



Advanced Contact Center Solutions Lab

Avaya Aura® Contact Center CS1000 (AML) to Aura (SIP) Migration Guidelines

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Abstract

This application note documents a set of guidelines to help customers migrate from a CS1000 (AML) based Avaya Aura® Contact Center to a SIP (Avaya Aura®) Contact Center. The details covered in this application note are intended for use by anyone involved in the upfront planning of these migrations.

The following AML sources are supported:

- Avaya NES Contact Center Release 6.0 or Release 7.0 (AML)
- Avaya Aura Contact Center Release 6.2, 6.3 or 6.4 (AML)
- Avaya Aura Contact Center Release 7.0, 7.0.x (AML)

Avaya Aura Contact Center Release 7.0.x

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1. Introduction

Call Center applications have traditionally been embedded in the software DNA of a host Telephony PBX/Switch platforms. The PBX/switch provides the platform required to run a call center solution – two key features provided by the PBX are (i) PSTN trunks and gateways for callers to be able to reach the contact center and (ii) agent stations and associated station functionality required by agents to perform their work.

The embedded call center application provides the required routing, work assignment and agent management required by our customers. In reality there is a symbiotic relationship between the PBX platform and the call center application – the two have traditionally worked hand in hand.

Avaya Aura® Contact Center is an evolution of these PBX embedded call center applications but with fundamental differences – the key one being that it has evolved “outside” of the PBX (Avaya Aura® Communication Manager) core and has adopted Session Initiation Protocol (SIP) at its core. This has enabled it to move customers into SIP (Session Initiation Protocol) based architectures in line with the emerging migration of both UC and CC enterprise applications to the SIP paradigm.

Avaya Aura® Contact Center has introduced a fundamental change to the traditional call center architecture with its software based media component called Avaya Media Server. The Avaya Media Server component is used to “anchor” all contact center calls and, in effect, take them off the traditional PBX gateways and associated DSP’s. This means that the PBX, in effect, becomes a gateway for customer calls, in that control of the customer call is now with the Avaya Aura® Contact Center application. The PBX still hosts the agent and supervisor telephones or stations which are also controlled and monitored by the Avaya Aura® Contact Center application.

These fundamental differences between the traditional Contact Center deployment (AML based on CS1000) and its’ evolution to a SIP (Avaya Aura) based solution present challenges when attempting to migrate systems from AML to SIP. Detailed planning and a clear understanding of the manual steps involved are required.

This application note describes the key migration steps required when moving from an AML (CS1000) based AACC to a SIP (Aura) AACC.

The AACC 7.0 software includes a number software tools to allow CS1000 Contact Center data (AML) to be migrated to a newly deployed SIP based solution. The data that can be included in the migration along with these enhancements are detailed in this application note.

IMPORTANT: *Migrating to AACC 7.0 SIP (Avaya Aura) is assumed*

It should be noted that this document is intended as supplementary information to the existing Avaya Aura® Contact Center published user guides. It highlights any additional

configuration, features and limitations but should not be viewed as a replacement to the existing product documentation.

2. Solution Overview

Avaya Aura® Contact Center can be deployed with a CS1000 switch type (AML based contact center) OR with an Aura UC environment – SIP based contact center. Although many of the key principles and features are shared between the two installation types, there are some variations in terms of feature delivery and configuration to consider when you are planning to migrate from a CS1000 (AML based) to Aura (SIP based) solution.

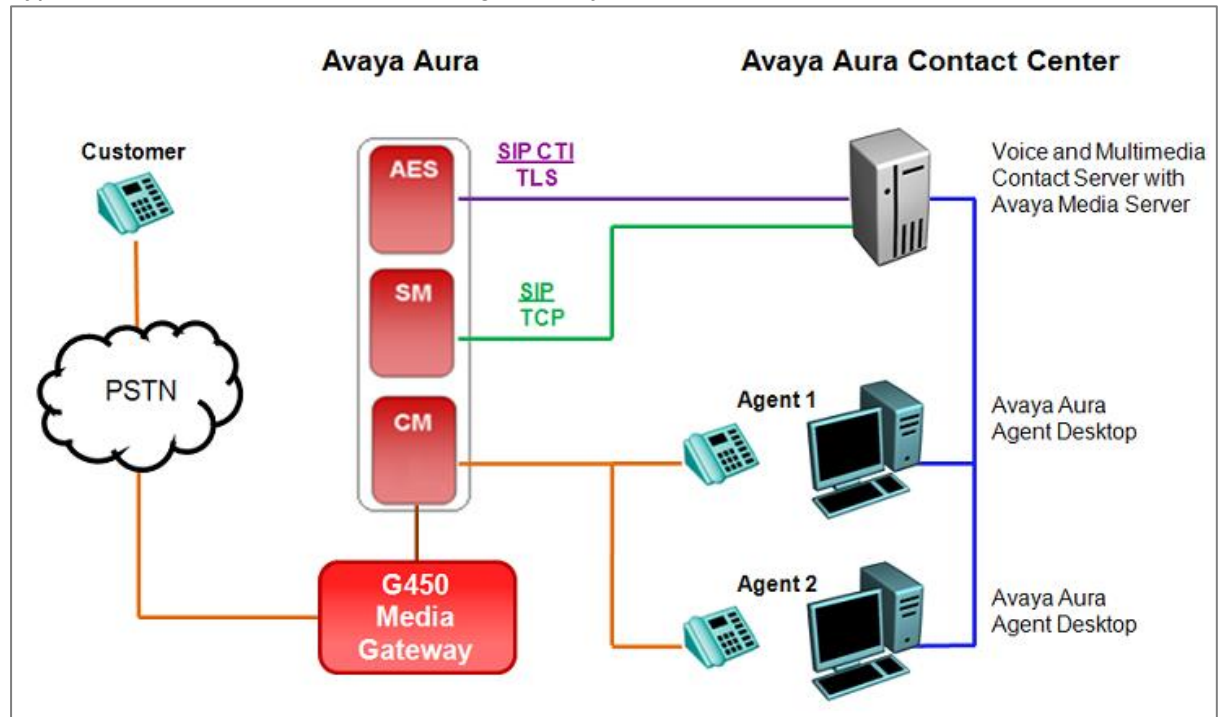
AACC 7.0 provides a range of tools and procedures to migrate AACC data (historical and configuration) from an AML based solution to a SIP based environment. Before this can take place, the target SIP solution must be available in advance of any migration activity. For this reason, the recommended approach is to deploy a new AACC 7.0 SIP solution, on new hardware to communicate with your newly deployed Aura UC infrastructure. This must be planned for, deployed and provisioned as per the existing documented procedures including any additional network and environment requirements.

It is expected that this activity is done on a phased basis with both systems (AML and newly deployed SIP/Aura) available in parallel. This is to allow for appropriate acceptance testing without impacting any live contact center activity.

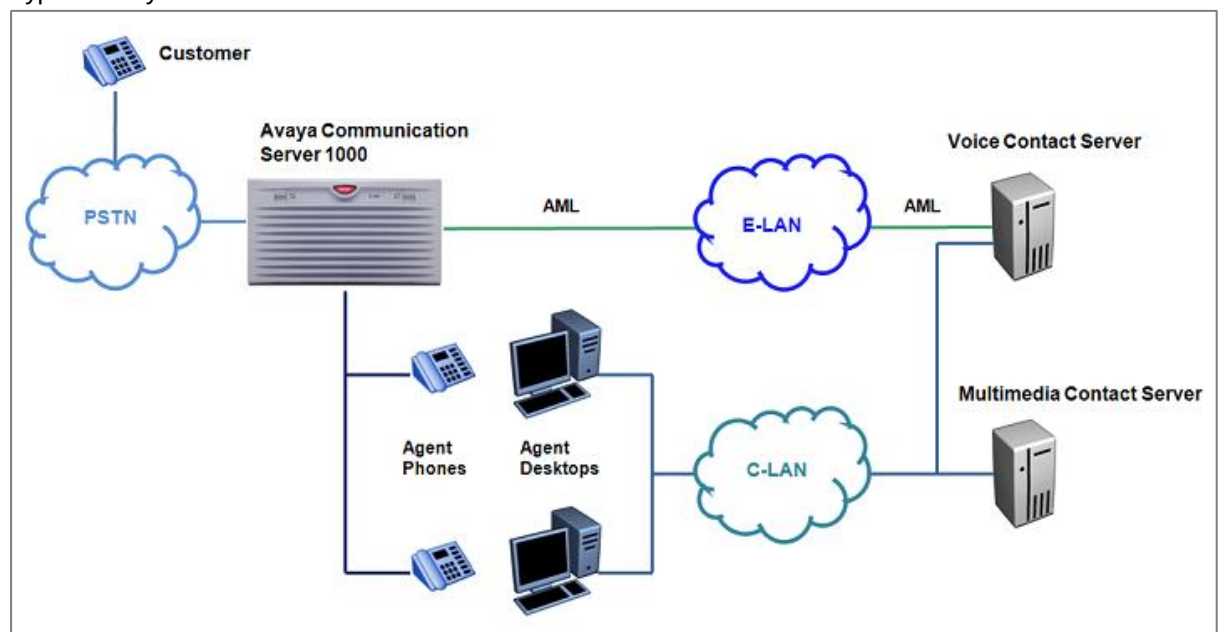
This document is not a task-based or procedural guide. It is intended to be used as a reference guide for groups involved in the pre-sales and planning stages of a CS1000 to SIP migration. Any areas that require special consideration during this planning phase are highlighted and readers are redirected to existing, published AACC documentation. This application note must be used in conjunction with existing AACC guides, beginning with the *Avaya Aura Contact Center Overview and Specification* guide.

2.1 AACC deployments on Avaya Aura UC platform OR Communication Server 1000

Typical SIP-enabled contact center using the Avaya Aura® Unified Communications:



Typical Avaya Communication Server 1000 AML-based contact center:



3. Document Structure and Goals

This document is intended as an up-front planning guide when migrating an Avaya Aura Contact Center deployed CS1000 (AML deployment) to a SIP based solution on Aura CM.

The document begins with a chapter covering the tools provided with AACC 7.0 and guides the reader through the main steps required to migrate their AML data to a SIP based deployment

Next, a reference customer deployment is presented. This allows the reader to review at a high level the main changes they can expect to make when deploying an equivalent contact center in a SIP environment. The featured deployment has been chosen to highlight the most significant differences in terms of configuration and solution deployment.

As there are a large range of options and configurations available with AACC (both CS1000 & Aura SIP), more detailed information is contained in the '*Migration Considerations – Detailed Analysis*' chapter:

This contains a series of sections, structured in table format and provides a detailed view of feature and functional differences to consider when migrating from AML to SIP. This is provided across a broad range of AACC capabilities and solution integration points. In many cases, similar functionality is provided in SIP as AML but delivered differently, e.g. Agent Greeting may be configured in an AML contact center but is also available in SIP enabled contact centers using the Avaya Media Server, these changes are called out in this chapter.

Where appropriate, each section also contains answers to commonly asked questions and guidance to readers on how best to deploy their newly installed SIP solution.

IMPORTANT:

Both SIP and AML based contact centers are possible within CS1000 deployments. This document describes migration considerations and procedures when moving from an AML based system only. Wherever the term CS1000 is used, an AML implementation should be assumed.

The assumed target deployment is a SIP based AACC 7.0 deployed with Avaya Aura.

4. AACC 7.0 Contact Center Data Migration Tools & Procedures

4.1 Overview

This chapter provides an overview of the steps involved when migrating from AML (CS1000) to a SIP based (Avaya Aura) AACC; this is intended as a planning aid in advance of the migration. For detailed, procedural steps please refer to ***Upgrading and Patching Avaya Aura Contact Center: Part 7: AML to SIP Migrations***.

The following data is included in the migration:

Core Contact Center Data:

- Historical Contact Center data
- Agent & Supervisor profiles and accounts (including Supervisor assignments)
- Skillsets and Skillset assignments
- Route-Points (CDNs) and DNIS etc.
- Call Presentation Classes, Activity codes etc.
- Orchestration Designer Flows, scripts and variables

Administration Data:

- Administration and Supervisor accounts (CCMA)
- Report Groups, Partitions & Access Classes
- Private / Custom reporting
- Filters and displays

Multimedia and Agent Desktop Data (if applicable):

- Multimedia Historical Data
- Multimedia Configuration data – includes Rules, Groups, auto-responses etc. etc.
- Agent Desktop configuration and desktop settings
- Agent Desktop Phone-book data
- Windows login credentials for Agent Desktop agents

Before a migration can take place, it is assumed that a newly installed AACC 7.0 deployment is in place and commissioned as per published documentation. This allows full validation of the new deployment prior to the migration.

To ensure that any SIP specific configuration is not overwritten on the target server, the following information is safe-guarded by the migration tools and will not be overwritten when the migration is run.

Server Configuration:

- Local Settings
- Licensing
- SIP Network Settings
- SIP Local Subscriber

Contact Center Configuration:

- Global Settings
- Media Server
- Media Service Routes
- Historical Statistics (system limits)

NOTE: Activation of Contact Summary reporting must be set on the newly deployed SIP AACC server. This will not be migrated from the source server.

4.2 Up-front Planning and Pre-requisites

This section looks at some specific up-front planning when performing a migration – This information should be reviewed in conjunction with the Avaya Aura Contact Center Overview and Specification guide.

Supported Migration Paths

The following table summarizes supported versions and topologies

Source (CS 1000 AML)	Target (Avaya Aura SIP)	Support
Source CS 1000 deployments:		
Avaya Aura Contact Center 7.0 (AML)	Avaya Aura Contact Center 7.0	Yes
Avaya Aura Contact Center 6.4 – Service Pack 15 (AML)	Avaya Aura Contact Center 7.0	Yes
AACC 6.2 or AACC 6.3 (AML)	Avaya Aura Contact Center 7.0	Yes
Avaya NES Contact Center 6.0 or 7.0 (AML)	Avaya Aura Contact Center 7.0	Yes
<i>*AACC 7.0 supports migrating from SIP CS1000 to Avaya Aura. This is outside the scope of this document and covered by a separate application note.</i>		
Network Topologies:		
Workgroup	Domain	Yes
Domain	Domain	Yes
Workgroup	Workgroup	Not Supported
Domain	Workgroup	Not Supported
License Types:		
Web LM licensing	Web LM licensing	Yes
PLIC Licensing	Web LM licensing	<i>All new SIP deployments use Web LM</i>
Web LM licensing	PLIC Licensing	Not Supported
PLIC Licensing	PLIC Licensing	Not Supported
<i>Note: Migrations to the ACCS 7.0 release are also possible but outside the scope of this document. This is covered by a separate application note.</i>		

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Supported target deployments & document references

As described earlier, the AML to SIP migration process makes use of a number of procedures and tools already provided with earlier versions of the AACC product. These procedures were previously intended for use when migrating contact center data when the UC type remained the same, e.g. CS1000 AML to CS1000 AML. Since AACC 6.4.2, these procedures have been enhanced to support AML to SIP migrations also.

Supported target deployments and applicable document reference are listed below:

Target Deployment	Document Reference
Voice and Multimedia Contact Server without Avaya Media Server	<i>Upgrade and patching Avaya Aura Contact Center: Part 7: AML to SIP migrations</i>
Voice and Multimedia Contact Server with Avaya Media Server	
Separate Voice and Multimedia Contact Servers	

System Backup after Ignition (IMPORTANT)

A full AACC backup must be taken after the ignition process has completed and before the system is commissioned or used.

The CCMA data can only be migrated to a system that does not contain any customer data. The CCMA migration will fail if the system is found to contain data other than what was injected by the Ignition Wizard. If the CCMA migration fails in this way, the solution is to go back to the post-ignition backup or re-install the system.

Staging a migration

During the AACC data migration, a data consistency check is automatically performed to ensure the integrity of the migrated data. If any issues are found during this process, they will be logged to the event viewer – (Refer to [Appendix A: Migration Event Codes](#) for details on events). In the case of agent validation errors being detected, it is necessary to run the 'Agent Configuration Validation Tool' to resolve these inconsistencies.

IMPORTANT: *The Agent Configuration Validation Tool is available for use by Avaya Support and Design personnel only. For this reason a staged migration should be run in advance of any planned migration to allow sufficient time to plan for Avaya Support engagement.*

Planning for a dry-run is a useful way to verify that everything is in place ahead of the actual migration and also allows for more accurate restore/migration timings to be calculated. The migration procedure can be re-run on the same system.

Migrating from a source AML system with more resources than can be supported on the target SIP based AACC deployment.

Careful consideration must be given to the supported capacity and resource limits of the target SIP based solution vs. the source AML deployment. Please refer to [Review AACC Capacity & Environment considerations](#) for more details.

If one of the goals of the migration is to reduce the size of the contact center – for example, reduced agent count or application/flow count, the following must be done as a prerequisite:

- Take a full AACC backup of the **source** deployment.
- Reduce the capacity of the **source** deployment prior to migrating to the new SIP deployment. For example, delete any excess resources (agents, applications) that are un-supported on the target system.
- Perform the migration as per the migration procedures and migrate the reduced set of resources and data to the **target** deployment.
- *Post-Migration:* If historical reporting of the previously deleted resources is required, it is possible to revert to the backup taken at the start of this procedure on the source AML server. The source AML system can then be retained for reporting purposes.

Additional Prerequisite Tools:

Depending on your source AML deployment, additional tools or resources may be required as part of the migration steps and procedures that are not available on the new AACC server. These are summarized here. Review these tools and ensure they are available prior to beginning the migration.

AML Source	Required Tools
Avaya NES 6.0 (AML)	1. Contact Center Migration Tool: Used to convert NES 6.0 databases to a format compatible with AACC migration Tools. This is available from the Avaya Web Site (http://www.avaya.com/support) 2. Microsoft Excel is required to use the SIP Spreadsheet (CCMA Consolidated Configuration Tool). This can be deployed on desktop with network access to the AACC server. Excel 2010 is described in the AACC documentation 3. NES 6.0 Documentation: Ensure you have the appropriate documentation for the release you are migrating from. This will be required when taking CCMA backups of the source server(s). 4. RCW Report Migration Tool: You can migrate custom historical reports built using Report Creation Wizard (RCW) in earlier releases to equivalent SSRS reports on Contact Center 7.0.

	Refer to the <i>AACC 7.0 Release Notes</i> for information on where to find this tool. Choose the version for NES 6
Avaya NES 7.0 (AML)	1. Microsoft Excel is required to use the SIP Spreadsheet (CCMA Consolidated Configuration Tool). This can be deployed on desktop with network access to the new AACC server. Excel 2010 is described in the AACC documentation 2. NES 7.0 Documentation: Ensure you have the appropriate documentation for the release you are migrating from. This will be required when taking backups of the source server(s). 3. RCW Report Migration Tool: You can migrate custom historical reports built using Report Creation Wizard (RCW) in earlier releases to equivalent SSRS reports on Contact Center 7.0. Refer to the <i>AACC 7.0 Release Notes</i> for information on where to find this tool. Choose the version for NES 7
AACC 6.2, AACC 6.3 or AACC 6.4 (AML)	1. Microsoft Excel is required to use the SIP Spreadsheet (CCMA Consolidated Configuration Tool). This can be deployed on desktop with network access to the new AACC server. Excel 2010 is described in the AACC documentation 2. AACC Documentation: Ensure you have the appropriate documentation for the release you are migrating from. This will be required when taking backups of the source server(s). 3. RCW Report Migration Tool (Applies to AACC 6.4 only): You can migrate custom historical reports built using Report Creation Wizard (RCW) in earlier releases to equivalent SSRS reports on Contact Center 7.0.

	Refer to the <i>AACC 7.0 Release Notes</i> for information on where to find this tool. Choose the version for AACC 6.4. <i>Minimum of AACC 6.4 (SP14) required.</i>
AACC 7.0 (AML)	1. Microsoft Excel is required to use the Spreadsheet (CCMA Consolidated Configuration Tool). This can be deployed on desktop with network access to the new AACC server. Excel 2010 is described in the AACC documentation 2. AACC Documentation: Ensure you have the appropriate documentation for the release you are migrating from. This will be required when taking backups of the source server(s).

Windows User Accounts:

Unless otherwise specified, login to the AACC server using the local Administrator account to run migration applications and procedures.

Required AML Contact Center Accounts:

Ensure that you have access to the sysadmin account for your source AML server. This will be required by one of the post migration steps.

AACC 7.0 Enhanced Security:

Refer to: *Upgrading and Patching Avaya aura Contact Center: Part 7: AML to SIP Migrations*, section 'Turning off Web Services security' to determine if it applies to your deployment.

4.3 Summary of Migration steps

How to migrate AML Contact Center data to AACC 7.0

STEP 1

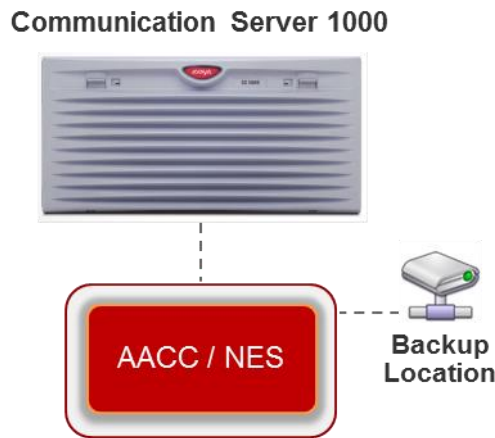
INSTALL AACC 7.0 for Avaya Aura:



- Install a SIP based (Avaya Aura) AACC solution in a Windows Domain environment as per existing AACC 7.0 published procedures
- The target server must be a non-HA system – High Availability options can be enabled once the migration is complete

IMPORTANT: A full AACC backup must be taken after the ignition process has completed and before the system is commissioned or used. It will be necessary to revert to the CCMA portion of this backup prior to migrating CCMA data.

- The new deployment must be fully provisioned and tested prior to attempting a migration of AML contact center data
- Run a complete AACC system backup, prior to running any migration tools if you wish to retain your test configuration.
- Web Services Security – Refer to the section 'Turning off Web Services security' to determine if it applies to your deployment. This is included in the AML to SIP Migration chapter of the document referenced at the beginning of this chapter.

STEP 2**BACKUP the existing source (AML) Contact Center Data:****On the source AML Server:**

- Perform a full backup of the databases from your AML-based Contact Center server(s), following the documentation for the release from which you are migrating.
- Ensure that the following components are included in the Backup:

Component	Additional Migration Notes:
CCMS	
CCT	
CCMM	Multimedia Data: Only applicable if you are using Multimedia on your AML system
CCMA	Administration Data: Note: If you are migrating from AACC 7.0 (AML), a CCMA backup can be taken using the Database Maintenance Application. For earlier versions, the CCMA Backup & Restore application must be used.

Migrating from NES 6.0:

If the source system is an NES 6.0 release, you must run the Contact Center Migration Tool. This is required to backup and convert the NES Contact Center 6.0 databases to a format compatible with AACC 7.0. The Contact Center Migration Tool is available from the Avaya Web Site (<http://www.avaya.com/support>)

Also note: There is no supported method of migrating archived Multimedia Data from an NES 6.0 deployment.

Refer to: *Upgrading and patching Avaya Aura Contact Center: Part 7: AML to SIP migrations for detailed procedures.*

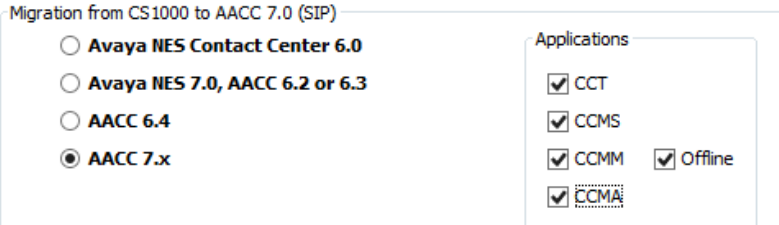
- Preparing for custom Report Creation Wizard reports migration: You can migrate custom historical reports built using Report Creation Wizard (RCW) in earlier releases to equivalent SSRS reports on Contact Center7.0. Export the original reports from Crystal format (.rpt) to intermediate files and save them in a network location. After installing the new server you can restore and convert the saved files to SSRS format. Refer to the AACC 7.0 Release Notes for information on where to find the RCW export utility. There are three versions of the utility, one for NES 6.0, NES 7.0 and AACC 6.4 and should be run on the source server. For more detailed information, refer to the Upgrade and Patching Avaya Aura Contact Center guide.

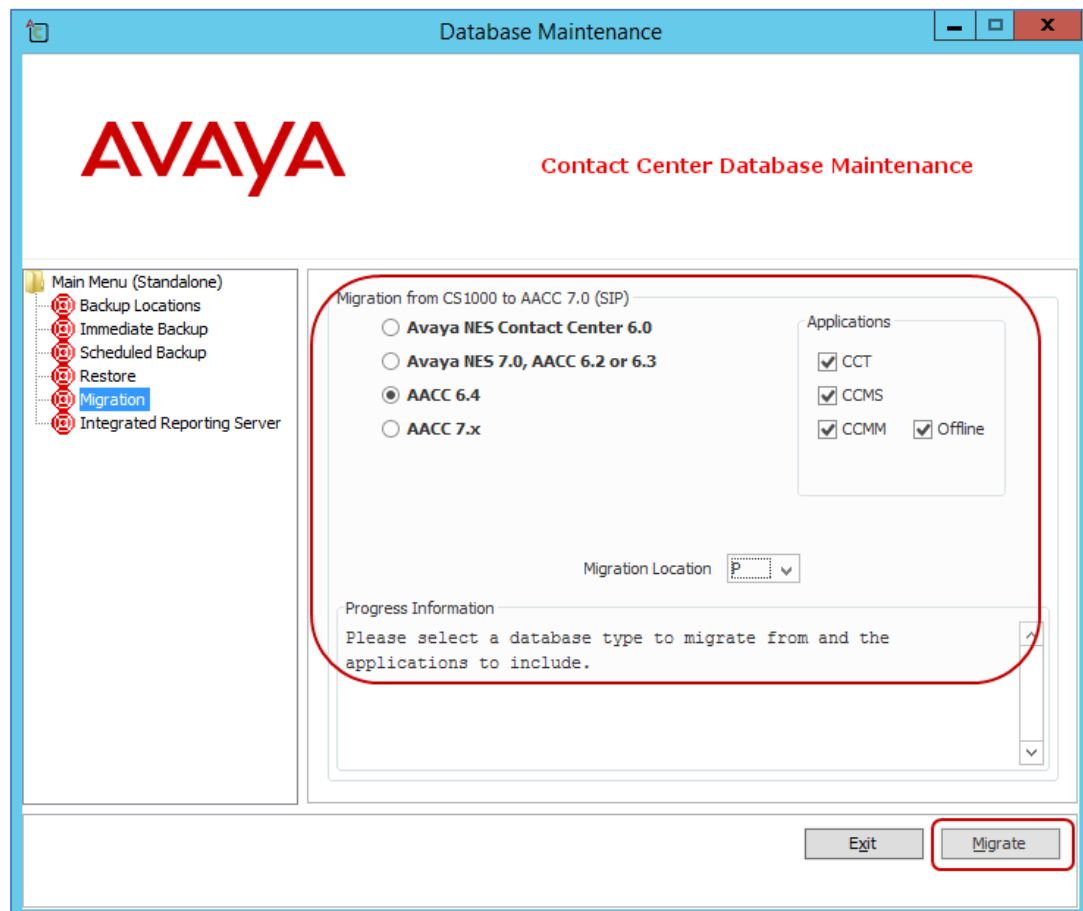
NOTE: The 6.4 version of the Report Creation Wizard export tool is supported on AACC 6.4 SP14 or later.

STEP 3**Migrate (AML) Contact Center Data:**

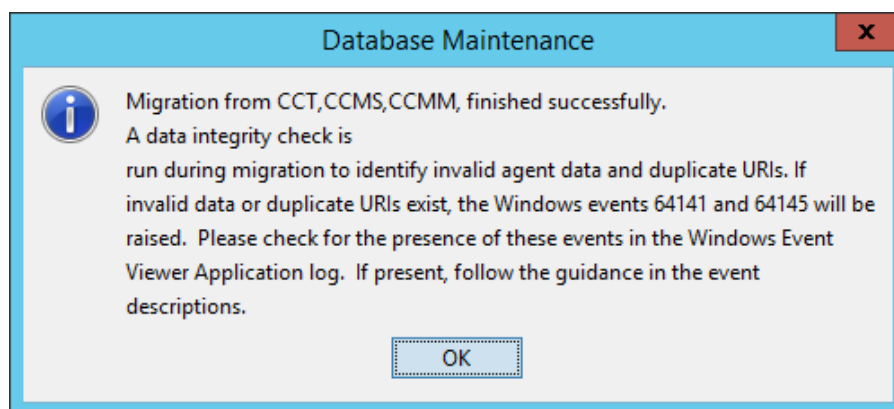
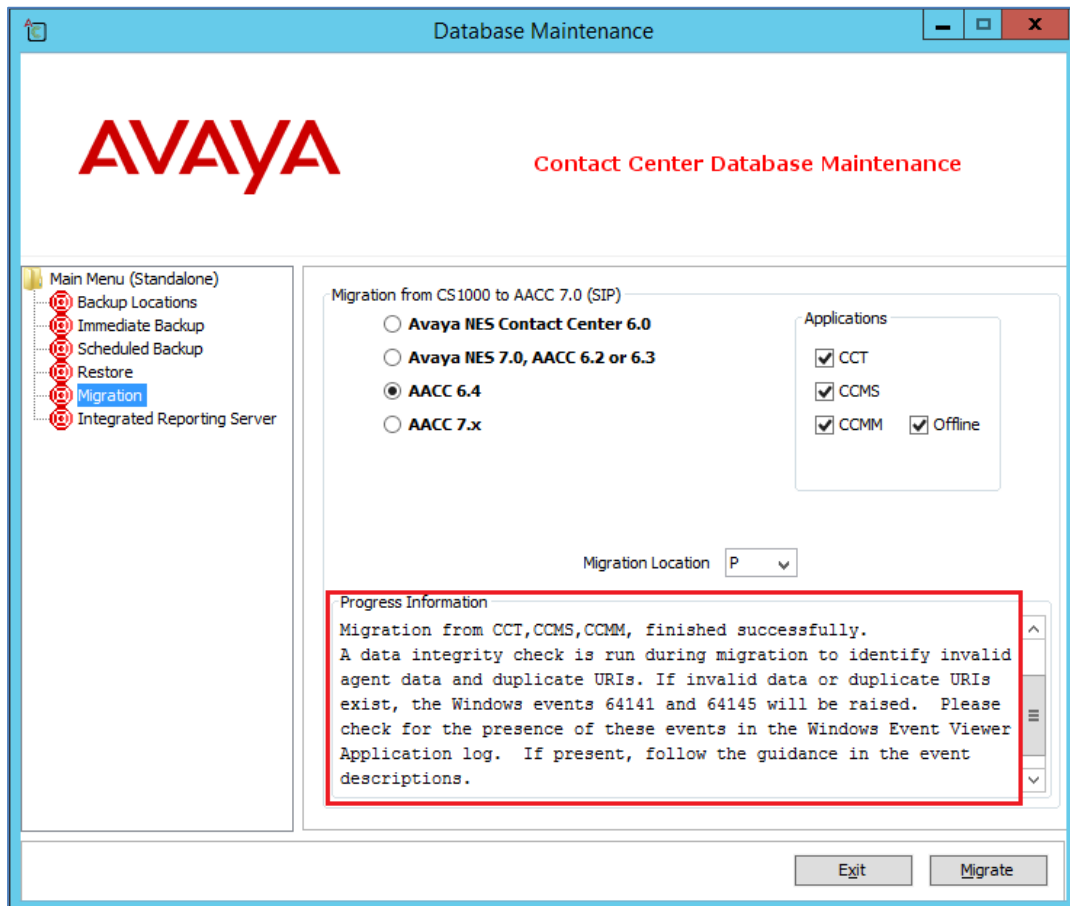
On the target SIP Server: Using the Database Maintenance application

- Launch the Database Maintenance application.
- From the *Migration* node, select the migration source that is applicable for your AML Contact Center version. The following options are supported:

Migration Option	Description
Avaya NES Contact Center 6.0	Legacy database – Contact Center Migration Tool must be used to create backup Administration Data (CCMA) must be restored separately
Avaya NES 7.0, AACC 6.2 or 6.3	Administration Data (CCMA) must be restored separately
AACC 6.4 (shown below)	Administration Data (CCMA) must be restored separately
AACC 7.0	Administration Data (CCMA) is restored using the common Contact Center Database Maintenance application: 



- The *Migration Location* must point to a network location containing the AML contact center backup. If you are migrating from an older release where CCT and CCMS data are distributed over multiple servers, each application will need to be migrated separately. This data will then be consolidated onto the newly deployed AACC 7.0 server.
- Click *Migrate* to migrate contact center data. This will migrate configuration and historical data.
- Progress will be displayed on screen:



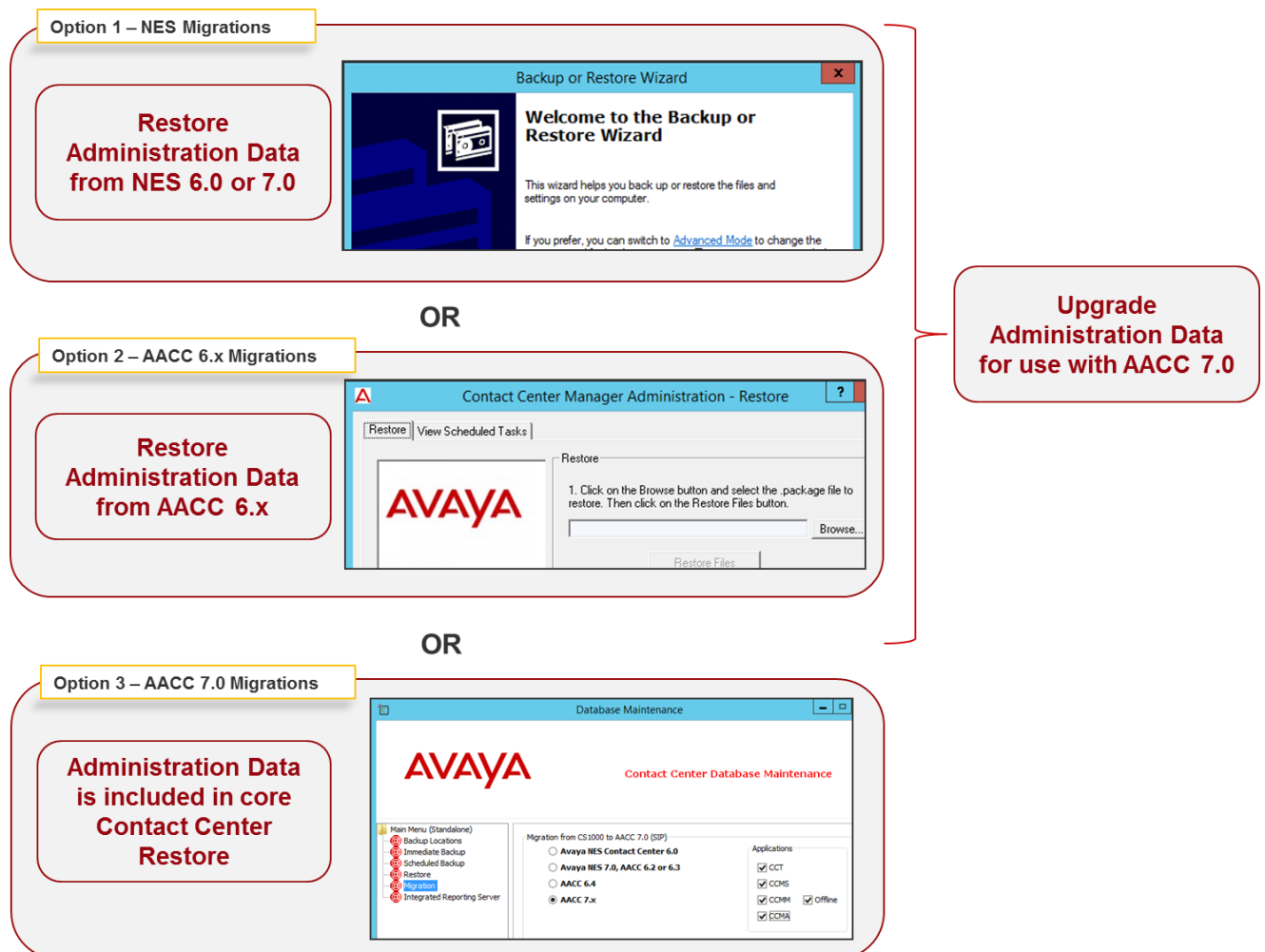
IMPORTANT: A data integrity check is run as part of the migration to guard against duplicate URIs being configured in the system. Once the migration completes, inspect the Windows

Event log for the presence of 64141 or 64145 events and follow the guidance in the event description. Refer to [Appendix A: Migration Event Codes](#) for details on events.

- Only information applicable to a SIP based contact center will be migrated.
- The resources listed in the table below will be updated with the correct SIP domain information as part of the migration. They will also be uniquely identified using either agent ID or resource number. These values can be changed later if required using the SIP Spreadsheet and Microsoft Excel. This is detailed in the next step.

Resource	Field	Migrated Value
Agent	Voice URI	agentID@sipdomainname e.g. 42000@accsl.avaya.com
CDNs (Route Points)	URI	CDN@sipdomainname e.g. 3201000@accsl.avaya.com
DNIS	URI	DNIS@sipdomainname
Landing Pads	URI	LandingPad@sipdomainname
If moving from a Workgroup environment, agents require a windows domain account to be configured. This can be done using the CCMA Consolidated Configuration Tool – Spreadsheet & Excel.		

On the target SIP Server: Migrating Administration (CCMA) data



Option 1: NES Migrations – Restoring & upgrading Contact Center Manager Administration data from NES 6.0 or NES 7.0

IMPORTANT: Before migrating CCMA data, you must restore the CCMA portion of the AACC 7.0 backup taken immediately after ignition. CCMA data from the source server can only be migrated to a system that does not contain any customer data.

For this option, it is necessary to restore and upgrade data from an older (Windows 2003) based backup. This requires use of the NTBackup utility on the newly deployed AACC 7.0 server, this utility is installed by Avaya Aura Contact Center 7.0.

Refer to the following chapters of the Upgrading and patching Avaya Aura Contact Center Release 7.0, Part 7: AML to SIP migrations for specific procedures to follow for this option:

- *Chapter 22: AML to SIP Migration – Restoring Contact Center Manager Administration data from NES 6.0 or NES 7.0*
- *Chapter 22: AML to SIP Migration – Upgrading the Contact Center Manager Administration NES database*

Option 2: AACC 6.x Migrations – Restoring & upgrading Contact Center Manager Administration data from any AACC 6.x release

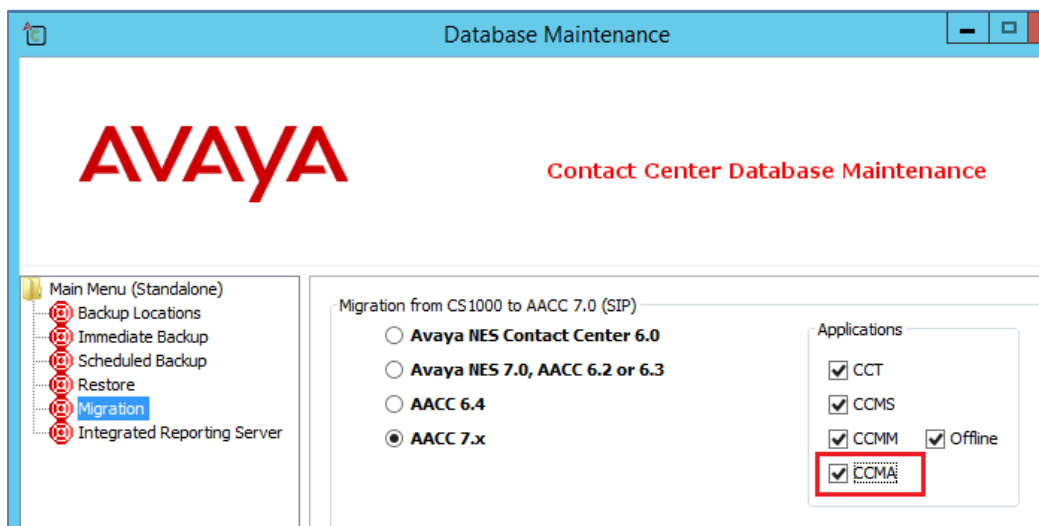
IMPORTANT: Before migrating CCMA data, you must restore the CCMA portion of the AACC 7.0 backup taken immediately after ignition. CCMA data from the source server can only be migrated to a system that does not contain any customer data.

Refer to the following chapters of the Upgrading and patching Avaya Aura Contact Center Release 7.0, Part 7: AML to SIP migrations for specific procedures to follow for this option:

- *Chapter 22: AML to SIP Migration – Restoring Contact Center Manager Administration data*
- *Chapter 22: AML to SIP Migration – Upgrading the Contact Center Manager Administration data*

Option 3: AACC 7.0 Migrations – Migrating Contact Center Manager Administration data from an AACC 7.0 release

For this option, the Administration data is already included in the core Contact Center backup (via the Database Maintenance utility) and is available as a migration option in the Database Maintenance utility:



STEP 4

Administration & Multimedia Configuration:

- After the AML contact center data has been migrated, a number of configuration steps are required.

Administration Configuration:

- In AACC Administration, the entry for the contact center servers (CCMS, CCT & CCMM – where applicable) must be updated to point to the newly installed SIP server. This can be done via the Administration, Configuration component using an administrator level account such as *webadmin*

AVAYA **Configuration**

Server Download Status Launchpad Help

☒ ACCSLGAL_MM1
☒ ACCSLGALAACC1
☒ ACCSLGAL1CCT

Server Properties

Type: CCMS

Server Name: SIPSERVER1

IP Address: 10.134.63.34

Display Name: SIPAACC

Login ID: sysadmin

Password: *****

DSN Prefix: CCMS

The following ODBC DSN will be automatically created for this system:
CCMS_10.134.63.34_DSN

Submit Reset

- Update Server Name and IP address information of the previously configured AML entries to point to the new SIP AACC. This must be done for CCMS, CCT and CCMM servers. If a Multimedia server was not previously part of the AML solution, a new CCMM entry is required to point to the local SIP based deployment.

IMPORTANT: When changing the server name and IP address for the CCMS entry, you must specify the sysadmin login ID and password as was previously set on the AML system sysadmin credentials are migrated with the cache database

- Refresh Servers to update licensed features and settings

Multimedia Configuration:

The following changes are performed using the Multimedia Dashboard:

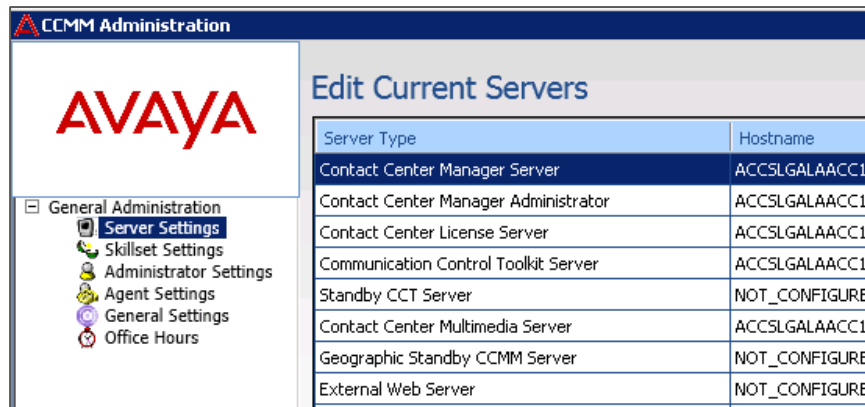
- Launch the Multimedia Dashboard on the new SIP AACC server
- Edit the Contact Center Manager Administrator entry and set local CCMA server name

Server Availability (Ping Status)	
Server	Computer Name
☒ Contact Center Manager Server	ACCSLGALAACC1.
☒ Contact Center Manager Administrator	ACCSLGALAACC1
☒ Contact Center License Server	ACCSLGALAACC1.
☒ Communication Control Toolkit Server	ACCSLGALAACC1.

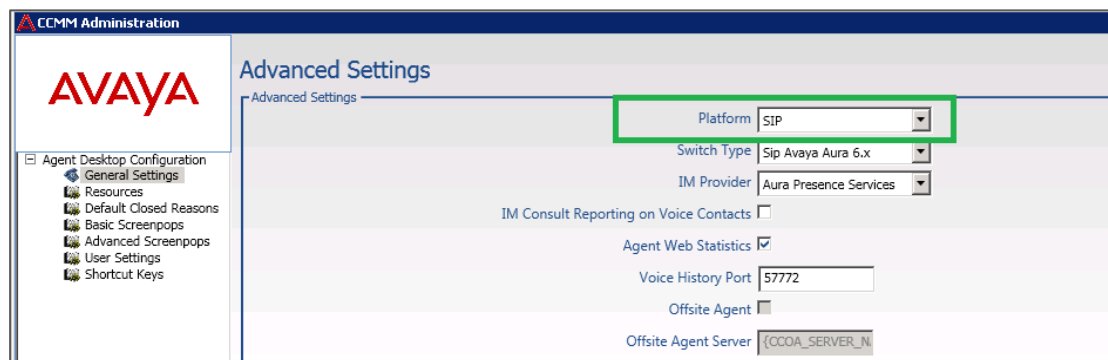
- Edit the Contact Center Multimedia Server entry and set the new CCMM server name.
- Click on the update button to ensure the changes have been added to the system

The following changes are performed using the CCMM Administration Application:

- Login to AACC Administration as *webadmin* OR a user account with Administrator privileges
- Launch the CCMM Administration application and choose *General Administration* and *Server Settings*



- Ensure the correct entries have been specified for all solution servers. This includes the Manager Server, License Server, Control Toolkit Server, Multimedia server and applicable inbound and outbound mail servers.
- Save changes
- Choose *Agent Desktop Configuration, Advanced Settings*:



- Set the Platform Type to **SIP**
- Set the Switch Type to **Sip Avaya Aura 6.x**

Agent Desktop:

When you are ready for agents to begin using the newly deployed AACC system, install Avaya Agent Desktop on each of your agent workstations. If any previous versions of either Avaya Aura Agent Desktop (where the source is AACC) OR Contact Center Agent Desktop (where the source is NES) are installed, remove them before installing Avaya Agent Desktop.

It is not possible to run two versions of Avaya Agent Desktop on the same workstation.

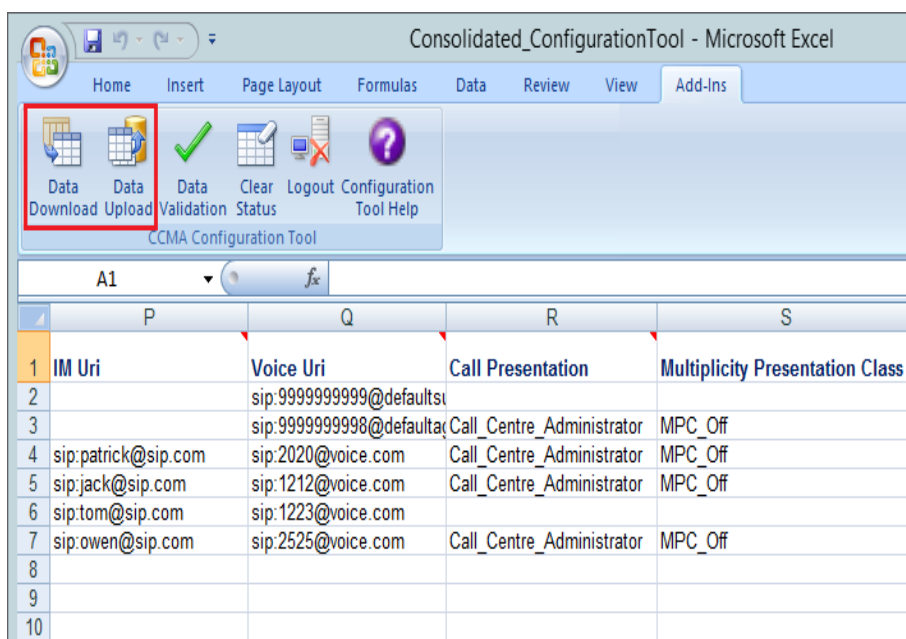
Converting custom RCW reports to AACC 7.0 format:

This is an optional step and only applicable if you have created custom reports using RCW on the source AML deployment. Copy the zip file of converted RCW reports (prepared during Step 2) to the newly deployed AACC 7.0 server and convert each report to SSRS

format. For more information on this procedure, refer to the *Upgrade and Patching Avaya Aura Contact Center* guide.

STEP 5

Update migrated Data for use in a SIP based AACC:



Use the SIP Spreadsheet and Microsoft Excel to:

- Download migrated contact center data from the new SIP based AACC
- Bulk change details for Agents, Supervisors, Route Points (CDNs), DNIS etc. The following table highlights the main items that must be updated, for complete details for fields that can be updated refer to: [CCMA Consolidated Configuration Tool](#)

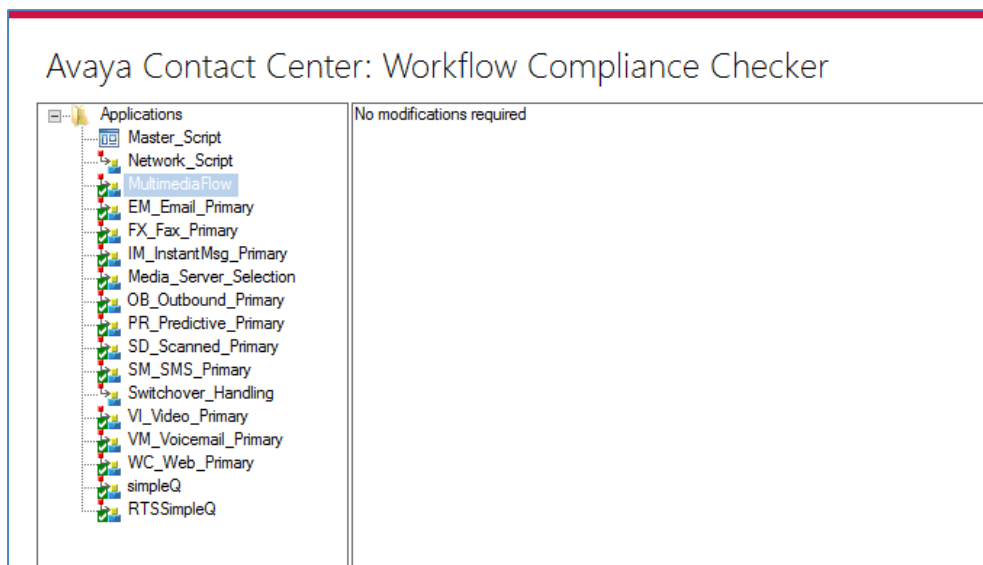
Resource	Setting/field												
Agent / Supervisor:	Voice URI												
	Call Presentation Classes (including Multiplicity)												
	Domain (Windows)												
	User ID (Windows domain account)												
CDNs (Route Points)	URI												
DNIS	URI												
Routes	Update the Name field on any migrated RAN routes to match the music/.wav location on your Media Server												
Call Presentation Classes	Invalid 'Presentation Options' must be updated to either <i>Call Force Delay</i> or <i>Return to Queue</i>. Any invalid options will be highlighted in red when viewed in Administration – Configuration <table><tr><th>Presentation Option</th><th>Call Force Delay</th></tr><tr><td>Call Force Delay</td><td>0</td></tr><tr><td>Return To Queue</td><td>0</td></tr><tr><td>Let Call Ring</td><td>0</td></tr><tr><td>Call Force Delay</td><td></td></tr><tr><td>Return To Queue</td><td></td></tr></table>	Presentation Option	Call Force Delay	Call Force Delay	0	Return To Queue	0	Let Call Ring	0	Call Force Delay		Return To Queue	
Presentation Option	Call Force Delay												
Call Force Delay	0												
Return To Queue	0												
Let Call Ring	0												
Call Force Delay													
Return To Queue													

- Upload the modified data to the Contact Center to allow agents to handle SIP contacts
- **IMPORTANT:** Updating DNIS and CDN URI's can only be done via the SIP Spreadsheet and Microsoft Excel. This requires a system restart to take effect. It is not possible to update these fields using the CCMA UI.

NOTE: Not all fields can be updated using Excel. For more information on supported operations please refer to: [CCMA Consolidated Configuration Tool](#)

STEP 6

Workflow Compliance Checker & Orchestration Designer:



- A Workflow Compliance Checker is available in AACC 7.0. This can be used to:
 - Analyze migrated customer workflows
 - Identify unsupported commands, blocks and intrinsics
 - Generate a report that provides guidance to the customer in making any necessary changes using Orchestration Designer.
- The application can be launched by browsing to: `D:\Avaya\Contact Center\Common Components\WorkflowComplianceChecker` and running the `Workflow Application.exe`

For sample call flows and guidance on how to implement equivalent functionality in a SIP based solution, please refer to [Reference implementation – Key Changes](#).

5. Reference Deployment – CS1000 Telephone Only, Voice Only Contact Center Deployment

5.1 Deployment Summary

The following deployment represents a sample voice-only CS1000 solution using only telephones (no agent desktop) for AACC agents. The objective is to migrate to a SIP based AACC with an Avaya Aura Unified Communications infrastructure.

To migrate to a SIP based AACC, it is necessary to install and deploy a new AACC 7.0 solution – new hardware required. This system and the associated Unified Communications components must be provisioned as per the existing AACC solution user guides. This section describes the key areas that should be planned for as part of this deployment:

CS1000 Deployment Details:

Solution and Environment & Capacity:

- Voice-only Contact Center deployment
- AACC server installed in Windows Workgroup
- Agents deployed with telephone sets only – Model 3904 desk-phone
- 400 agents / 7,000 calls per hour

Contact Routing & Flows:

- CDN and DNIS routing

Agent & Supervisor deployment:

- Phone only agents
- Using 'Answer by placing DN call on hold' presentation class

Reporting:

- Custom reports and report groups

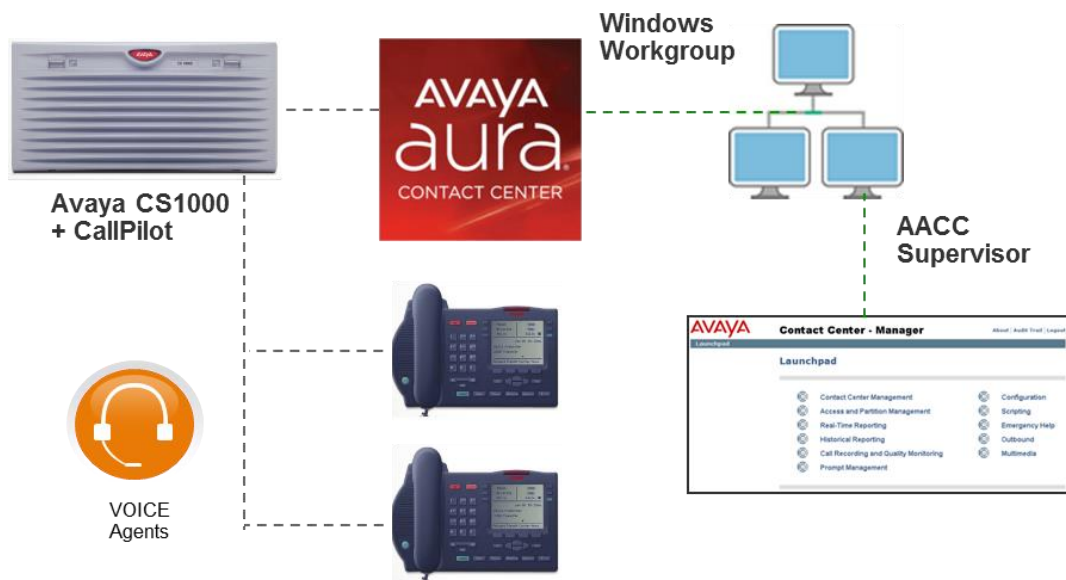
Voice Services – Call Pilot:

- Back-end integration via AACC flows, providing:
 - RAN announcements
 - Controlled Broadcast Announcements
 - Play Prompt / Collect Digits

Additional Features:

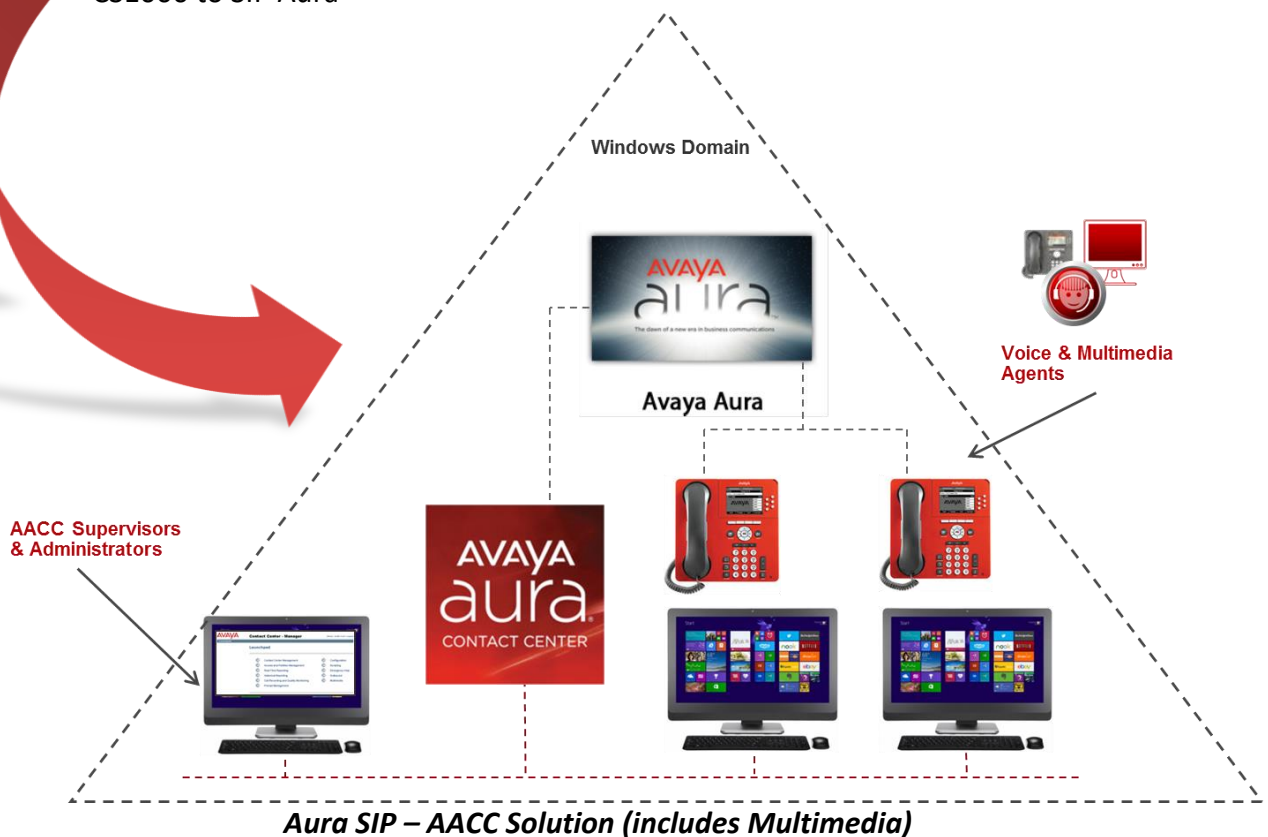
- Agent Greeting Card
- Avaya Call Recording
- Custom Wallboards using RSM feeds
- HDX integration

5.2 Reference implementation – Overview



CS1000 Phone-only solution

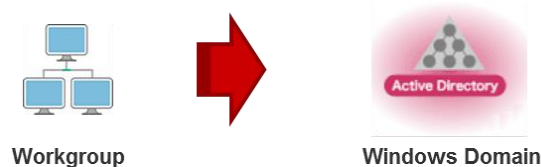
CS1000 to SIP Aura



5.3 Reference implementation – Key Changes

Solution and Environment

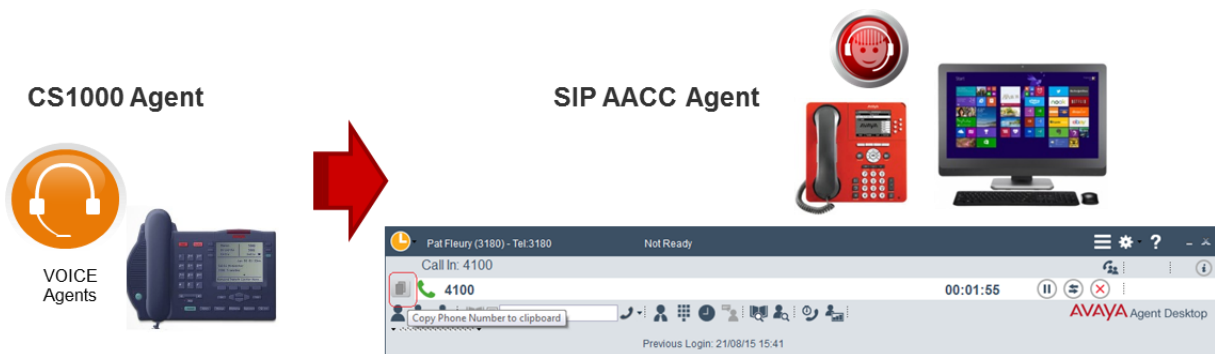
1) Network & Topology changes:



- SIP based (Avaya Aura) AACC is installed in a Windows Domain environment. This is applicable for AACC servers, agent/supervisor & administrator workstations.
- Allows for seamless hostname resolution between AACC servers and agent/supervisor workstations
- Supports single sign-on for agents to connect to Avaya Agent Desktop
- Allows for full Mission Critical High Availability if required

2) Agent Changes:

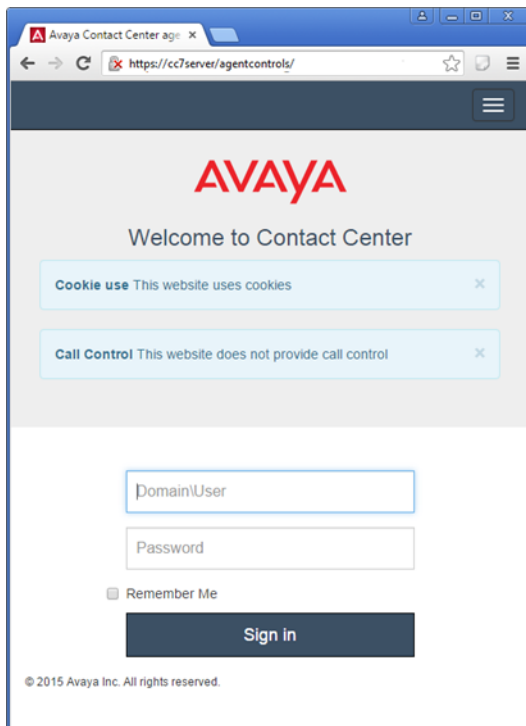
Option 1: Agent uses Avaya Agent Desktop



- SIP agents use Avaya Agent Desktop (Windows workstations required) to login to AACC and handle contacts
- Supports all AACC multimedia contact types. Agents can handle multimedia contacts if deployed with AACC in the future
- Provides a large range of agent features e.g. phonebook, Customer history, Presence integration, screen pop etc.

OR

Option 2: Agent uses new Agent Browser application and telephone



- If the target SIP based AACC will remain a voice only solution and the full Avaya Agent Desktop is not required (e.g. no Multimedia contact routing), the new Agent Browser application can be used in conjunction with a telephone.
- This provides the minimum SIP Contact Center functionality required for voice-only agents to handle voice contacts without needing to deploy full Agent Desktop.
- Its design means it can run in any modern browser meeting certain minimum requirements.

NOTE: The Contact Center Agent Browser application does not provide call control or multimedia features. Agents must use a desktop phone for call control. Please refer to the ***Using the Contact Center Agent Browser application*** document for full details

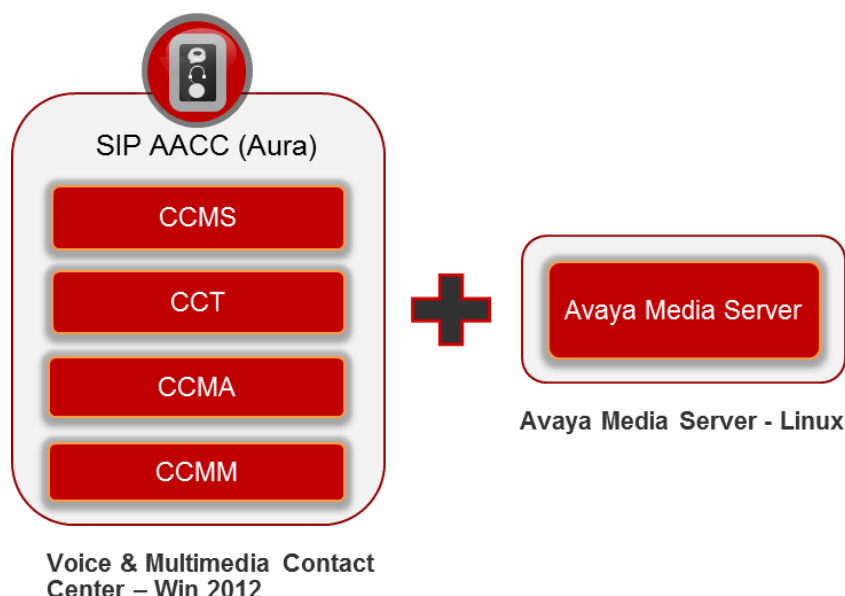
In both options:

- Agents move from a **3904** model digital desk-phone to a **9641G** model desk-phone (other phone and soft-phone options available). Refer to [Review Telephone Support](#)
- Agents in CS1000 have one key for contact center calls and one key for personal calls. In SIP, three lines (Third Line Appearance feature) are used to receive and make calls

3) Hardware Requirements:

For this solution (400 active agents), the following High-End solution is used:

- Voice and Multimedia Contact Server without Avaya Media Server
- Avaya Media Server (Linux) standalone



- For SIP based AACC Contact Centers, Multimedia (co-resident CCMM in this example) is always required. This is to allow administrators to configure and deploy Agent Desktop.
- In this example, a high-end server is specified for the Voice and Multimedia contact center. Please refer to the Avaya Aura Contact Center Overview and Specification Release 7.0 for more information on recommended server specifications

Note: Virtualized guest installs of AACC 7.0 (via DVD) are also possible. AACC 7.0 no longer supports software appliance (OVA) deployments. Consult published AACC guides for more information.

Configuration & Administration

4) Unified Communications – Resources

Unified Communication resources and components exist in a SIP enabled domain and the URI information detailed below refers to this domain.

Route-Points (CDNs) and DNIS:

- Dial patterns must be configured as part of the Aura (Session Manager) commissioning – This ensures that voice contacts are routed to AACC
- Route-Point and DNIS resources must be configured with URI information in AACC:
 - The migration process will automatically set the DNIS and CDN URI values to the format: 'sip:<resource number>@<SIP domain>' (e.g. sip:60000@acssl.avaya.com). Use the SIP Spreadsheet and Microsoft Excel to bulk change these URI values if required.
 - CS1000 vs. Aura specific screens are shown below:

Configuration					
CDNs (Route Points)					
CDNs	Open Queue	Landing Pads			
Name	Number	Call Type	Acquired?	Status	
CDN_60000	60000	Local	☒	Acquired	
CDN_60001	60001	Local	☒	Acquired	
*			☐		

CS1000: Sample Route-Point Configuration in AML (CCMA -> Configuration)

Configuration						Logged in user: webadmin
Launchpad Help						
CDNs (Route Points)						Server: CECGALA
CDNs	Open Queue			Acquire All CDNs	De-acquire All CDNs	Refresh Status
Name	Number	URI	Call Type	Acquired?	Status	
CDN_60000	60000	sip:60000@accsl.avaya.com	Local	☐	Not Acquired	
CDN_60001	60001	sip:60001@accsl.avaya.com	Local	☒	Acquired	
CDN_3203200	3203200	sip:3203200@accsl.avaya.com	Local	☒	Acquired	
*				☐		

Aura SIP: Sample Route-Point Configuration in SIP (CCMA -> Configuration)

- The SIP Spreadsheet and Microsoft Excel can be used to update URI information – System restart required. This cannot be done using the Administration UI.
- Once the correct URI information has been specified, the Contact Router can be used to apply any appropriate Flow/Applications.

5) Avaya Media Server Resources:

The Avaya Media server is an integral part of an AACC SIP based contact center and provides conferencing services for contact center calls – i.e. calls are anchored on the Avaya Media Server. Voice Services are also provided by the Media server (replaces CS1000 based CallPilot), allowing announcements, music and digit collection capability as part of AACC contact flows. Agent greeting (optional licensed feature) is provided by the Media Server and is integrated into AACC for administration.

In this example (given the capacity requirements), a standalone, Linux based Media Server will be used. This Media Server is configured in AACC, Configuration. Sample screens are illustrated below:

Configuration

[Launchpad](#) [Help](#)

Media Services and Routes

	Service Name	Treatment Address
▶	ACC_APP_ID	sip:eb8acaa5-509d-11e4-9125-00ff70260b11@accsl.avaya.com
	XDIALOG	sip:__sip-xdialog@domain.com
*		

Routes(Target Media Servers)-ACC_APP_ID

Available		Selected (1)	
	>> > < <<	accslaacc701	↑ Rank ↓
Submit			

Once configured, the Media Server must be assigned the CONF service for media anchoring. XDIALOG is optional and depends on configuration

Media Servers and Media Service resources are safe-guarded or protected as part of the AACC data migration. This means that any configuration and data entry made in these screens will remain in place after an AML migration. No data will be over written.

Integrated Prompt Management (launched via main AACC Administration Launchpad) is illustrated below:

AVAYA **Prompt Management** Logged in user: webadmin | [Logout](#)

Status Launchpad Help

Namespace: cec.avaya.com Content Group: en_us Server: cecgalaacc1

Media Content Groups

Name	Content Type	Size	Created	Modified
OutOfHours.wav	audio/x-wav	199820	2012-03-02 12:15:07	2012-03-02 12:15:07
OutOfService.wav	audio/x-wav	145716	2012-03-02 12:15:07	2012-03-02 12:15:07
PlayPrompt.wav	audio/x-wav	63308	2012-03-02 12:15:07	2012-03-02 12:15:07
PromptMenu.wav	audio/x-wav	163252	2012-03-02 12:15:07	2012-03-02 12:15:07
PromptSubMenu1.wav	audio/x-wav	183740	2012-03-02 12:15:07	2012-03-02 12:15:07
PromptSubMenu2.wav	audio/x-wav	188836	2012-03-02 12:15:07	2012-03-02 12:15:07
SecondRan.wav	audio/x-wav	58452	2012-03-02 12:15:07	2012-03-02 12:15:07
SMSThankYou	audio/x-wav	378855	2014-07-23 11:25:30	2014-07-23 11:25:30
Timeout.wav	audio/x-wav	24396	2012-03-02 12:15:07	2012-03-02 12:15:07
TimeoutSubMenu.wav	audio/x-wav	25652	2012-03-02 12:15:07	2012-03-02 12:15:07

Page size: 10 Number of items: 28

Media files stored on Media Server

Edit Media

Media Name: OutOfHours.wav

User Defined Partitions

This prompt is currently assigned to 0 partitions.

☐ test Supports partitioning

Admin Operations

Add Media Save Download Media Delete Media Delete Content Group Rename Content Group Cancel

Integrated Prompt Management

Allows AACC administrators and supervisors to:

- View and download announcements (WAV files)
- Add new media and announcements and delete existing files
- Manage Content groups.

The above actions are controlled by Access Classes and can be limited to a subset of prompts based on partitioning.

6) Agent Configuration:

In this example, agents are migrated from a phone-only solution to Avaya Agent Desktop – this is required by SIP deployments. All applicable agent, supervisor and related (skillsets, call presentation etc.) are automatically migrated as part of the process.

The following screens provide a summary of the important fields that must be set for each Agent and Supervisor/Agent – All of the highlighted items can be updated using the SIP Spreadsheet and Microsoft Excel.

CS1000: Agent configuration

Contact Center Management

Logged in user: webadmin | Logout

[Unchpad](#) [Help](#)

Agent Details: Vincent Browne **Server:** AMLServer

▼ [User Details](#)

First Name: *	Vincent	User Type:	Agent	Login ID can be retained for SIP Not Applicable for SIP
Last Name: *	Browne	Login ID:	* 3201800	
Title:	Mr	Personal DN:	3208888	
Department:	Sales	ACD Queue:		
Language:	English	ACD Queue Error:		
Comment:	Sample agent			

▼ [Agent Information](#)

Primary Supervisor: *	Kavanagh Patrick	Call Presentation:	Call_Centre_Administrator
Agent Key:		Multiplicity Presentation Class:	MPC_Off
Login Status	Logged Out	Threshold:	Agent_Template
		Tn Name:	

▶ [Contact Types](#)

▶ [Skillsets](#)

▶ [Partitions](#)

SIP (Aura): Agent Configuration

Contact Center Management

Logged in user: **webadmin** | [Logout](#)

[Inchpad](#) [Help](#)

Agent Details: **Vincent Browne** Server: **ACCSLGALAACC1**

User Details

First Name: Vincent	User Type: Agent				
Last Name: Browne	Login ID: * 3201800				
Title: Mr	Voice URI: sip:3201800@accs1.avaya.com ◀ Voice URI required for SIP				
Department: Sales	IM URI: sip:				
Language: English	Account Type: ☒ Create CCT Agent				
Comment: Sample Agent	CCT Agent Login Details <table border="0"> <tr> <td>Domain</td> <td>accs1</td> </tr> <tr> <td>User ID:</td> <td>vincent</td> </tr> </table> ◀ Domain account for AAAD login	Domain	accs1	User ID:	vincent
Domain	accs1				
User ID:	vincent				

▶ [Associate User Account](#) ◀ Choose Active Directory accounts for agent login here

Agent Information

Primary Supervisor: * King Martin	Call Presentation: Call_Centre_Administrator
Login Status	Multiplicity Presentation Class: MPC_Off
Logged Out	Threshold: Agent_Template

▶ [Agent Greeting](#) ◀ Agent Greeting is provided by Avaya Media Server

▶ [Offsite Agent](#) ◀ Offsite Agent is optional for SIP based AACC

▶ [Contact Types](#)

▶ [Skillsets](#)

▶ [Partitions](#)

- Agent Login ID's remain the same as those configured in CS1000.
- The migration process will automatically set the agent Voice URI value in the format: 'sip:<Login ID>@<SIP domain>' (e.g. sip:3201800@accsl.avaya.com). Use the SIP Spreadsheet and Microsoft Excel to bulk change these URI values if required.
- Agent login details are required. In this example 'accsl/vincent' is selected as the user account for this agent. When logging into the contact center (via Agent Desktop), this Windows domain account will be used – Agent Desktop also supports single sign-on with the Windows desktop

Note: The SIP Spreadsheet and Microsoft Excel can be used to bulk update any migrated agent & supervisor accounts. Complete details are included in the [CCMA Consolidated Configuration Tool](#) section of this document.

7) Agent Call Presentation Classes:

CS1000: Sample Call Presentation Class configuration:

Configuration							
Call Presentation Classes							
Name	Presentation Option	Call Force Delay Timer	Return To Queue After N Seconds	After Return to Queue, Make Phoneset	After Call, Break for N seconds	Answer By Placing DN Call On Hold	Agent Reserved for Network Call
Call_Centre_Admin	Return To Queue	0	18	Not Ready	0	☒	☒
AgentCallPresentati	Return To Queue	0	18	Not Ready	0	☒	☒
TestAML	Let Call Ring	0	N/A	N/A	0	☐	☐
*	Call Force Delay					☐	☐
	Return To Queue						
	Let Call Ring						

The following options are not applicable for SIP based contact centers:

- Let Call Ring
- Answer By Placing DN Call On Hold

Any migrated Call Presentation classes using the above options must be updated to an appropriate setting. Invalid options will be highlighted in red when viewed in Configuration.

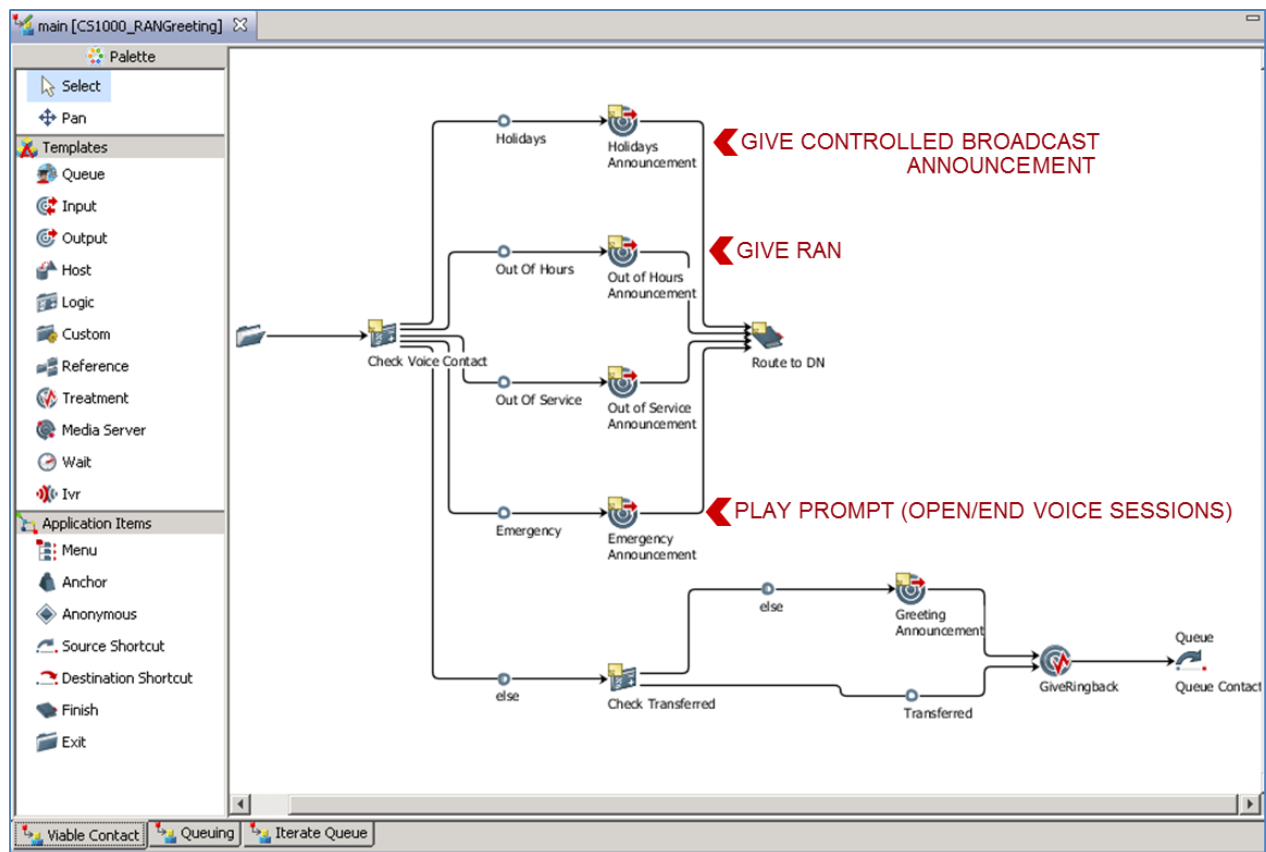
8) Contact Routing and Flows

A sample application flow for this reference deployment is called “**CS1000_RANGreeting**”. This was initially based on the CS1000 template flow: “**Simple Greeting RAN**” but has been enhanced for the purposes of this document to also include:

- GIVE CONTROLLED BROADCAST ANNOUNCEMENT
- OPEN/END VOICE SESSION with PLAY PROMPT

While commonly used operations in a CS1000 solution, these commands are not supported in a SIP based environment. This section describes how this functionality can be provided on a SIP based solution.

CS1000: Reference AML based flow



GIVE CONTROLLED BROADCAST ANNOUNCEMENT:



Holidays Announcement Output Block

main [CS1000_RANGreeting] Holidays Announcement [CS1000_RANGreeting]

Output Options:
☐ RAN ☒ Play Prompt ◀ CS1000 (AML) specific Play Prompt Option

Controlled Broadcast Announcement:
☒ Play as a Broadcast Announcement ◀ Broadcast
☐ Priority:
☒ Continuous

☒ Access IVR DN:

Prompt List:

Condition	Wording / Value	Voice Segment / Speak As
☒ Always	The Contact Center is closed during the holidays	cp_HolidayAnnouncement_gv

Voice Segment global variable

Selected Row Details:

Setup Prompts Processing Transitions

Script representation of above:

```

86SECTION HOLIDAYS_ANNOUNCEMENT
87/*
88    This Announcement needs to be configured to customers'
89requirements.
90
91Play a Holidays Announcement & route to a mailbox.
92*/
93GIVE CONTROLLED BROADCAST ANNOUNCEMENT 2001
94/* The Contact Center is closed during the holidays */
95PLAY PROMPT CONTINUOUS VOICE SEGMENT cp_HolidayAnnouncement_gv
96
97EXECUTE ROUTE_TO_DN
  
```

OPEN/END VOICE SESSION with PLAY PROMPT:



Emergency Announcement Output Block

main [CS1000_RANGreeting] Emergency Announcement [CS1000_RANGreeting]

Output Options
☐ RAN ☒ Play Prompt CS1000 (AML) specific Play Prompt Option

Controlled Broadcast Announcement
☐ Play as a Broadcast Announcement
☐ Priority:
☐ Continuous

☒ Access IVR DN:

Prompt List:

Condition	Wording / Value	Voice Segment / Speak As	Barge-in
☒ Always	Announce emergency message	cp_EmergencyAnnounceme...	true

Voice Segment
global variable

Selected Row Details

Setup Prompts Processing Transitions

Script representation of above:

```

122SECTION EMERGENCY_ANNOUNCEMENT
123/*
124Play an Emergency Announcement & route to a mailbox
125*/
126OPEN VOICE SESSION 2001
127/* Announce emergency message */
128PLAY PROMPT VOICE SEGMENT cp_EmergencyAnnouncement_gv
129END VOICE SESSION
130EXECUTE ROUTE_TO_DN
  
```

How to migrate the “CS1000 RANGreeting” flow to a SIP based solution:

AML based scripts and flows are automatically migrated to the newly deployed SIP AACC as part of the CS1000 to SIP migration process. They are not modified in any way as part of the migration and require manual updates to allow them to function in a SIP environment. The Workforce Validation tool can also be used to highlight invalid syntax and provide guidance. Please refer to the section: [AACC Scripting & Application Flows Considerations](#) for more details.

In this reference SIP based deployment, the Avaya Media Server can be used to provide voice services functions previously provided by CallPilot on CS1000. To do this, note the following changes:

- Any blocks providing RAN or MUSIC must be reviewed and any RAN or MUSIC routes must be updated to point to the correct announcement file on the Avaya Media Server. This is done via the CCMA – Configuration component.

Configuration		
Routes		
Name	Number	Threshold Class
rock1	10	Route_Template
pop	11	Route_Template
Sample_Music	511	Route_Template
Default_Route	999	Route_Template
*		

- Provided the route is configured to use the correct announcement file (Example pop WAV file), no scripting/flow changes are required.

How to migrate the Holidays Announcement & Emergency Announcement output blocks:

As already described, these blocks contain commands that are not supported in a SIP environment and require manual updates to use alternative options. The enhanced IVR block, used in conjunction with the Avaya Media Server is recommend for both options:

Once the AML flow has been imported, invalid options will be greyed out on screen and can no longer be modified. Errors will also be displayed in the ‘Problems’ tab as required Voice Segment variables are no longer present. This is illustrated below:

SIP (Aura): Invalid options are disabled and missing variables will report as errors.

The screenshot shows the 'Emergency Announcement [CS1000_RANGreeting]' configuration window. The 'Output Options' section has 'Play Prompt' selected, with a red arrow pointing to the text 'Options not available once migrated to SIP solution'. The 'Controlled Broadcast Announcement' section has three unchecked options: 'Play as a Broadcast Announcement', 'Priority' (set to 4), and 'Continuous'. The 'Access IVR DN' section is checked, with a value of '2001' and a red arrow pointing to the text 'Reference to CallPilot resources'. The 'Prompt List' table shows a single row for 'Always' with the condition 'Announce emergency message', voice segment 'cp_EmergencyAnnounceme...', and barge-in 'true'. A red arrow points to the voice segment with the text 'Not Valid in SIP'. The 'Selected Row Details' section is empty. The 'Setup' tab is selected, and the 'Problems' sub-tab shows 2 errors: 'Undefined variable:cp_EmergencyAnnouncement_gv' and 'Undefined variable:cp_HolidayAnnouncement_gv', both located in the 'Emergency Announcement' and 'Holidays Announcement' respectively.

Condition	Wording / Value	Voice Segment / Speak As	Barge-in
Always	Announce emergency message	cp_EmergencyAnnounceme...	true

Description	Resource	Path	Location	Type
Errors (2 items)				
Undefined variable:cp_EmergencyAnnouncement_gv			Emergency Announcement	Problem
Undefined variable:cp_HolidayAnnouncement_gv			Holidays Announcement	Problem

To resolve the above issues, it is recommended to use the enhanced IVR block to provide announcements – applicable for both Broadcast and Play Prompt options.

Note: The 'RAN' option could also be used in this scenario but the enhanced IVR block is described here as it provides a larger range of options.

Sample configuration screen shown below:

SIP (Aura): Enhanced IVR block – In this case the name of the prompt (.wav file) stored on the Avaya Media server is specified (or variable containing the prompt name).

Script representation of above:

```
4SECTION HOLIDAYS_ANNOUNCEMENT
5/*
6 Play an Holidays Announcement & route to a mailbox.
7*/
8ASSIGN CONTACT DATA "SIP_FROM_ADDRESS" TO c_sip_from_add_cv
9ASSIGN CONTACT DATA "SIP_TO_ADDRESS" TO c_sip_to_add_cv
10ASSIGN c_sip_to_add_cv TO vxmlfrom
11ASSIGN c_sip_from_add_cv TO vxmlto
12ASSIGN c_play_only_gv TO voicexml
13ASSIGN "Holidays" TO prompttoplay
14ASSIGN "info" TO method
15
16GIVE IVR WITH VXML TREATMENT voicexml PARAMETERS prompttoplay, vxmlfrom, vxmlto
17EXECUTE ROUTE_TO_DN
```

Note the following parameters and variables:

- SIP_FROM_ADDRESS: The SIP Contact Source is the value received in the From address in the SIP INVITE message. Example: customer@cc.avaya.com
- SIP_TO_ADDRESS: The value received in the To address in the SIP INVITE message.
- c_play_only_gv: By default this will specify 'PlayPrompt.vxml' option.
- Holidays.wav is assigned the prompt to play.

More advanced call-flows can be developed using the IVR block, these can also include digit collection. Refer to the 'Digit Collection' sample template for more detail OR review some of the sample options below:

SIP (Aura): Sample Play and Collect options using the IVR block. This is typically used as an alternative to Voice Sessions on CS1000

Play Prompt ☒ Play and Collect ☐ Send IM ☐ Free Form **Play and collect option selected**

Prompt Name: **Prompt to play e.g. on Avaya Media Server**

Return Value: **Collected digits will be stored in this variable**

☐ Service URI:

☐ Session Based

Digit Collection

☒ Number of Digits:

☒ Timeout (secs):

☒ Terminating Character:

☐ Interrupt IVR:

Digit Collection options - configurable

Setup Ivr Processing Transition

9) Agent Greeting

In a SIP based contact center, Agent Greeting is provided by AACC 7.0 and is integrated into AACC in terms of administration and configuration. As with CS1000, it is provided as an additional licensed feature. In the case of migrating from CS1000 to a SIP enabled AACC, it is recommended to deploy and provision Agent Greeting as per the existing documentation. Agents should re-record their greetings on the new system. Also refer to the [Review Voice Services & Self Service options](#) for more feature details.

The SIP Spreadsheet and Microsoft Excel can be used to bulk update agent properties with Agent Greeting settings.

Sample screens illustrated below:

SIP (Aura): Integrated Agent Greeting configuration screens. This is accessed via the AACC Configuration Launchpad option.

Configuration

Logged in user: **webadmin** | [Logout](#)

Status
Launchpad
Help

Agent Greeting Settings

Server: **cecgalacc1**

General Settings

Agent Greeting enabled

☒

Mute agent on greeting playback

☒

"No greeting" warning tone

☐

Allow an agent to interrupt the greeting

☒

Maximum allowed greeting length

seconds

Agent Greeting Recording Number

Prompt language

Time of Day Settings

Enable time of day greetings

☒

Morning start time

(hh:mm)

Afternoon start time

(hh:mm)

Evening start time

(hh:mm)

Playback position

SIP (Aura): Standard agent Details page includes admin details for Agent Greeting:

Contact Center Management

Agent Details: **Vincent Browne**

▼ User Details

First Name: **Vincent**

Last Name: **Browne**

Title:

Department:

Language:

Comment:

User Type:

Login ID:

Voice URI:

IM URI:

Account Type:

☒ Create CCT Agent

CCT Agent Login Details

Domain

accsl

User ID:

vincent

▶ [Associate User Account](#)

▶ [Agent Information](#)

▼ [Agent Greeting](#)

Agent Greeting Enabled ☒

Agent Greeting Password

Agent Greeting Recorded ☒

▶ [Offsite Agent](#)

▶ [Contact Types](#)

← Agent profiles can be enabled via Contact Center Management

10) Custom Applications & Interfaces

AACC provides a number of open interfaces that third-party developers can use to build applications that work with Contact Center Manager Server. In this sample deployment, custom wallboards and HDX integration are included. These are supported for both CS1000 and SIP (Aura) based applications.

- Real-Time Statistics Multicast (RSM) and Real-Time Interface (RTI) provide real-time information to applications such as wall boards.
- The Host Data Exchange (HDX) provides an interface for applications to communicate with the call processing script and workflow. This interface ensures the workflow can access information in an external database. With the Open Database Connectivity (ODBC) interface, an application can extract information from the Contact Center Manager Server database.

Please refer to the Dev Connect website for an overview of features provided and differences between releases of AACC:

https://www.devconnectprogram.com/site/global/products_resources/avaya_aura_contact_center/interfaces/ccms_programming_apis/releases/index.gsp

11) AACC Reporting:

One of the significant changes in AACC 7.0 (for both AML and SIP based solutions) is the replacement of SAP Crystal Reports with Microsoft SQL Server Reporting Services (SSRS) as the historical reporting runtime engine within AACC. Although this change and its implications are not unique to AML to SIP migrations, it represents an important consideration in terms of planning a migration. For more information refer to the following section of this document [Performance Management & Historical Reporting](#)

Standard Reports:

There are a number of minor changes in terms of AACC Historical Reporting between CS1000 based systems and SIP Aura. In terms of standard system provided reports, the following table summarizes the key differences:

Report Group	Report Name	CS1000 (AML)	Aura (SIP)
Configuration	IVR Queue and Port Properties	Yes	No
Configuration	Telephone Display Properties	Yes	No
Others	IVR Port First Login / Last Logout	Yes	No
Others	IVR Port Statistics	Yes	No

User Defined Reports:

User Defined Reports (i.e. Standard reports saved with user specific criteria) will continue to operate on the newly deployed SIP server. Note that some specified criteria may not be available on a SIP server.

Report Creation Wizard Reports:

When migrating from any version of NES or AACC prior to AACC 7.0 (AML), reports that have been created on the source AML server using the Report Creation Wizard can be migrated using a set of RCW migration tools. These tools are discussed in the following section of this document: [Summary of Migration steps](#)

User Created Reports – created outside of Report Creation Wizard:

When migrating from any version of NES or AACC prior to AACC 7.0 (AML), custom reports built outside of RCW (e.g. using Crystal Designer) cannot be migrated and must be re-created using either RCW or one of the third party report builder applications.

12) AACC & Avaya Call Recording:

Both CS1000 and SIP solutions are supported with Avaya Call Recording – some differences exist in terms of how contacts are recorded, for more details refer to the [Work Force Optimization & AACC](#) section of this document.

13) Licensing:

In this example, the newly deployed SIP based AACC solution will require new license files. This will be based on the Web LM license model.

6. Migration Considerations – Detailed Analysis

5.1 Upfront Assumptions & Prerequisites

This analysis is based on the Avaya Aura Contact Center 7.0 release however much of the information discussed here will be applicable to previous releases.

IMPORTANT: The following assumptions & restrictions apply:

- A phased migration is assumed. This means that new hardware is required for the newly deployed SIP based solution.
- All hardware, software and network requirements already specified in the AACC customer documentation must be followed for the newly deployed SIP based AACC.
- There are key configuration and provisioning differences when working with a SIP based AACC versus those on AML. Sufficient time must be provided to allow for this re-configuration before moving any live agents to the new system. Example: Application flows & scripts will require analysis and modification.
- Plan for and review all interactions with other applications (either Avaya or third party). In many cases these cannot be supported on SIP based deployments and alternatives will be required.
- This document is intended for use with Avaya Aura Contact Center only. Migrations to Avaya Contact Center Select (ACCS) are outside the scope of this document

5.2 Unified Communications Overview – Key considerations for AACC Integration

This section highlights important considerations when integrating AACC with Aura CM vs. CS1000. One of the key differences when moving to an Aura/CM deployment is that the Contact Center uses industry-standard SIP and CSTA (TR/87 over SIP) interfaces to communicate with the Unified Communications platform. Avaya Agent Desktop supports Avaya Aura® Unified Communications phones and continues to support voice, email, and Web chat contact types. AACC (including Avaya Media Server) requires the following UC components to allow this integration:

- Communication Manager
- System Manager
- Session Manager
- Application Enablement Services

Refer to the *Avaya Aura® Contact Center and Avaya Aura Unified Communications Platform Integration* guide for complete details on UC requirements with AACC.

Key UC considerations are listed here:

Feature / Characteristic	AML/CS1000	SIP/Aura CM
UC Environment:		
Avaya Communication Server 1000	CS1000 Release 7.6	N/A
Avaya Aura	N/A	Communication Manager, System Manager, Session Manager and Application Enablement Services required. Consult the Contact Center Overview and Specification guide for full details.
Third Line Appearance – Minimum support version	N/A	CM 6.3 FP3
Number of AACC servers per shared UC platform	3	3
AACC sharing of UC platform resources with other applications	Supported	AACC can share a CM with Elite. Not possible to share CM with CCE, IC and EMC.
Telephone Support	Refer to Review Telephone Support section of this document	
Communication Protocol	Proprietary AML	SIP with TR/87
Media Path/anchoring	Media maintained within CS 1000 core	Media anchoring is on Avaya Media Server
Dial Plan and Routing	Defined on CS1000 core	Dial Plan must be defined for Communication Manager. Routing must be defined on Session Manager via SMGR.
Dialed Number Identification Services (DNIS)	Provided by CS1000 core	Provided on Session Manager via number translations. Original customer DNIS number is retained in <i>SIP TO</i> header. See AACC UC Integration guide (referenced below) for more detail.
AACC is also supported with cost effective Midsize Enterprise or VE options. These are fully detailed in the Contact Center Overview and Specification guide.		

Feature / Characteristic	AML/CS1000	SIP/Aura CM
Aura Presence Server, Microsoft Lync Server 1013	N/A	Supported

One of the primary provisioning differences is the use of SIP URI's (Universal Resource Identifier) for each contact center agent and any CDNs, DNIS or Landing Pad resources. This must be carefully planned for so that this information is available during the migration and bulk update phases of the SIP deployment. Details below:

Resource Name	AML/CS1000	SIP/Aura CM
Routing Resources:		
Route Point (CDN) – Number	Number must match that configured on CS1000 (max 7 digits)	Number assigned to Route Point (Max 15 digits)

Resource Name	AML/CS1000	SIP/Aura CM
Route Point (CDN) – Universal Resource Identifier (URI)	N/A	The Universal Resource Identifier (URI) of the Route Point in Session Manager Routing. (Max 255 characters)
DNIS – Number	DNIS number must match that configured on CS1000 (max 7 digits)	Number assigned to the DNIS (3 to 15 digits)
DNIS – URI Universal Resource Identifier (URI)	N/A	The Universal Resource Identifier (URI) of the DNIS. (Max 255 characters)
Agent/Supervisor Resources:		
Voice URI	N/A	The SIP address of the TR87 controlled terminal dedicated to this agent, in the format sip:agent. For an Avaya Aura® PBX, use: sip:extension@sipdomain
IM URI	N/A	The agent's SIP address as configured on the Presence Server. For example, sip:12345@sipccocs.com
Login ID	Required	Required (when Logging into AAAD)
Personal DN	Optional	N/A

Unified Communications Reference Material:

For more detailed information, please refer to the following customer documentation:

- Avaya Aura® Contact Center Overview and Specification
- Avaya Aura® Contact Center Configuration Sections – Avaya Aura® Contact Center and Avaya Aura® Unified Communications Platform Integration
- DNIS support using Session Manager Configuration – Avaya Aura® Contact Center and Avaya Aura® Unified Communications Platform Integration

5.3 Review AACC Capacity & Environment considerations

Questions answered:

Question	Answer	Comments
Can I run my new AACC server on Aura/CM at the same capacity and call rates as my AML system?	Dependent on configuration	In most cases, Yes. There are variations in supported upper limits which must be considered for larger capacity systems. Review the table below for more information:

The following table provides a detailed matrix of supported capacity limits which should be considered when moving from an AML based AACC to a new deployment on Aura CM.

Important Capacity considerations

Capacity Item	AML/CS1000	SIP/Aura CM
Agents per AACC node	Inbound voice 5000 active (concurrent) and 10,000 configured (administered) multimedia 3000	Inbound voice / Multimedia: 3000 for Aura 6.2/7.0 (Max 10,000 configured)
Agents per network of nodes	150,000 (30 nodes x 5,000)	90,000 (30 nodes x 3,000 agents)
Supervisors per server	600 active & configured	600 active & configured (Licensed Resource)
Skillsets	1500	1500
Network skillsets	1500	1500
Skillsets per agent	150	150
Calls per Hour (cph)	100,000	45,000
Multimedia contacts per hour (cph)	12,000	12,000
Number of applications (that is, exit points from the Master_Script) The product contains five predefined applications. Therefore, you can create 800 applications in SIP contact centers.	1005 (1000)	805 (800)
Co-resident Multimedia server Inbound Multimedia contacts per hour	2400	2400 ¹
Max number of Route Points (CDNs)	2000 ²	5000

¹Assumes the AACC server is NOT co-res with Avaya Media Server

²Increased from 1000 to 2000 since AACC 6.4.2

Note 1: The capacity/resource profile of the source AML data must be within the range of what can be supported on the target SIP solution. Example: If you have 950 applications configured in AML, reduce this to 800 before the migration.

Note 2: Above limits are based on a full Aura UC stack. Other options are supported – MES and VE solutions but at reduced agent counts.

Environment considerations

Item	AML/CS1000	SIP/Aura CM
AACC servers – Workgroup & Domain support?	Workgroup OR Domain	SIP based AACC is supported in a domain only
Agent Workstation – Workstation & Domain support?	Phone-only mode is available. Agent workstation is optional	Agent Desktop required. Agent Workstations must be part of a domain. Either AACC server domain OR domain in the same forest. Full (two-way) trust enabled ¹

Reference Material:

For more detailed information, please refer to the following customer documentation:

- Avaya Aura® Contact Center Overview and Specification Release 7.0. *Refer to Part 4: Performance specifications.*

5.4 Review Telephone Support

AACC deployed in a SIP based environment has inherent support for an embedded softphone with Agent Desktop. Where a hard-phone is required, the following information highlights compatible phones that may be used.

Questions answered:

Question	Answer	Comments
Can I reuse my CS1000 agent sets for my SIP AACC agents?	No	Refer to table below for supported sets. <i>Note: Specific set types are required to support third line appearance feature. Third Line feature requires CM 6.3 FP3</i>
Can I reuse my CS1000 agent sets on my CM as knowledge worker sets (non AACC agents)?	Check set type	Some sets can be used for this purpose. The following models can be used: 1120E, 1140E, 1165E, 1220 and 1230
My AML solution is deployed as phone only – no Agent Desktop. Can I continue to use this mode on SIP CM?	Agent Browser Application OR Avaya Agent Desktop are needed	1. Agent Browser Application: (browser based) provides the minimum SIP Contact Center functionality required for voice-only agents to handle calls without needing to deploy the full Avaya Agent Desktop. Agents can handle calls with just this application and their desktop phone. 2. Avaya Agent Desktop: Required to use a workstation with Avaya Agent Desktop for all contact center activities. Recommended that the agent uses AAAD for all telephony functions as well. For features, refer to section Agent Desktop and Common Agent Features

The following table provides a detailed matrix of supported telephone set options on CS1000 and Aura/CM when used as AACC agent sets. Within a SIP environment, an AAAD integrated softphone is also supported.

Supported AACC agent telephones on Aura CM

Telephone Model/Type	AML/CS1000	SIP/Aura CM	Comments
H.323 Telephones:			
Avaya 1600 Series IP deskphones	No	Partial	No support for Third line Appearance
Avaya 4600 Series IP deskphones	No	Yes	
Avaya 9600 Series IP deskphones	No	Yes	
Avaya 96x1 Series IP deskphones	No	Yes	

Telephone Model/Type	AML/CS1000	SIP/Aura CM	Comments
Avaya Aura® Agent Desktop embedded softphone	No	Yes	Ensure you provision an IP_Agent license on the Communication Manager for each softphone used by the contact center. The Communication Manager station associated with the Avaya Aura® Agent Desktop embedded softphone must not be configured as an Avaya 96x1 Series deskphone. Configure the Communication Manager station associated with the embedded softphone as one of the other supported H.323 phone types.
Avaya one-X® Communicator Release 5.2 or later	No	Yes	Avaya one-X® Communicator not supported running concurrently with AAAD
Avaya one-X® Communicator Release 6.x	No	Yes	Avaya one-X® Communicator not supported running concurrently with AAAD
Digital Telephones:			
Avaya 24xx Series deskphones	No	Partial	No support for Third line Appearance
Avaya 64xx Series deskphones	No	Partial	No support for Third line Appearance
SIP Telephones:			
Avaya 96x0 Series IP deskphones	No	Yes	
Avaya 96x1 Series IP deskphones	No	Yes	
Avaya 9608 IP deskphone	No	Yes	
Avaya 9611G IP deskphone	No	Yes	
Avaya 9621G IP deskphone	No	Yes	
Avaya 9641G IP deskphone	No	Yes	

Supported AACC agent telephones for Communication Server 1000

Telephone Model/Type	AML/CS1000	SIP/Aura CM	Comments
Avaya 39xx Digital Deskphone (CS1000):			
Avaya 3904 Digital Deskphone	Yes	No	
Avaya 3905 Digital Deskphone	Yes	No	
IP phones and softphones (CS1000):			
Avaya 1110 IP Deskphone	Yes	No	
Avaya 1120E IP Deskphone	Yes	No	Can be reused as knowledge worker
Avaya 1140E IP Deskphone	Yes	No	Can be reused as knowledge worker
Avaya 1150E IP Deskphone	Yes	No	
Avaya 1165E IP Deskphone	Yes	No	Can be reused as knowledge worker
Avaya 1200 Series IP Deskphone	Yes	No	1220 and 1230 types can be reused as knowledge worker

Telephone Model/Type	AML/CS1000	SIP/Aura CM	Comments
Avaya 2002 IP Deskphone	Yes	No	
Avaya 2004 IP Deskphone	Yes	No	
Avaya 2007 IP Deskphone	Yes	No	
Avaya 2050 IP Softphone	Yes	No	
<i>Support of specific types of phones can change with each software release of the call server (Avaya Communication Server 1000).</i> <i>Consult the Avaya Communication Server 1000 documents for an up-to-date list of supported phone types for the software release in use.</i> <i>When moving from a digital phone type (e.g. 3904) to an IP based phone type, the appropriate IP network infrastructure must be in place to allow these sets to be deployed.</i>			

5.5 Review Voice Services & Self Service options

Questions answered:

Question	Answer	Comments
I am currently using CallPilot to provide Voice Services for AACC. This is unsupported on Aura/CM, are there other options available?	Yes	Refer to table below for details of how similar features and integrations can be provided in a SIP based solution, e.g. Avaya Media Server
Can I reuse my AACC application flows and scripts when I migrate to a SIP based contact center?	Yes – Changes Required	AML based scripts and flows are stored in the AACC database and are migrated as part of the data migration procedures. A new Workforce Validation Tool can be run against the migrated flows to analyze flows and highlight unsupported commands, blocks and intrinsics
Can I reuse or move my audio prompts from my AML CallPilot to my new Avaya Media Server?	Re-Record or Conversion may be required	<i>For Avaya Media Server, Optimum Playback performance Recoding Media File Format: WAV file encoded as Linear 16-bit PCM, 8 KHz Mono with a sampling rate of 128 kbits/sec</i>
Can I reuse my agent greeting prompts when I move to agent Greeting based on Avaya Media Server?	No	Agent greetings must be recorded as part of the new AACC provisioning.

CallPilot Integration

CallPilot is typically deployed with AACC on CS1000 to provide voice services and is not applicable for use in a SIP based deployment. The following table summarizes how equivalent functionality can be provided using Avaya Media Server or Experience Portal applications.

Note: Any AACC application flows or scripts must be re-written to support SIP specific syntax.

CallPilot Feature	AML CS1000	SIP Aura/CM	Comments
Contact Center Voice Services on Avaya CallPilot®	Supported	Voice Services are provided via Avaya Media Server	<i>SIP based contact centers can provide Voice services using the Avaya Media Server</i>
Prompt recording	Use phone with CallPilot to modify prompts	Use a recording application to modify prompts	<i>For Avaya Media Server, Optimum Playback performance Recoding Media File Format: WAV file encoded as Linear 16-bit PCM, 8 KHz Mono with a sampling rate of 128 kbits/sec.</i>

CallPilot Feature	AML CS1000	SIP Aura/CM	Comments
IVR Integration (GIVE IVR script/application command)	Yes – Integration with AACC scripts. GIVE IVR is used when not using Voice Sessions	Use GIVE IVR with VXML – provides treatments to callers by playing recorded announcements or by collecting caller information from VXML scripts residing on the media server.	For SIP enabled contact centers, different syntax, parameters are required. Different behaviors and enhanced functionality is available. Refer to system provided 'Simple Greeting' application template in Orchestration Designer for examples of how this can be implemented
Broadcast announcement capability	Yes	No Broadcast capability – Larger port capacity on Avaya Media Server to accommodate this	CallPilot supports one to many (up to a maximum of 50), while Avaya Media Server requires a one to one mapping of ports.
Broadcast: Max number of simultaneous calls per port	50	N/A	Not applicable for SIP contact centers. Avaya Media Server is provisioned with suitable port capacity
Digit Collection	Yes – using the COLLECT DIGITS script/application command	Provided by Avaya Media Server – Using enhanced GIVE IVR for SIP enabled contact centers	Sample 'digit collection' flow/template in AACC. Full details of these commands are available from the 'Using Contact Center Orchestration Designer Release 7.0 guide'
Digit Collection: Maximum number of digits that can be collected:	16	80	
Support for Voice Sessions	Yes	Provided by Avaya Media Server – Using enhanced GIVE IVR for SIP enabled contact centers	SIP Contact Centers do not use Voice Sessions. Refer to the Digit Collection and MultiLevelMenu sample applications in Orchestration Designer. Refer to: Using Contact Center Orchestration Designer Release 7.0 guide This provides examples of how to build flows using the 'GIVE IVR' command.
Play Prompt (within Voice sessions)	Yes	N/A	The alternative in SIP based contact centers is the enhanced GIVE IVR command. See the Simple Greeting application template for more information.

CallPilot Feature	AML CS1000	SIP Aura/CM	Comments
Front-End IVR Applications (CallPilot)	Supported – Calls terminate directly on an Avaya CallPilot® voice menu and are not controlled by Contact Center Manager Server until transferred the call to a CDN	No Front-End Support with Avaya Media Server. Functionality can be provided by AAEP or AAM	<i>NOTE: There is no call context in AML/CS1K deployments, i.e. Call Steering can be implemented but no routing decisions can be made within AACC. Recommendation is that applications are re-deployed on Avaya Aura Experience Portal.</i>
Menu Applications (CallPilot)	Yes	Provided by Avaya Media Server – Using enhanced GIVE IVR for SIP enabled contact centers	<i>Sample menu flow Multi Level Menu provided in AACC. Using Avaya Media Server means that menu applications are controlled by AACC.</i>
Integrated Prompt Management within AACC	N/A	Yes – integrated prompt management for Avaya Media Server	<i>Integrated prompt Management delivered as part of AACC 7. This allows management of recorded prompts and music.</i>
Integrated Partitioning of prompts for AACC reporting	N/A	Yes – AACC partitioning of prompts for Avaya Media Server	<i>CCMA integration: delivered as part of AACC 7 release</i>
Expected Wait time and position in Queue	Yes	Yes – using Avaya MS & UC	<i>SIP Based Contact Centers use Avaya Media server and Aura UC.</i>
<i>AACC application flows/scripts must be re-written to provide SIP specific commands</i>			

Integrated Recorder Announcer Integration

Integrated Recorder Announcer (formally MIRAN) is not supported in a SIP based contact center. Similar functionality can be provided using the Avaya Media Server included as part of the AACC solution. Application flows/scripts will need to be re-written.

Integrated Recorder Feature	AML CS1000	SIP Aura/CM	Comments
RAN and MUSIC functions	Supported	Provided by Avaya Media Server or AAEP	<i>Also possible to use GIVE IVR with VXML – provides treatments to callers by playing recorded announcements or by collecting caller information from VXML scripts residing on the media server. Sample flows are provided within AACC – Orchestration Designer to</i>

Integrated Recorder Feature	AML CS1000	SIP Aura/CM	Comments
Telephone User Interface (TUI) Ability to use telephony interface to record and play prompts etc.	Supported	No	Use a recording application to modify them: Optimum Playback performance Recording Media File Format: WAV file encoded as Linear 16-bit PCM, 8 KHz Mono with a sampling rate of 128 kbits/sec.
Integrated Prompt Management within AACC	N/A	Yes – integrated prompt management for Avaya Media Server	Integrated prompt Management as part of AACC 7.0
Integrated Partitioning of prompts for AACC reporting	N/A	Yes – CCMA partitioning of prompts for Avaya Media Server	AACC Integration: Part of AACC 7.0
AACC application flows/scripts must be re-written to provided SIP specific commands			

Avaya Aura Experience Portal Integration

Avaya Experience Portal provides self-service / IVR Integration including front-end, mid call & call attached data. In an Avaya Communication Server 1000 AML-based solution, Avaya Aura® Contact Center supports Landing Pads for integration with Avaya Aura® Experience Portal. SIP based solutions additionally support SIP header Information or Contact Intrinsic as call attached data.

Experience Portal Feature	AML CS1000	SIP Aura/CM	Comments
Landing Pad Transfer	Yes	Yes – possible but not recommended. Use P-Intrinsic SIP header instead	For AML CS1000, front-end integration is the only supported option
Data Directed routing using Landing Pads	No	Recommend using P-Intrinsic SIP header	For AML/CS1000 deployments, Landing Pads can only be used for front-end scenarios with a one-time drop of data. No routing decisions can be made by attached data. Note: The CCAD_IVR intrinsic has the data and can be viewed in AAAD when selecting 'Raw Intrinsic'

Experience Portal Feature	AML CS1000	SIP Aura/CM	Comments
UUI SIP Header Transfer	No	Yes	SIP-enabled systems can use User-to-User Information (UUI) to transmit small amounts of data between systems within SIP header messages. Voice XML applications can use SIP header information to collect, store, and transport customer call-related information. Voice XML application can use customer interview data to modify the SIP header, and then pass the customer call along with updated header data to the next application in the solution. Voice XML applications can also use SIP header information to make processing decisions about a customer call. Examples of SIP header UUI data include a customer account number obtained during a self-service customer interview. Avaya Aura® Agent Desktop and Avaya Aura® Contact Center Orchestration Designer can also modify User-to-User Information. This SIP header UUI data can be used to support Avaya Aura® Application Sequencing.
P-Intrinsic SIP header	No	Yes	Avaya Aura® Contact Center supports the custom P-Intrinsics private header. In a contact center solution, you can use SIP headers to transfer small amounts of call-related information between SIP-enabled applications. The application receiving this SIP message reads these headers and performs some action based on the contents of the headers. SIP header information can provide additional data about a call that applications can use to process that call.
SIP INFO message body using Context Creation	No	Yes	Suitable for mid call IVR sessions
Any other SIP based/3 rd party application support	No	No	Not Supported for CS1000 OR SIP based deployments

Media Processing Server (MPS) integration

Media Processing Server is a self-service solution that integrates with AACC/CS1000 solutions. There is no MPS integration support SIP/Aura based solutions of AACC and alternative solutions based on Avaya Experience Portal or Avaya Media Server are recommended.

MPS Feature (MPS 500)	AML CS1000	SIP Aura/CM	Comments
Front End Call Steering	Yes	Avaya Experience Portal Recommended	<i>For SIP based solutions, Front-End Applications can be developed (Via Orchestration Designer) in AAEP. These applications can be used to 'steer' contacts to appropriate AACC route-points, application flows and skillsets.</i>
Front End with attached call data	Yes	Avaya Experience Portal Recommended	<i>Routing decisions within AACC (Flows & Applications) possible based on call data</i>
Backend integration – integration with AACC flows and scripting	Yes	Avaya Experience Portal Recommended	<i>Back-End integration can be achieved using AACC application flows. Use enhanced GIVE IVR commands.</i>
AACC Real-Time statistics (RTS) integration	Yes	No	<i>No equivalent RTS connector available in AAEP</i>
CTI Connector	Yes	No	<i>No equivalent RTS connector available in AAEP</i>
CM & Meridian Link Services (MLS)	Yes	No	<i>No MLS support in SIP Aura deployments</i>
<i>AACC application flows/scripts (if applicable) must be re-written to integrate with AAEP</i>			

Reporting on MPS resources

In some CS1000 based deployments, richer AACC reporting has been achieved by configuring MPS ports as AACC agents and as agent sets on the CS1000. This allowed AACC supervisors and administrators to make use of agent specific reporting and statistics when running reports on port usage – e.g. Agent performance reports.

This configuration also facilitated back-end integration of the Contact Center with MPS by using the *QUEUE* block in application flows or *QUEUE TO SKILLSET* script command, i.e. contacts could be queued to a skillset consisting of only MPS ports (configured as agents and logged into AACC).

IMPORTANT: This configuration is not supported in a SIP based contact center as IVR ports are not configured or reported on within AACC. When planning a migration from an AML/CS1000 based solution to SIP/Aura, it is important to consider in advance how MPS integrations can be replaced with either Avaya Experience Portal or Avaya Media Server. New reporting must also be changed and planned for to make use of the reporting features provided by AAEP or Avaya Media Server.

Agent Greeting

Agent Greeting is provided on both AML CS1000 and SIP Aura systems. It is important to note that this functionality is provided using two different applications and configurations. AML contact centers makes use of an integrated Agent Greeting card as part of the CS1000, SIP based systems provide Agent Greeting via the Avaya Aura Contact Center.

Important: Agent Greeting for CS1000 is now End of Sale and no longer an available option.

Agent Greeting Feature	AML CS1000 Avaya Agent Greeting	SIP Aura CM Avaya Media Server	Comments
Agent Greeting	Provided by Agent Greeting for CS1000	Provided via Avaya Media Services (Licensed Feature)	<i>For SIP, this is a licensed feature and available via the already integrated Avaya Media Server.</i>
Integrated admin with AACC	No	Yes	<i>In SIP/Aura solutions, AACC integration is provided by CCMA (Configuration & Contact Center Management) & Agent Desktop</i>
Mute agent on greeting playback	via CS1000 'Mute On agent answer' class of service	Yes	<i>Option to automatically mute the agent while the greeting plays. Configured via AACC Administration</i>
Greeting warning tone (optional)	Yes	Yes	<i>Option to ensure a warning tone is played, audible to the agent and the caller, if an agent answers a call for which no greeting is available. Configured via CCMA.</i>
Allow an agent to interrupt the greeting	Yes	Yes	<i>Option to allow an agent to terminate the greeting to speak directly to the caller. For more information on interrupting the greeting, see Using Agent Desktop for AACC Release 7.0 .</i>
Supported greeting length	Dependent on PC card flash memory limits	1 to 30 seconds	
Prompt language options	Yes	Yes	<i>Allows an administrator to choose a language for the application prompts. Agents hear the selected language when they dial into the recording application. Configurable via CCMA: Configuration.</i>
Time of Day settings support	Yes – via Agent Greeting 3.01. Support for morning, afternoon and evening	SIP (Aura) deployments support morning, afternoon and evening start times	<i>In SIP/Aura solutions, Time of Day settings are provided via Avaya Media Server integration and configurable via CCMA.</i>
Time of day settings: Playback position	Before main Greeting OR After Main greeting	Before main Greeting OR After Main greeting	

IP Media Services

IP Media Services (based on Avaya Media Server) is available on CS1000 to provide voice services but is limited to GIVE RAN and GIVE MUSIC only. As with previously discussed options, the recommendation is to use the integrated Avaya Media Services as part of SIP AACC to replace this component.

Callback Assist

New in release 7.0, AACC now interoperates with Avaya Callback Assist (CBA) and provides an enhanced call back functionality for Contact Center applications. CBA is an application hosted on the Avaya Aura Experience portal and provides the following:

- Front End Avaya Experience Portal + Callback Assist: In this case the estimated wait time is checked (prior to arriving at AACC) and if the wait time is above a threshold, the customer is played a prompt offering them the choice of waiting or leaving details for a call back.
- Back End Integration: In this case the call arrives into AACC and the contact center flow queries the expected wait time and offers the customer a choice of remaining in a queue or requesting a call back. If a callback is requested by the customer, the call is routed to the CBA application.

Voice Services Reference Material:

For more detailed information, please refer to the following customer documentation:

- Refer to the [AACC Scripting & Application Flows Considerations](#) section of this document for more information on how to migrate your existing application flows and scripts.
- Avaya Aura® Contact Center Client Administration, *refer to Agent Greeting & Prompt Management chapters*.
- Using Contact Center Orchestration Designer, *refer to Application migration section*.

5.6 Agent Desktop and Common Agent Features

The use of Avaya Agent Desktop is one of the key changes when moving from an AML to a SIP based contact center, if you are moving from a phone only solution on AML. A summary of important requirements is listed below along with a number of tables with a detailed analysis of important agent functions.

Summary:

- For SIP based AACC deployments, agents typically require Agent Desktop to login to the contact center. This allows agents to manage all contact types from one unified application. Since AACC 7.0, agents can alternatively use the new Agent Browser Application along with their telephone, to login and handle voice contacts. **There is no support for phone-only mode agents in AACC in a SIP environment.**
- All agents running Agent Desktop will require access to a Windows workstation (Windows 7 or 8.1 supported) on which they can install and run the desktop application.
- Agent workstations must be added to a domain. This must be either the same domain as the AACC Contact Center OR in the same forest with a full (two-way) trust with the AACC domain.
- Agent Desktop deployments must adhere to specific latency requirements. These are referenced at the end of this chapter.

Questions answered

Question	Answer	Comments
My AML solution is deployed in phone-only mode. Can I continue to use this mode in AACC on Aura/CM?	Agent Browser Application OR Avaya Agent Desktop are needed	1. Agent Browser Application provides the minimum SIP Contact Center functionality required for voice-only agents to handle calls without needing to deploy the full Avaya Agent Desktop. Agents can handle calls with just this application and their desktop phone. 2. Avaya Agent Desktop: Required to use a workstation with Avaya Agent Desktop for all contact center activities. Recommended that the agent uses AAAD for all telephony functions as well. For features, refer to section Agent Desktop and Common Agent Features
I am already using Agent Desktop in my AML deployment. What steps do I need to perform?	Uninstall + Re-install of SIP AAAD version	The AML version must be uninstalled and a new version installed from the SIP Aura/CM deployment

Agent Browser Application

The Agent Browser Application is a new agent interface in AACC 7.0 which is based on cross-platform web technologies. It is intended for use by agents not requiring the Avaya Agent Desktop, e.g., voice only agents moving from AML to SIP Contact Center and had not previously used Agent Desktop that do not require multimedia contacts or screen pop features.

- It provides the minimum SIP Contact Center functionality required for voice-only agents to handle calls without needing to deploy the full Agent Desktop.
 - Agents can handle calls with just this application and their desktop phone.
- Its design means it can run in any modern browser meeting certain minimum requirements.
- Provides a mobile-friendly interface that responds to different screen sizes and orientations by automatically changing the layout.
- The Browser-based, Agent Browser Application is a single-frame web page hosted within IIS8 on the ACC server.
- It connects via WebSockets to the new Integration Portal on the ACC server to receive real-time updates and to facilitate the invocation of agent operations such as login, logout, ready, not ready, etc.

Sample screen: Agent is logged in and on a voice call:

The screenshot displays the Agent Browser Application interface. At the top, there is a navigation bar with 'Guide' and 'Settings' links, and a user profile section for 'Agent One'. Below this, a status bar shows 'Agent One' with a green checkmark and a 'Ready' button. The main content area is divided into two sections. The first section displays agent information: 'Agent ID' (8111671), 'Phone Number' (8111671), and 'Supervisor' (Supervisor One). Below this, it shows the 'Last successful connection' as '8/19/2015 4:04:13 PM'. The second section contains a row of buttons: 'Ready', 'Not Ready', a dropdown arrow, 'Call Supervisor', and 'Log out'. Below this, a 'Contacts' section shows 'Contact Center Call' with details: 'Skillset' (Skill1), 'Caller' (sip:8111673@sipccgal.com), and 'Activity' (SUV Sales). At the bottom, there is a row of buttons: 'Activity', 'After Call Work', and 'Emergency'. The footer of the interface states '© 2015 Avaya Inc. All rights reserved.'

Supported Operations (Agent Browser Application):

- Logging on and off
- Changing the agent status
- Setting not ready reason codes
- Setting activity codes
- Calling your supervisor

- Handling an emergency
- Setting after call work codes

NOTE: The Contact Center Agent Browser application does not provide call control or multimedia features. Agents must use a desktop phone for call control. Please refer to the ***Using the Contact Center Agent Browser application*** document for full details

Browser & OS support (Agent Browser Application):

Browser	Supported Versions	Operating System
Internet Explorer	11.0	Windows 7 Windows 8.1 Windows 10
Google Chrome	43.0.23	Windows 7 Windows 8.1 Windows 10 Android 4.4.2 Android 5.0.1 iOS 8.3
Mozilla Firefox	38.0.5	Windows 7 Windows 8.1 Windows 10
Safari	8.0.5	iOS 8.3 OS X Yosemite 10.10.3

Avaya Agent Desktop Login Options

Agent Login Options	AML CS1000	SIP Aura/CM	Comments
AAD application support	Yes	Yes	<i>Same codebase between AML/SIP. Different configuration files are deployed – reinstall of AAAD required when moving from AML based deployment to SIP/Aura</i>
Voice login via phoneset (Phone only)	Yes	No	<i>In a SIP contact center – Agents login using AAAD.</i>
Voice login via phoneset at any ACD phone	Yes	No	<i>Refer to Agent Desktop and Common Agent Features section for more information on hot-desking.</i>
Voice login via AAD	Yes	Yes	<i>Mandatory for SIP/Aura AACC deployments (unless using Agent Browser application)</i>
Multimedia login via AAD	Yes	Yes	
Length of Agent ID	4 to 16 digits	4 to 16 digits	
Validation of agent login	AACC validation	AACC validation	
Avaya Aura® Agent Desktop supports three voice modes:			
Desk Phone	Yes	Yes	

Agent Login Options	AML CS1000	SIP Aura/CM	Comments
My Computer (softphone)	N/A	Yes	
Other Phone (Telecommuter mode)	N/A	Yes	<i>For AML equivalent, see CS1000 Offsite Agent – detailed later in this chapter</i>

Avaya Agent Desktop compatibility with other applications

The following applications cannot be run concurrently with Avaya Agent Desktop:

- Avaya one-X® Communicator
- Avaya one-X® one X Agent
- Microsoft Office Communicator client
- Any non-Avaya softphone

Commonly used agent functions

Agent Function	AML CS1000	SIP Aura/CM	Comments
Max call appearance lines per agent station/phone	2 (typical)	3 (Last line restricted) Requires CM 6.3 FP3	<i>For AML/CS1000 deployments, agents are configured with an 'InCalls' key for contact center calls and a personal Directory Number (DN). In the case of SIP/Aura, each agent is configured with up-to 3 line appearances, the last of which is restricted and reserved for outbound calls or conferencing.</i>
Ability to receive a 2 nd voice call whilst on an AACC voice contact	Yes	Yes	<i>IMPORTANT: In SIP environments, an agent can receive a second call (in addition to an AACC voice contact) but the second call must NOT be another AACC voice contact. i.e. a personal call only. Only one AACC voice contact may be handled at a time.</i>
Ability to make a 2 nd voice call whilst on an AACC voice contact	Yes	Yes	
Walkaway trigger	Headset removal	No	<i>Plugging out the headset will be triggered to CS1K/AACC and flagged in RTD's. No walkaway in SIP deployments.</i>
Activity Codes: Entry and reporting	Enter using AAAD or Phoneset	Enter using AAAD only	<i>In SIP/Aura deployments, Activity codes must be entered via AAAD.</i>

Agent Function	AML CS1000	SIP Aura/CM	Comments
After Call work / Not Ready Codes: Entry and Reporting	Must use AAAD if entering code while working a contact	Enter using AAAD only	<i>In an AML environment, you can use a phone to enter a code. However, if you want to enter an ACW code while you are working on a contact, you must use Agent Desktop and not a phone. If you use a phone and enter a non-ACW code, the time is not reported as ACW time or Post Call Processing time. Also, entering an ACW code or Not Ready code, while active on a contact might not be reflected correctly in reporting.</i>
Activity Codes (Including Not Ready & After Call Work): Supported range of codes	1 to 32	1 to 32	
Agent phonebook with LDAP integration (Agent Desktop)	Yes	Yes	<i>No migration possible of agent contacts list. This is stored in the Multimedia database and is linked to the agent ID</i>
Agent Greeting	Provided by Avaya Agent Greeting 3.0x and later	Provided by Avaya Media Server (Licensed Feature)	<i>Integrated Agent greeting is provided by Avaya Media Server integration for SIP/Aura contact centers. Please refer to the Review Voice Services & Self Service options section for more information on this feature</i>
Whisper Skillset	No	Yes – provided by Whisper Skillset feature in AACC 7.0	<i>In a SIP-enabled Contact Center, Avaya Aura® Contact Center supports an optional configuration to provide Whisper Skillset to agents. The Whisper Skillset feature allows a brief skillset-specific announcement to be played to agents before they are connected to a customer call. For more information on the Whisper Skillset feature, see Avaya Aura® Contact Center Server Administration</i>
Coverage path to voice mail (Avaya Aura® Messaging, Modular Messaging, CM Messaging)	N/A Alternative Voicemail options available on CS1000	Yes	<i>CM: Coverage path for Avaya Communication Manager Messaging, Avaya Modular Messaging and Avaya Aura Messaging. In a SIP-enabled Contact Center using a Communication Manager, Avaya Aura® Contact Center supports a limited configuration of the Communication Manager – Coverage Path feature. For more information, see Avaya Aura® Contact Center Configuration – Avaya Aura® Unified Communications Platform Integration</i>
Call Force / Zip Tone	Yes	Yes – provided by Avaya Media Server	<i>SIP/Aura systems: Only applies to agents configured to use a 'Call Force' presentation class. If both Zip Tone and Whisper skillset features are applicable for a given call, Zip tone takes precedence.</i>

Agent Function	AML CS1000	SIP Aura/CM	Comments
Single step transfer	Yes	Can be partially accomplished by simply completing the transfer	<i>In a SIP-enabled Contact Center, blind (single-step) transfers are not supported. However, early transfer completion is supported. Early transfer completion is the ability to complete a supervised (two-step) transfer while the far end is still ringing. In a SIP enabled Contact Center, references to blind transfer mean supervised transfer with early transfer completion.</i>
Phone display support	Yes	No	<i>Skillset, CDN#, Route, DNIS, CLID, Route#, DNIS#, CLID#, Queue, CDN, Trunk Note: Communication Manager phonesets can be configured to display UUI data.</i>
Support for Call Park	Yes	No	
Completion of transfer while far end is ringing	If the far end address is out-of-provider (not monitored by CCT), the remote connection state transitions immediately from the Alerting state to Established.	Yes	
Emergency help / Emergency key	Yes	Yes – via Agent Desktop Emergency key	<i>Note for the Emergency help Real Time Display: In a SIP-enabled Contact Center, after a supervisor releases an Emergency call, the Emergency report displays the status of the Emergency call as “Closed”. In an AML-based Contact Center, after a supervisor releases an Emergency call, the Emergency report continues to display the status of the Emergency call as “Active” until the agent releases the call. Refer to AACC Client Administration.</i>
Message Waiting indicator	Yes	No	<i>Not available via AAAD. Can be configured on desk-phone.</i>
Display Waiting Calls (DWC) key	Yes	No	<i>Used for Contact Center Network ACD calls</i>

Agent Function	AML CS1000	SIP Aura/CM	Comments
<i>In SIP deployments, Agents must not use their desk phone or Avaya Agent Desktop to phone, transfer a call or conference a call to a phone number that is:</i> - <i>Routed to a Route Point Example: a Virtual Directory Number (VDN) routed to a Route Point</i> - <i>Converted to a Route Point</i> - <i>Call forwarded to a Route Point</i> <i>Agents can use their desk phone or AAAD to transfer, or conference or phone directly to a Route Point</i> <i>Please refer to the Call Redirection App Note (referenced at the end of this chapter) for more information on supported transfer & conferencing scenarios.</i>			

Hot Desking and Remote Agent options

Hot Desking and Remote Agent Options	AML CS1000	SIP Aura/CM	Comments
Hot Desking / Agent Roaming (CCT Hot Desking for AML based contact centers)	Yes	N/A (See SIP Free Seating below)	<i>In SIP deployments, agent enters their telephone credentials first. This allows roaming.</i>
Offsite Agent (CS1000 AML Contact centers only)	Yes	Telecommuter mode (No Offsite Agent server required for equivalent functionality)	<i>Requires an Offsite Agent server. Only available for CS1K AML deployments. Recommend using Telecommuter as an alternative</i>
AAAD Telecommuter Mode	No	Yes	<i>SIP only. Based on CM Telecommuter functionality provides simple architecture for remote agent working</i>
Ability to login to any phoneset as a specific agent ID to enable free seating / hot desking	Yes	Yes – Must logon to station	<i>Free-seating or “hot-desking”: Agent can login to any configured phone-set (supply login credentials and then login via AAAD for AACC)</i>

Screen Pops on Avaya Agent Desktop

Screen pop functionality can be configured using the Multimedia Administration tool – The new (target) SIP based AACC 7.0 install will include Multimedia components as standard. Agent Desktop displays screen pops when a contact alerts or is answered. Administrators can configure these screen pops to display relevant information about the alerting or answered contact based on the intrinsic, such as Skillset, To Address, and DNIS associated with the contact.

Screen pop features	AML CS1000	SIP Aura/CM	Comments
Basic Screen Pop	Yes	Yes	
Advanced Screen Pop	Yes	Yes	
System Intrinsic set on CDN calls	Yes	Yes	<i>Intrinsics for AML contact centers introduced in AACC 6.4. System intrinsics contain system-wide information about skillsets, time, traffic, and voice contacts. Avaya Aura® Contact Center (AACC) automatically creates and maintains system intrinsic. Intrinsics are available only to query data about the system within OD applications, not to modify data. Some examples of system intrinsics: Call intrinsic – CDN, Skillset intrinsic – POSITION IN QUEUE, Traffic intrinsic – TOTAL ACTIVE CALLS.</i>
System Intrinsic set on personal calls	No	Yes	

Screen pop features	AML CS1000	SIP Aura/CM	Comments
SIP Contact Intrinsic	N/A	Yes	<i>In a SIP-enabled Avaya Aura® Contact Center, Contact Intrinsic data makes it easy to develop screen pops, reducing the time, effort and cost required to launch new capabilities. In a SIP-enabled contact center solution, each contact has associated SIP Contact Intrinsic data. This Contact Intrinsic data may contain data relevant to that call, the calling customer and other information retrieved by self-service or third party applications.</i>
Intrinsics from Open Networking Web Service	No	Yes	<i>Only applicable to SIP based contact centers</i>

Call Presentation Choices

Some Call presentation classes are not available on SIP Aura CM based contact centers. The complete list is reviewed below. These are defined and available to configure in AACC Administration, Configuration component.

Call Presentation	AML CS1000	SIP Aura/CM	Comments
Call Force Delay	Yes	Yes	<i>In a SIP-enabled contact center, you can configure the Call Force Answer Zip Tone feature. If this is configured in a contact center where the Agent's phone ringer is disabled, in order to reduce customer wait time, Avaya recommends setting the Call Force Delay Timer to zero seconds.</i>
Return To Queue	Yes	Yes	<i>IMPORTANT: In SIP-enabled Contact Centers, Voice and IM contacts return to the queue after 30 seconds.</i>
Let Call Ring	Yes	No	<i>No support in SIP/Aura deployments for this option</i>
After Return to Queue, Make phoneset Logout	Yes	Yes	
After Return to Queue, Make phoneset Not Ready	Yes	Yes	
After Call break for n seconds	Yes	Yes	<i>Valid values: 0–300 (seconds)</i>
Answer by placing DN Call on Hold	Yes	No	<i>No support in SIP/Aura deployments for this option</i>
Agent Reserved for Network Call	Yes	Yes	<i>Select this check box to display the message Reserved on the agent's phone when the server reserves the agent for a network call.</i>

Caller Name information in Avaya Agent Desktop:

If Agent Desktop receives a call from an internal extension (e.g. knowledge working on the same Avaya Aura) which is not defined in AACC, the agent will see the extension number displayed in agent Desktop. In contrast, if a call received from another agent or

Supervisor/Agent, the agent name will be displayed in Agent Desktop. This is provided via the AACC friendly name feature.

Agent Desktop Reference Material:

For more detailed information, please refer to the following customer documentation:

- Avaya Aura® Contact Center Overview and Specification: Refer to the Agent Desktop client requirements chapter. This contains information on the following important consideration:
 - Localized languages
 - Client hardware requirements
 - Client operating system requirements
 - Network infrastructure requirements
 - VMware Horizon View VDI support
 - Client Citrix support
 - Agent Desktop tick client deployment
 - Third-party software requirements
- Using agent Desktop for AACC: This contains a complete list of feature options available in Agent Desktop.
- Call Redirection with Avaya Aura® Contact Center 6.4 Application Note – Available from <http://www.avaya.com/support>

5.7 AACC Licensing Considerations

Different license types, mechanisms and supported feature options apply for AACC contact centers deployed in an AML environment vs. SIP. For this reason, new license files are required when moving to a SIP contact center. Important differences are details below:

Licensing Types

Licensing Types	AML CS1000	SIP Aura/CM	Comments
Nodal Enterprise Licensing (WebLM & PLIC licensing mechanisms)	Yes	Yes (Web LM only for new installations)	<i>The Nodal Enterprise license type controls the licensing for a single Avaya Aura® Contact Center node. Contact Center License Manager uses Nodal Enterprise licensing to control a single installation of Contact Center Manager Server (CCMS), Contact Center Manager Administration (CCMA), Contact Center Multimedia (CCMM), and Communication Control Toolkit (CCT). The Nodal Enterprise license type is supported by the WebLM and PLIC licensing mechanisms. In SIP-enabled Avaya Aura® Contact Center solutions that use nodal WebLM licenses, Contact Center License Manager pushes licenses to the Avaya MS servers configured as Media Servers in Contact Center Manager Administration.</i>
Corporate Licensing (PLIC only)	Yes	No	<i>You can use Corporate Enterprise licensing to distribute licenses to multiple servers so they can share licenses from a single pool.</i>
Nodal NCC Licensing	Yes	Yes	
Corporate NCC Licensing	Yes	No	
Essential Licensing	<i>Avaya Aura® Contact Center Release 7.0 does not support Essential licensing for new installations.</i>		<i>Recommend moving to a full license type if moving to SIP/Aura</i>

Licensing Mechanisms

Licensing Mechanism	AML CS1000	SIP Aura/CM	Comments
WebLM License file	Yes	Yes	<i>IMPORTANT: WebLM does not support Corporate licensing</i>
Avaya WebLM server	Yes	Yes	<i>IMPORTANT: 1. WebLM does not support Corporate licensing. 2. Avaya Aura® Contact Center deployments that use Avaya WebLM server do not support the High Availability (HA feature).</i>
PLIC license file	Yes	No – uses Web LM licensing	<i>PLIC is supported for upgrades only where no additional features are required that depend on Web LM.</i>

Voice Services using Avaya Media Services licensing

The following table provides a summary of license requirements for a typical migration of voice services functionality from Call Pilot to AACC and Avaya Media Services:

Note: The licensing detailed below is in addition to standard Avaya Media Services instance and agent license requirements.

IVR Feature	AML CS1000	SIP Aura/CM	License Requirements
Call Pilot:			
IVR Integration (GIVE IVR script/application command)	Yes – Integration with AACC scripts. GIVE IVR is used when not using Voice Sessions	Use GIVE IVR with VXML – provides treatments to callers by playing recorded announcements or by collecting caller information from VXML scripts residing on the media server.	AACC inc. Avaya Media Services: Consumes an Announcement (ANNC) license if <u>only</u> playing an announcement.
Support for Voice Sessions	Yes	Provided by Avaya Media Server – Using enhanced GIVE IVR for SIP enabled contact centers	AACC inc. Avaya Media Services: Consumes a Dialog (DIALOG) license if playing an announcement <u>and</u> collecting digits.
Integrated Recorder Announcer (formally MIRAN):			
GIVE RAN	Yes	Yes	<i>The following script commands do not consume Announcement OR Dialog licenses:</i> - GIVE RINGBACK - GIVE RAN - GIVE MUSIC
GIVE MUSIC	Yes	Yes	

Licensing Reference Material:

For more detailed information, please refer to the following customer documentation:

- Avaya Aura® Contact Center Overview and Specification. Refer Part 3: Licensing requirements. The following are discussed:
 - o License types
 - o Licensing mechanisms
 - o How to obtain an Avaya Aura® Contact Center license
 - o License Manager installation location in a solution
 - o Avaya Aura Media Server licensing considerations
 - o Announcement and Dialog treatment licensing
 - o Licensed packages and features
 - o About the license file
 - o Licensing requirements for Agent Desktop features
 - o Licensing grace period
 - o License manager statistics
 - o Supported License Manager servers

5.8 AACC Scripting & Application Flows Considerations

Application Flows or scripts perform two major functions: they define the path a contact follows (contact routing) and they provide treatments to a contact as the contact moves through Contact Center (such as music or ring-back). You can also use the applications to track and record information about each step in the progress of a contact, and use this information to analyze how your contact center functions to improve service.

AACC application flows and scripts are very customizable and include dependencies with a range of PBX resources (Route Points, DNIS, agent ID etc.), in addition to this, interactions exist with many other solution components e.g. Voice Services and self-service options. For these reasons, flows used in an AML environment must be modified or re-written before they can be used in a SIP based contact center. In some cases, it may be sufficient to simply update any referenced resources within the migrated flows. In other cases it may be best to recreate the flows from new on the target SIP server.

In addition to script commands and intrinsics that are not applicable in a SIP contact center (these are detailed in the tables below), the following commonly used resources must be reviewed if you plan to reuse any migrated application flows or scripts:

- Route Points (CDNS's), DNIS and Landing pad numbers will need to be updated to reflect the dialing plan chosen for use with the SIP contact center.
- Recorded announcement (RAN) routes
- Music routes
- Skillset names and agent ID's that may be used in Queue To Agent or Queue to Agent syntax
- The agents in each skillset
- Contact center working hours and holidays
- Interactive Voice Response (IVR) queues
- Call treatments
- Any interactions with other applications, e.g. Voice Mail etc.

Questions answered

Question	Answer	Comments
Can I migrate and reuse my AACC application flows and scripts when I move to a SIP based contact center?	Yes	AACC flows and scripts are stored in the AACC database and are migrated as part of the AML to SIP migration tools and procedures. Migrated flows and scripts must be analyzed and updated to remove invalid syntax (AML specific) and reference SIP resources
I am using the Contact Router on my AML system. Will my Contact Router be migrated for use in a SIP based AACC?	Yes	A Contact Router configuration from an AML system will be migrated to the new AACC
Are there any other features or applications within AACC that can help me to update my application flows in SIP?	Yes	The Workflow Compliance Checker (provided with AACC 7.0) can be used to identify unsupported commands, blocks or intrinsics. Validation reports can be

Question	Answer	Comments
		generated to provide guidance on changes required. <i>Note: although syntax changes can be highlighted it will still be necessary to review all resources referenced by any migrated flows. E.g. IVR parameters, routes etc. to ensure they are correctly pointing to SIP resources</i> Orchestration Designer also allows you to generate documents on individual flows and scripts. These can be generated on the AML system and can be very useful when planning new/equivalent flows in SIP based contact centers. More details below.

Scripting Commands – Voice Processing

Command	AML CS1000	SIP Aura/CM	Comments
COLLECT DIGITS	Yes	N/A Replaced by enhanced GIVE IVR with VXML	Refer to DigitCollection sample flow/template in AACC to achieve digit collection contact flows
GIVE CONTROLLED BROADCAST ANNOUNCEMENT	Yes	N/A	Not supported on SIP deployments. Can be replaced using an IVR block
GIVE IVR	Yes	Replaced by enhanced GIVE IVR with VXML	Replaced by GIVE IVR with VMXL. Refer to Simple Greeting sample flow/template in AACC.
OPEN/END VOICE SESSION	Yes	N/A Replaced by enhanced GIVE IVR with VXML	Replaced by GIVE IVR with VMXL.
PLAY PROMPT	Yes	N/A Replaced by enhanced GIVE IVR with VXML	Replaced by GIVE IVR with VMXL.
Each of the above commands MUST be changed when moving flows to a newly deployed SIP system.			

Scripting Commands – Networking

Command	AML CS1000	SIP Aura/CM	Comments
CHANGE PRIORITY IN NACD	Yes	N/A	For use with CS1000 Network Automatic Call Distribution packaged feature
QUEUE TO NACD	Yes	N/A	For use with CS1000 Network Automatic Call Distribution packaged feature
REMOVE FROM NACD	Yes	N/A	For use with CS1000 Network Automatic Call Distribution packaged feature

Call Intrinsic

Intrinsic	AML CS1000	SIP Aura/CM	Comments
CALL FORWARD	Yes	No	<i>The Call is forwarded</i>
CALL FORWARD BUSY	Yes	No	<i>Call is forwarded because the original destination was busy.</i>
CALL FORWARD DO NOT DISTURB	Yes	No	<i>The call is forwarded because the Do Not Disturb was enabled at the original destination</i>
CALL FORWARD NO ANSWER	Yes	No	<i>The call is forwarded because the Call was not answered at the original destination</i>
CLID	Yes	Yes	<i>Support for the CLID intrinsic was introduced in AACC 7.0. This applies to voice contacts only. WILDCLID is also fully supported for SIP deployments. See sample custom script usage below</i>
Example WildCLID test <pre> /* Note: wildclid_cv is a WildCLID type */ ASSIGN 600@ TO wildclid_cv IF (CLID = wildclid_cv) THEN LOG "Got CLID is the 6000 range" ELSE LOG "Did not get a CLID in the 6000 range" END IF </pre>			
CONFERENCED	Yes	No	<i>Consultation call for a conference</i>
DIRECT CALL	Yes	No	<i>A direct call to a CDN.</i>
INTERNATIONAL CALL	Yes	No	<i>Returns the status of the call as an international call</i>
LOC	Yes	No	<i>Returns the Location Code of the call</i>
NETWORK CALL	Yes	No	<i>Indicates if the call is a networked call</i>
NPA	Yes	No	<i>Returns the Number Plan Area</i>
NPANXX	Yes	No	<i>Returns the Number Plan Area and the Local Exchange Code</i>
NXX	Yes	No	<i>Returns the Local Exchange Code</i>
TRANSFERRED	Yes	No	<i>Returns true if the call is transferred</i>

Script Command: Route Call to Voice mail – Special Considerations

This section reviews the ROUTE CALL script/application command when used in the context of sending a customer voice contact to the supported voice mail system. The characteristics and expected behavior of routing a caller to voice mail can be different on a SIP based contact center vs. a CS1000 (AML) based solution.

Consider the following use case:

In a CS1000 (AML) based contact center, Logic is applied to the AACC flows/scripts to re-direct customer voice contacts to voice mail (CallPilot) if agents are not available within a particular time period. To do this, a CallPilot Messaging CDN and a default mailbox for that CDN are created.

Configuration Summary:

CallPilot Messaging CDN:	9777
CallPilot mailbox:	9600
AACC Route Point (CDN):	9600

Voice Contact Flow:

- Customer arrives at Route Point 9600 and is queued to the Sales skillset
- After a period of time the customer is offered the option of being routed to a Voice mail system
- The **ROUTE CALL 9777** command is executed within the AACC flow/script
- The customer is routed to the 9600 mailbox
- This based on the initial 9600 route point entry for the customer

As already discussed in this document, Avaya Aura Messaging is typically used in SIP based deployments to provide Voice Mail functionality. In this case, the ROUTE CALL flow/application command will not behave in the same way if the caller is routed to the Avaya Aura Messaging Access Number.

Consider the same use case for SIP:

Configuration Summary:

Avaya Aura Messaging Access Number:	9777
AAM Mailbox:	9600
AACC Route Point (CDN):	9600

Voice Contact Flow:

In this case using the script command **ROUTE CALL 9777** will not route the call to the AAM mailbox 9600. Instead, the caller will be presented with the standard welcome options provided by Avaya Aura Messaging. Reasons for this are:

- When the Route Call command is encountered in the script, Avaya Aura® Contact Center will REFER the customer to Avaya Aura Messaging. This means that the

signaling is removed from the Avaya Aura® Contact Center, and media from the AMS.

- Avaya Aura Messaging requires that the History information (contained in the SIP Header) be populated with the Route Point information in order to make an association between the caller entry point and the desired mailbox. This is currently not supported by AACC.

Summary

The above use case represents one specific scenario and may not be applicable to your Contact Center or application flows. It is highlighted in this document so that alternative configuration options can be planned for in advance of any migration taking place. To provide similar behavior, it is important that the Communication Manager performs the final divert to the voice mail system and thereby sets the required History information to the SIP Header. A number of options are possible here and you should review the best solution for your contact center flows. Options include:

- Calls can be routed to a specific station with coverage path enabled to redirect callers to the desired mailbox.
- Calls can be routed to a Communication Manager VDN & Vector which routes the call to the voice mail system CDN. In this case the History information will also be present and the Caller can be routed to the appropriate mailbox.

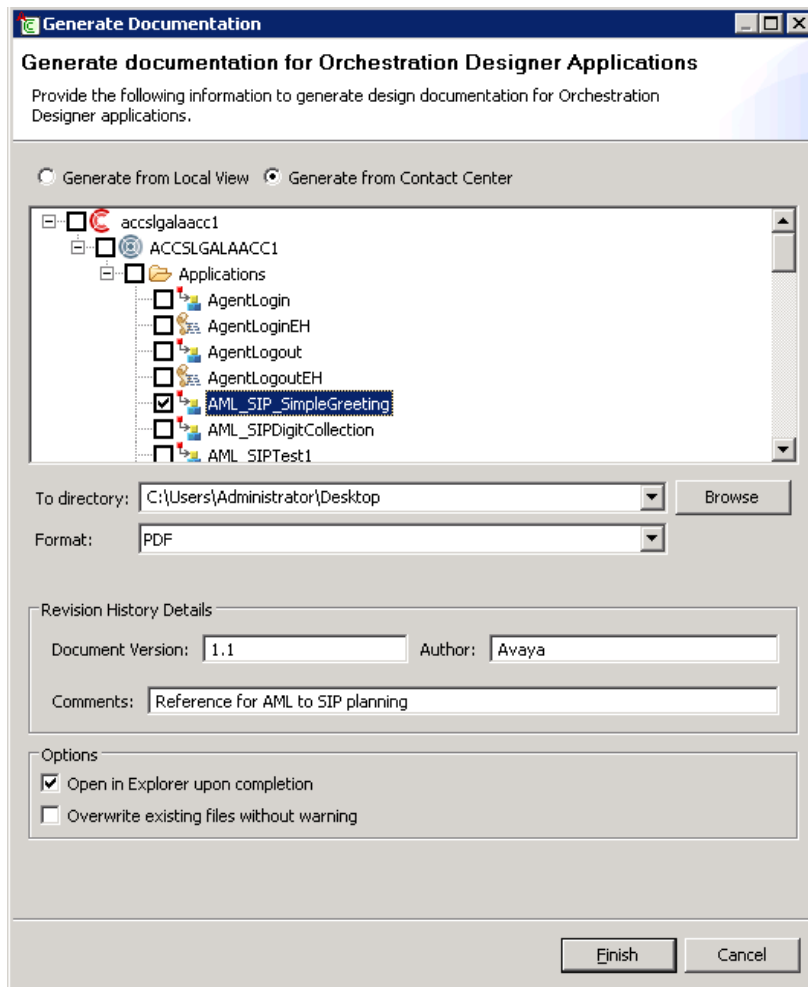
Toll Free Queuing in AACC

For a SIP based AACC without Toll Free Queuing enabled, the system connects the call immediately. This means that billing for the call starts immediately. When Toll Free Queuing is enabled, billing starts either when the call is answered by an agent or when an IVR Play and Collect is played or a set time in queue of max up to 2 minutes. For details of the Toll Free Queuing feature refer to AACC customer documentation. Toll Free Busy functionality (BUSY sent back to Intelligent Network for the call to receive an alternate treatment) is not available in SIP AACC if Toll Free Queuing is not enabled.

Generating Application flow documentation

Using Orchestration Designer to generate documentation of existing AML flows or scripts is a useful way of planning new flows in SIP based contact centers that replicate equivalent routing logic.

This can be run from the following screen within Orchestration designer:



The generated document provides a comprehensive summary of each operation and script command including graphical views of the contact flow. For more detailed information on generating these reports, refer to:

- Using Contact Center Orchestration Designer: *Refer to Chapter 10: Contact Center and Local view operations*

Reference Material:

Please refer to the following customer documentation:

- Avaya Aura® Contact Center Overview and Specification.
- Using Contact Center Orchestration Designer.
- Call Redirection with Avaya Aura® Contact Center 6.4 Application Note – available from <http://www.avaya.com/support>

5.9 Supervisor and Administrator functions

This section describes key considerations in terms of administration and supervisor functions and is primarily focused on AACC Administration operations. This is important during the migration and bulk update phases.

Summary information:

- AACC allows configuration data to be migrated from an AML Contact Center to a newly deployed SIP server. Once migrated, Agent, Supervisor and UC related resources must be updated with SIP specific details – This can be done in bulk using the SIP Spreadsheet and Excel. Relevant settings are detailed here.
- Agents must be configured with a Voice URI and a CCT login with associated active directory domain account to allow login via Agent Desktop or Agent Browser application.
- Call Presentation class and contact type options will differ in SIP
- Route-Points (CDNs), DNIS and landing pads must be configured to use URI's. This can be done using the SIP Spreadsheet and Microsoft Excel
- New options are available and must be provisioned as part of the standard commissioning procedures, e.g. Avaya Media Servers, CCT and CCMM servers are required for Agent Desktop login.
- Refer to the tables below for more detailed information.

Questions answered

Question	Answer	Comments
Can I migrate my Contact Center data from the AML server to my newly deployed SIP server	Yes	All applicable AML configuration and historical data can be migrated to AACC 7.0 Migrated data will need to be updated to include SIP specific details e.g. SIP URI address for agents, CDNs and DNIS
Are there tools within AACC that can help me update my migrated Contact Center data?	Yes	CCMA bulk update tools (described later in this document) can be used to bulk update many contact center resources
What CCMA data is migrated as part of the migration process	See list	- Users, Partitions, Access classes, Report Groups, Reporting Filters and Server entries - Reports: User-defined & Schedules - Real-time report information
Are there differences between Supervisor users in AML vs. SIP based deployments	Yes	Supervisor users in SIP based AACC are used to login to CCMA, run reports and manage agents. It is not possible to login to AAAD as a Supervisor user. To use Supervisor features such as Observe/Barge in and login to AAAD you must change accounts to be Supervisor/Agents.

Contact Center Administration and Management

Administration Function	AML CS1000	SIP Aura/CM	Comments
Contact Center Management – Agent & Supervisor Configuration			
Agent Login Details	Login ID, Personal DN and ACD Queue configured	Login ID, URI and CCT Agent information required	For SIP/Aura deployments, agent must login using Avaya Agent Desktop OR Simple Agent Desktop (voice only)
Agent Greeting configuration	No	Yes	Licensed feature using Avaya Media Server
Offsite Agent configuration	No	Yes	
Contact Types	No Support for IM Instant Message Contact Type	IM Supported in SIP deployments	
Telephony/Port information (Supervisor/agent only)	Yes	No	
Agent & Skillset assignments	Yes	Yes	
Access and Partition Management			
CCMA User roles	Yes	Yes	Administrator and Standard user roles supported
Access Classes	Yes	Yes	Access class options will vary based on supported features
User Defined partitions	Yes	Yes	NOTE: SIP/Aura systems also support partitioning of Avaya Media Server prompts
Standard Partitions (including standard report groups)	Yes	Yes	Report group contents will vary based on the deployed system. SIP systems also allow configuration of an 'All Prompts' standard partition.
User Defined Report Groups	Yes	Yes	
Prompt Management:			
Integrated Prompt management for Avaya Media Server prompts	N/A	Yes	Accessed via the Administration Launchpad
Emergency Help:			
Integrated Emergency Help reporting	Yes	Yes	Note for the Emergency help Real Time Display: In a SIP-enabled Contact Center, after a supervisor releases an Emergency call, the Emergency report displays the status of the Emergency call as "Closed". In an AML-based Contact Center, after a supervisor releases an Emergency call, the Emergency report continues to display the status of the Emergency call as "Active" until the agent releases the call. Refer to AACC Client Administration.
Call Recording and Quality Monitoring:			

Administration Function	AML CS1000	SIP Aura/CM	Comments
Link to CRQM server from AACC Administration	Yes	Yes	
Scripting: Please refer to the AACC Scripting & Application Flows Considerations section for complete information			
Configuration:			
Activity Codes	Yes	Yes	
Agent Greeting	No	Yes	
Call Presentation classes	Refer to AAAD and Phoneset section to review full list of supported options		<i>Let Call Ring and Answer by placing DN call on hold are not supported in SIP environments</i>
Call Recording and Quality Monitoring:	Yes	Yes	
CDNs (Route Points)	Based on Number	Based on Number & SIP URI	<i>Refer to Capacity tab for limits and maximum values</i>
DNIS	Based on Number	Based on Number & SIP URI	<i>Refer to Capacity tab for limits and maximum values</i>
Formulas	Yes	Yes	
Global Settings	Yes	Yes	<i>Refer to AACC Client administration guide AML or SIP specific settings</i>
Historical Statistics	Yes	Yes	
IVR ACD-DNs	Yes	N/A	<i>Replaced by Media Services and Routes below</i>
Media Servers	N/A	Yes	<i>For SIP, All Contact Center Calls are anchored on the Avaya Media Server and must be configured. Also provides voice services, Agent Greeting options etc.</i>
Media Services & Routes	N/A	Yes	<i>For SIP Media Services must be configured.</i>
Multiplicity Presentation Classes	Yes	Yes	
Networking Communication Parameters	Yes	Yes	
Phoneset Displays	Yes	N/A	<i>AACC does not display data on CM phones</i>
Phonesets	Yes	N/A	
Real-time Statistics	Yes	Yes	
Routes	Routes must be acquired	Avaya Media Server resources	
Skillsets	Yes	No Map ACD-DN Number configured for SIP/Aura	
Threshold Classes	Yes	Yes	

Note: When creating and configuring agents on a SIP based AACC solution, CCT address and terminals do not need to be configured. SIP URI's are required instead.

Supervisor Tools

Command	AML CS1000	SIP Aura/CM	Comments
Observe and Barge-In of inbound Contact Center Calls via AAAD	N/A In CS1000 Supervisors can observe from their phoneset	Yes – Applies only to Agent Supervisors	<i>Observing and Barging-in on a voice contact is possible only in a SIP-enabled Contact Center. You can observe only incoming Avaya Aura® Contact Center voice contacts. Avaya Aura® Contact Center does not support Observe or Barge-in for Outbound contacts or any other voice contacts.</i>
Observe and Barge-In of Agent initiated contact center calls. Example: An agent dials a CDN number for the contact center	N/A	Yes – Applies only to Agent Supervisors	<i>agent to agent calls cannot be observed OR agent calls to external numbers</i>
Observer and Barge-In of AACC web chat contacts	N/A	Yes – Applies only to Agent Supervisors	<i>AACC SIP also supports configurable thresholds for voice and web chat contacts</i>
Force agent login / Change agent state	Yes	Yes	
<i>The number of supported CCMA Supervisors on a SIP based AACC is defined by the license. Typically this is 10% of the agent count.</i>			

Reference Material:

Please refer to the following customer documentation:

- Avaya Aura® Contact Center Client Administration:
 - Contains complete information on creating and administering contact center data.
 - Refer to Part 2: Contact Center Management for information on creating agents & supervisors including 'Create Copy' and 'Add Many' options.

5.10 Performance Management & Historical Reporting

One of the significant changes in AACC 7.0 (for both AML and SIP based solutions) is the replacement of SAP Crystal Reports with Microsoft SQL Server Reporting Services (SSRS) as the historical reporting runtime engine within AACC. Although this change and its implications are not unique to AML to SIP migrations, it represents an important consideration in terms of planning a migration. Note the following highlights:

- Crystal report templates (.rpt) have been replaced by SSRS report templates (.rdl)
- The report 'look and feel' is consistent with previous Crystal based templates
- The report export options supported by SSRS are limited to pdf, Microsoft Word & Microsoft Excel
- XML and CSV formats are not supported
- Restrictions have been added within the CCMA application to limit the size of reports that can be generated.
 - This limit is designed to restrict the amount of memory that the CCMA Report Service will consume while the report is being generated
 - The guardrails limit the number of records that can appear on a report to 50,000

This section details differences in historical reporting for AML and SIP solutions. The number of report template changes for standard reports is low.

IMPORTANT: Activation of Contact Summary reporting must be set on the newly deployed SIP AACC server. These settings will not be migrated from the source server.

Questions answered

Question	Answer	Comments
Can I migrate my Contact Center Historical data from the AML server to my newly deployed SIP server	Yes	AML Historical data can be migrated to the target SIP AACC 7.0 deployment using the AML to SIP migration tools and procedures
What does the change from Crystal Reports to SSRS mean for my migration?	See Comments	1. Not applicable to AACC 7.0 (AML) to AACC 7.0 (SIP Aura) migrations. 2. AACC 7.0 provides 'like for like' SSRS templates to replace the standard Crystal templates 3. During a migration, any scheduled jobs referring to standard templates will be updated to run the SSRS (or .rdl) version of the report. 4. Custom reports built using RCW can be migrated using a set of RCW migration tools. These tools are discussed in the section Summary of Migration steps and also covered in detail in the <i>Upgrading and patching Avaya Aura Contact Center</i> document.

Question	Answer	Comments
		5. No support for custom reports built outside of RCW (e.g. using Crystal Designer). These reports cannot be migrated and must be re-created using either RCW or one of the third party report builder applications. 6. The report export options supported by SSRS are limited to PDF, Microsoft Word & Microsoft Excel. XML and CSV formats are not supported
Can I continue to use my user defined reports on the new SIP server?	Yes	Report templates are included in the Administration migration and can be used on the new server. See note at the end of this chapter for more information.
Aside from the new SSRS engine, are there other differences specific to an AML to SIP migration?	Yes	In SIP, there is no reporting of IVR ports & queues. Telephony display properties are not applicable. No trunk information.
Are there any differences to Real-Time Reporting	Yes	Agent Map Real Time Display supports 'Display arranged by Agent ID' only. Position ID is not applicable.

AACC Reporting Templates - Includes Networking & Multimedia

Standard Report	AML CS1000	SIP Aura/CM	Comments
Administration:			
Access Classes	Y	Y	
Report Groups	Y	Y	
User Defined Partitions	Y	Y	
Users	Y	Y	
Agent Performance:			
Activity Code By Agent	Y	Y	
Agent Average Calls Per Hour	Y	Y	
Agent Average Calls Per Hour, Bottom 5	Y	Y	
Agent Average Calls Per Hour, Top 5	Y	Y	
Agent By Activity Code	Y	Y	
Agent By Application Performance	Y	Y	
Agent By Skillset Performance	Y	Y	
Agent DN Performance	Y	Y	<i>In SIP based AACC, only calls made to OR received from an AACC agent or Supervisor/Agent are pegged as Internal. All Other calls are regarded as External.</i>

Standard Report	AML CS1000	SIP Aura/CM	Comments
Agent DN Performance Calls Answered, Bottom 5	Y	Y	<i>In SIP based AACC, only calls made to OR received from an AACC agent or Supervisor/Agent are pegged as Internal. All Other calls are regarded as External.</i>
Agent DN Performance Calls Answered, Top 5	Y	Y	<i>In SIP based AACC, only calls made to OR received from an AACC agent or Supervisor/Agent are pegged as Internal. All Other calls are regarded as External.</i>
Agent Efficiency	Y	Y	
Agent Efficiency By Contact Type	Y	Y	
Agent Efficiency By skillset	Y	Y	
Agent Login / Logout	Y	Y	
Agent NACD Activity	Y	Y	
Agent Performance	Y	Y	
Agent Performance by Supervisor	Y	Y	
Agent Performance Calls Answered, Bottom 5	Y	Y	
Agent Performance Calls Answered, Top 5	Y	Y	
Agent Short Calls	Y	Y	
Agent Transferred/Conferenced Activity	Y	Y	
Estimated Revenue By Agent	Y	Y	
Not Ready Reason Codes By Agent	Y	Y	
Skillset By Agent Performance	Y	Y	
Call By Call:			
Call By Call Statistics	Y	Y	
Configuration:			
Config – Activity Code Properties	Y	Y	
Config – Agent By Supervisor Properties	Y	Y	
Config – Agent Properties	Y	Y	
Config – Agent Skillset Assignment	Y	Y	
Config – Agent Skillset Properties	Y	Y	
Config – Agent Supervisor Assignment	Y	Y	
Config – Application Script Properties	Y	Y	
Config – Application Template Properties	Y	Y	
Config – CDN (RoutePoint) Properties	Y	Y	
Config – Database View Definitions	Y	Y	
Config – DNIS Properties	Y	Y	
Config – Formula Properties	Y	Y	

Standard Report	AML CS1000	SIP Aura/CM	Comments
Config – Historical and Real Time Statistics	Y	Y	
Config – IVR Queue and Port Properties	Y	N/A	<i>This report is not applicable to SIP systems</i>
Config – Logged In Agent	Y	Y	
Config – Multimedia	Y	Y	
Config – Route Properties	Y	Y	
Config – Script Variable By Script	Y	Y	
Config – Script Variable Properties	Y	Y	
Config – Skillset Properties	Y	Y	
Config – Supervisor Properties	Y	Y	
Config – Telephone Display Properties	Y	N/A	
Config – User Access Privilege	Y	Y	
Contact Summary:			
Activity Code By Address	Y	Y	
Agent By Address	Y	Y	
Contact Duration	Y	Y	
Contact Duration By Agent	Y	Y	
Contact Summary	Y	Y	<i>Reporting of Source of Call Disconnect for SIP Based AACC was introduced in AACC 6.4. Avaya Media Server Details only appear for SIP</i>
Contacts By Agent	Y	Y	
Originator By Disposition	Y	Y	
Media Server Call Distribution	No	Y	<i>Only applicable to SIP based solutions</i>
Multimedia:			
Contacts Closed By Reason Code	Y	Y	
Contacts Closed By Skillset	Y	Y	
Contacts Details	Y	Y	
Contacts Outstanding Detail	Y	Y	
Contacts Outstanding Summary	Y	Y	
Contacts Received by Skillset	Y	Y	
NCC:			
Config – Network Site and Application Properties (NCC)	Y	Y	
Config – Network Skillset Routing Properties (NCC)	Y	Y	
Config – Network Table Routing Assignment (NCC)	Y	Y	
Network Call By Call	Y	Y	
Network Consolidated Application Performance	Y	Y	

Standard Report	AML CS1000	SIP Aura/CM	Comments
Network Consolidated DNIS Statistics	Y	Y	
Network Consolidated Incoming Calls	Y	Y	
Network Consolidated Outgoing Calls	Y	Y	
Network Consolidated Route Performance	Y	Y	
Network Consolidated Skillset Call Distribution	Y	Y	
Network Consolidated Skillset Performance	Y	Y	
Nodal Consolidated Application Delay Before Abandon	Y	Y	
Nodal Consolidated Application Delay Before Answer	Y	Y	
Nodal Consolidated Application Performance	Y	Y	
Networking:			
Config – Network Site and Application Properties	Y	Y	
Config – Network Skillset Routing Properties	Y	Y	
Crosstab – Network Incoming Calls	Y	Y	
Crosstab – Network Outgoing Calls	Y	Y	
Network Application Performance	Y	Y	
Network DNIS Statistics	Y	Y	
Network Incoming Calls	Y	Y	
Network Outgoing Calls	Y	Y	
Network Route Performance	Y	Y	
Network Skillset Performance	Y	Y	
Network Skillset Timeline	Y	Y	
Network Skillset Timeline By Contact Type	Y	Y	
Others:			
Activity Code By Application	Y	Y	
Application By Activity Code	Y	Y	
Application By Skillset	Y	Y	
Application Call Treatment	Y	Y	
Application Delay Before Abandon	Y	Y	
Application Delay Before Answer	Y	Y	
Application Performance	Y	Y	
CDN (Route Point) Statistics	Y	Y	Avaya Media Server Details are available for SIP solutions
Crosstab – Application Performance	Y	Y	
Crosstab – CDN (Route Point) Statistics	Y	Y	Avaya Media Server Details are available for SIP solutions

Standard Report	AML CS1000	SIP Aura/CM	Comments
Crosstab – DNIS Statistics	Y	Y	
Crosstab – Route Performance	Y	Y	
Crosstab – Skillset Performance	Y	Y	
Crosstab – Trunk Performance	Y	Y	
DNIS Statistics	Y	Y	
IVR Port First Login / Last Logout	Y	N	
IVR Port Statistics	Y	N	
IVR Queue Statistics	Y	Y	
Music/RAN Route Statistics	Y	Y	
Route Performance	Y	Y	
Skillset By Application	Y	Y	
Skillset Performance	Y	Y	
Skillset Timeline	Y	Y	
Skillset Timeline By Contact Type	Y	Y	
Trunk Performance	Y	Y	
Outbound:			
Campaign Call Details	Y	Y	
Campaign Performance	Y	Y	
Campaign Script Results Details	Y	Y	
Campaign Summary	Y	Y	
Script Summary	Y	Y	
<i>Note: Activation of Contact Summary reporting must be set on the newly deployed SIP AACC server. This will not be migrated from the source server.</i>			

User Defined Reports (i.e. Standard reports saved with user specific criteria) will continue to operate on the newly deployed SIP server. Note that some specified criteria may not be available on a SIP server.

Report Creation Wizard Reports:

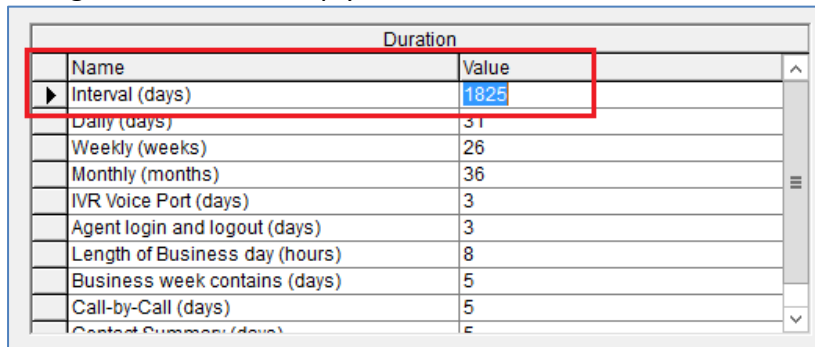
When migrating from any version of NES or AACC prior to AACC 7.0 (AML), reports that have been created using RCW can be migrated using a set of RCW migration tools. These tools are discussed in the section [Summary of Migration steps](#) and also covered in detail in the Upgrading and patching Avaya Aura Contact Center document.

User Created Reports – created outside of Report Creation Wizard:

When migrating from any version of NES or AACC prior to AACC 7.0 (AML), custom reports built outside of RCW (e.g. using Crystal Designer) cannot be migrated and must be re-created using either RCW or one of the third party report builder applications. These reports will not be available on the target 7.0 server and if required, reports should be re-created in advance of any migration.

Storage of Historical Reporting Statistics Limited to 1825 days (5 years):

The limit on the number days/weeks/months and Interval (days) for which historical statistics can be stored within AACC has been set to 1825 (e.g. "Interval (days)" in the screenshot below). These values are configured within the Configuration – Historical Statistics page of CCMA. An attempt to enter a value greater than 1825 will result in an error on the page. Systems upgrading from earlier releases where the limit was higher must now reduce the configured value to comply with this limit.



Duration	
Name	Value
Interval (days)	1825
Daily (days)	31
Weekly (weeks)	26
Monthly (months)	36
IVR Voice Port (days)	3
Agent login and logout (days)	3
Length of Business day (hours)	8
Business week contains (days)	5
Call-by-Call (days)	5
Contact Summary (days)	5

Reference Material:

Please refer to the following customer documentation:

- Using Avaya Aura Contact Center Reports and Displays
- Contact Center Performance Management Data Dictionary

5.11 Multimedia & Outbound considerations

Multimedia contact routing is an optional feature within AACC and may not be part of your solution. Multimedia components can be used to configure, administer and report on multimedia contacts but are also used to configure and deploy and administer Avaya Agent agent desktop – required on all SIP/Multimedia contact centers.

Multimedia data and configuration details exist in their own database, however there are intersections with other core AACC data, e.g. skillsets. For this reason the following questions are important:

Note: In a voice only-solution, the Multimedia Administration application must still be used to configure Avaya Agent Desktop features and functions.

Questions answered

Question	Answer	Comments
Can I migrate my Multimedia configuration and historical data to my new AACC 7.0 SIP server?	Yes	Multimedia data can be migrated as part of the AML to SIP migration process in AACC 7.0 Note: <i>There is no supported method of migrating archived Multimedia Data from an NES 6.0 deployment.</i>
Can Agent phonebooks and contact lists (available via Agent Desktop) be migrated?	Yes	This data is included in the Multimedia Backup and can be migrated as part of the AML to SIP migration process Note: <i>Manual Multimedia steps are required after a migration and Agent Desktop will need to be re-installed</i>
Are there new Multimedia features available to me when I move to SIP	Yes	Instant Message Contact types can be routed as an additional routed contact type. Enterprise Web Chat is available as an additional licensed feature.
Is POM supported on SIP Aura/CM	Yes	POM is supported on both solutions. Agents also have the option to use embedded softphone with AAAD.

Multimedia Features

Command	AML CS1000	SIP Aura/CM	Comments
E-mail as routed media type	Yes	Yes	
Web Chat as routed media type	Yes	Yes	
Enterprise Web Chat	No	Yes	<i>Enhanced capacity Web Chat – This is a licensed feature and requires custom desktop development via Enterprise Web Chat SDK.</i>
SMS as routed media type	Yes	Yes	
Fax as routed media type	Yes	Yes	
Federated Instant messaging as routed media type	No	Yes	<i>Avaya Presence Server, Lync 2013</i>

Command	AML CS1000	SIP Aura/CM	Comments
Expert Presence display and consultation	No	Yes	
Peer to Peer/Expert agent IM (not routed contact type)	No	Yes	
Social Media as routed media type	Yes	Yes	<i>APS Offer</i>
Multiplicity	Yes	Yes	<i>1 Voice + 1-5 non voice contacts</i>
Customer Contact history	Yes	Yes	<i>Requires Multimedia Install</i>

Outbound Features

Command	AML CS1000	SIP Aura/CM	Comments
Preview / Progressive outbound dialing and integrated campaign management and reporting	Yes	Yes	
Predictive Outbound – blended web services integration with Proactive Contact 5.0	Yes	Yes	<i>APS custom integration</i>
Scheduled web callback	Yes	Yes	
POM: Avaya Proactive Outreach Manager (Release 3.0 + AACC 7.0)	Yes	Yes	<i>Possible to make use of embedded softphone in AAAD with SIP.</i>

Reference Material:

Please refer to the following customer documentation:

- Avaya Aura® Contact Center Server Administration Release:
 - Part 7 Contact Center Multimedia
- Also refer to the [Agent Desktop and Common Agent Features](#) section of this document.

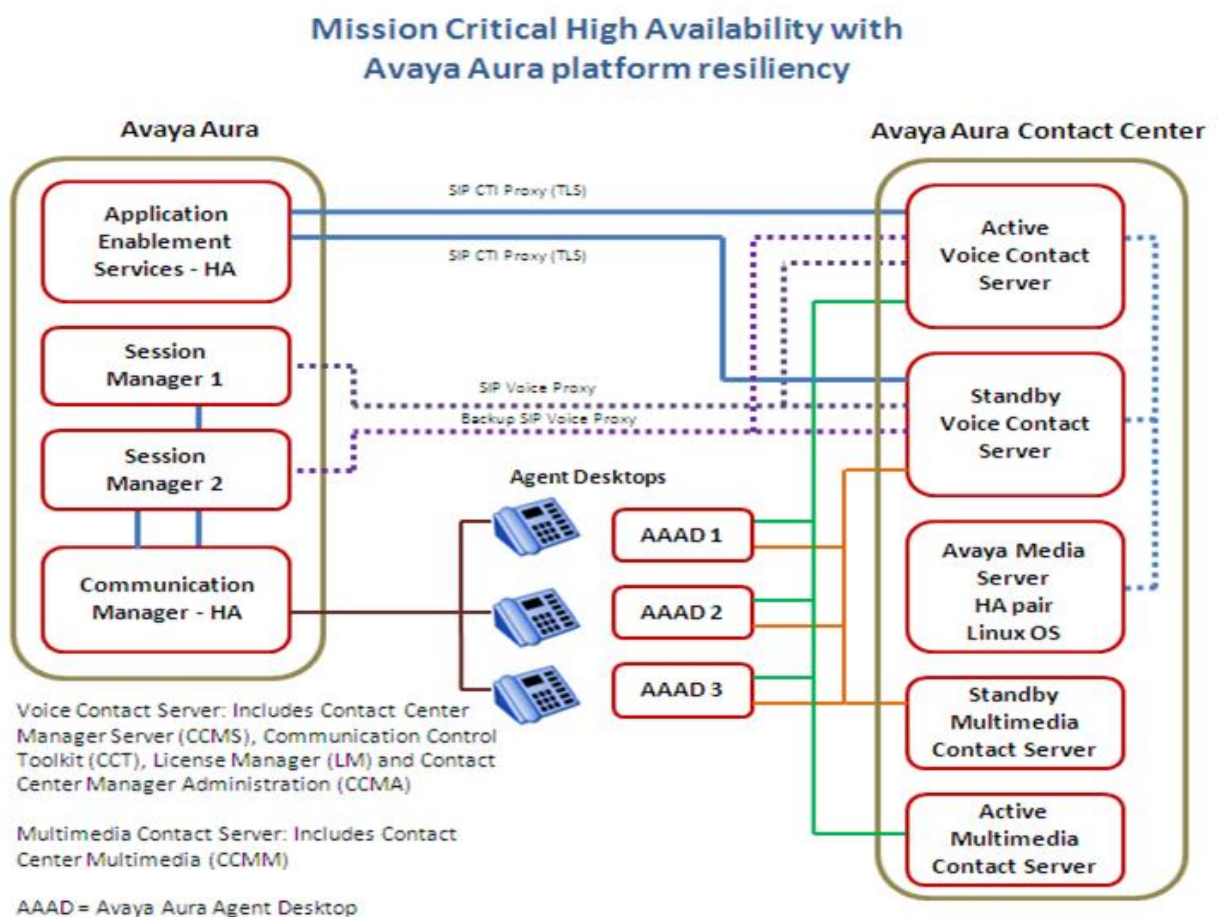
5.12 AACC High Availability

Contact Center High availability is offered on both AML/CS1000 systems and on SIP with Aura CM. The way in which HA is implemented and the expected behavior for agents and supervisors in the event of switch-over varies between AML and SIP solutions. There are also considerations in terms of network topologies and dependencies with the UC infrastructure. As this is a complex area, this document does not attempt to describe all high availability options that exist, however, important differences are highlighted in the tables contained in this chapter.

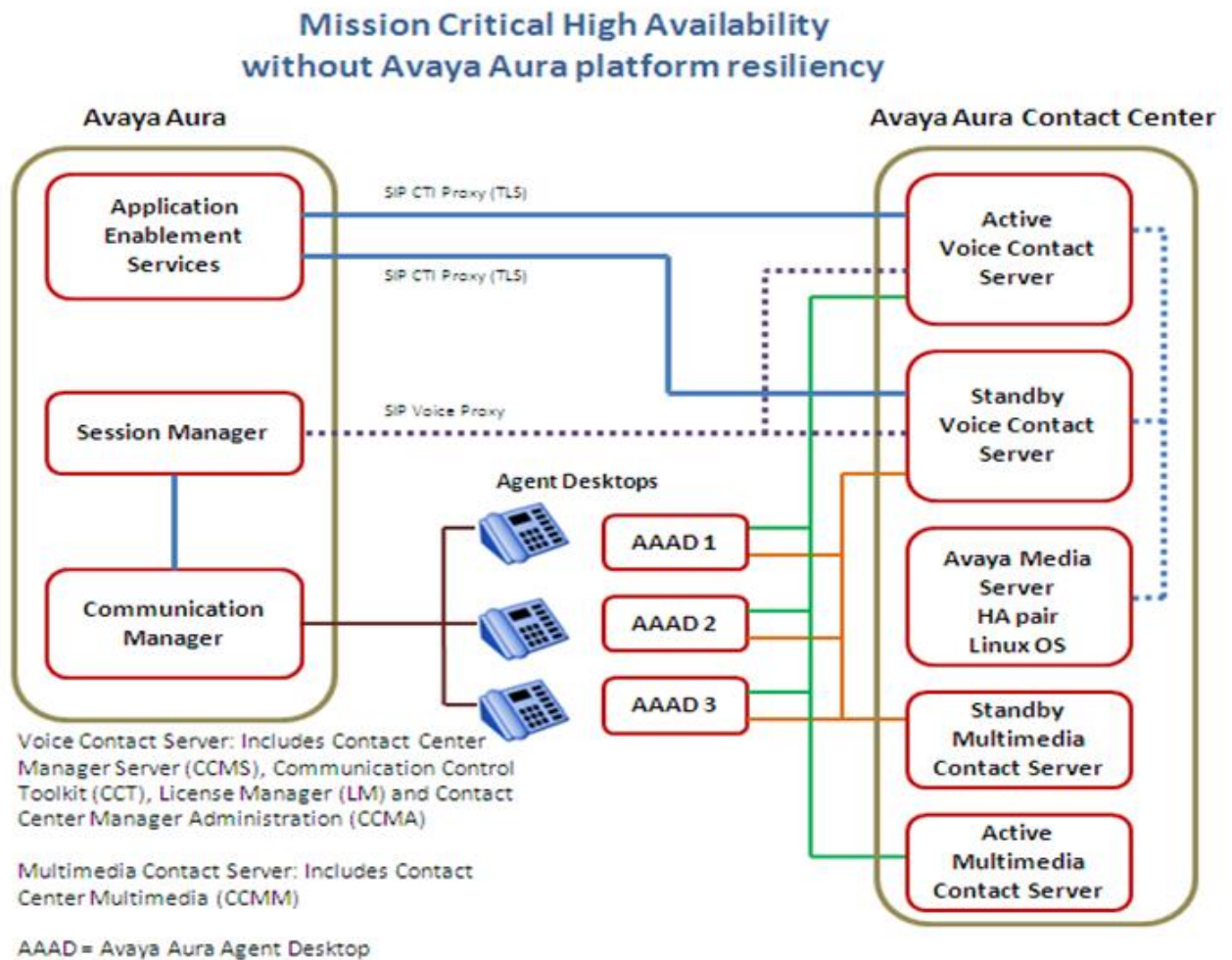
Readers are advised to review the chapters covering High Availability in the Contact Center Overview and Specification guide for complete information when planning a deployment.

SIP based contact centers deployed in Aura/CM are supported with mission critical HA (Campus) which has a number of advantages over other HA options – these are highlighted in this chapter. Supported configurations for use with Aura/CM are listed below:

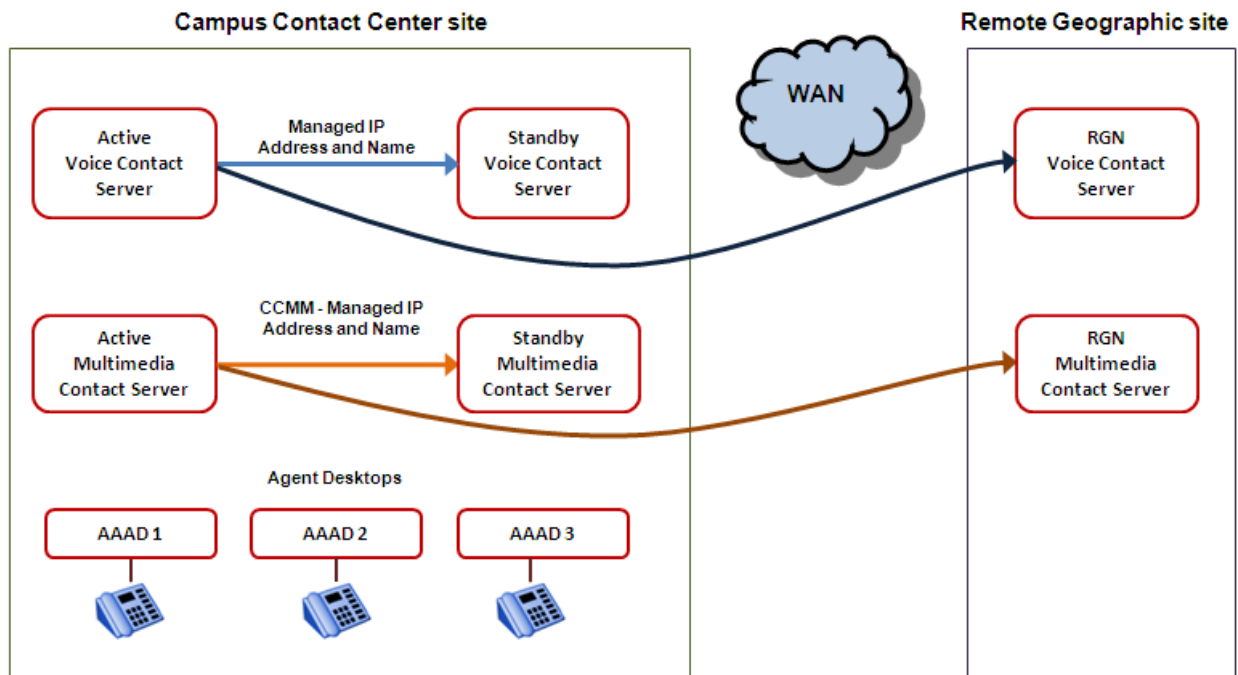
Option1: Mission Critical HA with platform resiliency



Option2: Mission Critical HA without platform resiliency



Option3: Mission Critical HA (Campus) with Remote Geographic Node



Voice Contact Server: Includes Contact Center Manager Server (CCMS), Communication Control Toolkit (CCT), License Manager (LM) and Contact Center Manager Administration (CCMA)

Multimedia Contact Server: Includes Contact Center Multimedia (CCMM)

AAAD = Avaya Aura Agent Desktop

Fallback & Fallback ACD options

Default ACD fallback is provided by the Communication Server for AML contact centers. This allows ACD call routing in the event of an AACC outage when not deployed using HA. Comparable options for Aura/SIP contact centers are detailed below. More details available in the Avaya Aura® Contact Center – Avaya Aura® Unified Communications Integration guide

AACC Fallback options	AML CS1000	SIP Aura/CM	Comments
Fallback ACD	Yes	N/A	CS1000 Specific ACD
Fallback to Avaya Aura Communication Manager Hunt Group	N/A	Yes	If Avaya Aura® Contact Center is unable to process voice contacts, Avaya Aura® Session Manager can reroute customer voice contacts intended for Contact Center to an Avaya Aura® Communication Manager Hunt Group.

AACC Fallback options	AML CS1000	SIP Aura/CM	Comments
Fallback to Avaya Aura Call Center Elite skill	N/A	Yes	<i>In solutions with Avaya Aura® Call Center Elite, Avaya Aura® Session Manager can also reroute customer voice contacts intended for Contact Center to an Avaya Aura® Communication Manager split or Elite skill.</i> Note: agents must not login to Elite and AACC simultaneously

Campus High Availability

AML based CS1000 systems support Hot Standby high availability when deployed in Campus environment. SIP/Aura solutions are supported with Mission Critical high availability. The feature differences are highlighted in this section: More details in: Avaya Aura® Contact Center Overview and Specification guide

HA Feature	AML CS1000	SIP Aura/CM	Comments
Provisioning & Configuration Tools	No SMMC. Configured using HA UI & managed by Master Service	Centrally managed using SMMC, High Availability UI & System tray options	<i>Mission Critical HA is centrally managed by the System Management and Monitoring Component (SMMC). This monitors network communications, network latency, and contact center components. If a Contact Center component or network link fails, SMMC detects this failure and takes appropriate action.</i> <i>SMMC (SIP/Aura only) ensures that both active and standby servers are fully synchronized prior to allowing any switch-overs to occur.</i>
HA enabled characteristics	Only active server AACC services are running. Standby services in a stopped state	Both active and standby services are running	<i>For SIP/Aura systems a number services will not be running on the standby server. E.g. CCMA RTD service. These will automatically start upon switch-over.</i>
Data Replication: Cache replication of CCMS, CCT, CCMA & CCMM data. (Configuration and contact data)	Yes	Yes	<i>Both AML & SIP/Aura systems rely on Cache database replication</i> <i>Administration (CCMA) data was added to Cache in AACC 7.0</i>
Data Replication: Real-time replication of contact center state information.	No	Yes	<i>In SIP/Aura deployments real-time contact/session state is synchronized to the standby server using CMF replication.</i>

HA Feature	AML CS1000	SIP Aura/CM	Comments
Switchover: Contact handling	In treatment calls handled by ACD. Existing calls maintained	In treatment calls handled by switchover handling script. Existing calls maintained	The switchover handling script (Mission Critical HA) can be customized to suit customer needs
Switchover: Performance	Time dependent on configuration and service start-up times	Seconds level switchover	<i>In SIP/Aura HA deployments, switchover occurs in a number of seconds. AACC services already running. State information is actively replicated to the standby server and AACC agents are already 'pre-authenticated' and logged in.</i>
Switchover: Agent Experience	Agents log back on	Agent state (Agent Desktop) is maintained	<i>In SIP/Aura HA deployments, AAAD and CCT maintain agent state. No requirement to login.</i>
Environment: Managed Name/IP support	Yes	Yes	<i>Managed name/ip is required for SIP/Aura systems</i>
Environment: Domain membership required	Yes	Yes	<i>Domains are standard for all SIP based AACC</i>
Environment: Supported Deployments	Campus deployment in same subnet. Remote locations are possible	Mission critical HA is supported in a Campus (one subnet) deployment only	<i>Mission Critical requires (at a minimum) two Voice Contact servers (CCMS, CCT, CCMA & LM), two Multimedia contact servers (if using Multimedia), two Avaya Media Server Linux-based servers, redundant Ethernet switches and a Windows AD domain controller & DNS. In terms of UC integration, Mission critical HA can be deployed with or without Avaya Aura platform resiliency</i>
Voice Services HA	Call Pilot HA	Full HA for Avaya Media Servers	<i>For SIP/Aura: Two Avaya Media Server Linux servers are required and must be configured as a HA pair</i>

Remote Geographic Node

RGN Support	AML CS1000	SIP Aura/CM	Comments
RGN Support for Campus High Availability	Yes – Manual intervention required	Yes – Manual intervention required	<i>SIP/Aura deployments: RGN servers can only be deployed with a Mission Critical HA with full Avaya Aura platform resiliency in place. An Additional UC infrastructure is required at the RGN node.</i>

Reference Material:

Please refer to the following customer documentation:

- Avaya Aura® Contact Center Overview and Specification:
 - Part 1: Overview: Refer to High Availability Chapters
 - Part 2: Interoperability

- Avaya Aura® Contact Center and Avaya Aura® Unified Communications Platform Integration Release 7.0.
- Application Note: Failover Solutions for Avaya Aura® Contact Center to Avaya Aura® Communication Manager Hunt Groups and/or Avaya Aura® Call Center Elite with Vector Variables. Available from <http://www.avaya.com/support>

5.13 Solution Integration Points

This section of the document provides information on common integrations with AACC. The products and features referenced here are typically additional licensed features or add-ons to the core AACC product and may not be applicable to your solution.

Questions answered

Question	Answer	Comments
I have applications that use Meridian Link Services as part of my existing AML solution. Can I use these applications when I move to SIP?	No	There is no MLS support on SIP based AACC
Can I reuse my NCC server when I move to a SIP based solution?	See Notes	Only possible if the NCC server is at an appropriate software release and hardware specification for AACC 7.0. e.g. if migrating from AACC 7.0 (AML) to AACC 7.0 (SIP/Aura) Please refer to the Contact Center Overview and Specification guide for more information.
While staging my migration, Can I network mixed nodes (SIP & AML) in my enterprise?	Yes	This is possible. Refer to the Contact Center Overview and Specification guide for more information

Complementary & Packaged Applications

Application	AML CS1000	SIP Aura/CM	Comments
Meridian Link Services (MLS)	Yes	No	Self-service/IVR integration inc. front end. There is no support for MLS in SIP based solutions.
Host Enhanced Routing	Yes	N/A	CS1K specific functionality
Knowledge Worker Server (CCT Install Options)	Yes	No	Only supported in CS1000/AML deployments. Knowledge Worker solutions support only voice calls, they do not support multimedia contacts. Knowledge Worker solutions do not support routed voice contacts or skill-based routing. A Knowledge Worker terminal cannot log on to an ACD queue or answer calls routed to a queue. A Knowledge Worker terminal can make and answer regular calls.

AACC Networking

Important information regarding Universal Networking when migrating to AACC 7.0 from earlier contact center releases:

Although not specific to AML to SIP migrations, the following information is applicable to any network (Universal Networking) of AACC servers upgrading/migrating to AACC 7.0.

A new 10.1 version of Gigaspaces is deployed with AACC 7.0. This is not compatible with earlier versions of Contact Center using this feature. Before adding an AACC 7.0 to an existing 6.x network of contact centers, the network must first be upgraded with a UNE alignment patch. All 6.x AACC nodes must first be upgraded to include this alignment patch.

Networking is supported in both AML and SIP environments but implemented using difference features and configuration.

Networking Feature	AML CS1000	SIP Aura/CM	Comments
Network Skill-Based Routing (NSBR)	Provided by NACD package on CS1000 OR Universal Networking. DNIS routing requires Universal Networking enabled	Provided via Universal Networking feature	
Ability to network multiple Contact Center Manager Servers	Yes	Yes	
NCC Support for mixed (AML & SIP-enabled nodes)	Yes	Yes	<i>Avaya Aura® Contact Center supports networked calls between AML-based and SIP-enabled nodes. The Network Control Center (NCC) software must be the most recent release, relative to the other nodes on the network. For example, in a network with Avaya Aura® Contact Center Release 6.4 and Avaya Aura® Contact Center Release 6.3 nodes, the NCC must be Release 6.4. All nodes in the network must be Avaya Aura® Contact Center Release 6.2 or later.</i>
Networking statistics and displays	Yes	Yes	
Network ACD for Fallback or Queue to NACD	Yes	No	<i>NACD requires MCDN trunks</i>
ACD for Fallback/Defaulting	Yes	No	<i>Only applicable where the agent is based on a CS1000 AML system</i>
Network Message Service	Yes	No	<i>NMS requires MCDN trunks</i>
Network Attendant Service	Yes	No	<i>NAS requires MCDN trunks</i>

Networking Feature	AML CS1000	SIP Aura/CM	Comments
Attached Data Networking for Telephone Sets	Yes	No	<i>AML uses TAPI Networking</i>
Attached Data Networking for CTI/Soft-phones/Toolbars	Yes	Yes	

Integration with Security Framework

AACC supports security framework deployments that use System Manager as their primary security server. This can be the System Manager of either an Avaya Aura® Communication Manager PABX or an Avaya Communication Server 1000 PABX. You can upgrade an existing Contact Center security framework deployment, where the primary security server is a Unified Communications Management (UCM) server or a standalone Contact Center Security Framework server. However, you can no longer install a Security Framework server from the Contact Center DVD.

“Security Framework server” refers to a legacy AACC Windows-based standalone server installed from the Contact Center DVD. The phrase “security framework” refers to a deployment of Contact Center that uses a primary security server to authenticate AACC Administration log on requests

Customer Relationship Management (CRM) connectors

CRM Connector	AML CS1000	SIP Aura/CM	Comments
Salesforce Integration: Simple URL based Out-of-the-box screen pop for Salesforce.com	Yes	Yes	Enhanced list of available intrinsics are available with SIP based deployments
Support for other CRM connectors	<i>Refer to DevConnect program for 3rd party connectors, including those from AMC Technology</i>		

Avaya Presence Server & Microsoft Lync integration

SIP-enabled Avaya Aura® Contact Center supports Presence services provider and Microsoft Lync Server as an Instant Messaging (IM). Avaya Aura® Contact Center can process the IM messages it receives from Microsoft Lync Server and then route the messages to available Avaya Aura® Agent Desktop agents. This is only available in SIP contact centers.

IM Application	AML CS1000	SIP Aura/CM	Comments
Avaya Aura Presence Server 6.1 – Presence and IM as routed contact type	No	Yes	Agent profiles will need to be updated (Administration – Contact Center Management) to include the IM contact type and the appropriate skillset assignments.

IM Application	AML CS1000	SIP Aura/CM	Comments
Microsoft Lync Server 2013 – Presence and IM routed as contact type	No	Yes	Agent profiles will need to be updated (Administration – Contact Center Management) to include the IM contact type and the appropriate skillset assignments.

SDK's, Open Interfaces & CTI Integration:

All of the packages listed below are available for download from Dev Connect.

IMPORTANT: Although SDK's may be supported on AML/CS1000 deployments and SIP Aura/CM, applications using open interfaces may need to change to provide comparable functionality in a SIP environment – example CCT open interfaces. Please refer to the appropriate SDK documentation for the list of supported operations.

Package	AML CS1000	SIP Aura/CM	Comments
CCMA Open interfaces and SDK	Yes	Yes	
CCMM Open Interfaces (Agent Web Services SDK)	Yes	Yes	
CCMS Host Data Exchange HDX Interface SDK	Yes	Yes	Host Data Exchange is supported in both AML (CS1000) and SIP (Aura) solutions. If moving from an AML to a SIP environment, ensure that any OD/scripting commands, variables and intrinsics used by the HDX interaction are also supported in SIP. As with any other scripts/OD applications, these must be imported and re-validated manually when moving from AML to SIP. Complete details of HDX integrations are included in the following guide: Avaya Aura® Configuration—Orchestration Designer Application Development
CCMS Open Networking SDK	Yes	Yes (Not Recommended for AAEP integration with SIP based AACC – Other options detailed in AAEP Integration section)	The Open Networking Open Interface enables a 3rd Party Application to perform call transfers between different nodes in a network. Any data associated with the call intact will be intact. 3rd Party Applications will be able to reserve a Landing Pad on the target node enabling the call to be transferred with data attached. The web services will also provide the functionality to cancel the reserving of a Landing Pad freeing it up for other calls to be transferred across the network.

Package	AML CS1000	SIP Aura/CM	Comments
CCMS Open Queue SDK	Yes	Yes	
CCMS Real-Time statistics Multicast SDK	Yes	Yes	
CCT .NET SDK	Yes	Yes	<i>Refer to SDK and SDK documentation for information of supported interfaces and operations</i>
CCT Open Interfaces SOAP SDK	Yes	Yes	<i>Refer to SDK and SDK documentation for information of supported interfaces and operations</i>
CCT Open Interfaces REST SDK	Yes	Yes	<i>Refer to SDK and SDK documentation for information of supported interfaces and operations</i>
Outbound Web Services	Yes	Yes	
Web Communication SDK	Yes	Yes	
Enterprise Web Chat – Web Chat SDK & Custom Desktop SDK	No	Yes	<i>Only supported on Multimedia Contact server + Voice Contact Server (separate servers)</i> <i>Not supported on co-res Voice and Multimedia servers</i>

Remote Agent Observe

This feature is only supported on AML (CS1000) deployments. Remote Agent Observe allows supervisors and observers to remotely observe and record agent calls without having to install any physical equipment at the observer's location.

Remote Agent Observe features	AML CS1000	SIP Aura/CM	Comments
Agent Observe criteria	Integrates with AACC based on Position ID, Agent ID, Skillset ID, DNIS, Application ID & CDN	N/A	<i>Refer to the Supervisor Tools section of this document for more information on supported observe options</i>
Support for multiplicity enabled AACC agents	No	N/A	<i>Remote Agent Observe will not operate with multiplicity enabled agents in an AACC contact center. This is due to a change in the RSM stream that Nortel Remote Agent Observe was not designed to handle. If an individual agent is not multiplicity enabled, Remote Agent Observe will operate correctly for that agent. Ref: Remote Agent Observe Planning, Installation and Administration Guide</i>
Call Recording	Yes	N/A	

5.14 Work Force Optimization & AACC

Workforce Optimization including Call recording are commonly deployed with AACC. Important variations between AML and SIP in terms of features and configurations are highlighted here.

Supported Versions

WFO Support	AML CS1000	SIP Aura/CM	Comments
Avaya Aura Workforce Optimization (WFO) Versions 12.0 & 12.1	Yes – CTI integration is via MLS	Yes	

Avaya Contact Recorder

AML and SIP contact centers support Contact Recording. Features include:

ACR Feature	AML CS1000	SIP Aura/CM	Comments
Duplicate Media Streaming	Yes	N/A	<i>Customers need to move to a DMCC model (below) when moving to an Aura/CM environment. Need to consider and plan for the introduction of gateways to support moving to Aura/CM</i>
Single-step conferencing on DMCC (Non AACC Calls)	N/A	Yes	<i>This option produces a single merged recording. Gateway infrastructure required. For Contact Center Calls AMS is used by ACR.</i>
Trunk-side Tap	Yes	Yes	<i>In the case of both switch types where an AACC is present, recording via AMS will take precedence over the points:</i> CS1000 1). Internal calls cannot be recorded. This makes it unsuitable for quality monitoring. 2). An Avaya Contact Center Manager Server is required Aura/CM 1). Internal calls cannot be recorded. This makes it unsuitable for quality monitoring. 2). An Avaya Contact Center Manager Server is NOT required
Passive IP Tap	N/A	Yes	
TDM Station-side Tap	Yes	Yes	

ACR Feature	AML CS1000	SIP Aura/CM	Comments
Beep Tone	Uses CS1000 / MLS	Provided by AACC / Avaya Media Server OR ACR	CS1000 – If the beep tone is set on: The beep tone is recorded The beep tone is heard by all parties on the call SIP/CM – 2 different approaches here depending if the recording is of a contact center call (AACC/AMS) or a non-contact center call. 1). <u>AACC/Avaya Media Server</u> – This is turned on/off at the AMS level. - The beep tone is recorded - The beep tone is heard by all parties on the call 2). <u>Non-AACC</u> – Parameter on ACR that, when set, will force the ACR to inject a beep tone. - The beep tone is recorded - The beep tone is not heard by parties on the call
Record on Demand	Yes	Limited Support	Possible to implement by conferencing in a recording port. No possible via AAAD.
Save Conversation	Yes	No	
Secure Call Recording (SRTP)	Yes	No	Not supported on SIP Agent/Avaya Media Server contacts
Multi-DN Recording – Ability to choose between recording AACC calls OR personal DN calls	Yes	All calls a recorded	
Supported OS for ACR	Windows Only	Windows OR Linux	TDM Tap is not supported on Linux
Support for Contact Recording Desktop (CRD)	Yes	No – AIM (Agent Initiated Monitoring) is supported which provides partial functionality	

Outbound Dialer Support

Dialer	AML CS1000	SIP Aura/CM	Comments
SER	Limited availability of SER dial on pre-AACC6.4	N/A	<i>POM is recommended on AACC 7.0 for AML OR SIP deployments</i>
POM	Yes	Yes	

5.15 CCMA Consolidated Configuration Tool

CCMA Consolidated Configuration Tool (previously SCT tools) are a useful method of bulk download and upload of commonly used Contact Center Data. These tools are Microsoft Excel based and can be used to populate agent, supervisor, Route Point (CDN) and DNIS etc. information.

The Spreadsheet allows many Agent properties/fields to be updated with new information. Updates are also supported for additional fields in CDN and DNIS resources. This represents an important phase in an AML to SIP migration as it allows migrated data (AML) to be updated in bulk for use in a SIP based deployment. Example: Bulk configuration of URI information for agents, CDNs and DNIS entries.

Fields supporting update operations are detailed in the Matrix below.

Summary information:

- CS 1000 and SIP specific spreadsheets are available from CCMA Configuration component.
- Possible to download existing OR upload new contact center data.
- Possible to update/modify a subset of data (detailed below)
- Not all data is included in the Tools. It may be necessary to manually add items before running. Example: Multiplicity Call Presentation Classes.

Field Matrix

The following table highlights important fields that will be required if using the Configurations Tools when migrating to SIP. Ensure these are available and URI information has been defined prior to uploading data to the AACC server.

Differences between supported and required fields for AML vs. SIP solutions are documented below:

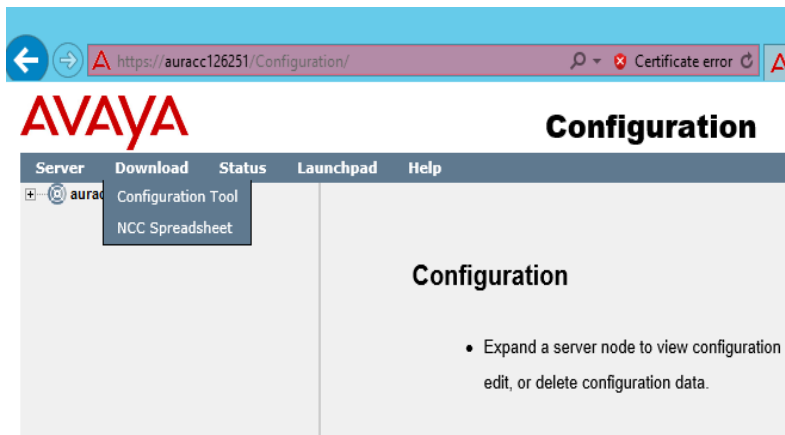
Field	AML CS1000	SIP Aura/CM	Update Support	Comments
Agent/Supervisors:				
FirstName	Required	Required	Yes	
LastName	Required	Required	Yes	
UserCapability	Required	Required	No	
LoginID	Required	Required	No	
ServerUtilityUserID	Required	N/A	N/A	
Language	Required	Required	Yes	
Title	Optional	Optional	Yes	
Department	Optional	Optional	Yes	
Comment	Optional	Optional	Yes	
AccessClass	Optional	N/A	N/A	
Telephony/Port Address	Required	N/A	N/A	
PersonalDN	Optional	N/A	N/A	

Field	AML CS1000	SIP Aura/CM	Update Support	Comments
IMUri	N/A	Optional	Yes	Multimedia / Instant Message requirement only
VoiceUri	N/A	Optional	Yes	This field will be defaulted by the Tool but should be set to a correct value for voice agents
CallPresentation	Required	Required	Yes	Call Presentation Classes must be added prior to creating an agent
MultiplicityPresentation	Required	Required	Yes	This must be specified for any new agents but cannot be added using the SIP Spreadsheet. Multiplicity should be created via CCMA in advance of running this Tool
Threshold	Required	Required	Yes	
AgentKey	Optional	N/A	N/A	
DefaultACDQueueNumber	Optional	Optional	N/A	
DefaultACDQueueErrorCode	Optional	Optional	N/A	
PrimarySupIP	Required	Required	Yes	
AgentGreetingEnabled	N/A	Optional	Yes	Avaya Media Server Agent Greeting only
AgentGreetingPassword	N/A	Optional	Yes	Avaya Media Server Agent Greeting only
AgentGreetingRecorded	N/A	N/A	N/A	Avaya Media Server Agent Greeting only
OffsiteAgentEnabled	N/A	Optional	Yes	
OffsiteAgentPhone1	N/A	Optional	Yes	
OffsiteAgentPhone2	N/A	Optional	Yes	
OffsiteAgentAlternatePhone	N/A	Optional	Yes	
NonSkillsetCallMonitoring	N/A	Optional	Yes	
Domain	N/A	Required for SIP	Yes	Required to login to AAAD
DomainUsername	N/A	Required for SIP	Yes	Required to login to AAAD
Assignment of contact Types	Required	Required	Yes – contact types can be assigned but not un-assigned	Note: Setting a contact type value to blank will not un-assign the contact type.

Field	AML CS1000	SIP Aura/CM	Update Support	Comments
Assignment of skillsets	Required	Required	Yes – skillsets can be assigned but not un-assigned	When updating skillset assignments, existing priorities can be updated by changing the priority value in the relevant cell. Note: Setting a priority value to blank will not un-assign that skillset.
Skillsets:				
Skillset Name (Name)	Required	Required	Yes	
Skillset Type (Type)	Optional	Optional	No	
Default Activity Code (DfltAC)	Required	Required	Yes	
Threshold Class (ThrsdCls)	Required	Required	Yes	
Call Source Preference (CallSourcePref)	Required	Required	Yes	
Call Age Preference (CallAgePref)	Required	Required	Yes	
Map ACD-DN Number (MapACDDN)	Optional	N/A	N/A	This field is not required for SIP Contact Centers
Out of Service Mode (OutOfSrvMode)	Required	Required	Yes	
Call Request Queue Size (CallReqQueueSize)	Optional	Optional	No	
Flow Control Threshold (FlowCntThreshold)	Optional	Optional	No	
Include Local Node (IncLocalNode)	Optional	Optional	No	
TargetServiceLevel	Optional	Optional	Yes	
ServiceLevelRouting	Optional	Optional	Yes	
Comment	Optional	Optional	Yes	
Route Point (CDNs):				
Name	Required	Required	Yes	
Number	Required	Required	No	
URI	No	Required	Yes	Requires System Restart to take effect

Field	AML CS1000	SIP Aura/CM	Update Support	Comments
Type	Required	Required	No	AML: Must be: Local / Open Queue Local / Open Queue Network / Landingpad / MCDN Network / DNIS Network SIP: must be LOCAL / MCDN Network
Status	Optional	Optional	No	
Call Presentation Class:				
Call Presentation Class Name	Required	Required	Yes	
Presentation Option	Required	Required	Yes	AML: Let Call Ring / Return To Queue / Call Force Delay SIP: Return To Queue / Call Force Delay
Call Force Delay Timer	Optional	Optional	Yes	Only applicable to Call Force Delay option
Return To Queue Wait Interval	Optional	Optional	Yes	
Return To Queue Mode	Optional	Optional	Yes	
After Call, Break for N Seconds	Optional	Optional	Yes	
Answer By Placing DN Call on Hold	Optional	N/A	N/A	
Agent Reserved for Network Call	Optional	N/A	N/A	
DNIS:				
DNIS Name	Required	Required	Yes	
DNIS Number	Required	Required	No	
Service Level Threshold	Required	Required	Yes	
Description	Optional	Optional	Yes	
URI	N/A	Required	Yes	Requires System Restart to take effect

Tools are available from CCMA – Configuration component:



Limitations:

Please note the following limitations:

- Agents and Supervisors: User type change is not supported. Any changes to the UserCapability column for an existing user will be ignored.
- Updating an agent profile that is currently logged in is not supported.
- Un-assigning contact types. Clearing the value of a contact type cell in the spreadsheet does not un-assign the contact type from the agent.

Reference Material:

Please refer to the following customer documentation:

- Avaya Aura® Contact Center Client Administration: *Chapter 28: Configuration Tool procedures*

References and Documentation

The following links contain all of the required standard documentation on the platforms described in this applications note. It is assumed that the reader(s) of this document are fully knowledgeable in the products described and their integration with each other.

Avaya Aura® Contact Center 7.0 Documentation Suite:

Available from <http://www.avaya.com/support>

Call Redirection with Avaya Aura® Contact Center 6.4 – Application Note

<https://downloads.avaya.com/css/P8/documents/100179689>

Coverage to Voice Mail with Avaya Aura® Contact Center – Application Note

<http://downloads.avaya.com/css/P8/documents/100154616>

Appendix A: Migration Event Codes

Summary of possible Windows events, generated by the Database Maintenance application during an AML to AACC migration:

Windows Event ID	Event Description
64141 Purpose: Alert on invalid agent records.	Description: Agent Validation found errors on the agent records. Impact: AACC may not process contacts/display real time reporting correctly. Action: Run RCW/CCMA report using view dbo.UserValidationStatus & run Agent Configuration Validation Tool to fix affected agent data.
64145 Purpose: Alert on duplicate URIs found among Agent, CDN and DNIS records.	Description: Post Migration/Restore duplicate URI found. Impact: User management on CCMA, Agent login, Contact processing, or Real time reporting. Cause: Duplicate URI configured for Agent, CDN and DNIS. Action: 1. Create and run an external report on the CCMS reporting view dbo.URIExceptions in the Namespace: CCMS_STAT for details. 2. Get the list of duplicate URIs from the report. These can be Agents, Supervisor/Agents, CDNs (Route-points) or DNIS. 3. Using System Control and Monitor Utility (SCMU), ensure that the AACC server is fully up and running. 4. Modify the Agent OR Supervisor Agent URI to a value other than the DNIS or CDN URI value that was causing the conflict. This can be done via CCMA using Contact Center Management agent properties. 5. If there are a large number of conflicts the CCMA Consolidated Configuration Tool can be used. <i>Note: It is also possible to change the URI of the CDN or DNIS and resolve the conflict in this way. CDN and DNIS URIs can be updated using the CCMA Consolidated Configuration Tool only. They cannot be modified using the CCMA Configuration web page.</i> 6. Restart AACC server for changes to take effect.
<i>Note: The Agent Configuration Validation Tool is an internal Avaya Tool and can only be run by Avaya Support</i>	

Document Release History

Issue	Date	Author	Description
1.0	29-July-14	WM	Released 1.0 version of AML to SIP migration guidelines application note. Based on AACC 6.4 (SP13)
2.0	18-Dec-14	WM	Released Version 2.0. This is based on AACC 6.4.2 (SP14)
3.0	15-Dec-15	WM	Published version 3.0 for use with AACC 7.0
4.0	31-Jul-18	MM	Published version 4.0 for use with AACC 7.0.x

Glossary of Terms

AACC	Avaya Aura® Contact Center
AAAD	Avaya Aura® Agent Desktop
AES	Application Enablement Server
AMS	Avaya Media Server
API	Application Programming Interface
CCT	Communication Control Toolkit
CDN	Controlled Directory Number
CLAN	Customer Local Area Network
CM	Communication Manager
CTI	Computer Telephony Integration
CR	Change Requests
DN	Directory Number
FP	Feature Pack
GA	Generally Available
HA	High Availability
H/W	Hardware
ICP	Interactive Communications Portal
IM	Instant Message
MC	Mission Critical
MPHA	Machine Preserving High Availability
OCMT	Outbound Campaign Management Tool
RTP	Real-Time Transport Protocol
SES	Session Enablement Manager
SIP	Session Initiation Protocol
SM	Session Manager
SP	Service Pack
SSRS	SQL Server Reporting Services
TLS	Transport Layer Security
VSP	Virtual Services Platform (Avaya)

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