



## Avaya Solution & Interoperability Test Lab

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# Application Notes for ASC Marathon Evolution Call Recording Solution with Avaya Communication Manager – Issue 1.0

### Abstract

These Application Notes describe the configuration steps required for ASC Marathon Evolution to successfully interoperate with Avaya Communication Manager 2.2.

Marathon Evolution is a Call Recording solution able to capture audio from Communication Manager using a variety of integration mechanisms.

Marathon Evolution uses Computer Telephony Integration (CTI) to extract call event information and supports passive trunk tapping and active station side recording.

An Avaya S8300 Media Server within an Avaya G350 Media Gateway running Avaya Communication Manager 2.2 was used as the hosting PBX. Features and functionality were validated and performance testing was conducted to verify operation under light load.

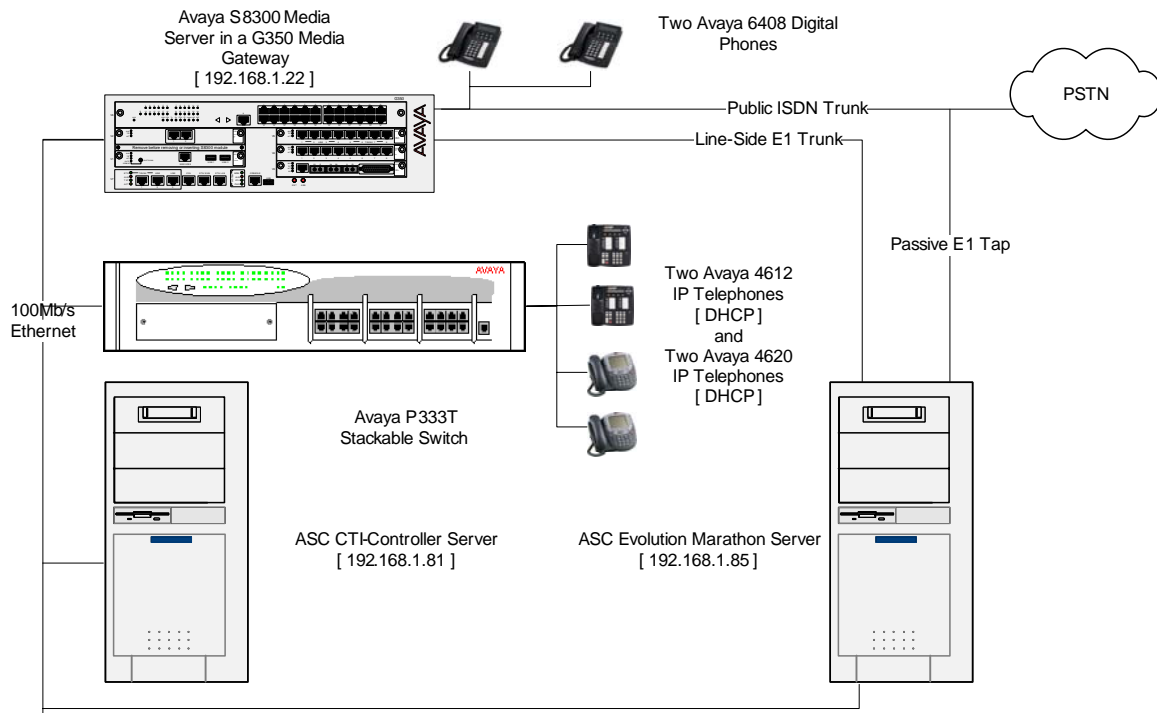
Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the *DeveloperConnection* Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These Application Notes describe the compliance-tested configuration using an ASC Marathon Evolution call recorder, an ASC CTI-Controller server, and an Avaya Communication Manager system.

Marathon Evolution supports passive trunk tapping as well as active station-side recording.

The solution as tested is shown below.



**Figure 1: Tested Avaya Communication Manager System with ASC Marathon Evolution Server and ASC CTI-Controller Server**

## 2. Equipment and Software Validated

The tested configuration is detailed below.

Equipment	Version Information
Avaya S8300 Media Server within an Avaya G350 Media Gateway	Communication Manager 2.2 (R012x.02.0.111.4)
Avaya MM710 T1/E1 Media Module	HW04 FW009
Avaya P333T Stackable Switch	V4.0.17
ASC Marathon Evolution	R4.0
ASC CTI-Controller	R2.0

## 3. Configure Avaya Communication Manager

Different features of Communication Manager need to be configured for the two recording modes to be tested. Please refer to the Administration Guide for Communication Manager for further details – Avaya Document 555-233-506 [1]. The specific options are detailed below.

### 3.1. Configure the CTI Link

Regardless of the mode of audio recording, a CTI link is required to provide call details for each recording. Since ASC provides a version of Avaya Computer Telephony within their CTI-Controller server, it is only necessary to define a CTI link. This requires that the co-resident DLG feature of Communication Manager be enabled as well as Computer Telephony Adjunct Links as shown:

Display System-Parameters Customer-Options (only the relevant page is shown)

OPTIONAL FEATURES	
Abbreviated Dialing Enhanced List? y	Audible Message Waiting? y
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? y
ARS/AAR Partitioning? y	<b>Computer Telephony Adjunct Links? y</b>
ARS/AAR Dialing without FAC? y	<b>Co-Res DEFINITY LAN Gateway? y</b>
ASAI Link Core Capabilities? n	Cvg Of Calls Redirected Off-net? y
ASAI Link Plus Capabilities? n	DCS (Basic)? n
Async. Transfer Mode (ATM) PNC? n	DCS Call Coverage? n
Async. Transfer Mode (ATM) Trunking? n	DCS with Rerouting? n
ATM WAN Spare Processor? n	Digital Loss Plan Modification? y
ATMS? n	DS1 MSP? n
Attendant Vectoring? n	DS1 Echo Cancellation? n

A CTI Link must be configured to provide the logical connection between Communication Manager and the external CTI Server. The **Type** field must be set to “ADJ-IP” for this configuration. The extension number must be valid in the dialplan of the PBX but is otherwise not important.

### Display CTI-Link 1

```

Switch name: ASC_Avaya - CTI LINK
CTI Link: 1
Extension: 2999
Type: ADJ-IP
Name: Avaya CT Link
COR: 1
FEATURE OPTIONS
Event Minimization? n      Special Character for Restricted Number? n

```

The Node-Names form must be configured with the name and IP Address of the CTI-controller server as shown below:

### Display Node-Names

```

Switch name: ASC_Avaya - NODE NAMES
Type      Name          IP Address
IP        CTI            192.168.1 .81
IP        default       0 .0 .0 .0
IP        procr       192.168.1 .22

```

The IP Services must be configured to enable the co-resident DLG option and to define a link to the CTI-Controller server as shown below:

### Display IP Services

```

Switch name: ASC_Avaya - IP SERVICES
Service   Enabled   Local      Local      Remote     Remote
Type      Type      Node       Port       Node       Port
DLG       y         procr      5678
CTI Link  Enabled   Client Name  Client Link  Client Status
1         y         CTI          1            in use

```

### 3.2. Configure A Trunk To Be Used With Passive Monitoring

No special configuration of the E1 trunk to allow passive monitoring is required. The details of the DS1, Signaling Group, and Trunk Group configuration are provided for information only and will vary based on customer needs. The tapped trunk was connected to a Euro-ISDN 30 service from British Telecom.

#### Display DS1 1v2

```
Switch name: ASC_Avaya - DS1 CIRCUIT PACK

Location: 001V2                               Name: BT ISDN Link
Bit Rate: 2.048                               Line Coding: hdb3

Signaling Mode: isdn-pri
Connect: network
TN-C7 Long Timers? n                          Country Protocol: etsi
Interworking Message: PROGRESS                Protocol Version: b
Interface Companding: alaw                    CRC? y
Idle Code: 01010100
                                           DCP/Analog Bearer Capability: 3.1kHz

                                           T303 Timer(sec): 4

Slip Detection? n                             Near-end CSU Type: other
```

#### Display Signaling Group 91

```
Switch name: ASC_Avaya - SIGNALING GROUP

Group Number: 91                             Group Type: isdn-pri
Associated Signaling? y                      Max number of NCA TSC: 0
Primary D-Channel: 001V216                  Max number of CA TSC: 0
                                           Trunk Group for NCA TSC: 91

Trunk Group for Channel Selection: 91
Supplementary Service Protocol: a           Network Call Transfer? n
```

#### Display Trunk Group 91

```
Switch name: ASC_Avaya - TRUNK GROUP

Group Number: 91                             Group Type: isdn          CDR Reports: y
Group Name: BT ISDN Link                     COR: 1                   TN: 1           TAC: 791
Direction: two-way                          Outgoing Display? n     Carrier Medium: PRI/BRI
Dial Access? y                               Busy Threshold: 255     Night Service:
Queue Length: 0
Service Type: public-ntwrk                   Auth Code? n           TestCall ITC: rest
                                           Far End Test Line No:

TestCall BCC: 4
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): enbloc/overlap

Trunk Hunt: cyclical

                                           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
```

TRUNK FEATURES

ACA Assignment? n Measured: none Wideband Support? n  
 Maintenance Tests? y  
 Data Restriction? n NCA-TSC Trunk Member: 1  
 Send Name: y Send Calling Number: y  
 Used for DCS? 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? y  
 Send UUI IE? y Modify Tandem Calling Number? n  
 Send UCID? n  
 Send Codeset 6/7 LAI IE? y Dsl Echo Cancellation? n  
 US NI Delayed Calling Name Update? n  
 SBS? n Network (Japan) Needs Connect Before Disconnect? n

INCOMING CALL HANDLING TREATMENT

Service/ Feature	Called Len	Called Number	Del	Insert	Per Call CPN/BN	Night Serv
public-ntwrk	6	5474	4	10		

TRUNK GROUP

Administered Members (min/max): 1/30

GROUP MEMBER ASSIGNMENTS

Total Administered Members: 30

Port	Code	Sfx	Name	Night	Sig	Grp
1:	001V201	MM	710		91	
2:	001V202	MM	710		91	
3:	001V203	MM	710		91	
4:	001V204	MM	710		91	
5:	001V205	MM	710		91	
6:	001V206	MM	710		91	
7:	001V207	MM	710		91	
8:	001V208	MM	710		91	
9:	001V209	MM	710		91	
10:	001V210	MM	710		91	
11:	001V211	MM	710		91	
12:	001V212	MM	710		91	
13:	001V213	MM	710		91	
14:	001V214	MM	710		91	
15:	001V215	MM	710		91	

TRUNK GROUP

Administered Members (min/max): 1/30

GROUP MEMBER ASSIGNMENTS

Total Administered Members: 30

Port	Code	Sfx	Name	Night	Sig	Grp
16:	001V217	MM	710		91	
17:	001V218	MM	710		91	
18:	001V219	MM	710		91	
19:	001V220	MM	710		91	
20:	001V221	MM	710		91	
21:	001V222	MM	710		91	
22:	001V223	MM	710		91	
23:	001V224	MM	710		91	
24:	001V225	MM	710		91	
25:	001V226	MM	710		91	
26:	001V227	MM	710		91	
27:	001V228	MM	710		91	
28:	001V229	MM	710		91	
29:	001V230	MM	710		91	
30:	001V231	MM	710		91	

### 3.3. Configure the Active Line-Side E1 Trunk

Station-Side monitoring is provided using a Line-Side E1 trunk. This configuration allows up to 30 virtual extensions to be configured with the real hardware being at the remote end of an E1 link. In this configuration, the far-end is the ASC Marathon Evolution server. This allows the ASC server to record audio streams by using CTI to Single-Step Conference one of these virtual extensions.

A Line-Side E1 trunk does not have an associated Signaling Group or Trunk Group, but is simply administered as a DS1 card and extensions.

The DS1 configuration and a single extension ( 2001 ) are shown below. For the testing, extensions 2001 through 2030 were configured with only the extension number and port being different for each virtual device:

#### Display DS1 1v3

```
Switch name: ASC_Avaya - DS1 CIRCUIT PACK

Location: 001V3                               Name: SO Monitor 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
```

#### Display Station 2001

```
Switch name: ASC_Avaya - STATION

Extension: 2001                                Lock Messages? n        BCC: 0
Type: DS1FD                                   Security Code:          TN: 1
Port: 001V301                                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
```

## 4. Configure the Avaya P333T Ethernet Switch

No configuration of the P333T Ethernet Switch was required.

## 5. Configure the ASC CTI-Controller Server

This server consists of both an installed version of Avaya Computer Telephony and the ASC CTI-Controller software. The former is well defined in the Avaya documentation supplied with the server software. The latter operates as a slave process to the Marathon Evolution software. As such, it does not require any configuration.

## 6. Configure the ASC Marathon Evolution Server

The Marathon Evolution Server is supplied pre-installed with a copy of the Marathon Evolution software and suitable defaults. A Web interface is used to configure the solution.

The default values chosen for the majority of the application mean that little configuration is required for the recorder to perform the bulk recording used in the integration testing. In production environments, additional configuration may be required, reflecting unique customer requirements.

For the integration testing, the first 30 recording channels were configured for Active Station-Side recording and the second 30 channels were configured for Passive Trunk-Side recording.

Please refer to ASC's website for additional information including Web-Based Installation Training [3].

### 6.1. Configure the Station-Side Recording Channels

Channel 1 is shown in Figure 2 and Figure 3. Note that this is actually one screen, separated into two figures.

The "RecordStartMode" needs to be set to "HOST" meaning that the external CTI-Controller application is responsible for supplying event information indicating that a voice call should be recorded.

The input parameters "InputSource1" and "InputType1" must be configured correctly. A value of "PCM30" for InputSource1 identifies the physical E1 card to be used for recording. A value of "PRI\_ACTIVE\_TIMESLOT" for InputType1 means that the recorder is terminating an E1 trunk in an active fashion. The InputSource1 field must correspond to the timeslot for the channel. Hence for the example shown, which is channel 1, the value is also 1.



ASC DataManager-Portal - Microsoft Internet Explorer

Address: http://192.168.1.85/ADM/index.php?language=656e5f4742

## ASC DataManager

- ASC DataManager
- User Administration
- Configuration
  - System
  - Alarm notification
  - Channels**
  - Rule based recording
  - Recorder information
- Archive Client
- SDDM Client
- Registry
- Versions

### Channels

State	ChannelDescription	ChannelID
OK	Channel 001	409NMW1001
OK	Channel 002	409NMW1002
OK	Channel 003	409NMW1003
OK	Channel 004	409NMW1004
OK	Channel 005	409NMW1005

### Configuration of Channel 001

State	Name	Description	Value(s) (De-/Select all)
	<i>RecordStartMode</i>	Start recording by:	<input type="checkbox"/> HOST (External application) <input checked="" type="checkbox"/> <input type="checkbox"/> CONTINUOUS (Always recording.) <input type="checkbox"/> <input type="checkbox"/> VOX (Signal level) <input type="checkbox"/> <input type="checkbox"/> COR (Contact operation) <input type="checkbox"/>
	<i>RecordStopMode</i>	Stop recording by:	<input checked="" type="checkbox"/> - (Use the triggers from recording start) <input type="checkbox"/> <input type="checkbox"/> HOST (External application) <input type="checkbox"/> <input type="checkbox"/> VOX (Signal level) <input type="checkbox"/> <input type="checkbox"/> COR (Contact operation) <input type="checkbox"/>
	<i>StorageMode</i>	Storage mode	<input checked="" type="checkbox"/> COMPLETE_CALL_INFO (Store when all call) <input type="checkbox"/>
	<i>VoxLevel</i>	Threshold value for sensitivity of signal detection. Range from 0dB (max sensitive) to 62dB (least sensitive).	<input checked="" type="checkbox"/> 20 dB <input type="checkbox"/>
	<i>Timespan_Until_Deletion</i>	Time to keep a call in the database (YY:MM:DD:HH:mm).	<input checked="" type="checkbox"/> 99:00:00:00 <input type="checkbox"/>
	<i>CLJEnable</i>	Enable CLI detection	<input checked="" type="checkbox"/> No <input type="checkbox"/>
	<i>DTMFEnable</i>	Enable DTMF detection	<input checked="" type="checkbox"/> Yes <input type="checkbox"/>
	<i>PreTrigger</i>	PreTrigger to use by record start. [0..51]*100ms.	<input checked="" type="checkbox"/> 20 <input type="checkbox"/>
	<i>Compression</i>	Compression to use for audio data	<input checked="" type="checkbox"/> ADPCM_16 (16 kbps) <input type="checkbox"/>
	<i>VoxPostTime</i>	Minimum duration for silence before recording stop in conjunction with VOX trigger. 100ms+[0..1023]*100ms	<input checked="" type="checkbox"/> 79 <input type="checkbox"/>
	<i>VoxTimeMin</i>	Minimum signal duration before recording start in conjunction with VOX trigger.	<input checked="" type="checkbox"/> 1000 ms <input type="checkbox"/>
	<i>IdlePostTime</i>	Minimum duration for silence before recording stop in conjunction with IDLE WORD trigger. 100ms+[0..1023]*100ms	<input checked="" type="checkbox"/> 49 <input type="checkbox"/>
	<i>IdleTimeMin</i>	Minimum signal duration before recording start in conjunction with IDLE WORD trigger.	<input checked="" type="checkbox"/> 500 ms <input type="checkbox"/>

ASC DataManager Copyright ASC telecom AG 2004. All rights reserved. Version 1.03.04

Session will be closed in 29:50 without activity!

**Figure 2: Configuring a Line-Side E1 Trunk Recording Channel – Part I**

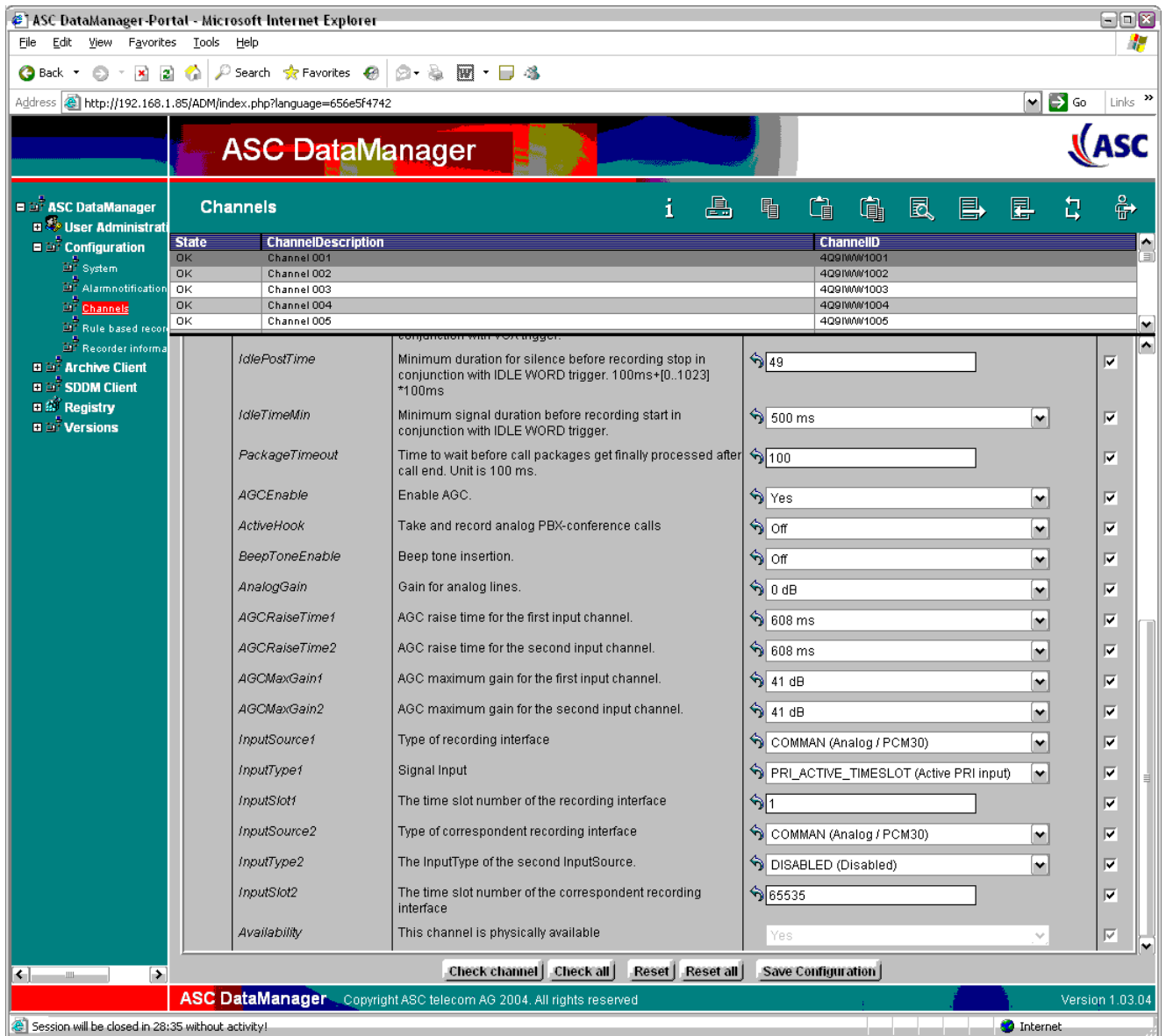


Figure 3: Configuring a Line-Side E1 Trunk Recording Channel – Part II

## 6.2. Configure the Passive Trunk Tapping Recording Channels

Channel 31 is shown in Figure 4 and Figure 5. Note that this is actually one screen, separated into two figures.

The "RecordStartMode" needs to be set to "HOST" meaning that the external CTI-Controller application is responsible for supplying event information indicating that a voice call should be recorded.

The input parameters “InputSource1” and “InputType1” must be configured correctly. A value of “DP\_XXXX Passive” for InputSource1 means that a physical E1 card is used for recording, but that no signaling should be applied to this trunk. A value of “AUDIO\_STREAM” for InputType1 means that the recorder is not terminating an E1 trunk in an active fashion, and therefore should not monitor the signaling. The InputSource1 field must correspond to the actual timeslot for the channel.

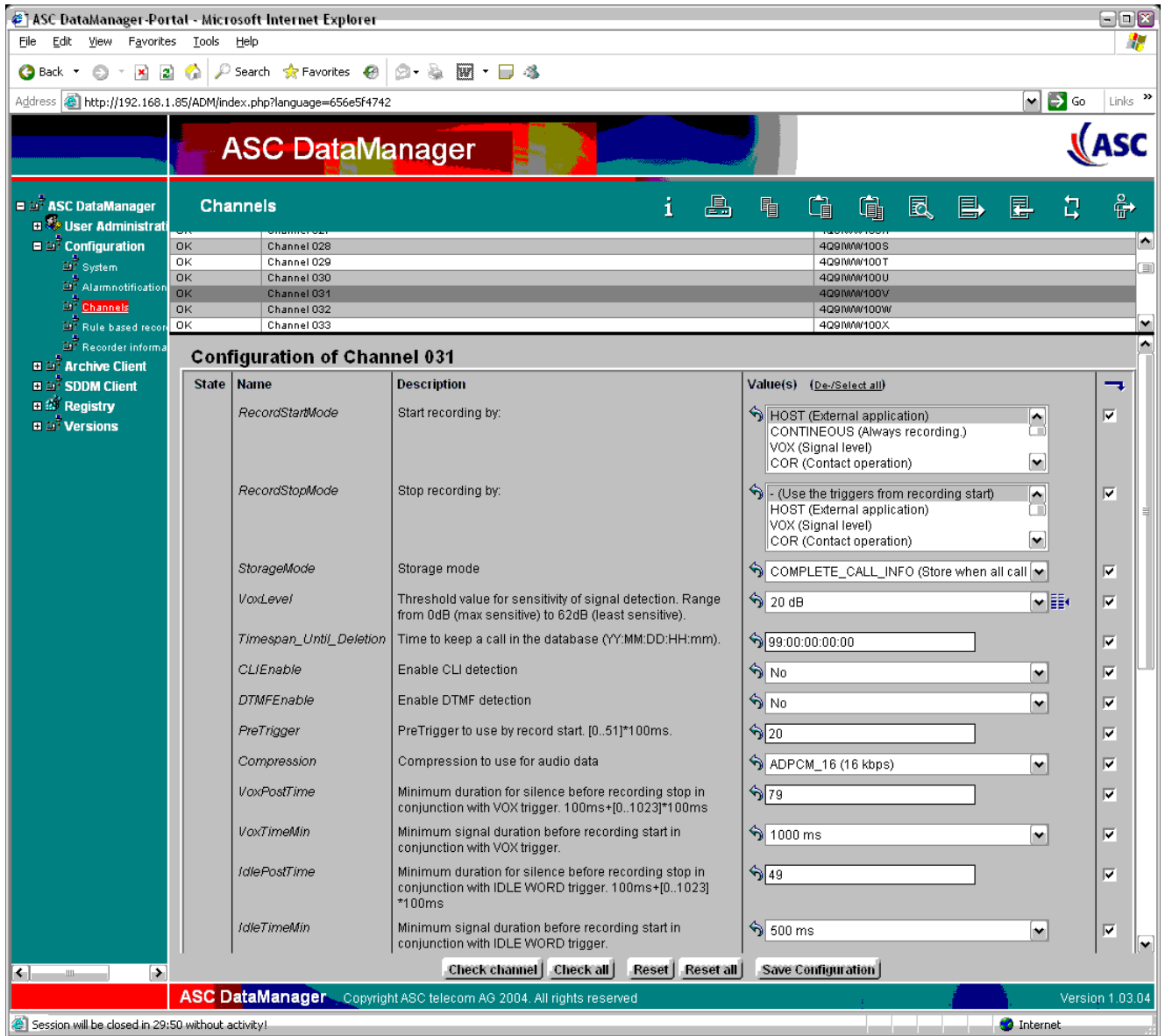


Figure 4: Configuring a Passive E1 Trunk Tapping Recording Channel – Part I

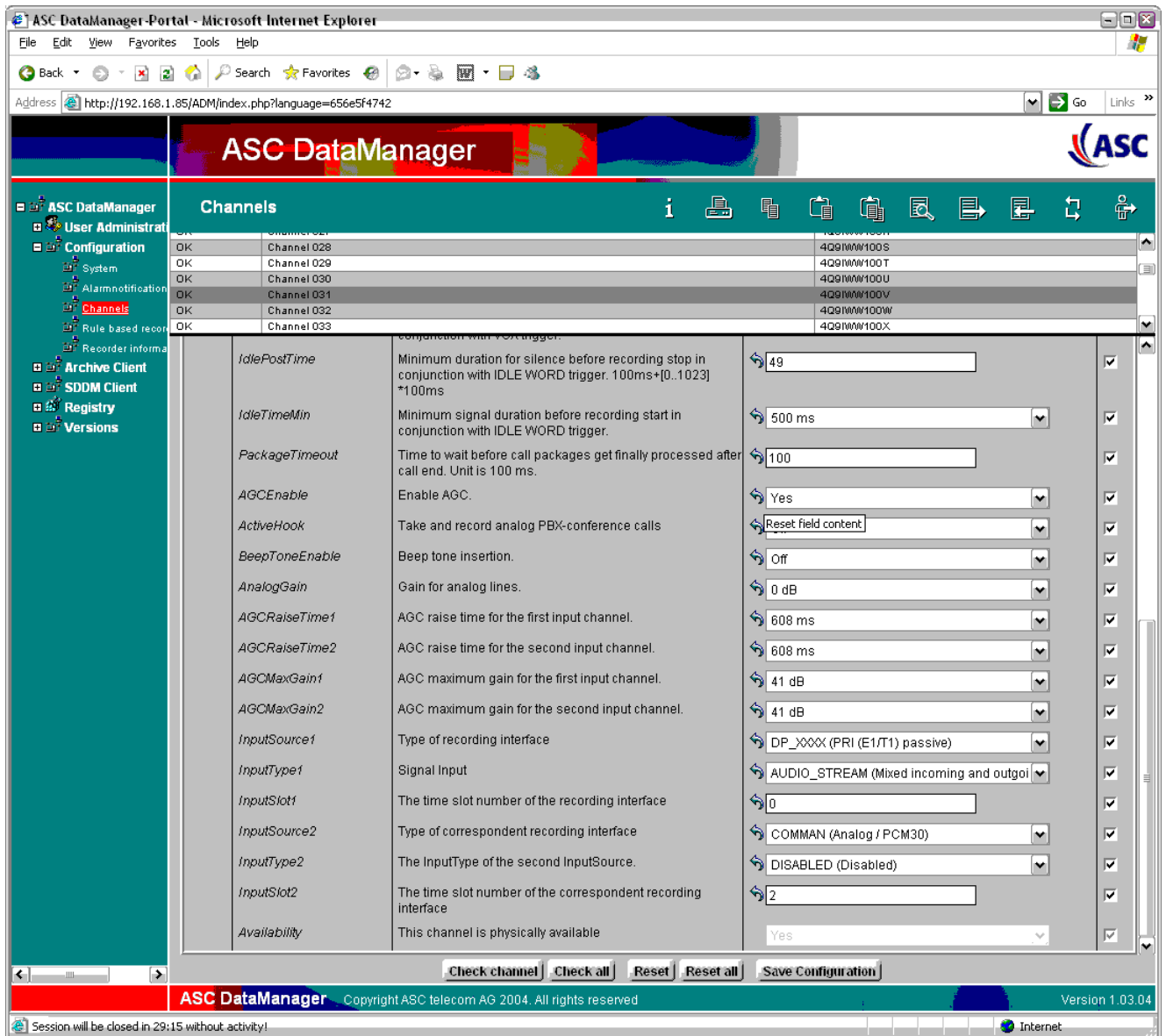


Figure 5: Configuring a Passive E1 Trunk Tapping Recording Channel – Part II

## 7. Interoperability Compliance Testing

### 7.1. General Test Approach

Testing included validation of correct operation of typical Voice Recording functions including Inbound, Outbound, Blind Transfer, Attended Transfer, and Conference calls. These tests were repeated for both tested recording modes. Light load testing and link integrity testing were also carried out.

## 7.2. Test Results

All tests passed.

## 8. Verification Steps

The following verification steps can be used to isolate problems in the field and to ensure that the CTI link is correctly passing data between the various components of the solution.

Since the CTI-Controller server contains an Avaya CT Server, the following can be used to verify the connectivity from the CTI-Controller server to Avaya Communication Manager.

1. Avaya CT is shipped with a simple TSAPI application called "TSTEST". This utility allows connection to a server and the origination of a call to verify CTI connectivity. There is also a small application called "TSSPY" which can be used to trace the messages to and from the Avaya CT Server. These two in conjunction are able to ensure that the CTI link is operating correctly. Hence the only required verification step for CTI is to use "TSTEST" to initiate a call from one known physical extension to another. Having made the CTI call, ensure that the physical devices are indeed trying to call each other, manually answer the call, and then use "TSTEST" to clear the call.
2. ASC has an Error Manager, which monitors the status of all of the configured trunks. This will alarm if any one is reporting a failure. This can be easily used to validate physical connectivity.
3. The status of the Line-Side E1 extensions, if this recording mode is being used, can be tested from Communication Manager as with a conventional station. If the status is "disconnected", then the E1 trunk is not operating correctly. The "Test DS1 xxxx" command can be used to check that the DS1 card is connected correctly. The first test is physical connectivity. Please refer to the Avaya Communication Manager manuals for details of other error messages that may be displayed when using this command.

## 9. Support

If technical support is required for the ASC Evolution Marathon solution, then please contact their Technical Support Hotline:

Email: [hotline@asc.de](mailto:hotline@asc.de)

## 10. Conclusion

These Application Notes describe the configuration steps required for ASC Marathon Evolution to successfully interoperate with Avaya Communication Manager 2.2. An Avaya S8300 Media Server within an Avaya G350 Media Gateway running Avaya Communication Manager 2.2 was used as the hosting PBX. Features and functionality were validated and performance testing was conducted in order to verify operation under light load. The configuration described in these Application Notes has been successfully compliance tested.

## 11. Additional References

- [1] Administrators Guide for Communication Manager (Doc ID: 555-233-506) can be found at <http://support.avaya.com>.
- [2] Installation Guide for Avaya Computer Telephony can also be found at <http://support.avaya.com>.
- [3] Additional Product Information can be obtained from ASC's website at [http://www.asctelecom.com/english/index\\_e.html](http://www.asctelecom.com/english/index_e.html) including Web-based installation training.

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