which displays the following screen. Verify that the AES server installation has a TSAPI service license. If this is not the case, please contact an Avaya representative regarding licensing.

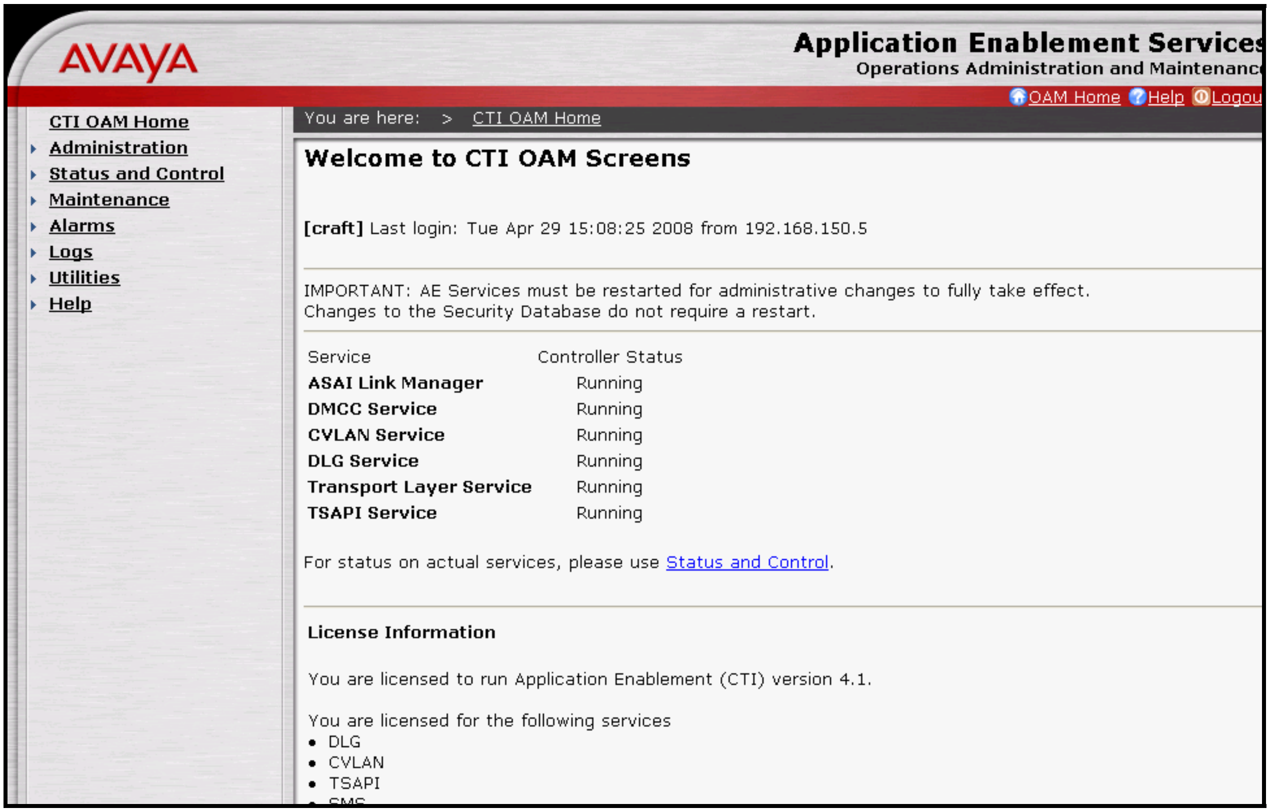

Figure 25: AES CTI OAM Welcome Screen

Navigate to **Administration->Switch Connections**. Enter the name of the Switch Connection to be added, and click on the "Add Connection" button. This name should match that which is used by the ASC DataManager in **Figure 50**.

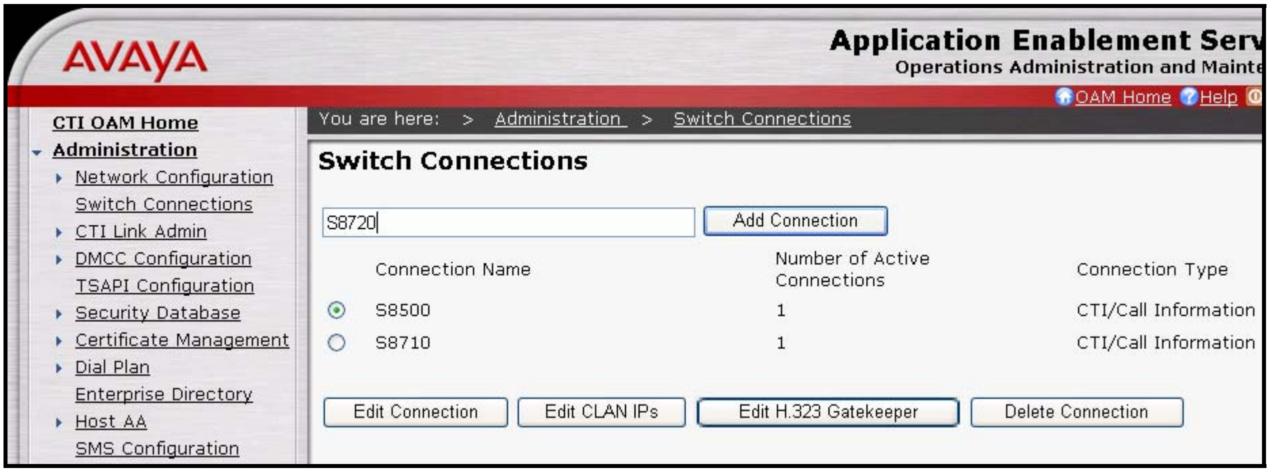

Figure 26: Switch Connection Screen

This causes the following screen to be presented. At this point, enter the screen fields as described in the following table, and click the “Apply” button.

Parameter	Usage
Switch Connection Type	Specify a type of CTI/Call Information.
Switch Password	The Switch Password must be the same as was entered into the Avaya Communication Manager AE Services Administration form via the “change ip-services” command, described in **Figure 11**. Passwords must consist of 12 to 16 alphanumeric characters
SSL	SSL (Secure Socket Layer) is enabled by default. Keep the default setting unless you are adding a Switch Connection for a DEFINITY Server CSI

Table 16: Configuration of Switch Password

Figure 27: Set Switch Password Screen

From the **Administration->Switch Connections** screen, click the "Edit CLAN IPs" button to display the screen show below. Enter the IP address of the CLAN with which AES is to use for communication with the switch, and click the "Add Name or IP" button.

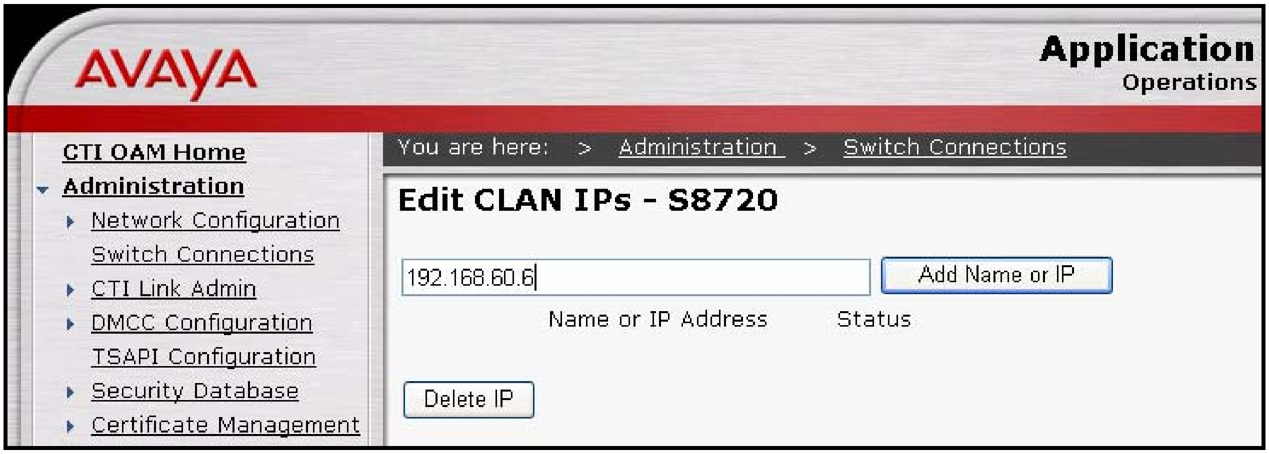

Figure 28: CLAN Screen

On the left margin of the screen, navigate to **Administration->CTI Link Admin->TSAPI Links.** The following screen is displayed. Click the "Add Link" button.

Figure 29: TSAPI Links Screen

Fill in the parameters for the link to be added. The "Link" parameter must be a value between 1 and 16 which is not assigned to another link. The "Switch Connection" parameter should be the name of the Avaya Server which is to be controlled by this link. The value for the TSAPI "Switch CTI Link Number" must be a value between 1 and 64, and must be the same as was used in the Avaya Communication Manager "add cti-link" configuration command in **Figure 12**. Click the "Apply Changes" button.

Figure 30: Add TSAPI Link Screen

Navigate to **User Management->Add User**.

The "CT User" field for this user must be set to "Yes". In this case, the AES user is the MARATHON EVOLUTIONCT application, which uses AES to monitor stations and initiate switching operations. The "User Id" and "User Password" must be the same as those configured for ASC DataManager in **Figure 51**.

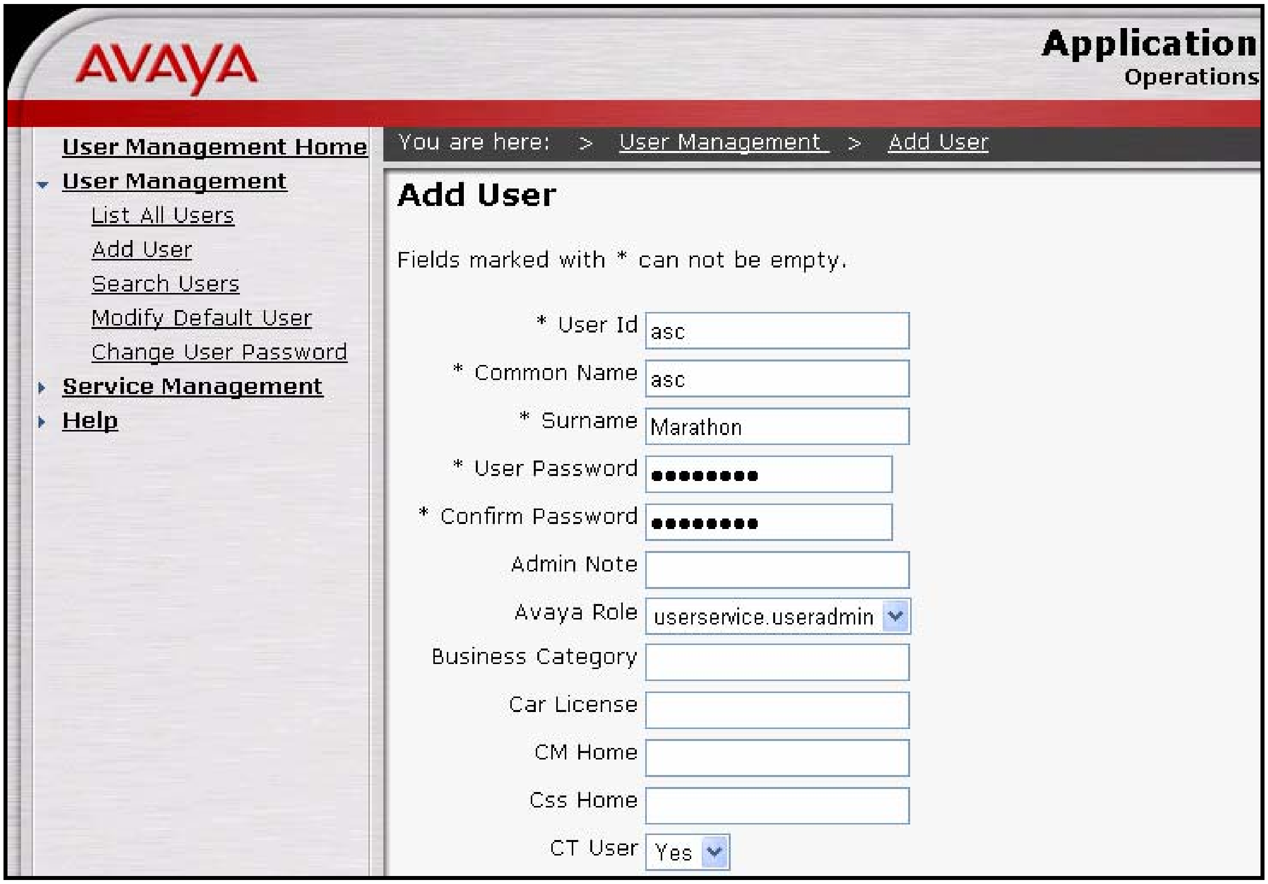

Figure 31: Add User Screen

Navigate to **Administration -> Network Configuration -> Ports** and configure the DMCC Server Ports as shown in the following table.

Parameter	Usage
Unencrypted Port	Set this port to 4721, enabled to match the value in **Figure 51**.
Encrypted Port	Set this port to 4722, enabled to match the value in **Figure 51**.

Table 17: DataManager AES Server Interface Parameters

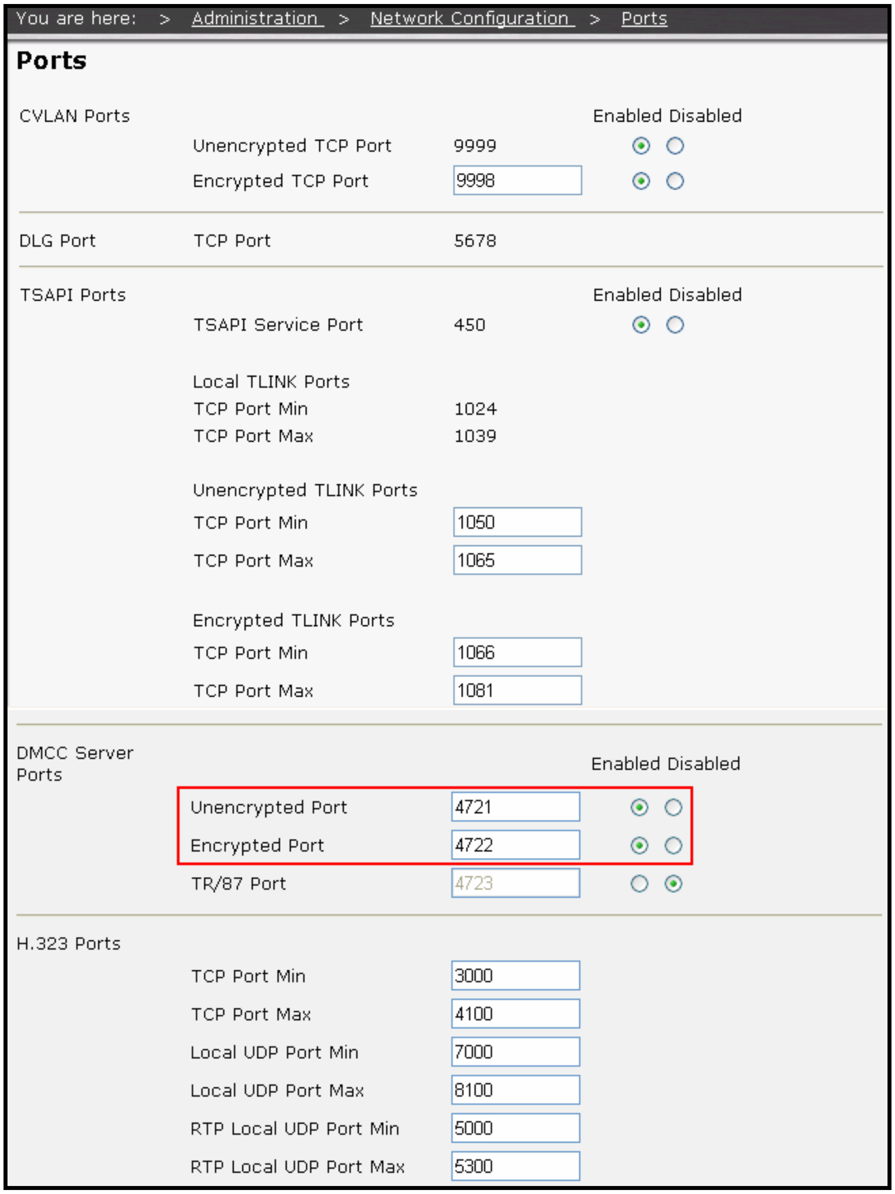

Figure 32: AES Port Configuration

3.3. Avaya SIP Enablement Services

Configure SES by entering "<SES IP Address>/admin/" in a web browser. After entering the administrator name and password, the following screen content is displayed. Select "Launch Administration Web Interface".

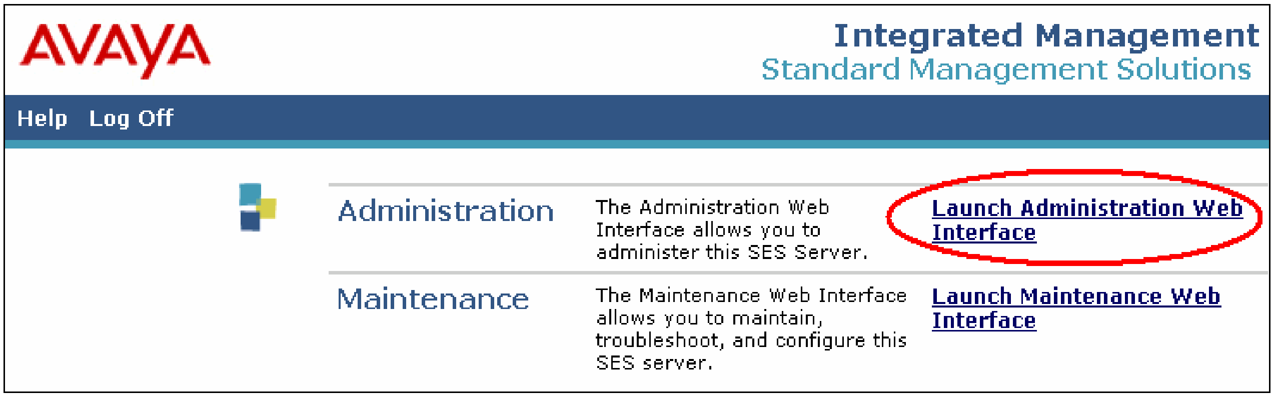

Figure 33: Launch Maintenance Web Interface Screen

3.3.1. Install License

From the "Top" menu navigate to **Server Configuration -> License**.

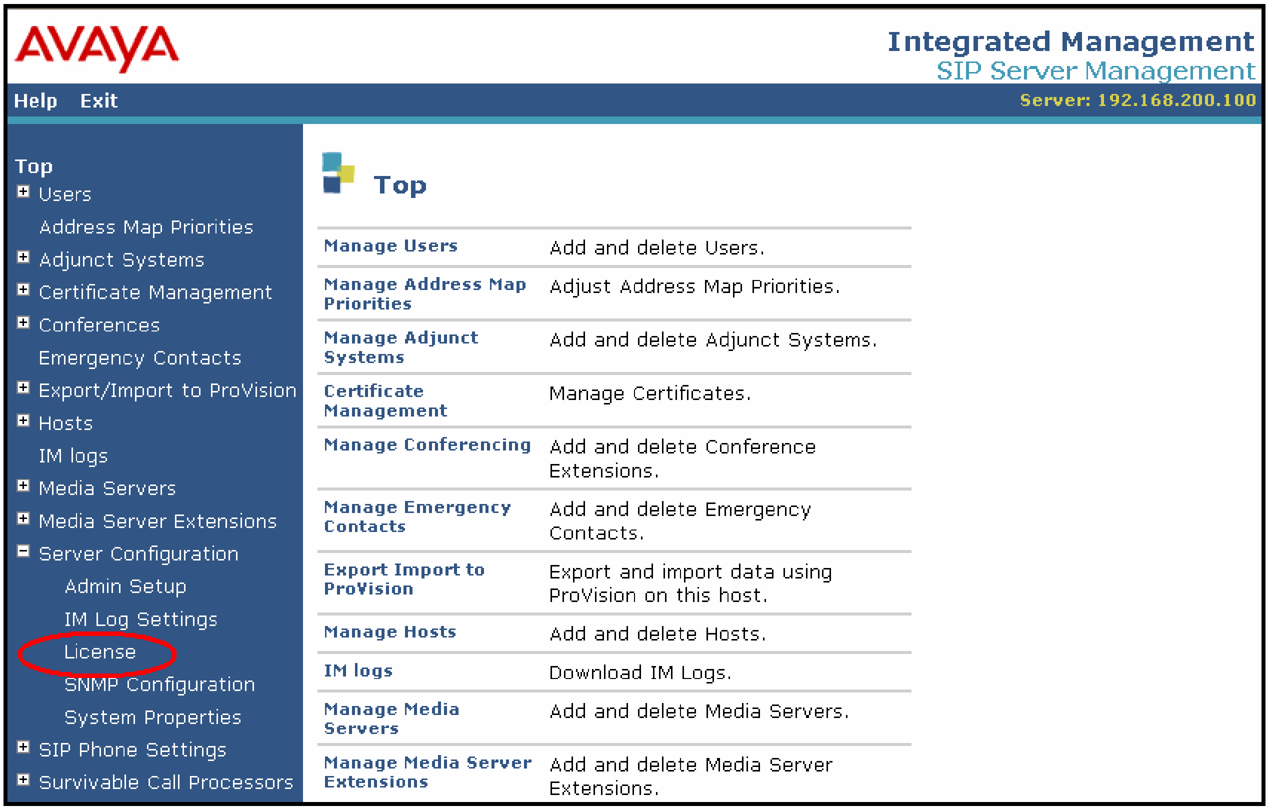

Figure 34: Select License from Top SES Screen

Click “Access WebLM”.

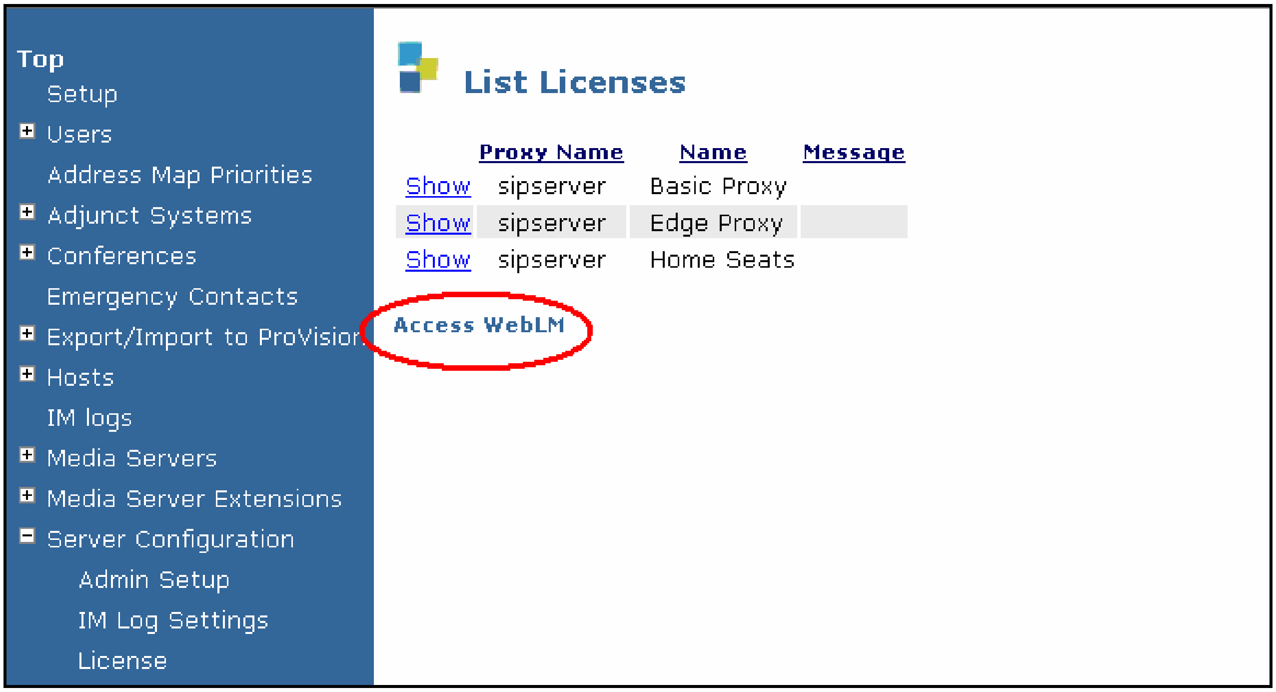

Figure 35: Select WebLM from License Screen

Log in to WebLM with the appropriate administrative user name and password. Read the license file when instructed by WebLM. Exit WebLM. Re-enter the URI “http:<SES IP address>/admin” into the Web browser.

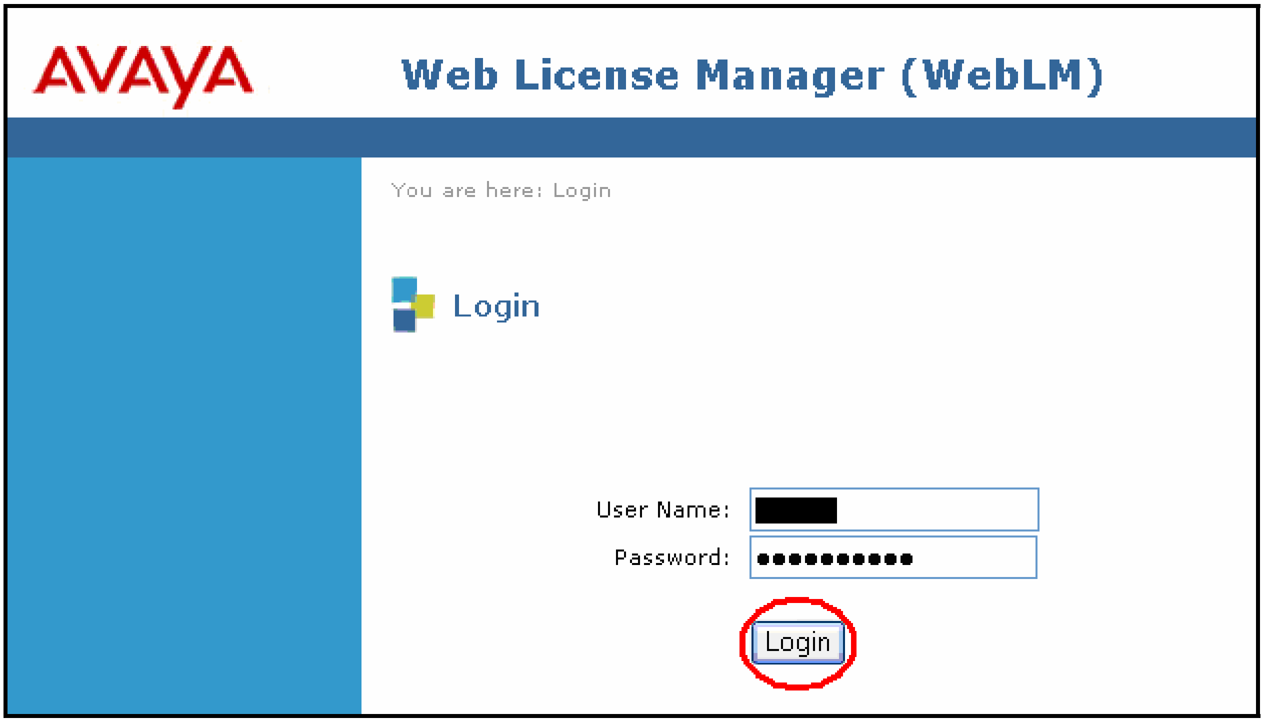

Figure 36: WebLM Login Screen

Click “Setup” on the Top SES menu.

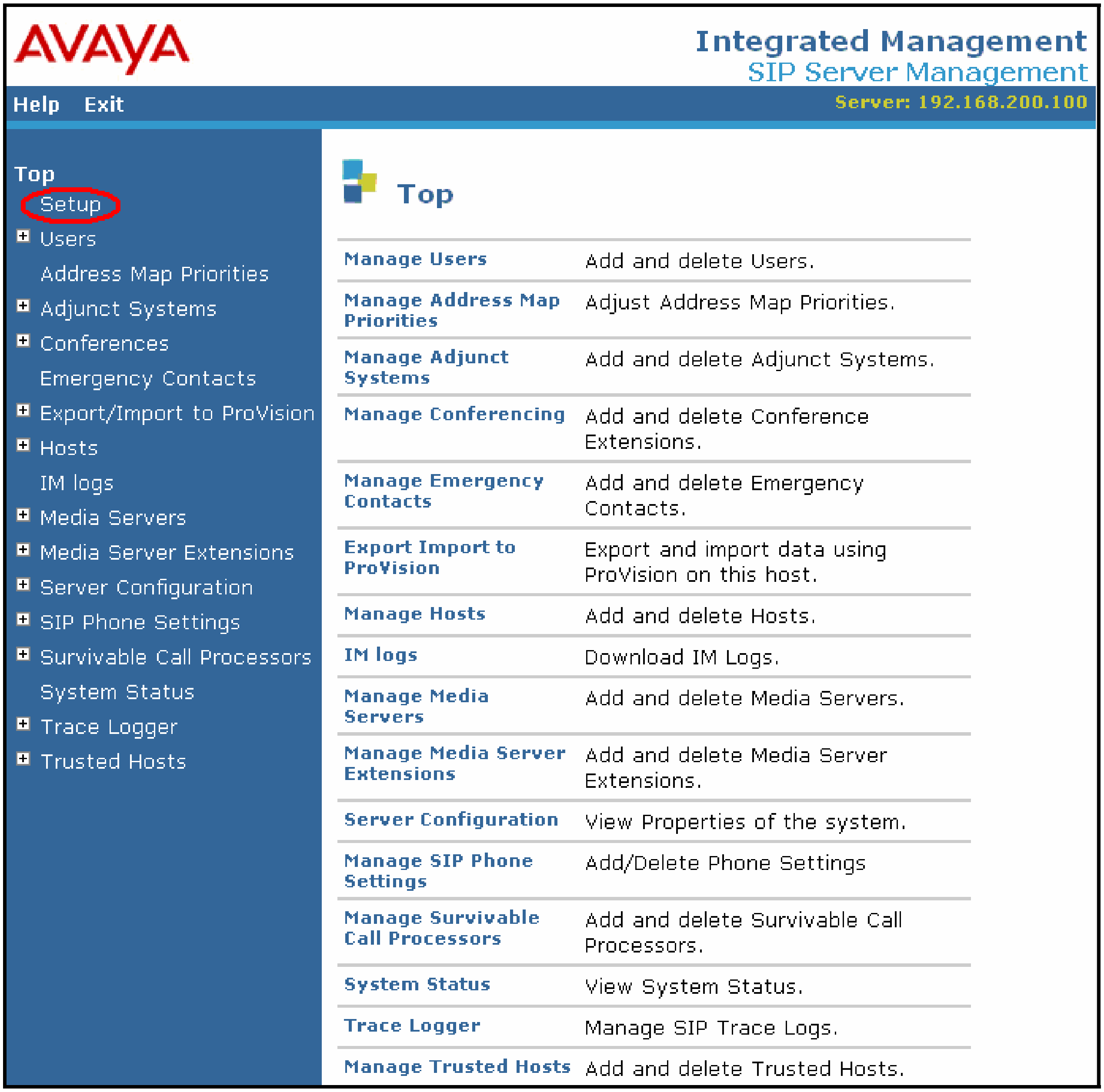

Figure 37: Initiate “Setup” from Top SES Configuration Screen

3.3.2. Setup Dataservice

Click “Setup Dataservice”.

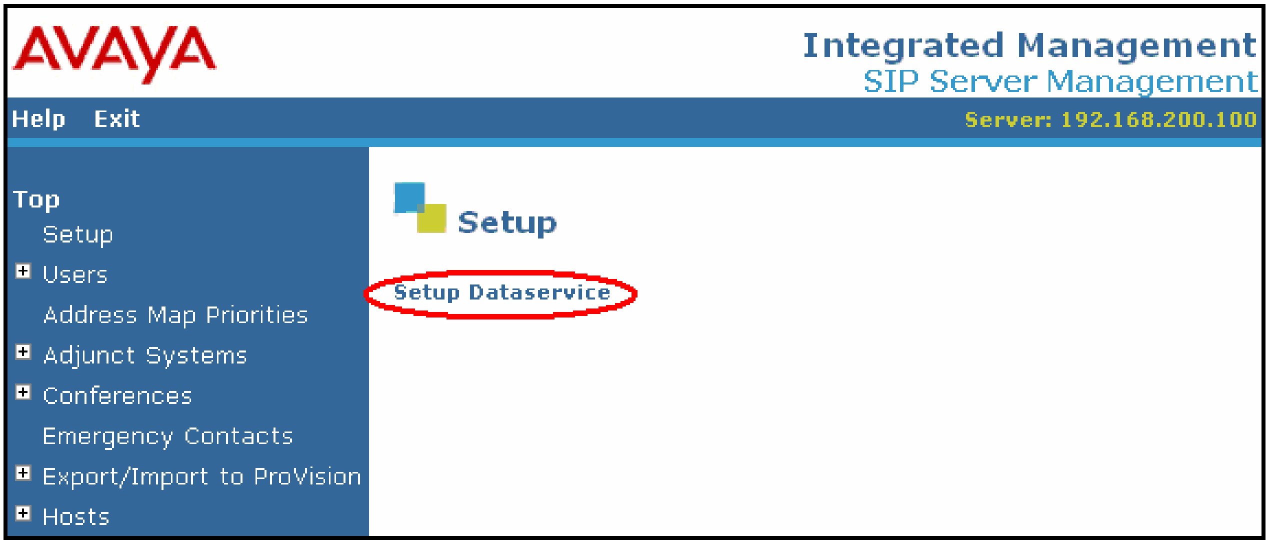

Figure 38: Initiate “Setup Dataservice” from Top Setup Screen

Select “This server is the SES Master Administration System for the SES Network”, and click “Setup”, and “Continue” for the screen that follows.

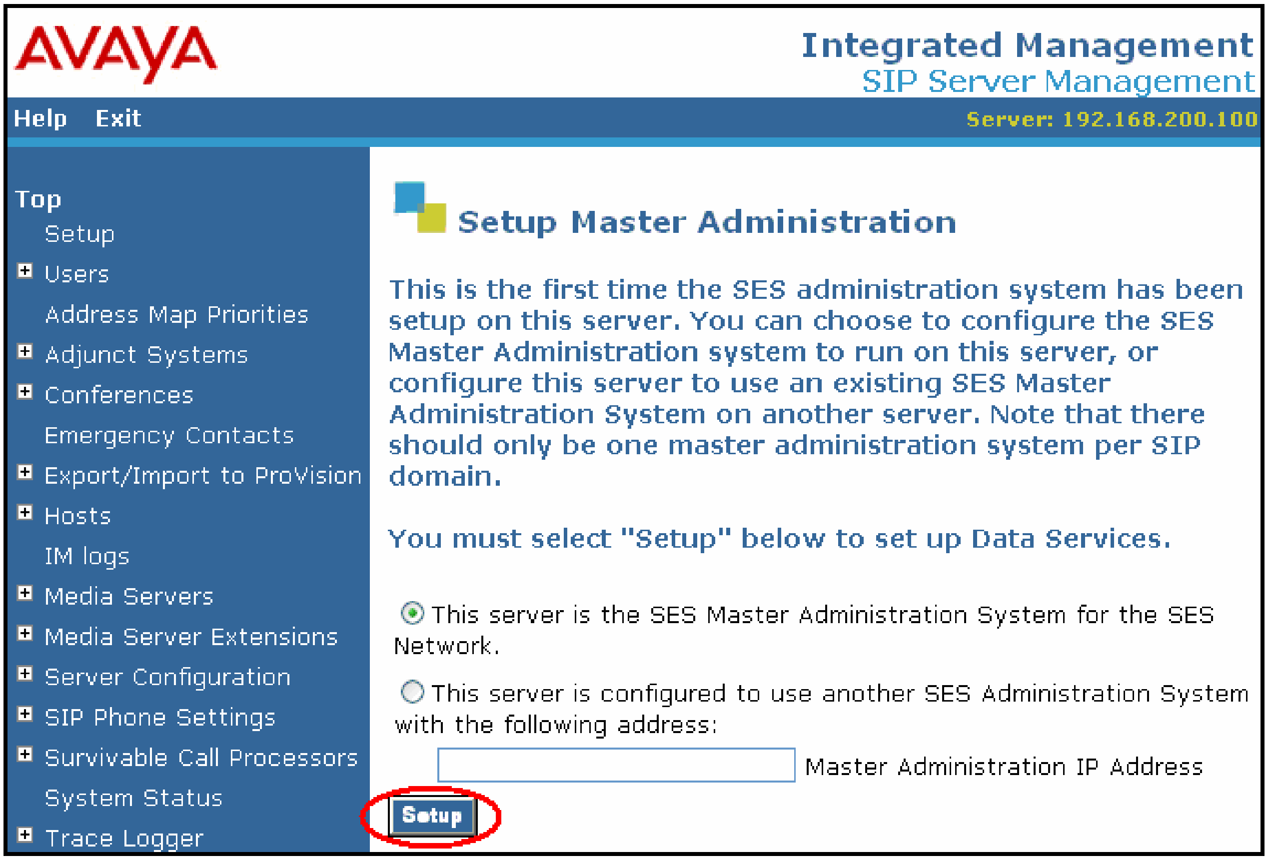

Figure 39: “Setup Master Administration” Screen

3.3.3. Setup SIP Domain

Click “Setup SIP domain”.

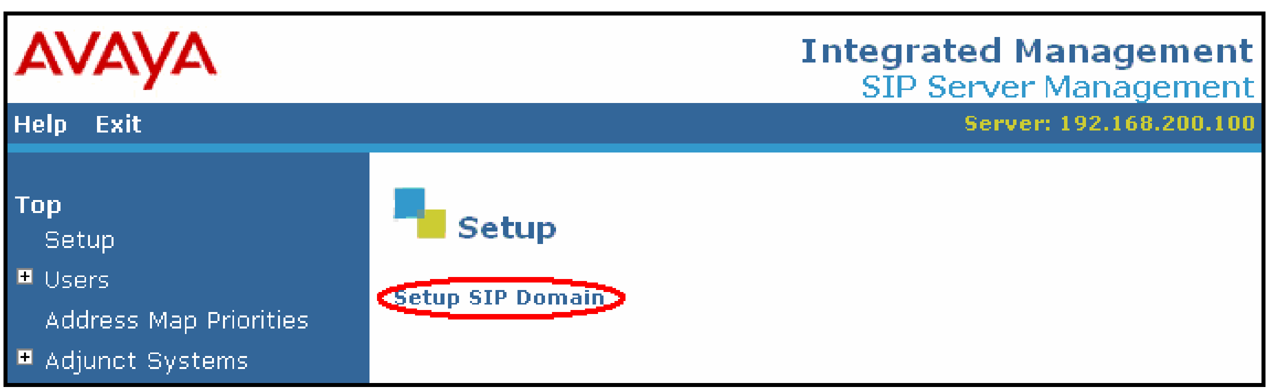

Figure 40: “Setup SIP Domain” Screen

Enter values in this screen as shown in the following table, and click "Update", followed by "Ok" for the following screen.

Parameter	Usage
SIP Domain	Enter the same value as was used for "Authoritative Domain" in **Figure 8**.
License Host	Enter the IP address of the license host, in this case the IP address of the SES server.

Table 18: Parameters for System Properties

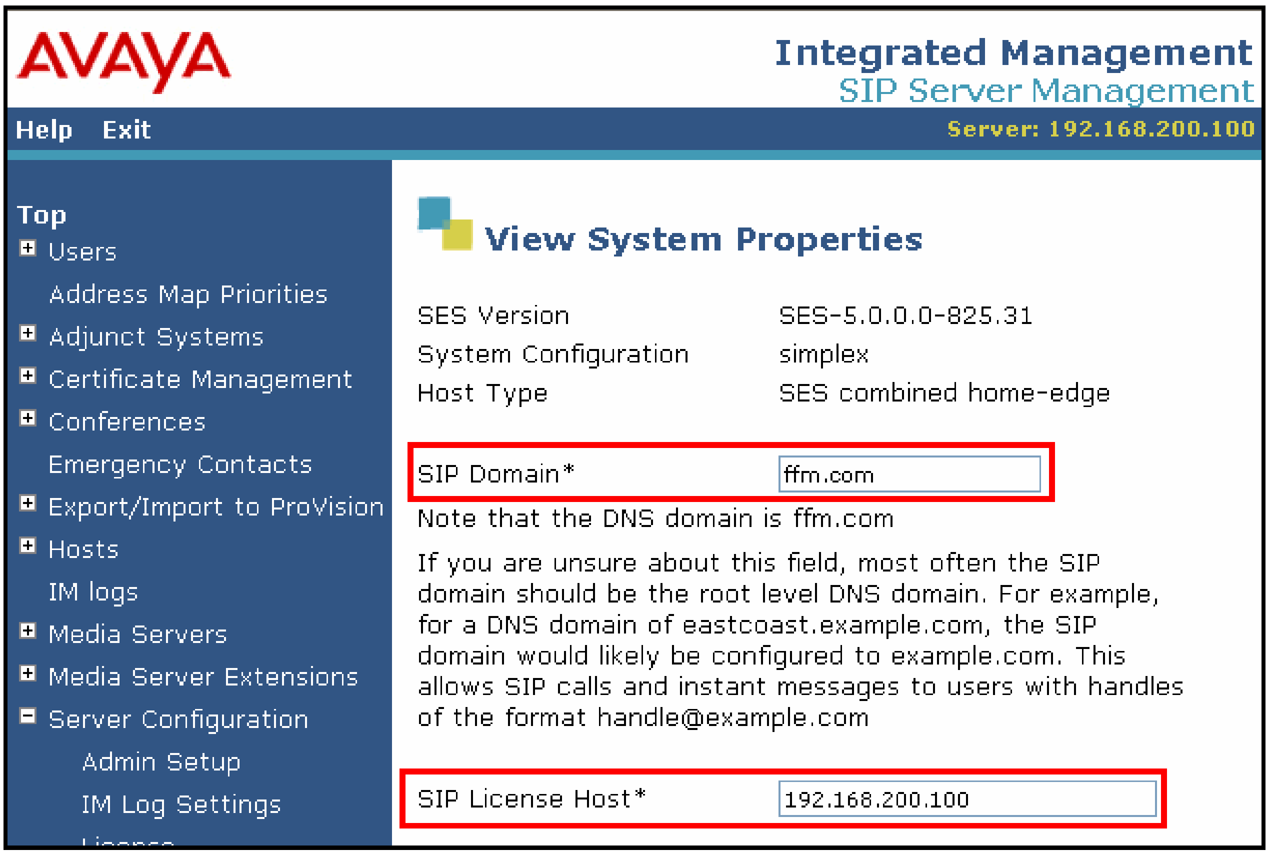

Figure 41: System Properties Screen

3.3.4. Add Media Server Interface

Navigate to **Media Servers → Add** from the "Top" level menu shown in **Figure 34**, and specify the interface parameters as shown in the following table.

Parameter	Usage
Media Server Interface Name	Enter a descriptive name for this interface.
SIP Trunk IP Address	Enter the IP address of the S8300 Server, or the address of the CLAN interface if an G650 gateway is used.
Media Server Admin Address	Enter the IP address of the S8300 Server administration interface.
Media Server Admin Login	Enter an administrator user ID for the media server.
Media Server Admin Password	Enter the password for the above user.

Table 19: "Add Media Server Interface" Parameters

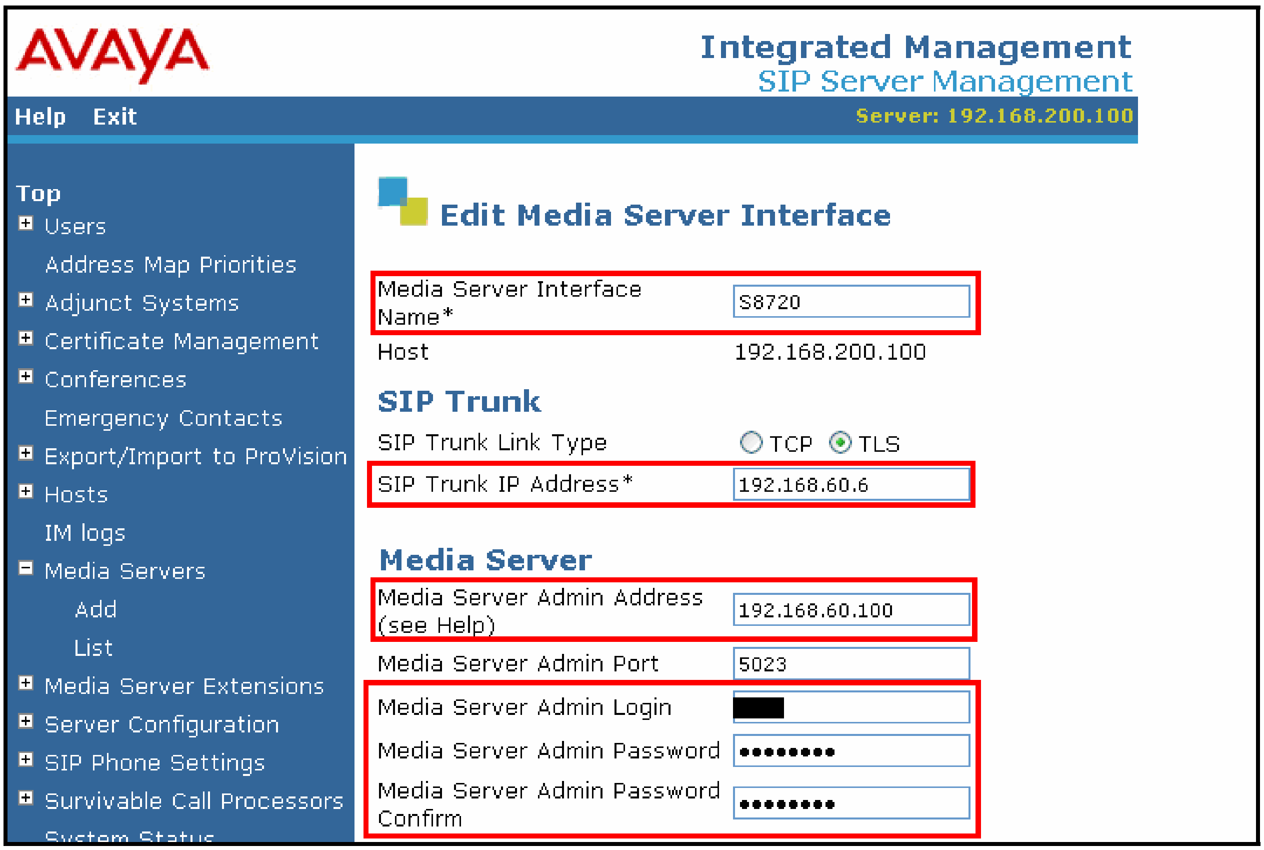

Figure 42: SES Add Media Server Interface Screen

3.3.5. Add Hosts

Navigate to **Hosts → Add Host** from the top level screen shown in **Figure 34**. Enter values in this screen as shown in the following table, accepting the default values for those parameters which are not listed. Click the "Add" button upon completion and the "Continue" button when the following screen is displayed.

Parameter	Usage
Host IP Address	Enter the IP address of the SES server.
Profile Service Password	Enter the password which was entered from the initial setup script when SES was installed.

Table 20: "Add Host" Parameters

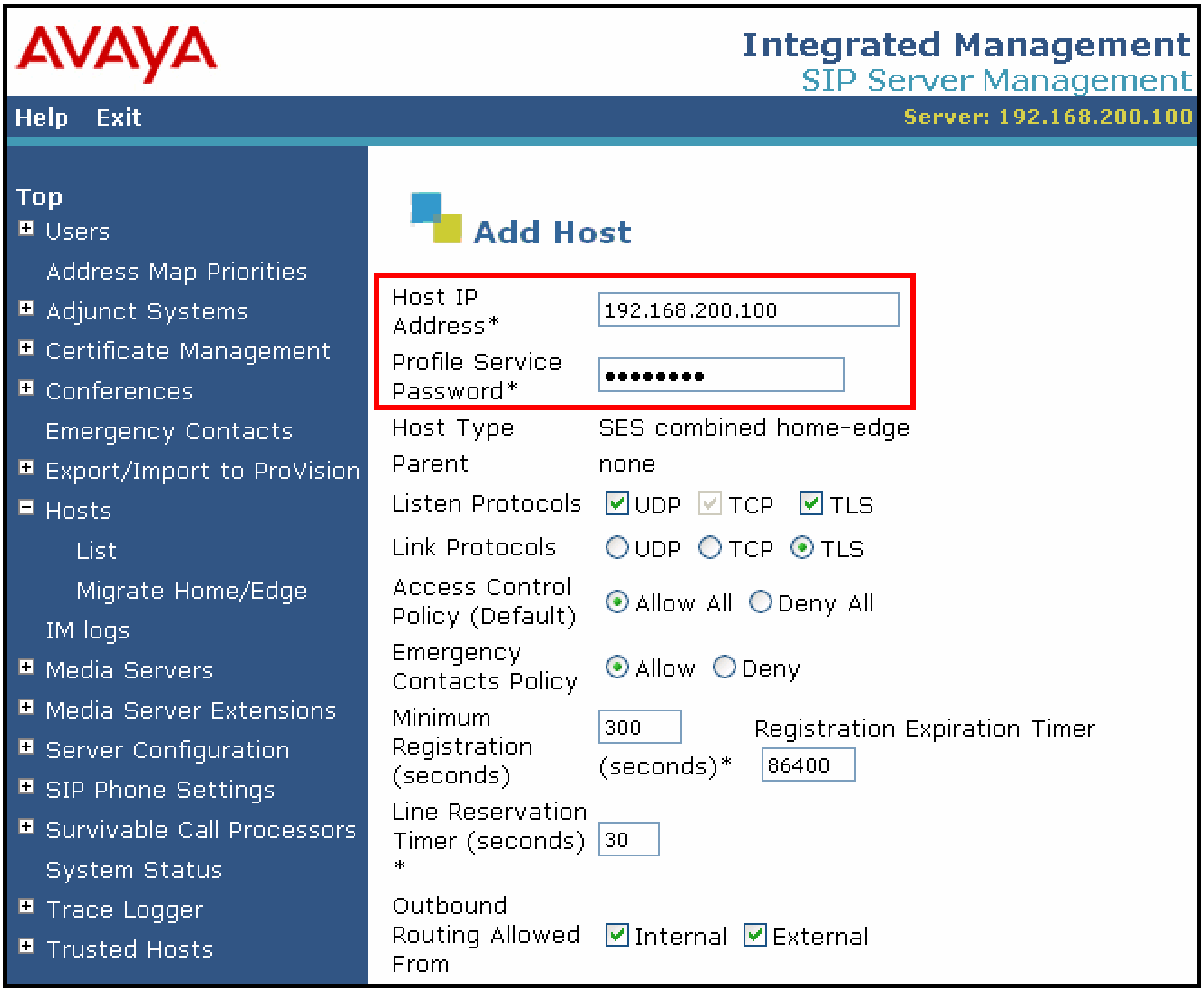

Figure 43: SES Add Host Screen

3.3.6. Add Users

From the “Users” menu in the left frame, click “Add”, and enter the parameters shown in the following screen, for each of the SIP telephones shown in **Table 1**.

Parameter	Usage
Primary Handle	Enter the extension to be assigned to the user.
User ID	Enter the extension to be assigned to the user.
Password / Confirm	Enter the password to be assigned to the telephone.
First / Last Name	Enter a name for identification purposes.
Add Media Server Extension	Check this box, to add an extension for this user.

Table 21: User Configuration Parameters

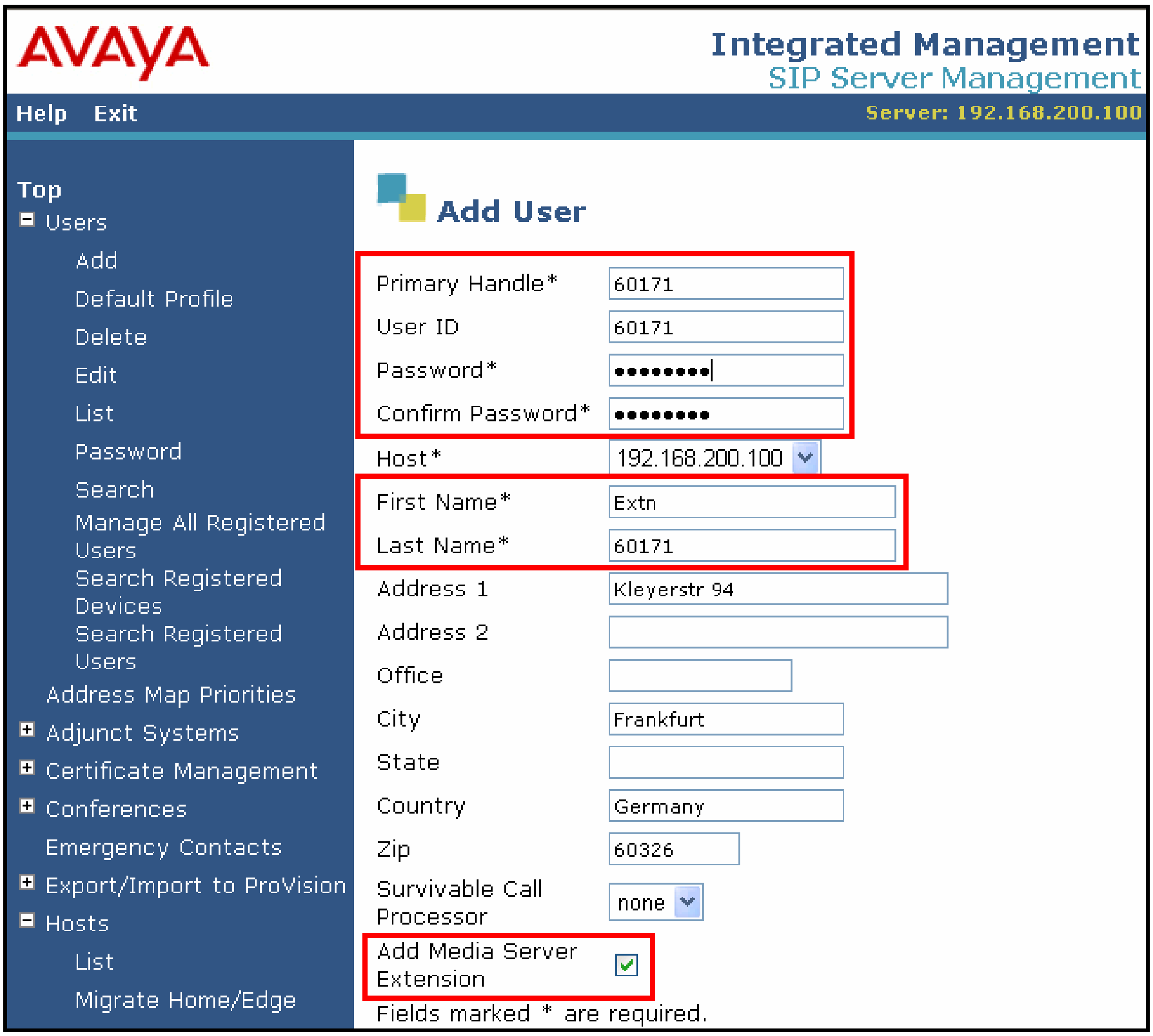

Figure 44: SES Add User Screen

The “Add Media Server Extension” screen will appear after the user has been added in the previous step. Enter the “Extension” for the SIP telephone from **Table 1** for the user which was created in the previous step, select the corresponding “Media Server” from the drop-down list, and click “Add”. Note that the extensions must match those which were allocated to the stations added in **Figure 14**.

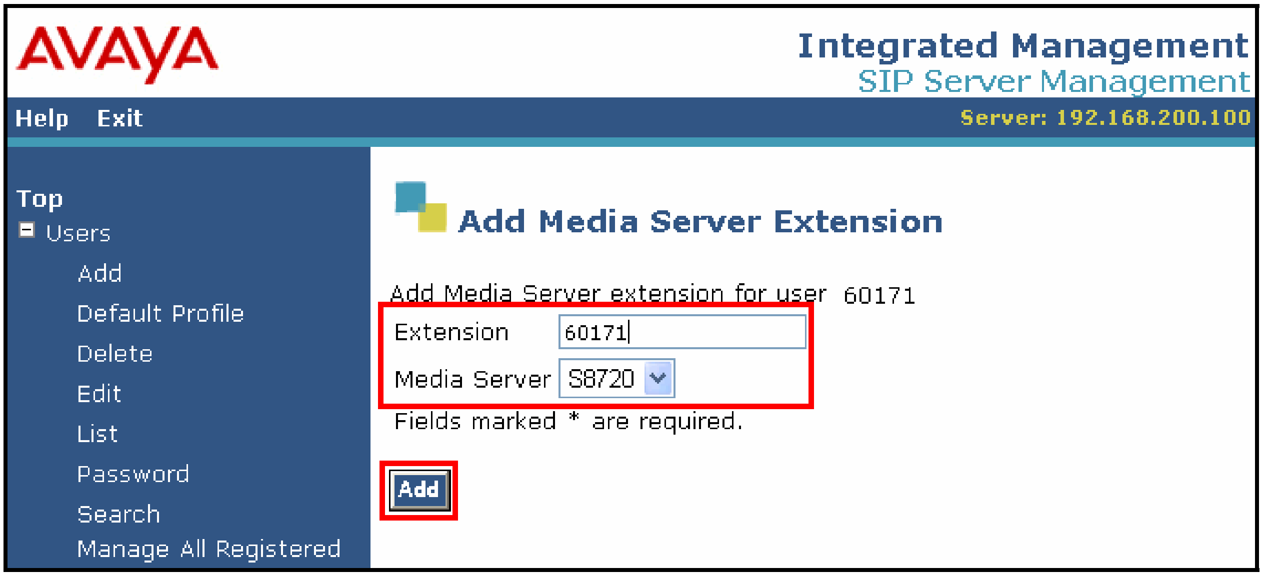

Figure 45: SES Add Media Server Extension Screen

3.4. Configure MARATHON EVOLUTION Server

The ASC MARATHON EVOLOUTION Voice Recorder has an integrated web server and can be configured remotely via a web browser by selecting its IP address as the target URL from the browser. Selection of this URL causes the following to be displayed.

The Marathon configuration tool is available by selecting the **ASC DataManager** application on the right.

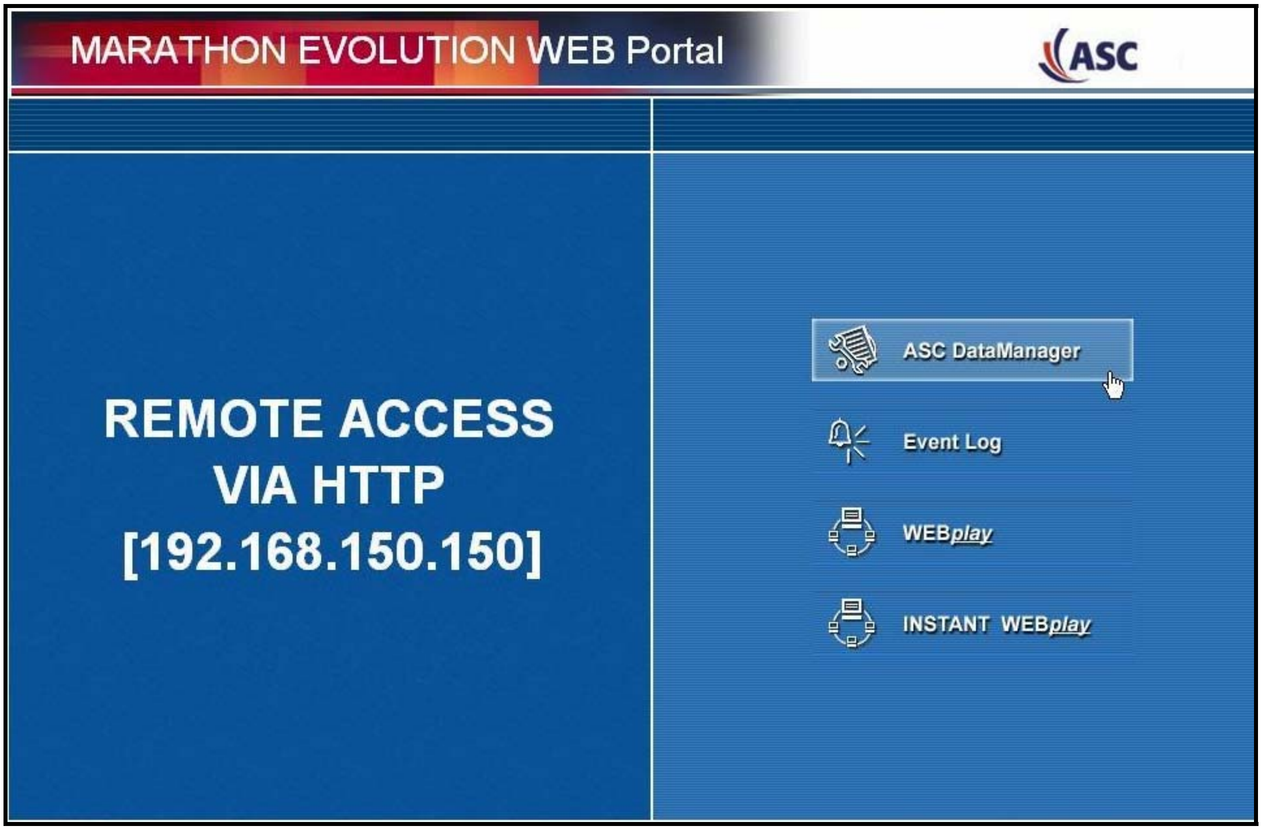

Figure 46: MARATHON EVOLUTION Welcome Screen

Click to expand the “ASC Datamanager” -> “Configuration” ->“System” menu item in the left frame of the screen.

Figure 47: DataManager Welcome Screen

Select “AvayaInterface” from the “Select module” drop-down menu.

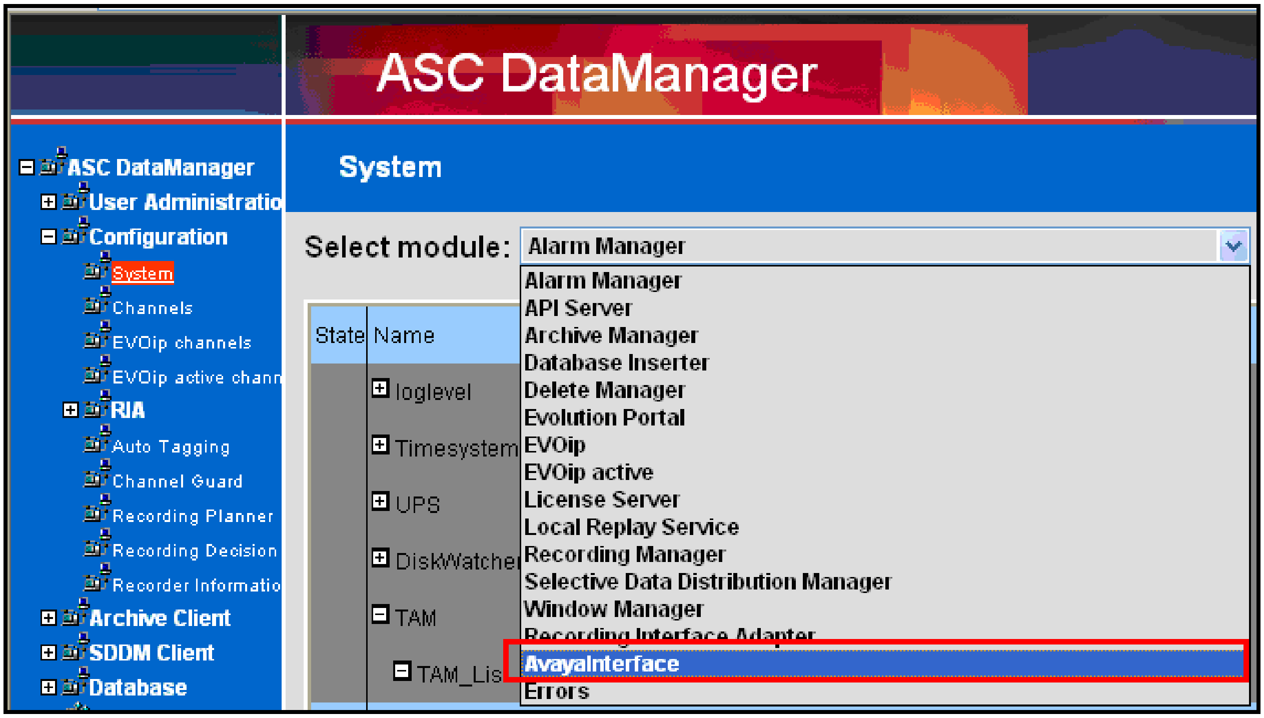

Figure 48: DataManager Select AvayaInterface Mode

Set “State” parameters as described in the following table.

Parameter	Usage
Activate	Set this parameter to “Yes”.
OperationMode / CTI	Set this parameter to “Yes” for Single Step Conference operation.

Table 22: DataManager AvayaInterface Operation Mode Parameters

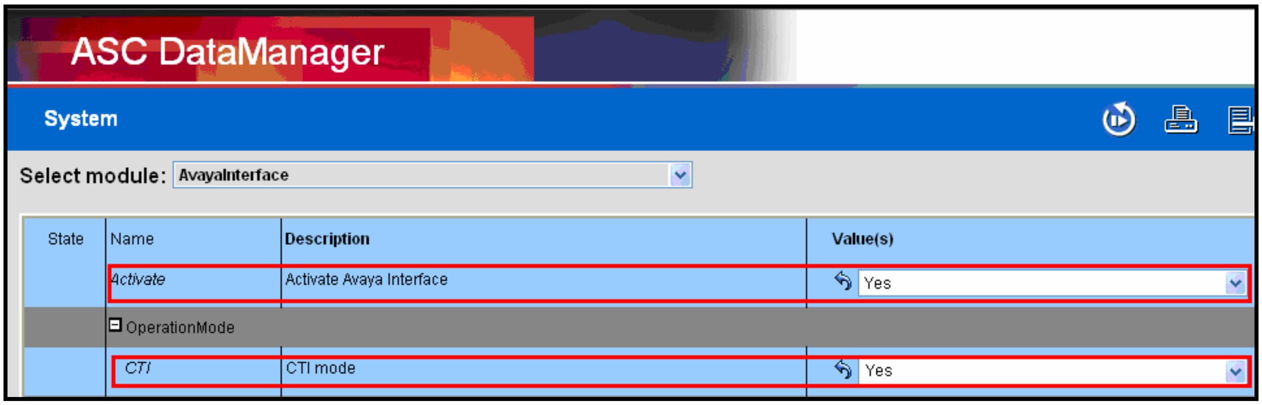

Figure 49: DataManager AvayaInterface Operation Mode

Configure the "AvayaInterface" "Communication Manager" parameters as shown in the following table.

Parameter	Usage
IP	Enter the IP address of the Communication Manager clan interface as configured in **Figure 7**.
Name	Enter the name which was assigned to the switch connection in **Figure 26**.

Table 23: Configuration IP Stations

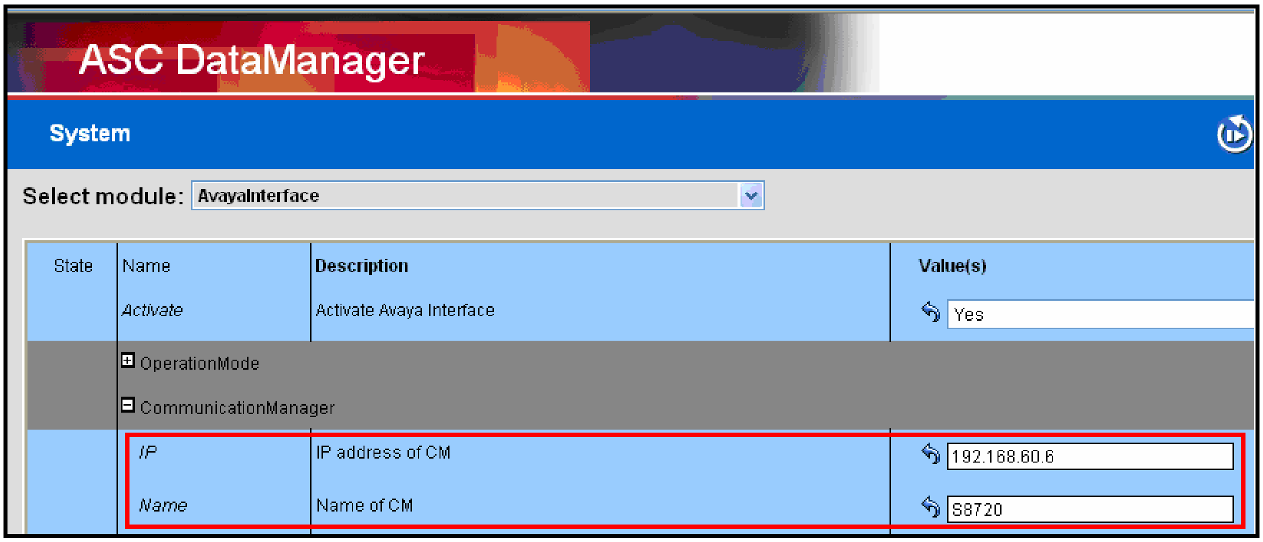

Figure 50: DataManager AvayaInterface Communication Manager

Configure the "AvayaInterface" "AES Server" parameters as shown in the following table.

Parameter	Usage
IP	Enter the IP address of the AES Server.
PortEncrypted	Enter the same value which is specified in **Figure 32**.
PortUnencrypted	Enter the same value which is specified in **Figure 32**.
Secure	Enter "No".
User	Enter the same value which is specified in **Figure 31**.
Password	Enter the same value which is specified in **Figure 31**.

Table 24: DataManager AES Server Interface Parameters

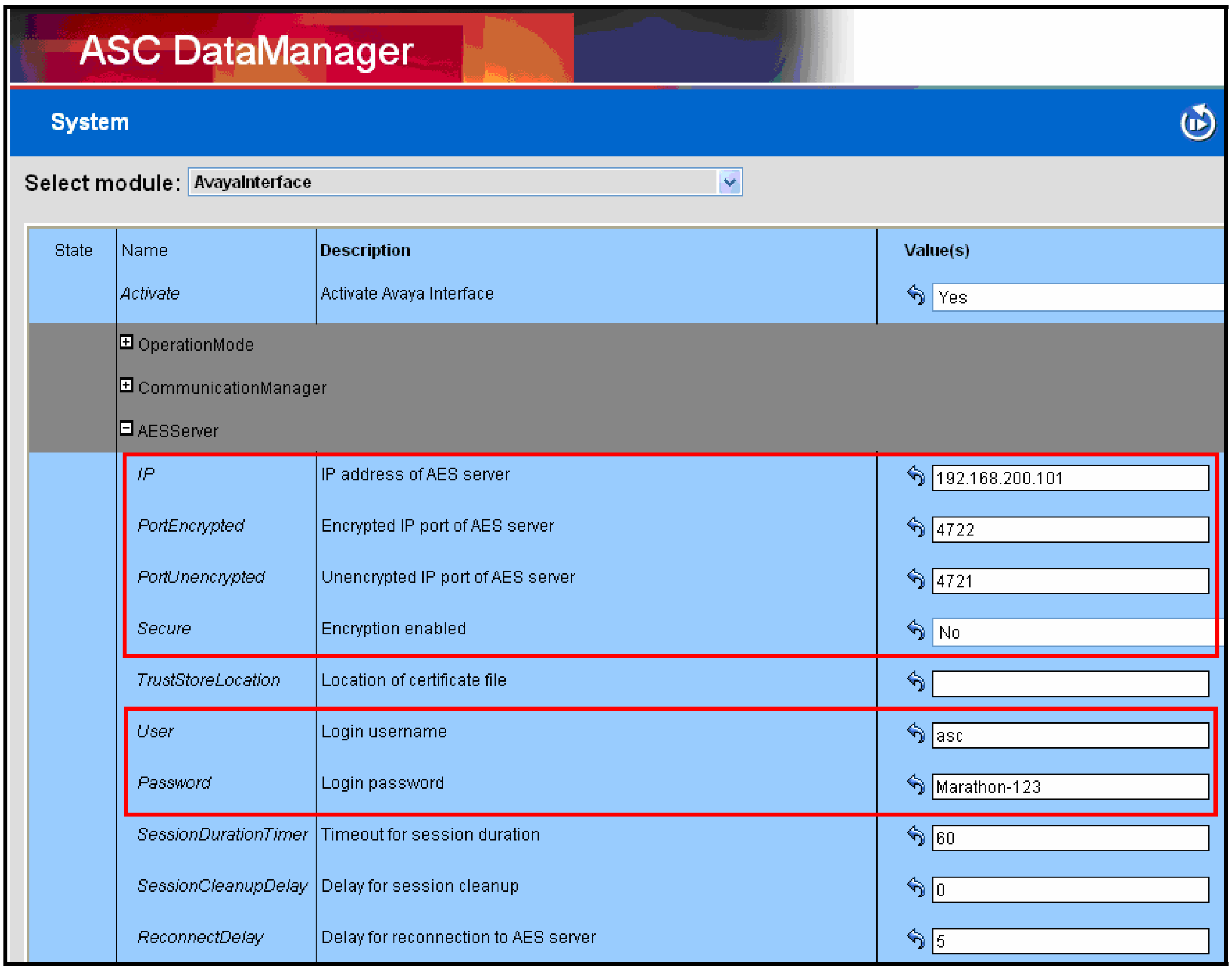

Figure 51: DataManager AES Server Interface Screen

Configure the "AvayaInterface" "Softphones" parameters as shown in the following table.

Parameter	Usage
RangeStart	Enter the extension of the first Virtual CTI Station which was allocated for monitoring, as defined in **Figure 17.**
RangeLen	Enter the number of Virtual CTI Stations used for monitoring.

Table 25: DataManager Softphones Parameters

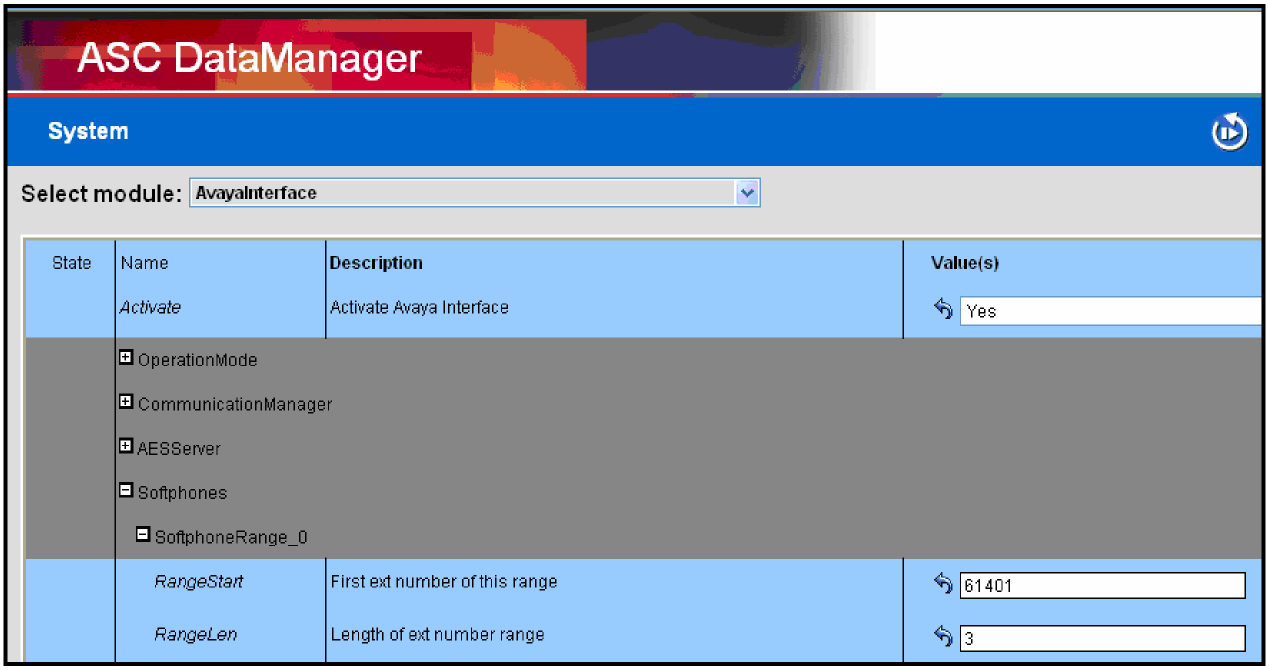

Figure 52: DataManager Softphones Screen

Configure the "AvayaInterface" "Recorder" parameters as shown in the following table.

Parameter	Usage
IP	Enter the IP address of the MARATHON EVOLUTION server.
ReconnectDelay	Enter "15".

Table 26: DataManager Recorder Parameters

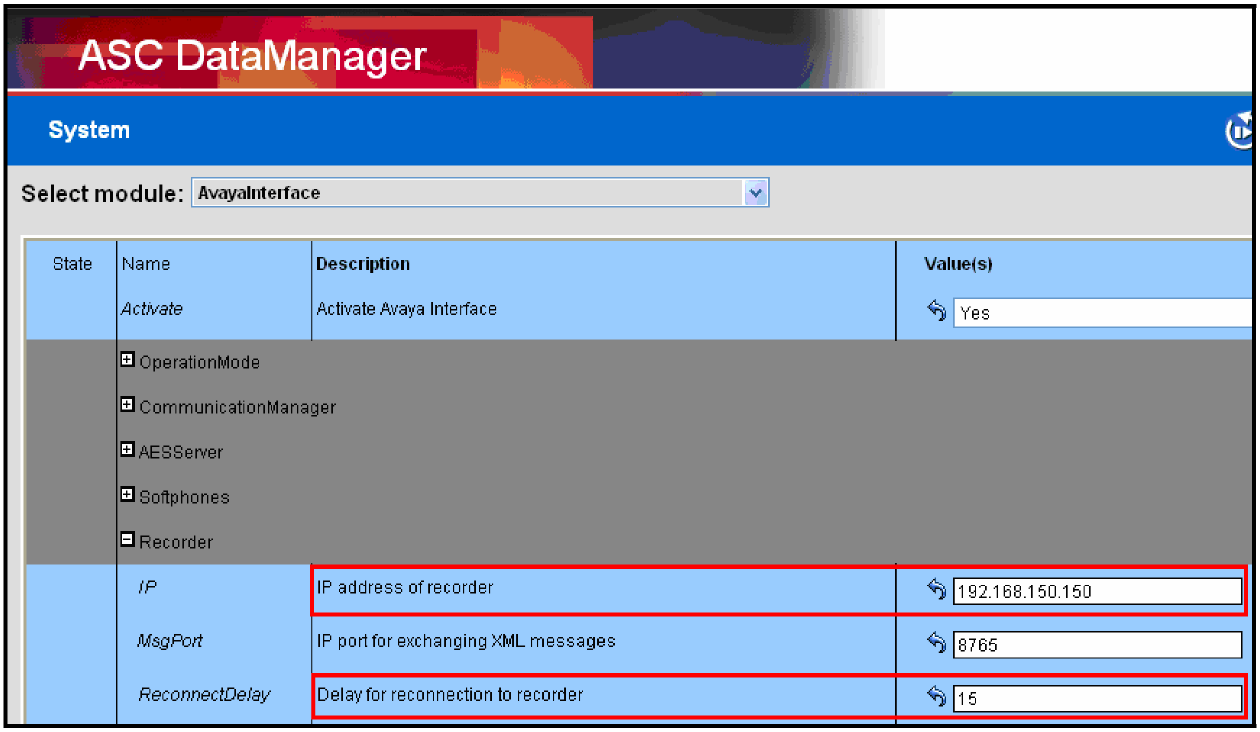

Figure 53: DataManager Recorder Screen

From the DataManager top-level menu, select the “EVOip” module.

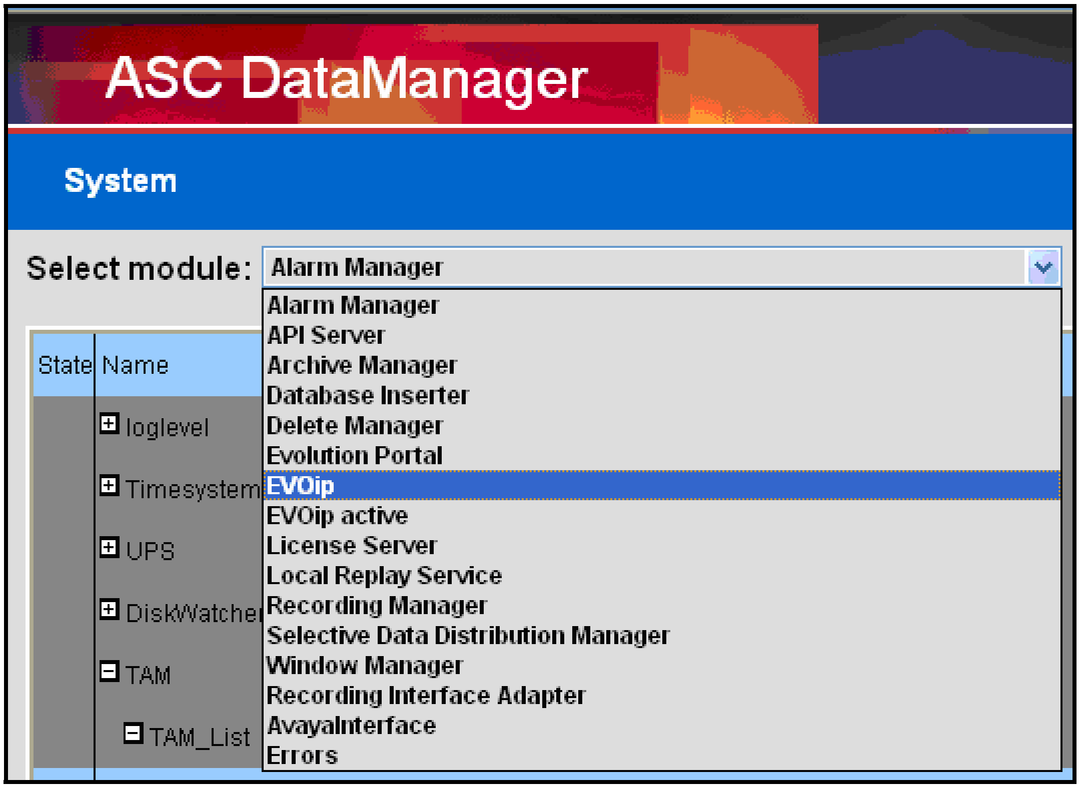

Figure 54: Select EVOip Screen

Set the “Calldata” “EnableCallTagging” parameter to “No” for SSC operation.

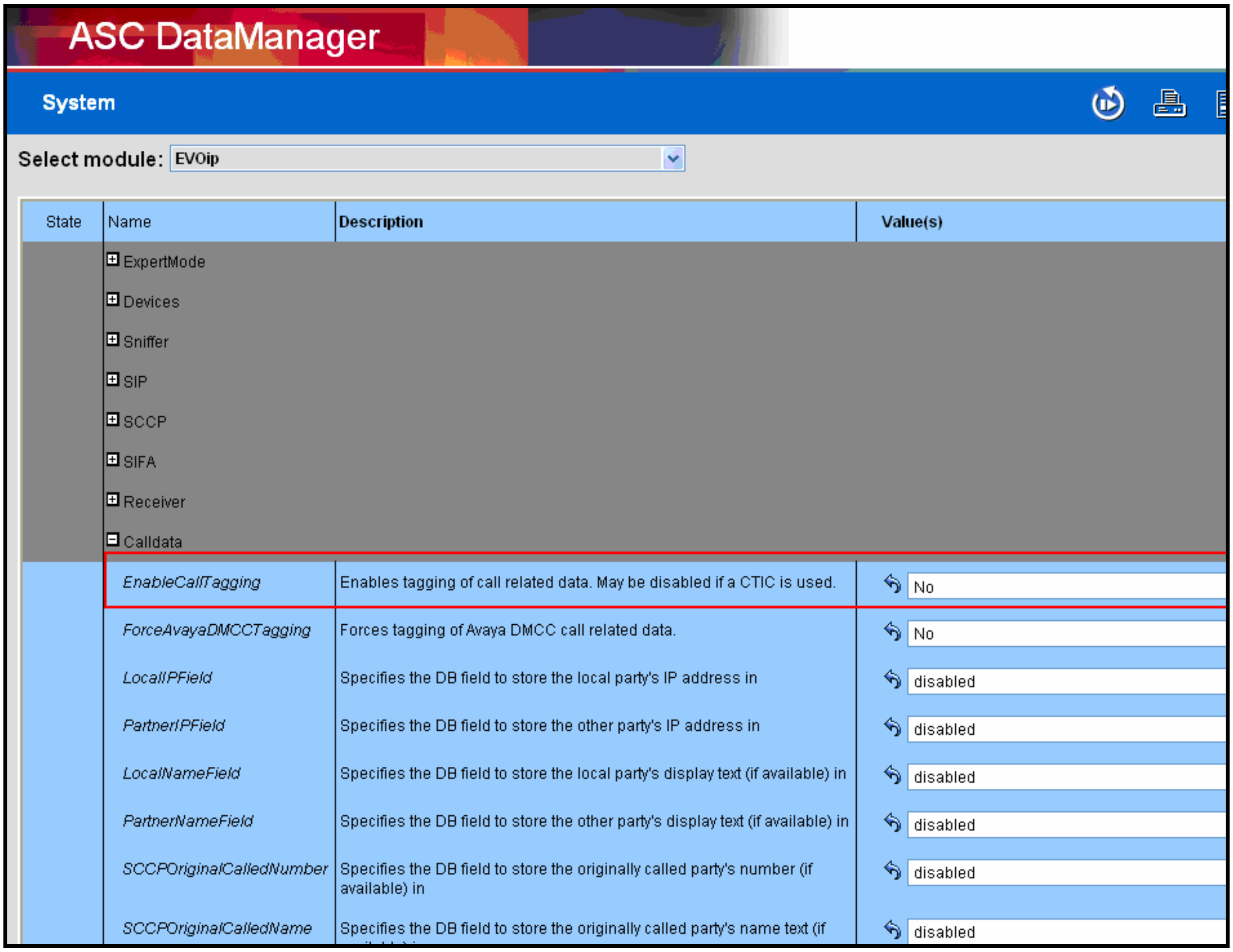

Figure 55: DataManager EVOip Calldata Screen

Select the EVOip channels menu point from the main menu.

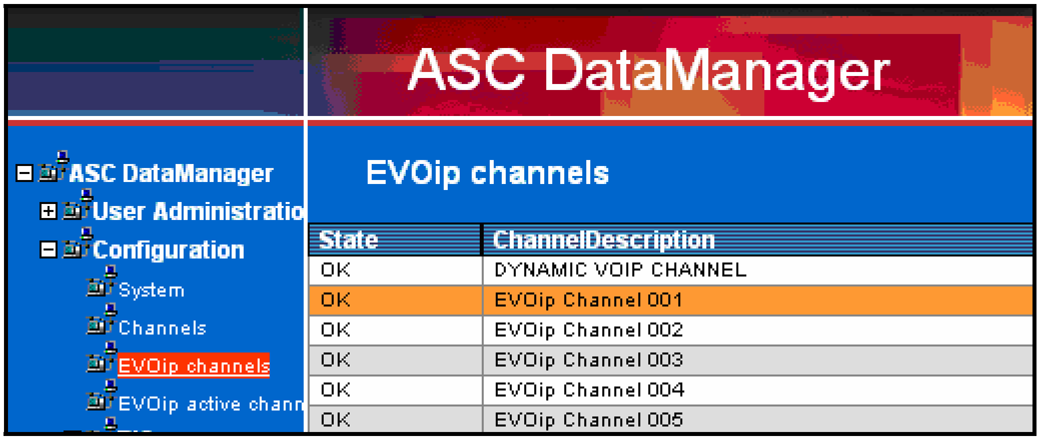

Figure 56: DataManager Channels Screen

Configure each of the allocated channels as shown in the screen below. Repeat this for the other extensions in **Table 1** which are to be monitored by channels 2 and 3.

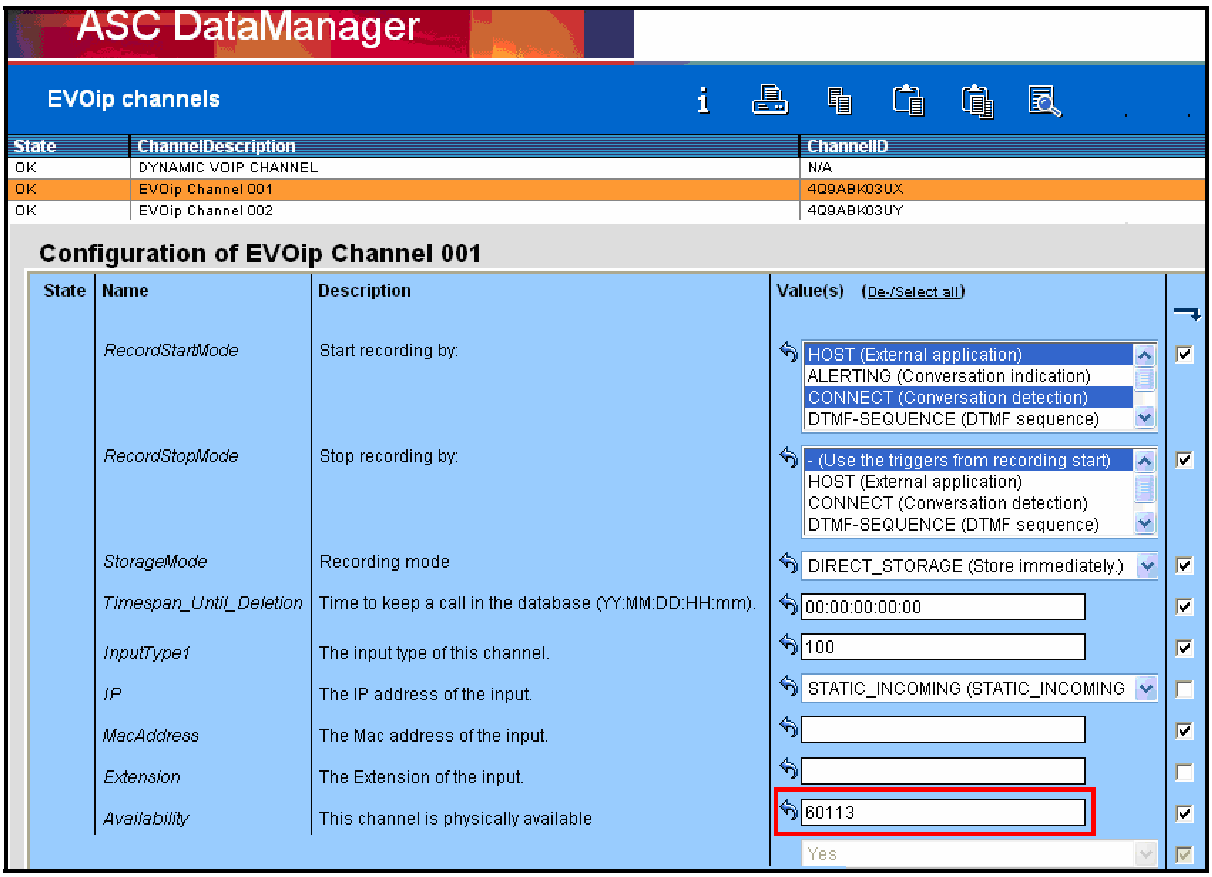

Figure 57: EVOip Channel Configuration Screen

Select “Configuration” -> “RIA” -> “RIAactive” from the left frame, and enter the parameters shown in the following table.

Parameter	Usage
NumInstances	Enter “1”.
Enabled	Set this parameter to “Yes”.
Name	Accept the default name.
LogLevel	Accept the default log level of “Medium”.

Table 27: DataManager RIA*active* Parameters

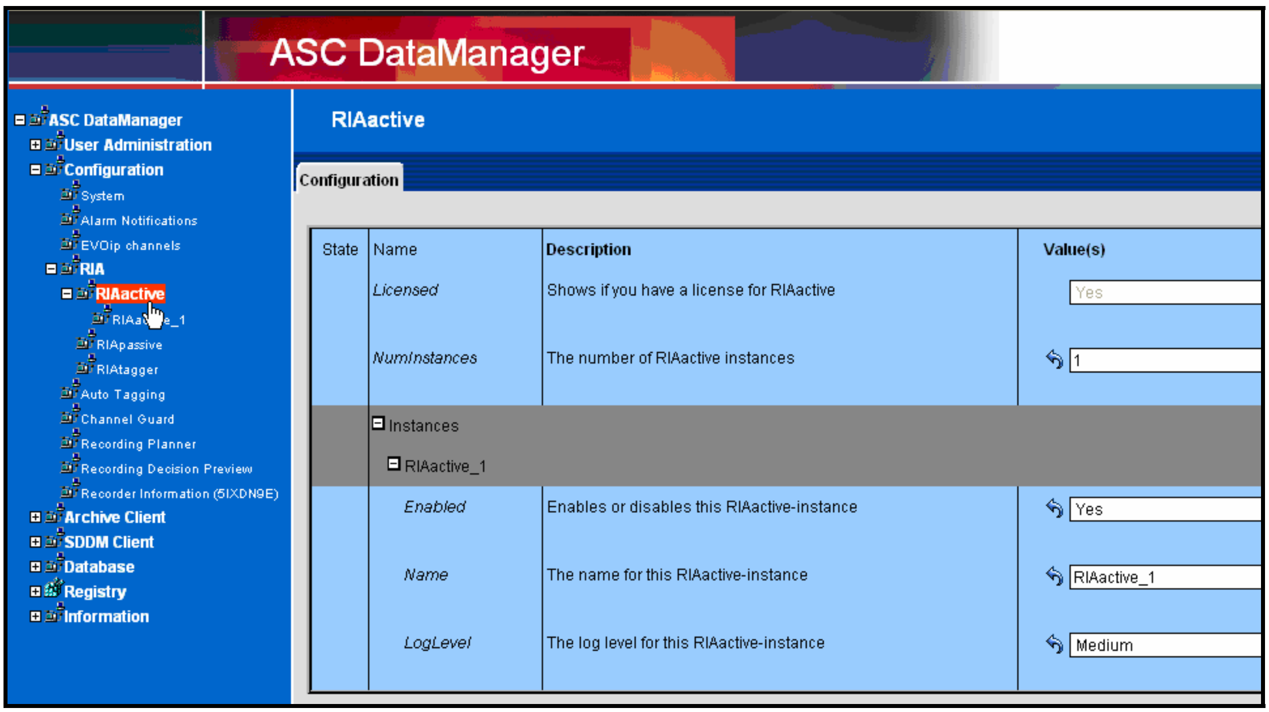

Figure 58: DataManager RIA*active* Screen

Select “Configuration” -> “RIA” -> “RIAactive” -> “RIAactive _1” from the left frame, and enter the parameters shown in the following table.

Parameter	Usage
Statemachine	Enter “RIAServer”.
EventBuffer	Enter “1000 Events”.
PIF-Type	Enter “PIFAvayaCM”.
Connection-Type	Enter “TCP”.
Server-IP	Enter the address assigned to the RIA*active* server.
Server-Port	Enter the port number assigned in **Figure 65.**
Inactivity-Timeout	Enter “60”.
LogBinaryData	Enter “No”.

Table 28: DataManager RIAactive_1 Parameters

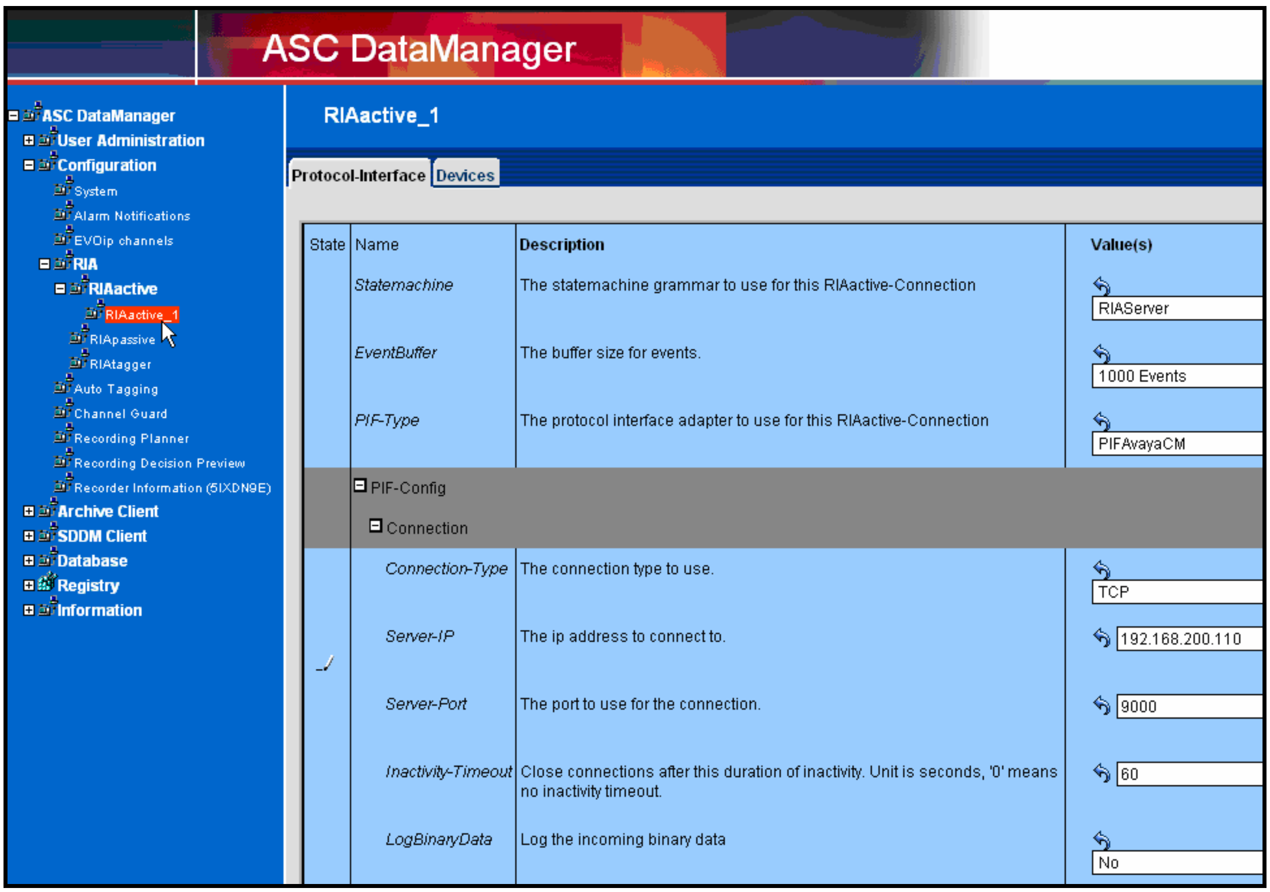

Figure 59: DataManager RIAactive_1 Protocol Interface Screen

Select “Configuration” -> “RIA” -> “RIAactive” -> “RIAactive_1” from the left frame, select the “Devices” tab, and set the “PhysicalPbxID” for “EVOip Channel 001” through “EVOip Channel 003” to “active”.

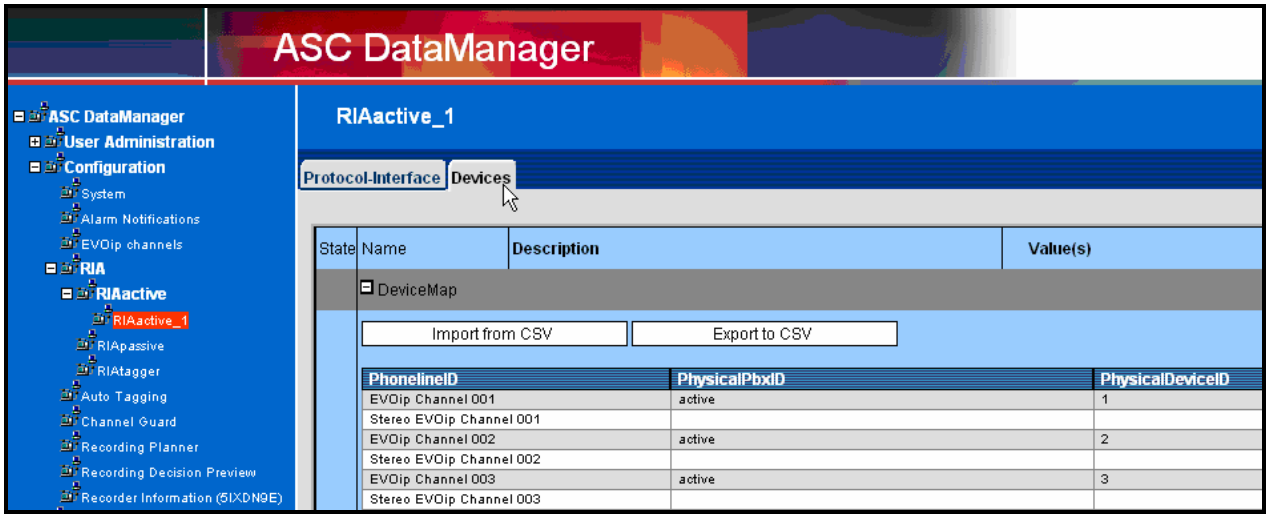

Figure 60: DataManager RIAactive_1 Devices Screen

3.5. Configure Marathon RIA*active*

Install the RIA*active* software from the distribution medium, and reboot the system. Access the RIA*active* server via web browser either from the local system, or remotely and log in with the appropriate user name and password.

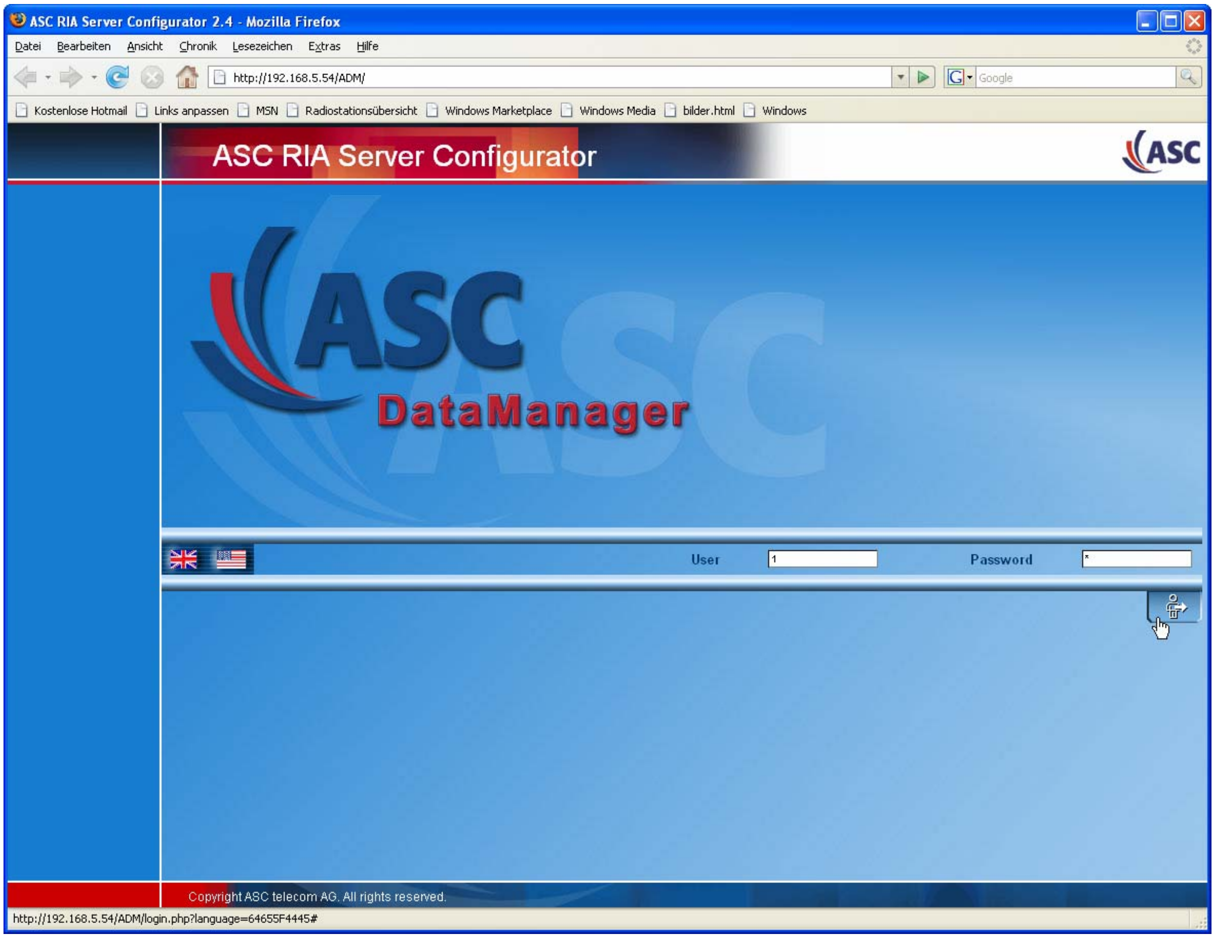

Figure 61: RIA*active* Login Screen

Select “Avaya CM” -> “Avaya PBX” from the left frame of the screen, and enter the parameters shown in the following table.

Parameter	Usage
PBX Name	Enter a name for identification purposes.
Hostname	Enter the name allocated to Communication Manager by AES, which is composed of the four text strings in **Table 30**, which are separated from another with a “#” character.
Port	Enter “1”.
Login	Enter the user name assigned by AES in **Figure 31**.
Password	Enter the password name assigned by AES in **Figure 31**.

Table 29: RIA*active* PBX Parameters

Strings	Usage
AVAYA	This is a fixed value.
S8720	This is the name that was assigned to the Switch Connection, which was assigned on the Avaya AES Administration -> Switch Connections screen (see **Figure 26**).
CSTA	This is a fixed value.
AES-SERVER1	This is the name that was assigned to the Avaya AES server when the Avaya AES software installation was performed.

Table 30: Composition of the “hostname” Parameter

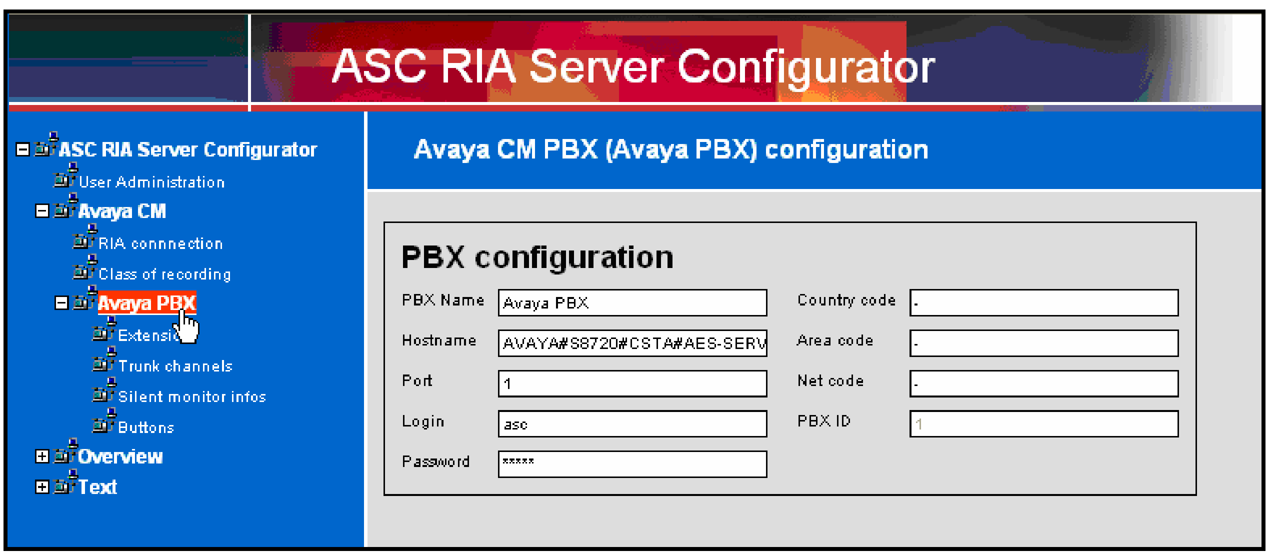

Figure 62: RIA*active* PBX Configuration Screen

For each of the extensions to be monitored, Select “Avaya CM” -> “Avaya PBX” -> “Extensions” from the left frame of the screen, and enter the parameters shown in the following table.

Parameter	Usage
Extension type	Select “Extension”.
Language	Select the language used by the user of the extension.
Class of recording	Select “Standard class of recording”.
ExtensionNumber	Leave the default extension set to the default value.
Monitorpoint enabled	Set this parameter to “yes”.

Table 31: RIA*active* PBX Parameters

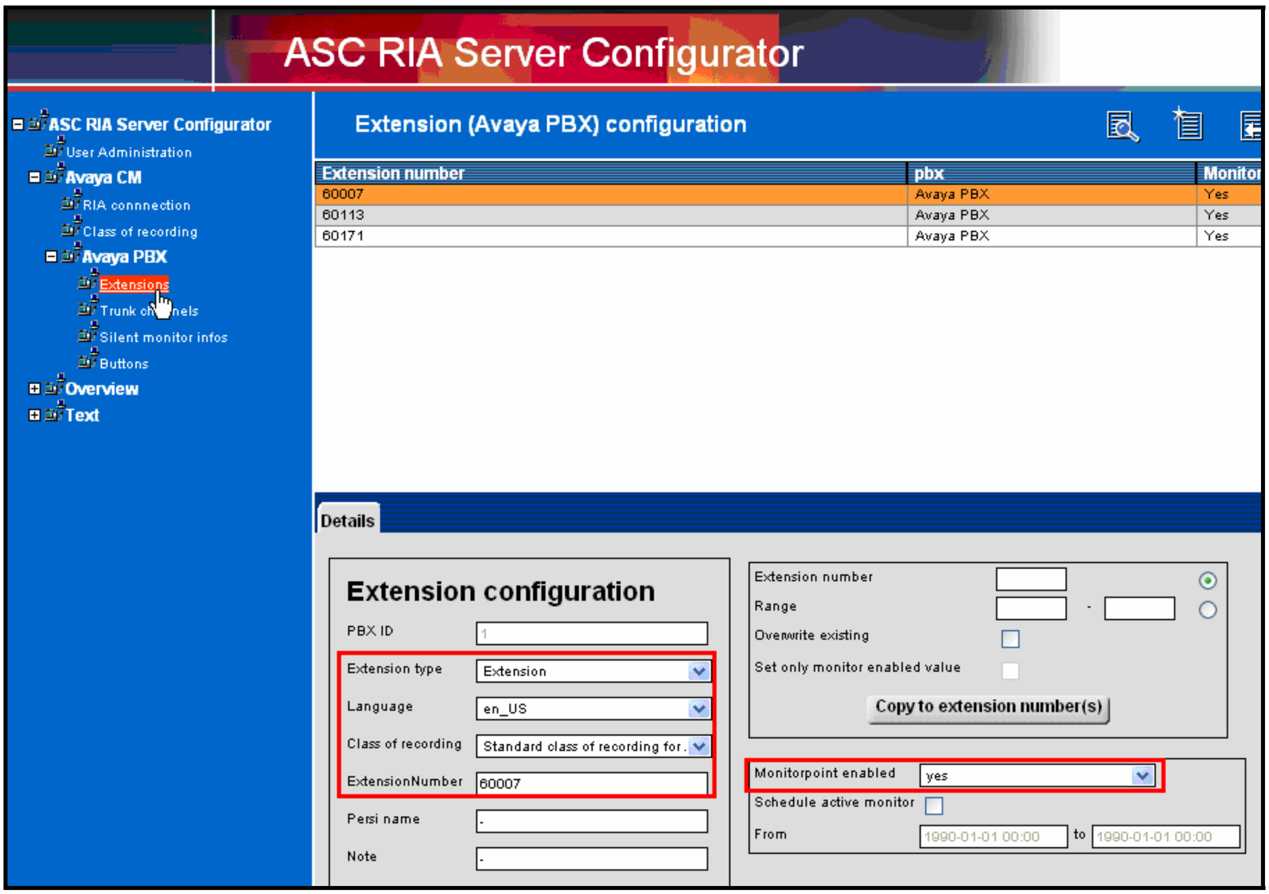

Figure 63: RIA*active* Extension Configuration Screen

For each of the extensions to be monitored, Select “Avaya CM” -> “Avaya PBX” -> “Silent monitor infos” from the left frame of the screen, and enter the parameters shown in the following table.

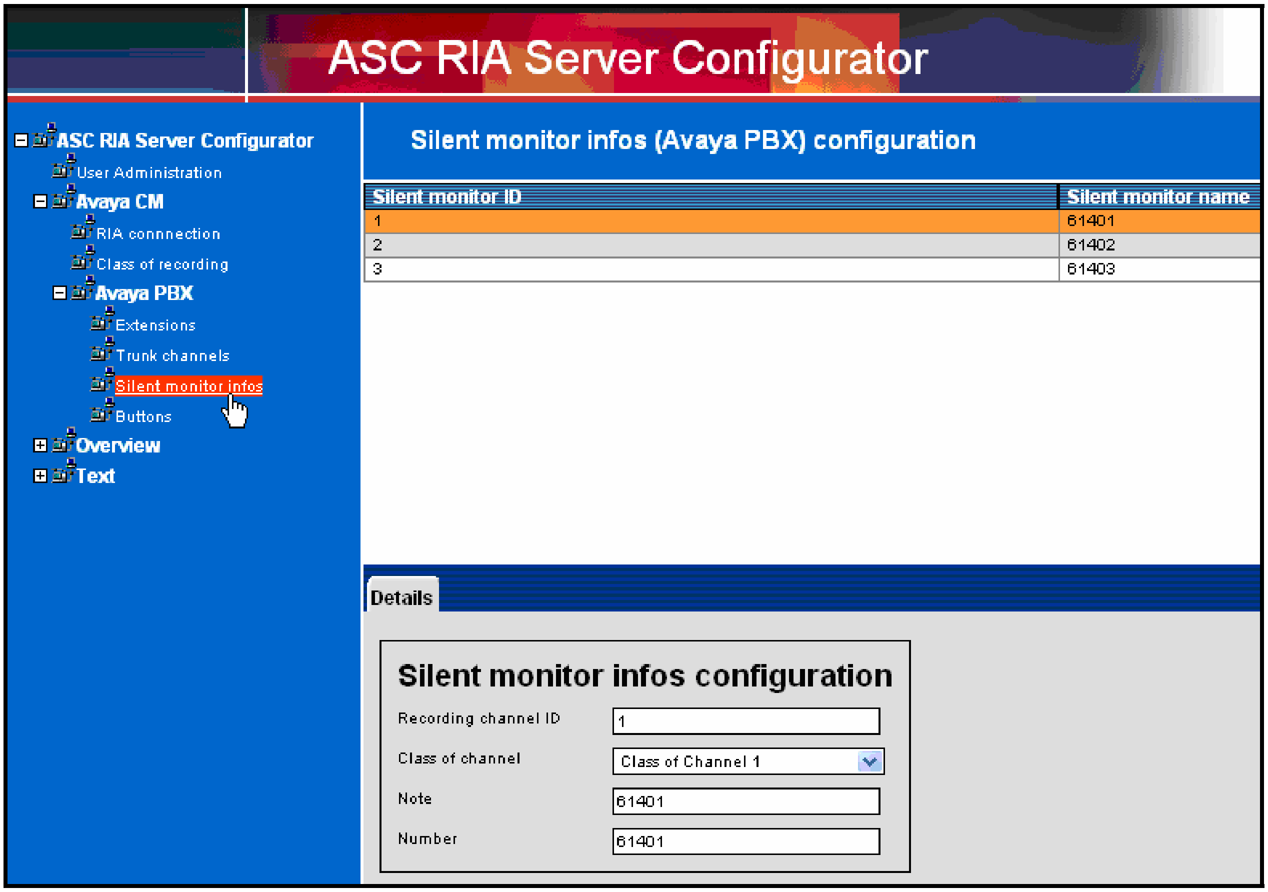

Figure 64: RIA*active* Silent Monitor Configuration Screen

Select “Avaya CM” -> “RIA connect” from the left frame of the screen, and set the ServerPort to an otherwise unused value. This must be the same value set in **Figure 59**.

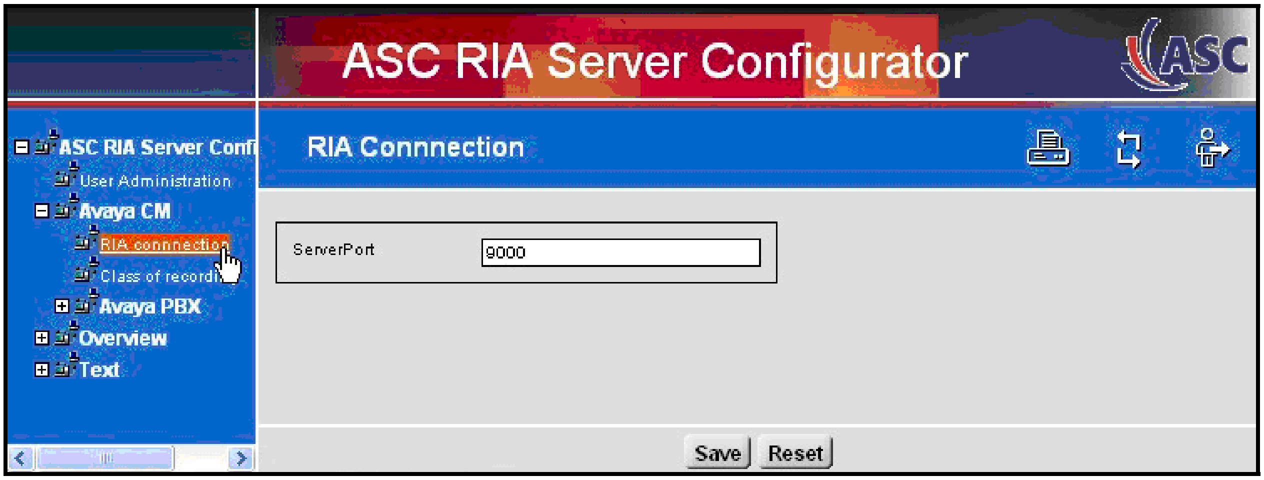

Figure 65: RIA*active* Server Port Screen

4. Interoperability Compliance Testing

The objective of the compliance testing done on the ASC telecom MARATHON EVOLUTION product was to verify that it is compatible with Avaya Communication Manager. This includes verifying that the essential MARATHON EVOLUTION features function properly when used with Avaya Communication Manager, and that Avaya Communication Manager features are not hindered by the interaction with MARATHON EVOLUTION. Furthermore, MARATHON EVOLUTION's robustness was verified.

4.1. General Test Approach

The test method employed can be described as follows:

- Avaya Communication Manager was configured to support various local IP telephones, as well as a networked PBX connection and a PSTN connection.
- A simulated PSTN interface was attached to Avaya Communication Manager, equipped with telephones that were used to simulate external callers.
- The MARATHON EVOLUTION was configured to monitor various telephones and trunks attached to Avaya Communication Manager.
- The major MARATHON EVOLUTION features and functions were verified using the above-mentioned local and external telephones, including the ability to monitor
  - Locally attached IP and digital telephones
  - Telephones attached to the PSTN
  - Telephones attached to a networked PBX
- The following MARATHON EVOLUTION methods were verified in these tests:
  - Single step conferencing
- The following test scenarios were used to test the various MARATHON EVOLUTION features:
  - Basic call
  - Hold/retrieve
  - Transfer
  - Blind transfer
  - Conferencing
  - Hunt group calls
- MARATHON EVOLUTION's robustness was tested by verifying its ability to recover from interruptions to its external connections including:
  - The LAN connection between and the MARATHON EVOLUTION and the network
  - The LAN connection between and the ASC RIA*active* and the network
- MARATHON EVOLUTION's robustness was further tested by verifying ability to recover from power interruptions to the following components:
  - The MARATHON EVOLUTION server
  - The ASC CTI Controller
  - The Avaya Communication Server to which the MARATHON EVOLUTION is attached.

All testing was performed manually. The tests were all functional in nature, and no performance testing was done.

4.2. Test Results

All tests which were performed produced the expected result.

5. Verification Steps

The following steps can be performed to verify the correct installation and configuration of MARATHON EVOLUTION:

- Verify that the Avaya AES and MARATHON EVOLUTION systems can ping each other.
- Verify that the various telephones can call each other.
- Log into the Avaya AES as described in Section 3.2 and perform the following:
    - Verify that CTI OAM Status and Control “Switch Connection Summary” shows that the connection between Avaya AES and Avaya Communication Manager is operational.
    - Verify that CTI OAM Status and Control “Services Summary” shows that TSAPI service is operational.

6. Support

Support for MARATHON EVOLUTION is available at:

ASC telecom AG
Seibelstrasse 2-4
63768 Hoesbach
Germany
Phone +49 6021 5001-0
Fax +49 6021 5001-310
E-Mail hq@asctelecom.com
http://www.asctelecom.com

7. References

[1] *Administrator Guide for Avaya Communication Manager*, January 2008, Issue 4.0, Document Number 03-300509.

[2] *Feature Description and Implementation for Avaya Communication Manager*, January 2008, Issue 6, Document Number 555-245-205.

[3] *Installing and Administering SIP Enablement Services*, January 2008, Issue 5.0, Document Number 03-600768.

[4] *SIP Enablement Services (SES) Implementation Guide*, January 2008, Issue 5, Document Number 16-300140.

[5] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Issue 7, Document Number 555-233-507.

[6] ASC telecom product descriptions: http://www.asctelecom.com/english/index_e.html

[7] "*RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals* ", May 2000, RFC 2833, available at http://www.ietf.org/rfc.html

8. Conclusion

These Application Notes describe the conformance testing of the ASC telecom MARATHON EVOLUTION voice recorder with Avaya Communication Manager. Silent monitoring via the single step conferencing recording method offered by the MARATHON EVOLUTION was tested. A detailed description of the configuration required for both the Avaya and the ASC telecom equipment is documented within these Application Notes. The MARATHON EVOLUTION passed all of the tests performed, which included both functional and robustness tests.

9. Change History

Issue	Date	Reason
1.1	8/15/2008	Remove unrequired license entries from **Table 3** and **Figure 3**. Change Security Code description in **Table 11**.
1.0	8/8/2008	Initial issue