Avaya Solution & Interoperability Test Lab

Application Notes for Kofax Communication Server with Avaya Aura® Communication Manager and Avaya Aura® Session Manager Interconnection via 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 and Avaya Aura® Session Manager. Kofax Communication Server communicates with Session Manager via a SIP trunk. Kofax Communication Server provides fax server functionality.

Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the member's test lab.

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1. Introduction

These Application Notes describe the configuration used to enable Kofax Communication Server to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Kofax Communication Server offers various telephony features. However, the only feature included in the Avaya conformance-tested feature set is fax functionality.

Kofax Communication Server fax features allow fax messages to be sent/received to/from both local and PSTN fax endpoints, and be subsequently printed or archived.

2. General Test Approach and Test Results

The compliance testing between Kofax Communication Server and Communication Manager 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 telephones, as well as a SIP connection to Session Manager.
- The Session Manager was configured to connect to both Communication Manager and Kofax Communication Server via SIP trunks.
- Kofax Communication Server was configured to connect to Session Manager.

2.1. Interoperability Compliance Testing

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

- The following test scenarios were used to test the various fax features of the Kofax Communication Server:
- Basic fax send via local and PSTN connection with G.711A and G.711MU codecs.
- Basic fax receive via local and PSTN connection with G.711A and G.711MU codecs.
- Forwarding of a fax from a local extension to the Kofax Communication Server.
- Transfer of a fax call from a local extension to the Kofax Communication Server.
- Verification of correct Transmitting Subscriber Identification composition for sent and received fax messages.
- Kofax's robustness was tested as follows:
- Verifying its ability to recover from power interruptions to the server.
- Verifying its ability to recover from interruptions to the LAN connection between the Kofax and the network.

2.2. Test Results

The test requirements specified in **Section 2.1** were tested successfully.

2.3. Support

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

3. Reference Configuration

The following diagram shows the configuration used for compliance testing.

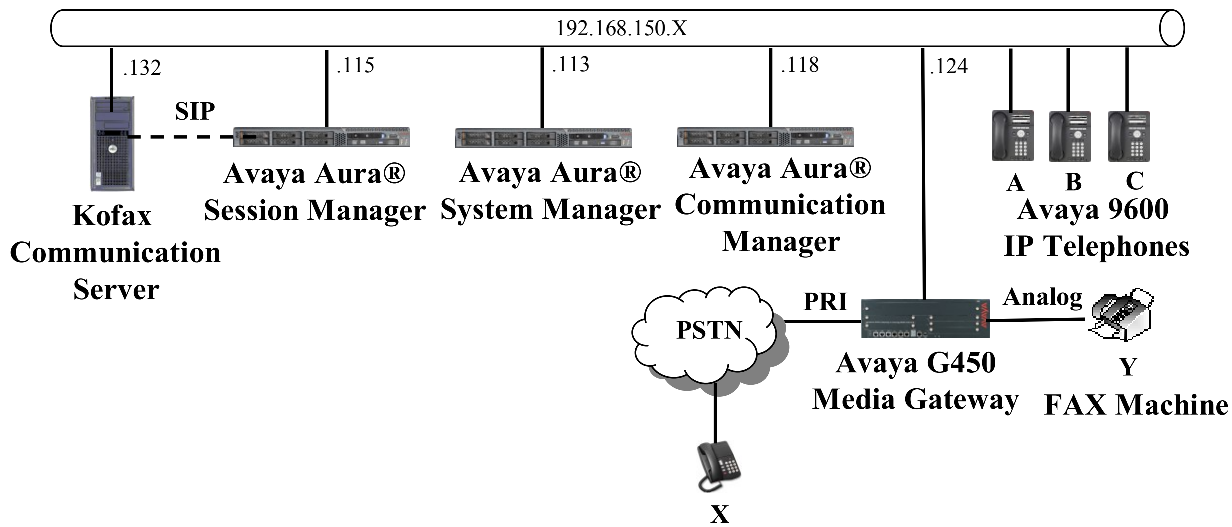

Figure 1: **Kofax Test Configuration**

Kofax Communication Server uses a SIP trunk interface to Session Manager and thus does not register individual SIP endpoints.

The endpoint extension numbers used for testing are shown in the following table.

Endpoint	Type	Extension	PSTN Number
A	Avaya 9640G SIP Phone	2370	+49 1111111111 2370
B	Avaya 9640G SIP Phone	2371	+49 1111111111 2371
C	Avaya 9640G H.323 Phone	2372	+49 1111111111 2372
X	ISDN PSTN Phone		+49 222222 6174
Y	Group 3 Fax Machine	2366	+49 1111111111 2366

Table 1: Endpoint Extension Assignment

4. Equipment and Software Validated

Component	Version
Avaya Aura® Communication Manager	CM 6.0.1, GA load 510.1, with 18621
Avaya Aura® Session Manager	SM software 6.1.0.0.610023
Avaya Aura® System Manager	System Manager software 6.1.4.0 Patch 06_01_SP0_r873
Avaya G450 Media Gateway	31.18.1
Avaya MM710AP PRI interface	HW05 / FW021
Avaya MM711AP analog interface	HW27 / FW073
Avaya 96x0 SIP Phones	2.6.4
Avaya 96x0 H.323 Phones	3.1.1
Kofax Communication Server	9.1
Kofax IP Call Control	3.0.5

Table 2: Hardware/Software Component Versions

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section were all performed using the Communication Manager System Administration Terminal (SAT).

Only those configuration details concerning the interface to Kofax Communication Server are shown within this section. Thus, the details regarding the configuration of the interface to Avaya Aura® Session Manager have been omitted.

5.1. Verify system-parameters customer-options

Use the **display system-parameters customer options** command to verify that Communication Manager is configured to meet the minimum requirements to run Kofax Communication Server. Those items shown in **bold** indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Parameter	Usage
Maximum Administered SIP Trunks Stations (Page 2)	The number of available licensed SIP trunks must be sufficient to accommodate the number of trunk members assigned to the trunk group used to interface to Session Manager in **Figure 8**.

Table 3: System-Parameters Customer-Options Parameters

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 4000  0
           Maximum Concurrently Registered IP Stations: 2400  1
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 50    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 4000  10
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0
```

Figure 2: System-Parameters Customer-Options Screen, Page 2

5.2. Node Names

Use the **change node-names ip** command to configure the node name for the Session Manager SIP trunk.

Parameter	Usage
Name / IP Address	Enter an appropriate name to identify the Session Manager SIP trunk, along with the IP address of the trunk.

Table 4: Node-Names IP Parameters

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
asset               192.168.150.115
default             0.0.0.0
procr               192.168.150.118
procr6              ::
```

Figure 3: Node-Names IP Form

5.3. Dialplan

Use the **change dialplan analysis** command to configure the dial plan using the parameters shown below.

Dialed String	Usage
2	Make an entry for Avaya terminal extensions.
6	Make an entry for Kofax terminal extensions.
*8	Make an entry for the Trunk Access Code used in the SIP trunk group defined in **Figure 8**.

Table 5: Dialplan Analysis Parameters

```
change dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 4

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   2           4   ext
   6           4   ext
   *8          4   dac
```

Figure 4: Dialplan Analysis Form

5.4. Configure Network Region

Use the **change ip-network-region** command to assign an appropriate domain name to be used by Communication Manager. This name is also used in **Figure 17**.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: aura.dcffm
    Name: local
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
```

Figure 5: IP Network Region Form

5.5. Configure IP-Codec

Use the **change ip-codec-set 1** command to designate that the G.711A codec set used to communicate with Session Manager.

```
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.711A            n         2        20
 2:
```

Figure 6: IP-Codec-Set Form

5.6. Configure SIP Interface to Session Manager

Use the **add signaling-group** command to configure the Signaling Group parameters for the SIP trunk group. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type	Enter the Group Type as "sip".
Transport Method	Enter "tcp".
Near-end Node Name	Enter "procr" to designate the Processor Ethernet interface.
Near-end Listen Port	Enter "5060".
Far-end Node Name	Enter the name assigned to the SIP trunk to Session Manager configured in **Figure 3**.
Far-end Listen Port	Enter "5060".
Far-end Domain Name	Enter the domain name assigned to the network region in **Figure 5**.
Direct IP-IP Audio Connections	Enter "y" to turn on "shuffling".

Table 6: Signaling-Group Parameters for SIP Interface

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                           SIP Enabled LSP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: asset
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1

Far-end Domain: aura.dcffm
                                             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
```

Figure 7: Signaling Group Form

Use the **add trunk-group** command to configure the SIP interface to Session Manager. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type (page 1)	Specify the Group Type as “sip”.
Group Name (page 1)	Select an appropriate name to identify the device.
TAC (page 1)	Specify a trunk access code that can be used to provide dial access to the trunk group.
Service Type (page 1)	Designate the trunk as a “tie” line to a peer system.
Signaling Group (page 1)	Enter the number assigned to the SIP signaling group shown in **Figure 7**.
Number of Members (page 1)	Specify sufficient number of members to support the maximum simultaneous connections required.
Numbering Format (page 3)	Enter “private”.

Table 7: Trunk-Group Parameters for the SIP Interface

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

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: Local-to-CM                COR: 1        TN: 1        TAC: *801
   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: 10
```

Figure 8: Trunk Group Form, page 1

```
add 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
```

Figure 9: Trunk Group Form, page 3

5.7. Call Routing to Fax Server

Use the **change uniform-dialplan 0** command. Assign values for this command as shown in the following table.

Parameter	Usage
Matching Pattern	Enter the leading digit of the extensions assigned to the Kofax terminals.
Len	Enter the length of the extensions assigned to the Kofax terminals.
Net	Enter "aar".

Table 8: Uniform-Dialplen Parameters

```
change uniform-dialplan 0                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                Percent Full: 0

  Matching                   Insert              Node
  Pattern       Len Del      Digits      Net Conv Num
 6               4   0                   aar  n
```

Figure 10: Uniform-Dialplan Form

Use the **change aar analysis 0.** Assign values for this command as shown in the following table.

Parameter	Usage
Dialed String	Enter the leading digit of the extensions assigned to the Kofax terminals.
Min / Max	Enter the length of the extensions assigned to the Kofax terminals.
Route Pattern	Enter the number of the route pattern described in **Figure 12**.
Call Type	Enter "aar".

Table 9: AAR Analysis Parameters

```
change aar analysis 0                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 2

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    6                     4    4      6        aar          n
```

Figure 11: AAR Analysis Form

Use the **change route-pattern <n>** command, where **<n>** is the route pattern to route calls for Kofax terminals from Communication Manager to Session Manager. Assign values for this command as shown in the following table.

Parameter	Usage
Pattern Name	Enter a descriptive name to identify the route pattern.
Grp No	Enter the number of the SIP trunk which connects to Session Manager, which is defined in **Figure 8**.

Table 10: Route-Pattern Parameters

```
change route-pattern 6                                          Page   1 of   3
                    Pattern Number: 6   Pattern Name: FAX Server
                             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: 1    0                                                              n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     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
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

Figure 12: Route-Pattern Form

Use the **change ars analysis 0** command to select a route pattern for calls to the PSTN, as shown in the following table.

Parameter	Usage
Dialed String	Enter the leading digit of the extensions assigned for outgoing PSTN calls.
Min / Max	Enter the length PSTN numbers.
Route Pattern	Enter the number of the route pattern described in **Figure 12**.
Call Type	Enter “pubu”.

Table 11: ARS Analysis Parameters

6. Configure Avaya Aura® Session Manager

This section illustrates relevant aspects of the Avaya Aura® Session Manager configuration used in the verification of these Application Notes.

Session Manager is managed via Avaya Aura® System Manager. Using a web browser, access "https://<ip-addr of System Manager>/SMGR". In the **Log On** screen, enter appropriate **Username** and **Password** and press the **Log On** button.

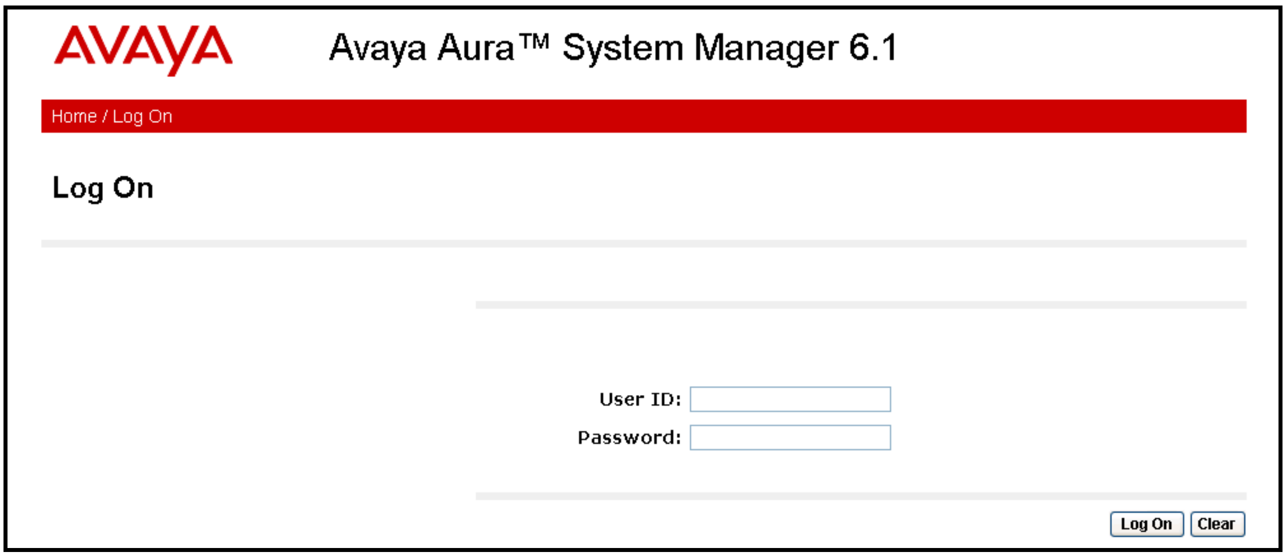

Figure 13: System Manager Login Screen

Once logged in, a **Home Screen** is displayed.

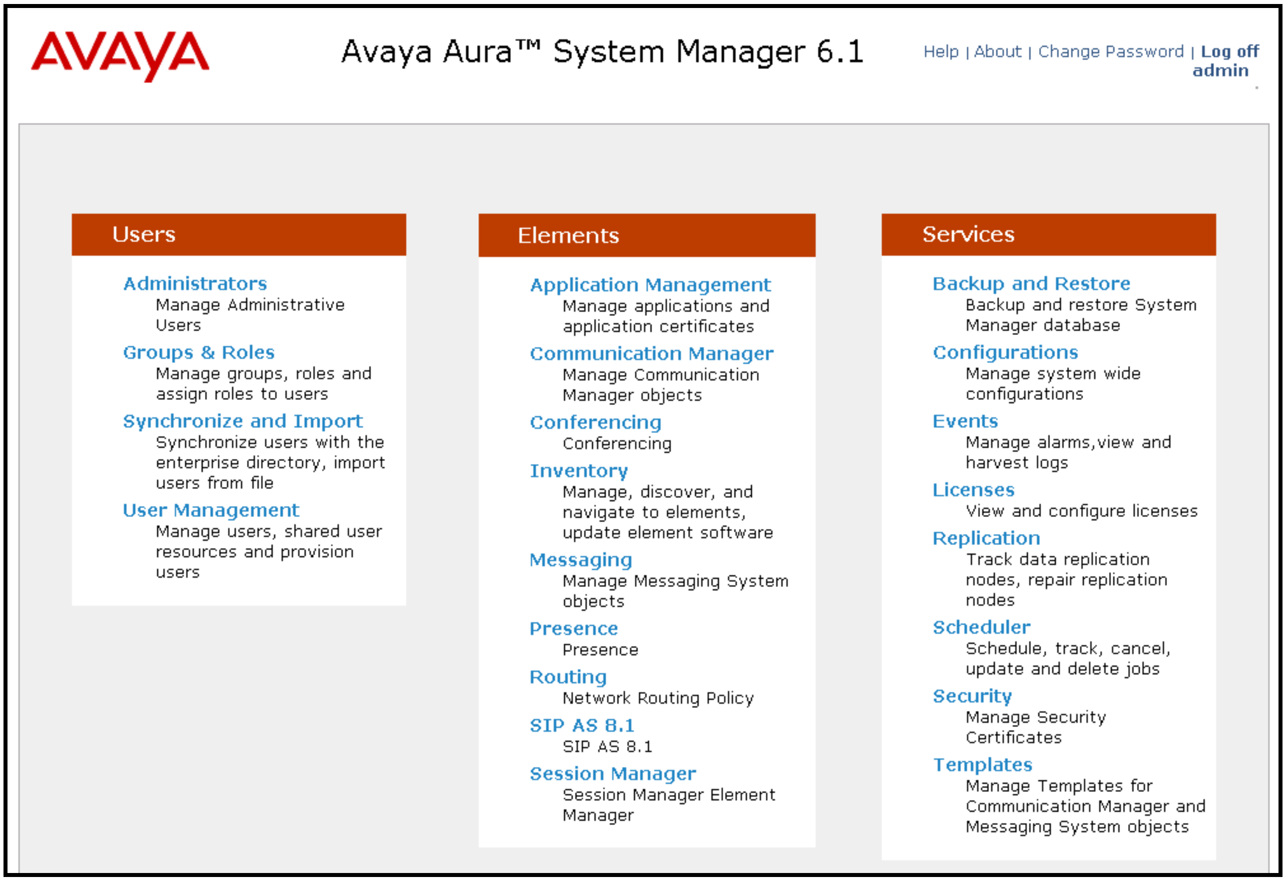

Figure 14: System Manager Home Screen

6.1. Routing

When Routing is selected, the right side outlines a series of steps.

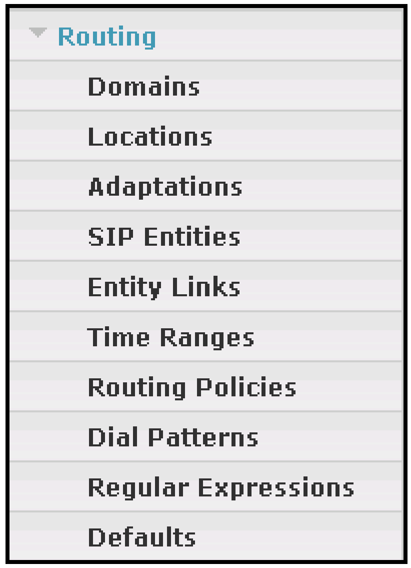

Figure 15: System Manager Call Routing Menu

The sub-sections that follow are in the same order as the steps outlined under **Introduction to Network Routing Policy (NRP)** in the abridged screen shown below. In these Application Notes, all these steps are illustrated with the exception of Step 9, since "Regular Expressions" were not used.

Introduction to Network Routing Policy

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).

Step 2: Create "Locations"

Step 3: Create "Adaptations"

Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"

Step 5: Create the "Entity Links"

- Between Session Managers
- Between Session Managers and "other SIP Entities"

Step 6: Create "Time Ranges"

- Align with the tariff information received from the Service Providers

Step 7: Create "Routing Policies"

- Assign the appropriate "Routing Destination" and "Time Of Day"

(Time Of Day = assign the appropriate "Time Range" and define the "Ranking")

Step 8: Create "Dial Patterns"

- Assign the appropriate "Locations" and "Routing Policies" to the "Dial Patterns"

Step 9: Create "Regular Expressions"

- Assign the appropriate "Routing Policies" to the "Regular Expressions"

Each "Routing Policy" defines the "Routing Destination" (which is a "SIP Entity") as well as the "Time of Day" and its associated "Ranking".

Figure 16: System Manager Introduction to Routing Policy

6.1.1. Domains

To view or change SIP domains, select **Routing → Domains**. Click on the checkbox next to the name of the SIP domain and **Edit** to edit an existing domain, or the **New** button to add a domain. Click the **Commit** button after changes are completed. The domain name to be configured should be the same as was configured for the Communication Manager network region in **Figure 5**.

The following screen shows the list of configured SIP domains.

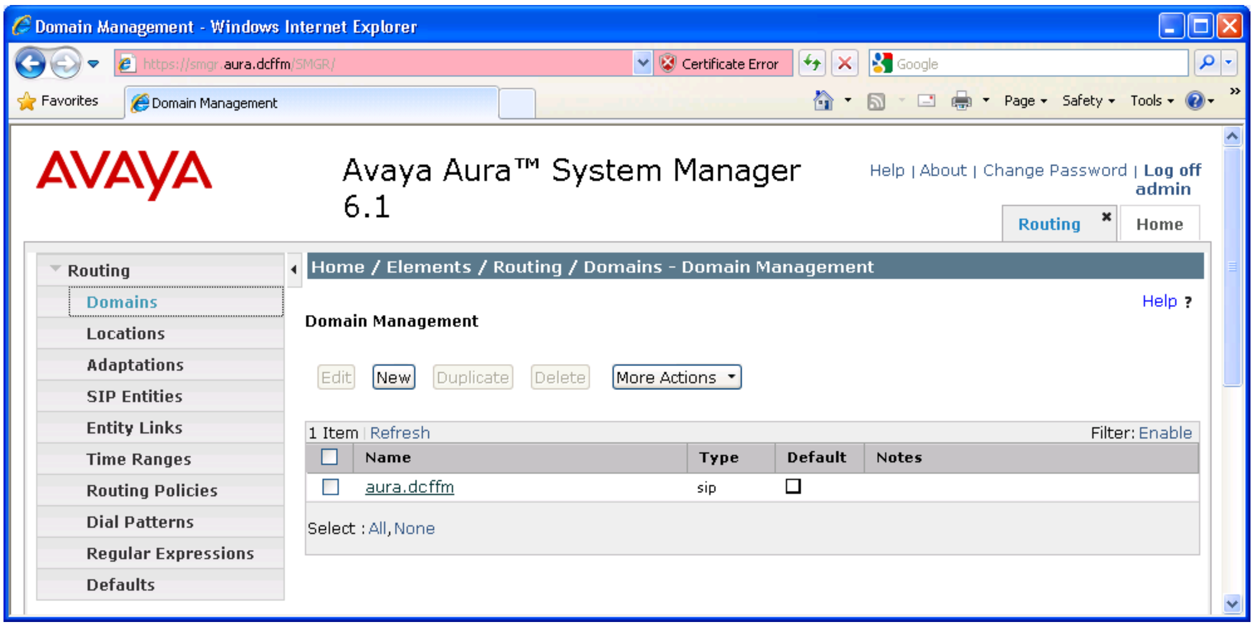

Figure 17: Session Manager Domains

6.1.2. Locations

To view or change locations, select **Routing → Locations**. The following screen shows an abridged list of configured locations. Click on the checkbox corresponding to the name of a location and **Edit** to edit an existing location, or the **New** button to add a location. Click the **Commit** button after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control.

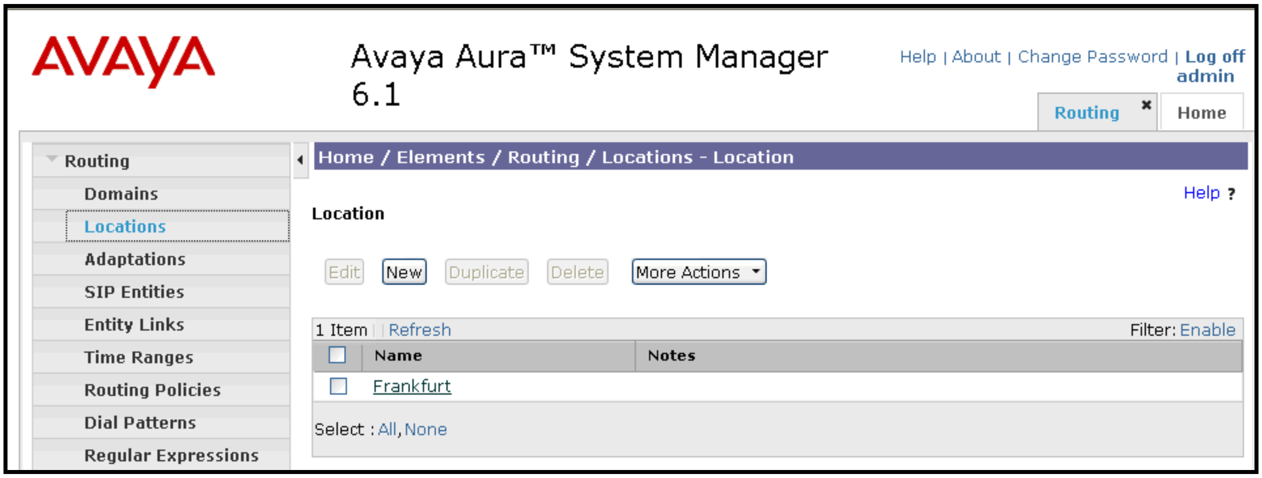

Figure 18: Session Manager Locations

6.1.3. SIP Entities

To view or change SIP elements, select **Routing → SIP Entities**. Click the checkbox corresponding to the name of an element and **Edit** to edit an existing element, or the **New** button to add an element. Assign values for this command as shown in the following table.

Parameter	Usage
Name	Enter an appropriate name to identify the SIP entity.
FQDN or IP Address	Enter the Kofax Communication Server IP address.
Location	Select the location defined in **Figure 18** from the drop-down menu.
Time Zone	Select the proper time zone from the drop-down menu.

Table 12: Route-Pattern Parameters

Click the **Commit** button after changes are completed.

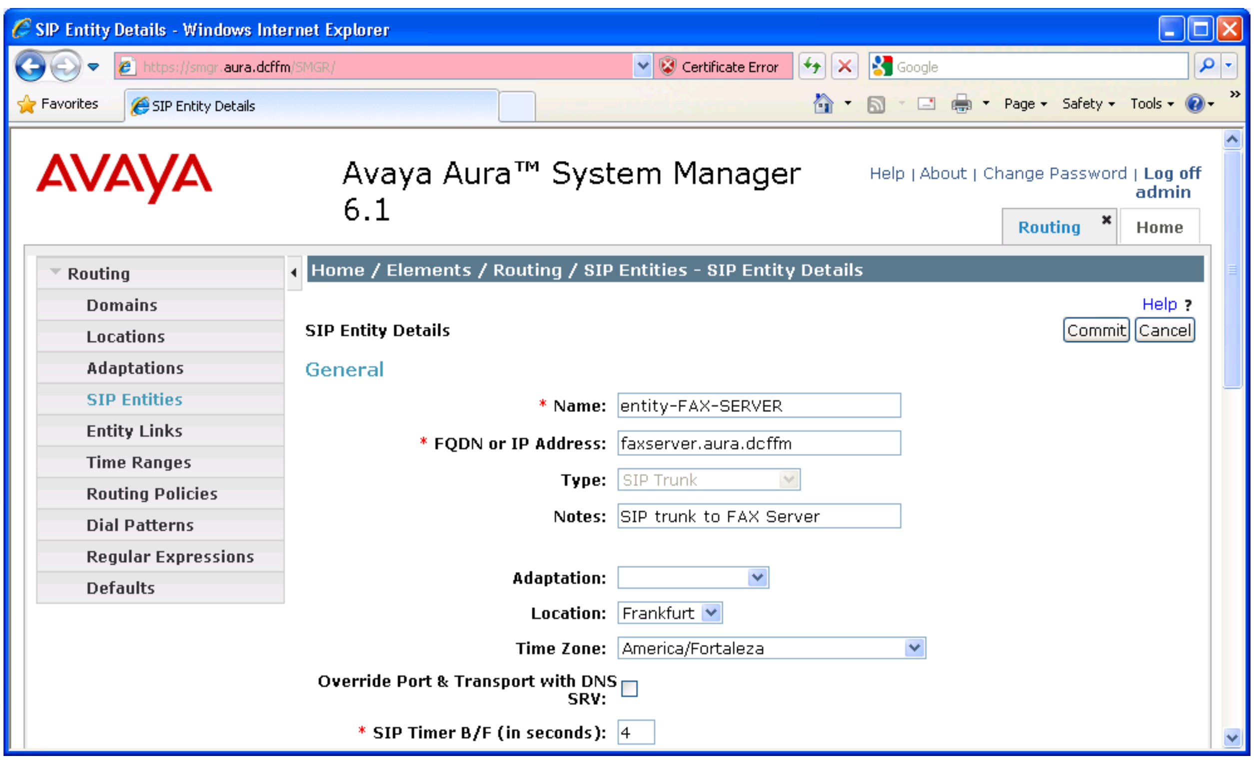

Figure 19: Session Manager SIP Entity for Kofax SIP Trunk

6.1.4. Entity Links

To view or change Entity Links, select **Routing → Entity Links**. Click on the checkbox corresponding to the name of a link and **Edit** to edit an existing link, or the **New** button to add a link. Assign values for this command as shown in the following table.

Parameter	Usage
Name	Select the SIP entity for Kofax server created in **Figure 19** from the drop-down menu.
SIP Entity 1 / Protocol / Port	Select the SIP entity for Session Manager, with the appropriate protocol and port.
SIP Entity 2 / Port	Select the SIP entity for the Kofax Communication Server ,with the appropriate port.
Trusted	Check this box.

Table 13: Entity Link Parameters

Click the **Commit** button after changes are completed.

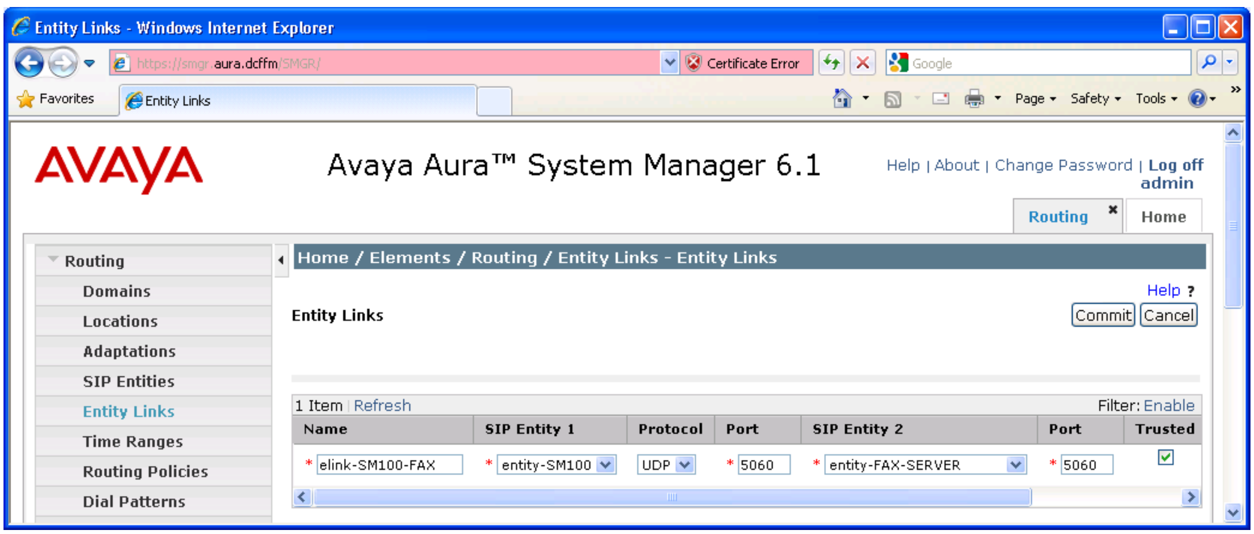

Figure 20: Session Manager Entity Link for Kofax SIP Trunk

6.1.5. Time Ranges

To view or change Time Ranges, select **Routing → Time Ranges**. The Routing Policies shown subsequently will use the "24/7" range since time-based routing was not the focus of these Application Notes.

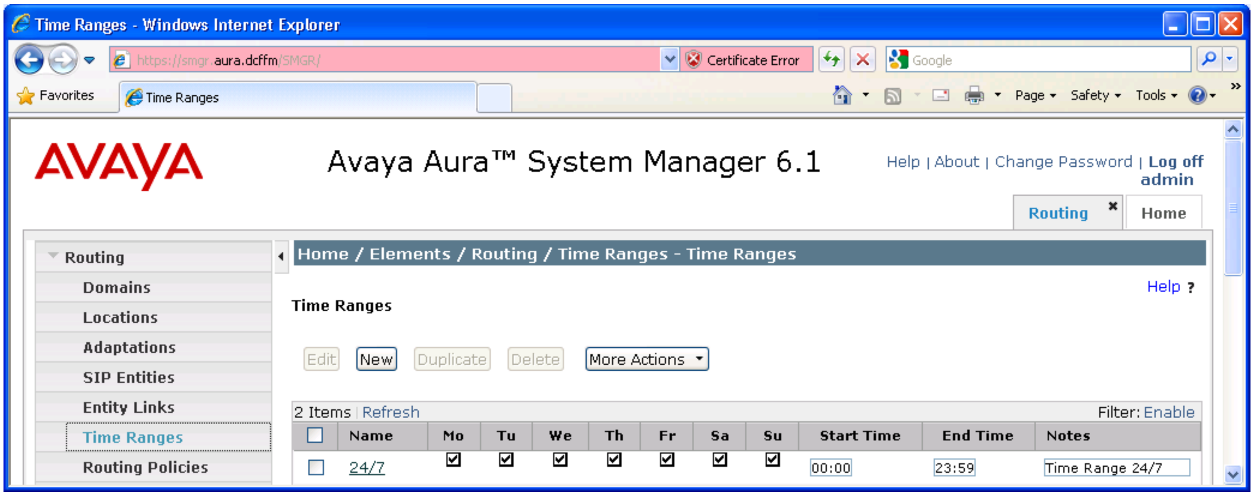

Figure 21: Session Manager Time Ranges

6.1.6. Routing Policies

To view or change routing policies, select **Routing → Routing Policies**. Click on the checkbox corresponding to the name of a policy and **Edit** to edit an existing policy, or **New** to add a policy. Enter a descriptive name for the routing policy, and select the Kofax server as the route destination by clicking "Select".

Click the **Commit** button after changes are completed.

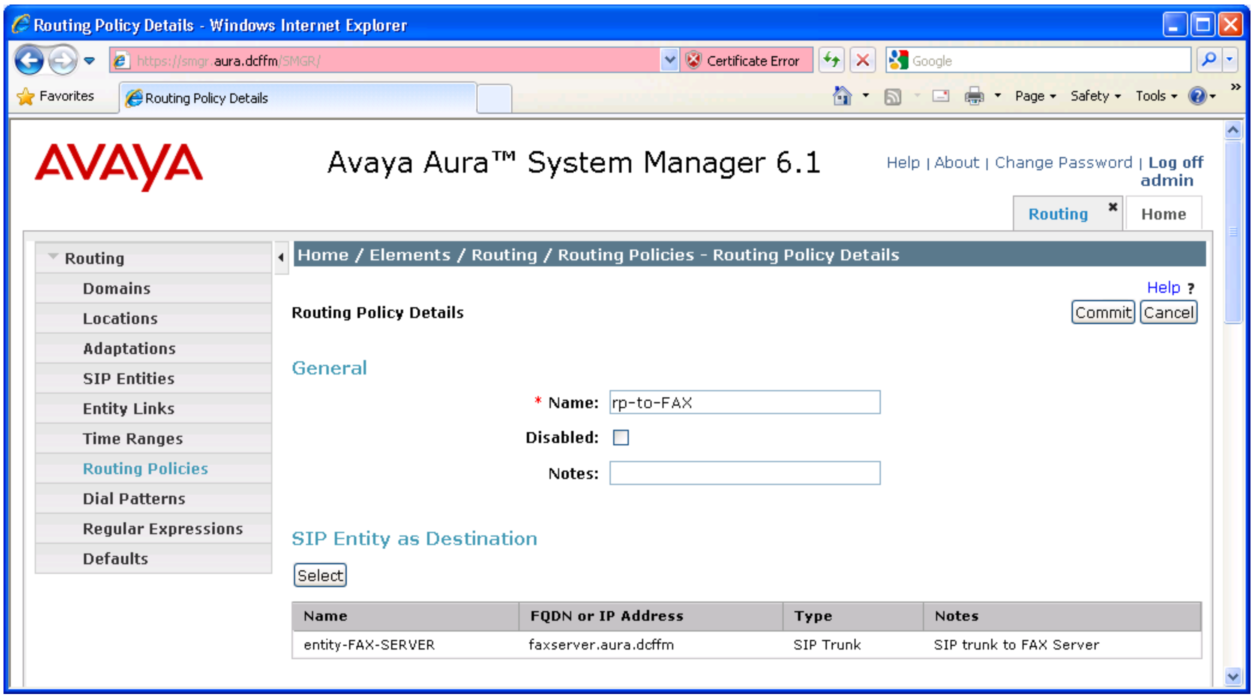

Figure 22: Session Manager Routing Policy for Calls to Kofax Endpoints

6.1.7. Dial Patterns

To view or change dial patterns, select **Routing → Dial Patterns**. Click on the checkbox corresponding to the name of a pattern and **Edit** to edit an existing pattern, or **New** to add a pattern. Assign values for this command as shown in the following table.

Parameter	Usage
Pattern	Enter the leading digits of the Kofax endpoint extensions.
Min	Enter the length of the Kofax endpoint extensions.
Max	Enter the length of the Kofax endpoint extensions.
SIP Domain	Select "aura.dcffm" from the drop-down menu.

Table 14: Dial Pattern Parameters

Click the "Add" button, select the originating location of "All", and the routing policy defined in **Figure 22**, and click the **Commit** button.

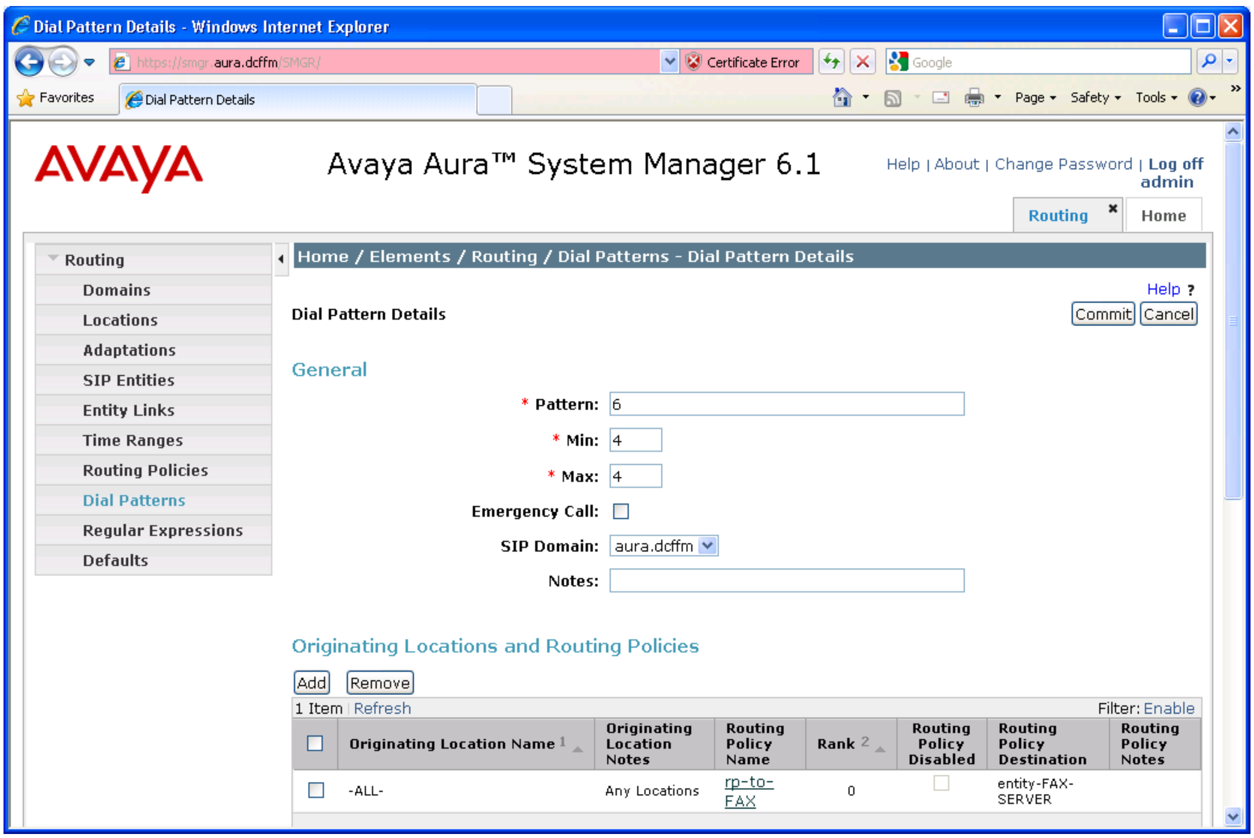

Figure 23: Session Manager Dial Pattern for Calls To Kofax Terminals

7. Configure Kofax Communication Server

Only those configuration details concerning the interface to Avaya are shown within this section.

For this purpose, the Web-based KCS FoIP configuration utility was used to configure the Kofax Communication Server.

Open the KCS FoIP configuration utility

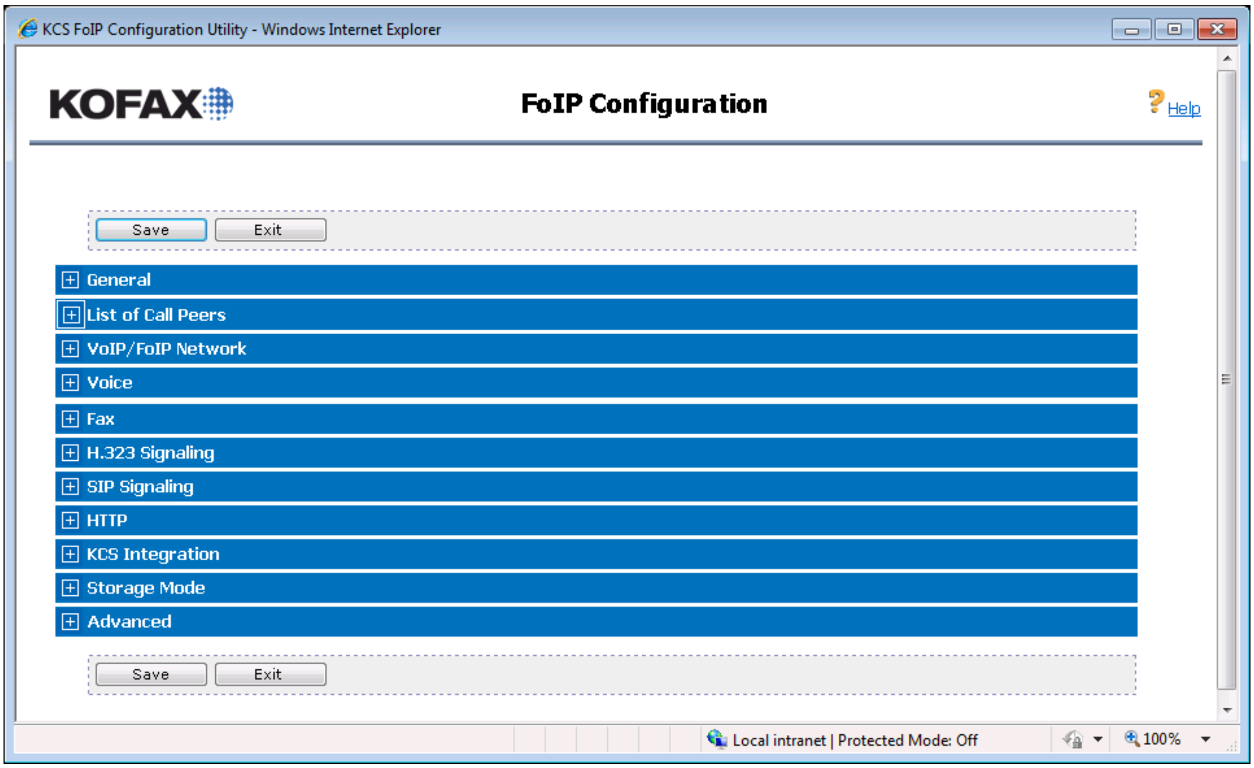

Figure 24: FoIP Configuration Initial Screen

Open the "List of Call Peers" menu item. For one of the elements which has a "Host" entry of "0.0.0.0", enter the items shown in the following table in the row.

Parameter	Usage
Enabled	Check this box.
Protocol	Select "SIP" from the drop-down menu.
Host	Enter the IP address of the Session Manager Ethernet interface.

Table 15: List of Call Peers Parameters

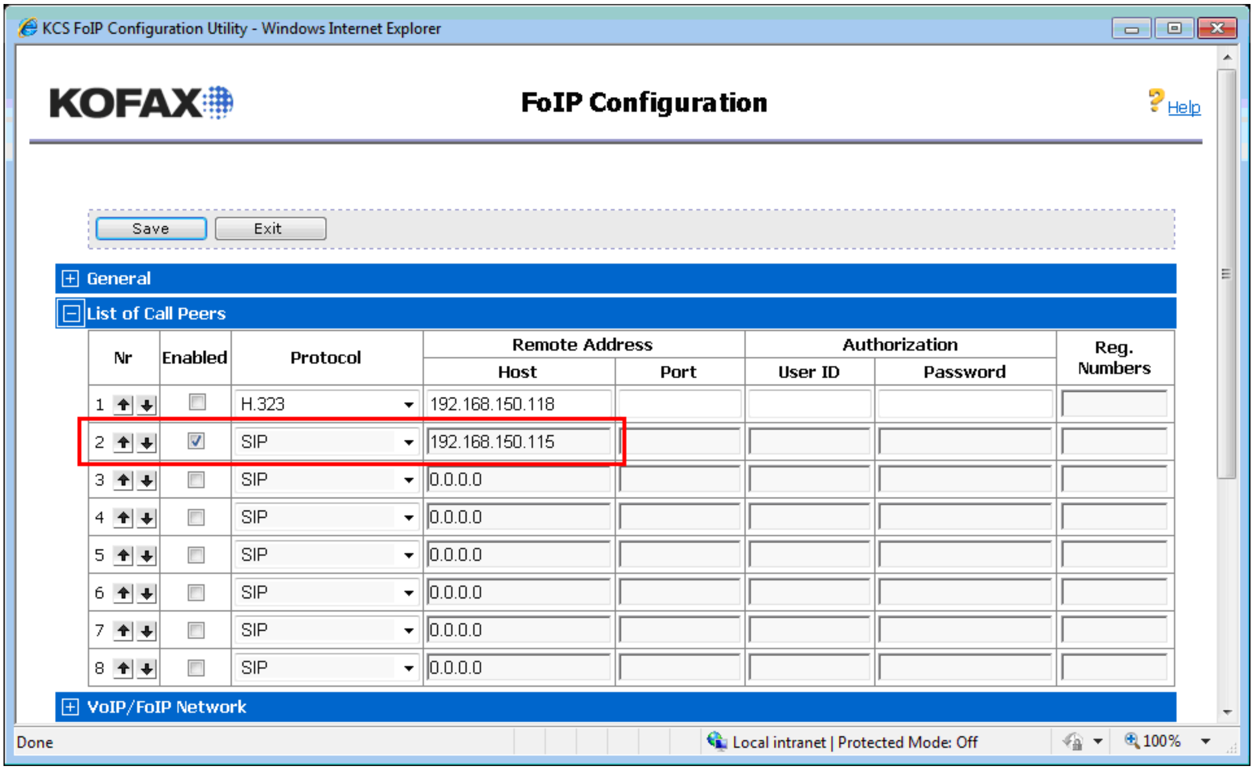

Figure 25: List of Call Peers Screen

From the top level menu, open the SIP Signaling menu item and check the “EnableRtpNte” check box.

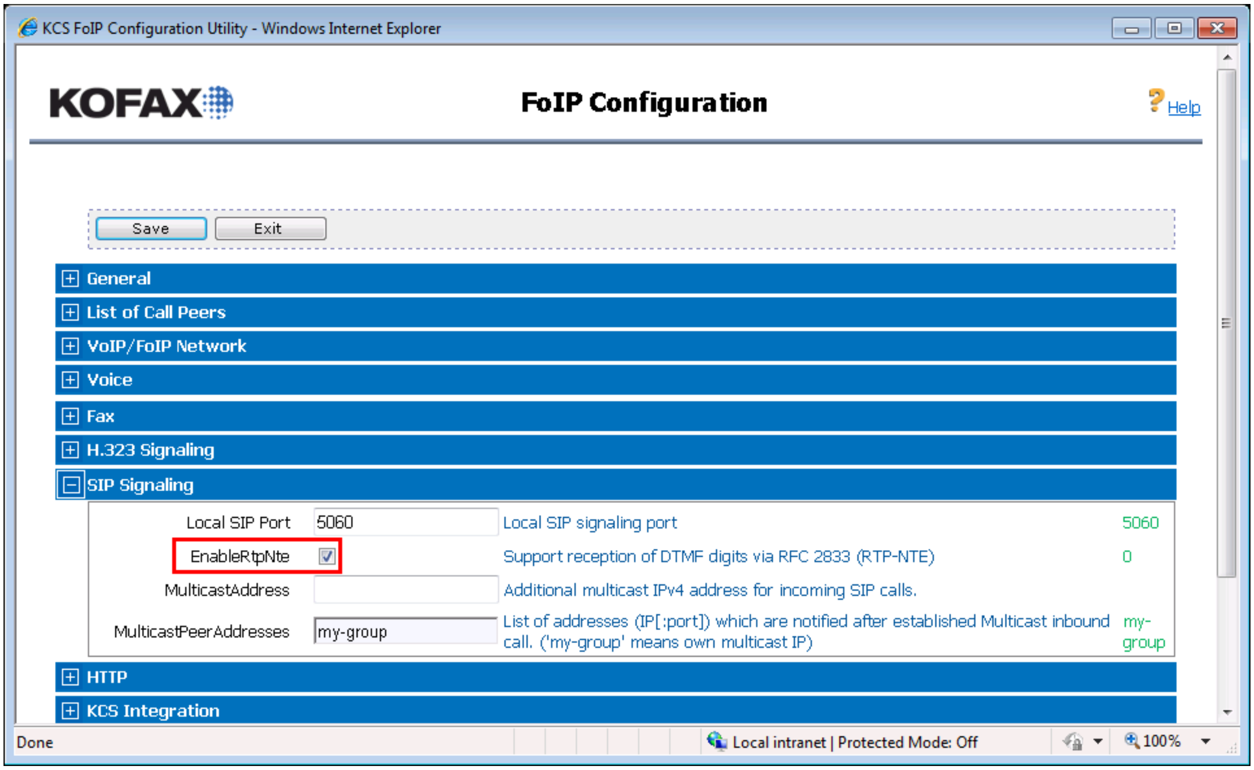

Figure 26: SIP Signaling Screen

8. Verification Steps

The correct installation and configuration of the Kofax Communication Server can be verified by performing the following steps shown below. 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 signaling group status is “in-service”.

```
status signaling-group 1
                         STATUS SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

Figure 27: Signaling Group Status

Start the Kofax Communication Server monitor and verify that the SIP status is in the “running” state.

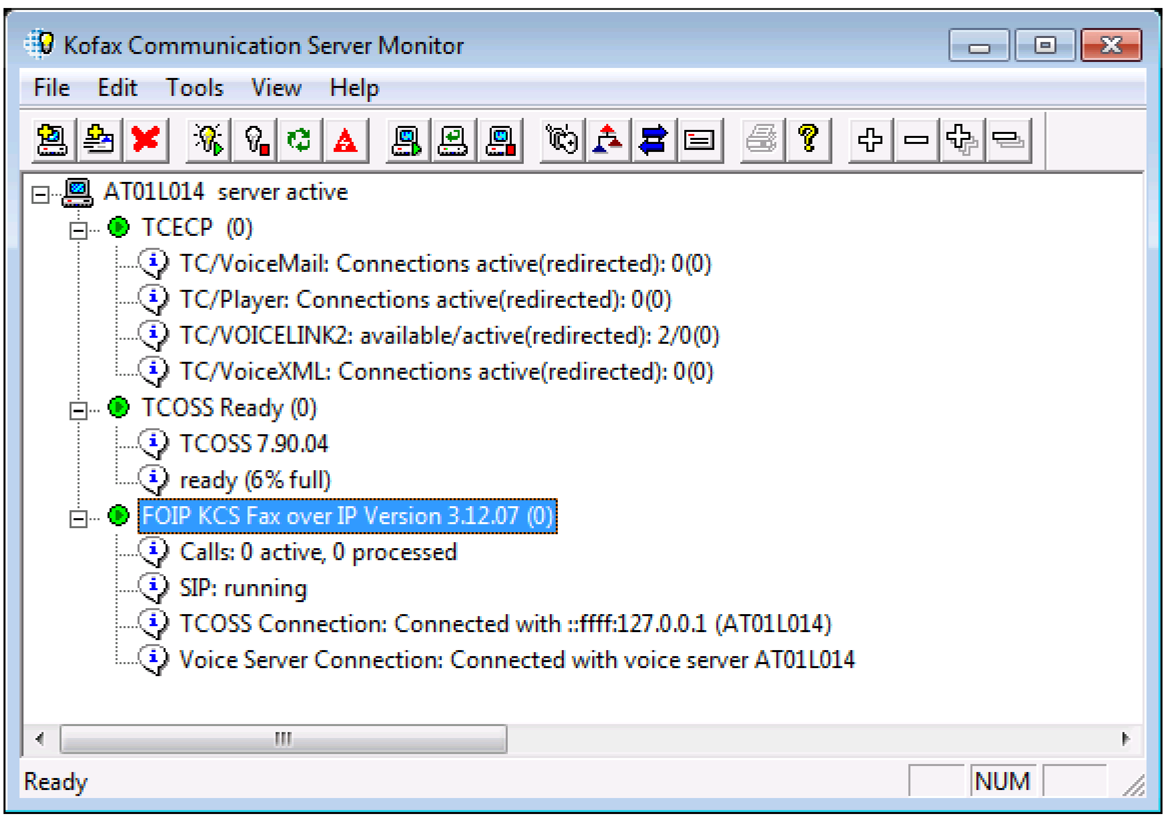

Figure 28: Session Manager Dashboard

9. Conclusion

These Application Notes describe the compliance testing of the Kofax Communication Server with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The fax functionality of Kofax Communication Server was tested. A detailed description of the configuration required for both the Avaya and the Kofax equipment is documented within these Application Notes. Kofax passed all of the tests performed.

10. References

This section references documentation relevant to these Applications. Avaya product documentation, including the following, is available at http://support.avaya.com.

Information regarding Kofax products is available at http://www.kofax.com/business-communication-software/.

[1] *Installing and Configuring Avaya Aura® Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010 available at http://support.avaya.com/css/P8/documents/100089133

[2] *Administering Avaya Aura® Communication Manager*, Doc ID 03-300509, Issue 6.0 June 2010 available at http://support.avaya.com/css/P8/documents/100089333

[3] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100082630

[4] *Installing and Configuring Avaya Aura® Session Manager,* Doc ID 03-603473 Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089152

[5] *Maintaining and Troubleshooting Avaya Aura® Session Manager,* Doc ID 03-603325, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089154

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.