Avaya Solution & Interoperability Test Lab

Application Notes for ASC Telecom MARATHON EVOLUTION Voice Recorder with Avaya Communication Manager – Issue 1.0

Abstract

These Application Notes describe the conformance testing of the ASC Telecom MARATHON EVOLUTION voice recorder with Avaya Communication Manager. These Application Notes contain an extensive description of the configurations for both MARATHON EVOLUTION and Communication Manager that 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 Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

The objective of these Application Notes is to verify that ASC Telecom MARATHON EVOLUTION voice recording server can interoperate with Avaya Communication Manager and Avaya Application Enablement Services. The MARATHON EVOLUTION voice recorder offers the following methods of voice recording:

1. Passive trunk recording
2. Station side recording
3. H.323 passive recording
4. Silent monitoring via single step conferencing
5. H.323 active recording

For the purpose of this test, only methods 1 and 4 are used.

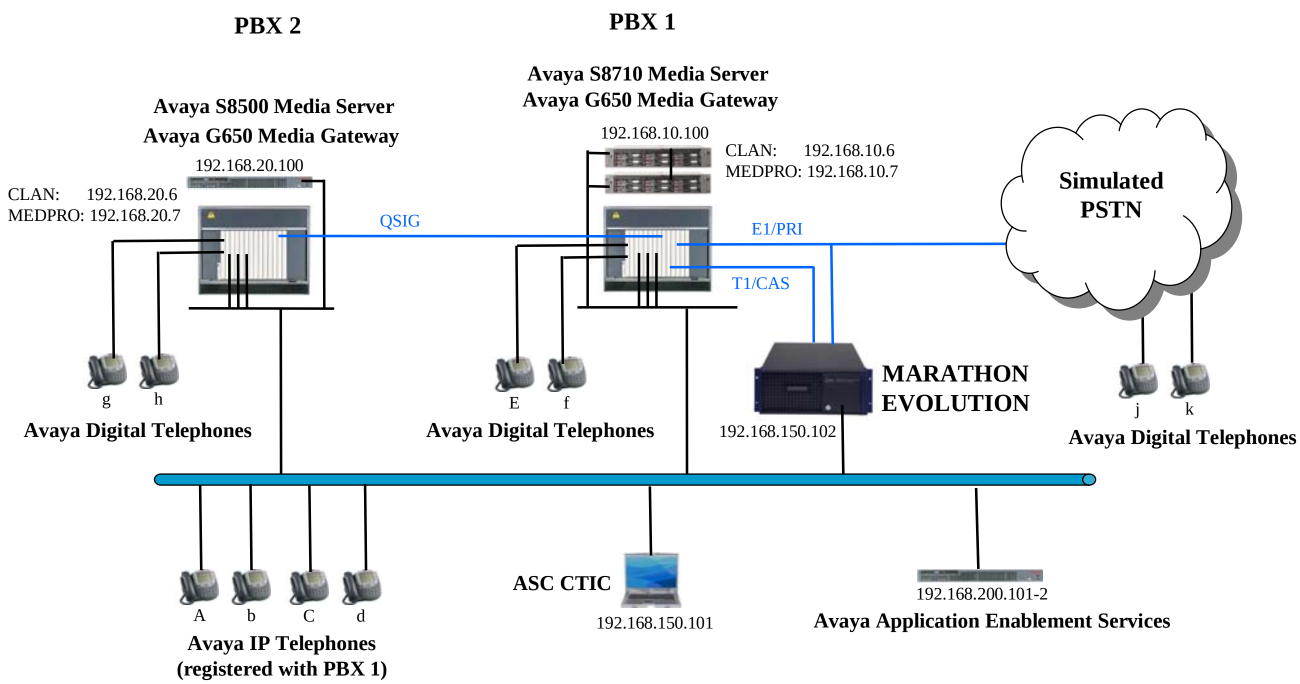


Figure 1: MARATHON EVOLUTION Test Configuration

The ASC CTIC 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 TSAPI 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 a T1 interface.

Additionally, the MARATHON EVOLUTION is attached via a “tap” to the E1/PRI interface which connects PBX 1 to the simulated Public Switched Telephone Network (PSTN). This tap provides an input path to the MARATHON EVOLUTION for communication information sent from PBX 1 to the simulated PSTN as well as that sent from the simulated PSTN to PBX 1, thus allowing the MARATHON EVOLUTION to unobtrusively monitor the communication information flowing in either direction.

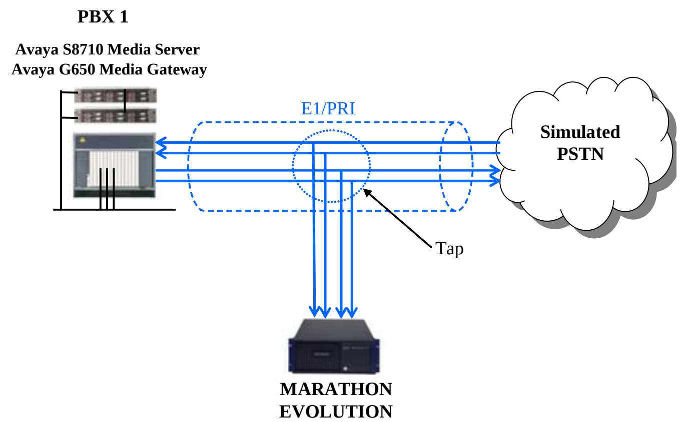


Figure 2: E1/PRI Tap Connection

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.

Phone	Monitored	Model	Extension
A	*	Avaya 4620SW IP	1000127
b		Avaya 4610SW IP	1000115
C	*	Avaya 4610SW IP	1000114
d		Avaya 4610SW IP	1000116
E	*	Avaya 2402	1000002
f		Avaya 2420	1000015
g		Avaya 2410	2000007
k		Avaya 2420	2000015
j		Avaya 2410	3000013
k		Avaya 2420	3000014

Table 1: Device Monitor Configuration

When a call is to be recorded, the MARATHON EVOLUTION voice recorder can initiate a single-step conference, and thus include itself in calls which it wishes to record. The voice stream for such calls is received via the Marathon’s T1 interface to PBX 1.

If one of the participants in a call is external, the MARATHON EVOLUTION voice recorder can alternatively monitor the call’s voice stream via its tap on the PSTN interface to PBX 1. The choice of which of these recording methods is used is governed by configuration parameters.

The PBX 2 system is attached to PBX 1 via a T1/QSIG interface, and is used as a networked PBX system. This allows remote networked telephones (g, h) to be included in the test.

The “PSTN Simulation” system is used to simulate a T1 connection to the PSTN, and allows external telephones (j, k) 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.

2. Equipment and Software Validated

Hardware/Software Component	Version
Avaya S8500 and S8710 Communication Servers	R013x.01.0.628.6
Avaya G650 Communication Gateways • Avaya TN799DP C-LAN interface • Avaya TN2302AP IP Media Processor • Avaya TN2464CP DS1 Interface • Avaya TN2214CP Digital Line	 HW01/FW017 HW20/FW110 HW01/FW018 HW08/FW015
Avaya Application Enablement Services	3.1
Avaya 4610SW IP telephone	2.3
Avaya 4620SW IP telephone	2.3
ASC MARATHON EVOLUTION SW	6.10.08
ASC Marathon platform OS	SuSE Linux 9.0
ASC CTIC	4.0
ASC CTIC platform OS	Win XP pro + SP2

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 PBX 1 (S8710)

The configuration and verification operations illustrated in this section were all performed using the Avaya Communication Manager SAT terminal via telnet port 5023.

3.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer** command to verify that the 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.

The value configured for "Maximum Concurrently Registered IP Stations" must be sufficient to support the total number of IP stations used.

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

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 10    0
           Maximum Concurrently Registered IP Stations: 50    10
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     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: 20    20

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     0
                    Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

The "ASAI Link Core Capabilities", "ASAI Link Plus Capabilities", and "Computer Telephony Adjunct Links" parameters must be set to "y" for MARATHON EVOLOUTION to access AES.

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES

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

The “IP Stations” parameter must be set to “y” so that IP stations can be attached.

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

   Emergency Access to Attendant? y                                 IP Stations? y
           Enable 'dadmin' Login? y               Internet Protocol (IP) PNC? y
           Enhanced Conferencing? y                        ISDN Feature Plus? n
                  Enhanced EC500? y            ISDN 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? n                     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 Appl. Server Interface (MASI)? n
            Hospitality (Basic)? y        Multimedia Call Handling (Basic)? n
 Hospitality (G3V3 Enhancements)? n      Multimedia Call Handling (Enhanced)? n
                       IP Trunks? y

           IP Attendant Consoles? n
```

The “CTI Stations” parameter must be set to “y”.

```
display system-parameters customer-options                      Page   9 of  11
                           ASAI ENHANCED FEATURES

                               CTI Stations? y
           Increased Adjunct Route Capacity? n
                              Phantom Calls? y






                         ASAI PROPRIETARY FEATURES

                               Agent States? n
```

The value configured for “IP Phone” must be sufficient to support the total number of IP stations used.

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

Product ID  Rel. Limit          Used
IP_API_A       : 0              0
IP_API_B       : 0              0
IP_API_C       : 0              0
IP_Agent       : 1              0
IP_IR_A        : 0              0
IP_Phone       : 12000          10
IP_ROMax       : 12000          0
IP_Soft        : 5              0
IP_eCons       : 0              0
```

3.1.2. Configure Dial Plan

3.1.2.1 Configure Dial Plan Analysis

Use the **change dialplan analysis** command to specify that dialed strings which begin with "1", "2", or "3" are extensions. Include the strings "*02" and "*03" which are as Trunk Access Codes, as described in sections 3.1.4.3 and 3.1.5.3.

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

      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        1       7    ext
        2       7    ext
        3       7    ext
        *02     3    dac
        *03     3    dac
```

3.1.2.2 Configure Uniform Dial Plan

Use the **change uniform-dialplan** command to specify that extensions with a leading digit of "2", which are allocated to the Avaya S8500, and those with a leading digit of "3", which are connected to the PSTN, are to be processed by Auto Alternate Routing (aar). The inserted digits are used by aar as the routing pattern selection criteria (see section 3.1.2.3)

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

  Matching         Insert          Node   Matching         Insert          Node
  Pattern   Len Del Digits Net Conv Num    Pattern  Len Del Digits Net Conv Num
  2         7   0   002   aar  n                                        n
  3         7   0   003   aar  n                                        n
```

3.1.2.3 Configure Auto Alternate Routing

Use the **change aar analysis** command to select a route pattern, using the inserted digits specified by the Uniform Dial Plan (see section 3.1.2.2) are used the selection criteria. The "002" string is used to select the route pattern 2 for the S8500 (see section 3.1.4.4) and the "003" string to select the route pattern 3 for PSTN simulation (see section 3.1.5.4).

```
change aar analysis 0                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    1

          Dialed           Total     Route     Call    Node   ANI
          String          Min  Max  Pattern    Type    Num    Reqd
    002                   10   10    2         aar            n
    003                   10   10    3         aar            n
```

3.1.3. Configure Interface to MARATHON EVOLUTION voice recorder

Use the **change ds1 <x>** command, where <x> is the port address of the DS1 interface board, to configure the DS1 circuit pack which is to be used as the E1/CAS interface to the MARATHON EVOLUTION voice recorder. Accept the default values, except for those which are highlighted in the screen below. The following table describes the usage of the various parameters which must be supplied to this command, as shown below.

Parameter	Usage
Bit Rate	"2.048" is the bit rate required by an E1 interface.
Line Coding	"hdb3" is the line coding required by the MARATHON EVOLUTION voice recorder E1 interface.
Signaling Mode	Channel Associated Signaling (CAS) specifies that signaling information is to be carried separately in each channel.
Interface Companding	The "alaw" voice encoding is to be used by this interface.
CRC?	The MARATHON EVOLUTION voice recorder assumes the presence of a Cyclic Redundancy Check sequence, so this parameter must be set to "y".

Table 3: Hardware/Software Component Versions

```
add ds1 01a07                                                   Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01A07                           Name: Line-Side E1
            Bit Rate: 2.048                    Line Coding: hdb3

      Signaling Mode: CAS

        Interconnect: pbx                 Country Protocol: 1

Interface Companding: alaw                            CRC? y
           Idle Code: 11111111




      Slip Detection? n                 Near-end CSU Type: other

   Echo Cancellation? n
```

3.1.4. Configure Interface to Avaya S8500

3.1.4.1 Configure DS1 Interface to Avaya S8500

Use the **change ds1 <x>** command, where <x> is the port address of the DS1 interface board, to configure the DS1 circuit pack which is to be used as QSIG interface to the Avaya S8500. Accept the default values, except for those which are highlighted in the screen below. The following table describes the usage of the various parameters which must be supplied to this command.

Parameter	Usage
Bit Rate	"2.048" is the bit rate required by an E1 interface.
Line Coding	"hdb3" is the line coding which is configured for the corresponding S8500 interface port.
Signaling Mode	Specify a signaling mode of "isdn-pri" to specify ISDN Primary Rate, as required by QSIG.
Connect	This interface is to act as the "pbx".
Interface	The interface is to act as "peer-slave".
Interface Companding	"alaw" voice encoding is to be used by this interface.
CRC?	The Cyclic Redundancy Check (CRC) parameter is set to "y" to specify that a CRC sequence is to be generated for the data sent via this interface port.
Idle Code	The idle code must be set to "01010101".
Channel Numbering	Channel numbering is by time slot.

Table 4: Configuration for DS1 Interface to Avaya S8500

```
add ds1 01a06                                                Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01A06                          Name: QSIG-8500
            Bit Rate: 2.048                   Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: pbx                       Interface: peer-slave
   TN-C7 Long Timers? n                     Peer Protocol: Q-SIG
Interworking Message: PROGress                       Side: b
Interface Companding: alaw                           CRC? y
           Idle Code: 01010101          Channel Numbering: timeslot
                              DCP/Analog Bearer Capability: 3.1kHz

                                          T303 Timer(sec): 4


      Slip Detection? n                Near-end CSU Type: other

   Echo Cancellation? n
```

3.1.4.2 Configure Signaling Group for the Interface to Avaya S8500 (PBX 2)

Use the **add signaling-group <x>** command, where <x> is a free signaling group number, to create a signaling group which is to be used to connect to the Avaya S8500 (PBX 2). Accept defaults for parameters, except for those which are highlighted.

Parameter	Usage
Primary D-Channel	Port 16 of the DS1 card used by the trunk must be assigned as the D channel port which is "01A0616"
Trunk Group for Channel Selection	Use trunk group "2" which is associated with this signaling group.
Supplementary Service Protocol	Specify a protocol of "b", as required by QSIG private network.

Table 5: Configuration Signaling Group for Interface to Avaya S8500

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

 Group Number: 2              Group Type: isdn-pri
                    Associated Signaling? y          Max number of NCA TSC: 5
                       Primary D-Channel: 01A0616     Max number of CA TSC: 5
                                                   Trunk Group for NCA TSC: 2
       Trunk Group for Channel Selection: 2
          Supplementary Service Protocol: b          Network Call Transfer? N
```

3.1.4.3 Configure Trunk Interface to Avaya S8500

Use the **add trunk-group <x>** command, where <x> is a free trunk group number, to create a trunk group which is to be used to connect to the Avaya S8500. Accept defaults for parameters, except for those which are highlighted.

Parameter	Usage
Group Type	Specify a type of "isdn", as required by QSIG
TAC	Set the Trunk Access Code to "*02".
Carrier Medium	Specify a carrier medium of "PRI/PRI2 to specify Primary Rate/Basic Rate.
Service Type	Specify the trunk is used as a "tie" line to another PBX.
Supplementary Service Protocol	Specify a protocol of "b", as required by QSIG private network.
Digit Handling	Specify bi-directional overlapped digit handling. Set to "overlap/overlap".

Table 6: Configuration Parameters for Trunk Interface to Avaya S8500

```
add trunk-group 2                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 2                    Group Type: isdn          CDR Reports: y
  Group Name: S8500                       COR: 1        TN: 1        TAC: *02
   Direction: two-way        Outgoing Display? y         Carrier Medium: PRI/BRI
 Dial Access? y                Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

```
add trunk-group 2                                            Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): overlap/overlap
       Digit Treatment:                                        Digits:
            Trunk Hunt: cyclical                     QSIG Value-Added? n
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
```

Assign sufficient channels to handle the traffic to be carried by the trunk.

```
add trunk-group 2                                               Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/30
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  29

       Port    Code Sfx Name        Night          Sig Grp
  1: 01A0601  TN2464 C  port-01                      1
  2: 01A0602  TN2464 C  port-02                      1
  3: 01A0603  TN2464 C  port-03                      1
  4: 01A0604  TN2464 C  port-04                      1
  5: 01A0605  TN2464 C  port-05                      1
  6: 01A0606  TN2464 C  port-06                      1
  7: 01A0607  TN2464 C  port-07                      1
  8: 01A0608  TN2464 C  port-08                      1
  9: 01A0609  TN2464 C  port-09                      1
 10: 01A0610  TN2464 C  port-10                      1
 11: 01A0611  TN2464 C  port-11                      1
 12: 01A0612  TN2464 C  port-12                      1
 13: 01A0613  TN2464 C  port-13                      1
 14: 01A0614  TN2464 C  port-14                      1
 15: 01A0615  TN2464 C  port-15                      1
```

3.1.4.4 Configure Routing Pattern to Avaya S8500 (PBX 2)

Use the **change route-pattern** command to specify that first three digits of the extension number used for this route pattern should be deleted, and trunk group 2 used to route the call. The leading digits "002" were added by the uniform-dialplan entry for this dial pattern, as shown is section 3.1.2.2.

Parameter	Usage
Grp No	Specify trunk group "2" which is associated with this routing pattern
No. Del Dgts	Specify "3" to delete the three digits which were added via the Uniform Dial Plan (see section 3.1.2.2)

Table 7: Configuration Parameters Routing Pattern to Avaya S8500

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: to S8500
                             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: 2    0                    3                                         n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 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
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

3.1.5. Configure Interface to PSTN

3.1.5.1 Configure DS1 Interface to PSTN

Use the **change ds1 <x>** command where <x> is the port address of the DS1 interface board, to to configure the DS1 circuit pack which is to be used as the E1 Primary Rate interface to the PSTN. Accept the default values, except for those which are highlighted in the screen below. The following table describes the usage of the various parameters which must be supplied to this command.

Parameter	Usage
Bit Rate	"2.048" is the bit rate required by an E1 interface.
Line Coding	"hdb3" is the line coding required by the PSTN interface.
Signaling Mode	"isdn-pri" specifies that ISDN primary rate signaling is to be used.
Interface Companding	The "alaw" voice encoding is to be used by this interface.
CRC	The PSTN simulator to which this trunk is attached assumes the presence of a Cyclic Redundancy Check sequence. Set to "y".
Idle Code	The idle code must be set to "01010101".

Table 8: Configuration Parameters for DS1 Interface to Avaya PSTN

```
add ds1 01a10                                                   Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01A10                           Name: PRI-PSTN
            Bit Rate: 2.048                    Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: pbx                        Interface: user
   TN-C7 Long Timers? n                   Country Protocol: 1
Interworking Message: PROGress            Protocol Version: a
Interface Companding: alaw                            CRC? y
           Idle Code: 01010101
                              DCP/Analog Bearer Capability: 3.1kHz

                                           T303 Timer(sec): 4


      Slip Detection? n                 Near-end CSU Type: other

   Echo Cancellation? N
```

3.1.5.2 Configure Signaling Group for Interface to PSTN

Use the **add signaling-group <x>** command, where <x> is a free signaling group number, to create a signaling group which is to be used to connect to the PSTN. Accept defaults for parameters, except for those which are highlighted.

Parameter	Usage
Primary D-Channel	Port 16 of the DS1 card used by the trunk must be assigned as the D channel.
Trunk Group for Channel Selection	The use the trunk group associated with this signaling group.
Supplementary Service Protocol	Specify a protocol of “b”, as required by PSTN.

Table 9: Configuration Signaling Group for Interface to Avaya PSTN

```
change signaling-group 3                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 3              Group Type: isdn-pri
                    Associated Signaling? y          Max number of NCA TSC: 5
                       Primary D-Channel: 01A1016     Max number of CA TSC: 5
                                                   Trunk Group for NCA TSC: 3
       Trunk Group for Channel Selection: 3
          Supplementary Service Protocol: b          Network Call Transfer? y
```

3.1.5.3 Configure Trunk Interface to PSTN

Use the **add trunk-group <x>** command, where <x> is a free trunk group number, to create a trunk group which is to be used to connect to the PSTN. Accept defaults for parameters, except for those which are highlighted.

Parameter	Usage
Group Type	Specify a type of “isdn”, as required by the PSTN interface
TAC	Set the Trunk Access Code to “*03”.
Dial Access	Specify the trunk is to have dial access. Set to “y”.
Service Type	Specify the trunk is used as a tie line to the PSTN
Digit Handling	Specify bi-directional overlapped digit handling. Set to “overlap/overlap”.
Send Calling Number	Specify that the calling number should be sent. Set to “y”.
Send Connected Number	Specify that the connected number should be sent. Set to “y”.

Table 10: Configuration Parameters for Trunk Interface to PSTN

```
add trunk-group 3                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 3                   Group Type: isdn          CDR Reports: y
  Group Name: PSTN                      COR: 1        TN: 1        TAC: *03
   Direction: two-way        Outgoing Display? y        Carrier Medium: PRI/BRI
 Dial Access? y                Busy Threshold: 255      Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n           TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

```
add trunk-group 3                                               Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): overlap/overlap
       Digit Treatment:                                       Digits:
            Trunk Hunt: cyclical                     QSIG Value-Added? n
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

```
add trunk-group 3                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member: 1
                                      Send Name: n      Send Calling Number: y
            Used for DCS? n                            Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: y
Network Call Redirection: none                  Hold/Unhold Notifications? n
             Send UUI IE? y                   Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                        Ds1 Echo Cancellation? n

                                          US NI Delayed Calling Name Update? n

                             Network (Japan) Needs Connect Before Disconnect? n
                                                        Apply Local Ringback? n
```

Assign sufficient channels to handle the traffic to be carried by the trunk.

```
add trunk-group 3                                                 Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/15
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  15

       Port    Code Sfx Name        Night           Sig Grp
  1: 01A1001  TN2464 C                                3
  2: 01A1002  TN2464 C                                3
  3: 01A1003  TN2464 C                                3
  4: 01A1004  TN2464 C                                3
  5: 01A1005  TN2464 C                                3
  6: 01A1006  TN2464 C                                3
  7: 01A1007  TN2464 C                                3
  8: 01A1008  TN2464 C                                3
  9: 01A1009  TN2464 C                                3
 10: 01A1010  TN2464 C                                3
 11: 01A1011  TN2464 C                                3
 12: 01A1012  TN2464 C                                3
 13: 01A1013  TN2464 C                                3
 14: 01A1014  TN2464 C                                3
 15: 01A1015  TN2464 C                                3
```

3.1.5.4 Configure Routing Pattern to PSTN

Use the **change route-pattern** command to specify that first three digits of the extension number used for this route pattern should be deleted, and trunk group 3 used to route the call. The leading digits "003" were added by the uniform-dialplan entry for this dial pattern, as shown is section 3.1.2.2.

Parameter	Usage
Grp No	Specify trunk group "3" which is associated with this routing pattern
No. Del Dgts	Specify "3" to delete the three digits which were added via the Uniform Dial Plan (see section 3.1.2.2)

Table 11: Configuration Parameters Routing Pattern to PSTN

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: PSTN
                             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: 3    0                    3                                         n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 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
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

3.1.6. Configure Interface to AES

The Avaya Application Services server TSAPI interface provides MARATHON EVOLOUTION with a means of communicating with Avaya Communication Manager to perform telephony operations. Avaya Communication Manager requires the configuration parameters shown in this section.

Use the **change node-names ip** command to define the address of the "clan" interface and the Avaya Enablement Services server.

```
change node-names ip                                            Page   1 of   1
                                IP NODE NAMES
    Name              IP Address          Name              IP Address
clan                192.168.10.6                          .   .   .
default             0 .0 .0 .0                          .   .   .
ipsi                192.168.10.5                          .   .   .
medpro              192.168.10.7                          .   .   .
procr                 .   .   .                         .   .   .
```

Use the **add ip-interface** command to allocate a call control interface. The port value specified should be that of the Clan interface. The value used as "Node Name" must be one of the names from the list defined by the **change node-names ip** command. The "Subnet Mask" and "Gateway Address" should be assigned to the values used by the Ethernet network to which the Clan is attached.

```
add ip-interface 01a02                                          Page   1 of   1
                                  IP INTERFACES


                  Type: C-LAN
                  Slot: 01A02
           Code/Suffix: TN799  D
             Node Name: clan
            IP Address: 192.168.10.6
           Subnet Mask: 255.255.255.0                            Link:
       Gateway Address: 192.168.10.254
  Enable Ethernet Port? y                        Allow H.323 Endpoints? y
        Network Region: 1                         Allow H.248 Gateways? y
                  VLAN: n                           Gatekeeper Priority: 5

 Target socket load and Warning level: 400
       Receive Buffer TCP Window Size: 8320
                                ETHERNET OPTIONS
                  Auto? y
```

Use the **change ip-services** command to set the parameters for **AESVCS** service as shown below for the C-LAN which was defined above to serve as the interface to the AES server.

```
change ip-services                                              Page   1 of   4

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

An entry for the AES server must be made in the list in the screen shown below. The name assigned to the AES server when it was installed must be entered in the "AE Services Server" field for that entry. The "Password" entry must the same as was assigned to the switch connection, as shown in section 3.2 of this document.

```
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
```

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 "Add / Edit TSAPI Links" configuration screen shown in section 3.2 of this document. 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   2
                                    CTI LINK
 CTI Link: 4
Extension: 1999996
     Type: ADJ-IP
                                                                     COR: 1
     Name: AES-devcon223-tsapi
```

Use the **add data-module <x>** command, where <x> is an unassigned extension, to allocate an extension to be used as the data interface for the clan module. The value used as "extension" can be any free extension. The "Name" value is only used for identification purposes. The "Type" field must be "ethernet". The "Port" should be assigned to port 17 of the Clan interface. The "Link" number should be assigned a value between 1 and 99.

```
add data-module 1000000                                         Page   1 of   1
                               DATA MODULE

  Data Extension: 1000000        Name: clan
            Type: ethernet
            Port: 01A0217
            Link: 1




Network uses 1's for Broadcast Addresses? y
```

3.1.7. Configure Phantom Monitoring Stations

Use the **add station** command to create "phantom" stations to be used by the MARATHON EVOLOUTION Voice Recorder to monitor calls. These stations are assigned to ports on the DS1 interface which is connected to the MARATHON EVOLOUTION Voice Recorder, and are used as participants in single step conferences which are initiated by the MARATHON EVOLOUTION Voice Recorder to monitor calls. Accept the default values, except for those which are highlighted in the screens which follow. The table below describes the usage of the various parameters which must be supplied to this command.

Parameter	Usage
Extension	Use an unused extension which is compatible with the dial plan.
Type	Use a type of "DS1FD" to specify the DS1 interface.
Port	Specify one of the ports on the DS1 interface which is attached to the MARATHON EVOLUTION voice recorder. The configuration of this interface is described in section 3.1.3.
Auto answer	Specify a value of "all", as required by the MARATHON EVOLUTION voice recorder.

Table 12: Configuration Phantom Monitoring Stations

```
add station 1002001                                             Page   1 of   3
                                     STATION

Extension: 1002001                          Lock Messages? n          BCC: 0
     Type: DS1FD                            Security Code:             TN: 1
     Port: 01A0701                        Coverage Path 1:            COR: 1
     Name: SiMo DS1 01                    Coverage Path 2:            COS: 1
                                          Hunt-to Station:          Tests? y

STATION OPTIONS
              Loss Group: 4
    Off Premises Station? y
       R Balance Network? n


          Survivable COR: internal
   Survivable Trunk Dest? Y
```

```
add station 1002001                                           Page   2 of   3
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: all
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                        Call Waiting Indication: y
 Per Button Ring Control? n                   Att. Call Waiting Indication: y
                                                Distinctive Audible Alert? y
        Switchhook Flash? y                           Adjunct Supervision? y
    Ignore Rotary Digits? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic
    MWI Served User Type:
              AUDIX Name:
                                                Coverage After Forwarding? s


 Emergency Location Ext: 1002001
```

Repeat this for the stations 100202 to 100205 using sequential DS1 ports.

3.1.8. Configure IP Stations A through D

Use the **add station** command to create a digital station for extensions 1000127, 1000114, 1000115, 1000116, using the values shown below.

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.
Security Code	Assign a string of decimal digits as a Security Code. For convenience, the reverse of the extension is used.
Name	Any alphanumeric string can be assigned as an extension name.

Table 13: Configuration IP Stations

```
add station 1000127                                             Page   1 of   4
                                     STATION

Extension: 1000127                          Lock Messages? n          BCC: 0
     Type: 4620                             Security Code: 7210001     TN: 1
     Port: IP                             Coverage Path 1:            COR: 1
     Name: ext 1000127                    Coverage Path 2:            COS: 1
                                          Hunt-to Station:

STATION OPTIONS
              Loss Group: 19              Personalized Ringing Pattern: 1
                                                      Message Lamp Ext: 1000127
            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? n




                                                   Customizable Labels? Y
```

3.1.9. Configure Digital Stations e and f

Use the **add station** command to create a digital station for extension 1000002, using the values shown below. Repeat this for extension 1000015.

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.
Port	Assign the port of the TN2214CP Digital Line circuit pack to which the digital station is attached.
Name	Any alphanumeric string can be assigned as an extension name.

Table 14: Configuration IP Stations

```
add station 1000002                                             Page   1 of   4
                                     STATION

Extension: 1000002                         Lock Messages? n          BCC: 0
     Type: 2402                            Security Code:             TN: 1
     Port: 01A0502                       Coverage Path 1:            COR: 1
     Name: ext 1000002                   Coverage Path 2:            COS: 1
                                         Hunt-to Station:

STATION OPTIONS
              Loss Group: 2              Personalized Ringing Pattern: 1
             Data Module? n                         Message Lamp Ext: 1000002
            Speakerphone: 1-way                  Mute Button Enabled? y
        Display Language: english

                                                   Media Complex Ext:
                                                        IP SoftPhone? n
```

3.1.10. 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 1000127 and 1000114 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 1000127 and 1000114 are used.

Table 15: Configuration IP Stations

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

            Group Number: 1                              ACD? n
              Group Name: ASC Test HG                  Queue? n
         Group Extension: 1001301                     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:
```

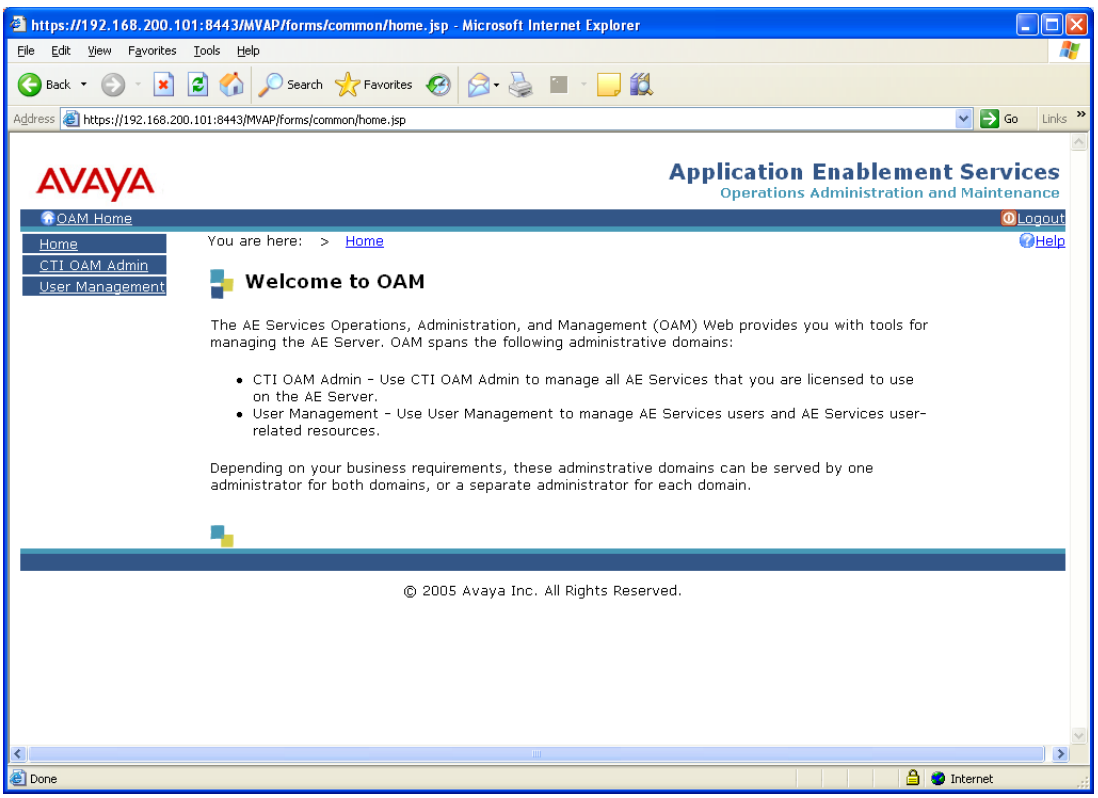
```
add hunt-group 1                                                Page   3 of  60
                                  HUNT GROUP
          Group Number: 1       Group Extension: 1001301   Group Type: ucd-mia
  Member Range Allowed: 1 - 1500       Administered Members (min/max): 1   /2
                                            Total Administered Members: 2
GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)            Ext      Name (24 characters)
   1: 1000127  ext 1000127                 14:
   2: 1000114  ext 1000114                 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
```

3.2. Configure Avaya AES

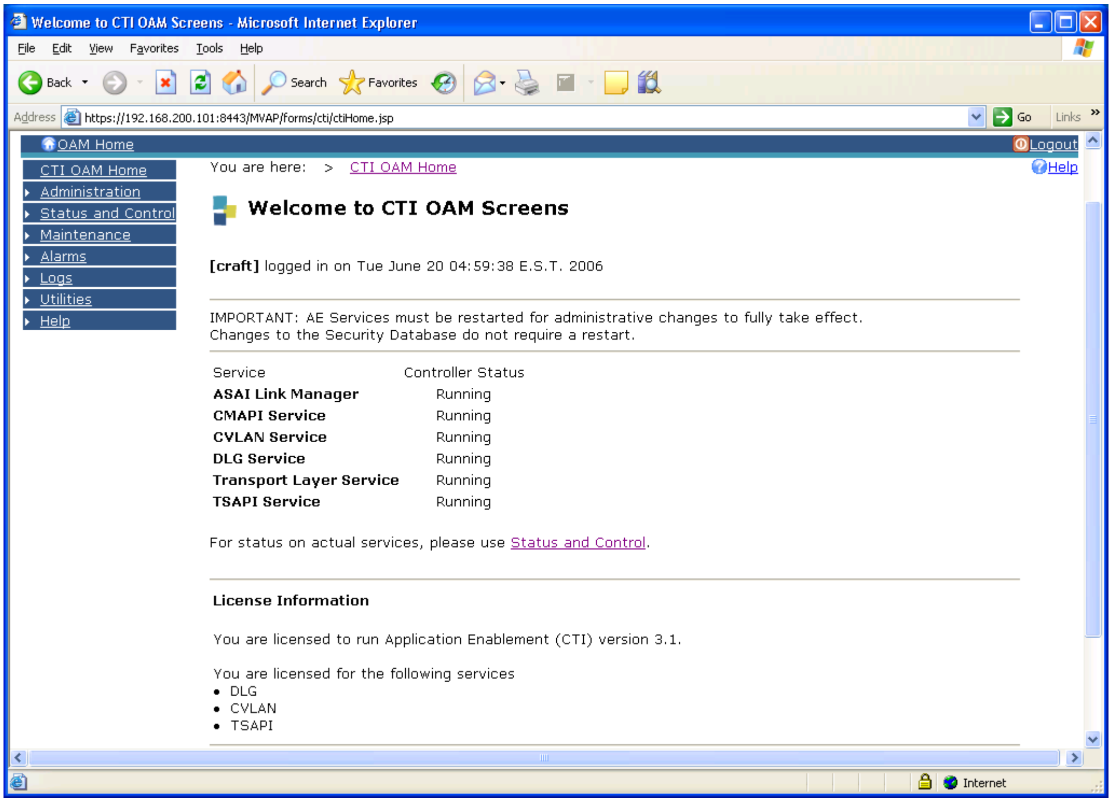
The AES server is configured via a web browser by accessing the following URL:

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

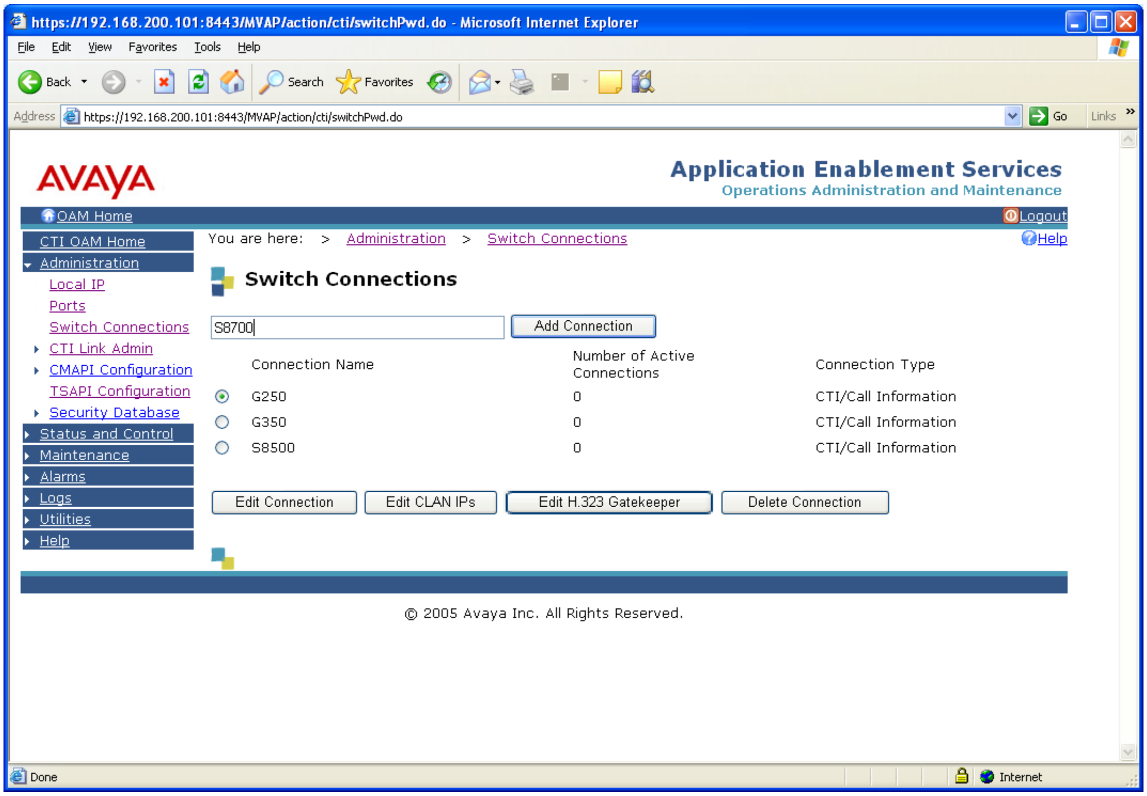
Once the login screen appears, enter either the OAM Admin login ID/password for perform administrative activities on the AE Server or the User Management ID/password to manage AE Services users and AE Services user-related resources. AE Server administrative activities have been partitioned into two administrative domains to enable each to be administered by separate administrators, should business requirements so dictate. To change from one of these domains to the other, first log out and then log in again with the user name/password which corresponds to the domain to be accessed (do not forget the "***s***" on "http***s***", or the login will not succeed).



After logging in with the OAM Admin user ID/password, 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.



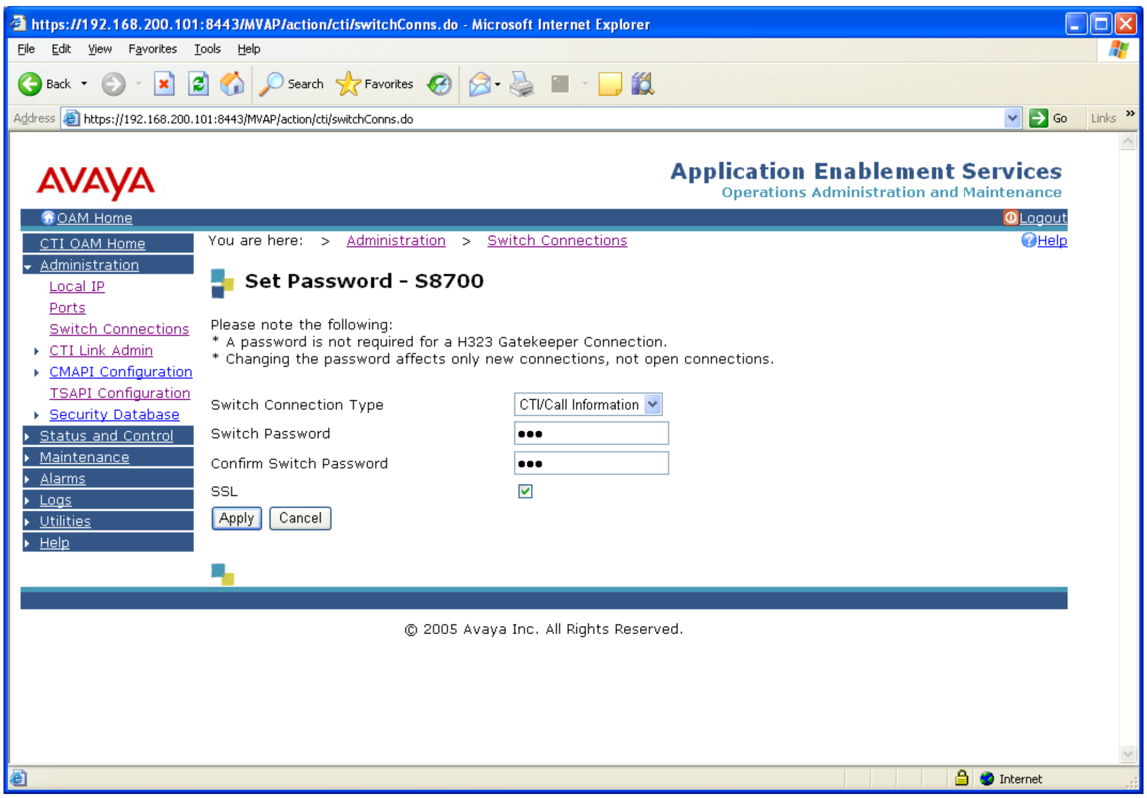
Navigate to **Administration->Switch Connections**. Enter the name of the Switch Connection to be added, and click on the "Add Connection" button. This name should match that which is used in the cti.ini configuration described in table 22 of this document.



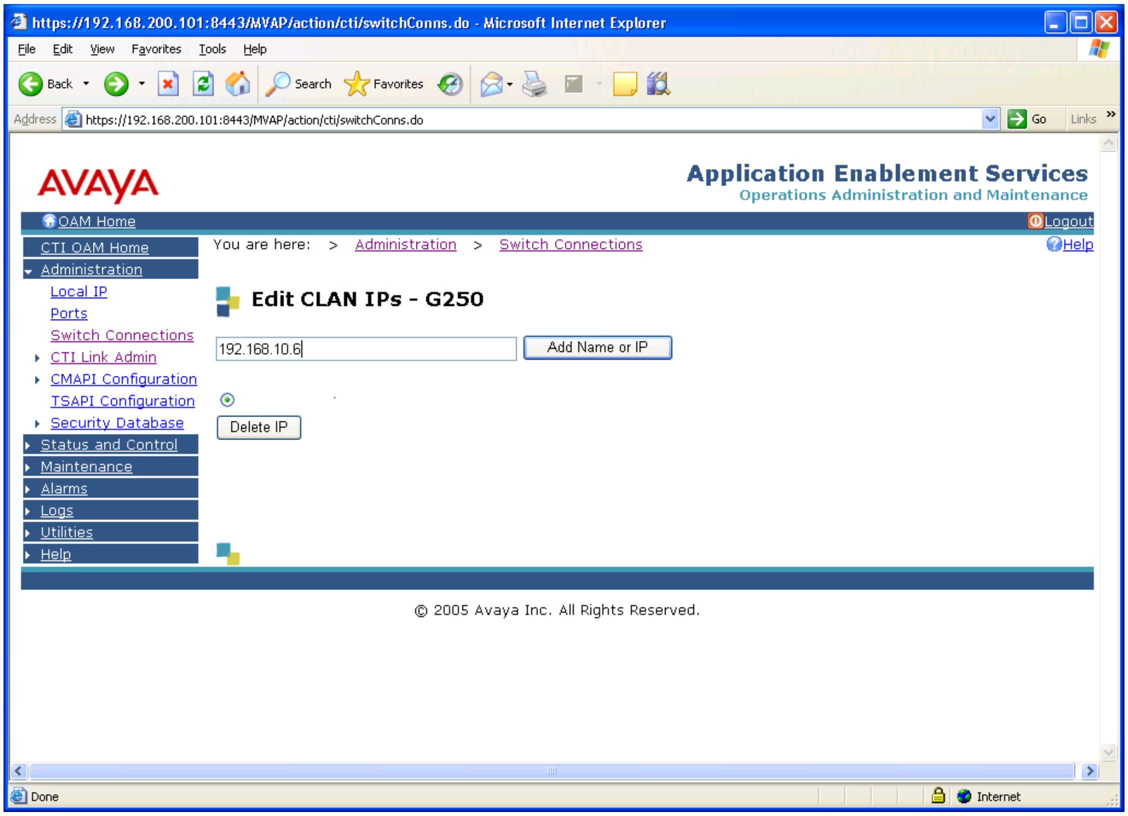
This causes the following screen to be presented. At this point, enter the screen fields as described in the following table, and click the “Apply” button.

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 section 3.1.6. 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

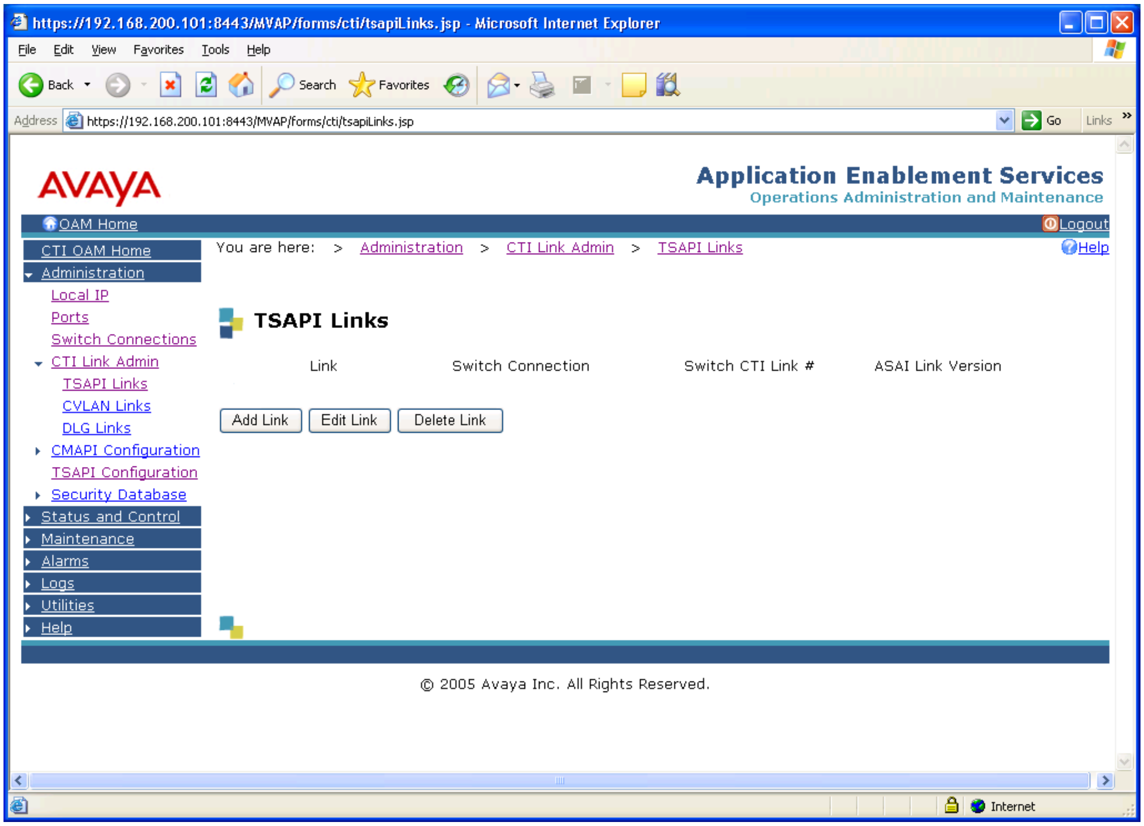
Table 16: Configuration of Switch Password



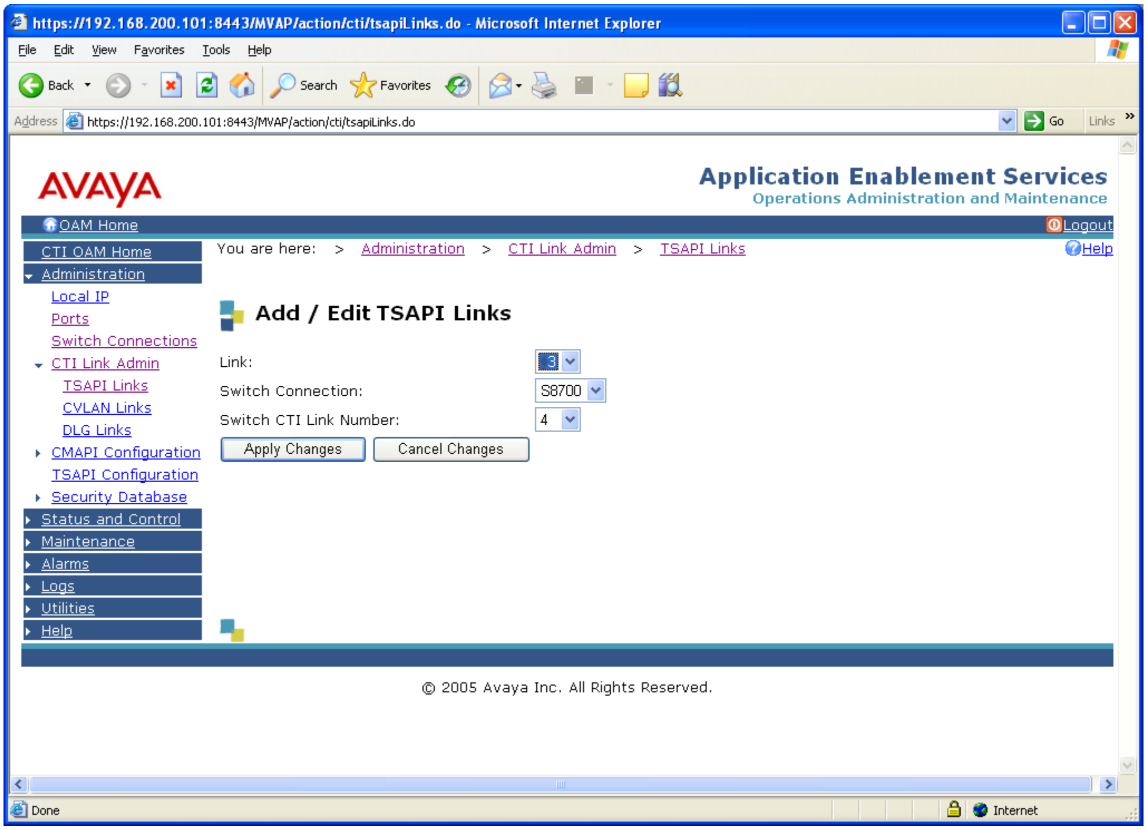
From the **Administration->Switch Connections** screen, click the "Edit CLAN IPs" button to display the screen show below. Enter the IP address of the CLAN with which AES is to use for communication with the switch, and click the "Add Name or IP" button.



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.



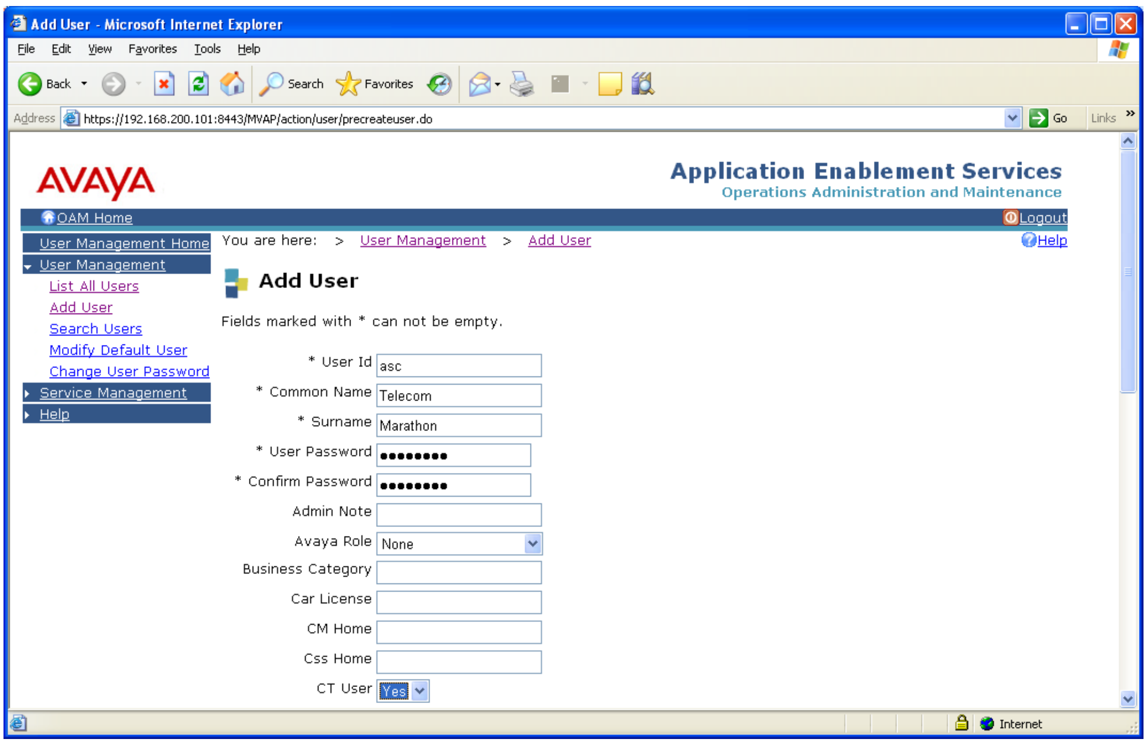
Fill in the parameters for the link to be added. The "Link" parameter must be a value between 1 and 16 which is not assigned to another link. The "Switch Connection" parameter should be the name of the Avaya Media Server which is to be controlled by this link. The value for the TSAPI "Switch CTI Link 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 section 3.1.1. Click the "Apply Changes" button.



Log out and log in again with the user administration ID/password, which will cause the "OAM Welcome" screen to be displayed just as after the previous login.

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

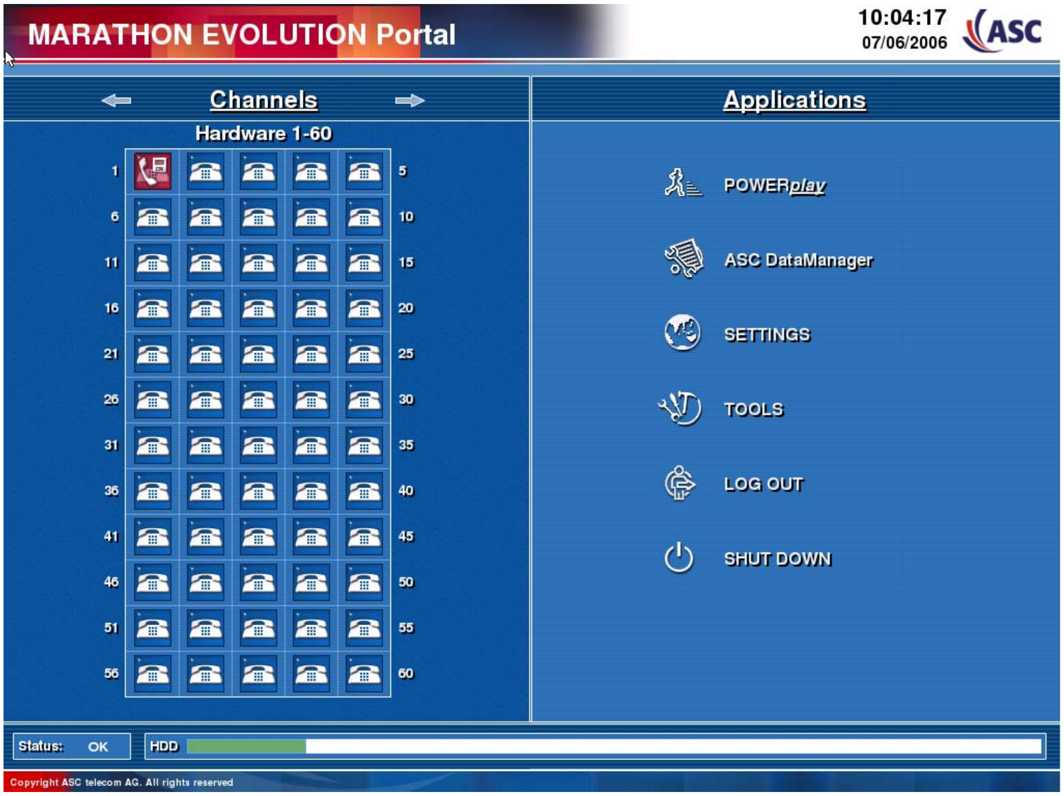
The "CT User" field for this user must be set to "Yes". In this case, the AES user is the MARATHON EVOLUTIONCT application, which uses AES to monitor stations and initiate switching operations. The values chosen for the "User Id" and "User Password" fields must be the same as those contained cti.ini configuration file, as described in table 22 of this document. Upon completion, scroll down and select the "Apply" button. The "User Id" and "User Password" must also be added to the cti.ini configuration file, as described in table 22 of this document.



3.3. 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.

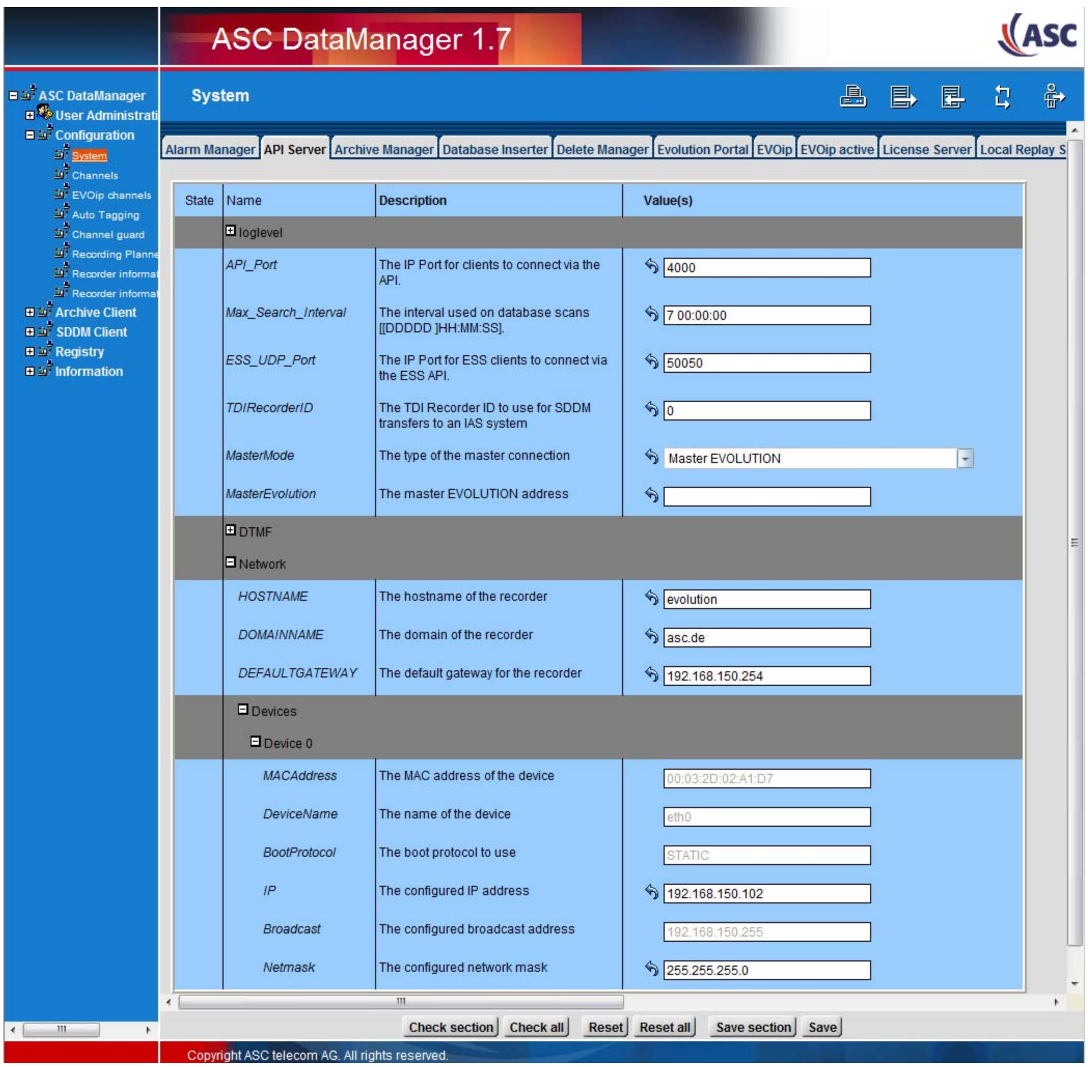
The Marathon configuration tool is available by selecting the **ASC DataManager** application on the right.



The following parameters must be set within the API Server tab. Upon completion, click the "Save section" button.

Parameter	Value
DEFAULTGATEWAY	The default gateway to be used by the MARATHON EVOLOUTION Voice Recorder. In this case, 192.168.150.254.
IP	The IP address to be assigned to the MARATHON EVOLOUTION Voice Recorder. In this case, 192.168.150.102.
Netmask	The net mask to be used by the MARATHON EVOLOUTION Voice Recorder. In this case, 255.255.255.0.

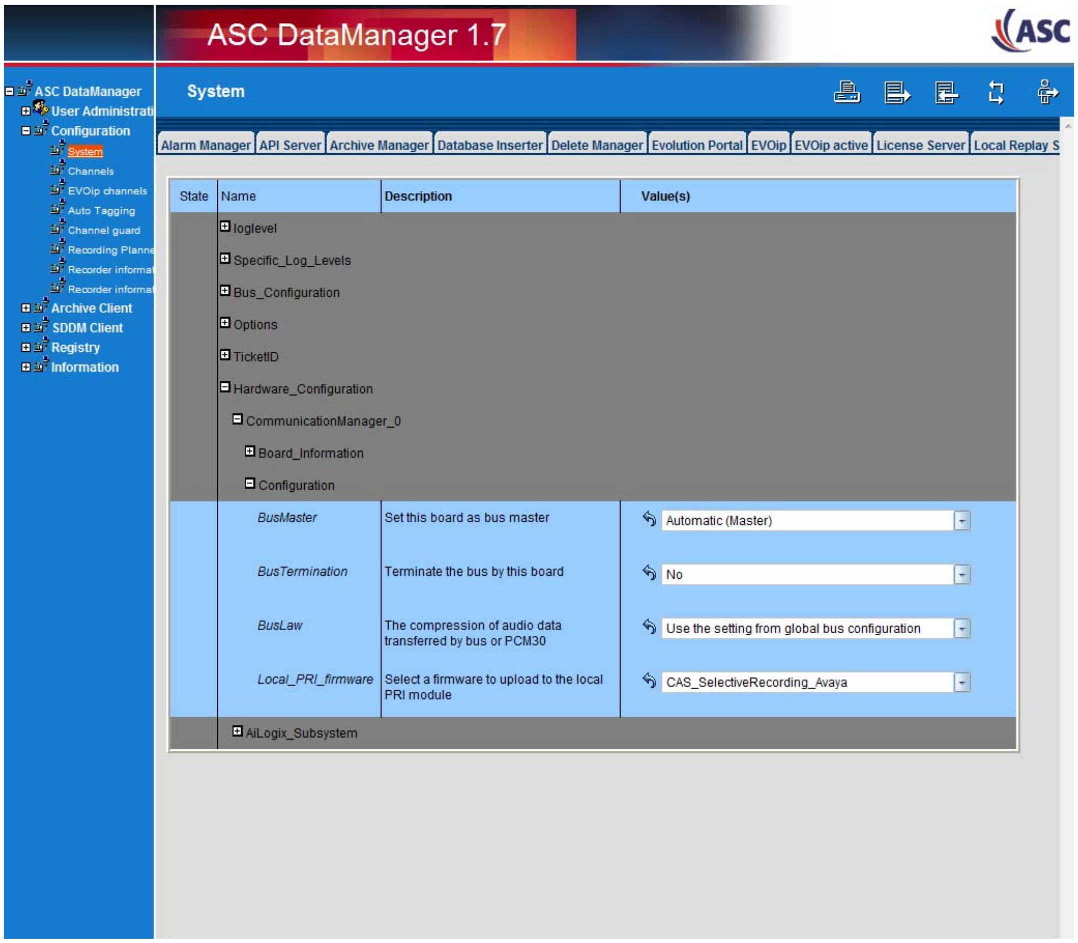
Table 17: Configuration API Server



Scroll to the right and select the "Recording Manager" tab (not shown), and set the parameters as shown in the following table.

Parameter	Value
Local_PRI_firmware	"CAS_SelectiveRecording_Avaya"

Table 18: Configuration Recording Manager



Navigate to **ASC Data Manager->Configuration->Channels**. For each of the channels used by the Marathon for single-step conference recording, the following parameters must be set for the screen shown on the next page.

Parameter	Value
RecordStartMode	HOST (External application)
StorageMode	EXTERN_DELETE or COMPLETE_CALL_INFO
InputSource1	COMMAN (Analog / PCM30)
InputType1	PRI_ACTIVE_TIMESLOT (Active PRI Input)
InputSlot1	1

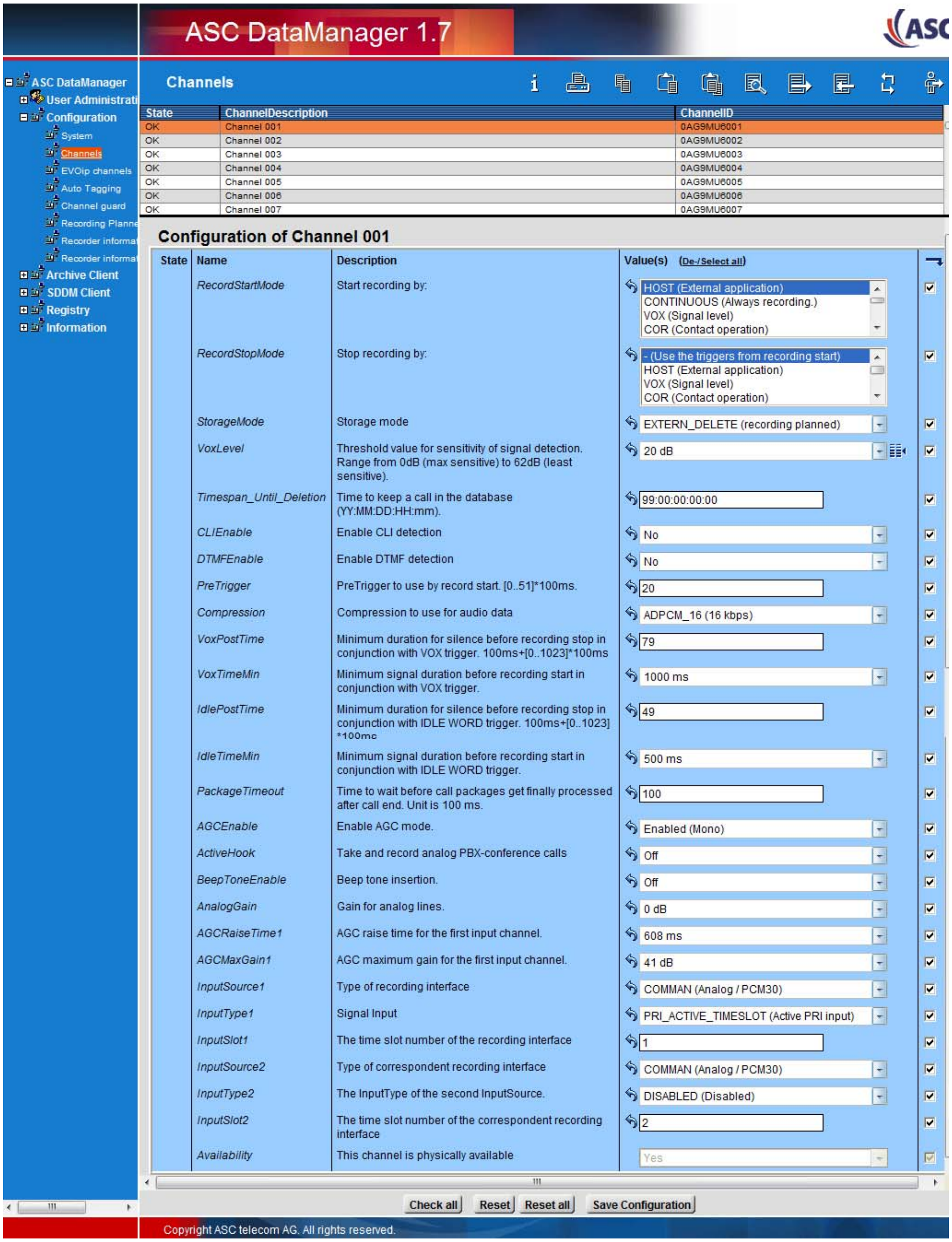
Table 19: Channel Configuration

Note that each of the channels to be used for recording must be configured. This task is simplified greatly via the availability of the channel copy and paste functions, which enable the settings for one channel to be pasted to multiple channels via a single operation, as follows:

- Selecting the channel to be copied
- Pressing the "channel copy" control
- Selecting those channels to which the parameters are to be copied
- Pressing the "channel paste" control

The time slots (InputSlot1) assigned to each channel are set individually by the paste function, eliminating the need for subsequent editing.

Upon completion, click the "**Save Configuration**" button to save changes.

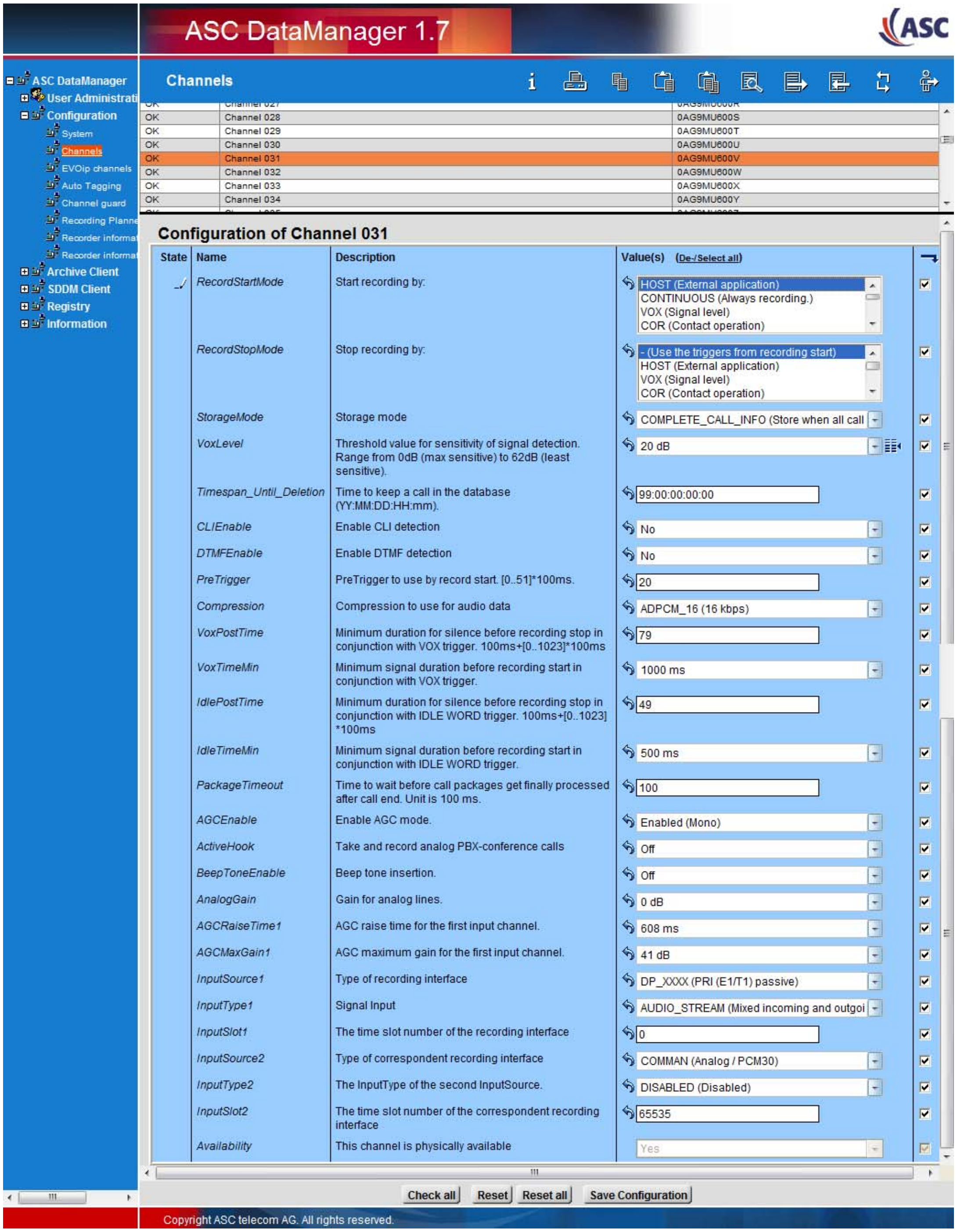

ASC DataManager 1.7
ASC DataManager
User Administrati
Configuration
System
Channels
EVOip channels
Auto Tagging
Channel guard
Recording Planne
Recorder informat
Recorder informat
Archive Client
SDDM Client
Registry
Information
Channels
State
ChannelDescription
ChannelID
OK Channel 001 0AG9MU6001
OK Channel 002 0AG9MU6002
OK Channel 003 0AG9MU6003
OK Channel 004 0AG9MU6004
OK Channel 005 0AG9MU6005
OK Channel 006 0AG9MU6006
OK Channel 007 0AG9MU6007
Configuration of Channel 001
State
Name
Description
Value(s) (De-/Select all)
RecordStartMode
Start recording by:
HOST (External application)
CONTINUOUS (Always recording.)
VOX (Signal level)
COR (Contact operation)
RecordStopMode
Stop recording by:
- (Use the triggers from recording start)
HOST (External application)
VOX (Signal level)
COR (Contact operation)
StorageMode
Storage mode
EXTERN_DELETE (recording planned)
VoxLevel
Threshold value for sensitivity of signal detection. Range from 0dB (max sensitive) to 62dB (least sensitive).
20 dB
Timespan_Until_Deletion
Time to keep a call in the database (YY:MM:DD:HH:mm).
99:00:00:00:00
CLIEnable
Enable CLI detection
No
DTMFEnable
Enable DTMF detection
No
PreTrigger
PreTrigger to use by record start. [0..51]*100ms.
20
Compression
Compression to use for audio data
ADPCM_16 (16 kbps)
VoxPostTime
Minimum duration for silence before recording stop in conjunction with VOX trigger. 100ms+[0..1023]*100ms
79
VoxTimeMin
Minimum signal duration before recording start in conjunction with VOX trigger.
1000 ms
IdlePostTime
Minimum duration for silence before recording stop in conjunction with IDLE WORD trigger. 100ms+[0..1023] *100ms
49
IdleTimeMin
Minimum signal duration before recording start in conjunction with IDLE WORD trigger.
500 ms
PackageTimeout
Time to wait before call packages get finally processed after call end. Unit is 100 ms.
100
AGCEnable
Enable AGC mode.
Enabled (Mono)
ActiveHook
Take and record analog PBX-conference calls
Off
BeepToneEnable
Beep tone insertion.
Off
AnalogGain
Gain for analog lines.
0 dB
AGCRaiseTime1
AGC raise time for the first input channel.
608 ms
AGCMaxGain1
AGC maximum gain for the first input channel.
41 dB
InputSource1
Type of recording interface
COMMAN (Analog / PCM30)
InputType1
Signal Input
PRI_ACTIVE_TIMESLOT (Active PRI input)
InputSlot1
The time slot number of the recording interface
1
InputSource2
Type of correspondent recording interface
COMMAN (Analog / PCM30)
InputType2
The InputType of the second InputSource.
DISABLED (Disabled)
InputSlot2
The time slot number of the correspondent recording interface
2
Availability
This channel is physically available
Yes
Check all
Reset
Reset all
Save Configuration
Copyright ASC telecom AG. All rights reserved.

For each of the channels used by the Marathon trunk side recording the following parameters must be set for the screen shown on the next page:

Parameter	Value
RecordStartMode	HOST (External application)
StorageMode	EXTERN_DELETE or COMPLETE_CALL_INFO
InputSource1	DP_XXXX (PRI(E1/T1) passive)
InputType1	AUDIO_STREAM
InputSlot1	0

Table 20: Channel Configuration for Trunk Side Recording

Here again, the channel copy and channel paste functions can be used to simplify the task of setting each of the channels to be used for recording. Upon completion click on the "**Save Configuration**" button to save changes.


ASC DataManager 1.7
ASC
ASC DataManager
User Administrati
Configuration
System
Channels
EVOip channels
Auto Tagging
Channel guard
Recording Planne
Recorder informat
Recorder informat
Archive Client
SDDM Client
Registry
Information
Channels
OK Channel 028 0AG9MU600S
OK Channel 029 0AG9MU600T
OK Channel 030 0AG9MU600U
OK Channel 031 0AG9MU600V
OK Channel 032 0AG9MU600W
OK Channel 033 0AG9MU600X
OK Channel 034 0AG9MU600Y
Configuration of Channel 031
State Name Description Value(s) (De-/Select all)
RecordStartMode Start recording by: HOST (External application) CONTINUOUS (Always recording.) VOX (Signal level) COR (Contact operation)
RecordStopMode Stop recording by: - (Use the triggers from recording start) HOST (External application) VOX (Signal level) COR (Contact operation)
StorageMode Storage mode COMPLETE_CALL_INFO (Store when all call
VoxLevel Threshold value for sensitivity of signal detection. Range from 0dB (max sensitive) to 62dB (least sensitive). 20 dB
Timespan_Until_Deletion Time to keep a call in the database (YY:MM:DD:HH:mm). 99:00:00:00:00
CLIEnable Enable CLI detection No
DTMFEnable Enable DTMF detection No
PreTrigger PreTrigger to use by record start. [0..51]*100ms. 20
Compression Compression to use for audio data ADPCM_16 (16 kbps)
VoxPostTime Minimum duration for silence before recording stop in conjunction with VOX trigger. 100ms+[0..1023]*100ms 79
VoxTimeMin Minimum signal duration before recording start in conjunction with VOX trigger. 1000 ms
IdlePostTime Minimum duration for silence before recording stop in conjunction with IDLE WORD trigger. 100ms+[0..1023] *100ms 49
IdleTimeMin Minimum signal duration before recording start in conjunction with IDLE WORD trigger. 500 ms
PackageTimeout Time to wait before call packages get finally processed after call end. Unit is 100 ms. 100
AGCEnable Enable AGC mode. Enabled (Mono)
ActiveHook Take and record analog PBX-conference calls Off
BeepToneEnable Beep tone insertion. Off
AnalogGain Gain for analog lines. 0 dB
AGCRaiseTime1 AGC raise time for the first input channel. 608 ms
AGCMaxGain1 AGC maximum gain for the first input channel. 41 dB
InputSource1 Type of recording interface DP_XXXX (PRI (E1/T1) passive)
InputType1 Signal Input AUDIO_STREAM (Mixed incoming and outgoi
InputSlot1 The time slot number of the recording interface 0
InputSource2 Type of correspondent recording interface COMMAN (Analog / PCM30)
InputType2 The InputType of the second InputSource. DISABLED (Disabled)
InputSlot2 The time slot number of the correspondent recording interface 65535
Availability This channel is physically available Yes
Check all
Reset
Reset all
Save Configuration
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3.4. Configure Marathon CTIC

The Marathon CTIC software should installed from the distribution media, accepting all defaults. The configuration parameters for Marathon CTI Controller are contained in the file ctic.ini, an ASCI text file which is installed in the "C:\<Program Files Folder>\ASC\CTI Tenovis" directory locally on the CTIC during product installation. This file can be edited with a text editor.

The following is a list of those sections within this file which must be edited for use with Avaya Communication Manager and/or Avaya Application Enablement Services. The full text of this file after it has been edited is shown at the culmination of this section. Those entries which are highlighted in bold correspond to parameters which have been configured for Avaya Communication Manager and/or Avaya Application Enablement Services.

The following is a description of the various sections within this file:

`LogModule`

The content of this section is reserved for future use.

`AlarmMan`

The content of this section is reserved for future use.

`Licenseserver`

The content of this section is reserved for future use.

`CONTROLLER-1`

The usage of the entries within this section is as follows:

Parameter	Value
`EssServerAddress`	The IP address of the Marathon server.
`EssClientAddress`	The IP address or DNS name of the CTIC.

Table 21: cti.ini File Configuration for CONTROLLER-1 Section

The remainder of the entries within this section should be set as specified.

PBX-1

The usage of the entries within this section is as follows:

Parameter	Value
ServiceName	AVAYA#S8700#CSTA#AES_SERVER_1
Login	The AES login ID to be used by this application. This ID must be added to AES user database, as described in the "Add User" screen of section 3.2 of this document.
Password	The AES login password for the above login ID. This password must be added to AES user database, as described in the "Add User" screen of section 3.2 of this document.

Table 22: cti.ini File Configuration for PBX-1 Section

The ServiceName parameter consists of the following fields, separated by "#" characters:

Parameter	Value
AVAYA	This is a fixed value.
S8700	This is the name that was assigned to the switch connection which was assigned to the PBX within the AES Administration Switch Connections screen shown in section 3.2 of this document.
CSTA	This is a fixed value.
AES_SERVER_1	This is the name that was assigned to the AES server when the AES software installation was performed. This name is contained in the AES ip-services table, as described in section 3.1.6.

Table 23: ServiceName Parameter Components

The remainder of the entries within this section should be set as shown.

PBXDRIVER-1

The entries within this section should be set as shown.

SIMOCHNL-1...

There is a separate section in the configuration file for each station which was created for the purpose of participating in single step conferences. The number of such stations is the limit of the number of conversations which can be monitored simultaneously via single step conference.

The usage of the entries within each of these sections is as follows:

Parameter	Value
`SimoDeviceMap`	The extension which was configured with Communication Manager for the purpose of participating in single step conferences for the purpose of monitoring conversations which are to be recorded.

The remainder of the entries within these sections should be set as shown.

`TSCHNL-1...`

There is one entry of this type in the configuration file for each channel in the E1/PRI line which was "tapped" for the purpose of monitoring. The entries in each of these sections should be set as shown.

Parameter	Value
`TrunkMap`	<trunk group>,<channel number>

The remainder of the entries within this section should be set as shown.

`MONITORPOINT-1...`

There is one entry of this type in the configuration file for each station that is to be monitored. The usage of the entries within this section is as follows:

Parameter	Value
`MonitorDeviceID`	The extension of the device to be monitored.
`RecordingMode`	This parameter is set to either SIMO or TS. SIMO indicates "Silent Monotoring" via single step conference.

The remainder of the entries within this section should be set as shown.

```
; Possible Log Levels
; -1 - off
;  0 - always out
;  1 - info out
;  2 - debug out
;  3 - debug all out

[LogModule]
Global Level      = -1
Logfile Level     = 3
Logfile Size      = 4000000
Console Level     = 3
Eventlog Level    = 3
Debugout Level    = 3

[AlarmMan]
Address           = localhost
Port              = 0
send              = 60000
wait              = 10000
reconnectIntervall= 1000

[Licenseserver]
Address       = 192.168.1.1
Port     = 7000
Protocol         = CCOPEN
send              = 60000
wait              = 10000
reconnectIntervall= 1000

[CONTROLLER-1]
ControllerName=QMSConroller
ControllerDll=CTIC_CTRL_QMS.dll
EssServerAddress=192.168.150.102
EssServerPortNo=50050
EssClientAddress=CTICLORENZ
EssClientPortNo=10001
EssClientApiDllName=ESSClientAPI.dll
StringMultiplePartners=MULTIPLE
StringNotAvailable=N/A
QMSUsedPbx=1
QMSServerIP=0.0.0.0
QMSServerPort=10000
QMSInternalRecordingDecision=1
TagSplit=AsciiField3
TagOtherParty=AsciiField1
TagCalledParty=AsciiField2

[PBX-1]
PbxID=1
PbxName=Avaya
DriverIDForPbx=1
ServiceName=AVAYA#S8700#CSTA#AES_SERVER_1
Login=asc
Password=asc

[PBXDRIVER-1]
DriverName=AvayaTsapiDriver
DriverDll=CTIC_PBXD_AvayaTSAPI.dll
RecoveryTimerInterval=40000

[SIMOCHNL-1]
ControllerID1=1
MarathonID=0
ChannelID=1
SimoDeviceMap=1002001
PbxID=1
```

```
[SIMOCHNL-2]
ControllerID1=1
MarathonID=0
ChannelID=2
SimoDeviceMap=1002002
PbxID=1

[SIMOCHNL-3]
ControllerID1=1
MarathonID=0
ChannelID=3
SimoDeviceMap=1002003
PbxID=1

[SIMOCHNL-4]
ControllerID1=1
MarathonID=0
ChannelID=4
SimoDeviceMap=1002004
PbxID=1

[SIMOCHNL-5]
ControllerID1=1
MarathonID=0
ChannelID=5
SimoDeviceMap=1002005
PbxID=1

[TSCHNL-1]
ControllerID=1
MarathonID=0
ChannelID=31
ControlMode=SS
TrunkMap=3,1
PbxID=1

[TSCHNL-2]
ControllerID=1
MarathonID=0
ChannelID=32
ControlMode=SS
TrunkMap=3,2
PbxID=1

[TSCHNL-3]
ControllerID=1
MarathonID=0
ChannelID=33
ControlMode=SS
TrunkMap=3,3
PbxID=1

[TSCHNL-4]
ControllerID=1
MarathonID=0
ChannelID=34
ControlMode=SS
TrunkMap=3,4
PbxID=1

[TSCHNL-5]
ControllerID=1
MarathonID=0
ChannelID=35
```

```
ControlMode=SS
TrunkMap=3,5
PbxID=1

[TSCHNL-6]
ControllerID=1
MarathonID=0
ChannelID=36
ControlMode=SS
TrunkMap=3,6
PbxID=1

[TSCHNL-7]
ControllerID=1
MarathonID=0
ChannelID=37
ControlMode=SS
TrunkMap=3,7
PbxID=1

[TSCHNL-8]
ControllerID=1
MarathonID=0
ChannelID=38
ControlMode=SS
TrunkMap=3,8
PbxID=1

[TSCHNL-9]
ControllerID=1
MarathonID=0
ChannelID=39
ControlMode=SS
TrunkMap=3,9
PbxID=1

[TSCHNL-10]
ControllerID=1
MarathonID=0
ChannelID=40
ControlMode=SS
TrunkMap=3,10
PbxID=1

[TSCHNL-11]
ControllerID=1
MarathonID=0
ChannelID=41
ControlMode=SS
TrunkMap=3,11
PbxID=1

[TSCHNL-12]
ControllerID=1
MarathonID=0
ChannelID=42
ControlMode=SS
TrunkMap=3,12
PbxID=1

[TSCHNL-13]
ControllerID=1
MarathonID=0
ChannelID=43
ControlMode=SS
TrunkMap=3,13
PbxID=1
```

```
[TSCHNL-14]
ControllerID=1
MarathonID=0
```

```
ChannelID=44
ControlMode=SS
TrunkMap=3,14
PbxID=1
```

```
[TSCHNL-15]
ControllerID=1
MarathonID=0
ChannelID=45
ControlMode=SS
TrunkMap=3,15
PbxID=1

[TSCHNL-16]
ControllerID=1
MarathonID=0
ChannelID=46
ControlMode=SS
TrunkMap=3,16
PbxID=1

[TSCHNL-17]
ControllerID=1
MarathonID=0
ChannelID=47
ControlMode=SS
TrunkMap=3,17
PbxID=1

[TSCHNL-18]
ControllerID=1
MarathonID=0
ChannelID=48
ControlMode=SS
TrunkMap=3,18
PbxID=1

[TSCHNL-19]
ControllerID=1
MarathonID=0
ChannelID=49
ControlMode=SS
TrunkMap=3,19
PbxID=1

[TSCHNL-20]
ControllerID=1
MarathonID=0
ChannelID=50
ControlMode=SS
TrunkMap=3,20
PbxID=1
```

```
[TSCHNL-21]
ControllerID=1
MarathonID=0
ChannelID=51
ControlMode=SS
TrunkMap=3,21
PbxID=1

[TSCHNL-22]
ControllerID=1
MarathonID=0
ChannelID=52
ControlMode=SS
TrunkMap=3,22
PbxID=1

[TSCHNL-23]
ControllerID=1
MarathonID=0
ChannelID=53
ControlMode=SS
TrunkMap=3,23
PbxID=1

[TSCHNL-24]
ControllerID=1
MarathonID=0
ChannelID=54
ControlMode=SS
TrunkMap=3,24
PbxID=1

[TSCHNL-25]
ControllerID=1
MarathonID=0
ChannelID=55
ControlMode=SS
TrunkMap=3,25
PbxID=1

[TSCHNL-26]
ControllerID=1
MarathonID=0
ChannelID=56
ControlMode=SS
TrunkMap=3,26
PbxID=1
```

```
[TSCHNL-27]
ControllerID=1
MarathonID=0
ChannelID=57
ControlMode=SS
TrunkMap=3,27
PbxID=1

[TSCHNL-28]
ControllerID=1
MarathonID=0
ChannelID=58
ControlMode=SS
TrunkMap=3,28
PbxID=1

[TSCHNL-29]
ControllerID=1
MarathonID=0
ChannelID=59
ControlMode=SS
TrunkMap=3,29
PbxID=1

[TSCHNL-30]
ControllerID=1
MarathonID=0
ChannelID=60
ControlMode=SS
TrunkMap=3,30
PbxID=1

[MONITORPOINT-1]
ControllerID1=1
PbxID=1
MonitorDeviceID=1000127
RecordingMode=SIMO

[MONITORPOINT-2]
ControllerID1=1
PbxID=1
MonitorDeviceID=1000114
RecordingMode=SIMO

[MONITORPOINT-3]
ControllerID1=1
PbxID=1
MonitorDeviceID=1000002
RecordingMode=SIMO
```

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 Communication Manager, and that 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 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:
  - Trunk side recording
  - 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 CTI Controller and the network
  - The connection between the MARATHON EVOLUTION T1/CAS connection to PBX 1.
- 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] “Installation Manual CTI-Controller for Avaya TSAPI”, Version 01, ASC Telecom
[2] “Service Manual CTI-Controller for AVAYA TSAPI”, Version 04, ASC Telecom
[3] ASC Telecom product descriptions: http://www.asctelecom.com/english/index_e.html
[4] “Feature Description and Implementation for Avaya Communication Manager”, 555-245-205, Issue 3, June 2005

8. Conclusion

These Application Notes describe the conformance testing of the ASC Telecom MARATHON EVOLUTION voice recorder with Avaya Communication Manager. Both the passive trunk recording and silent monitoring via single step conferencing recording methods offered by the MARATHON EVOLUTION were 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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Developer*Connection* Program at devconnect@avaya.com.