Avaya Solution & Interoperability Test Lab

Application Notes for Biscom FAXCOM Server with Avaya Aura^{TM} Communication Manager and Avaya Aura^{TM} SIP Enablement Services Using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Biscom FAXCOM Server to interoperate with Avaya Aura^{TM} Communication Manager and Avaya Aura^{TM} SIP Enablement Services using SIP trunks. Biscom FAXCOM Server is a fax solution that uses the SIP trunk interface from Avaya Aura Communication Manager via Avaya Aura SIP Enablement Services to send and receive fax.

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 steps required for Biscom FAXCOM Server to interoperate with Avaya Aura^{TM} Communication Manager and Avaya Aura^{TM} SIP Enablement Services using SIP trunks. Biscom FAXCOM Server is a fax solution that uses the SIP trunk interface from Avaya Aura Communication Manager via Avaya Aura SIP Enablement Services to send and receive fax.

Biscom FAXCOM Server utilizes the Dialogic Brooktrout SR140 Virtual Fax Board to support T.38 fax over the IP network, and integration with Avaya Aura Communication Manager and Avaya Aura SIP Enablement Services is achieved through the SIP trunk interface.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following on the Biscom FAXCOM Server:

- Proper handling of faxes via the SIP trunks including send/receive, intra-site, inter-site over ISDN (PRI), inter-site over IP (H.323), different media processor boards, enable/disable media shuffling, simultaneous with bi-directional faxes, and miscellaneous failure scenarios.

- Proper handling of faxes with different pages, resolution, complexity, format, and data rates.

- No adverse impact on the inter-site VoIP calls during VoIP faxes.

The serviceability testing focused on verifying the ability of the Biscom FAXCOM Server to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable and stopping/starting the fax service on the Biscom FAXCOM Server.

1.2. Support

Technical support on Biscom FAXCOM Server can be obtained through the following:

- **Phone:** (978) 250-8355
- **Web:** www.biscom.com/support

2. Reference Configuration

As shown in **Figure 1**, both the Local and Remote sites have a Biscom FAXCOM Server. SIP trunks are used to connect each Biscom FAXCOM Server with the local Avaya Aura Communication Manager via the local Avaya Aura SIP Enablement Service server. Routing between the two sites include both ISDN PRI and H.323 trunks.

The Local site consists of two Avaya G650 Media Gateways, with each media gateway configured as a separate port network in a separate IP network region.

The detailed administration of routing between the two sites and basic connectivity between Avaya Aura Communication Manager and Avaya Aura SIP Enablement Services are not the focus of these Application Notes and will not be described.

The administration procedures in these Application Notes are shown for the Local site. Unless specified otherwise, the same procedures need to apply to the Remote site using appropriate values for the Remote site from **Figure 1**.

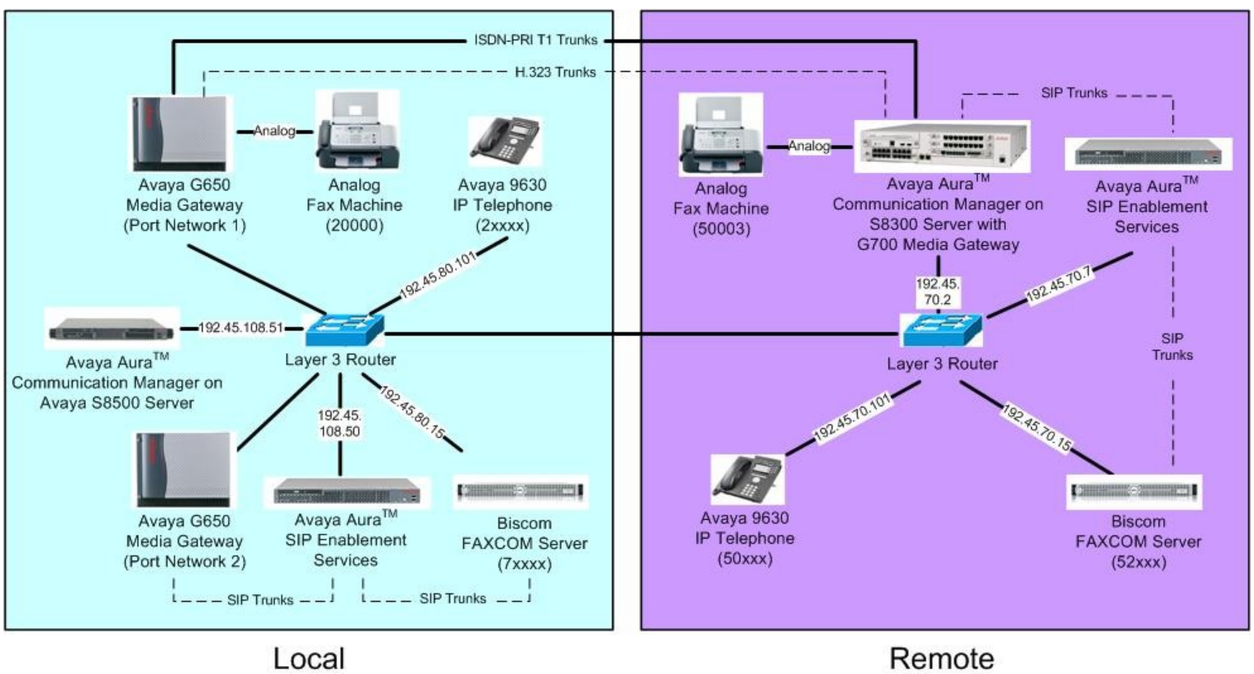


Figure 1: Biscom FAXCOM Server with Avaya Aura Communication Manager and Avaya Aura SIP Enablement Services Using SIP Trunks

3. Equipment and Software Validated

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

Equipment	Software
Avaya S8500 Server	Avaya Aura Communication Manager 5.2, R015x.02.0.947.3
Avaya G650 Media Gateways • TN799DP C-LAN Circuit Pack • TN2302AP IP Media Processor • TN2602AP IP Media Processor	 HW01 FW024 HW20 FW118 HW02 FW040
Avaya Aura SIP Enablement Services	SES-5.2.0.0-947.3a
Avaya 9600 Series IP Telephones (H.323)	3.0
Biscom FAXCOM Server with Dialogic Brooktrout Virtual Fax Board • Boston Bfv API • Boston Driver • Boston SDK	6.1.3.0 with fapiconfig version 6.1.4.0 V6.0.00 B11 V6.0.00 B7 V6.0.00 B11

4. Configure Avaya Aura™ Communication Manager

This section provides the procedures for configuring Avaya Aura Communication Manager. The procedures include the following areas:

- Verify Communication Manager license
- Administer IP codec set
- Administer IP network region
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer route pattern
- Administer public unknown numbering
- Administer AAR analysis
- Administer IP network map

4.1. Verify Communication Manager License

Log in to the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 800   44
           Maximum Concurrently Registered IP Stations: 18000 1
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 800   130
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    0
```

4.2. Administer IP Codec Set

Use the “change ip-codec-set n” command, where “n” is an existing codec set number that will be used for integration with the Biscom FAXCOM Server. Enter the audio codec type in the **Audio Codec** field. The only applicable codec types are G.711MU and G.711A. Retain the default values in the remaining fields.

```
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:
```

Navigate to **Page 2**, and enter “t.38-standard” for the **FAX Mode** field. Retain the default values in the remaining fields.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n



                    Mode                Redundancy
    FAX             t.38-standard        0
    Modem           off                  0
    TDD/TTY         US                   3
    Clear-channel   n                    0
```

4.3. Administer IP Network Region

Use the "change ip-network-region n" command, where "n" is an existing network region that will be used for integration with the Biscom FAXCOM Server. For the **Authoritative Domain** field, enter the SIP domain name of the SIP Enablement Services server, in this case "avayatest.com". For the **Codec Set** field, enter the codec set number from **Section 4.2**.

```
change ip-network-region 2                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:          Authoritative Domain: avayatest.com
    Name: PN1
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                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

Navigate to **Page 3**, and specify the codec set to use for inter-regions. In the compliance testing, the SIP trunks in the Local site that connect the Biscom FAXCOM Server to the SIP Enablement Services server are in network region 2, the SIP trunks in the Local site that connect the SIP Enablement Services server to the Communication Manager are in network region 1, and the ISDN and H.323 trunks in the Local site that connect to the Remote site are in network region 1.

```
change ip-network-region 2                                     Page   3 of  19

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    1      y   NoLimit                                             n all
 2    1                                                                all
```

Similar inter-region setting needs to be applied to the other network region, as shown below.

```
change ip-network-region 1                                     Page   3 of  19

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    1                                                                all
 2    1      y   NoLimit                                             n all
```

4.4. Administer SIP Trunk Group

Administer a SIP trunk group to interface with the local Biscom FAXCOM Server. Use the "add trunk-group n" command, where "n" is an available trunk group number. Set the **Group Type** to "sip", and **Service Type** to "tie". Enter a descriptive **Group Name**, and an available trunk access code for the **TAC** field.

```
add trunk-group 7                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 7                    Group Type: sip           CDR Reports: y
  Group Name: ToBiscomFax                 COR: 1        TN: 1        TAC: *007
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group:
                                                     Number of Members: 0
```

Navigate to **Page 3**, and enter "public" for the **Numbering Format** field as shown below.

```
add trunk-group 7                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y


                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n



 Show ANSWERED BY on Display? y
```

4.5. Administer SIP Signaling Group

Administer a SIP signaling group for the new trunk group to use for signaling. Use the "add signaling-group n" command, where "n" is an available signaling group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Near-end Node Name:** An existing C-LAN node name from the local port network.
- **Far-end Node Name:** The existing SIP Enablement Services node name.
- **Far-end Network Region:** The IP network region number from **Section 4.3**.
- **Far-end Domain:** The IP address of the local Biscom FAXCOM Server.

In the compliance testing, "CLAN2A" is a C-LAN located in port network 2 and pre-configured with network region 2.

```
add signaling-group 7                                             Page   1 of   1
                                SIGNALING GROUP

 Group Number: 7              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n




   Near-end Node Name: CLAN2A                 Far-end Node Name: SES
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 2
Far-end Domain: 192.45.80.15

                                             Bypass If IP Threshold Exceeded? 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? n                        Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

4.6. Administer SIP Trunk Group Members

Administer SIP trunk group members for the newly added SIP trunk group. Use the "change trunk-group n" command, where "n" is the trunk group number added in **Section 4.4**. Enter the corresponding signaling group number from **Section 4.5** into the **Signaling Group** field. Enter the desired number of trunk group members into the **Number of Members** field.

```
change trunk-group 7                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 7                   Group Type: sip           CDR Reports: y
  Group Name: ToBiscomFax                COR: 1        TN: 1        TAC: *007
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                      Signaling Group: 7
                                                    Number of Members: 6
```

4.7. Administer Route Pattern

Create a route pattern to use for the newly created SIP trunk group. Use the "change route-pattern n" command, where "n" is an available route pattern. Enter a descriptive **Pattern Name**. In the **Grp No** field, enter the trunk group number from **Section 4.4**. In the **FRL** field, enter a level that allows access to this trunk with "0" being least restrictive.

```
change route-pattern 7                                            Page   1 of   3
                    Pattern Number: 7   Pattern Name: ToFaxServer
                             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: 7    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
```

4.8. Administer Public Unknown Numbering

Use the "change public-unknown-numbering 0" command, to define the calling party number to send to the local Biscom FAXCOM Server. Add an entry for the trunk group defined in **Section 4.4**. In the example shown below, all calls originating from a 5-digit extension beginning with 2 and routed over any trunk group will result in a 5-digit calling number. The calling party number will be in the SIP "From" header.

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                     Total Administered: 2
 5  2                                       5           Maximum Entries: 9999
```

4.9. Administer AAR Analysis

This section provides a sample AAR routing used for routing calls with dialed digits 7xxxx to the local Biscom FAXCOM Server. Note that other methods of routing may be used. Use the "change aar analysis 0" command, and add an entry to specify how to route calls to 7xxxx. In the example shown below, calls with digits 7xxxx will be routed as an AAR call using route pattern "7" from **Section 4.7**.

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

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    7                     5    5     7         aar          n
```

4.10. Administer IP Network Map

Use the "change ip-network-map" command to assign the network region number from **Section 4.3** for incoming fax calls to the local Biscom FAXCOM Server, as shown below.

```
change ip-network-map                                          Page   1 of  63
                               IP ADDRESS MAPPING

                                               Subnet Network     Emergency
 IP Address                                    Bits   Region VLAN Location Ext
 --------------------------------------------- ------ ------ ---- -------------
 FROM: 192.45.80.15                            /      2      n
   TO: 192.45.80.15
 FROM:                                         /             n
   TO:
```

5. Configure Avaya Aura™ SIP Enablement Services

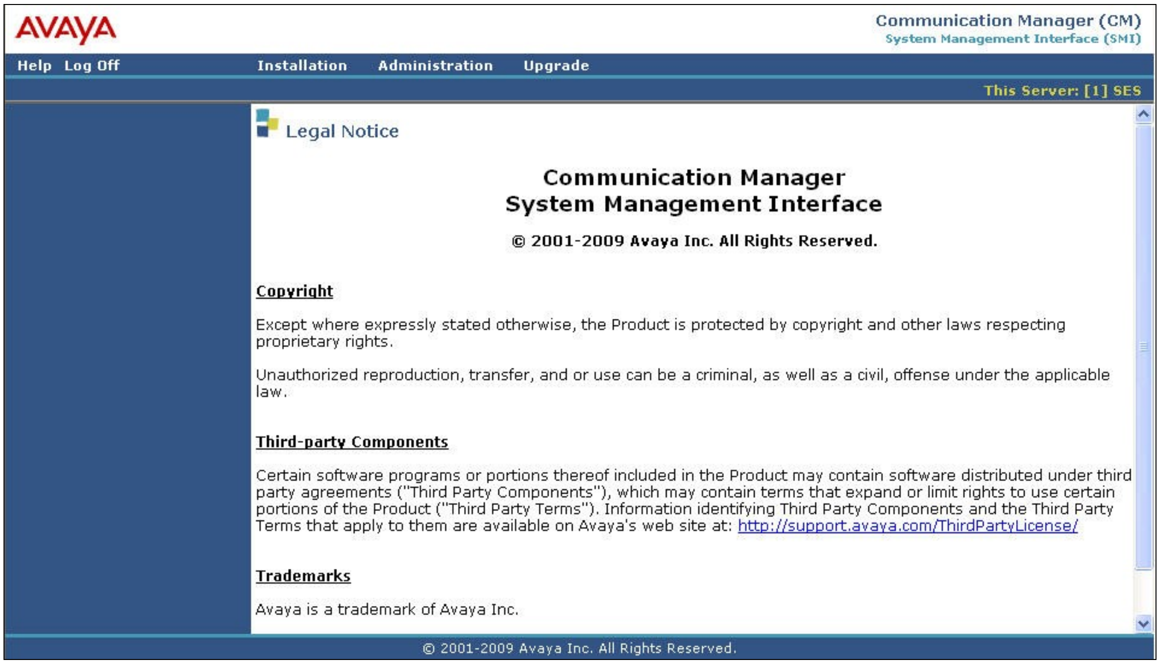
This section provides the procedures for configuring Avaya Aura SIP Enablement Services. The procedures include the following areas:

- Launch administration interface
- Administer Communication Manager servers map
- Administer trusted host

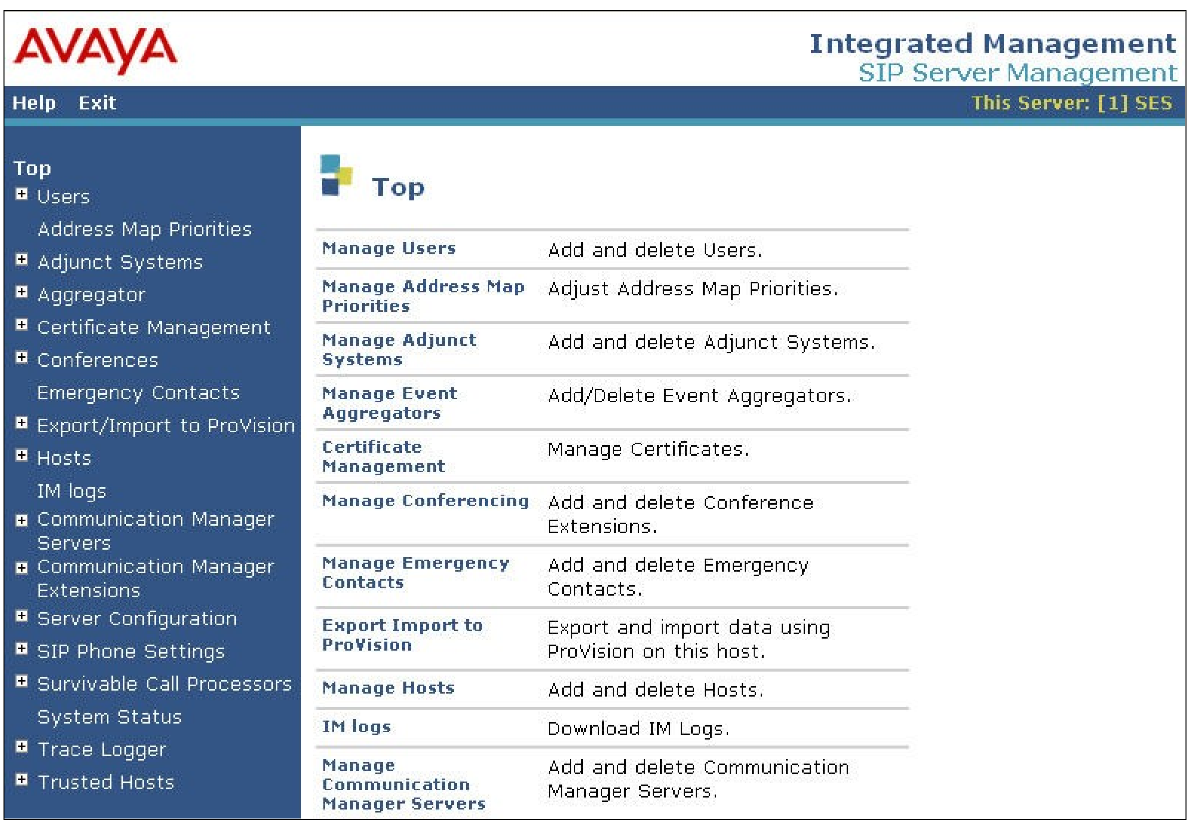
5.1. Launch Administration Interface

Access the SIP Enablement Services web interface by using the URL "http://ip-address/admin" in an Internet browser window, where "ip-address" is the IP address of the SIP Enablement Services server. Log in with the appropriate credentials.

In the subsequent screen, select **Administration > SIP Enablement Services** from the top menu.

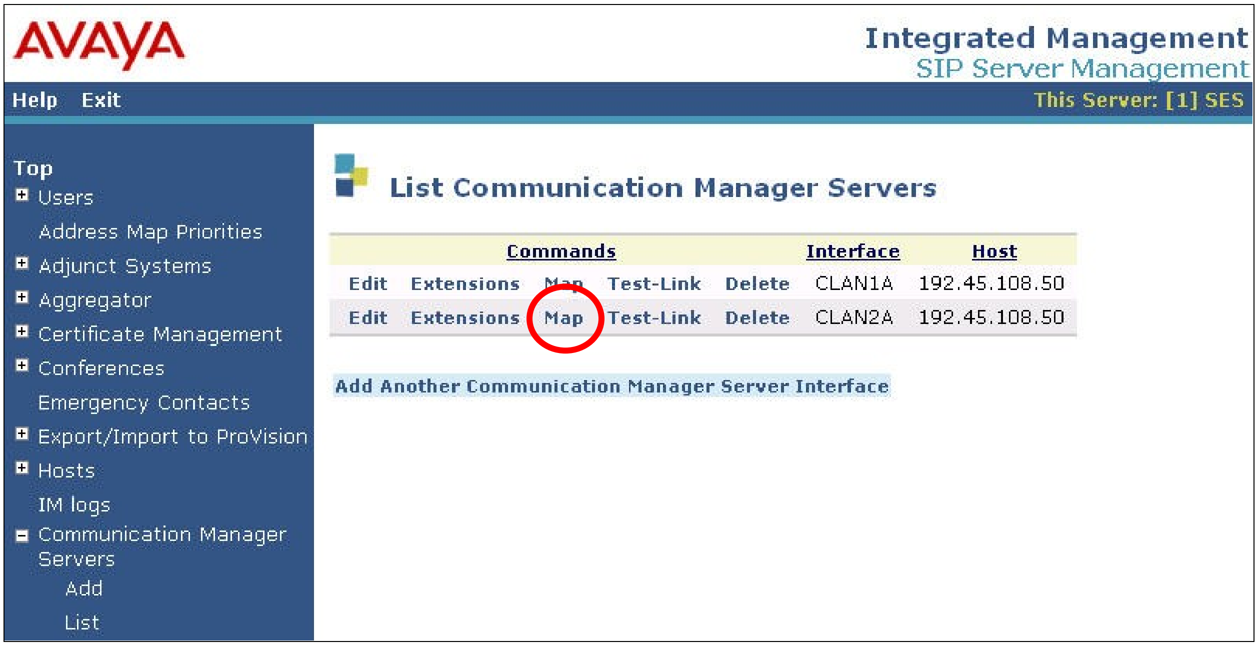


The **Top** screen is displayed next.



5.2. Administer Communication Manager Servers Map

Select **Communication Manager Servers > List** from the left pane. The **List Communication Manager Servers** screen is displayed. Click the **Map** link corresponding to the appropriate **Interface**, in this case “CLAN2A”, which interfaces to port network 2 on the local Avaya Aura Communication Manager.



In the **List Communication Manager Server Address Map** screen below, click on the **Add Map In New Group** link.

The **Add Communication Manager Server Address Map** screen is displayed next. This screen is used to specify which calls are to be routed by the local Avaya Aura Communication Manager. A new address map needs to be added, so that fax calls from the local Biscom FAXCOM Server to the Remote site will be routed by the local Avaya Aura Communication Manager.

For the **Name** field, enter a descriptive name to denote the routing. For the **Pattern** field, enter an appropriate syntax for address mapping. For the compliance testing, a pattern of "^sip:5[0-9]{4}" is used to match to any extensions of 5xxxx at the Remote site. Click **Add**.

5.3. Administer Trusted Host

Select **Trusted Hosts > Add** from the left pane (not shown). The **Add Trusted Host** screen is displayed. For the **IP Address** field, enter the IP address of the local Biscom FAXCOM Server. Enter a desired description for the **Comment** field.

6. Configure Biscom FAXCOM Server

This section provides the procedures for configuring the Biscom FAXCOM Server. The procedures include the following areas:

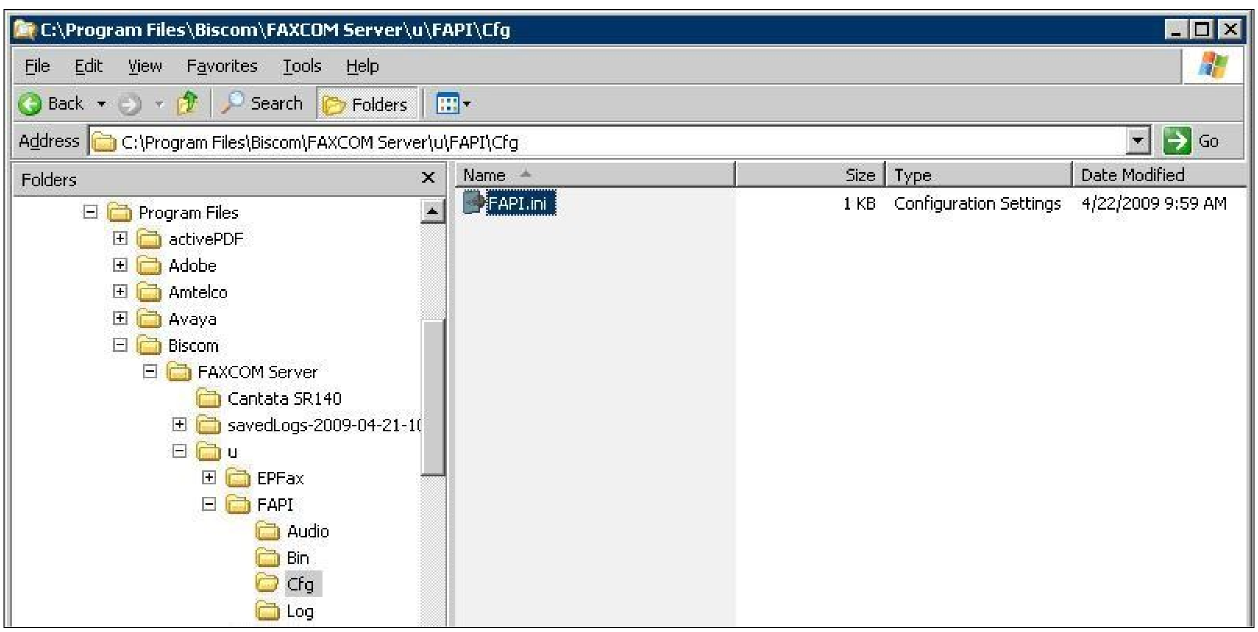
- Execute configuration script
- Administer FAPI.ini
- Start fax service

6.1. Execute Configuration Script

From the Biscom FAXCOM Server, launch the **Command Prompt** window and enter "fapiconfig sip" as shown below to initialize for SIP. The initialization is complete when "SR140" is returned, as shown below.

6.2. Administer FAPI.ini

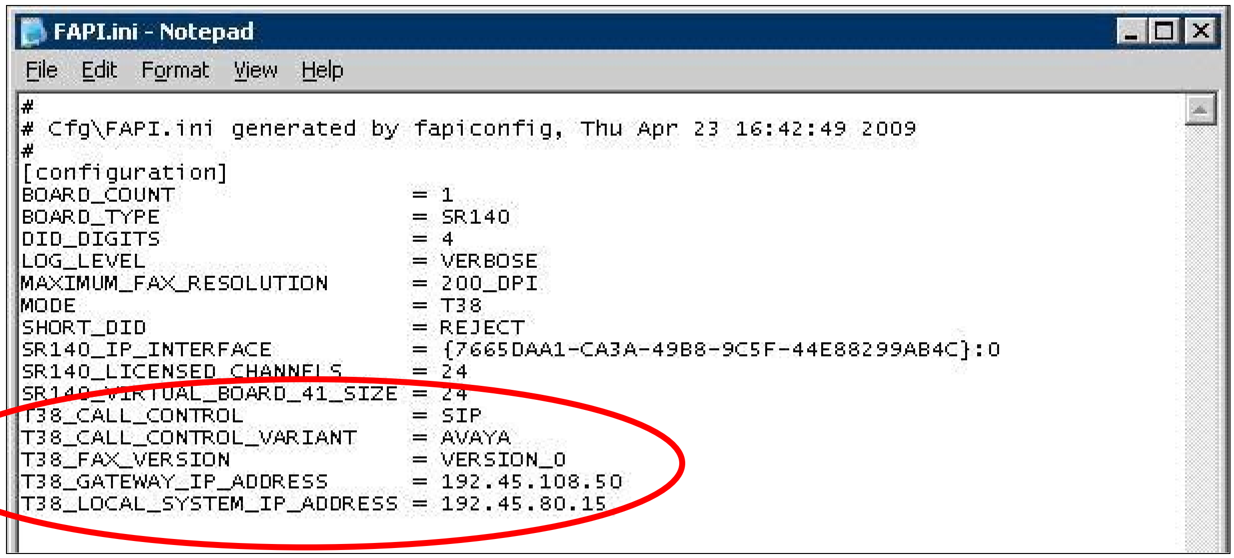
Navigate to the **Cfg** directory to edit the **FAPI.ini** file, as shown below.



The **FAPI.ini** file contains a list of configurable parameters. Enter the following values for the specified fields, and retain the default values for the remaining fields.

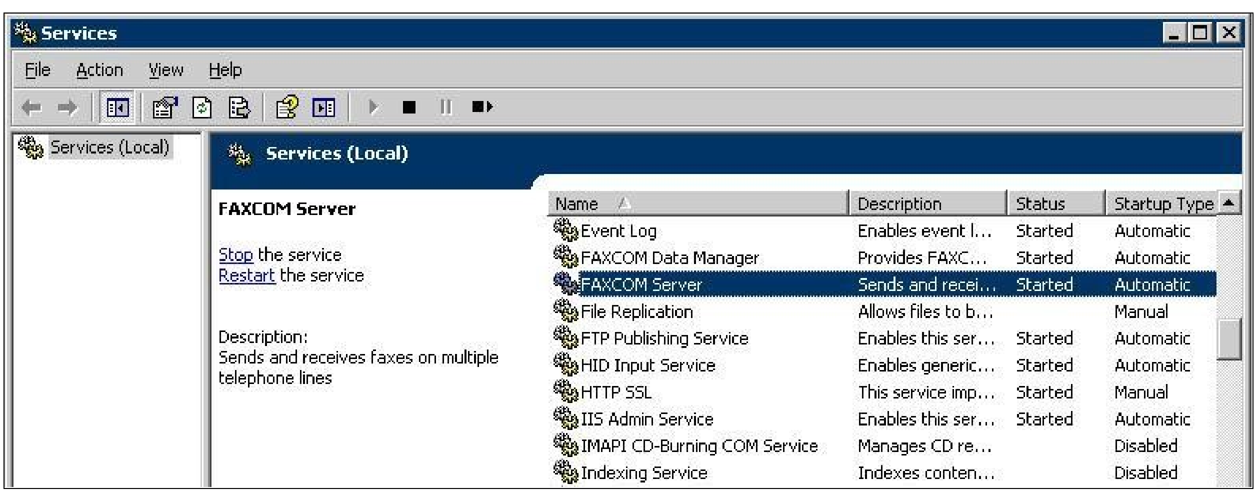
- **T38_CALL_CONTROL:** "SIP"
- **T38_CALL_CONTROL_VARIANT:** "AVAYA"
- **T38_GATEWAY_IP_ADDRESS:** The local SIP Enablement Services server.
- **T38_LOCAL_SYSTEM_IP_ADDRESS:** The local FAXCOM Server.

In the event that the local Biscom FAXCOM Server has multiple Ethernet cards, then follow [5] to manually obtain the value for the **SR140_IP_INTERFACE** parameter shown below.



6.3. Start Fax Service

Launch the **Services** window, and right-click on **FAXCOM Server** and select "Start".



7. General Test Approach and Test Results

The feature test cases were performed manually. Intra-site and inter-site fax calls to and from the local Biscom FAXCOM Server were made. The fax calls were sent and received by using the Send A Test Fax utility at the local Biscom FAXCOM Server and the analog fax machine at the Remote site. The Biscom FAXCOM Server at the remote site was used for testing simultaneous send/receive of fax calls.

The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet cables and stop/start the fax service on the Biscom FAXCOM Server.

All test cases were executed and passed.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, SIP Enablement Services, and the Biscom FAXCOM Server.

8.1. Verify Avaya Aura TM Communication Manager

On Communication Manager, verify the status of the local SIP trunk group by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 4.4**. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 7                                                   Page   1

                            TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0007/001 T00133   in-service/idle    no
0007/002 T00134   in-service/idle    no
0007/003 T00135   in-service/idle    no
0007/004 T00136   in-service/idle    no
0007/005 T00137   in-service/idle    no
0007/006 T00138   in-service/idle    no
```

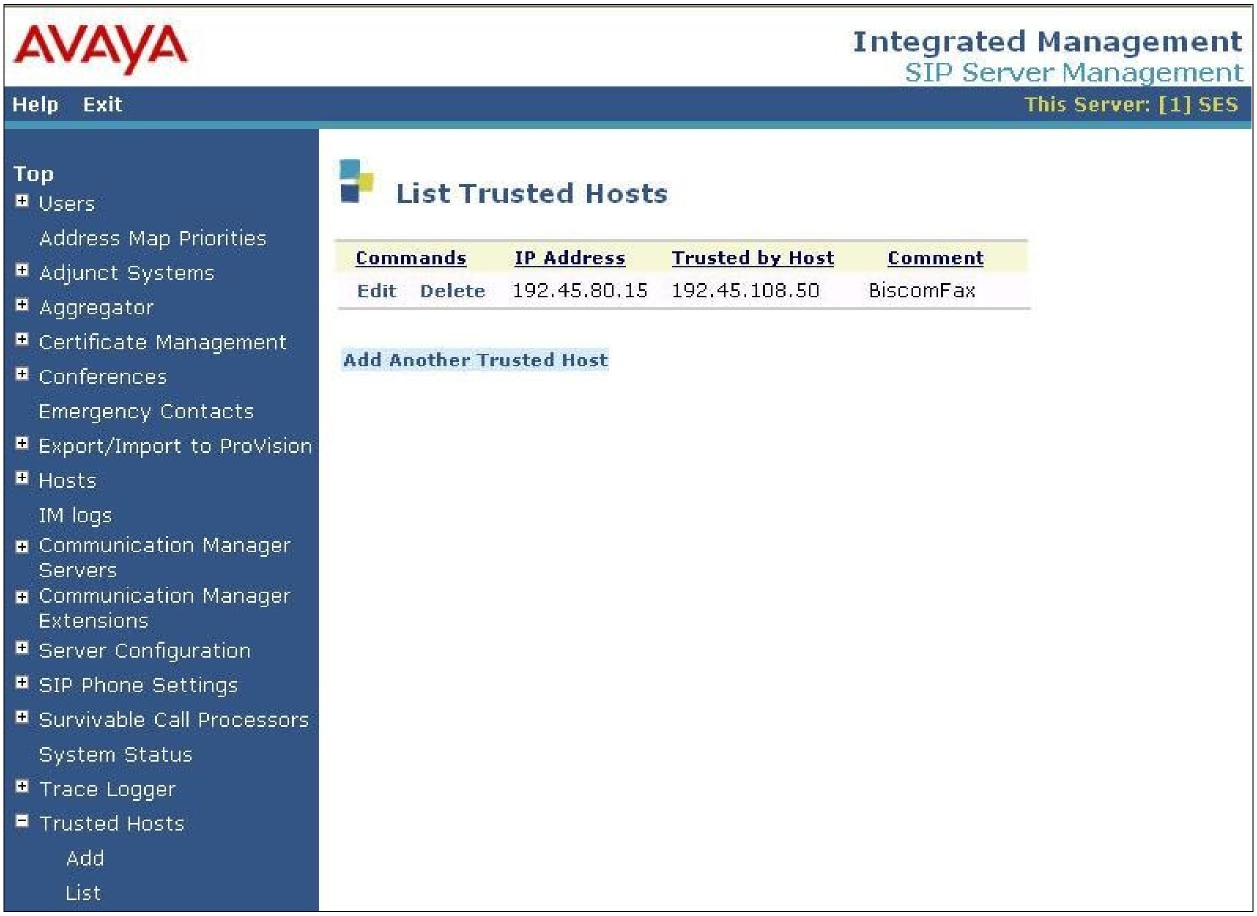
Verify the status of the SIP signaling group by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 4.5**. Verify that the signaling group is "in-service" as indicated in the **Group State** field shown below.

```
status signaling-group 7
                         STATUS SIGNALING GROUP

        Group ID: 7                            Active NCA-TSC Count: 0
      Group Type: sip                           Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

8.2. Verify Avaya Aura™ SIP Enablement Services

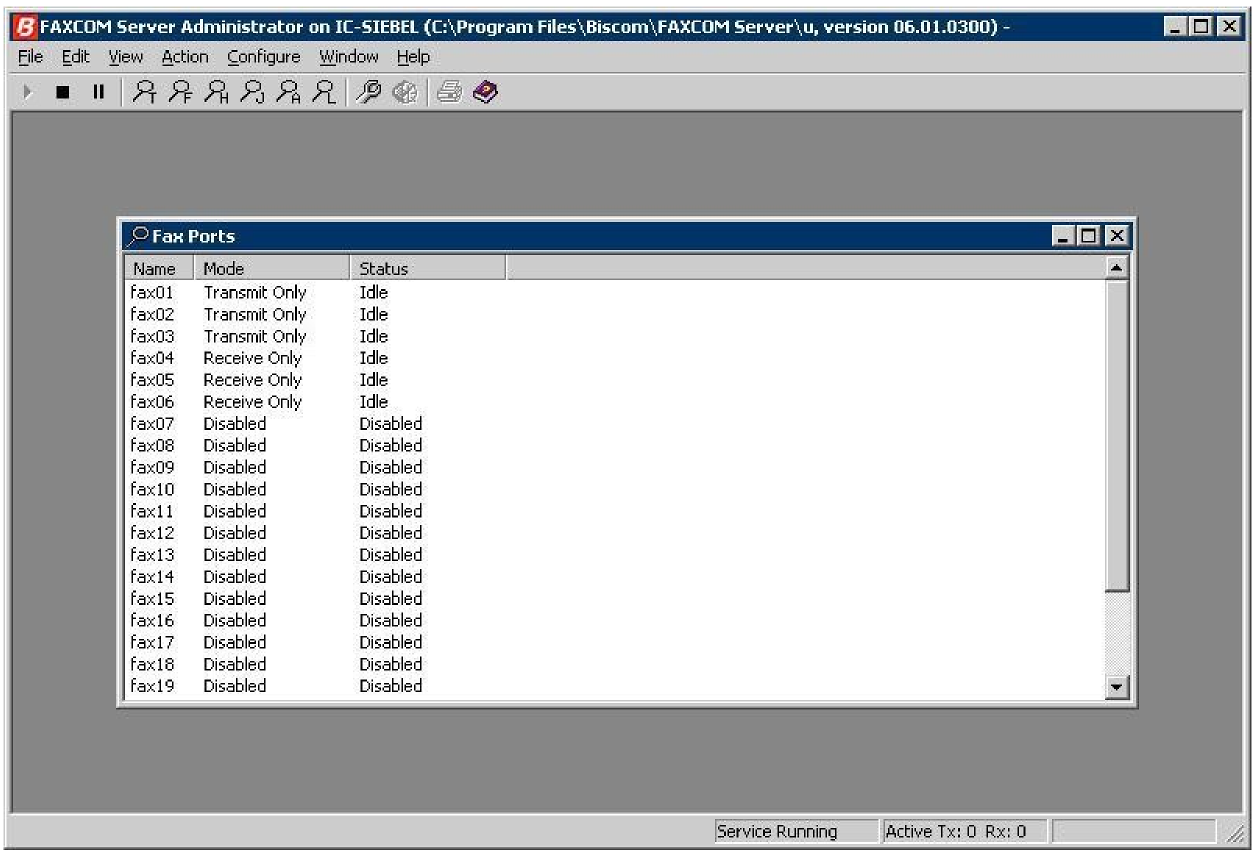
On SIP Enablement Services, select **Trusted Hosts > List** from the left pane, and verify that the Biscom FAXCOM Server appears as a trusted host, as shown below.



8.3. Verify Biscom FAXCOM Server

From the Biscom FAXCOM Server, select **Start > All Programs > FAXCOM > FAXCOM Server > Administrator**. The **FAXCOM Server Administrator** screen is displayed, as shown below. Select **View > Fax Ports** from the top menu.

The **FAXCOM Server Administrator** screen is updated with a **Fax Ports** pane. Verify that the status of all configured ports is "Idle". In the compliance testing, six fax ports were pre-configured on the FAXCOM Server.



9. Conclusion

These Application Notes describe the configuration steps required for Biscom FAXCOM Server to successfully interoperate with Avaya Aura Communication Manager and Avaya Aura SIP Enablement Services using SIP trunks. All feature and serviceability test cases were completed.

10. Additional References

This section references the product documentation relevant to these Application Notes.

1. *Administering Avaya Aura™ Communication Manager*, Document 03-300509, Issue 5.0, Release 5.2, May 2009, available at http://support.avaya.com.

2. *SIP Support in Avaya Aura™ Communication Manager*, Document 555-245-206, Issue 9, May 2009, available at http://support.avaya.com.

3. *Installing, Administering, Maintaining, & Troubleshooting SIP Enablement Services*, Document 03-600768, 6.0, June 2008, available at http://support.avaya.com

4. *FAXCOM Server Administrator's Guide*, February 2009 Revised Edition, available from Biscom Technical Support.

5. *KB Avaya 20090424*, Knowledge Base article under "SR140 Avaya", available from Biscom Technical Support.

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.