



**Application Notes for Biscom FAXCOM Server with Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager Using SIP Trunks – Issue 1.0**

**Abstract**

These Application Notes describe the configuration steps required for Biscom FAXCOM Server to interoperate with Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager using SIP trunks. Biscom FAXCOM Server is a fax solution that utilizes a SIP trunk to Avaya Aura™ Session Manager to send and receive faxes.

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™ Communication Manager and Avaya Aura™ Session Manager using SIP trunks. Biscom FAXCOM Server is a fax solution that utilizes a SIP trunk to Avaya Aura™ Session Manager to send and receive faxes.

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™ Session Manager is achieved through SIP trunks.

## 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 SIP trunks, including: sending/receiving faxes, intra-site faxes, inter-site faxes over ISDN (PRI), inter-site over IP (SIP), different media processor boards, simultaneous bi-directional faxes, and miscellaneous failure scenarios.
- Proper handling of faxes with different pages, resolution, complexity, format, and data rates.
- No adverse impact on intra and inter-site VoIP calls during 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](http://www.biscom.com/support)

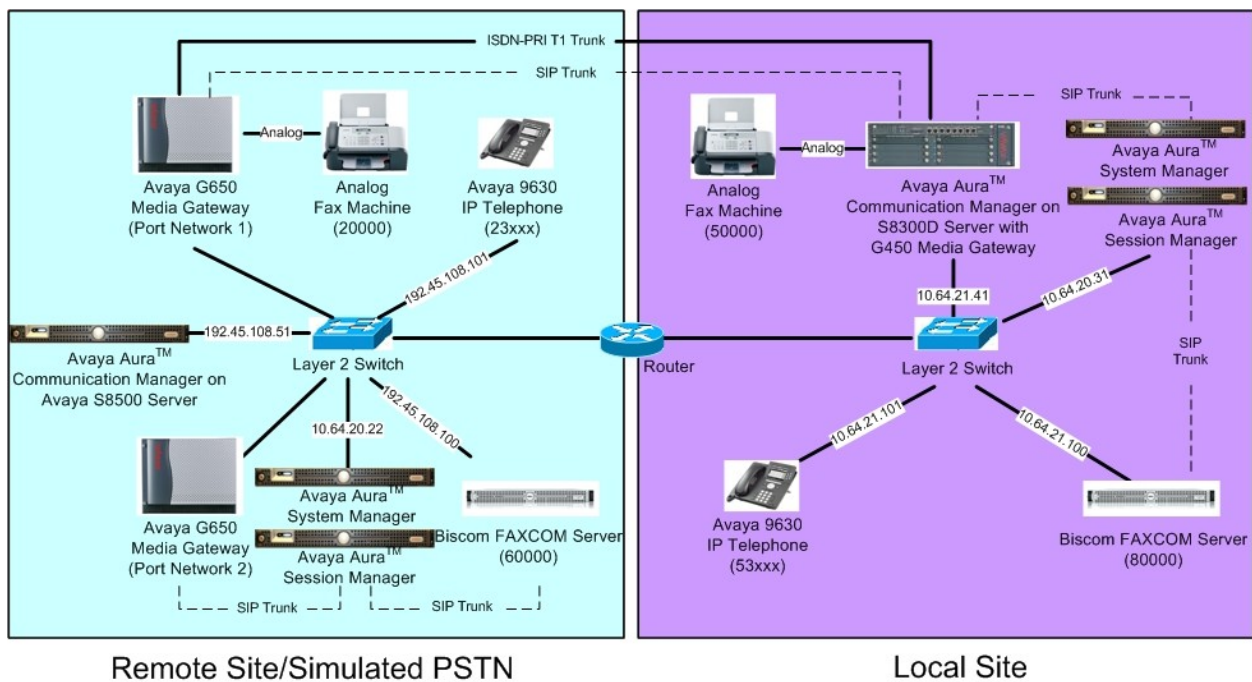
## 2. Reference Configuration

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™ Session Manager server. Routing between the two sites include both ISDN PRI and SIP trunks.

The Remote 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™ Session Manager 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 Communication Manager and Session Manager Using SIP Trunks**

### 3. Equipment and Software Validated

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

Equipment	Software
Avaya S8300 Server with Avaya G450 Media Gateway (Local site)	Avaya Aura™ Communication Manager 6.0, R016x.00.0.345.0, Update 18444 (Avaya Aura™ System Platform: 6.0.1.0.5)
Avaya S8800 Server (Local site)	Avaya Aura™ System Platform: 6.0.1.0.5 Avaya Aura™ System Manager: 6.0.7.0
Avaya S8800 Server (Local site)	Avaya Aura™ System Platform: 6.0.1.0.5 Avaya Aura™ Session Manager 6.0.0.0.600020
Avaya S8500 Server (Remote site)	Avaya Aura™ Communication Manager 5.2.1, R015x.02.1.016.4, Update 18433
Avaya G650 Media Gateways (Remote Site) <ul style="list-style-type: none"> <li>• TN799DP C-LAN Circuit Pack</li> <li>• TN2302AP IP Media Processor</li> <li>• TN2602AP IP Media Processor</li> </ul>	HW01 FW024 HW20 FW120 HW02 FW051
Avaya 9600 Series IP Telephones (H.323)	3.1.1
Biscom FAXCOM Server	6.1.9
Dialogic Brooktrout SR140 SDK	6.2.4

## 4. Configure Avaya Aura™ Communication Manager

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

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

### 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		<b>USED</b>
Maximum Administered H.323 Trunks:	4000	100
Maximum Concurrently Registered IP Stations:	2400	2
Maximum Administered Remote Office Trunks:	4000	0
Maximum Concurrently Registered Remote Office Stations:	2400	0
Maximum Concurrently Registered IP eCons:	68	0
Max Concur Registered Unauthenticated H.323 Stations:	100	0
Maximum Video Capable Stations:	2400	0
Maximum Video Capable IP Softphones:	2400	0
<b>Maximum Administered SIP Trunks:</b>	<b>4000</b>	<b>280</b>
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:	300	0

(NOTE: You must logoff & login to effect the permission changes.)

## 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 “G.711MU” in the **Audio Codec** fields. 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

FAX           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 **Codec Set** field, enter the codec set number from **Section 4.2**. Set **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** to “no” (this is required for inter-site fax calls over the SIP trunks).

```
change ip-network-region 1                               Page 1 of 20
                                                    IP NETWORK REGION
  Region: 1
  Location: 1      Authoritative Domain: avaya.com
    Name: Compliance Testing
  MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: no
    Codec Set: 1      Inter-region IP-IP Direct Audio: no
      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
  H.323 IP ENDPOINTS      AUDIO RESOURCE RESERVATION PARAMETERS
      RSVP Enabled? n
    H.323 Link Bounce Recovery? y
    Idle Traffic Interval (sec): 20
    Keep-Alive Interval (sec): 5
    Keep-Alive Count: 5
```

### 4.4. Administer IP Node Names

Use the “change node-names ip” command, and add an entry for the local Session Manager. In this case, “SM2” and “10.64.20.31” are entered as **Name** and **IP Address**. Note the “procr” / “10.64.21.41” entry, which is the node name to the processor board, and will be used later to configure the SIP trunk to Session Manager.

```
change node-names ip                                     Page 1 of 2
                                                    IP NODE NAMES
  Name      IP Address
  AcmePacket      10.64.20.106
  CM-B1      192.45.108.55
  CM-B2      192.45.108.57
  FaxServer      10.64.21.100
  SES-A      10.64.21.61
  SM1      10.64.40.42
  SM2      10.64.20.31
  VoicePortal      10.64.10.32
  default      0.0.0.0
  procr      10.64.21.41
  procr6      ::
```

## 4.5. Administer SIP Signaling Group

Administer a SIP signaling group for a new trunk that will be created for the connection between Communication Manager and Session Manager. 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:** Processor node name from **Section 4.4.**
- **Far-end Node Name:** Session Manager node name from **Section 4.4.**
- **Far-end Network Region:** The IP network region number from **Section 4.3.**
- **Far-end Domain:** “avaya.com”

```
change signaling-group 8                               Page 1 of 1
                                                    SIGNALING GROUP
Group Number: 8                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: SM2
Near-end Listen Port: 5060                Far-end Listen Port: 5060
Far-end Network Region: 1

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



## 4.6. Administer SIP Trunk Group

Administer a SIP trunk group to interface with the Session Manager. 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. Set **Member Assignment Method** to “auto”, **Signaling Group** to the signaling group number from **Section 4.5**, and enter a desired value for number of trunk group members for **Number of Members**.

```
add trunk-group 8                                     Page 1 of 21
                                                    TRUNK GROUP
Group Number: 8                                     Group Type: sip          CDR Reports: y
  Group Name: demoSM (avaya.com)                   COR: 1                 TN: 1           TAC: 108
  Direction: two-way                               Outgoing Display? y
  Dial Access? n                                   Night Service:
  Queue Length: 0
  Service Type: tie                                Auth Code? n
                                                    Member Assignment Method: auto
                                                    Signaling Group: 8
                                                    Number of Members: 10
```

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

```
add trunk-group 8                                     Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n                               Measured: none
                                                    Maintenance Tests? y

  Numbering Format: public
                                                    UII Treatment: service-provider
                                                    Replace Restricted Numbers? n
                                                    Replace Unavailable Numbers? n

  Modify Tandem Calling Number: no

  Show ANSWERED BY on Display? y
```

## 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 4.6**. In the **FRL** field, enter a level that allows access to this trunk with “0” being least restrictive.

change route-pattern 8													Page 1 of 3	
											Pattern Number: 8		<b>Pattern Name: ToFaxServer</b>	
											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:	8	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						Dgts	Format
					Request								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

## 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 Session Manager. Add an entry for the trunk group defined in **Section 4.6**. In the example shown below, all calls originating from a 5-digit extension beginning with 5 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
Ext  Ext          Trk      CPN          Total
Len  Code         Grp(s)   Prefix     CPN
5    5                               5
                                         Total Administered: 1
                                         Maximum Entries: 240
```

## 4.9. Administer AAR Analysis

This section provides a sample AAR routing used for routing calls with dialed digits 8xxxx to the local Session Manager. 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 8xxxx. In the example shown below, calls with digits 8xxxx will be routed as an AAR call using route pattern “8” from **Section 4.7**.

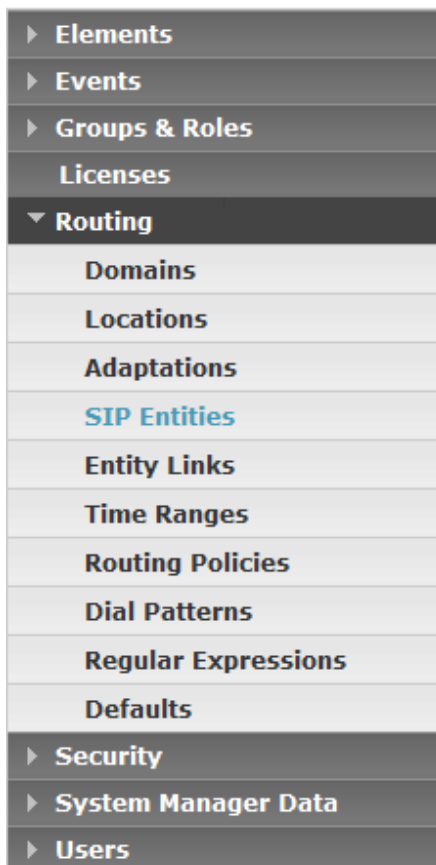
```
change aar analysis 0                                             Page 1 of 2
                        AAR DIGIT ANALYSIS TABLE
                        Location: all                               Percent Full: 2
Dialed          Total      Route   Call   Node  ANI
String          Min  Max   Pattern Type  Num  Reqd
2              5    5     1      aar   7    n
6              5    5     1      aar   6    n
8              5    5     8      aar   5    n
```

## 5. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Avaya Aura™ Session Manager. The procedures include adding the following items:

- Administer Domain
- SIP Entities corresponding to the SIP telephony systems and Avaya Aura™ Session Manager
- Entity Links, which define the SIP trunk parameters used by Avaya Aura™ Session Manager when routing calls to/from SIP Entities
- Time Ranges during which routing policies are active
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed.

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura™ System Manager, using the URL “http://<ip-address>/SMGR”, where “<ip-address>” is the IP address of Avaya Aura™ System Manager. Log in with the appropriate credentials and accept the Copyright Notice. The left menu shown below is displayed. Expand the **Routing** Link on the left side as shown. The sub-menus displayed in the left column below will be used to configure all of the above items (**Sections 5.1** through **5.6**).



## 5.1. Specify Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., “avaya.com”).
- **Notes:** Descriptive text (optional).

Click **Commit**.

The screenshot shows the Avaya Aura System Manager 6.0 interface. The top navigation bar includes the Avaya logo, the product name 'Avaya Aura™ System Manager 6.0', and user information: 'Welcome, admin Last Logged on at September 2, 2010 10:40 AM'. There are links for 'Help | About | Change Password | Log off'. The breadcrumb trail is 'Home / Routing / Domains'. The left sidebar has a tree view with 'Routing' expanded and 'Domains' selected. The main content area is titled 'Domain Management' and contains a table with one row for 'avaya.com'. The table has columns for 'Name', 'Type', 'Default', and 'Notes'. The 'Name' field contains 'avaya.com', 'Type' is 'sip', and 'Default' is an unchecked checkbox. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the table area. A red asterisk and the text '\* Input Required' are visible at the bottom left of the table area.

## 5.2. Add SIP Entities

A SIP Element must be added for Avaya Aura™ Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration a SIP Element is added for the Session Manager, Communication Manager, and the Biscom FAXCOM Server. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the SM or the signaling interface on the telephony system.
- **Type:** Select “Session Manager” for Avaya Aura™ Session Manager, “CM” for Communication Manager, and “other” for the fax server.

Under *SIP Link Monitoring*:

- **SIP Link Monitoring:** Select “Use Session Manager Configuration” or “Link Monitoring Enabled”

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The following screen shows addition of Avaya Aura™ Session Manager. The IP address used is that of the virtual SM-100 Security Module.

Under **Ports**, add the required listening port(s) for Session Manager.

**Port**

3 Items   <a href="#">Refresh</a>		Filter: <a href="#">Enable</a>		
<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	TCP	avaya.com	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	UDP	avaya.com	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	TLS	avaya.com	<input type="text"/>

Select : All, None

**\* Input Required**

The following screen shows addition of Communication Manager.

The screenshot displays the Avaya Aura System Manager 6.0 interface. The top navigation bar includes the Avaya logo, the product name "Avaya Aura™ System Manager 6.0", and a user status message: "Welcome, admin Last Logged on at September 2, 2010 10:40 AM". A secondary navigation bar contains links for "Help | About | Change Password | Log off". The main content area is titled "SIP Entity Details" and features a left-hand sidebar with a tree view of configuration categories: Elements, Events, Groups & Roles, Licenses, Routing (expanded), Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, Defaults, Security, System Manager Data, and Users. Below the sidebar is a "Help" section with links for "Help for SIP Entity Details fields" and "Help for Committing configuration changes". The main configuration area is divided into two sections: "General" and "SIP Link Monitoring". The "General" section includes fields for Name (mainLabS8300D), FQDN or IP Address (10.64.21.41), Type (CM), Notes, Adaptation, Location (.21 Subnet), Time Zone (America/Denver), and an unchecked checkbox for "Override Port & Transport with DNS SRV". It also features a SIP Timer B/F (in seconds) set to 4, a Credential name field, and Call Detail Recording set to none. The "SIP Link Monitoring" section includes a dropdown for "SIP Link Monitoring" set to "Link Monitoring Enabled", a Proactive Monitoring Interval (in seconds) of 900, a Reactive Monitoring Interval (in seconds) of 120, and a Number of Retries of 1. "Commit" and "Cancel" buttons are located in the top right corner of the configuration area.

The following screen shows addition of the Biscom FAXCOM Server.

This screenshot shows the Avaya Aura System Manager 6.0 interface for configuring a Biscom FAXCOM Server. The layout is identical to the previous screenshot, showing the "SIP Entity Details" page. The "General" section is configured with Name: FaxServerSite1, FQDN or IP Address: 10.64.21.100, Type: Other, and the same Location (.21 Subnet) and Time Zone (America/Denver) as the previous entity. The "SIP Link Monitoring" section is also configured with Link Monitoring Enabled, a Proactive Monitoring Interval of 900 seconds, a Reactive Monitoring Interval of 120 seconds, and 1 retry. The "Commit" and "Cancel" buttons are visible in the top right corner.

### 5.3. Add Entity Links

A SIP trunk between Avaya Aura™ Session Manager and a telephony system is described by an Entity link. To add an Element Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Avaya Aura™ Session Manager.
- **Port:** Port number to which the other system sends SIP requests  
In the sample configuration, TCP Protocol was used.
- **SIP Entity 2:** Select the name of the other system.
- **Port:** Port number on which the other system receives SIP requests
- **Trusted:** Check this box. **Note:** If this box is not checked, calls from the associated SIP Element specified in **Section 5.2** will be denied.

Click **Commit** to save each Entity Link definition.

The following screens illustrate adding the two Entity Links for:

1. Communication Manager
2. Biscom FAXCOM Server

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at September 2, 2010 10:40 AM  
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Entity Links

Entity Links

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* demoSM-mainLabS8	* demoSM	TCP	* 5060	* mainLabS8300D	* 5060	<input checked="" type="checkbox"/>	

\* Input Required



Note, UDP must be selected as the protocol between Session Manager and the Biscom FAXCOM Server.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at September 2, 2010 12:45 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Entity Links

Entity Links Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* demoSM-FaxServerS	* demoSM	UDP	* 5060	* FaxServerSite1	* 5060	<input checked="" type="checkbox"/>	

\* Input Required Commit Cancel

## 5.4. Add Time Ranges

Before adding routing policies (see next section), time ranges must be defined during which the policies will be active. In the sample configuration, one policy was defined that would allow routing to occur at anytime. To add this time range, select **Time Ranges**, and click on the left and click on the **New** button on the right. Fill in the following:

- **Name:** A descriptive name (e.g., “24/7”).
- **Mo through Su** Check the box under each of these headings
- **Start Time** Enter 00:00.
- **End Time** Enter 23:59

Click **Commit** to save this time range.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at September 2, 2010 12:45 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Time Ranges

Time Ranges Edit New Duplicate Delete More Actions Commit

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
<input type="checkbox"/>	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

## 5.5. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 5.2**. One routing policy must be added for routing calls to the Biscom FAXCOM Server. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP Entity to which this routing policy applies.

Under *Time of Day*:

Click **Add**, and select the time range configured in the previous section.

Defaults can be used for the remaining fields. Click **Commit** to save Routing Policy definition.

The following screen shows the Routing Policy for routing calls to the Biscom FAXCOM Server.

The screenshot displays the Avaya Aura System Manager 6.0 interface. The top navigation bar includes the Avaya logo, the system name 'Avaya Aura™ System Manager 6.0', and user information: 'Welcome, admin Last Logged on at September 2, 2010 12:45 PM'. A secondary navigation bar shows 'Home / Routing / Routing Policies / Routing Policy Details' and utility links: 'Help | About | Change Password | Log off'. On the left, a sidebar menu lists various system components, with 'Routing Policies' highlighted. The main content area is titled 'Routing Policy Details' and contains three sections: 'General', 'SIP Entity as Destination', and 'Time of Day'. The 'General' section includes fields for 'Name' (filled with 'to Fax Server Site 1'), 'Disabled' (checkbox), and 'Notes'. The 'SIP Entity as Destination' section has a 'Select' button and a table with one entry: 'FaxServerSite1' with FQDN '10.64.21.100' and Type 'Other'. The 'Time of Day' section has 'Add', 'Remove', and 'View Gaps/Overlaps' buttons, followed by a table with one row for a '24/7' time range. The table columns include Ranking, Name, days of the week (Mon-Sun), Start Time, End Time, and Notes.

Name	FQDN or IP Address	Type	Notes
FaxServerSite1	10.64.21.100	Other	

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

## 5.6. Add Dial Patterns

Define dial patterns to direct calls to the appropriate SIP Entity. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right. Fill in the following, as shown in the screen below, which corresponds to the dial pattern for routing calls to the Biscom FAXCOM Server:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **Domain** SIP domain specified in **Section 5.1**
- **Notes** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

The screenshot shows the Avaya Aura System Manager 6.0 interface. The top header includes the Avaya logo, the system name 'Avaya Aura™ System Manager 6.0', and a welcome message for user 'admin' last logged on at September 2, 2010, 12:45 PM. The navigation menu on the left includes 'Elements', 'Events', 'Groups & Roles', 'Licenses', 'Routing' (expanded), 'Domains', 'Locations', 'Adaptations', 'SIP Entities', 'Entity Links', 'Time Ranges', 'Routing Policies', 'Dial Patterns' (selected), 'Regular Expressions', 'Defaults', 'Security', 'System Manager Data', and 'Users'. The main content area is titled 'Dial Pattern Details' and contains the following sections:

- General:**
  - \* **Pattern:** 8
  - \* **Min:** 5
  - \* **Max:** 5
  - Emergency Call:**
  - SIP Domain:** avaya.com
  - Notes:** Fax Server at Site 1
- Originating Locations and Routing Policies:**
  - Buttons: Add, Remove
  - 1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	to Fax Server Site 1	0	<input type="checkbox"/>	FaxServerSite1	

  - Select: All, None
- Denied Originating Locations:**
  - Buttons: Add, Remove
  - 0 Items Refresh Filter: Enable

<input type="checkbox"/>	Originating Location	Notes
<input type="checkbox"/>		

At the bottom of the page, there is a '\* Input Required' message and 'Commit' and 'Cancel' buttons.

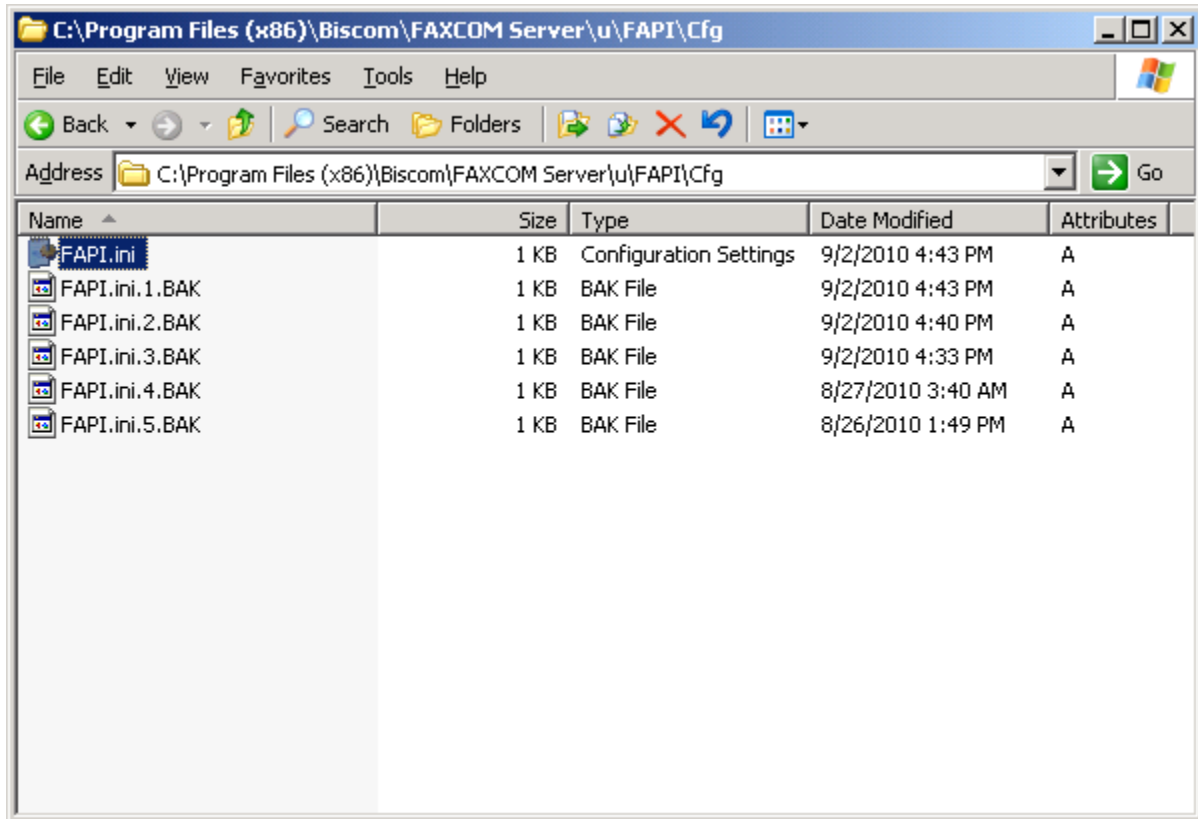
## 6. Configure Biscom FAXCOM Server

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

- Administer FAPI.ini
- Start fax service

### 6.1. Administer FAPI.ini

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



The **FAPL.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.

- **FOIP\_CALL\_CONTROL:** “SIP”
- **FOIP\_GATEWAY\_IP\_ADDRESS:** The Session Manager IP address.
- **FOIP\_LOCAL\_SYSTEM\_IP\_ADDRESS:** The local FAXCOM Server IP address.

```
#
# cfg\FAPL.ini generated by fapiconfig, wed Aug 25 09:39:34 2010
#
[configuration]
BOARD_COUNT           = 1
BOARD_TYPE            = SR140
DID_DIGITS            = 4
FOIP_CALL_CONTROL     = SIP
FOIP_CALL_CONTROL_VARIANT = DEFAULT
FOIP_GATEWAY_IP_ADDRESS = 10.64.20.31
FOIP_LOCAL_SYSTEM_IP_ADDRESS = 10.64.21.100
LOG_LEVEL             = BASIC
MAXIMUM_FAX_RESOLUTION = 200_DPI
MODE                  = T38
SHORT_DID             = REJECT
SR140_IP_INTERFACE    = DEFAULT
SR140_LICENSED_CHANNELS = 2
SR140_VIRTUAL_BOARD_41_SIZE = 2
T38_FAX_VERSION       = VERSION_0
```

## 6.2. Start Fax Service

From the Biscom FAXCOM Server, select **Start → All Programs → FAXCOM → FAXCOM Server Administrator**. If the service is not already running, click the “Start or Resume Service” button in the top left corner.

The screenshot displays the FAXCOM Server Administrator application window. The title bar reads "FAXCOM Server Administrator on SYCTAG-GRJ2YC1 (C:\Program Files (x86)\Biscom\FAXCOM Server\u, version 06.01.0900)". The menu bar includes File, Edit, View, Action, Configure, Window, and Help. The toolbar contains several icons, with the first icon (a play button) circled in red. The main window is divided into several sections:

- Job Statistics:** Contains "Data Selection" with radio buttons for "Cumulative Count" (selected) and "Hourly Average". It also has "Time Span" (radio buttons for "System Lifetime", "Since Counter Reset", "Last Hour") and "Job Type" (radio buttons for "All" (selected), "Transmit", "Receive"). Below this, it shows "System Lifetime Started at: 08/23/2010 16:29 -0600 Elapsed: 10 Days 00 Hours 28 Minutes" and "Counter Last Reset at: 09/02/2010 16:44 -0600 Elapsed: 0 Days 00 Hours 13 Minutes".
- Active Fax Ports:** Shows "Total: 0", "Transmitting: 0", and "Receiving: 0". There are "Reset Counter" and "Performance Monitor" buttons.
- Table:** A table with columns: Fax Port, Attempts, Pages, Successful, Conn Errors, Non-conn Errors. The row "All Fax Ports" shows 0 for all values.
- Tasks:** A table with columns: Task ID, Source, Fax Port, Status. It lists several tasks, including "FAX ports enabled (2 tx, 2 rx)", "FAXCOM workflow enabled", "FAXCOM service active via TRAN:6001", and "FAXCOM service active via TCP:6000".

The status bar at the bottom indicates "Service Running" and "Active Tx: 0 Rx: 0".

## **7. General Test Approach and Test Results**

### **7.1. General Test Approach**

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.

### **7.2. Test Results**

All test cases were executed. The one observation noted from the compliance test is that for inter-site fax calls over the SIP trunks, the media shuffling for the SIP trunk between the two sites has to be turned off.

## 8. Verification Steps

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

### 8.1. Verify Avaya Aura™ Communication Manager

On Communication Manager, 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 13
                        STATUS SIGNALING GROUP

      Group ID: 13                Active NCA-TSC Count: 0
      Group Type: h.323          Active CA-TSC Count: 0

Group State: in-service
```

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 4.6**. Verify that all trunks are in the “in-service/idle” state as shown below.

```
status trunk 13
                                                    Page 1

                        TRUNK GROUP STATUS

Member   Port      Service State      Mtce Connected Ports
                               Busy

0013/001 T00377   in-service/idle    no
0013/002 T00378   in-service/idle    no
0013/003 T00379   in-service/idle    no
0013/004 T00380   in-service/idle    no
0013/005 T00381   in-service/idle    no
0013/006 T00382   in-service/idle    no
0013/007 T00383   in-service/idle    no
0013/008 T00384   in-service/idle    no
0013/009 T00385   in-service/idle    no
0013/010 T00386   in-service/idle    no
0013/011 T00387   in-service/idle    no
0013/012 T00388   in-service/idle    no
0013/013 T00389   in-service/idle    no
0013/014 T00390   in-service/idle    no
```



## 8.2. 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. Verify that the status of all configured ports is “Idle”. During compliance testing, two fax ports were pre-configured on the FAXCOM Server.

The screenshot displays the FAXCOM Server Administrator application window. The title bar reads "FAXCOM Server Administrator on SVCTAG-GRJ2YC1 (C:\Program Files (x86)\Biscom\FAXCOM Server\u, version 06.01.0900)". The interface includes a menu bar (File, Edit, View, Action, Configure, Window, Help) and a toolbar with various icons. The main area is divided into several sections:

- Job Statistics:** Contains "Data Selection" options for Calculation Method (Cumulative Count, Hourly Average), Time Span (System Lifetime, Since Counter Reset, Last Hour), and Job Type (All, Transmit, Receive). It also shows system and counter reset times, active fax port counts (Total: 0, Transmitting: 0, Receiving: 0), and a table for Fax Port statistics.
- Fax Ports:** A table with columns Name, Mode, and Status. Two ports are listed: fax01 and fax02, both in "Transmit/Receive" mode and "Idle" status. The "Idle" status for both ports is circled in red.
- Tasks:** A table listing task IDs, sources, fax ports, and their descriptions.

At the bottom of the window, a status bar indicates "Service Running" and "Active Tx: 0 Rx: 0".

Name	Mode	Status
fax01	Transmit/Receive	Idle
fax02	Transmit/Receive	Idle

Task ID	Source	Fax Port	Status
0004			FAX ports enabled (2 tx, 2 rx)
0001			FAXCOM workflow enabled
0002	host2		FAXCOM service active via TRAN:6001
0003	host1		FAXCOM service active via TCP:6000

## 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™ Session Manager 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 6.0, Release 6.0, August 2010, available at <http://support.avaya.com>.
2. *Administering Avaya Aura™ Session Manager*, Document 03-603324, Issue 3, Release 6.0, August 2010, available at <http://support.avaya.com>
3. *FAXCOM Server Administrator's Guide*, February 2010 Revised Edition, available from Biscom Technical Support.
4. *KB Avaya 20100518*, Knowledge Base article under “SR140 Avaya 6.x”, available from Biscom Technical Support.

---

**©2010 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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](mailto:devconnect@avaya.com).