Avaya Aura™ Communication Manager TSAPI

Integration Guide

Encore Workforce Optimization Solution
Version 7.2 or later

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For Dealer and Customer Use Only

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Introduction

The Encore system integrates with the Avaya PBX. This integration allows the Encore system to successfully perform the following functions:

- Audio Collection – Capture the audio that needs to be recorded.
- Recording Control – Receive the necessary events that signal when the Encore must start and stop recording.
- Data Capture – Receive data associated with the call.

The Encore system can record calls on an Avaya system without the TSAPI integration, but the recording controls and data capture are limited; configuration for this integration is not covered in this document.

Supported Data Capture

The following is a list of the supported data elements that can be collected with each recording. Not every element is applicable for each call. For a description of each data element, refer to "Appendix 1: Glossary" on page 33.

- ACD Name
- ACD Number
- Agent ID*
- ANI
- Call ID
- Call Direction
- Call Type
- Consultation Call
- Dialed Number
- DNIS
- Extension
- Other Call ID
- Other Party Name
- Other Party Number
- Recorded Party Name
- Recorded Party Number
- Trunk
- Universal Call ID
- User to User Information
- Recorded Party Disconnect

*If using the internal ACD feature of Avaya Communication Manager, CT Gateway must monitor the ACD split or the EAS skill extension in which the agent is logged in to capture this information.

The Avaya Aura Contact Center (AACC) does not provide recording control but does capture additional data. The following additional data can be captured for the first monitored agent that answers an ACD call and is logged into the AACC system:

- Agent ID
- Agent Name^
- CDN
- Skillset
- AACC Caller ID
- AACC Caller Display

^This is stored in the Recorded Party field; it overwrites the Recorded Party Name that was captured by TSAPI for ACD calls.

Supported Recording Features

The following matrix should be used to determine which audio collection is best for your business needs. If you find that more than one collection method will work for you, talk to your Encore representative about which method is more cost-effective. For a description of each feature, refer to "Appendix 1: Glossary" on page 33.

RECORDING FEATURE	AUDIO COLLECTION METHOD				
	STATION-SIDE TDM	TRUNK-SIDE TDM [5]	STATION-SIDE RTP PACKET CAPTURE (PASSIVE INTERFACE) [5]	SUBSCRIPTION-BASED DMCC AUDIO STREAM	TRUNK-SIDE SIP PACKET CAPTURE [5]
Max. Recording Ports per Server [1]	192	288	500	500	500
Record External Calls	YES	YES	YES	YES	YES
Record Internal Calls	YES		YES	YES	
Record Encrypted Calls	N/A	YES		YES	YES [4]
Suspend/Resume on Hold	YES	YES	YES	YES	
ACD Split Recording [2,3]		YES	YES	YES	YES
Related Call Lookup	YES	YES	YES	YES	
AACC Data Collection	YES			YES	
Capture User to User Information	YES	YES	YES	YES	

1. Small Business Servers are limited to 72 ports.

2. Cannot be used if any audio collection method is used in conjunction with Station-side TDM.

3. Cannot be used if more than one audio collection method is used.

4. If the encryption occurs at the station, Encore can record the encrypted calls. If the encryption occurs at the trunk Encore cannot record encrypted calls.

5. Cannot be used if AACC is used to collect additional data.

Software and Hardware Requirements

SYSTEM	SOFTWARE REQUIREMENTS
Avaya system	• All audio collection methods ○ Avaya Aura Communication Manager v5.2.x or later ○ Avaya Aura Application Enablement Services (AES) v5.2.x or later ○ AES TSAPI Client v6.3.3 ○ TSAPI Basic User Licenses; to calculate the number needed read below. – If using the Encore feature that allows you to record all agents

SYSTEM	SOFTWARE REQUIREMENTS
	logged into an ACD skill: • Calculate the number of simultaneous agents logged into the ACD skill. If more than one skill is monitored, then the number of simultaneous agents logged in across all monitored skills. • Calculate the number of supervisors or agents to be recorded that do not log into an ACD skill. – If not using the ACD skill feature, you need one license for each phone used by agents and supervisors to be recorded. • Station-side TDM ○ If using AACC, use AACC v6.2 • Station-side RTP Packet Capture ○ VoIP phones must use the G.711MU, G.711A, and G.729 codec types • Subscription-based DMCC Audio Stream ○ For each concurrent Encore Recording License, you need the following: – 1 DMCC_DMC license or 1 IP_API_A license (Effective with Communication Manager Release 6.0; all new DMCC licenses will be added only to the AE Services license file VALUE_DMCC_DMC field.) – 1 IP_STA license – 1 STA license – 1 MedPro resource – 1 TDM slot resource ○ Emulated softphones for the recorded ports must use the G.711MU codec. The total number of emulated softphones must equal the total number of ports that can be simultaneously recorded (this is noted in the **Maximum number of ports** field in the Softphone Recording Server node which is configured in the CenterPlus Server Configuration; see the *Avaya Aura Communication Manager TSAPI Installation Addendum*). ○ If using AACC, use AACC v6.2
Encore system	• If using Station-side TDM and AACC, use the following: ○ AACCBridge.exe 1.1 ○ EncoreUtilities.dll v1.2 ○ Log4Dvs.dll v3.0.6.3471 • If using Subscription-based DMCC Audio Stream and AACC, use the following: ○ AACCBridge.exe 1.1 ○ EncoreUtilities.dll v1.2 ○ Log4Dvs.dll v3.0.6.3471 ○ .NET Framework 4.0 or .NET Framework 4.0 Client Profile

SYSTEM	HARDWARE REQUIREMENTS

Avaya system	• Station-side TDM
	  ○ Must allow Encore to tap at the punchdown block
	• Trunk-side TDM
	  ○ Must allow Encore to tap at the trunk
	• Station-side RTP Packet Capture
	  ○ Span port on network to route all RTP traffic for recorded stations to Encore server
	  ○ DHCP IP address reservation or static IP assignment for each station to be recorded
	• Subscription-based DMCC Audio Stream
	  ○ Each recording consumes an IP media processor; additional CLAN may be required.
	• Trunk-side SIP Packet Capture
	  ○ This depends on the customer's environment; see the Note in the "Trunk-side SIP Packet Capture" section on page 13.

Encore system	• Station-side TDM
	  ○ AudioCodes NGX PCIe card
	• Trunk-side TDM
	  ○ AudioCodes DP PCIe card

Compliance Testing

As of January 2020, Encore has been compliance tested and is approved to record using DMCC Service Observe and the TSAPI interface with the following Avaya equipment and software.

EQUIPMENT	SOFTWARE
Avaya Aura® Communication Manager in Virtual Environment	8.1 (8.1.0.1.1.890.25517)
Avaya G650 Media Gateway	N/A
Avaya Aura® Media Server in Virtual Environment	8.0.1.121
Avaya Aura® Application Enablement Services in Virtual Environment	8.1 (8.1.0.0.0.9-1)
Avaya Aura® Session Manager in Virtual Environment	8.1 (8.1.0.0.810007)
Avaya Aura® System Manager in Virtual Environment	8.1 (8.1.0.0.079814)
Avaya 1608-I IP Deskphone	1.3120
Avaya 9611G IP Deskphone (H.323)	6.8202
Avaya 9641G IP Deskphone (SIP)	7.1.6.1.3
DVSAnalytics Encore on Windows Server 2016 Avaya TSAPI Windows Client (csta32.dll) Avaya DMCC XML	7.1 Standard 8.1.0.9 6.1

As of December 2017, Encore has been compliance tested and is approved to record using DMCC Service Observe and the TSAPI interface with the following Avaya equipment and software.

EQUIPMENT	SOFTWARE
Avaya Aura® Communication Manager in Virtual Environment	7.1.1 (7.1.1.0.0.532.23985)
Avaya G650 Media Gateway	N/A
Avaya Aura® Media Server in Virtual Environment	7.8.0.333
Avaya Aura® Application Enablement Services in Virtual Environment	7.1.1 (7.1.1.0.0.5-0)
Avaya Aura® Session Manager in Virtual Environment	7.1.1 (7.1.1.0.711008)
Avaya Aura® System Manager in Virtual Environment	7.1 .1 (7.1.1.0.046931)
Avaya 9611G & 9641G IP Deskphone (H.323)	6.6506
Avaya 9621G IP Deskphone (SIP)	7.1.0.1.1
DVSAnalytics Encore on Windows Server 2012 R2 Avaya TSAPI Windows Client (csta32.dll) Avaya DMCC XML	6.0.6 Standard 6.3.3.103 6.1

As of October 2016, Encore has been compliance tested and is approved to record using DMCC Service Observe and the TSAPI interface with the following Avaya equipment and software.

EQUIPMENT	SOFTWARE
Avaya Aura® Communication Manager in Virtual Environment	7.0.1.1 (7.0.1.1.0.441.23169)
Avaya G650 Media Gateway	N/A
Avaya Aura® Media Server in Virtual Environment	7.7.0.334
Avaya Aura® Application Enablement Services in Virtual Environment	7.0.1 (7.0.1.0.2.15-0)
Avaya Aura® Session Manager in Virtual Environment	7.0 .1 (7.0.1.0.701007)
Avaya Aura® System Manager in Virtual Environment	7.0 .1 (7.0.1.0.064859)
Avaya 9611G & 9641G IP Deskphone (H.323)	6.6229
Avaya 9621G IP Deskphone (SIP)	7.0.1.1.5
DVSAnalytics Encore on Windows Server 2012 R2 Avaya TSAPI Windows Client (csta32.dll) Avaya DMCC XML	6.0.5 Standard 6.3.3.103 6.1

As of March 2014, Encore has been compliance tested and is approved to record using digital station taps or DMCC Service Observe with the following Avaya Aura® Contact Center equipment and software.

EQUIPMENT	SOFTWARE
Avaya Aura® Communication Manager running on S8800 Server with an Avaya G650 Media Gateway	6.3
Avaya Aura® System Manager running on S8800 Server	6.3
Avaya Aura® Session Manager running on S8800 Server	6.3
Avaya Aura® Application Enablement Services running on S8800 Server	6.3
Avaya Aura® Contact Center running on S8800 Server	6.3
Avaya 9670G IP Deskphone (H.323)	S3.1
Avaya 9608 IP Deskphone (H.323)	6.2313

Documentation Overview

This document provides integration information for a specific phone system. It helps a user to understand the features and benefits of the integration as well as what needs to be configured on the phone system. Conventions used in this guide include:

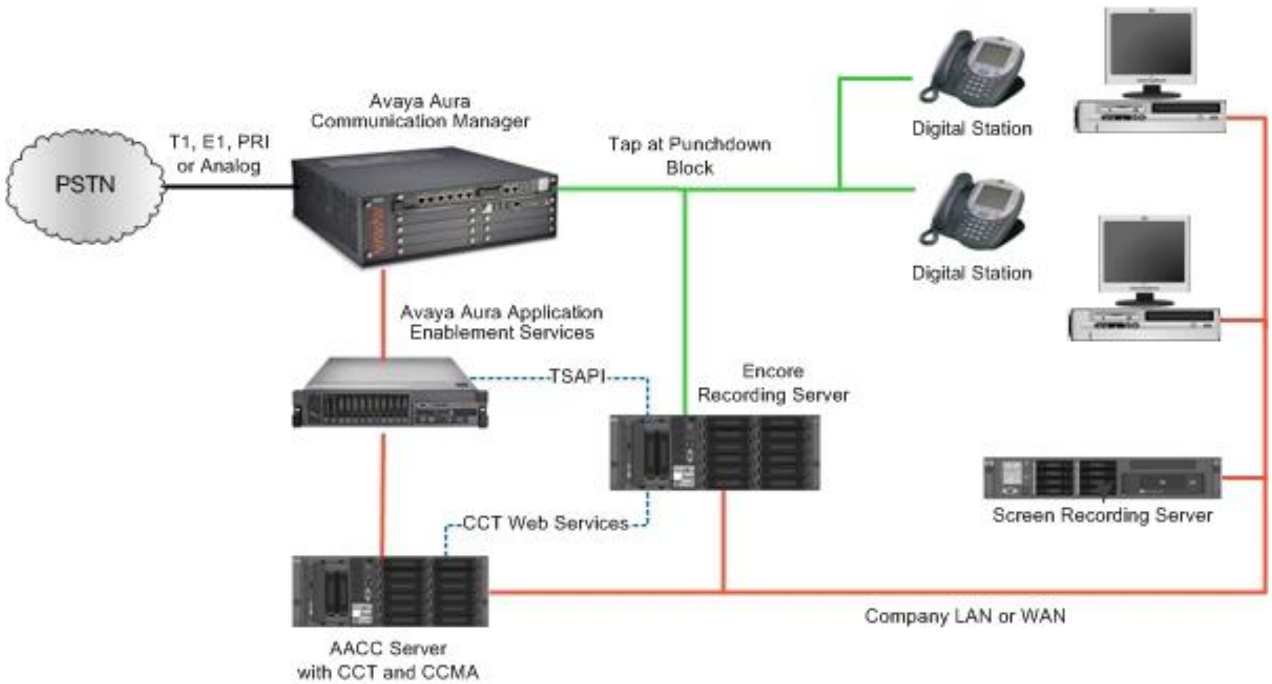
1. Computer commands needed to complete a task appear like this: **Sample** (in black)
2. Keyboard strokes that need to be entered appear like this: [Sample]

Overview

This section provides an overview of each audio collection method. For simplicity sake, the diagrams only display a single Encore server but there can be multiple Encore servers depending on the number of stations to be recorded.

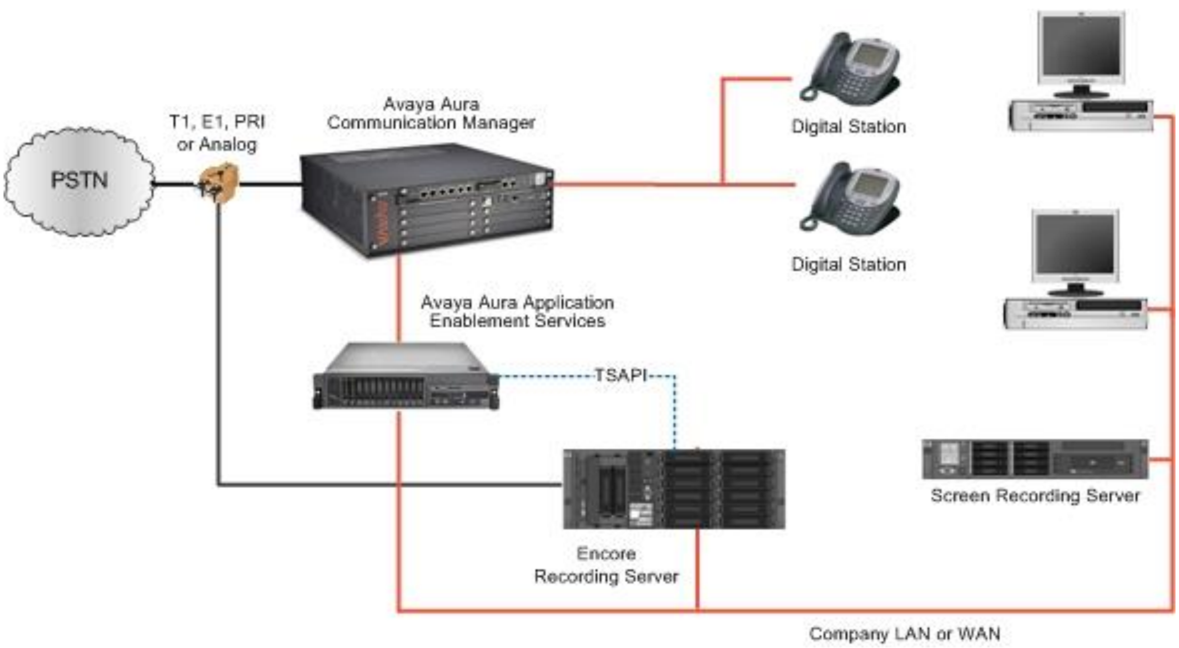
Station-side TDM

The Station-side TDM method uses a passive tap on the phones connected to the Avaya Aura Communication Manager to collect audio. These may be analog or digital phone sets. This passive tap is connected to the recording boards in the Encore server. The Avaya Aura Application Enablement Services monitors events on the Communication Manager and forwards the events to the AES TSAPI client installed on the Encore server. Based on events received from the TSAPI interface, the Encore server starts and stops recording, collects the audio on the recording boards, and collects the data associated with the call.


Avaya Aura
Communication Manager
T1, E1, PRI
or Analog
PSTN
Tap at Punchdown
Block
Digital Station
Digital Station
Avaya Aura Application
Enablement Services
TSAPI
Encore
Recording Server
Screen Recording Server
CCT Web Services
Company LAN or WAN
AACC Server
with CCT and CCMA

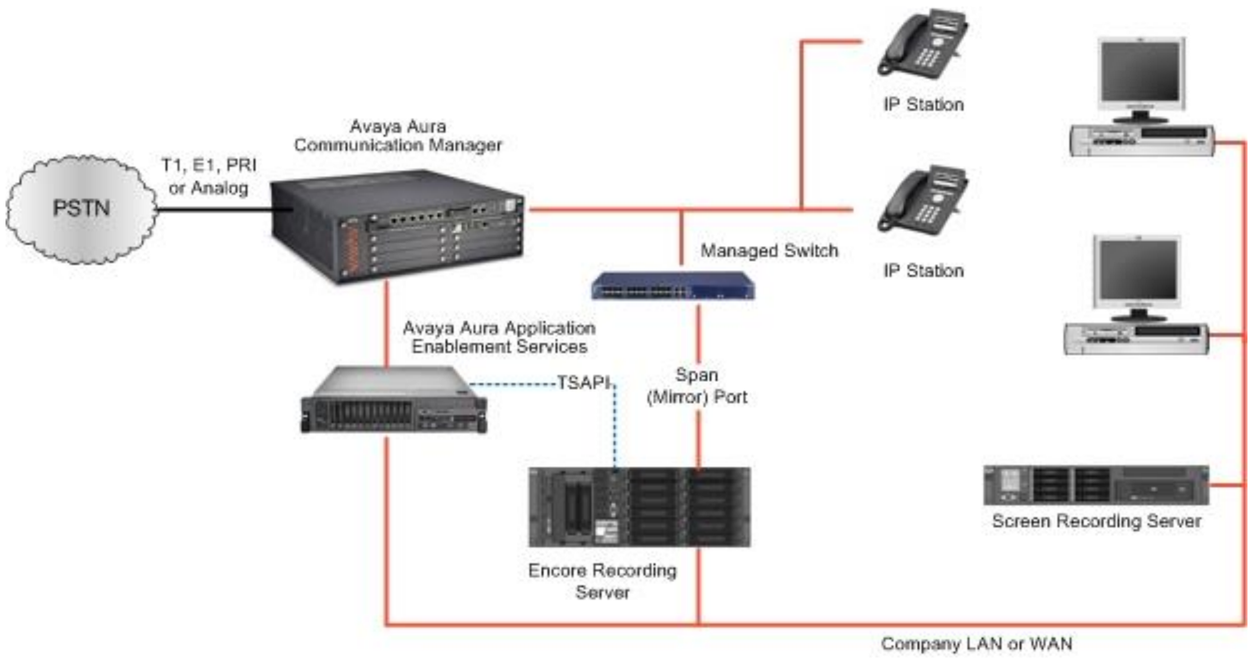
Trunk-side TDM

The Trunk-side TDM method uses a passive tap on the telephony trunks that connect the Avaya Aura Communication Manager to the PSTN. The trunks can be T1, E1, or Analog. This passive tap is connected to the recording boards in the Encore server. The audio is collected via the passive tap. The Avaya Aura Application Enablement Services monitors events on the Communication Manager and forwards the events to the AES TSAPI client installed on the Encore server. Based on events received from the TSAPI interface, the Encore server collects the audio on the recording boards and the data associated with the call.



Station-side RTP Packet Capture

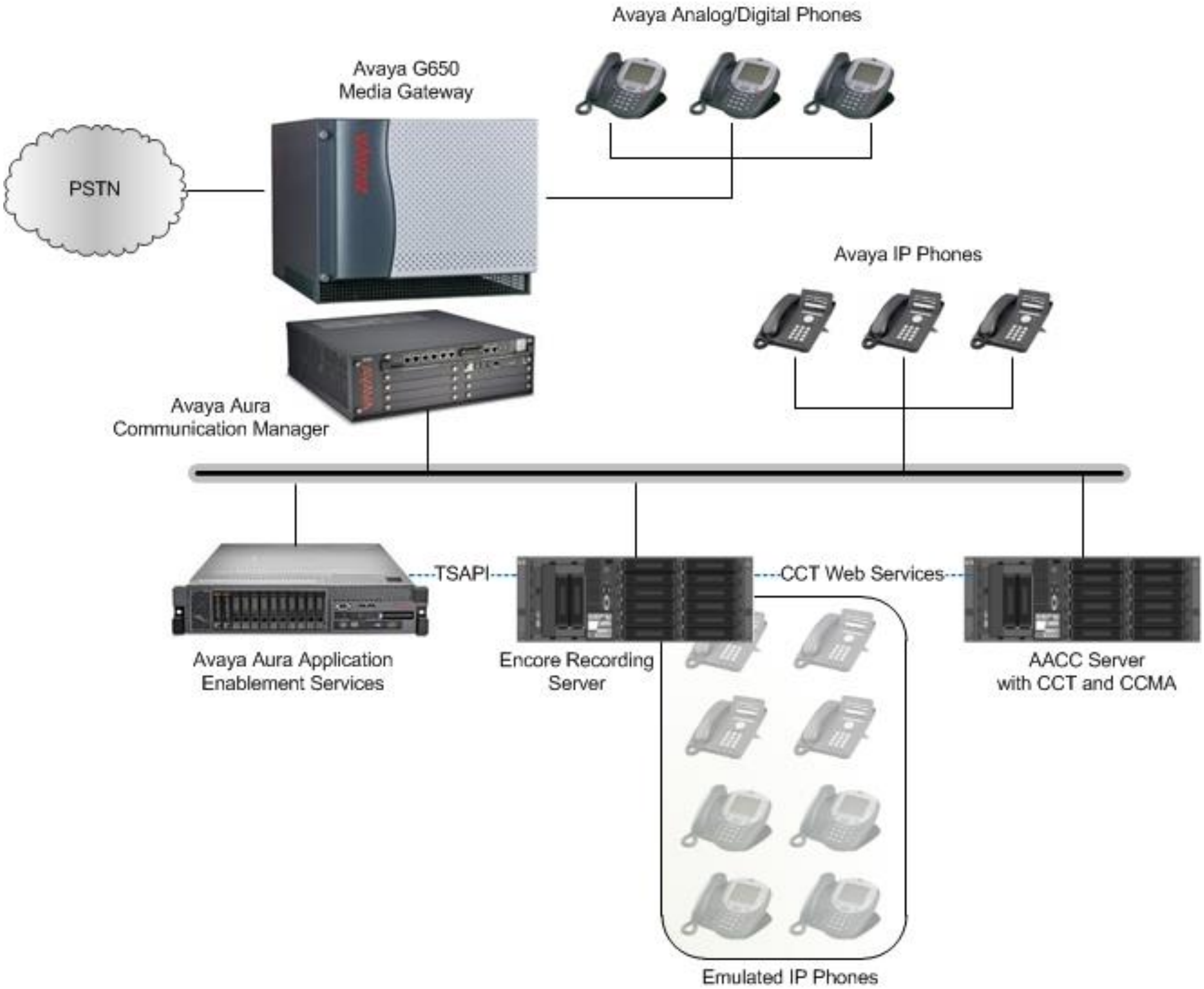
The Station-side RTP Packet Capture method uses a span port to collect the RTP audio packets directly from the network segment that includes the VoIP traffic. The Avaya Aura Application Enablement Services monitors events on the Avaya Aura Communication Manager and forwards the events to the AES TSAPI client installed on the Encore server. Based on events received from the TSAPI interface, the Encore server collects the RTP packets for a specific IP or MAC address and converts the RTP data to an audio recording file. Encore collects data associated with the call from the TSAPI messages.



Subscription-based DMCC Audio Stream

Encore uses TSAPI from the Avaya Aura Application Enablement Services to monitor skill groups and agent stations on the Avaya Aura Communication Manager. It also uses the Service Observing feature via the Avaya Aura Application Enablement Services Device, Media, and Call Control (DMCC) interface to capture the audio associated with the monitored stations.

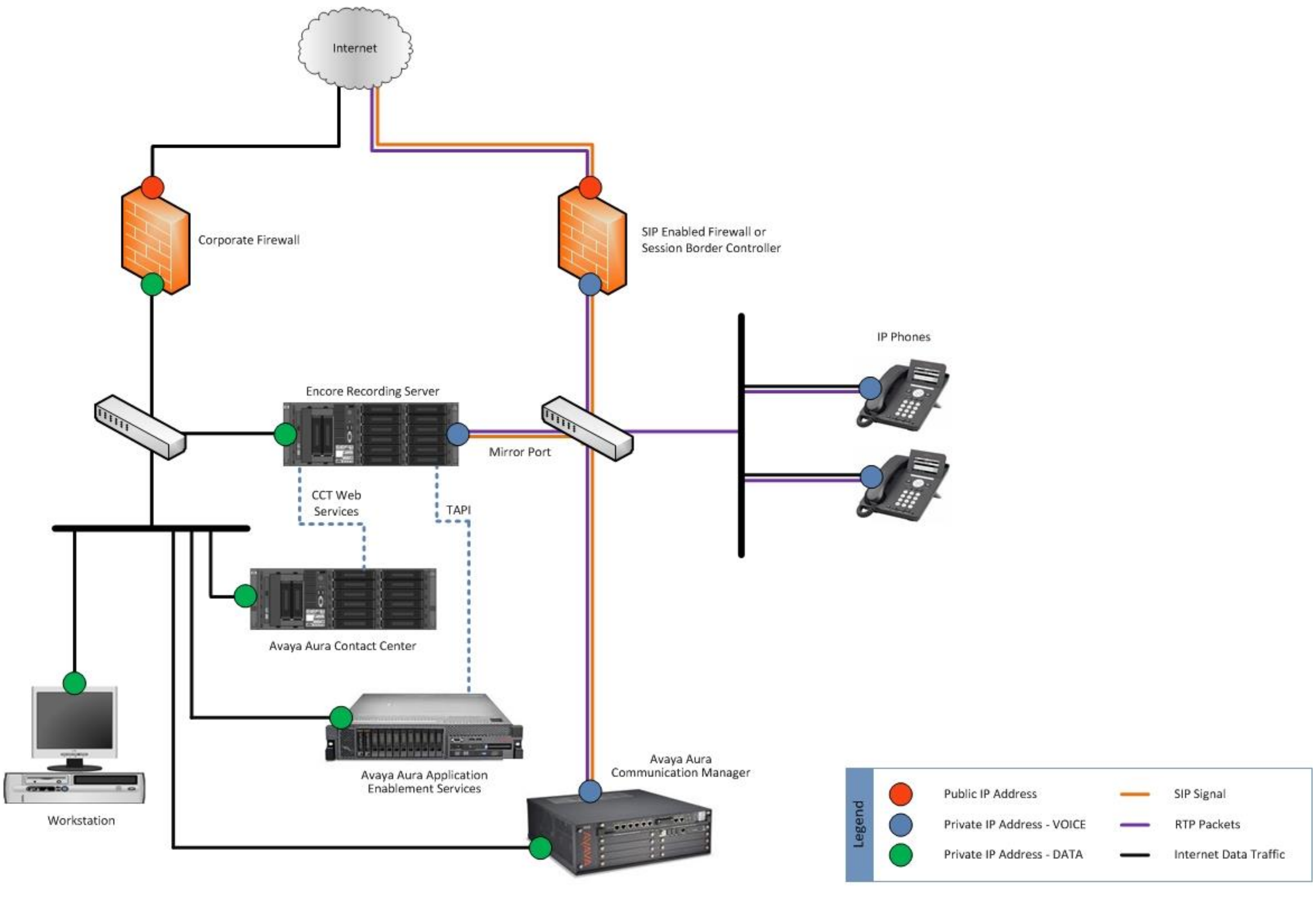
When a call is active on a monitored station, event reports are sent to Encore via TSAPI. Encore starts recording by sending a Service Observing button press from a virtual IP softphone via the DMCC interface to observe the active call and uses the Media Control Events from the DMCC interface to obtain the audio from the virtual IP softphone. The recording stops based on events received from TSAPI.



Trunk-side SIP Packet Capture

The Trunk-side SIP Packet Capture method uses a span port to collect the SIP and RTP audio packets directly from the network segment that includes SIP trunk traffic. The Avaya Aura Application Enablement Services (AES) monitors events on the Avaya Aura Communication Manager and forwards the events to the AES TSAPI client installed on the Encore server. Based on events received from SIP signaling and the TSAPI interface, the Encore server collects the RTP packets for a specific SIP trunk call and converts the RTP data to an audio recording file. Encore collects data associated with the call from the TSAPI messages when the call is terminated to a station of interest, or from the SIP signaling when the call is terminated elsewhere on the PBX.

> **NOTE** When recording SIP trunks, DVSAnalytics prefers that all SIP trunk traffic goes through a Session Border Controller (SBC), such as a Cisco CUBE, Ingate SIParator, etc. The SBC's LAN-side port must terminate to a network switch that can provide a SPAN/mirror port to the Encore server.
>
> If an SBC cannot be provided, then the SIP trunk provider must be able to guarantee that a single IP Address will be used for all SIP and RTP media packets for both inbound and outbound calls. The LAN-side port of the device used for SIP trunk traffic must terminate to a network switch that can provide a SPAN/mirror port to the Encore server.



Configure Avaya System

The steps to configure the Avaya system are listed in this section. It is assumed that the reader has a working knowledge of Avaya system and only needs specific configuration assistance.

Some steps and screenshots were taken from the *Application Notes for dvsAnalytics Encore 7.1 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Application Enablement Services 8.1 using Service Observing* document provided by Avaya Inc.

Configure Avaya Aura Communication Manager

Complete the steps in this section for the audio collection method you are using.

Complete for all systems

Add a CTI link using the “add cti-link n” command, where “n” is an available CTI link number. Enter an available extension number in the **Extension** field. Note that the CTI link number and extension number may vary. Enter **ADJ-IP** in the **Type** field, and a descriptive name in the **Name** field. Default values may be used in the remaining fields.

```
add cti-link 1                                                Page   1 of   3
                                    CTI LINK
 CTI Link: 1
Extension: 60111
     Type: ADJ-IP
                                                                  COR: 1
     Name: AES CTI Link
Unicode Name? n
```

The Universal Call ID (UCID) feature must be enabled so that it is received in the event messages. This feature may have already been enabled, but it is important to check the following settings to verify they have been set correctly to enable this feature:

1. On the **Avaya Site Administrator** command window enter this command “change system-parameters features”.
2. Enable the following features:
3. Set **Create Universal Call ID (UCID)** to **y**. This allows a UCID to be generated for each call.
4. Set **UCID network Node ID** to **1** (or any number between 1 and 32767, and that is unique to the switch in the network).
5. Set **Send UCID to ASAI** to **y**. This enables the transmission of UCID information.
6. On the **Avaya Site Administrator** command window enter this command “change trunk-group N”.
7. Enable the **Send UCID** field from **Page 3** for each trunk group that is used to communicate between CM systems. This field specifies whether the trunk should transmit UCIDs. Sending UCIDs in a network of CMs allows an application server that works with all of them to know it has already handled a call when the call is transferred to an agent on another switch.

8. Log into the System Access Terminal (SAT) to verify that the Communication Manager license has the proper permissions. Use the “display system-parameters customer-options” command to verify that the **Computer Telephony Adjunct Links** option is set to **y** on **Page 4** of the Avaya system. If this option is not set to **y** then contact your Avaya representative for a proper license file.

```
display system-parameters customer-options                      Page   4 of  12
                                OPTIONAL FEATURES

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

CAUTION If the UCID feature was enabled while CT Gateway was running, restart CT Gateway.

NOTE If using Station-side or Trunk-side TDM audio collection methods, the Configuration Manager setup is complete. Turn to “Configure Avaya Aura Application Enablement Services” on page 20.

Complete for Station-side RTP Packet Capture or Subscription-based DMCC Audio Stream systems

Complete for systems using the ACD Split feature

If the customer is using one or more ACD splits to determine which agent to record, be sure to assign all agents that must be recorded to an ACD split. Then provide the name(s) of the ACD split(s) to your Encore installer.

Administer IP codec set

Use the “change ip-codec-set n” command, where “n” is an existing codec set number used for integration with Encore. Enter the desired audio codec types in the **Audio Codec** fields. Encore supports G.711MU, G.711A, and G.729 codecs. If your network uses a combination of G.711 and G.729, Avaya recommends you work with your Avaya reseller to determine the best codec to use based on adequate resource on the Avaya system to manage the work load. Provide the Codec to your Encore installer.

For customer networks that uses encrypted media, make certain that **none** is included for **Media Encryption**, and that **Encrypted SRTP** is set to **best-effort**, these settings are needed for support of non-encrypted media from the virtual IP softphones used by Encore.

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

                          IP MEDIA PARAMETERS
    Codec Set: 1

    Audio         Silence       Frames    Packet
    Codec         Suppression   Per Pkt   Size(ms)
 1: G.711MU           n           2         20
 2:
 3:
 4:
 5:
 6:
 7:

     Media Encryption                    Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: aes
 3: none
 4:
```

NOTE If using the Station-side RTP Packet Capture method, the Configuration Manager setup is complete. Turn to "Configure Avaya Aura Application Enablement Services" on page 20.

Complete for Subscription-based DMCC Audio Stream systems

Verify service observing

Navigate to **Page 7** and verify that the **Service Observing (Basic)** customer option is set to **y**.

```
display system-parameters customer-options                        Page    7 of   12
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 8.0

                                ACD? y                        Reason Codes? y
                       BCMS (Basic)? y             Service Level Maximizer? n
         BCMS/VuStats Service Level? y           Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? y   Service Observing (Remote/By FAC)? y
                 Business Advocate? n             Service Observing (VDNs)? y
                   Call Work Codes? y                            Timed ACW? y
     DTMF Feedback Signals For VRU? y                   Vectoring (Basic)? y
                  Dynamic Advocate? n               Vectoring (Prompting)? y
```

Set system parameter features

Use the "change system-parameters features" command and navigate to **Page 11**. Set **Service Observing: Warning Tone** to the needed setting per customer requirements, and enable **Allow Two Observers in Same Call**, as shown below.

```
change system-parameters features                               Page  11 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
  EAS
         Expert Agent Selection (EAS) Enabled? y
        Minimum Agent-LoginID Password Length:
          Direct Agent Announcement Extension:                  Delay:
    Message Waiting Lamp Indicates Status For: station
                           Work Mode On Login: aux
  VECTORING
                     Converse First Data Delay: 0      Second Data Delay: 2
                 Converse Signaling Tone(msec): 100        Pause (msec): 70
                       Prompting Timeout(secs): 10
                   Interflow-qpos EWT Threshod: 2
    Reverse Star/Pound Digit For Collect Step? n
          Available Agent Adjustments for BSR? n
                             BSR Tie Strategy: 1st-found
   Store VDN Name in Station's Local Call Log? n
  SERVICE OBSERVING
               Service Observing: Warning Tone? n     or Conference Tone? n
    Allowed with Exclusion: Service Observing? n                 SSC? n
              Allow Two Observers in Same Call? y
```

Service Observe Physical Set – Optional

If the phones to be recorded have bridged call appearances and these bridged call appearances must be recorded, then make this change. Use the "change system special-applications" command and enable **(SA7900) - Service Observe Physical Set** which is located on **page 2**.

```
change system-parameters special-applications                   Page   2 of   9
                             SPECIAL APPLICATIONS

           (SA7710) - Enhanced Display on Redirected Calls? n

  (SA7776) - Display Incoming Digits for ISDN Trunk Groups? n
              (SA7777) - Night Service on DID Trunk Groups? n
                        (SA7778) - Display UUI Information? n
                           (SA7779) - Enhanced DID Routing? n
                 (SA7852) - # and * in Vector Collect Step? none

                  (SA7880) - ASAI Internally Measured Data? n
                    (SA7900) - Service Observe Physical Set? y
 (SA7933) - Busy Tone with SAC and No Available Cvg Points? n

                                  (SA7963) - Dial By Name? n
                    (SA7991) - Variable Length Account Code? n
   (SA7994) - Incr Station Busy Ind to 25,000 (Linux only)? n
                        (SA8052) - ISDN Redirecting Number? n
                 (SA8077) - Russian Power Industry Feature? n
             (SA8122) - QSIG CPC Conversion for Code Set 5? n
                      (SA8140) - Attendant Dial 0 Redirect? n
```

Administer Class of Restriction

Enter the "change cor n" command, where "n" is the class of restriction (COR) number for each of the following station types:

- For agent stations that need to be recorded, modify their COR with the **Can Be Service Observed** set to **y**.
- For the virtual softphone stations used by Encore to record, modify their COR with the **Can Be A Service Observer** set to **y**.

The screen shown below is for reference. For each COR #, only one of the bolded fields should be set to **y**.

```
change cor 2                                                Page   1 of  43
                              CLASS OF RESTRICTION

               COR Number: 2
          COR Description:

                      FRL: 0                                APLT? y
  Can Be Service Observed? y          Calling Party Restriction: none
Can Be A Service Observer? y           Called Party Restriction: none
        Time of Day Chart: 1      Forced Entry of Account Codes? n
         Priority Queuing? n               Direct Agent Calling? n
     Restriction Override: none       Facility Access Trunk Test? n
     Restricted Call List? n                Can Change Coverage? n
```

Administer Agent Stations

Use the "change station n" command, where "n" is a non-SIP agent station to be recorded. For **COR**, enter the COR number selected from the previous step for stations that are to be recorded.

Repeat this section to administer all non-SIP agent stations that will be recorded.

```
change station 65001                                        Page   1 of   5
                                     STATION

Extension: 65001                       Lock Messages? n              BCC: 0
     Type: 9611                        Security Code: *               TN: 1
     Port: S00103                    Coverage Path 1: 1              COR: 2
     Name: CM7 Station 1             Coverage Path 2:                COS: 1
                                  Hunt-to Station:                 Tests? y
```

Administer virtual IP softphones

Add a virtual softphone using the “add station n” command, where “n” is an available extension number. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Type: 4610**
- **Name:** A description name
- **Security Code:** A desired value
- **COR:** The class of restriction number from “Administer Class of Restriction“ on page 18
- **IP Softphone: y**

```
add station 65991                                               Page   1 of   6
                                     STATION

Extension: 65991                         Lock Messages? n              BCC: 0
     Type: 4610                          Security Code: 65991           TN: 1
     Port: IP                              Coverage Path 1:            COR: 1
     Name: Encore Virtual #1             Coverage Path 2:              COS: 1
                                         Hunt-to Station:
STATION OPTIONS
                                             Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                   Message Lamp Ext: 65991
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? y

                                                 IP Video Softphone? n
                              Short/Prefixed Registration Allowed: default

                                                Customizable Labels? y
```

Navigate to **Page 4** and add a **serv-obsrv** button as shown below.

```
change station 65991                                            Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                       Cord Length: 0
   Building:                                          Set Color:

ABBREVIATED DIALING
     List1:                    List2:                     List3:

BUTTON ASSIGNMENTS
 1: call-appr                          7:
 2: call-appr                          8:
 3: call-appr                          9:
 4: serv-obsrv                        10:
 5:                                   11:
```

Repeat this section to administer the desired number of virtual softphones.

Configure Avaya Aura Application Enablement Services

The steps in this section configure the Application Enablement Services (AES). This includes the following steps:

Step 1: Launch Application Enablement Services Management Console

Step 2: Verify license

Step 3: Administer TSAPI Link

Step 4: Administer H.323 Gatekeeper

Step 5: Disable Security Database

Step 6: Restart TSAPI Service

Step 7: Obtain Tlink Name

Step 8: Administer Encore User

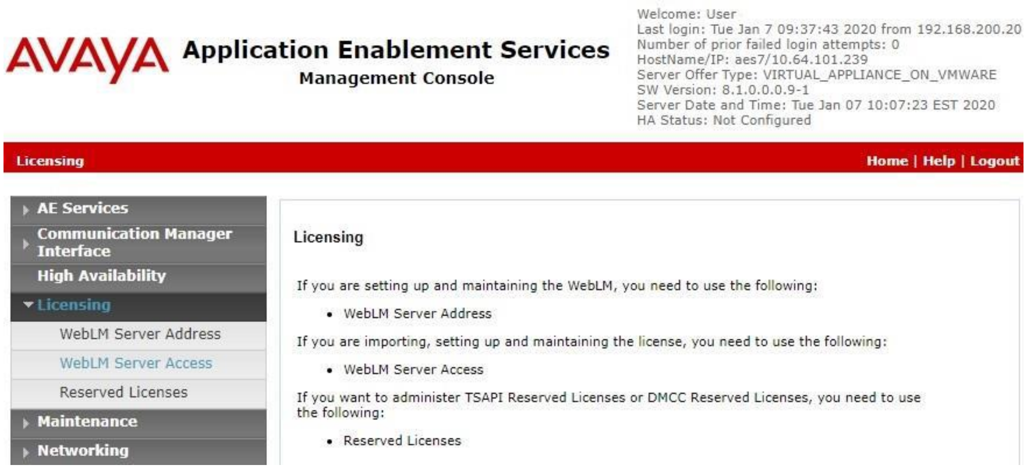
Step 9: Enable DMCC Unencrypted Port

Step 1: Launch Application Enablement Services Management Console

To log into the Application Enablement Services Management Console (Management Console), enter the fully qualified domain name or IP address of the AE Services server.

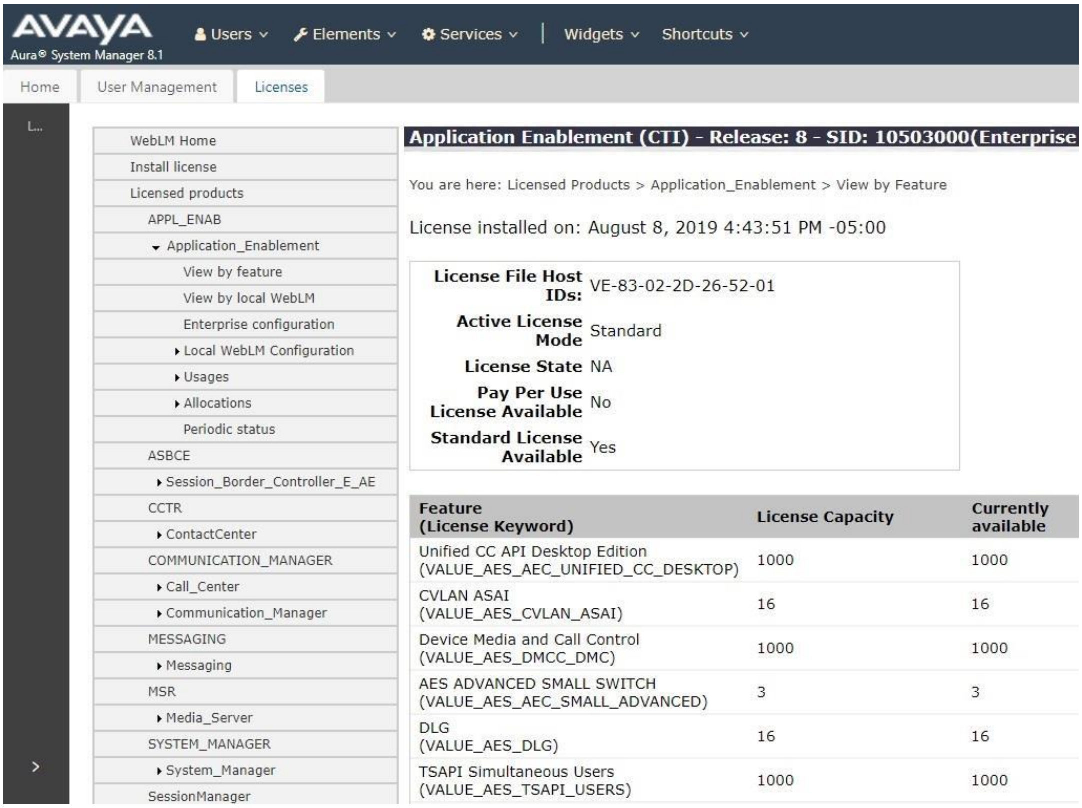
Step 2: Verify license

Select **Licensing | WebLM Server Access** in the left pane to display the **Web License Manager** window and log into it.



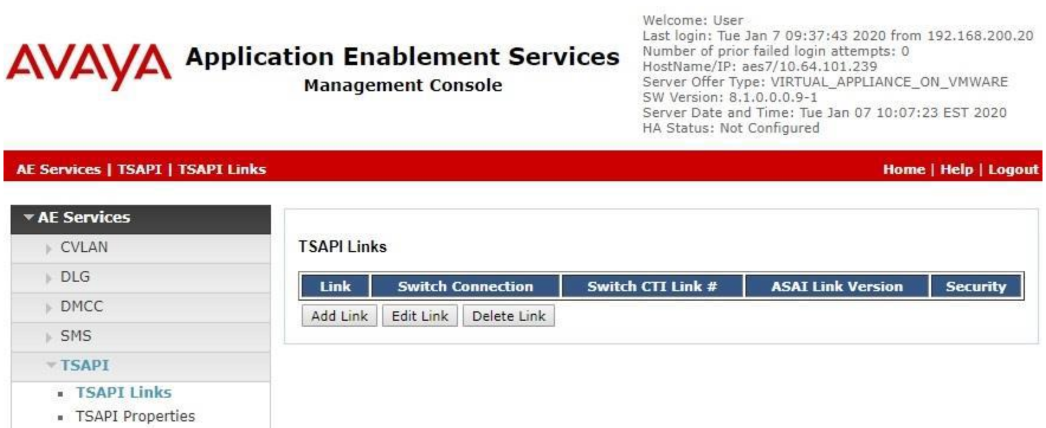
The **Web License Manager** window opens. Select **Licensed Products | APPL_ENAB | Application_Enablement** in the left pane to display the **Licensed Features** window in the right pane.

Verify that there are enough licenses for **TSAPI Simultaneous Users** and **Device Media and Call Control** as shown below. Note that the TSAPI license is used for device monitoring, and the DMCC license is used for the virtual IP softphones.



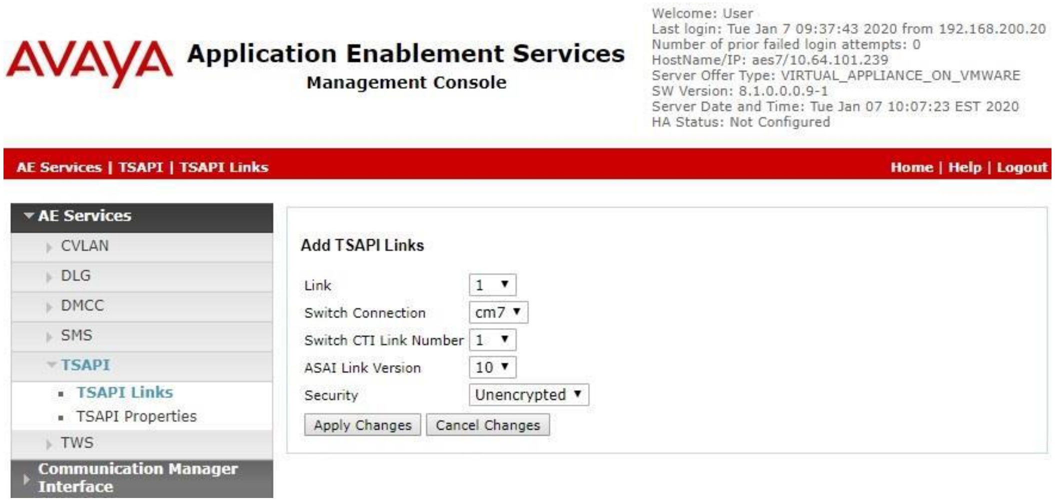
Step 3: Administer TSAPI Link

To administer a TSAPI link select **AE Services | TSAPI | TSAPI Links** from the left pane of the **Management Console**. The **TSAPI Links** window opens. Click **Add Link**.



The **Add TSAPI Links** window opens. The **Link** field is only local to the AES server and may be set to any available number. For **Switch Connection**, select the relevant switch connection from the drop-down list. In this example, the existing switch connection **cm7** is selected. For **Switch CTI Link**

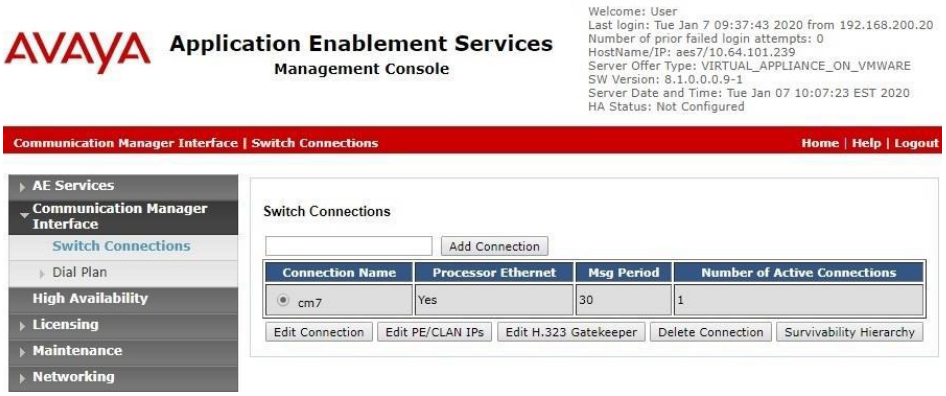
Number, select the CTI link number set up in “Complete for all systems” on page 14. Retain the default values in the remaining fields and click **Apply Changes**.



Step 4: Administer H.323 Gatekeeper

Select **Communication Manager Interface | Switch Connections** from the left pane. The **Switch Connections** window shows a listing of the existing switch connections.

Locate the **Connection Name** associated with the relevant Communication Manager, in this example **cm7**, and select the corresponding radio button. Click **Edit H.323 Gatekeeper**.

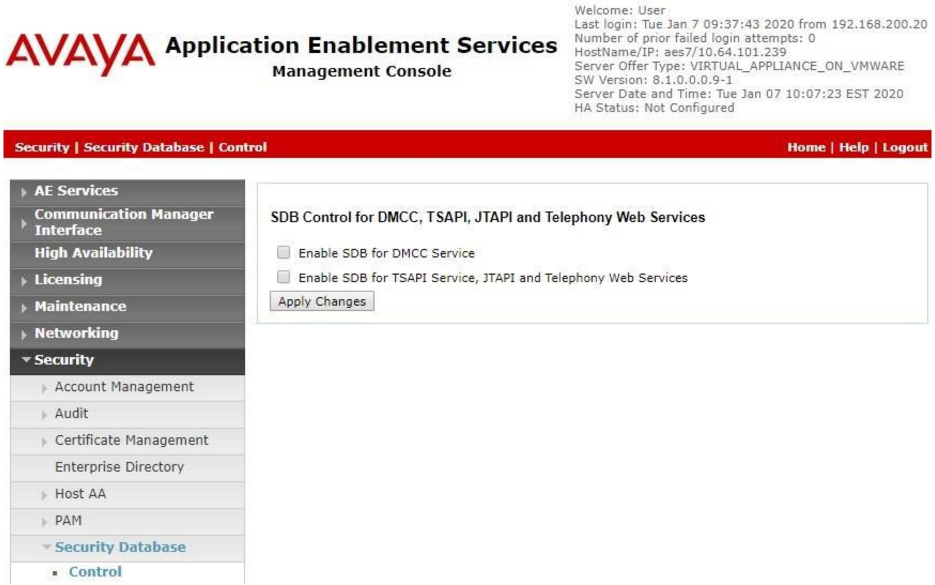


The **Edit H.323 Gatekeeper** window opens. Enter the IP address of a C-LAN circuit pack or the Processor C-LAN on Communication Manager to be used as H.323 gatekeeper, in this example **10.64.101.236** is shown below. Click **Add Name or IP**.

Step 5: Disable security database

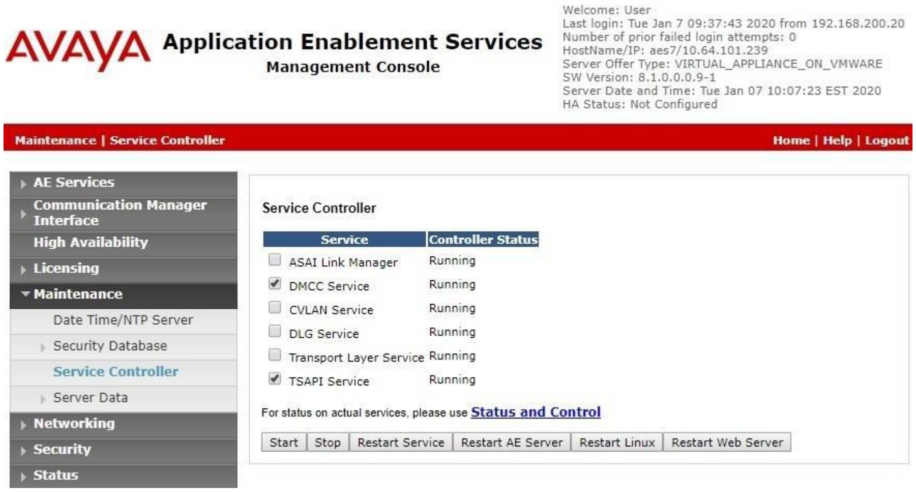
Select **Security | Security Database | Control** from the left pane, to display the **SDB Control for DMCC, TSAPI, JTAPI and Telephony Web Services** open in the right pane. Remove the check from both fields below and click **Apply Changes**.

> **NOTE** If the site is using the **Avaya Aura Contact Center (AACC)**, only the **Enable SDB for DMCC Services** should be unchecked.



Step 6: Restart the Services

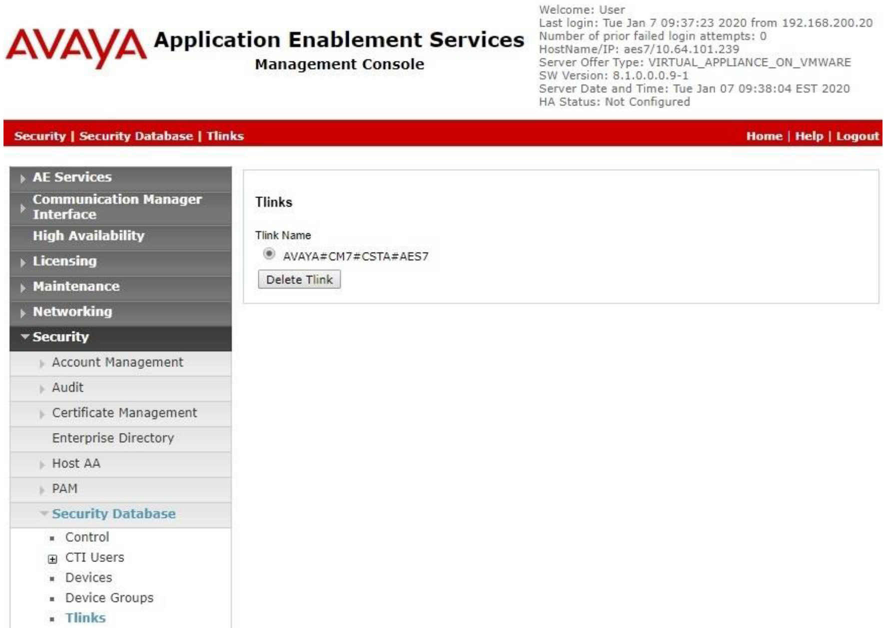
Select **Maintenance | Service Controller** from the left pane, to display the **Service Controller** window in the right pane. Check **DMCC Service** and **TSAPI Service** and click **Restart Service**.



Step 7: Obtain Tlink name

Select **Security | Security Database | Tlinks** from the left pane. The **Tlinks** window shows a listing of the Tlink names. A new Tlink name is automatically generated for the TSAPI service. Locate the Tlink name associated with the relevant switch connection, which would use the name of the switch connection as part of the Tlink name. Make a note of the associated Tlink name, to be used later for configuring Encore.

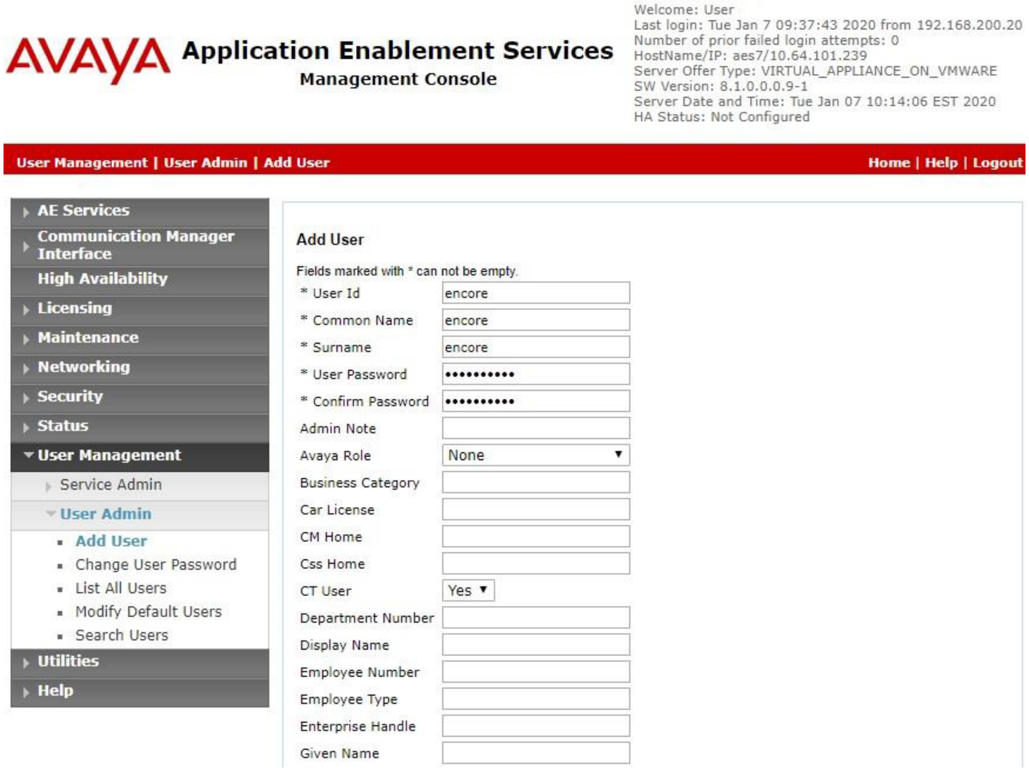
In this example, the associated Tlink name is **AVAYA#CM7#CSTA#AES7**. Note the use of the switch connection **CM7** from "Step 3: Administer TSAPI Link" on page 21.



Step 8: Administer Encore user

Select **User Management | User Admin | Add User** from the left pane, to display the **Add User** window in the right pane.

Enter desired values for **User Id**, **Common Name**, **Surname**, **User Password**, and **Confirm Password**. For **CT User**, select **Yes** from the drop-down list. Retain the default value in the remaining fields. Click **Apply** at the bottom of the window (not shown below).



NOTE If using the Station-side TDM, Trunk-side TDM, Station-side RTP Packet Capture, or Trunk-side SIP Packet Capture method, the AES server setup is complete. If using the Subscription-based DMCC Audio Stream method, complete the step on the next page.

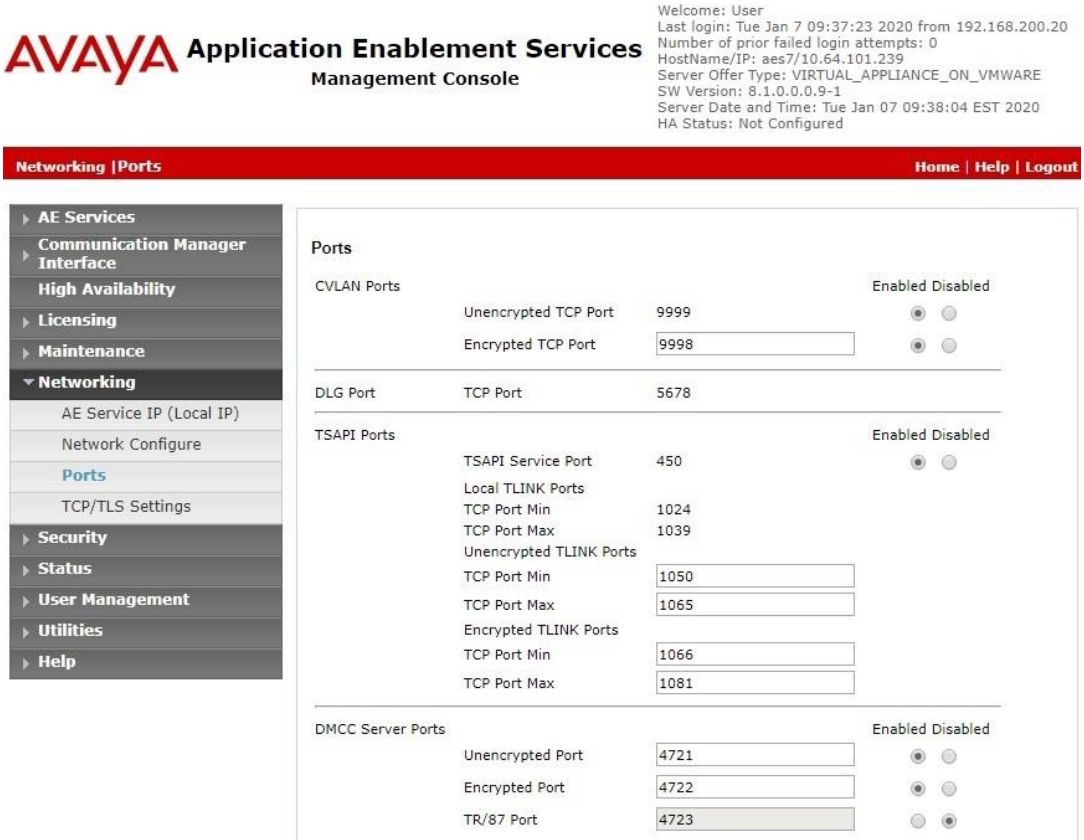
Step 9: DMCC server port

Only complete this step if using the Subscription-based DMCC Audio Stream audio collection method. If not using this method, skip this step.

Provide your dealer or DVSAnalytics installation tech with the following information:

- DMCC Server Port number
- DMCC Server Port type – Encrypted or Unencrypted

To find this information, select **Networking | Ports** from the left pane to display the **Ports** window in the right pane.



Configure Avaya Aura® Session Manager

Only complete this step if SIP agent stations are being recorded. If not using this method, skip this step.

Login to **System Manager** and select **Users → User Management** from the top menu. Select **User Management → Manage Users** from the left pane. Select the entry associated with a SIP agent station to be recorded and click **Edit**. The **User Profile | Edit** screen is displayed. Select the **Communication Profile** tab, followed by **CM Endpoint Profile**. Click the **Editor** icon next to the **Extension** field. In the screen that opens, set the **Type of 3PCC Enabled** to **Avaya**. If using the Subscription-based DMCC Audio Stream method, enter the Class of Restriction (COR) you previously setup in "Administer Class of Restriction" on page 18.

Repeat these steps for all SIP agent users.

Verify AACC Settings and Obtain Required Information

Only complete these steps if you are using AACC to capture additional ACD data.

Available licenses

In AACC, open the Contact Center Licensing and select the **Real Time Usage** page. Verify the SIP Call Recording license **LM_CONTACTRECN** is available. This listing corresponds to the license string **LM_CONTACTRECN** in the physical license file.

Also verify that the CCT Web Services are available; this is listed as **Open Interface Open Networking** on the **Real Time Usage** page. The listing corresponds to the license string **LM_OIN** in the physical license file.

Obtain information needed for the AACC integration

Provide your dealer or DVSAnalytics installation tech with the following information:

- CCT Server Name
- CCT Web Service Port, see below for instructions
- CCT User Name and Password
- AACC SIP Domain
- CCMA Server Name
- CCMA Web Service Port (Defaults to port **80**)
- CCMA User and Password (Use the default of **WebAdmin**, otherwise the provided account must have permission to list the properties of all agents configured on the system.)
- A list of the extensions used by recorded agents.
- List of extensions used by recorded agents; see the next page for instructions on how to obtain this list

CCT Web Service Port

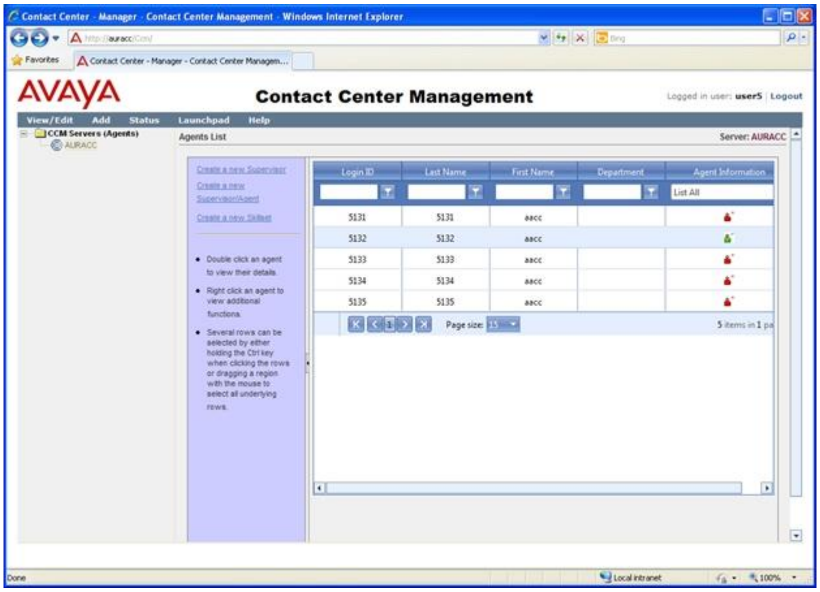
To obtain the CCT Web Service port, complete these steps:

1. Navigate to the AACC Web Services main page; it should be at **http://<*CCT Server Name*>:9090**.
2. Click the **Open Interface Splash Page** link.
3. Scroll to the bottom of the page that opens and click **UserService**.
4. The CCT web services port can be found in the **Wsdl Location**. In the sample below, the CCT web services port is 9084.

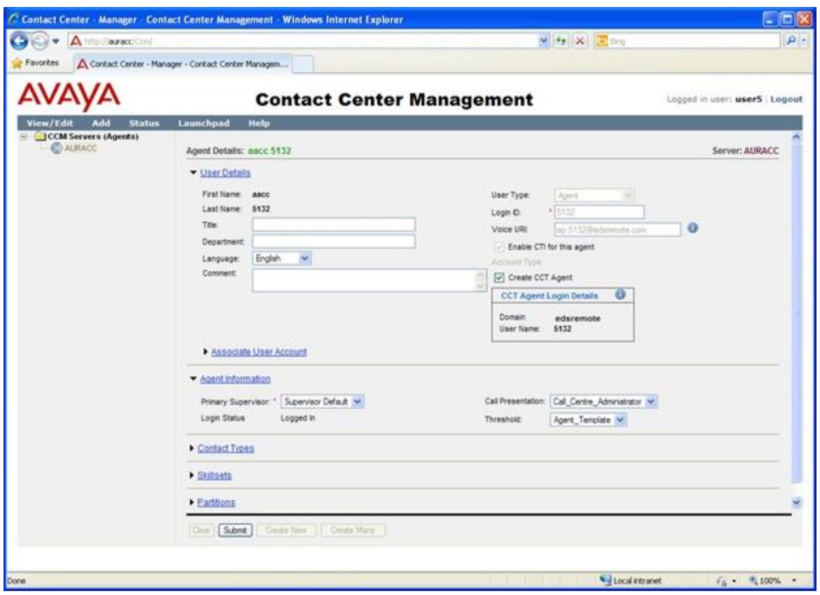
List of Extensions Used by Agents

To obtain the list of extensions used by agents, complete these steps:

1. Log into the Avaya Contact Center Manager, using the URL **http://<*CCMA Server Name*>**.
2. Select **Contact Center Management**.
3. Select **View/Edit** and select **Agents** from the drop-down menu.
4. In the tree view on the left select your Aura CC server. A list of agents opens.



5. Double-click a recorded agent.



6. The extension is shown in the **Voice URI** field. This parameter uses this format:

   sip:<*extension*>@<*SIPDomain*>

NOTE The domain should be the same as the AACC SIP Domain already obtained.

7. Return to the agent list by selecting your Aura CC server in the tree view on the left.
8. Repeat Steps 5−7 until all the extensions have been gathered for the agents that need to be recorded.

Call Handling Scenarios

This section explains how different calls are displayed in Encore. The samples in this section are from a station-side recording system and it is assumed that all stations involved in the calls are configured to be recorded.

Certain situations affect how recordings are created and how they can be located using the Related Call Lookup feature:

- Hold – When a call is put on hold, the recording is suspended. When the call is retrieved, the audio is appended to the recording to create one audio recording.
- Consultation Call – If an agent is on a call and then places a consultation call, the first call is put on hold and the recording is suspended. Assuming the called party is also using a recorded phone, the consultation call is recorded as two separate recordings – one for each extension. When the agent hangs up the consultation call and retrieves the caller, the two recordings end and the first recording resumes; the second portion of the recording is appended to the first portion. All three recordings have different Segment IDs (SID) and share the same Related ID (RID).
- Blind Transfer – When a call is blind transferred (also called an unannounced transfer), the first recording ends after the agent presses the transfer button and hangs up the handset. The second recording begins when the second agent answers the transferred call. The second recording ends when the second agent hangs up the call. A separate SID is associated with each recording and they usually share the same RID. If the call is transferred to an ACD queue or Hunt Group, it may not be possible to show the relationship between the recordings and the same RID may not be associated with both recordings.
- Conference Call – When an agent decides to bring a third party into a current call, the agent usually puts the caller on hold to first consult with the third party. The first recording of the agent and the outside caller suspends during the consultation call. Assuming the third party is using a recorded phone, the consultation call creates two recordings – one for the agent and another for the third party. After the consultation call ends and the three parties are joined into the conference, the first recording resumes and it ends when the agent hangs up. The recording of the third party continues until the third party hangs up.
- Internal Call – If both extensions are monitored by Encore, two recordings are created – one for each extension. The party who initiates the call is treated as the agent for data collection purposes.

External Inbound Call

Recordings: 1 | SID: 1 | RID: 1

Extension 5002 receives an external inbound call with SID 1 and hangs up when the call is complete. This call creates one recording and one RID even though no other calls are associated with it.

External Inbound Call with Supervised Transfer

Recordings: 3 | SID: 3 | RID: 1

1. Extension 5002 receives an external inbound call. Recording 1 begins with SID 1.
2. The agent presses the transfer button which puts the caller on hold and suspends Recording 1. The agent then makes a consultation call to extension 5025. Recording 2 for extension 5002 begins with SID 2 and Recording 3 begins for extension 5025 with SID 3. When extension 5002 hangs up to complete the transfer, Recordings 1 and 2 end.
3. Now the caller is transferred to the agent at extension 5025. Recording 3 continues.
4. When the agent at extension 5025 hangs up, Recording 3 ends.

The same RID is associated with all recordings to show they are related.

External Inbound Call to ACD

Recordings: 1 | SID: 1 | RID: 1

Extension 5002 answers an external ACD call. A recording with SID 1 begins and, when the call ends, the recording stops. The ACD number, ACD name, and Agent Login ID are associated with the recording.

External Outbound Call

Recordings: 1 | SID: 1 | RID: 1

Extension 5002 makes an external outbound call with SID 1 and hangs up when the call is complete. This call creates one recording and one RID even though no other calls are associated with it. The Call Direction for the recording shows as Outgoing. The dialed number is stored in the DNIS and Other Party Number fields.

Internal Call

Recordings: 2 | SID: 2 | RID: 1

Extension 5002 makes an internal call to extension 5009 (both extensions are monitored by Encore). A recording is created for each monitored extension and each recording is assigned a different SID. Both recordings are assigned the same RID to show they are related to each other.

External Inbound Call with Blind or Unannounced Transfer

Recordings: 2 | SID: 2 | RID: 1

1. Extension 5002 receives an external inbound call which starts Recording 1 with SID 1.
2. The agent transfers the caller to extension 5009 without consulting the agent at extension 5009. Recording 1 ends when 5002 hangs up his phone.
3. Recording 2 with SID 2 begins when 5009 answers the call. It ends when the agent hangs up her phone.

The same RID is associated with each recording to show they are related.

Consultation Call

Recordings: 3 | SID: 3 | RID: 1

1. Extension 5002 receives an external inbound call which starts Recording 1 with SID 1.
2. The agent puts the caller on hold, suspending Recording 1, and makes a consultation call to extension 5025 which starts Recording 2 with SID 2 to record extension 5002. This also starts Recording 3 with SID 3 to record extension 5025 in the consultation call.
3. When the agent at 5002 hangs up the consultation call, Recording 2 ends. When the agent at 5025 hangs up, Recording 3 ends.
4. The agent at extension 5002 then retrieves the original call and Recording 1 with SID 1 resumes.
5. When extension 5002 hangs up with the caller, Recording 1 ends.

The same RID is associated with all recordings to show they are related.

Conference Call

Recordings: 3 | SID: 3 | RID: 1

1. Extension 5010 receives an external inbound call which starts Recording 1 with SID 1.
2. The agent at extension 5010 puts the caller on hold and makes a consultation call to bring a supervisor at extension 5008 into the call. This suspends Recording 1. Recording 2 with SID 2 begins to record extension 5010 on the consultation call and starts Recording 3 with SID 3 to record the supervisor at extension 5008.
3. When the agent at extension 5010 joins the caller and the supervisor at extension 5008 into a three-party conference, Recording 2 ends. Recording 1 resumes and appends the audio to the first portion of the recording. Recording 3 continues.
4. When the supervisor at extension 5008 hangs up the call, Recording 3 ends.
5. When the agent at extension 5010 hangs up the call, Recording 1 ends.

The same RID is associated with all recordings to show they are related.

Appendix 1: Glossary

AACC data collection

The Avaya Aura Contact Center (AACC) does not provide recording control but does capture additional data. To see the data captured, turn to "Supported data capture" on page 3.

abandoned call

An incoming call which is answered by the ACD but terminated by the caller before it is answered by an agent.

ACD

Automatic Call Distributor. An application that answers calls and directs them to a predetermined queue, or line, of waiting calls. In most cases, the ACD ensures that the first call in is the first call answered. It also determines which agent receives a call based on predetermined criteria such as idle time or availability and generates reports on call volume and distribution.

ACD name

The ACD split name for the call.

ACD number

The ACD split number for the call.

ACD split

If the customer needs to control recording using an ACD split, every agent that needs to be recorded must be assigned to one or more ACD splits. Encore monitors the agents as they log into or out of the ACD. If the agent is a member of an ACD split that should be recorded, Encore records the conversation. No calls are required to be processed by the ACD split for recording to occur.

The ACD split feature is available on systems using the Station-side RTP Packet Capture and Subscription-based DMCC Audio Stream audio collection methods. ACD split may be used on systems that also need to record 10 or 11-digit extensions but the only audio collection method supported in this scenario is the Subscription-based DMCC audio collection method.

agent

A person who handles phone calls. Other variations include operator, attendant, representative, customer service representative (CSR), telemarketer, phone sales representative (TSR), and so on.

agent ID

The number assigned to an agent to identify the agent in the system. CT Gateway must monitor the ACD split or the EAS skill extension in which the agent is logged into to capture this information.

ANI

Automatic Number Identification. For inbound calls, this is the phone number from which the customer is calling (may not be supported by the trunk). For outbound calls, this is the dialed number. For a recording of a "Barge-In" call, the ANI will be incorrect.

automated attendant

A voice processing system that answers calls with a recording and then enables callers to press touch-tone buttons to navigate through a menu system to a person, department, or voice mail.

call direction

The direction is either incoming (inbound) or outgoing (outbound).

call record

An entry in a database that holds the data associated with a call.

call type

The call type is either internal, external, or conference.

CCMA

Contact Center Manager Administration

CCT

Communication Control Toolkit

consultation call

A call that is made while the customer (original call) is on hold. In the database, the **Consultation Call** field shows **Yes** when the recording is a consultation call.

Control Directory Number (CDN)

This is a special directory number not associated with any physical phone or equipment. The CDN specifies a destination ACD queue to which incoming calls are directed. Multiple CDNs can place calls into the same ACD queue. The parameters of the CDN, not those of the ACD queue, determine call treatment.

dialed number

Captured for outbound calls and is stored in the ANI field.

digital recording

A method of recording that converts analog sound into a series of pulses that are translated into binary code, which is read by computers.

DNIS

Dialed Number Identification Service. For inbound calls, this is the number the customer dialed or the agent’s extension number (may not be supported by the trunk). For a recording of a “Barge-In” call, the DNIS will be incorrect.

encrypted calls

Calls that have the audio RTP packets encrypted. This prevents 3rd party applications, such as the Encore system, from using the RTP packets for recording.

extension

The number associated with a person's station. Extension and station are sometimes used interchangeably.

external calls

In these calls, the calling or called parties are outside the PBX.

full-time recording

This method uses the Recording Engine to record all conversations for the defined endpoints.

hold duration capture

The sum of all hold durations that occurred during the recording.

inbound

Calls which are received/answered by a recorded party.

internal calls

Calls made between extensions on the same PBX.

obtain agent name from SIS

This feature is helpful in a free-seating environment where the extension name defined in the PBX is different than the agent name. Using the extension, Encore queries SIS to locate the agent's name and then attaches the agent name to the recording. If the customer is using this feature, every recorded agent must be setup with a Recording Account in Encore 3.

outbound

Calls which are initialed/placed by a recorded party.

other call ID

This is the Universal Call ID of related calls.

other party name

If the other party is logged into the internal ACD, this field contains the agent name defined in the Encore database. If the other party is logged into the AACC, this field contains the name associated with the monitored extension as defined in the PBX. For internal calls, this data is only available if the other party is also a recorded extension. This data is not available for external calls.

other party number

Number of the other party on the line with the person being recorded; if external and incoming call, this is an ANI.

pause/resume on hold

A method that pauses the recording of audio and screen when a call is placed on hold, and resumes recording when the hold is taken off.

PBX (PABX)

Private (Automated) Branch Exchange. The phone system to which the office phones are connected.

recorded party name

If the other party is logged into the internal ACD, this field contains the agent name defined in the Encore database. If the other party is logged into the AACC, this field contains the name associated with the monitored extension as defined in the PBX.

This data is only received when Encore begins monitoring the extension during startup or when a device is added to the ctisetup.ini file and the **Update Device List** option is used to update CT Gateway.

recorded party number

The extension of the monitored phone.

recorded party disconnect

If the recorded party disconnects the call, this field is set to 1. If the other party disconnects, this field is set to 0.

recording

The audio recording, screen recording, and database record associated with a single phone call or conversation.

scheduled recording

This method uses the ESO Engine 2 to only record the defined endpoints according to the recording schedule. For instance, Encore may only record 50% of the calls on the defined endpoints instead of 100% as is automatically done for full-time recording.

skill-based routing

A skillset is a label applied to a collection of abilities or knowledge required to process a request, such as language preference, product knowledge, or department knowledge. In skill-based routing, agents are assigned skillsets, and contacts are presented to available agents who have the skillset to serve the customer request.

skillset

A skillset is a group of abilities necessary to answer a specific type of contact. Skillsets are the basic building blocks of skill-based routing.

station

A phone connected to the PBX.

TSAPI

Telephony Server Application Programming Interface. This is an API that enables programmers to build telephony and CTI applications. TSAPI is similar to TAPI, but TSAPI runs on Netware platforms whereas TAPI runs on Windows operating systems. Another key difference is that TAPI can be used for both client- and server-based applications whereas TSAPI is strictly a server API.

trunk

The connection between the phone company and the PBX that carries incoming calls.

Universal Call ID

A unique identifier used by the database to locate each recording. The call identifier for a recording can be viewed in Encore.

User to User Information

The User to User Information (UUI) is obtained from the Delivered or Established event in the TSAPI message. Encore stores the data in a field in the call record as ASCII text.