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Avaya Solution & Interoperability Test Lab

Application Notes for CallTech CTMail® Unified Messaging system with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the CallTech CTMail® Unified Messaging system to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

CallTech CTMail® Unified Messaging system is an integrated automatic operator system, voicemail, audio text, fax server, Interactive Voice Response System and the most modern unified messaging server (fax, voicemail, email) with voice recognition and voice text.

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

1. Introduction

The overall objective of this interoperability compliance testing is to verify that the CallTech CTMail® Unified Messaging system software can interoperate with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3. CallTech CTMail® Unified Messaging system (herein referred to as CTMail) connects to Avaya Aura® Communication Manager via Avaya Aura® Session Manager.

CallTech CTMail® Unified Messaging system is an integrated automatic operator system, voicemail, audio text, fax server, Interactive Voice Response System and the most modern unified messaging server (fax, voicemail, email) with voice recognition and voice text.

CallTech CTMail® Unified Messaging system is integrated with telephone systems, digital, analog and IP as well as the majority of mail servers on the market.

An assumption is made that Avaya Aura® Session Manager, Avaya Aura® System Manager and Avaya Aura® Communication Manager are already installed and basic configuration has been performed.

Only steps relevant to this compliance test will be described in this document. In these Application Notes, the following topics will be described:
- Avaya Aura® Communication Manager – A SIP trunk configuration between Avaya Aura® Communication Manager and Avaya Aura® Session Manager.
- Avaya Aura® Session Manager – A SIP trunk configuration between Avaya Aura® Communication Manager and Avaya Aura® Session Manager.
- CallTech CTMail® Unified Messaging system – IP address configurations for CTMail Automatic Operator, CTMail Voicemail, CTMail IVR System, and Fax Server (FAXMail®)

2. General Test Approach and Test Results

The general test approach was to manually place PSTN calls to CTMail users or direct calls to CTMail and verify the welcome prompt for each feature (CTMail Automatic Operator, CTMail Voicemail and CTMail IVR System). During the compliance test, an analog port on Communication Manager was connected to an external fax machine to send/receive fax to/from CTMail.

For serviceability testing, physical and logical links were disabled/re-enabled, Avaya Servers were reset and CTMail was restarted.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by

DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability tests. The focus of the compliance testing was primarily on verifying the interoperability of CTMail in an Avaya telephony environment that includes Communication Manager, Session Manager, and various endpoints. During the CTMail solution test, the following features were verified:
- CTMail Automatic Operator
- CTMail Voicemail
- CTMail IVR System
- Fax Server (FAXMail®)

2.2. Test Results

All executed test cases passed successfully. CTMail successfully interoperated in an Avaya telephony environment. For testing CTMail, there are two ways (many-to-one and one-to-many) to accomplish the test.
- The first method is utilizing Vector Directory Numbers (VDNs) and a route-to feature in the vector. CTMail will inspect the History-Info header in the SIP trace coming into CTMail to find which Dialed Number Identification Service (DNIS) number the call is coming from. Depending upon the DNIS number, the call is redirected to an appropriate destination system mentioned in **Section 2.1**.
- The second method is providing a different extension for each system mentioned in **Section 2.1**. Thus a call coming into the CTMail will have a different called number. In this case, the History-Info header will have the called number.

During the compliance test, the member utilized the latter method to accomplish tasks.

2.3. Support

Technical support for CTMail can be obtained through the following:
- support@calltechsa.com
- +57 (1) 6 35 65 35 Ext: 911

3. Reference Configuration

Figure 1 illustrates a configuration used during the compliance test. For completeness, Avaya 96xx Series SIP IP Telephones on the Avaya S8300D Server and Avaya G450 Media Gateway side have been registered to Session Manager. These endpoints are included in Figure 1 to demonstrate calls between the SIP IP telephones that are going through the IP/PRI trunk between two Communication managers. The solution described herein is also extensible to other Avaya Servers and Media Gateways.

Note: Avaya S8300D Server with an Avaya G430 Media Gateway was included in the test only to provide an inter-switch scenario. Thus, there will not be any discussion on configuring Avaya S8300D Server with an Avaya G430 Media Gateway.
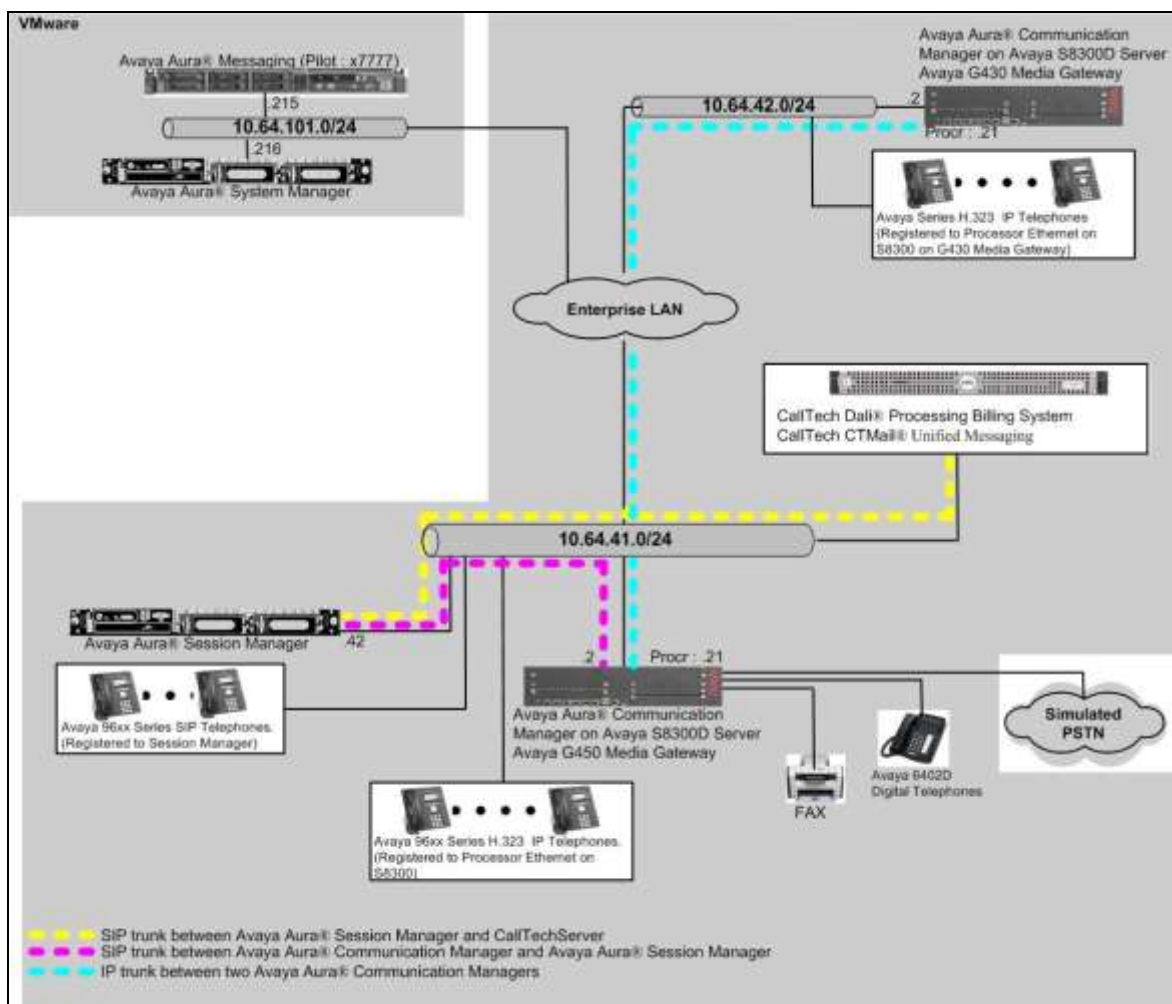
Figure 1. Test configuration of CTMail with Avaya Aura® Communication Manager

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya S8300D Server with Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.3 (R016x.03.0.124.0) with Patch 03.0.124.0-21754
Avaya Aura® System Manager	6.3.13
Avaya Aura® Session Manager	6.3.13.0.631304
Avaya S8300D Server with Avaya G430 Media Gateway	Avaya Aura® Communication Manager 6.01 (R015x.02.1.016.4)
Avaya 9600 Series SIP IP Telephone	
9620	2.6.12
9641G	6.4.1.25
Avaya 9600 and 96X1 Series H.323 IP Telephone	
9620	3.22
9621G	6.23
9650	3.23
CallTech CTMail® Unified Messaging system on Windows 8.1 Enterprise	5.2

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager. These steps are performed through the System Access Terminal (SAT). These steps describe the procedure used for the Avaya S8300D Server. All steps are the same for the other Avaya Servers. In this section, the following topics are discussed:

- **Configure IP Network Region**
- **Configure IP Node Name**
- **Config ure IP Codec**
- **Configure SIP Signaling**
- **Configure SIP Trunk**
- **Configure Route Pattern**

5.1. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and Session Manager. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- **Authoritative Domain** – Enter the appropriate name for the Authoritative Domain. Set to the appropriate domain. During the compliance test, the authoritative domain is set to "avaya.com".
- **Codec Set** – Set the codec set number as provisioned in the **IP Codec Set** form.

```
change ip-network-region 1                              Page   1 of  20
                            IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                  Intra-region IP-IP Direct Audio: yes
       Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                       IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

5.2. Configure IP Node Name

This section describes the steps for setting IP node name for Session Manager in Communication Manager. Enter the **change node-names ip** command, and add a node name for **SM-1** (Session Manager) along with its IP address.

```
change node-names ip                                    Page   1 of   2
                             IP NODE NAMES
    Name              IP Address
CTMail           10.64.43.249
default          0.0.0.0
procr            10.64.41.21
procr6           ::
rdtt             10.64.40.14
SM-1             10.64.41.42
```

5.3. Configure IP Codec

This section describes the steps for setting IP codec. Also included in this section is for configuring the fax configuration. During the compliance test, G.711MU and G.729 were utilized.

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

                              IP CODEC SET

    Codec Set: 1

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



     Media Encryption
 1: none
 2:
 3:
```

On Page 2 of the IP Codec form, the FAX field was configured "t.38-standard" mode, as shown below. During the compliance test, t.38-standard and pass-through was tested.

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

                              IP CODEC SET

                        Allow Direct-IP Multimedia? y
            Maximum Call Rate for Direct-IP Multimedia:  4096:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia:  4096:Kbits


                                                           Packet
                          Mode                Redundancy   Size(ms)
  FAX                     t.38-standard           0         ECM: y
  Modem                   off                     0
  TDD/TTY                 US                      3
  H.323 Clear-channel     n                       0
  SIP 64K Data            n                       0             20
```

5.4. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for signaling between Communication Manager and Session Manager. Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following:

- **Group Type** – Set to "sip".
- **Transport Method** – Set to "tls".
- **Near-end Node Name** - Set to "procr" as displayed in **Section 5.2**.
- **Far-end Node Name** - Set to the "SM-1" configured in **Section 5.2**.
- **Far-end Network Region** - Set to the region configured in **Section 5.1**.
- **Far-end Domain** - Set to "avaya.com".
- **Direct IP-IP-Audio Connections:** Set to "y"

```
add signaling-group 92                                         Page   1 of   2
                               SIGNALING GROUP

 Group Number: 92               Group Type: sip
  IMS Enabled? n             Transport Method: tls
        Q-SIP? n
    IP Video? y          Priority Video? y      Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n


   Near-end Node Name: procr                   Far-end Node Name: SM-1
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1


Far-end Domain:avaya.com
                                          Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate             RFC 3389 Comfort Noise? n
        DTMF over IP: rtp-payload         Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3              IP Audio Hairpinning? n
        Enable Layer 3 Test? y            Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n    Alternate Route Timer(sec): 6
```

5.5. Configure SIP Trunk

This section describes the steps for administering a trunk group in Communication Manager for trunking between Communication Manager and Session Manager. Enter the **add trunk-group** **<t>** command, where **t** is an unallocated trunk group and configure the following:

- **Group Type** – Set the Group Type field to "sip".
- **Group Name** – Enter a descriptive name.
- **TAC** (Trunk Access Code) – Set to any available trunk access code.
- **Signaling Group** – Set to the Group Number field value configured in **Section 5.4**.
- **Number of Members –** Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

```
add trunk-group 92                                      Page   1 of  21
                              TRUNK GROUP

Group Number: 92                    Group Type: sip        CDR Reports: y
  Group Name: SM_41_42                      COR: 1     TN: 1       TAC: 1092
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                         Member Assignment Method: auto
                                                   Signaling Group: 92
                                                 Number of Members: 10
```

5.6. Configure Route Pattern

Enter **change route-pattern <r>**, where **r** is the route-pattern number. The route pattern 92 routes calls to the trunk group 92, which is the SIP trunk to Session Manager.

```
change route-pattern 92                                       Page   1 of   3
                    Pattern Number: 92    Pattern Name: no IMS SIP trk
                             SCCAN? n     Secure SIP? n
   Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
   No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 92   0                                                           n    user
 2:                                                                  n    user
 3:                                                                  n    user

     BCC VALUE   TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                   Dgts Format
                                                            Subaddress
 1: y y y y y n  n             rest                                         none
 2: y y y y y n  n             rest                                         none
 3: y y y y y n  n             rest                                         none
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

It is assumed that Session Manager and System Manager have been installed, network connectivity exists between the two platforms, and following topics between Communication Manager and Session Manager are already configured:

- **SIP Domains**
- **Locations**
- **SIP Entities**
- **Entity Links**
- **Time Ranges**
- **Routing Policy**
- **Dial Patterns**
- **Manage Element**
- **Applications**
- **Application Sequence**

This section only discusses the configuration of below topics:

- **Configure SIP Entities for CTMail**
- **Configure Entity Links for CTMail**
- **Configure Routing Policy for CTMail**
- **Configure Dial Patterns for CTMail**

6.1. Configure SIP Entitiy for CTMail

Launch a web browser, enter **http://<IP address of System Manager>** in the URL, and log in with the appropriate credentials. Navigate to **Elements →Routing**.
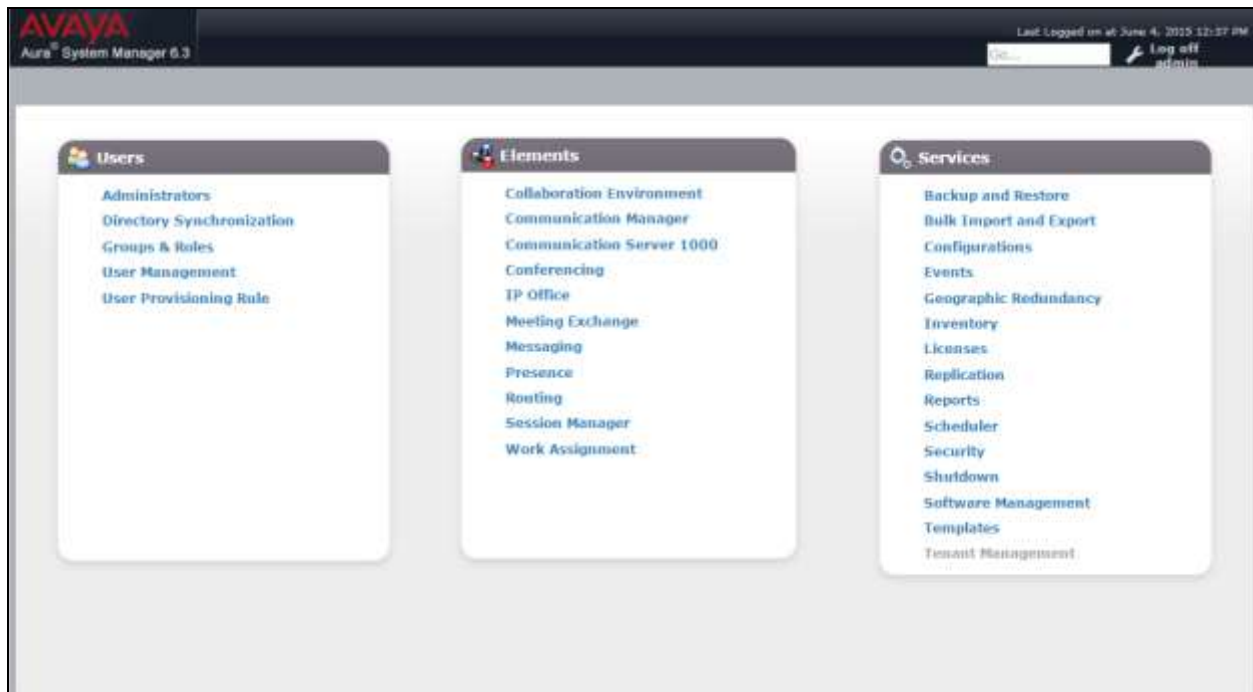
From the left pane, select **SIP Entities**. On the SIP Entities page, click the **New** button in the subsequent screen to add a new SIP entity for CTMail..
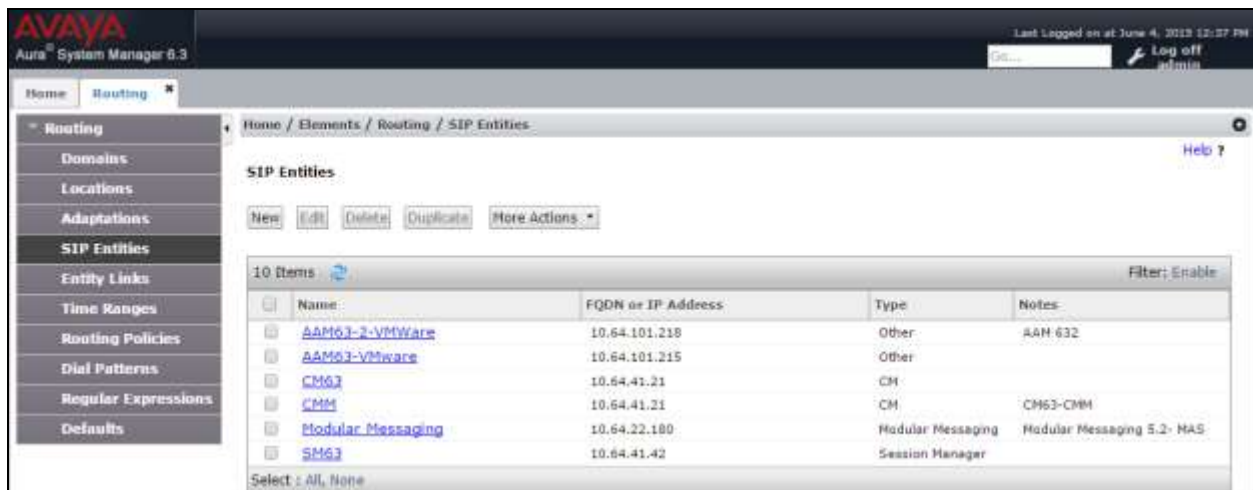
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:**   A descriptive name.
- **FQDN or IP Address:**   The IP address of the CTMail server.
- **Type:**   "Other".
- **Location:**   Select the CTMail location name.
- The Entity Links section will be automatically populated after the Entity Links page is configured.

Click the **Commit** button.

The following screen shows the SIP Entities page after the entity for CTMail was added.

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6.2. Configure Entity Link for CTMail

Select **Routing → Entity Links** from the left pane, and click **New** in the subsequent screen (not shown) to add a new entity link for CTMail.

The **Entity Links** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.
- **Name:**   A descriptive name.
- **SIP Entity 1:** The Session Manager entity name.
- **Protocol:**   The signaling group transport method. During the compliance test, the UDP protocol was used between Session Manager and CTMail.
- **Port:**   An appropriate signaling group listen port number is displayed.
  - Following is the default port
    - TLS – 5061
    - TCP/UCP - 5060
- **SIP Entity 2:** Select the CTMail entity name.
- **Port:**   An appropriate signaling group listen port number is displayed.
- **Connection :** Retain the default value, "trusted".
  Policy

Click the **Commit** button.
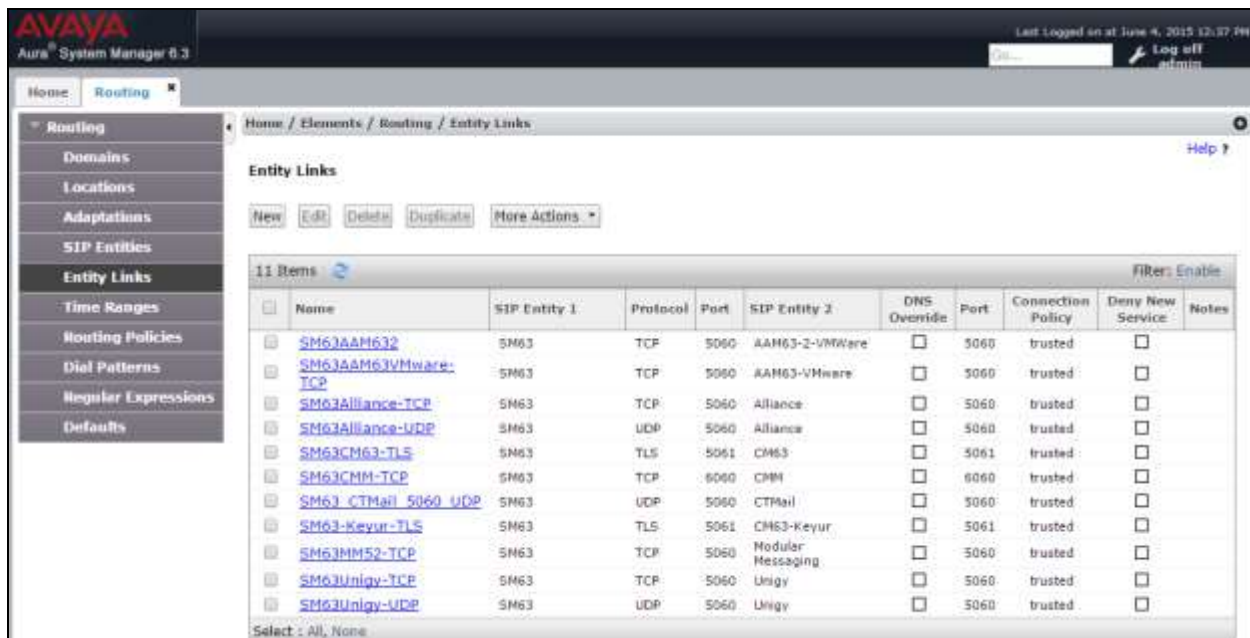
The following screen shows the Entity Links page after the entity link for CTMail was added.

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6.3. Configure Routing Policy for CTMail

Navigate to **Routing** → **Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for CTMail.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**.

In the **SIP Entity as Destination** sub-section, click **Select** and select the CTMail's entity name from **Section 6.1** in the listing (not shown).
The Dial Patterns section will be automatically populated after the Dial Patterns page is configured (**Section 6.4**).

Retain the default values in the remaining fields.

Click the **Commit** button.

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The following screen shows the Routing Policies page after the routing policy for CTMail was added.

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6.4. Configure Dial Patterns for CTMail

Navigate to **Routing → Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach CTMail.

The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match.
- **Min:** The minimum number of digits to be matched.
- **Max:** The maximum number of digits to be matched.
- **SIP Domain:** During the compliance test, "all" was selected for the sip domain.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy for reaching CTMail.

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In the **Originating Location** section, check the ''Apply The Selected Routing Policies to All
Originating Locations'' checkbox.
In the **Routing Policies** section, check an appropriate routing policy from **Section 6.3**.

Click the **Select** button.

After the **Originating Locations page** is completed, Click **Commit** on the Dial Pattern Details
page.

The following screen shows the Dial Pattern page after the Dial Patterns for CTMail were added.

	Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
☐	*	3	8	☐			-ALL-	
☐	12	10	12	☐			-ALL-	
☐	1303	10	12	☐			avaya.com	
☐	332	5	5	☐			-ALL-	To Alliance16 using SIP
☐	333	5	5	☐			-ALL-	To Alliance16 using QSIG via CM
☐	7200	5	5	☐			-ALL-	CM stations
☐	7201	5	5	☐			-ALL-	
☐	7205	5	5	☐			-ALL-	To Unigy using SIP
☐	7207	5	5	☐			-ALL-	
☐	7208	5	5	☐			-ALL-	
☐	7209	5	5	☐			-ALL-	
☐	7775	4	4	☐			-ALL-	
☐	7776	4	4	☐			-ALL-	
☐	7777	4	4	☐			-ALL-	
☐	7778	4	4	☐			-ALL-	

7. Configure CTMail

This section describes the configuration of CTMail. Installation of the CTMail software was performed by a CallTech engineer prior to the actual compliance test. From the Window 8.1 system, select **CTMail Server**.

This section only discusses the configuration of below topics:
- **Configure IP Address for CTMail**
- **Configure VoiceMail for CTMail**
- **Configure FAX for CTMail**

Note: *An Administrator can configure inbound extensions for FAX, IVR, Auto Attendant, and VoiceMail (7775/7777/7778/7779) using the same steps. Thus, in these App Notes only the FAX and VoiceMail configuration steps are discussed.*
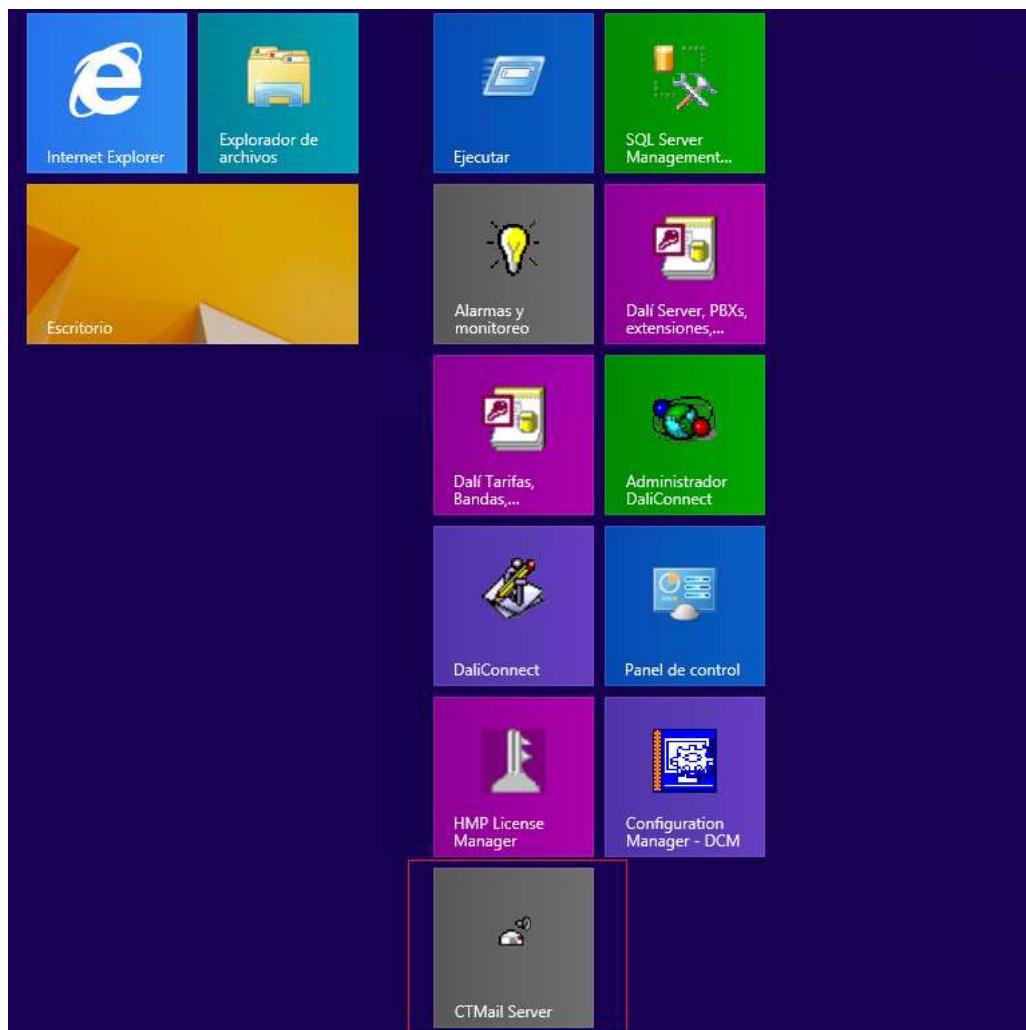
In the main page, the port activity is displayed.
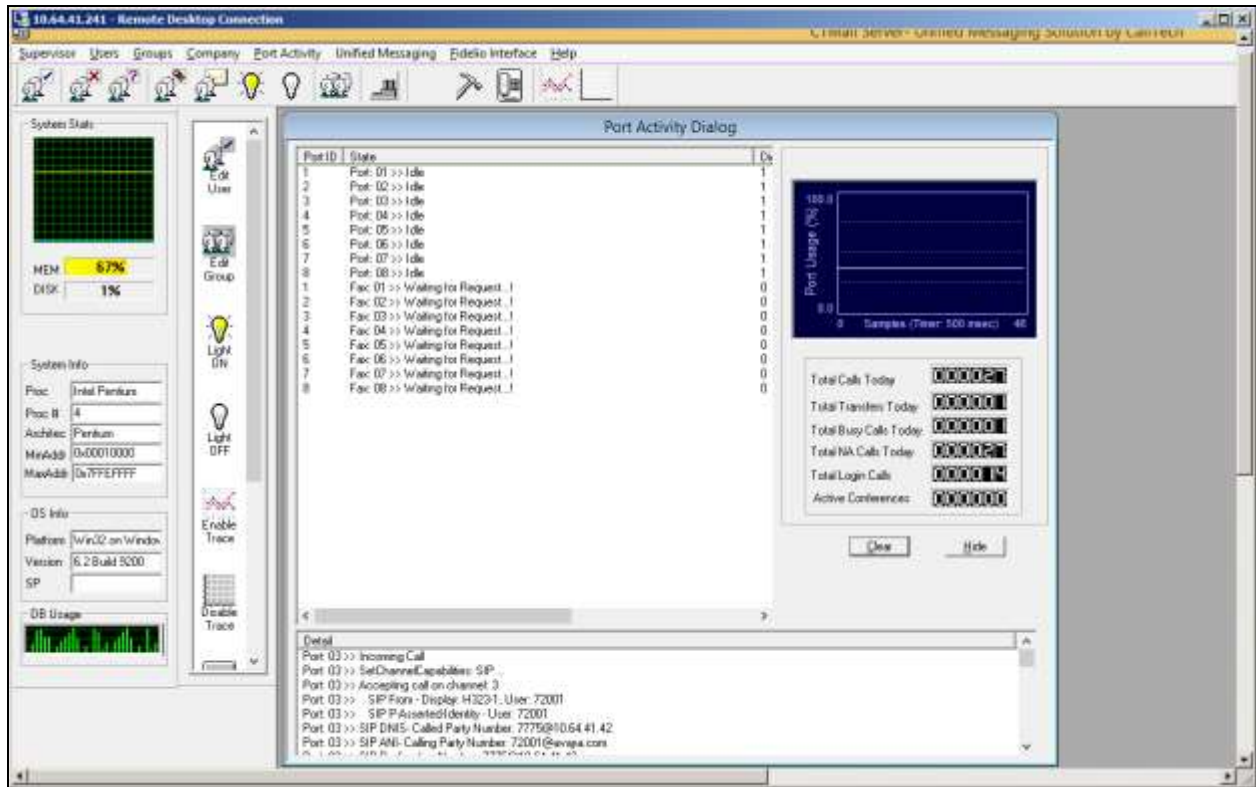
7.1. Configure IP address for CTMail

In the CTMail directory of the CTMail server, add/modify the parameters on the Parameters.txt file as shown below.

```
SIP Parameters
"Local IP Address" SIP_LOCAL_IP_ADDRESS 10.64.41.241
"Persisten Contact Header Type: 0 - disable 1 - line(all calls) 2 - line(single
session) 3 - on a call(single session)" SIP_PERSISTENT_CONTACT_HEADER_TYPE 1
"Persisten Contact Header Value" SIP_PERSISTENT_CONTACT_HEADER_VALUE
sip:7777@10.64.41.241
```

7.2. Configure VoiceMail for CTMail

Launch a web browser, enter **http://<IP address of CTMail server>/CTMail_Configurator** in the URL, and log in with the appropriate credentials.

Navigate to **Configuration → Ports → Ports Configurations** (not shown).

In the **Configuration** page, provide the following information:
- **Server Name**: Select the CTMail server, using the drop-down list.
- **PBX Name**: Select "Avaya CM", using the drop-down list.
- **Protocol**: Select "SIP", using the drop-down list.
- Retain the default values in the remaining fields.

Select **Update**.
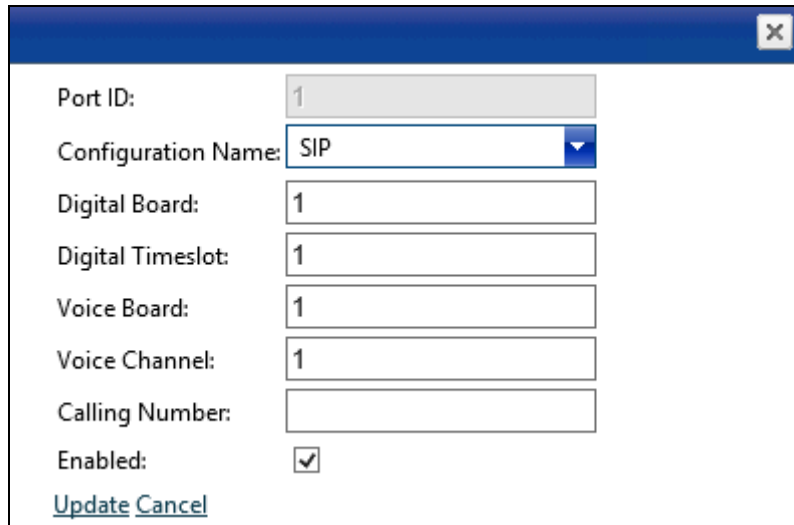
Navigate to **Configuration → Ports → Ports**. The following screen shows eight ports are enabled for receiving inbound calls. Each port can be configured using the **Edit** button.

The following screen shows the Edit process of the Port ID 1. Select "SIP", using the drop-down list, on the **Configuration Name** field. Retain the default values in the remaining fields. After the completion of the configuration, select **Update**.

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7.3. Configure Fax for CTMail

Navigate to **Configuration** → **Fax Ports** → **Fax Ports Configuration**. Select the **Edit** button.
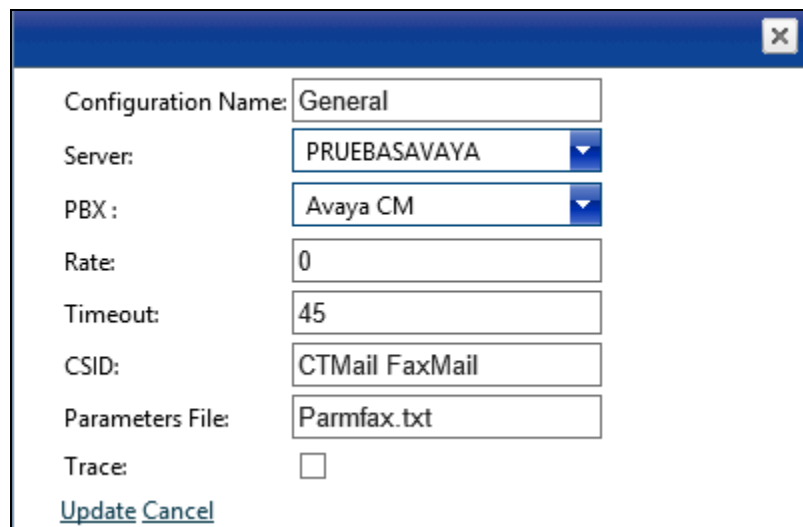
The **Fax Ports Configuration** screen displays. Provide the following information:
- **Server**: Select the name of the CTMail server, using the drop-down list.
- **PBX**: Select "Avaya CM", using the drop-down list.
- Retain the default values in the remaining fields.

Select **Update**.

Navigate to **Configuration → Fax Ports → Fax Ports**. Enable Fax ports by using the **Edit** button.

The following Fax port, Fax Port ID 3 and Fax Resource ID 2, was configured and enabled.

For the Codec configuration, navigate to **Configuration → SIP → Audio Capabilities**. Each codec can be enabled and disabled using the **Edit** button.

The following screen shows the sample configuration of G.711Alaw64k. During the compliance test, G.711Alaw was <u>not</u> utilized. Thus, the **Enabled** field is not checked.

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For the Fax Configuration, navigate to **Configuration → SIP → Data Capabilities**.

Fax can be configured using the **Edit** button. Please provide the following information:

- **Server Name:** Select the name of the CTMail server, using the drop-down list.
- **Capability**: Select "t38UDPFax", using the drop-down list.
- **Direction**: Select "Bidireccional", using the drop-down list.
- **Max Bit Rate Value**: Select "9600 bits/s", using the drop-down list.
- **Enabled**: Check the **Enabled** checkbox.

Select **Update**.

For the SIP Configuration, navigate to **Configuration → SIP → Parameters**. The following two screenshots display the SIP parameters that were used during the compliance test. The configuration changes will follow the same steps as before, using the **Edit** button.

8. Verification Steps

- Verify a correct prompt is played when a call is made to the CTMail server.
- Verify a fax can be received at CTMail, and a fax can be sent from CTMail.

9. Conclusion

These Application Notes describe the procedures for configuring CTMail with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. During the compliance test, feature tests on CTMail (*Fax, IVR, Auto Attendant, and VoiceMail*) passed successfully.

10. References

This section references the Avaya and CTMail documentation that are relevant to these Application Notes.

[1] *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 10 Release 6.3, available at http://support.avaya.com.
[2] *Administering Avaya Aura® Session Manager,* Release 6.3, Issue 7, September 2014, available at http://support.avaya.com

The CTMail Solution and Product information is available from CTMail. Visit http://www.calltechsa.com/CalltechHomePage/CTMail.aspx

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