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<td>Chapter 12: Third Party Call Extensions considerations</td>
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  - Avaya ACE Service Provider capacity considerations
  - Avaya ACE application capacity considerations
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  - Downloading ACE documents from support site
- **Chapter 8: Installation and deployment considerations**
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  - Certificate configuration is not updated on Upgrade to 6.2.1
  - ACE Server loses connectivity to Session Manager
  - Configuring NMS locations
  - IWA configuration fails when you backup and migrate from ACE VE 6.2 to 6.2.1
  - Applying the post installation patch
  - Applying patch to prevent the Shellshock security threat
- **Chapter 9: Provisioning considerations**
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  - Unable to edit provider on Avaya ACE GUI
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  - Provisioning of provider in OAMP on Google Chrome does not work
  - Unreliable certificate management
- **Chapter 11: Third Party Call considerations**
  - Third Party Call v2 and v3 do not support the whitespace character for the called party and second participant in the URI for WSDL requests
  - Display on the Calling and Called Party devices is incorrect when a TPC v2 call is initiated from ACE
- **Chapter 12: Third Party Call Extensions considerations**
  - Single step transfer operation on JTAPI provider in ACE succeeds but throws exceptions in the log
  - endCall does not clear all parties in a conference when the consultCall is made with conferenceOnly option
  - Missing notification events during singleStepTransfer
- **Chapter 13: Call Notification considerations**
  - Third Party Call Extensions (v2.0) related Call Notifications (v3.8) for interprovider calls have improper Called Party URI
  - Call Notifications (v3.8) for interprovider calls will have calling party appended with "phone-context=dialstring"
  - ACE sends Disconnect notification to listener when Busy on a call
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Chapter 1: Introduction

This Release Notes document provides a brief description of operational considerations for Avaya Agile Communication Environment™ Release 6.2.x
Introduction
Chapter 2: Avaya ACE GA load

Avaya ACE GA load

Download the Avaya ACE Release 6.2.1 files from the following locations:

- Confluence: https://confluence.forge.avaya.com/display/ACE/Download+Generally+Available

- PLDS:

  ACE000000031 - ACE Core 6.2 FP1 - WASND 8.0.0.5 V3 tar.gz
  ACE000000032 - ACE Core 6.2 FP1 - AMS (Media Server) load
  ACE000000033 - ACE Core 6.2 FP1 - Bare Metal (Non VE) tar.gz
  ACE000000034 - ACE Core 6.2 FP1 - VE load (ova)
  ACE000000035 - ACE Core 6.2 FP1 - IBM WAS for upgrade from 8.0.0.3 to 8.0.0.5
  ACE000000036 - ACE Core 6.2 FP1 - LINUX OS

- To deploy Avaya ACE in a bare metal configuration, download the file linux-iso-6.2.1.iso

- To deploy Avaya ACE in a Virtualized Environment, download the file ACE-6.2.1.29.31212-e51-00.ova
Chapter 3: Software load lineup

The following Avaya ACE Release 6.2.1 application and related software are available as iso images and tar files:

- Red Hat Enterprise Linux 6.0
- Avaya ACE Software and Installation Tools 6.2.1
- IBM WebSphere 8.0.0.5 V3
- Avaya Media Server (MS) 7.5.0.1024
- Avaya ACE upgrade software
- IBM Websphere upgrade software

*Note:*

The Websphere upgrade software is only available as tar file.

You can download the above files from the Avaya support site at [http://support.avaya.com/](http://support.avaya.com/).

Product compatibility

The following table lists the interoperability matrix for Avaya ACE Release 6.2.1.

**Avaya Aura load lineup**

<table>
<thead>
<tr>
<th>Product</th>
<th>5.2x</th>
<th>6.1</th>
<th>6.2 FP1</th>
<th>6.2 FP3</th>
</tr>
</thead>
<tbody>
<tr>
<td>Application Enablement Services</td>
<td>5.2.2 Superpatch 3</td>
<td>6.1 SP 2 (6.1.0.20)</td>
<td>6.2</td>
<td>6.3</td>
</tr>
<tr>
<td>Avaya Aura® Communication Manager</td>
<td>5.2.1 SP 7</td>
<td>6.0.1 SP 2 (00.1.510.1-1 8860)</td>
<td>6.2.0.0.3086</td>
<td>6.3</td>
</tr>
<tr>
<td>Avaya Aura® Session Manager</td>
<td>5.2 SP 3</td>
<td>6.1 SP1 (6.1.1.0.61102 3)</td>
<td>6.3.0.0-76008</td>
<td>6.3.2</td>
</tr>
<tr>
<td>Avaya Aura® System Manager</td>
<td>5.2 SP 2</td>
<td>6.1 SP1.1 (6.1.0.0.7345)</td>
<td>6.2</td>
<td>6.3.2</td>
</tr>
<tr>
<td>Product</td>
<td>5.2x</td>
<td>6.1</td>
<td>6.2 FP1</td>
<td>6.2 FP3</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>--------------</td>
<td>---------</td>
<td>---------</td>
<td>---------</td>
</tr>
<tr>
<td>G450 Branch Gateway</td>
<td>Firmware 30.X.Y</td>
<td>G450 Firmware 31.x.y (31.18.1)</td>
<td></td>
<td>6.3</td>
</tr>
<tr>
<td>Avaya Aura® Presence Services</td>
<td></td>
<td>6.1.5</td>
<td>6.2</td>
<td></td>
</tr>
</tbody>
</table>

**Non Aura products**

<table>
<thead>
<tr>
<th>Product</th>
<th>Supported releases</th>
</tr>
</thead>
<tbody>
<tr>
<td>Communication Server 1000</td>
<td>7.5, 7.6</td>
</tr>
<tr>
<td>Communication Server 2100/ CS2100 Core</td>
<td>SE 13.0</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager</td>
<td>8.6</td>
</tr>
<tr>
<td>Tandberg VCS</td>
<td>3.0</td>
</tr>
<tr>
<td>Avaya NES Contact Center</td>
<td>6.0 or later</td>
</tr>
</tbody>
</table>
## Chapter 4: Supported upgrade paths

The following migrate/upgrade paths are supported:

<table>
<thead>
<tr>
<th>Path</th>
<th>Migrate/Upgrade</th>
<th>Tool to run</th>
<th>What is migrated</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya ACERelease 6.2.0 Standalone (SA) to 6.2.1 High availability</td>
<td>Migrate</td>
<td>migrate62Standalone-62HA.sh</td>
<td>Database data</td>
</tr>
<tr>
<td>Avaya ACERelease 6.2.1 Standalone (SA) to 6.2.1 High availability</td>
<td>Migrate</td>
<td>migrate62Standalone-62HA.sh</td>
<td>Database data</td>
</tr>
<tr>
<td>Avaya ACERelease 6.2.0 HA to 6.2.1 HA</td>
<td>Upgrade</td>
<td>upgrade.sh</td>
<td>Database and configuration data</td>
</tr>
<tr>
<td>Avaya ACERelease 6.2.0 SA to 6.2.1 SA</td>
<td>Upgrade</td>
<td>upgrade.sh</td>
<td>Database and configuration data</td>
</tr>
<tr>
<td>Avaya ACERelease 3.0.x SA to 6.2.1 HA</td>
<td>Migrate</td>
<td>migrate30Standalone-62HA.sh</td>
<td>Database data</td>
</tr>
<tr>
<td>Avaya ACERelease 3.0.x SA to 6.2.1 SA</td>
<td>Migrate</td>
<td>migrate30to621.sh</td>
<td>Database and configuration data</td>
</tr>
<tr>
<td>Avaya ACERelease 3.0.x HA to 6.2.1 HA</td>
<td>Migrate</td>
<td>migrate30to621.sh</td>
<td>Database and configuration data</td>
</tr>
</tbody>
</table>
Supported upgrade paths
Chapter 5:  Issues resolved in release 6.2

The following issues, identified in the Release 3.0.2 Release Notes, have been resolved in Release 6.2.

**Table 1: Installation and deployment**

<table>
<thead>
<tr>
<th>ACECORE-7000</th>
<th>Cluster Suite does not failover when power removed from active host</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-8324</td>
<td>MOSFET security scan warning: unsafe file mode seen in root crontab</td>
</tr>
<tr>
<td>ACECORE-9211</td>
<td>After uninstalling ACE, the ACE software cannot be reinstalled</td>
</tr>
<tr>
<td>ACECORE-9673</td>
<td>Avaya ACE support for PLDS and WebLM licenses</td>
</tr>
</tbody>
</table>

**Table 2: Provisioning**

<table>
<thead>
<tr>
<th>ACECORE-10435</th>
<th>AS5300 is not a supported service provider in ACE release 3.0</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-4740</td>
<td>Limitations on Third party call (v2) and Third party call (v3) when configuring a service provider for SIP REFER or Media Server</td>
</tr>
</tbody>
</table>

**Table 3: ACE GUI and OAM**

<table>
<thead>
<tr>
<th>ACECORE-7228</th>
<th>Database not synchronized alarm is raised when databases are synchronized</th>
</tr>
</thead>
</table>

**Table 4: Third party call**

<table>
<thead>
<tr>
<th>ACECORE-7117</th>
<th>Endcall on held call causes exception</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-7654</td>
<td>Subsequent <code>makecall/makecallSession</code> requests to calling party with identical user and host will not recognize changes to password, port, parameter or header until ACE or service provider is restarted.</td>
</tr>
</tbody>
</table>

**Table 5: Third party call extensions**

<table>
<thead>
<tr>
<th>ACECORE-8117</th>
<th>Third party call extensions service metrics are pegged against third party call (v2) service metrics</th>
</tr>
</thead>
</table>

**Table 6: Call forwarding**

<table>
<thead>
<tr>
<th>ACECORE-6738</th>
<th>Cannot clear call forwarding errors remotely</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-10378</td>
<td>Answer notification lists wrong party as called in a network initiated call resolving to a call forwarded number</td>
</tr>
</tbody>
</table>

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Table 7: Foundation Toolkit

<table>
<thead>
<tr>
<th>Issue ID</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-7482</td>
<td>VES create dialog method does not throw exception for invalid URI</td>
</tr>
<tr>
<td>ACECORE-7484</td>
<td>Subsequent requests are lost when the session manager TLS port is 5061</td>
</tr>
<tr>
<td>ACECORE-7489</td>
<td>The media service <code>generateDtmf()</code> throws <code>UnsupportedOperation</code> exception</td>
</tr>
<tr>
<td>ACECORE-7491</td>
<td>Foundation toolkit does not send a 408 Request Timeout response</td>
</tr>
<tr>
<td>ACECORE-8743</td>
<td>Foundation toolkit client application registrations and subscriptions may not be automatically renewed after two consecutive failures</td>
</tr>
<tr>
<td>ACECORE-10348</td>
<td>Virtual endpoint service DialogTerminationCauses are incorrect in some circumstances</td>
</tr>
</tbody>
</table>

Table 8: Message Drop and Message Blast

<table>
<thead>
<tr>
<th>Issue ID</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-4255</td>
<td>Message Drop fails intermittently when both originating and terminating lines are off the same CS1000 PBX</td>
</tr>
</tbody>
</table>

Table 9: Avaya Aura service provider

<table>
<thead>
<tr>
<th>Issue ID</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-7219</td>
<td>ACE nad AES sporadically lose connectivity</td>
</tr>
<tr>
<td>ACECORE-7431</td>
<td>Limitations on Third party call (v2) operations when configuring an Avaya Aura service provider for SIP REFER</td>
</tr>
<tr>
<td>ACECORE-7769</td>
<td>Communication between ACE and AES fails after switchover or restart of AES</td>
</tr>
<tr>
<td>ACECORE-11847</td>
<td>Upgrade to ACE 3.0.2 does not preserve certificates</td>
</tr>
</tbody>
</table>

Table 10: CS 1000 service provider

<table>
<thead>
<tr>
<th>Issue ID</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-4262</td>
<td>The <code>getCallForwarding</code> WSDL operation returns truncated Avaya CS1000 phone numbers after ACE restart</td>
</tr>
<tr>
<td>ACECORE-4491</td>
<td>Calling party does not hear busy/invalid number tones during <code>makecall/makeCallSession</code> using third party call(v3) with SIP.</td>
</tr>
<tr>
<td>ACECORE-10695</td>
<td>CS1000 release 6.0 is now end of life and is no longer supported</td>
</tr>
</tbody>
</table>

Table 11: Turret

<table>
<thead>
<tr>
<th>Issue ID</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACECORE-6655</td>
<td>No provider out-of-service alarm for turret provider configured with wrong IP</td>
</tr>
<tr>
<td>ACECORE-4096</td>
<td>Users will be considered as non Avaya users if they have a duplicate IBM Lotus Sametime contact ID in their ACE user profile</td>
</tr>
</tbody>
</table>
Issues resolved in release 6.2
Chapter 6: Capacity and traffic considerations

Capacity model considerations

Determining the Avaya ACE system capacity model is based on the following:

- Number of devices that will be integrated with Avaya ACE for each user (for example, telephone, video device, soft clients)
- Number of users integrated with Avaya ACE
- What the users are doing which would require Avaya ACE resources
- How often they are doing it
- Capacity of the communication network

Actual traffic rates may vary depending on the type of hardware and operating system Avaya ACE is installed on, network conditions and other non-ACE related applications running on the Avaya ACE server. A typical Avaya ACE deployment with 15000 Avaya ACE users using ThirdPartyCall (v2), Presence and Call Notification (v2.3) with a 2-3 minute call duration can expect their traffic capacity to fall in the 6-15 calls per second range. Each Avaya ACE deployment should work with Avaya Professional Services (APS) to have an architect identify the addressable capacity for the deployment.

Avaya ACE capacity considerations

Avaya ACE can support a maximum of 45,000 SIP subscriptions (these are consumed by either SIP SUBSCRIBE or SIP REFER transactions) and 30,000 SIP call legs (these are consumed by SIP INVITE transactions and released by SIP BYE transactions). These limitations are used to determine how many SIP/TR87 contacts an Avaya ACE user can have and how many concurrent SIP calls can be made.

Avaya ACE on VMware capacity considerations

Avaya ACE on VMware supports a maximum of 15000 users with call rate of 18 call per second where each call consists of make call session, presence update and a call notification.
Avaya ACE Foundation Toolkit capacity considerations

Foundation Toolkit supports a call rate of 20 calls per second.

Avaya ACE Service Provider capacity considerations

**SIP B2BUA**

The maximum verified sustainable rate of Avaya ACE-initiated SIP B2BUA calls is 18 calls per second with 15,000 Avaya ACE users using the ThirdPartyCall (v2) web service or ThirdPartyCall (v3) web service with sufficient media server capacity and an average call duration of 3 minutes.

**TR87 (CTI)**

The maximum verified sustainable rate of Avaya ACE-initiated TR87 (CTI) calls is 18 calls per second with 15,000 Avaya ACE users using ThirdPartyCall (v2), Presence and Call Notification (v3.2 or v3.8) web services and an average call duration of 3 minutes.

**JTAPI**

The maximum verified sustainable rate of Avaya ACE-initiated calls is 6 calls per second on JTAPI (all service providers combined) per Avaya ACE server with 5000 Avaya ACE users using Third Party Call (v2) and Third Party Call extensions (v2) web services and an average call duration of 3 minutes.

**Cisco Unified Communication Manager using AXL**

The maximum verified sustainable rate of Avaya ACE-initiated AXL calls is 18 calls per second per Avaya ACE server with 15,000 Avaya ACE users using the Call Forward web services and average call duration of 3 minutes.

**Avaya NES Contact Center**

The maximum verified sustainable rate of Avaya ACE-initiated calls is 3 calls per second on the Avaya NES Contact Center using Third Party Call (v2), Call History and Call Notification (v3.8) web services using 5000 Avaya ACE-users with average call duration of 3 minutes.
Avaya ACE application capacity considerations

Message Drop and Message Blast

The Message Blast traffic rate is one call per second, with a maximum of 180 concurrent blast participants and a call duration of three minutes.

Click-to-Dial with Message Drop, Message Drop and Leave:

• 4 Click-to-Dials per second, where 0.5 calls per second is a Message Drop, 1.5 calls per second is a Message Drop & Leave and the call duration is 2.5 minutes.

• The traffic mixture for this model is:
  - CS 1000 Click-to-Dial: one call per second, where 0.25 calls per second is a Message Drop and 0.25 calls per second is a Message Drop & Leave.
  - Avaya Aura Click-to-Dial: 1 call per second call, where 0.25 calls per second is a Message Drop and 0.25 calls per second is a Message Drop & Leave.
  - Turret Click-to-Dial: 2 calls per second, where 1.0 call per second is a Message Drop & Leave.
Capacity and traffic considerations
Chapter 7: Documentation considerations

The Avaya Agile Communication Environment™ (ACE) software ships with the documentation available when the software is built. The following documents have been updated for the official ship date of Avaya ACE™ release 6.2.

- Avaya Agile Communication Environment™ Overview and Specification
- Avaya Agile Communication Environment™ Planning and Installation (NN10850-004)
- Avaya Agile Communication Environment™ Web Services (NN10850-007)
- Avaya Agile Communication Environment™ Troubleshooting (NN10850-026)

The following documents have been added:

- Avaya Agile Communication Environment™ User and Security Administration (NN10850-010)

The following document has been renamed:

from Avaya Agile Communication Environment™ Administration (NN10850-005)
to Avaya Agile Communication Environment™ Service Provider Administration (NN10850-005)

Updated documents can be identified by the issue number on the front cover, located after the document number. The second number indicates how many times the document has been reissued within a software release. For example, the original document is labelled XX.01. The second issue is labelled XX.02.

Current documents can be obtained from the Avaya support site at https://support.avaya.com. See Downloading ACE documents from support site on page 23.

---

**Downloading ACE documents from support site**

Use the following procedure to download ACE documents from the Avaya support site.

**Procedure**

1. On your web browser, enter the Avaya support site URL, [https://support.avaya.com](https://support.avaya.com).
2. Click **Downloads & Documents**.
3. Enter Avaya Agile Communication Environment in the Enter Your Product Here field.

4. In the Choose Release drop down menu, select the release.

5. In the Select the content type section, select Documents and click Enter.

6. In the resultant page, do one of the following:
   - Filter the documents displayed based on the type of document you require. To filter, select the type of document you want from the list.
   - Click Select All to display all the documents pertaining to the release.

7. From the documents displayed, click the document you want.
Chapter 8: Installation and deployment considerations

This section contains information about known issues related to Avaya Agile Communication Environment™ (ACE) installation and deployment.

Avaya ACE installer claims a port is in use when it is not

While Installing ACE, the installer displays an error message stating that a specific port is already in use and the installation fails even though the port is free. This happens if the ACE server has a connection established with a remote server and the remote server port matches with the specific port used by ACE.

**Expected result:**
Installation should not fail as the port is used on the remote server and not the ACE server.

**Actual result:**
Installation fails with the error that a specific port is already in use.

**Tracking number**
ACECORE-12885/ACE-11687

**Impact**
ACE cannot be installed if some port needed by ACE is used by any remote server to connect to ACE server.

**Workaround**

1. Reboot the ACE server.
2. Run the command: `netstat -nat | grep port`
   
   Ensure that the command returns no entries. This implies that the port is not assigned to any other service.
3. Restart the installation.
Certificate configuration is not updated on Upgrade to 6.2.1

Tracking number
ACE-12530

Description
Applicable when you upgrade from ACE 6.2.0 or ACE-6.2.1-SNAPSHOT to ACE 6.2.1. After the upgrade is completed, user-provisioned certificates from the previous release are not used. This is applicable for System Manager certificates and for self-signed certificates.

Impact
Service providers using certificates will go OOS after upgrade.

Workaround
Run the `/opt/IBM/WebSphere/AppServer/profiles/AppSrv01/bin/wsadmin.sh -user <was_admin_username> -password <was_admin_password> -f/opt/avaya/ace/bin/modifySSLConfig.py <host_type> <certificate_alias>` command in the Linux console of your ACE machine.

Here, was_admin_username is the IBM Websphere admin user name,

was_admin_password is the password for the IBM Websphere admin user

host_type is the Avaya ACE host type, as present in /etc/host_type

certificate_alias is the alias of the certificate to be used. The certificate is present in CellDefaultKeyStore. To view the certificates, on IBM Websphere GUI navigate to Security > SSL Certificate, Key Management > Key Stores, and Certificates > CellDefaultKeyStore > Personal certificates.

Note:
If you have configured the AAFT service provider, delete, and reconfigure the certificate from the Avaya ACE GUI.

ACE Server loses connectivity to Session Manager

Description
The ACE SIP component loses connectivity to Session Manager, though this is not propagated back. An alarm is not raised on ACE OAM. The SIP container threads go into a dead lock. This
scenario occurs when ACE connects to CS1k through Session Manager, which acts as a proxy.

**Tracking number**
ACE-12319

**Impact**
All applications using an ACE SIP provider will not be able to make any call related operation.

**Workaround**
You must install an additional IBM Websphere Application server patch. You must perform the following procedure post installation, post upgrade, and post an inrelease upgrade.

1. Download WASPatch621v3.tar.gz from PLDS. The ID is ACE00000039.
2. Create the /opt/avaya/was directory using the `mkdir /opt/avaya/was` command. Upload WASPatch621v3.tar.gz to this location.
3. Type the `tar -xvf WASPatch621v3.tar.gz` command to explode the tarball.
4. Type the `./installPatch6.2.1.sh` command to install the patch.

You must apply this patch to both the standalone and high-available deployments. For high availability scenarios, you must apply the patch to Node A and then Node B.

---

**Configuring NMS locations**

**Description**
While configuring NMS locations, non-numeric characters are accepted for Port Number.

**Tracking number**
ACE-12529

**Impact**
While configuring NMS locations, non-numeric characters are accepted for Port Number and notifications are not sent to the NMS locations.

**Workaround**
None. You must enter a numeric value for NMS port number.
IWA configuration fails when you backup and migrate from ACE VE 6.2 to 6.2.1

Description
IWA configuration fails when you backup and migrate from ACE VE 6.2, in which IWA is configured, to 6.2.1.

Tracking number
ACE-12532

Impact
When you upgrade from ACE VE 6.2, in which IWA is configured, to 6.2.1, the keytab file is not present in the backup file. As a result, IWA configuration fails.

Workaround

- Ensure that you backup the keytab file. Transfer the keytab file to a local computer before you power off the 6.2 virtual machine.
- Deploy the 6.2.1 OVA file.
- Transfer the keytab file from the local computer to the 6.2.1 OVA. Copy the keytab file at the same location as before.
- Upgrade to ACE 6.2.1.

Applying the post installation patch

⚠️ Note:
The following procedure is applicable only for users with Israel Daylight Saving Time.

About this task
You must perform the following procedure after you install or upgrade Agile Communication Environment™ 6.2.1.

To install Israel Daylight Saving Time settings on ACE 6.2.x, an additional RHEL rpm is required. You must also update the Java version.

Procedure

1. Download ACE_DST_Updatev1.tar.gz from PLDS. The ID is ACE000000040.
2. Create the /opt/avaya/dst directory, type the `mkdir` command.

3. Upload `ACE_DST_Updatev1.tar.gz` to /opt/avaya/dst.

4. Type the `tar -xvf ACE_DST_Updatev1.tar.gz` command to explode the tarball.


6. Copy the zip file to /opt/avaya/dst/ACE_DST_Updatev1 and type the `unzip jtzu-1.6.13d.zip` command to unzip `jtzu-1.6.13d.zip`.

7. Type the `cd /opt/avaya/dst/ACE_DST_Updatev1` command to change the directory.

8. Type the `./UpgradeACEDSTv1.sh` command to install the DST patch.

   **Note:**
   You must apply this patch to both standalone and high-available deployments. For high availability deployment, you must apply the patch to Node A and then Node B.

### Applying patch to prevent the Shellshock security threat

#### About this task
You must perform this task to prevent the security threat posed by Shellshock Bash injection.

**Note:**
You must apply this patch to both the standalone and high-available deployments. For high availability deployments, you must apply the patch to Node A and then to Node B.

#### Procedure

1. Download `ACEShellShockFix.tar.gz` from the PLDS website.
   The ID is `ACE000000050`.

2. Login to the Agile Communication Environment™ terminal by using PUTTY or a similar software.

3. Do one of the following:
   - Use sysadmin credentials to login to root, and enter `su` -.
   - If you do not use sysadmin credentials, login to the terminal using root credentials.
4. Go to cd /opt/avaya.

5. Upload the Agile Communication Environment™ tar ball to cd /opt/avaya.
   Use WinSCP or a similar client to upload the file.

6. Run the tar -xvf ACEShellShockFix.tar.gz command.

7. Go to cd /opt/avaya/ACEShellShockFix.

8. Run the ./fixShellShock.sh command.
Chapter 9: Provisioning considerations

This section contains information about known issues related to Avaya Agile Communication Environment™ (ACE) provisioning.

Some SIP interfaces do not use ssl

Currently some of the SIP interfaces on ACE do not use secure signaling.

The requirement for using secure signaling is applicable for the AS5300 and CS2100 network elements that ACE interfaces with.

It is not applicable to the Cisco Unified Communications Manager since it is not an Avaya product.

Tracking number
ACECORE-13417/ACE-11717

Impact
You cannot add a white space character in the URI request for the following parameters:

• called party parameter of a Third party Call (v2)
• second participant of a Third Party Call (v3)

Workaround
None

Unable to edit provider on Avaya ACE GUI

After posting the list of provisioned ACE service providers, unable to edit provider on Avaya ACE GUI by clicking the enabled Edit button, regardless of whether the provider status is In Service or Disabled.

In addition, after posting the edit page for some providers, the Update button on said page is enabled despite not having the ability to edit any settings.
Tracking number
ACECORE-12988/ACE-11595

Impact
The user thinks he can update the provider configuration but some provider configuration fields are not editable. This would misleading for user.

Workaround
None
Chapter 10: Avaya ACE GUI and OAM considerations

There are no known issues related to Avaya Agile Communication Environment™ (ACE) graphical user interface (GUI) and operations, administration, and maintenance (OAM) tasks performed using the ACE GUI.

Provisioning of provider in OAMP on Google Chrome does not work

Although the provisioning of a provider (AURA - SIP TLS) on the oamp is successful, the provider does not appear on OAM.

**Tracking number**
ACECORE-13289/ACE-11532

**Impact**
None.

**Workaround**
None. Google Chrome is not a supported browser. For a list of supported browsers, see, *Avaya Agile Communication Environment™ Planning and Installation* (NN10850–004).

Unreliable certificate management

ACE OAMP (GUI) became inaccessible after a certificate deletion from the GUI.

**Scenario:**

- Administrator logs on to ACE console via https://<ip_or_fqdn>:9449/oamp>
- Navigates to Security > Certificate management > System Manager Certificates.
- Clicks Delete on the page.

**Result:**
Page goes inaccessible

**Tracking Number**
ACE-11659

**Impact**
Administrator will not be able to use the Operation and Management console of ACE.

**Workaround**
Perform the following steps:

1. Connect to terminal session of the ACE machine. For example, through a PuTTY session.
2. Enter the credentials.
3. Type the following commands:
   a. cd /opt/avaya/ace/bin
   b. ./deleteCertificates.sh <was_admin_id> <was_admin_password>

   Where, *was_admin_id* is the Websphere administrator user name and *was_admin_password* is the associated password.

The above shell script displays success message on the console.

Key output statements to be looked at on the console are:

ACE ssl configuration is modified to default
SMGR CA certificate and SMGR personal certificate are deleted
Chapter 11: Third Party Call considerations

This section contains information about known issues related to Avaya Agile Communication Environment™ (ACE) Third Party Call web service.

---

**Third Party Call v2 and v3 do not support the whitespace character for the called party and second participant in the URI for WSDL requests**

For the called party in Third Party Call v2 and the second participant in Third Party Call v3, having a %20 (whitespace) in the URI will cause the URI to be recognized as invalid. For example:

<loc:calledParty>sip:steve@avaya.com?user=dave%20lee</loc:calledParty>

**Tracking number**
ACECORE-7030/ACE-11633

**Impact**
Customers will currently be unable to have whitespace for called party in Third Party Call v2 and second participant in Third Party Call v3 in the URI for WSDL requests.

**Workaround**
No workaround at this time.

---

**Display on the Calling and Called Party devices is incorrect when a TPC v2 call is initiated from ACE**

**Tracking number**
ACE-11516

**Impact**

The display on the Calling and Called Party devices is incorrect when a TPC v2 call is initiated from ACE. This is applicable for the TR/87 provider using AES / CS1K configuration.
Third Party Call considerations

This issue affects:

• Calls between SIP client to H.323 client. Here, only the SIP client has the issue.
• Call forward, redirect, consult & single step transfer scenarios as well.

This issue does not affect:

• Calls between a H.323 client to a SIP client
• Calls between two H.323 clients
• TPCv2 makeCall (H.323 to H.323) scenarios

**Workaround**

None
Chapter 12: Third Party Call Extensions considerations

This section contains information about known issues related to Avaya Agile Communication Environment™ (ACE) Third Party Call Extensions web service.

---

Single step transfer operation on JTAPI provider in ACE succeeds but throws exceptions in the log

**Scenario:**
- Set up CCM JTAPI service provider
- Make Call from A to B, send Answer Call on B
- Single Step Transfer B to C
- all operations work, but exception is thrown in cmf/JtapiCcm.log

**Tracking number**
ACECORE-13293

**Impact**
There is no functionality impact but exception gets printed in the log.

**Workaround**
None

---

endCall does not clear all parties in a conference when the consultCall is made with conferenceOnly option

In a consultCall operation using conferenceOnly as the consultOption with stations A, B and C.

A - CS1k, B - Aura SIP, C - Aura SIP.
ACE **endCall** does not clear all parties in the conference, leaving A and C still on call.

**Scenario:**

Perform the TPCv2 operation, **makeCall**, between Party A(CS1k) and Party B(Aura) (CallID 1) using ACE.

Perform the **TPCExtensions** operation, **consultTransfer** with conferenceOnly from Party B(Aura) to Party C(Aura)(CallID 2)

Perform the **TPCExtensions** operation, **consultComplete**

Perform **endCall**(CallID 1)

**Tracking number**

ACECORE-13400/ACE-11572

**Impact**

Applications using ACE **consultTransfer** or **consultComplete** using a CS1k DN as the Calling party will see two parties still on call (A and C) after using ACE **endCall**.

**Workaround**

None

---

**Missing notification events during singleStepTransfer**

**Setup:**

- Configure Aura TR87 service provider for **singleStepTransfer** and for callID-based notifications
- Configure CS1000 v2 SIP service provider for TPCv2 **makeCall**
- H.323 endpoints: DN2, DN3
- CS1000 endpoint: DN1
- Make TPCV2 **makeCall** between DN1 and DN2 and start callID based notifications **startCallNotification**
- Answer DN2 -Do **consultationCall** (DN2 to DN3), do not answer DN3
- Do **consultationComplete**, do not answer DN3
- Do an **endCall** using the callID from notification event for DN2

**Expected result:**

After **endCall**, entire call should be terminated and there should be no active call on DN1, DN2 and DN3
Actual result:

Call is terminated on DN1 but call is still active between DN2 and DN3. When DN2 is hung up DN3 still rings.

**Tracking number**
ACECORE-13315/ACE-4

**Impact**
The `endCall` operation does not terminate the entire call.

**Workaround**
Manually disconnect the call from the phone.
Third Party Call Extensions considerations
Chapter 13: Call Notification considerations

This section contains information about known issues related to Avaya Agile Communication Environment™ (ACE) Call Notifications web service.

Third Party Call Extensions (v2.0) related Call Notifications (v3.8) for interprovider calls have improper Called Party URI

Call Notifications regarding TPCV2 Extension services (ConsultCall, CallForward and SingleStepTransfer) have HLOC prefixed to called party number and missing URI scheme (tel).

Tracking number
ACECORE-6295/ACE-11774

Impact
For inter provider calls, ConsultCall, CallForward and SingleStepTransfer operations fail.

Workaround
Add Reverse Translation rule which removes HLOC and prefixes URI Scheme.

1. On the Avaya Avaya ACE GUI, under the Configuration menu, select Service Providers.
2. Select the service provider and then click Rules.
3. Select the Reverse Transformation check box and complete the required fields.
Call Notifications (v3.8) for interprovider calls will have calling party appended with “phone-context=dialstring”

When subscribing for notifications (DN based or callid based) using the default address direction Called, the calling party notifications have “;phone-context=dialstring” appended to the URI.

Tracking number
ACECORE-6294/ACE-11886

Impact
Call Notification (v3.8) display contains extra characters.

Workaround
No workaround at this time.

ACE sends Disconnect notification to listener when Busy on a call

When a call is made to a DN which is Busy (already in another call), Busy followed by Disconnect notifications are sent to the calling party listener. However, after about 25 seconds a Disconnect notification is also sent for the called party listener even though that party never entered the call. This Disconnect event has the call ID of the failed call.

Tracking number
ACE-11684

Impact
The application listening to events on the called party might be confused with the unexpected Disconnect event and might consider it to be a Disconnect for the ongoing call.

Workaround
Applications should consider the call ID field along with the type of event.
**TPCv2 and Call Notification v3.8 have issues when forwarded DN is busy on Aura TR/87 and SIP one-X clients**

In a **CallForward** scenario with Party B forwarded to Party C with Call Notification subscriptions to all parties, when an ACE **makeCall** is made between Part A and Party B, **getCallInformation** reports **CallingPartyBusy** as the Call termination reason.

**Tracking number**
ACECORE-12399/ACE-11519

**Impact**
Application listeners trying to verify the status of the call will receive wrong notifications about the status of Party A.

**Workaround**
None

---

**Unreachable event not received when called party is unreachable in Session ID based Call Notification v3.8**

On Aura TR87 and SIP One-X clients, an ACE MakeCall to an unreachable number using Session Id based embedded TPC notifications, an unreachable participant event is not notified to the listener.

**Tracking number**
ACECORE-12401/ACE-11604

**Impact**
Application listeners will not receive the NotReachable CallEvent

**Workaround**
None.
Call ID Participant based Call notifications contain multiple participant entries for PSTN numbers

**redirectCall** and **consult Call**

Call ID Participant based Call notifications contain multiple participant entries for PSTN numbers

**Tracking number**
ACECORE-13235/ACE-11587

**Impact**
There will be an additional participant in the call that might confuse to the Applications call modelling.

**Workaround**
None.

---

Duplicate notifications received when using H.323 clients in **singleStepTransfer**

In an Aura TR/87 ACE call with **singleStepTransfer** and CallId based notifications, after **singleStepTransfer**, there is a set of duplicate notifications sent to the client, with the new **participant**(transferred-to) as **CallParticipantInitial**.

**Scenario:**
Third Party Call v2**makeCall** (A(Aura) to B(Aura))

**startCallNotification** (addresses: callID, criteria: Participant)

Make sure call is connected

**singleStepTransfer** (B to C)

Make sure call is connected

**endCall**

**stopCallNotification**

**Tracking number**
ACECORE-13308/ACE-11675
Duplicate notifications received when using H.323 clients in singleStepTransfer

**Impact**
Client application listeners will see a set of duplicate notifications when Party C rings.

**Workaround**
No work around yet.

Applications can have a logic to ignore events if they are the same as the one received previously.
Call Notification considerations
Chapter 14: Call Forwarding considerations

This section contains information about known issues related to Avaya Agile Communication Environment™ (ACE) Call Forwarding web service.

Incomplete notifications for Call forwarding or redirectCall

Call ID based notifications for the call forwarded or redirected SIP calls are not supported. This means that the notification events will not contain the forwarded To party information.

Tracking number
ACECORE-13187/ACE-11570

Impact
Any SIP call forwarded calls subscribing for call-id based notifications will not get the forwarded To party information in the Call notification events.

For example:
- DN2 has set call forward or redirect to DN3
- DN1 calls DN2

Call notification events contain DN2 instead of DN3 that answered the call.

Note:
Avaya ACE does not support Call forwarding for SIP providers. The scenario mentioned here is when the Call Forwarding is done at the PBX level itself. For more information on services supported by various service providers, see Tables of Avaya ACE supported services by service provider type in Avaya Agile Communication Environment™ Web Services (NN10850–007).

Work around
None.
Call Forwarding considerations
Chapter 15: Presence considerations

There are no known issues related to Avaya Agile Communication Environment™ (ACE) Presence web service.
Chapter 16: Endpoint display considerations

This section contains information about all known issues related to endpoint displays.

Calling party phone does not display called party number in a TPCv2 makeCall

In an Aura TR/87 ACE makeCall between a CS1k DN (Calling party) and Aura DN (Called party), the display of Calling Party (CS1k) does not get updated properly.

Scenario:
makeCall from A (CS1k) to B (Aura), make sure call is connected.
endcall

Tracking number
ACECORE-13258/ACE-11716

Impact
Calling party phone does not display the called party number in ringing and connected state.

Workaround
None
Endpoint display considerations
Chapter 17: Foundation Toolkit considerations

This section contains information about known issues related to Avaya Agile Communication Environment™ (ACE) Foundation Toolkit.

⚠️ Note:
If the Foundation Toolkit is deployed with Avaya Aura® Communication Manager, the Communication Manager must be at release 6.0.1 or higher.

B2B Routing call fails when the calling and called party are sequenced to AAFT and CM

User setup:
calling party originating sequence: aaf client app + CM
calling party terminating sequence: CM
called party originating sequence: CM
called party terminating sequence: CM + aaf client app
AAFT client application uses B2B routing service.
When calling party calls called party, it gets a busy signal, or the call is setup briefly, then terminated.

Tracking number
ACECORE-10586/ACE-11601

Impact
The MAS media server does not support appending to a recording and therefore, this functionality cannot be exercised using the Foundation Toolkit API.

Workaround
None
Chapter 18: Avaya Aura® service provider considerations

This section contains information about known issues related to the Avaya Aura® service provider.

---

Avaya clients do not play Busy tone when a third party call is made to a busy line

Avaya clients do not play busy tone when a Third Party Call is established to a busy terminating line. This issue only affects Avaya service providers configured to use the "Invite;Answer;Refer" Make Call Sequence. Scenarios in which the terminating side of the call provides in-band busy treatment will also be unaffected.

Tracking number
ACECORE-4933/ACE-11631

Impact
The problem manifests as follows:

- Third Party Call is initiated through Avaya ACE, between an Avaya client and a busy subscriber.
- The originating Avaya client rings and the originating user answers. Call immediately drops and the originating client's display goes to an idle state, with no destination busy notification presented to the user.
- Desired behavior is that the Avaya client should present busy treatment to the user, in the form of a busy tone and optionally a message on the Avaya client's screen.

Workaround
None
SIP calls fail when AAFT and AURA SIP providers are both provisioned

SIP calls fail with Indeterminate originating entity error when AAFT and AURA SIP providers are both provisioned.

Tracking number
ACECORE-12641/ACE-11636

Impact
Network initiated SIP calls will fail for Aura SIP provider when an Aura FT provider is also provisioned on ACE. Though ACE initiated calls work fine it is recommended to disable Aura FT provider while using Aura SIP provider.

Workaround
• Disable Aura FT provider while making SIP calls using Aura SIP provider.
• When using AAFT applications ensure that the entity-link created from Session Manager uses the port 5063 on ACE.

Incorrect phone display when receiving call from thirdPartyCallController in a TPCv2/v3 makeCall

When a TPC call is made from ACE using Aura v2 SIP TLS service provider, calling party shows an incorrect display name.

Tracking number
ACECORE-13092/ACE-11520

Impact
It will be not known from where the call has come from.

Workaround
None. CM ticket raised for this issue.
First call after a CTI link has been restored always fails when the call is made from Browser add-in

First call fails from the browser add-in when CTI link in CM is brought down. End user must launch a second call before the functionality is restored.

Tracking number
ACECORE-13038/ACE-11580

Impact
First call will always fail after the CTI link is down.

Workaround
None.

Call forwarding and redirectCall have issues when using H.323 clients

When a CallForward is set on Party B to Party C and an ACE TPC v2 makeCall is made between Party A and Party B, the calling party display phone shows Party B as Called instead of Party C. The Called party, Party C, display phone shows Party B as Calling instead of Party A.

Tracking number
ACECORE-13133/ACE-11616

Impact
ACE applications and OneX soft or hard clients will be notified incorrectly about the other connection involved in the call.

Workaround
None.
Chapter 19: CS 1000 service provider considerations

This section contains information about known issues related to the Avaya Communication Server 1000 service provider.

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No busy notification for a network initiated call to a monitored DN on a CS 1000 TR/87 provider

If a user makes a network initiated call to a monitored DN, the call fails to establish when the DN is busy. However, no Call Notification Busy event is sent to the application monitoring this DN. Instead, a disconnect event is sent when the call fails.

This issue only occurs on the CS 1000 TR/87 provider.

Tracking number
ACECORE-7681

Impact
The application does not receive the Busy event, and does not know why the call was terminated.

Workaround
No workaround at this time.

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Failed to CallSender when ACE SIP Interface makecall from CS1K set to MM then CallSender to BCM set

Configuration:

1. Configure SIP trunk from CS1K7.6 to SM
2. ACE SIP Interface

Procedure:
1. Construct and send makeCallSession request to ACE to make call between CS1K DN and Modular Messaging

2. CS1K user logs in its mailbox on MM to check message then presses 88 to make call back to the called through CallSender application to BCM EndPoint that had left a voicemail

3. BCM DN answers the call.

Observation: Call is dropped immediately.

**Tracking number**
ACECORE-13161/ACE—11678

**Impact**
Can not initiate a call back through ACE between CS1K DN & MM to reach back to the caller.

**Workaround**
This issue doesn't happen when CS1K DN calls directly to MM without ACE.

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**Multiple transfers on a CS1000 provider results in inconsistent behavior**

**Description**
When multiple transfers are made on a CS 1000 service provider, the original participants of the call are shown as active participants in the call, even though they are no longer in the call.

**Tracking number**
ACE-12457

**Impact**
If the customer is using the GoogleApps integration feature of ACE with Esna software and if there are multiple transfers performed in the same call, the ILync client pop ups are inconsistent and end call operation from ILync client fails.

**Workaround**
None