Avaya Solution & Interoperability Test Lab

Application Notes for Kofax Communication Server with Avaya Aura® Communication Manager R7.0.1 and Avaya Aura® Session Manager R7.0.1 using a Transport Layer Security enabled SIP Trunk - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Kofax Communication Server to interoperate with Avaya Aura® Communication Manager R7.0.1 and Avaya Aura® Session Manager R7.0.1. Kofax Communication Server communicates with Avaya Aura® Session Manager via a Transport Layer Security enabled SIP trunk. This document provides configuration steps related to faxing capabilities of Kofax Communication Server.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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

These Application Notes describe the configuration used to enable Kofax Communication Server, from Kofax Ltd., to interoperate with Avaya Aura® Communication Manager R7.0.1 and Avaya Aura® Session Manager R7.0.1 using Transport Layer Security (TLS) on the SIP trunk connection. Kofax Communication Server offers a variety of telephony features. Kofax Communication Server fax features allow fax messages to be sent/received to/from both local and PSTN fax endpoints, and can subsequently be printed or archived. During compliance testing the fax feature and functionality was the sole focus.

2. General Test Approach and Test results

The general test approach was to simulate the configuration as implemented on customer premises. Compliance testing was between the Kofax Communication Server (Kofax Server) and Avaya Aura® Session Manager (Session Manager), and was performed manually. The tests were all functional in nature, and no performance testing was done. The test method employed can be described as follows, Communication Manager was configured to support various local IP (H.323) telephones and an analogue Fax Machine, as well as a SIP connection to Session Manager. Session Manager was configured to connect to both Communication Manager and Kofax Communication Server via SIP trunks using Transport Layer Security (TLS).

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Kofax Communication Server included TLS and SRTP.

2.1. Interoperability Compliance Testing

The following tests were performed as part of the compliance testing:

- Basic fax sending using T.38 and pass-through connection with G.711A and G.711MU codecs using SIPS and SRTP
- Basic fax receiving using T.38 and pass-through connection with G.711A and G.711MU

codecs using SIPS and SRTP

- Forwarding of a fax from a local Fax Machine to the Kofax Server via a local extension
- Forwarding of a fax from the Kofax Server to a local Fax Machine via a local extension
- Blind transfer of a fax from a local Fax Machine to the Kofax Server via a local extension Blind transfer of a fax from the Kofax Server to a local Fax Machine via a local extension
- Verification of correct status and Caller ID for sent and received fax messages
- Verification that Message Waiting Indication is sent to the correct phone extensions when faxes are received and subsequently turned off when the fax is accessed
- Successful recovery from network or power failure

2.2. Test Results

Tests were performed to insure full interoperability of a Kofax Communication Server when configured with TLS (using Session Manager). The tests were all functional in nature and performance testing was not included. All the test cases passed successfully.

2.3. Support

Support for Kofax Communication Server is available at: http://www.kofax.com/support/

3. Reference Configuration

Figure 1 illustrates the network configuration used during compliance testing. A SIP trunk was configured between the Kofax Communication Server (using TLS) and the Session Manager SIP Signaling interface. A SIP trunk was also configured between Communication Manager and Session Manager (using TLS). An analogue Fax Machine was connected to an MM714 Analog card on the G430 Media Gateway. An Avaya 9641G (H.323) telephone was also configured on Communication Manager so as to test faxes sent to phone extensions which had Call Forward enabled and also to transfer faxes to alternative Fax Machines, including to the Kofax Communication Server. An Avaya Aura® System Manager was used to manage the Session Manager.

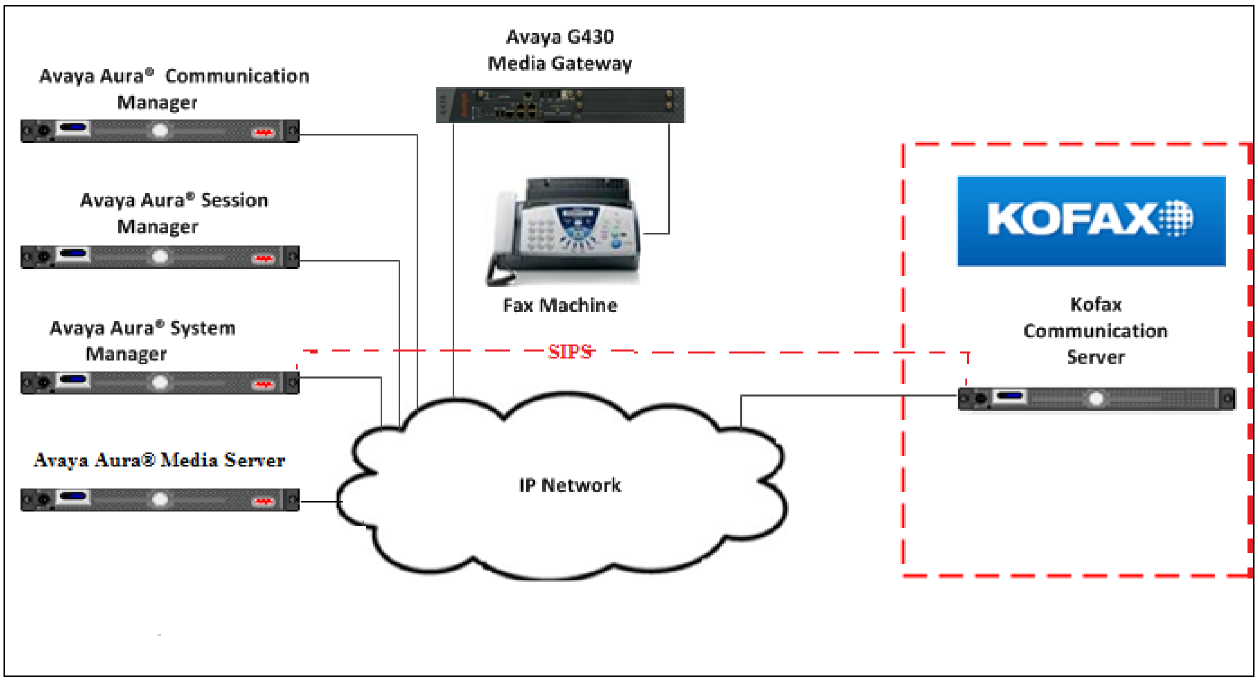


Figure 1: Avaya and Kofax Reference Configuration

4. Equipment and Software Validated

The hardware and associated software used in the compliance testing is listed below.

Avaya Equipment/Software	**Release/Version**
Avaya Aura® Communication Manager	R7.0 Build R017x.00.0.441.0 Update: 00.0.441.0-22856
Avaya Aura® Session Manager	R7.0.0.1.700102
Avaya Aura® System Manager	R7.0.0.1 Build 7.0.0.0.16266-7.0.9.7001011 Update 7.0.0.1.4212
Avaya Aura® Media Server	7.8.0.6 SP3
Avaya G430 Media Gateway Module MM714 (ANA)	Version 37.20.0 Version HW03 FW073
Kofax Equipment/Software	**Release/Version**
Kofax Communication Server KCS FoIP Application	Version 10.1 Version 3.26.11

Table 1: Hardware and Software Version Numbers

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

It is implied a working system is already in place. The configuration operations described in this section can be summarized as follows: (**Note**: during Compliance Testing all inputs not highlighted in bold were left as default).

- Check Media Encryption license
- Configure Session Manager Node
- Configure Signaling-Group (for information only)
- Configure Trunk Group (for information only)
- Configure Fax Station
- Configure Codecs

5.1. Check Media Encryption license

When using TLS to encrypt the signalling the Media will be encrypted using Secure RTP. On **Page 5** of the **system-parameters customer-options** screens check that **Media Encryption Over IP?** is set to **y**.

```
    display system-parameters customer-options                       Page   5 of  12
                                OPTIONAL FEATURES

      Emergency Access to Attendant? y                             IP Stations? y
                                  Enable 'dadmin' Login? y
              Enhanced Conferencing? y                       ISDN Feature Plus? n
                     Enhanced EC500? y         ISDN/SIP Network Call Redirection? y
       Enterprise Survivable Server? n                         ISDN-BRI Trunks? y
          Enterprise Wide Licensing? n                                ISDN-PRI? y
                 ESS Administration? y               Local Survivable Processor? n
             Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
        External Device Alarm Admin? y                 Media Encryption Over IP? y
     Five Port Networks Max Per MCC? n      Mode Code for Centralized Voice Mail? n
                                       Flexible Billing? n
      Forced Entry of Account Codes? y                 Multifrequency Signaling? y
         Global Call Classification? y          Multimedia Call Handling (Basic)? y
                Hospitality (Basic)? y       Multimedia Call Handling (Enhanced)? y
    Hospitality (G3V3 Enhancements)? y               Multimedia IP SIP Trunking? y
                                              IP Trunks? y

                                  IP Attendant Consoles? y
           (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Configure Session Manager Node

For Communication Manager to communicate with Session Manager a node must be configured. The screen shot below shows **SM71676** with IP address **10.10.16.77** was used.
Note: 10.10.16.77 IP address of the Session Manager SIP Signaling interface.

```
change node-names ip                                              Page   1 of   2
                              IP NODE NAMES
    Name              IP Address
AES63RP             10.10.16.78
SM71676             10.10.16.77
default             0.0.0.0
procr               10.10.16.211
procr6              ::
```

5.3. Configure Signaling Group

A signaling group is required before a trunk-group can be configured. Use the **add signaling-group** command followed by next available signaling-group number to configure the following:

- **Group Type:** Enter **sip**
- **Transport Method** Enter **tls**
- **Enforce SIPS URI for SRTP?** Enter **y**
- **Near-end Node Name:** Enter **procr**
- **Far-end Node Name:** Enter **SM71676** (Session Manager Node as configured in **Section 5.1**)
- **Far-end Network Region:** Enter the appropriate Network Region (i.e. 1)
- **Far End Domain:** Enter the appropriate Domain

When the configuration is complete, press **F3** to save.

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   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
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: SM71676
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: devconnect.local
                                             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
```

5.4. Configure Trunk Group

This section describes the Trunk Group configuration used during compliance testing. Use the **add trunk-group** command followed by next available Group number and configure the following:

- **Group Type:** Enter **sip**
- **Group Name:** Enter an informative name for the trunk (i.e. **To SM7.0 SIP)**
- **TAC** Enter a TAC number (i.e. **701)**
- **Service Type:** Enter **public-ntwrk**
- **Signaling Group:** Enter the Signaling Group number as configured in **Section 5.2**
- **Number of Members:** Enter the number of channels required to connect to the Session Manger (during compliance testing 30 channels were used)

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

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: To SM7.0 SIP               COR: 1        TN: 1        TAC: 701
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 30
```

Go to **Page 3** and enter **private** for **Numbering format**. When the configuration is complete, press **F3** to save.

```
display trunk-group 1                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n



                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

5.5. Configure Fax Station

The Fax Machine is configured as an Analog station **Type 2500** on Communication Manager and the **Extension** number used was **8270501**. The port used was an available port on a MM714 card on the G430 Media Gateway. Use the **add station** command to add the Fax machine. The screen shots below show the configuration used during compliance testing. When the configuration is complete, press **F3** to save.

```
add station 8270501                                          Page   1 of   4
                                     STATION

Extension: 8270501                       Lock Messages? n              BCC: 0
     Type: 2500                          Security Code: 1026            TN: 1
     Port: 002V301                     Coverage Path 1:                COR: 1
     Name: Fax Machine 8270501         Coverage Path 2:                COS: 1
                                       Hunt-to Station:              Tests? y
STATION OPTIONS
      XOIP Endpoint type: auto             Time of Day Lock Table:
              Loss Group: 1             Message Waiting Indicator: none
    Off Premises Station? n



          Survivable COR: internal
   Survivable Trunk Dest? y
                                              Remote Office Phone? n



Passive Signalling Station? N
```

```
add station 8270501                                          Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                    Coverage Msg Retrieval? y
  LWC Log External Calls? n                               Auto Answer: none
             CDR Privacy? n                          Data Restriction? n
   Redirect Notification? y                  Call Waiting Indication: y
 Per Button Ring Control? n             Att. Call Waiting Indication: y
   Bridged Call Alerting? n                 Distinctive Audible Alert? y
        Switchhook Flash? y                      Adjunct Supervision? y
    Ignore Rotary Digits? n
        H.320 Conversion? n        Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic                Audible Message Waiting? n
    MWI Served User Type:
              AUDIX Name:
                                            Coverage After Forwarding? s
                                              Multimedia Early Answer? n
                                          Direct IP-IP Audio Connections? Y
  Emergency Location Ext: 1026                      IP Audio Hairpinning? n
```

```
add station 8270501                                          Page   3 of   4
                                    STATION


   Bridged Appearance Origination Restriction? n







                             ENHANCED CALL FORWARDING
                                      Forwarded Destination         Active
 Unconditional For Internal Calls To:                                  n
                   External Calls To:                                  n
          Busy For Internal Calls To:                                  n
                   External Calls To:                                  n
      No Reply For Internal Calls To:                                  n
                   External Calls To:                                  n

            SAC/CF Override: n
```

```
add station 8270501                                          Page   4 of   4
                                    STATION
 SITE DATA
       Room:                                           Headset? n
       Jack:                                           Speaker? n
      Cable:                                          Mounting: d
      Floor:                                       Cord Length: 0
   Building:                                         Set Color:





ABBREVIATED DIALING
     List1:                   List2:                    List3:

HOT LINE DESTINATION
         Abbreviated Dialing List Number (From above 1, 2 or 3):
                                                     Dial Code:

     Line Appearance: call-appr
```

5.6. Configure Codecs

During compliance testing T.38 Fax was used. If using Pass-through Fax configuration, see **Appendix A**. To configure T.38 Fax, use the **change ip-codec-set x** command where x is the ip-codec-set being used. Configure the following on page 1:

- **Audio Codec (line 1)** Enter **G.711MU**
- **Silence Suppression** Enter **n**
- **Frames Per Pkt** Enter **2**
- **Audio Codec (line 2)** Enter **G.711A**
- **Silence Suppression** Enter **n**
- **Frames Per Pkt** Enter **2**
- **Media Encryption** Enter **2-srtp-aescm128-hmac32** and **1-srtp-aescm128-hmac80**
- **Encrypted SRTCP** Enter **enforce-unenc-srtcp**

Note: Kofax and Communication Manager currently do not support Encrypted SRTCP. It is recommended that **Encrypted SRTCP** is set to **enforce-unenc-srtcp** and the corresponding Kofax parameter MediaSecurityUnencryptedSrtcp to [2] offer only crypto with UNENCRYPTED_SRTCP, is set as shown in **Section 7.4**.
Should this Kofax parameter be set to [3] offer crypto with and without UNENCRYPTED_SRTCP, **Encrypted SRTCP** must be set to **best-effort.**

Note: The max baud rate is 9600 bits per second.

```
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: G.711A            n           2        20
 3:
 4:
 5:
 6:
 7:


     Media Encryption                   Encrypted SRTCP: enforce-unenc-srtcp
 1: 2-srtp-aescm128-hmac32
 2: 1-srtp-aescm128-hmac80
 3:
```

On Page 2 configure the following. Set **Fax Mode** to **off** All other inputs may be left at default. When the configuration is complete, press **F3** to save.

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

                          IP CODEC SET

                              Allow Direct-IP Multimedia? n

                                                                       Packet
                          Mode                      Redundancy         Size(ms)
    FAX                   t.38 standard                       0
    Modem                 off                        0
    TDD/TTY               US                         3
```

6. Configuring Avaya Aura® Session Manager

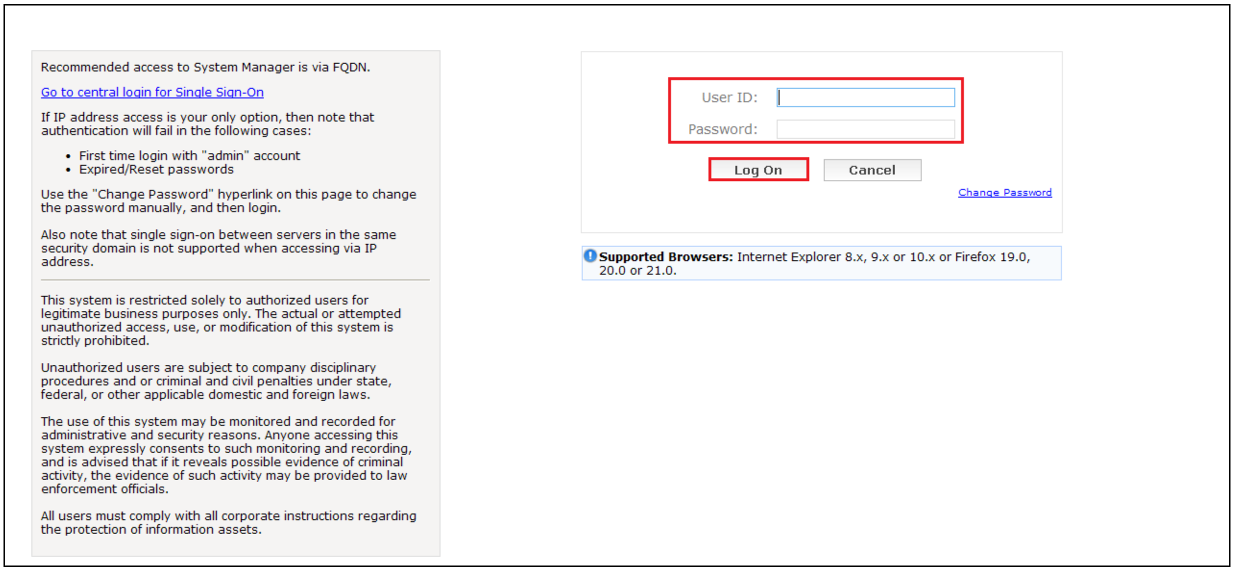
A number of configurations are required to enable the Session Manager to route faxes between Communication Manager and the Kofax Communication Server. All configurations of Session Manager are performed using System Manager. The configuration operations described in this section can be summarized as follows:

- Logging on to Avaya Aura® System Manager
- Administer SIP Domain
- Administer Locations
- Create Kofax Communication Server as a SIP Entity
- Create an Entity Link for Kofax Communication Server
- Create a Routing Policy Kofax Communication Server
- Create a Dial Pattern for Kofax Communication Server

Note: See **Appendix B** for a screen shot of the Entity Link used between Session Manager and Communication Manager.

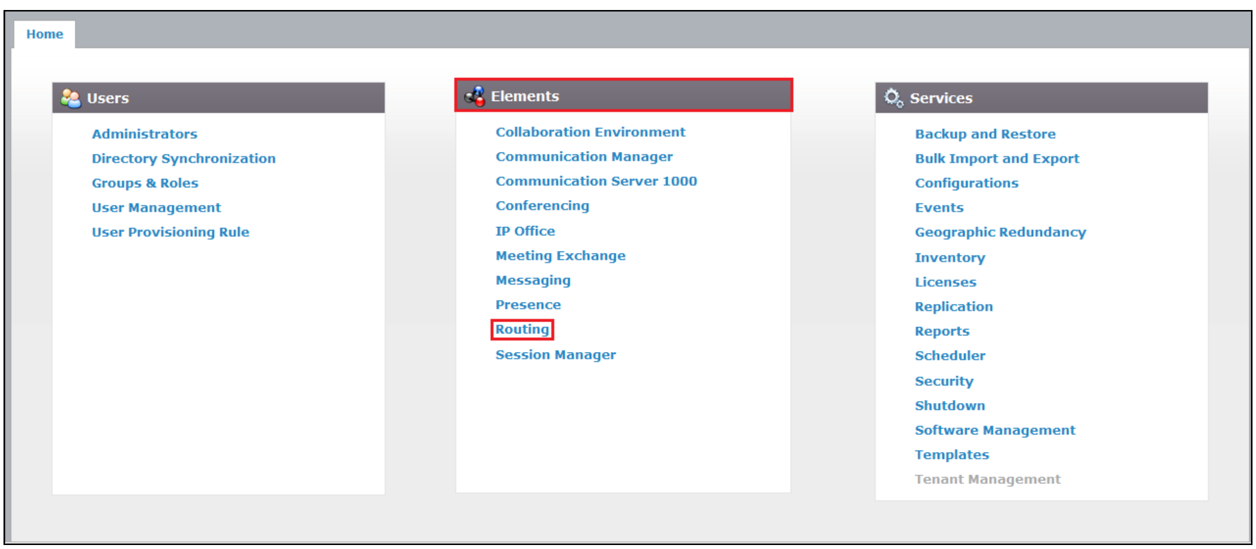
6.1. Logging on to Avaya Aura® System Manager

Log on by accessing the browser-based GUI of System Manager, using the URL "http://<fqdn>/SMGR" or "http://<ip-address>/SMGR", where:
"<fqdn> is the fully qualified domain name of the Avaya Aura® System Manager or the"<ip-address>" is the IP address of Avaya Aura® System Manager.
Once the System Manager web page opens log in with the appropriate credentials and click on the **Log on** button.

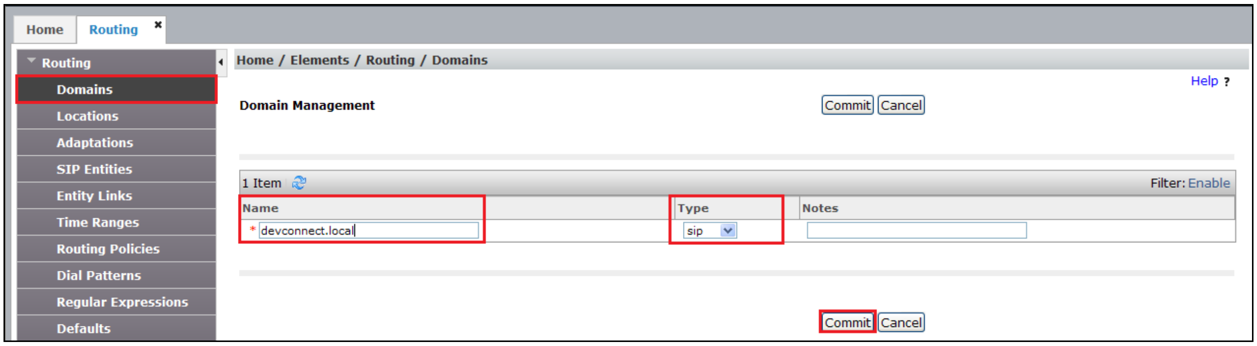


6.2. Administer SIP Domain

Once logged in, select **Routing** from under the **Elements** column.

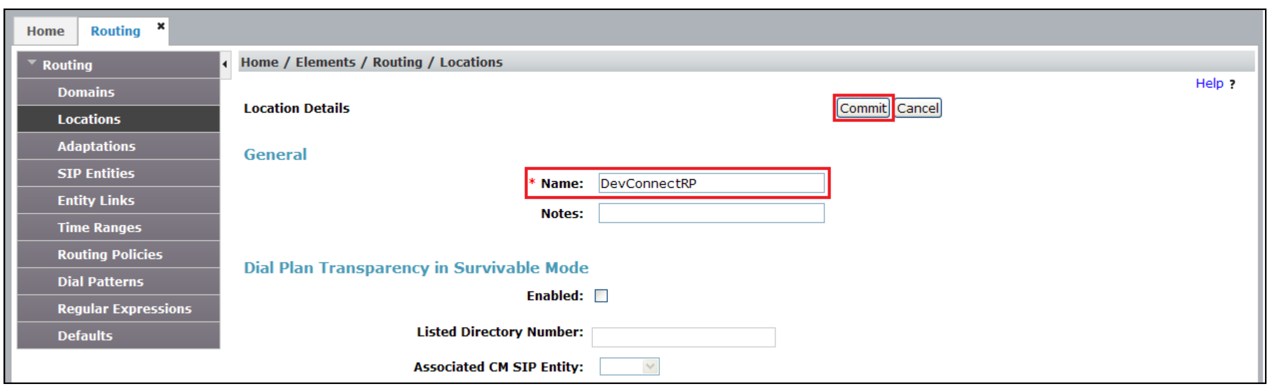


Select **Domains** on the left panel menu and then click on the **New** button (not shown). In the **Name** field enter the domain of the enterprise (i.e. **devconnect.local**) and select **sip** from the dropdown box. Click **Commit** to save changes.

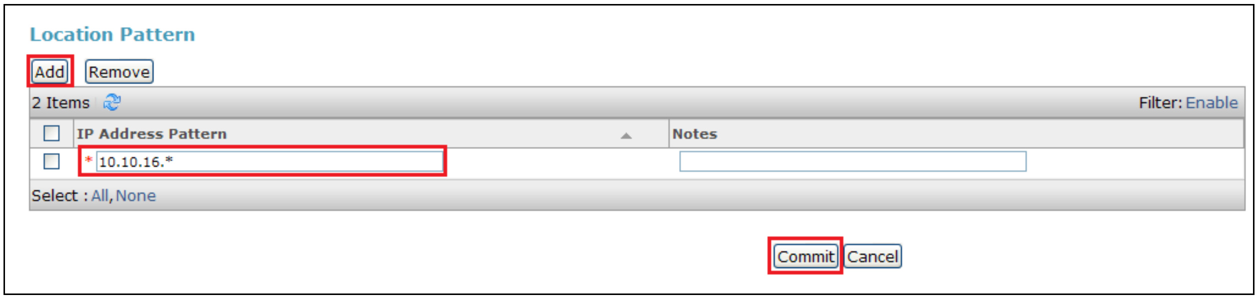


6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. Select **Locations** on the left panel menu and then click on the **New** button (not shown). In the **Name** field enter an informative name for the location (i.e. **DevConnectRP**). During compliance testing, all other fields were left at default values.



Scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string. Below is the location configuration used during compliance testing.



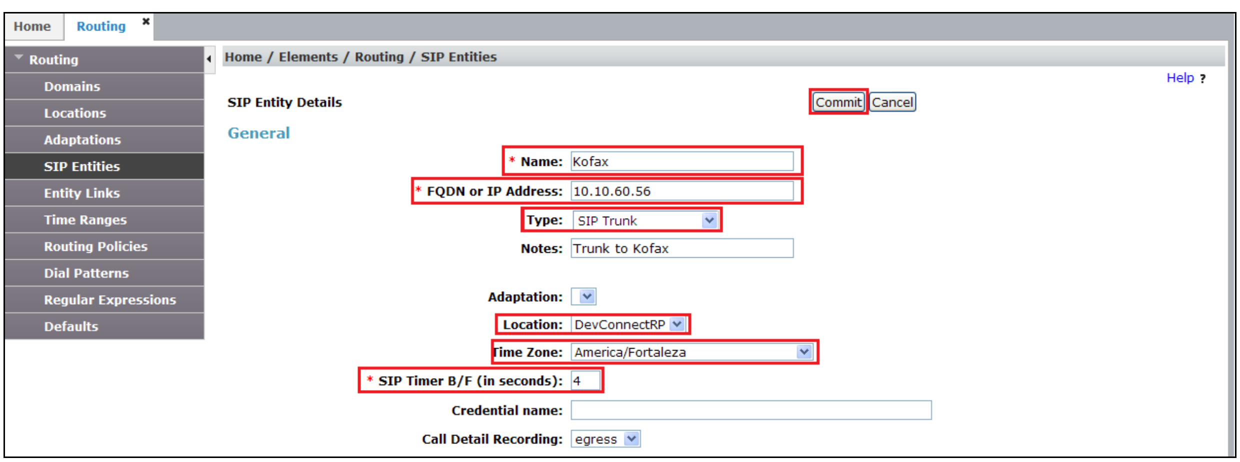
6.4. Create Kofax Communication Server as a SIP Entity

A SIP Entity must be added for the Kofax Server. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown).
Note: A SIP Entity was already configured for the Communication Manager and was called **CM63**.
Enter the following for the ApplianX SIP Entity:
Under **General** enter the following:

- **Name** Enter an informative name (e.g., **Kofax**)
- **FQDN or IP Address** Enter the IP address of the of the Kofax Server
- **Type** Select **SIP Trunk** from the dropdown box
- **Location** Select the location from the dropdown box that was configured in **Section 6.3**
- **Time Zone** Select Time zone for this location from the dropdown box
- **SIP Timer** Enter **4**

Once the correct information is entered click the **Commit** button.

Note: During compliance testing **Adaptation** was left blank.

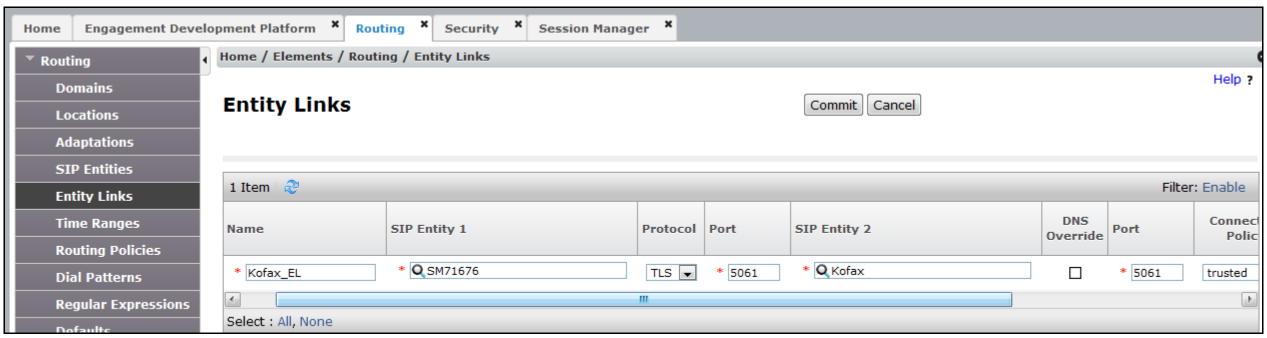


6.5. Create an Entity Link for Kofax Communication Server

The SIP trunk between Session Manager and the Kofax Server requires an Entity Link.
To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown) Enter the following:

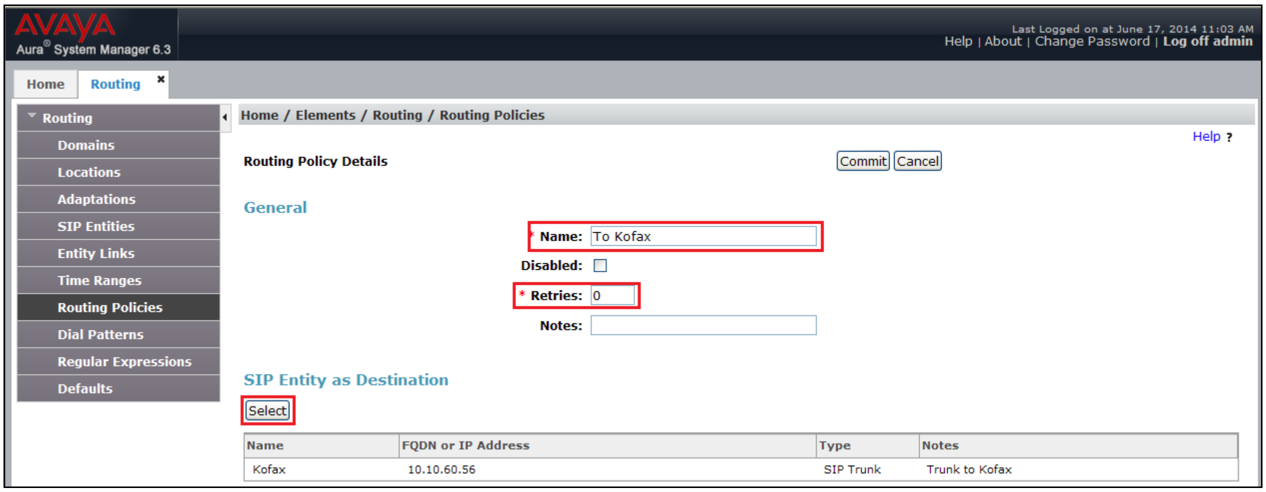
- **Name** An informative name, (e.g. **Kofax Link**)
- **SIP Entity 1** Select **SM71676** from the **SIP Entity 1** dropdown box
- **Protocol** Select **TLS** from the Protocol drop down box
- **Port** Enter **5061**
- **SIP Entity 2** Select **Kofax** from the **SIP Entity 2** dropdown box (configured in **Section 6.4)**
- **Port** Enter **5061** as the Port
- **Connection Policy** Select **trusted** from the dropdown box

Click **Commit** to save changes. The following screen shows the Entity Links used.

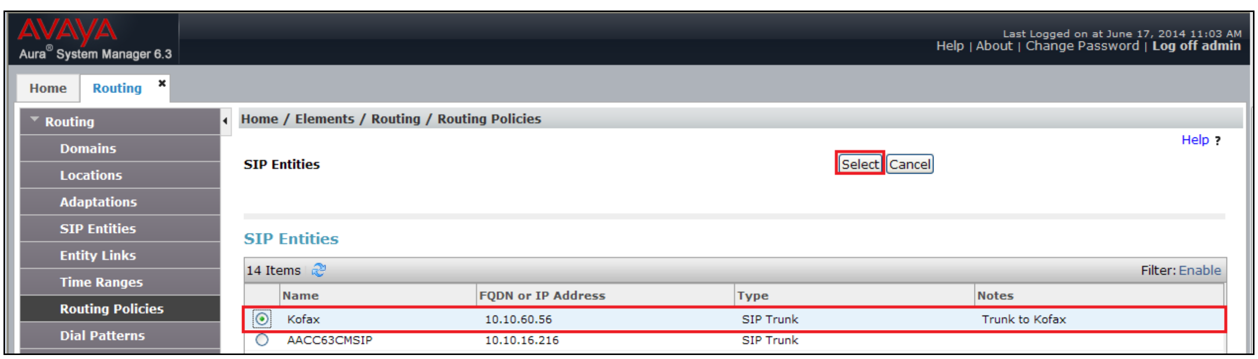


6.6. Create a Routing Policy for Kofax Communication Server

Create routing policies to direct calls to the Kofax Server via Session Manager. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown). In **Routing Policy Details** enter an informative name in the **Name** field (example, **To Kofax**) and enter **0** in the **Retries** field**.** At **SIP Entity as Destination**, click the **Select** button. A Routing Policy was also configured to direct calls to Communication Manager, but is outside the scope of these Application Notes.



Once the **SIP Entity** List screen opens, check the **Kofax** radio button. Click on the **Select** button to confirm the chosen options and then return to the **Routing Policies Details** screen and select the **Commit** button (not shown) to save.

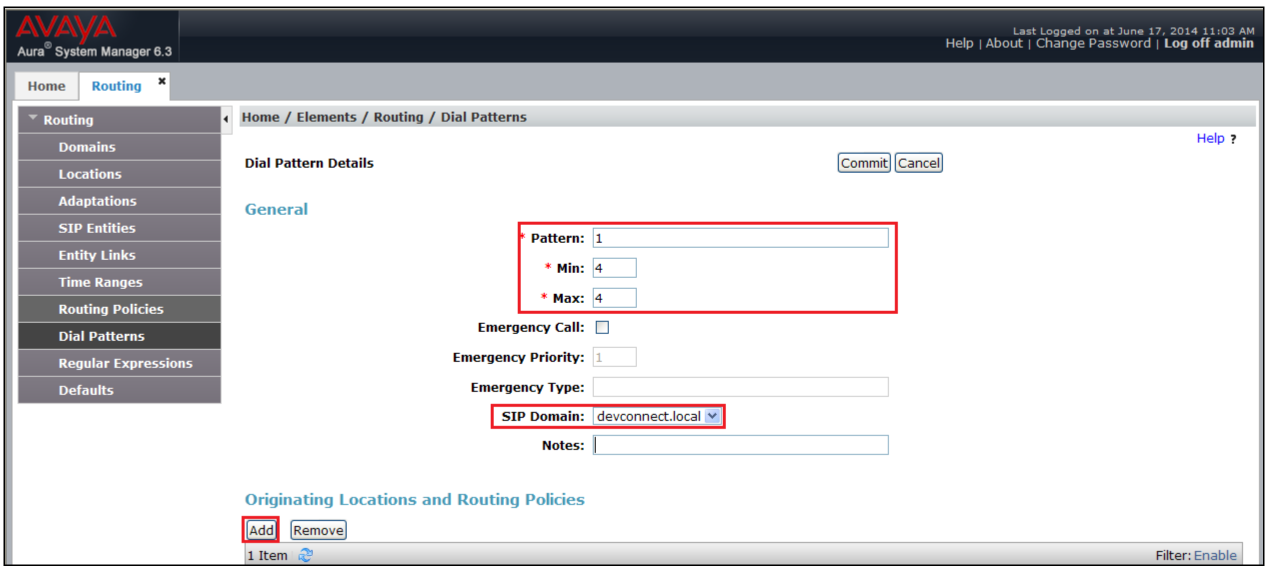


6.7. Create a Dial Pattern for Kofax Communication Server

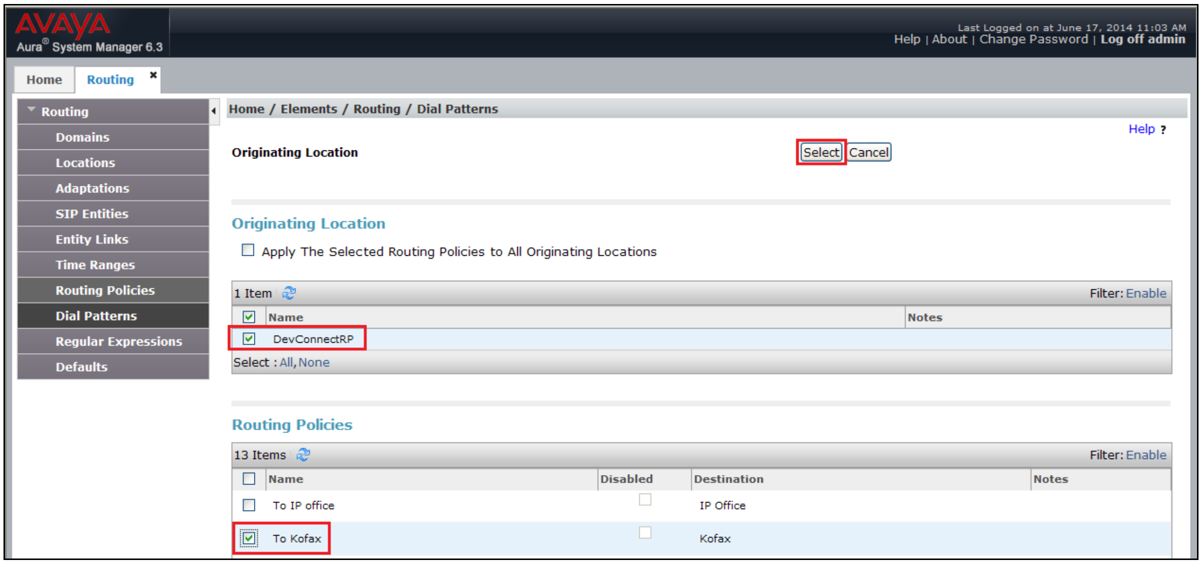
A dial pattern must be created on Session Manager to route calls to and from the Kofax Server. During compliance testing a number of dial patterns were used. The example below shows 1. To configure the Dial Pattern to route calls to the Kofax Server, select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown). A Dial Pattern was also configured to route calls to Communication Manager, but is outside the scope of these Application Notes. Under **General** enter out the following:

- **Pattern** Enter **1**
- **Min** Enter **4** as the minimum length of dialed number
- **Max** Enter **4** as the maximum length of dialed number
- **SIP Domain** Select **All** from the drop down box

Click the **Add** button in **Originating Locations and Routing Policies**.



In **Originating Location** check the **DevConnectRP** check box. Under **Routing Policies** check the **To Kofax** check box. Click on the **Select** button to confirm the chosen options and then be returned to the Dial Pattern screen (shown previously), select **Commit** button to save not shown.

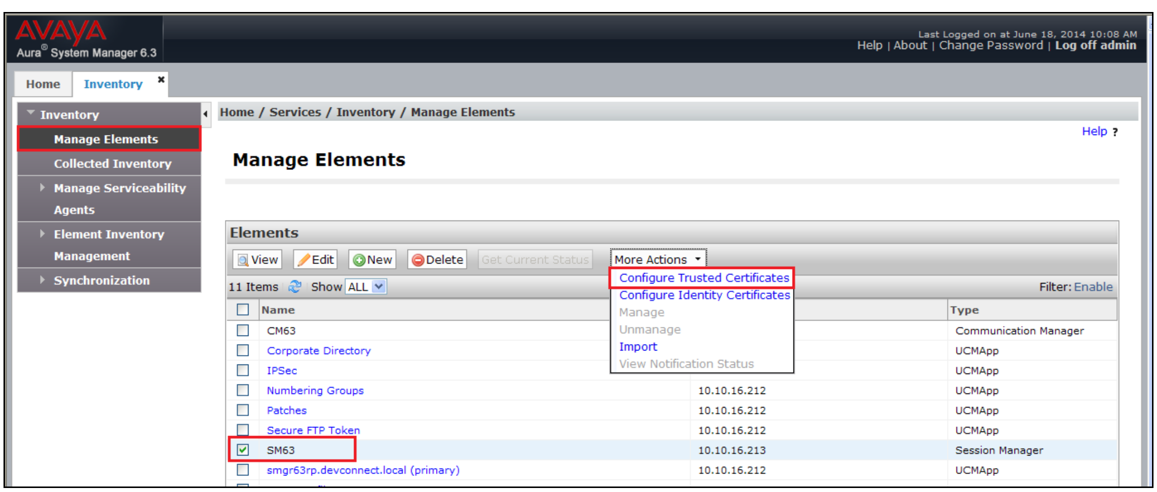


6.8. Manage Certificates

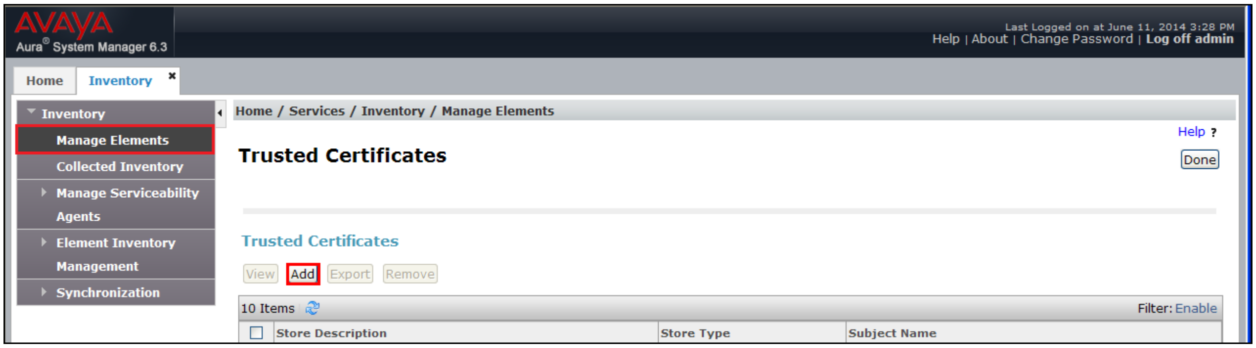
In order for Session Manager and the Kofax Server to successfully negotiate a TLS connection certificates must be exchanged and authenticated during the TLS handshake. For two-way authentication both Session Manager and the Kofax Server need to import each other's certificate. See **Appendix B** for information relating to exporting the Session Manager trusted certificates.

6.8.1. Adding Kofax Server Trusted Certificate

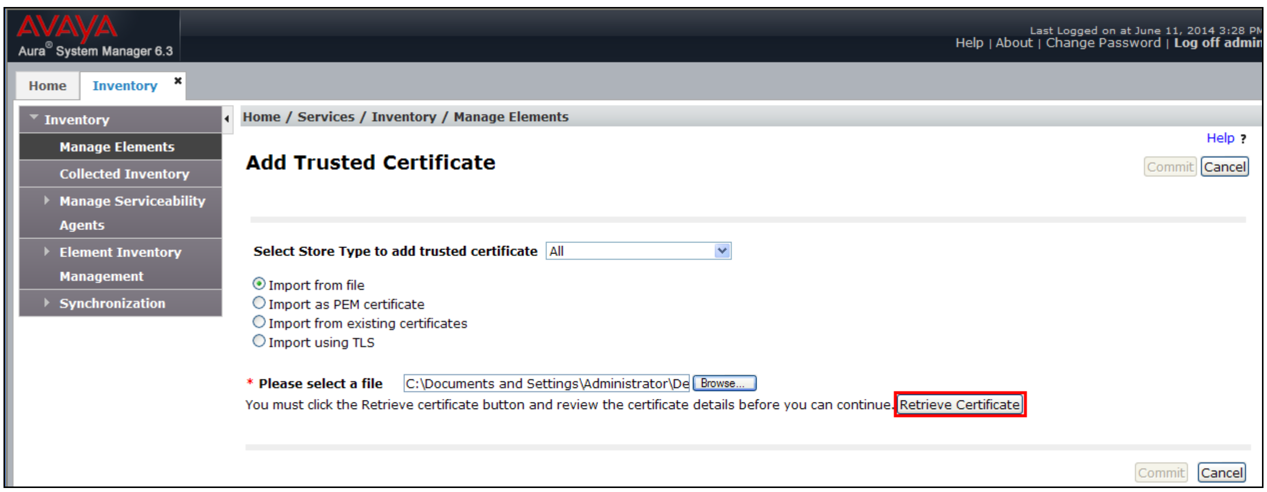
Before adding the trusted certificate it must first be placed in a location accessible by System Manager. To add the certificate select **Inventory** from the **Home** Screen under **Services** (not shown). Select **Managed Elements** and select the Session Manager you are using for the TLS SIP Trunk. From the More Actions drop down select **Configure Trusted Certificates**.



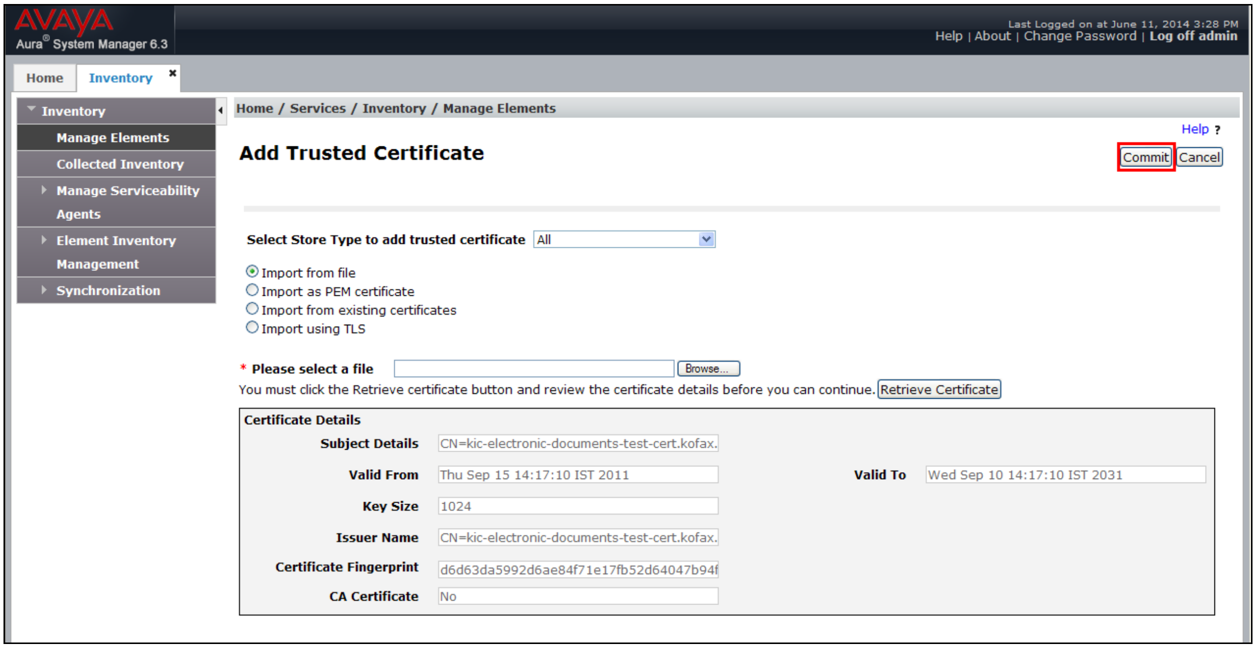
Once the **Trusted Certificate** screen opens, click on the **Add** button.



Once the **Add Trusted Certificate** screen opens select **All** from the **Select Store Type to add trusted certificate** dropdown. Select **Import from file** and **Browse** to the location of the certificate file supplied by Kofax beside **Please select a file**. Click on **Retrieve Certificate**.



Verify the certificate information and then click on **Commit** to store the certificate.



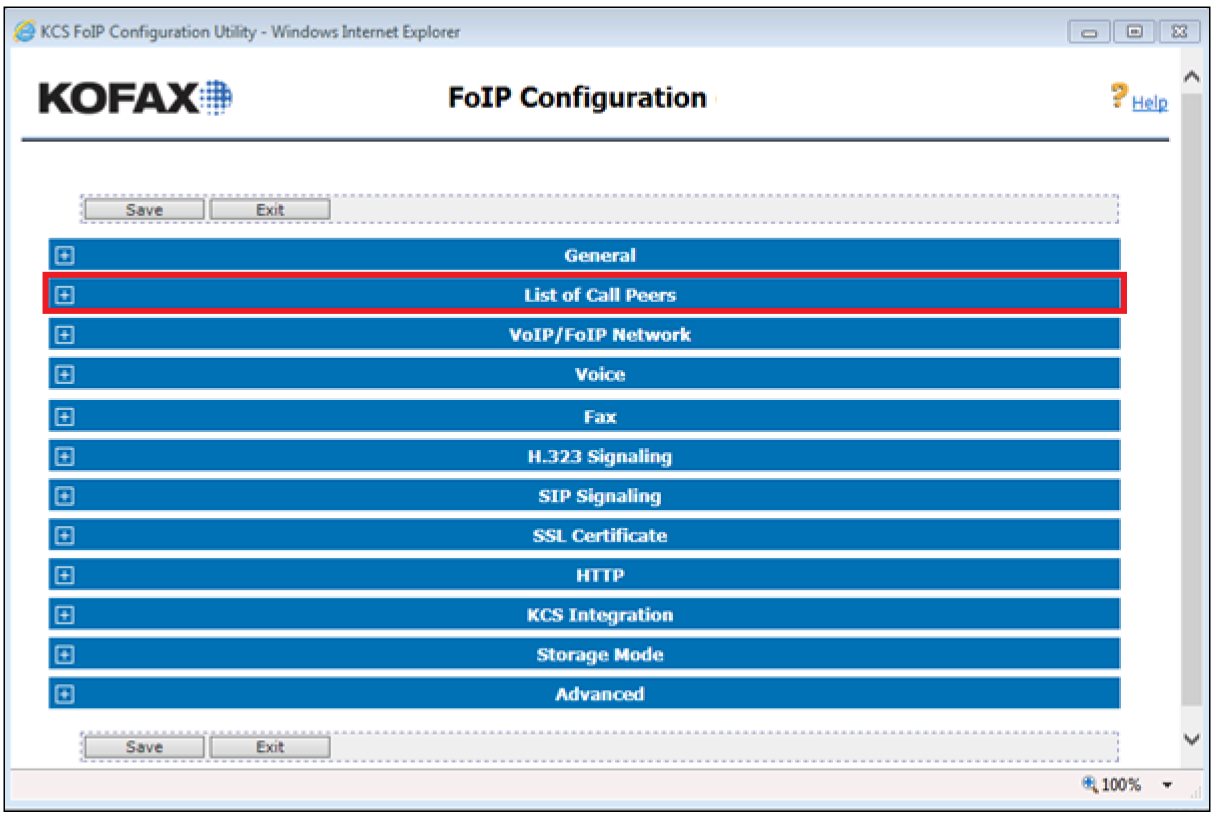
7. Configure Kofax Communication Server

The Kofax Server is provided, installed and implemented by Kofax. Only those configuration details concerning the interface to Avaya are shown within this section. The web-based Kofax Server FoIP configuration utility was used to configure the interface to Session Manager Open the KCS FoIP configuration utility from the shortcut on the Kofax Server desktop. The configuration operations described in this section can be summarized as follows:

- Configure List of Call Peers
- Configure Fax
- Configure SIP Signaling
- Configure KCS Integration

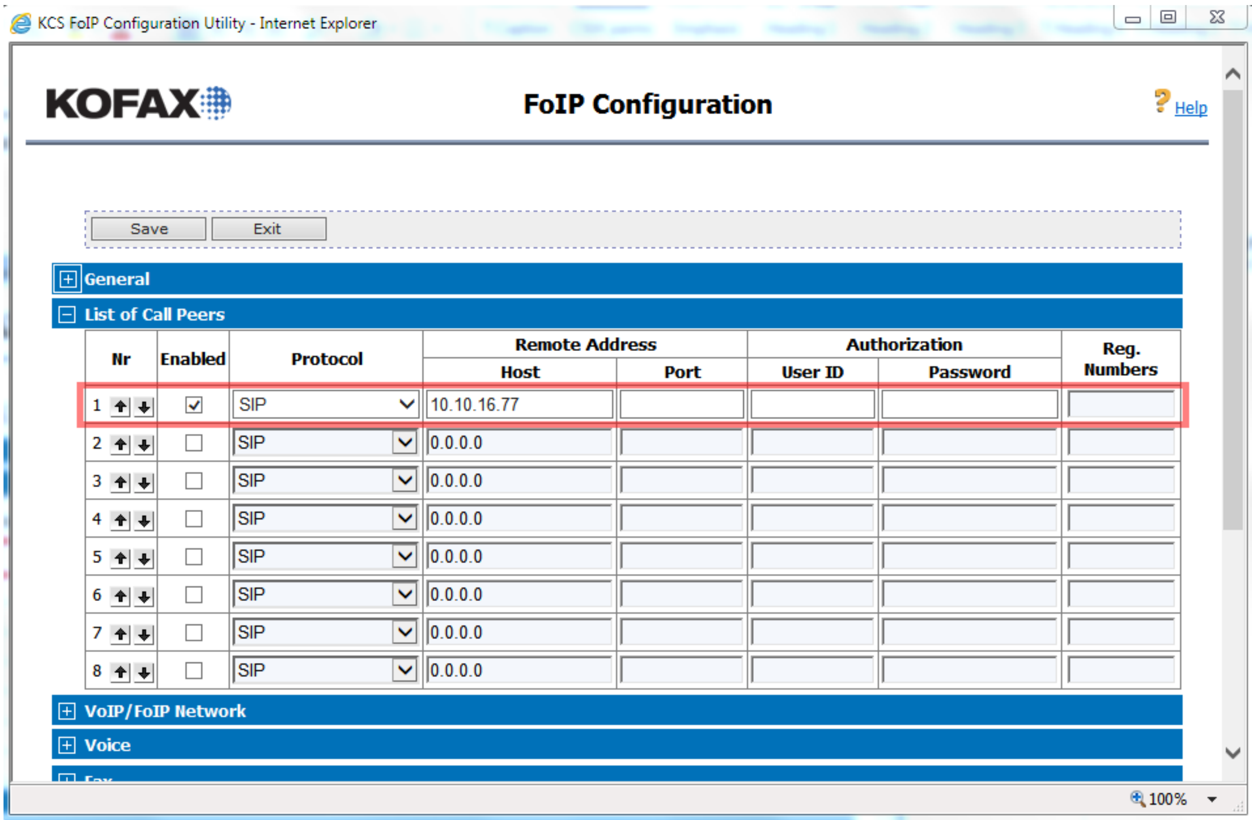
7.1. Configure List of Call Peers

Once the KCS FoIP configuration utility opens expand List of Call Peers menu item.



Once the **List of Call Peers** menu item opens complete the following for a free **Host**:

- **Enabled** Click on the Check box
- **Protocol** Select **SIP** from the dropdown box
- **Host** Enter the IP address of the Session Manager SIP Signaling Interface (see **Section 5.1**)



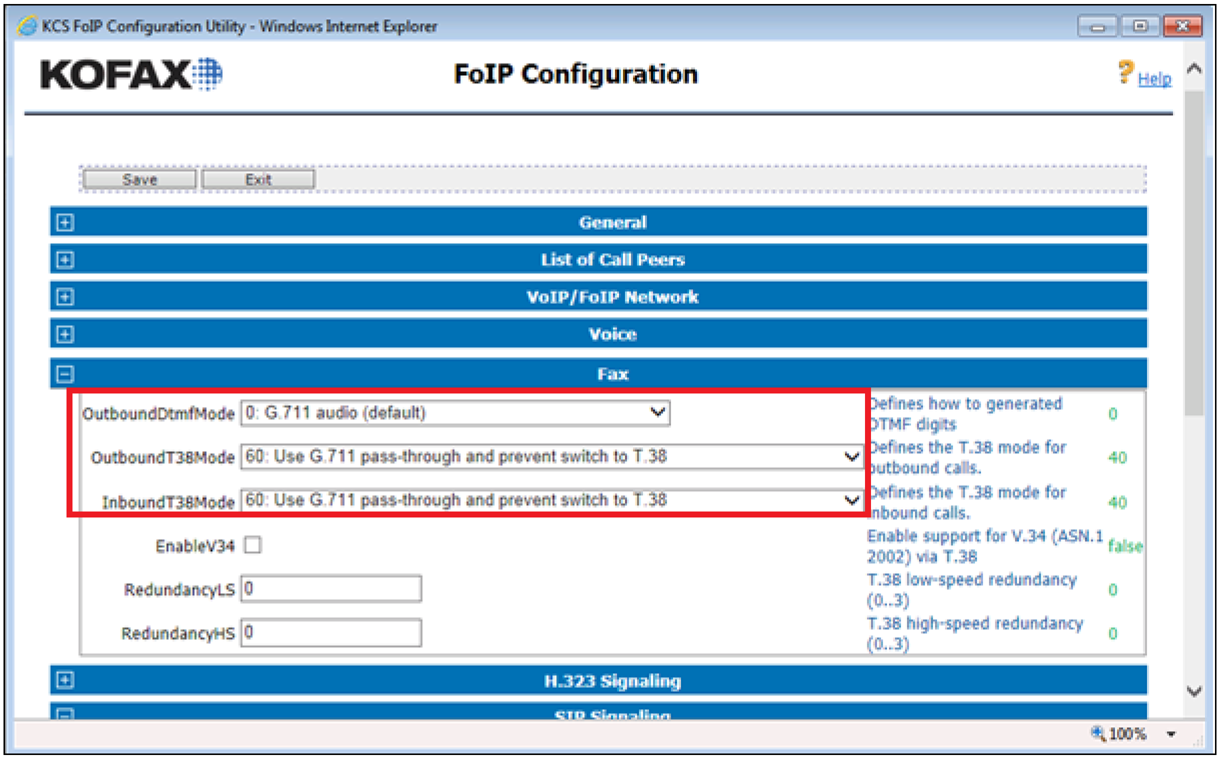
7.2. Configure Fax

Fax can be configured for **G.711 Pass-through**.

7.2.1. G.711 Pass-through

If only G.711 pass-through support is required, complete the following:

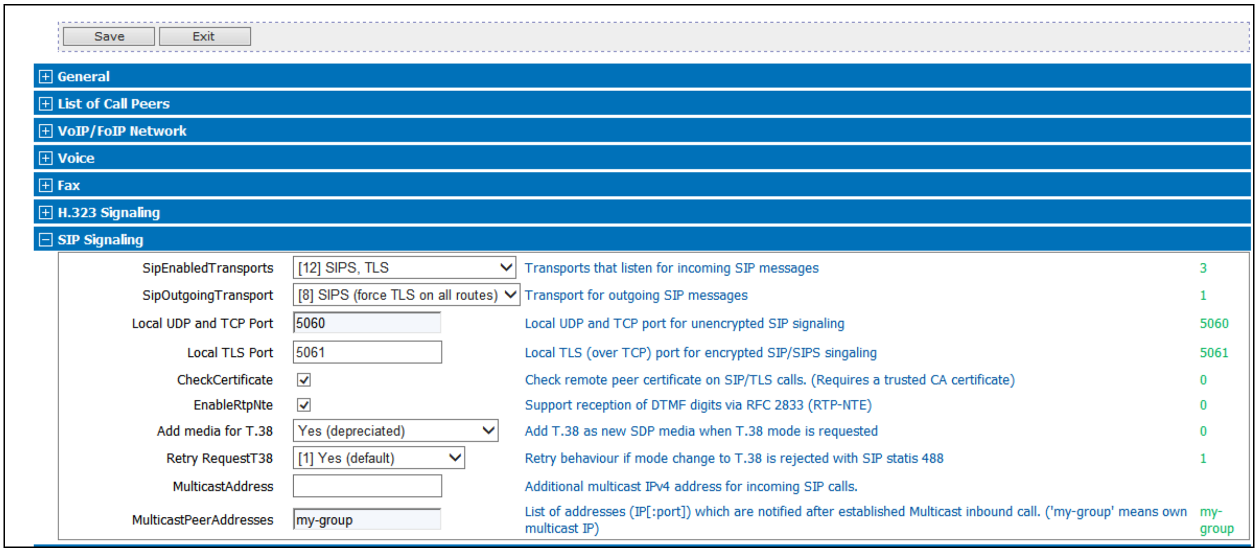
- **OutboundTDtmfMode** Select **0: G711 audio (default)** from the dropdown box
- **OutboundT38Mode** Select **60. User G.711 pass-through and prevent switch to T.38** from the dropdown box
- **InboundT38Mode** Select **60. User G.711 pass-through and prevent switch to T.38** from the dropdown box



7.3. Configure SIP Signaling

Open the **SIP Signaling** menu item and complete the following:

- **SipEnabledTransport** Select **[12] SIPS, TLS** from the dropdown box
- **SipOutgoingTransport** Select **[8] SIPS (force TLS on all routes)** from the dropdown box
- **Local TLS Port** Enter **5061**

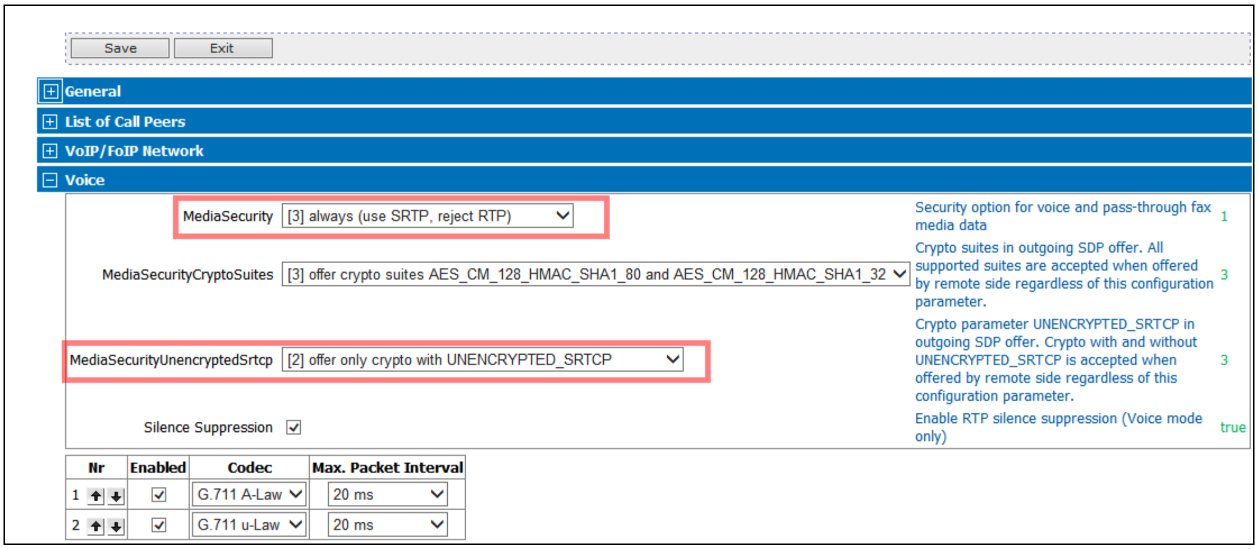


7.4. Configure Voice (Media Security)

Open the Voice menu item and complete the following:

- **MediaSecurity** Select **[3] always (use SRTP, reject RTP)**
- **MediaSecurityUnencryptedSrtcp** **Select [2] offer only crypto with UNENCRYPTED_SRTCP**

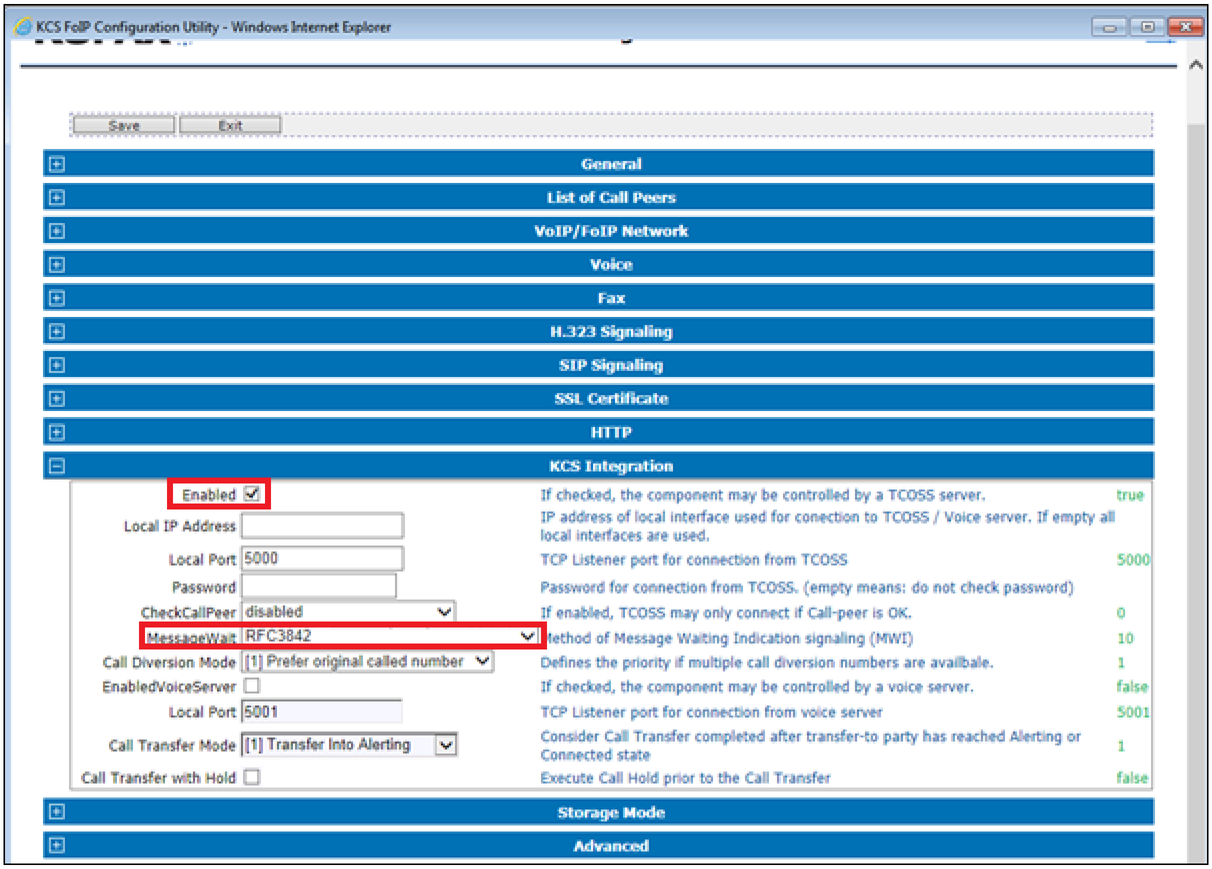
Note: Please read the notes on **Encrypted SRTCP** parameter in the **Section 5.6**



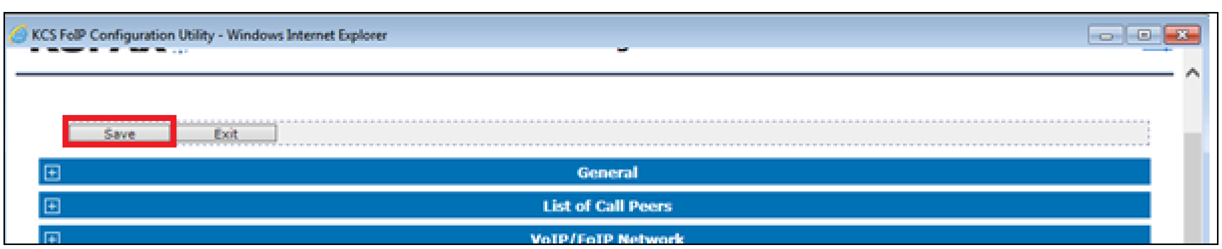
7.5. Configure KCS Integration

KCS Integration is configured if Message Waiting Indication is used to signal if a fax is in the fax recipient's in-box. Complete the following to configure KCS Integration:

- **Enabled** Check the check box
- **MessageWait** Select **RFC3842** from the dropdown box



Once the configuration is complete click on the **Save** button as shown in the screenshot below.



8. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the Avaya and Kofax Communication Server solution.

8.1. Verify the signaling group status

Using the SAT terminal, enter the **status signaling-group <n>** command, where **<n>** is the number of the SIP signaling group which connects to Session Manager. Verify that the **Group State** is **in-service**.

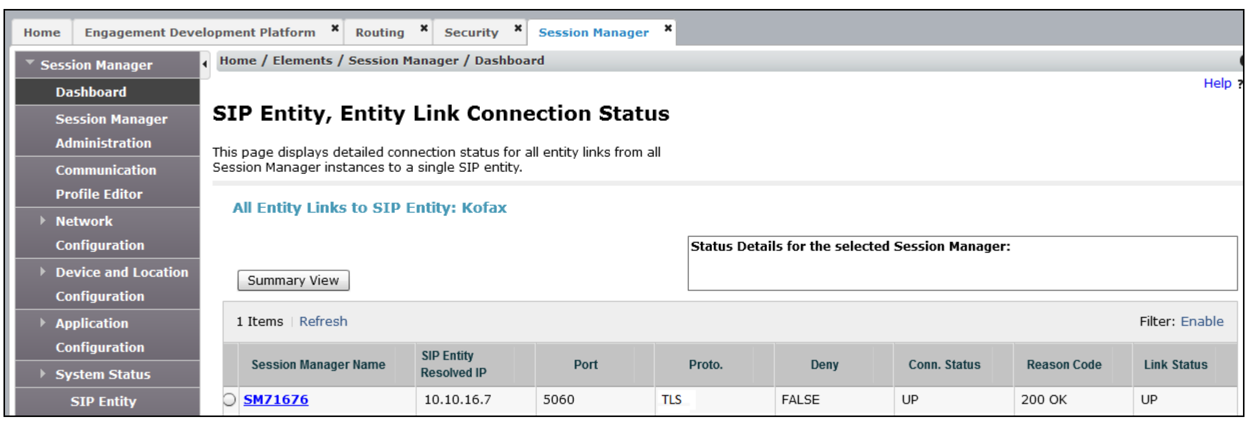
```
status signaling-group 1
                         STATUS SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

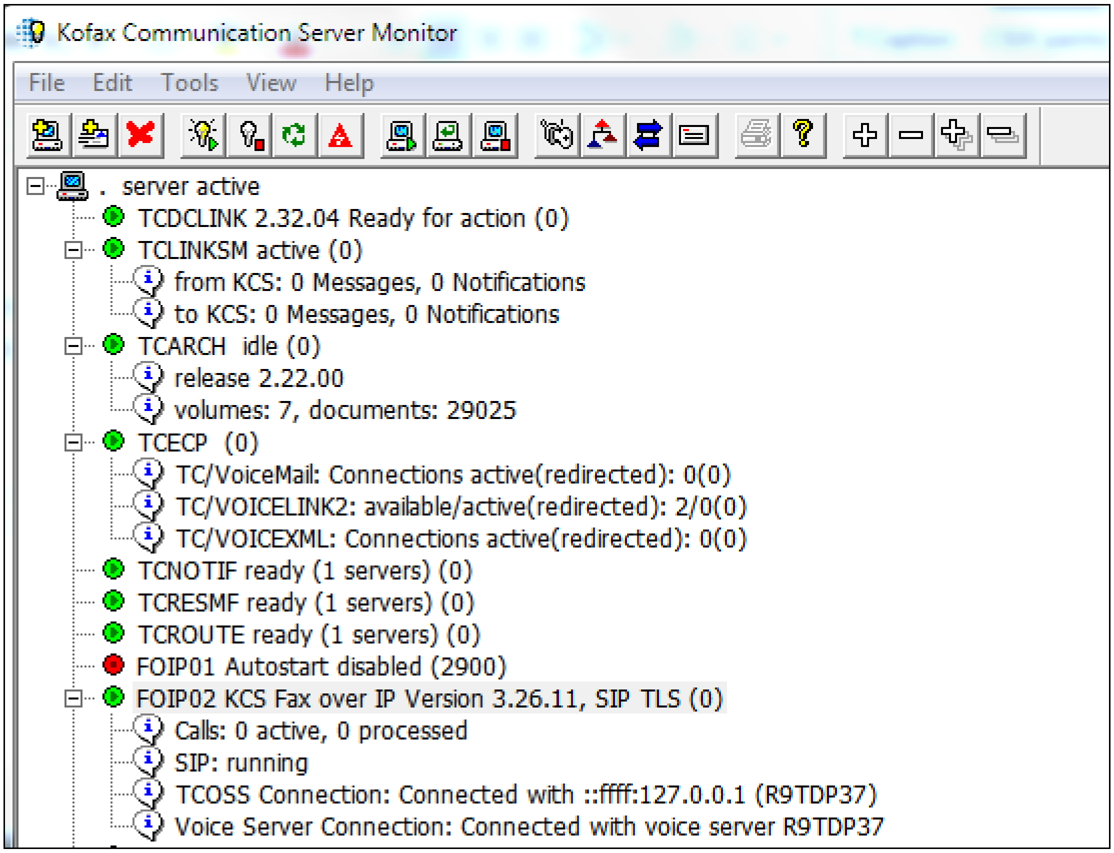
8.2. Verify the SIP Entity Link status for the Kofax Communication server

From System Manager select **Session Manager** from under the **Elements** column, (not shown). When the **Session Manager** tab opens select **System Status** followed by **SIP Entity Monitoring**, then click on Kofax SIP Entity created in **Section 6.4,** ensure that the **Conn. Status** is **Up**, the **Reason Code** is **200OK** and the **Link Status** is **Up**.



8.3. Verify Kofax Communication Server SIP Status

Start the Kofax Communication Server monitor and verify that **SIP** is in the **running** state.



8.4. Verify that faxes are sent and received from the Kofax Communication Server

Send and receive multipage faxes, ensure the faxes are successfully sent and received and are legible, confirm that the caller ID and fax details are correct.

9. Conclusion

These Application Notes describe the configuration steps required for Kofax Communication Server to interoperate with an Avaya Aura® Communication Manager 7.0.1 and Avaya Aura® Session Manager 7.0.1. All test cases have passed and met the objectives outlined in **Section 2.2**.

10. Additional References

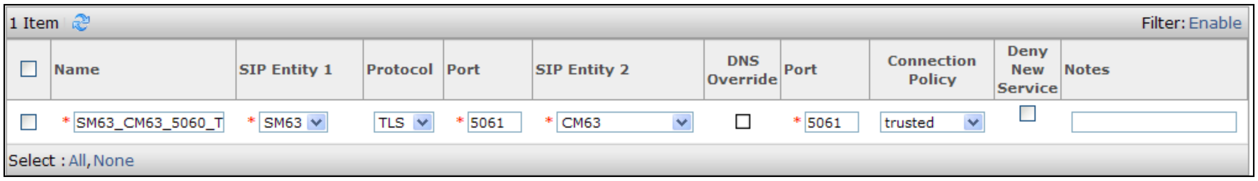
This section references the Avaya and Kofax documentation that is relevant to these Application Notes. Avaya product documentation, including the following, are available at: http://support.avaya.com

[1] *Administering Avaya Aura® Communication Manager*, Release 7.0.1, August 2017.
[2] *Administering Avaya Aura® Session Manager*, Release 7.0.1, 2017.
[3] *Administering Avaya Aura® System Manager*, Release 7.0.1, 2017.

Product Documentation for Kofax can be at the following location:
http://www.kofax.com/business-communication-software/

Appendix A

Entity Link between Session Manager and Communication Manager.



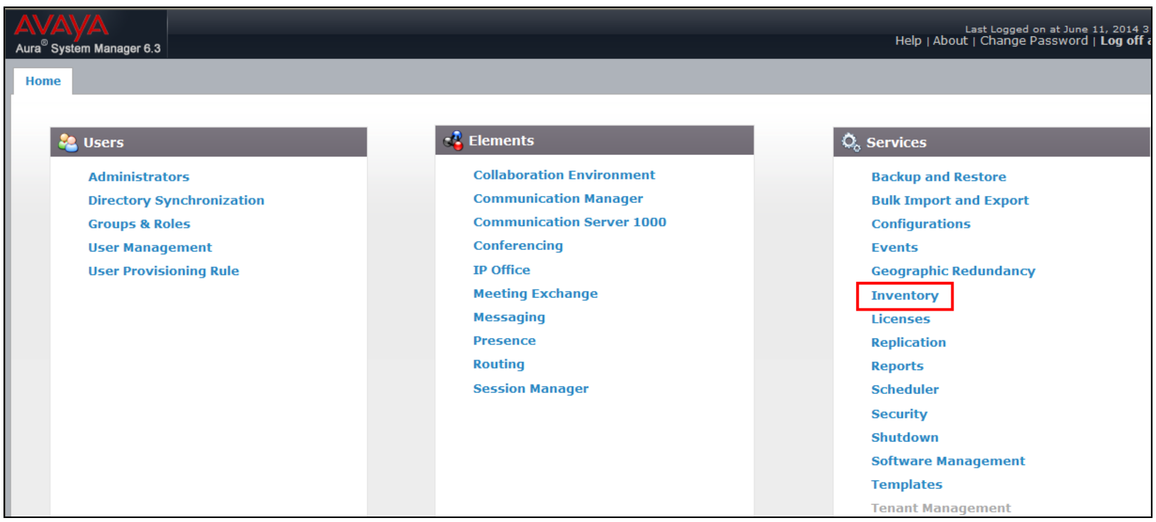
1 Item | Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
☐	* SM63_CM63_5060_T	* SM63	TLS	* 5061	* CM63	☐	* 5061	trusted	☐	

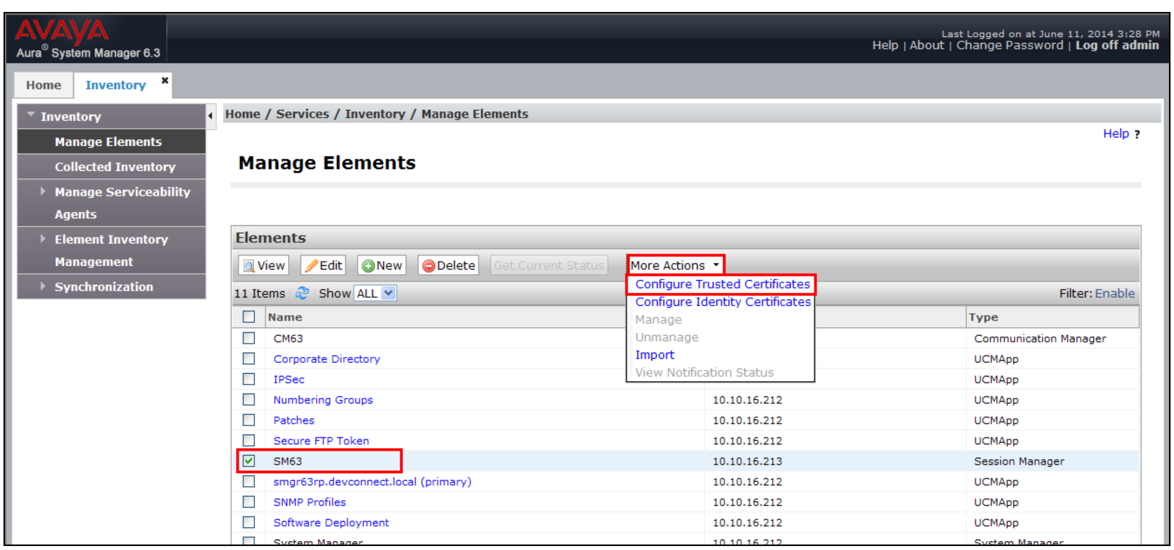
Select : All, None

Appendix B

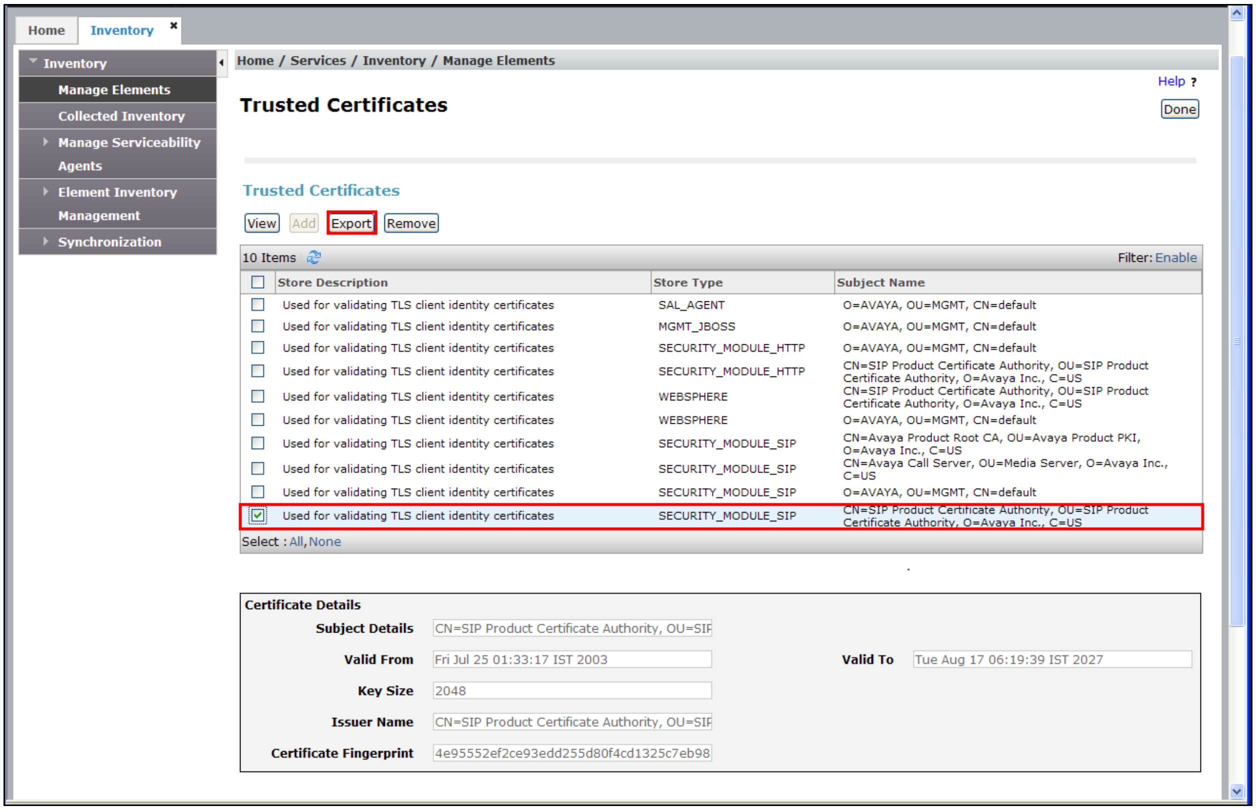
To export the Session Manager trusted certificates follow the steps below.
After logging into System Manager go to **Home → Services → Inventory**.



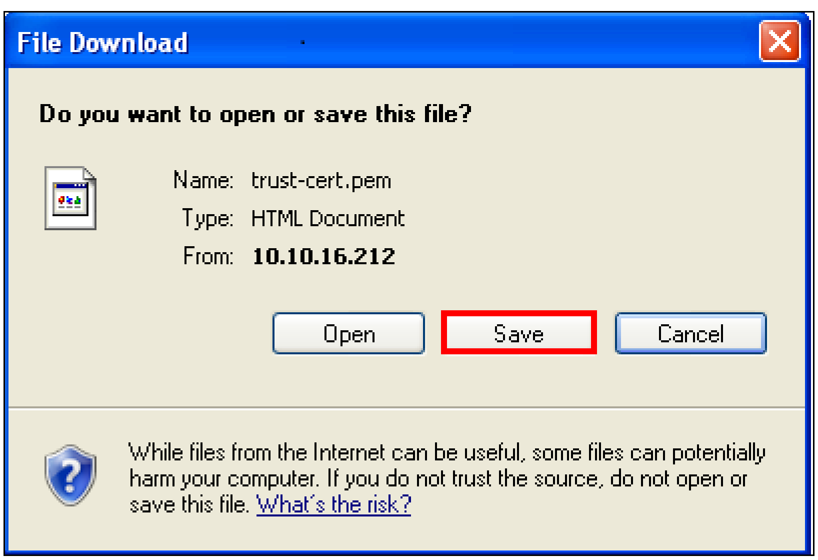
Select **Manage Elements** and click on Session Manager Element (i.e. SM63). From the **More Actions** dropdown box select **Configure Trusted Certificates**.



Once the **Trusted Certificates** screen open check the **CN=SIP Product Certificate Authority, OU=SIP Product Certificate, O=Avaya Inc., C=US** check box. Click the **Export** button to export the certificate.



When the **File download** window opens click on the **save** button and chose a location to store the Certificate. The file stored will then be required to be installed on the Kofax Server.



Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at *devconnect@avaya.com*.