Dialogic® Brooktrout® SR140 Fax Software with Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager 6.1.0

Installation and Configuration Integration Note

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1 Scope

This document is intended as a general guide for configuring a basic installation of the ***Avaya Aura® Communication Manager and Avaya Aura® Session Manager*** using a SIP trunk interface with the Dialogic® Brooktrout® SR140 Fax over IP (FoIP) software platform. The interoperability includes SIP call control and T.38/T.30 media.

The configuration information contained in this document is based on the following document from Avaya:

Avaya Solution & Interoperability Test Lab Report: Application Notes for Configuring Dialogic Brooktrout SR140 Fax Software with Avaya Aura® Communication Manager and Avaya Aura® Session using a SIP Trunk Interface - Issue 0.1.[6]

The Dialogic® Brooktrout® SR140 Fax Software is a host-based Fax over IP (FoIP) engine utilized by fax servers to send and receive fax calls over an IP network. In the tested configuration, Avaya Aura® Session Manager routed fax calls via a SIP trunk to and from a fax server utilizing the Dialogic® Brooktrout® SR140 Fax Software.

This document is not intended to be comprehensive, and thus does not replace the manufacturer's detailed configuration documentation. Users of this document should already have a general knowledge of how to install and configure the ***Avaya Aura® Communication Manager and Avaya Aura® Session Manager*** using a SIP trunk interface.

The sample configuration shown and/or referred in the subsequent sections was used for **Avaya DevConnect Certification Testing** performed at **Avaya DevConnect Labs**. Therefore, it is quite possible that the sample configuration will not match an exact configuration or versions that would be present in a deployed environment. However, the sample configuration does provide a possible starting point to work with the equipment vendor for configuring your device. Please consult the appropriate manufacturer's documentation for details on setting up your specific end user configuration.

For ease of reference, the Dialogic® Brooktrout® SR140 Fax Software will sometimes be denoted herein as SR140. The ***Avaya Aura® Communication Manager*** will be denoted herein as Avaya Aura CM and the ***Avaya Aura® Session Manager*** will be denoted Avaya Aura SM, or some other form thereof. All references to the SDK herein refer to the Dialogic® Brooktrout® Fax Products SDK.

2 Configuration Details

The following systems were used for the sample configuration described in the document.

2.1 Avaya Aura® Communication Manager & Aura® Session Manager with G650 & G450 Media Gateways

Vendor	***Avaya***
SW Model	***Aura® Communication Manager 6.0.1*** ***Aura® Session Manager 6.1.0***
Hardware	***Avaya G650 Media Gateway*** ***Avaya G450 Media Gateway*** ***Refer to Network Configuration Section for details***
IP Device	***Dialogic® Brooktrout® SR140 Fax Server***
Protocol to SR140 Fax Software	***SIP***
Additional Notes	***DSP resources required as noted below in the Avaya DSP Resource section***

Avaya DSP Resources
Fax calls consume DSP (Digital Signal Processing) resources for processing fax data on the TN2302AP IP Media Processor (MedPro) circuit pack and the TN2602AP IP Media Processor circuit pack in the Avaya G650 Media Gateway, and the integrated Voice over Internet Protocol (VoIP) engine of the Avaya G450 Media Gateway. To increase the capacity to support simultaneous fax calls, additional TN2302AP and/or TN2602AP MedPro circuit packs may need to be installed in the Avaya G650 Gateway, and additional Avaya MM760 Media Module or Modules may need to be installed in the Avaya G450 Media Gateway. The information contained in the table below indicates DSP capacities/usage in the Avaya media processors. Customers should work with their Avaya sales representatives to ensure that their fax solutions have adequate licenses and DSP resources to match the intended fax capacity/usage.

Platform Device	**DSP Resources per Platform Device**	**DSP Resources per FoIP Call**
TN2302, MM760	64	4
TN2602	64	1

2.2 Dialogic® Brooktrout® SR140 Fax Software

Vendor	***Dialogic***
Model	***Dialogic® Brooktrout® SR140 Fax Software***
Software Version	***Tested with SDK 6.2.4***
Protocol to Aura Session Manager	***SIP***
Additional Notes	***None***

2.3 Network System Configuration

The test configuration was designed to emulate two separate sites with multiple Port Networks at one site (Site 1), and modular Gateway resources at the other site (Site 2). **Figure 1** illustrates the configuration used in these Application Notes, with a focus on the configuration at Site 2. Communication Manager Servers and Gateways at the two sites were connected via SIP and ISDN-PRI trunks. Faxes were alternately sent between the two sites using these two facilities. The fax servers communicated directly with Session Manager via SIP. Each Session Manager was configured using a System Manager (not shown).

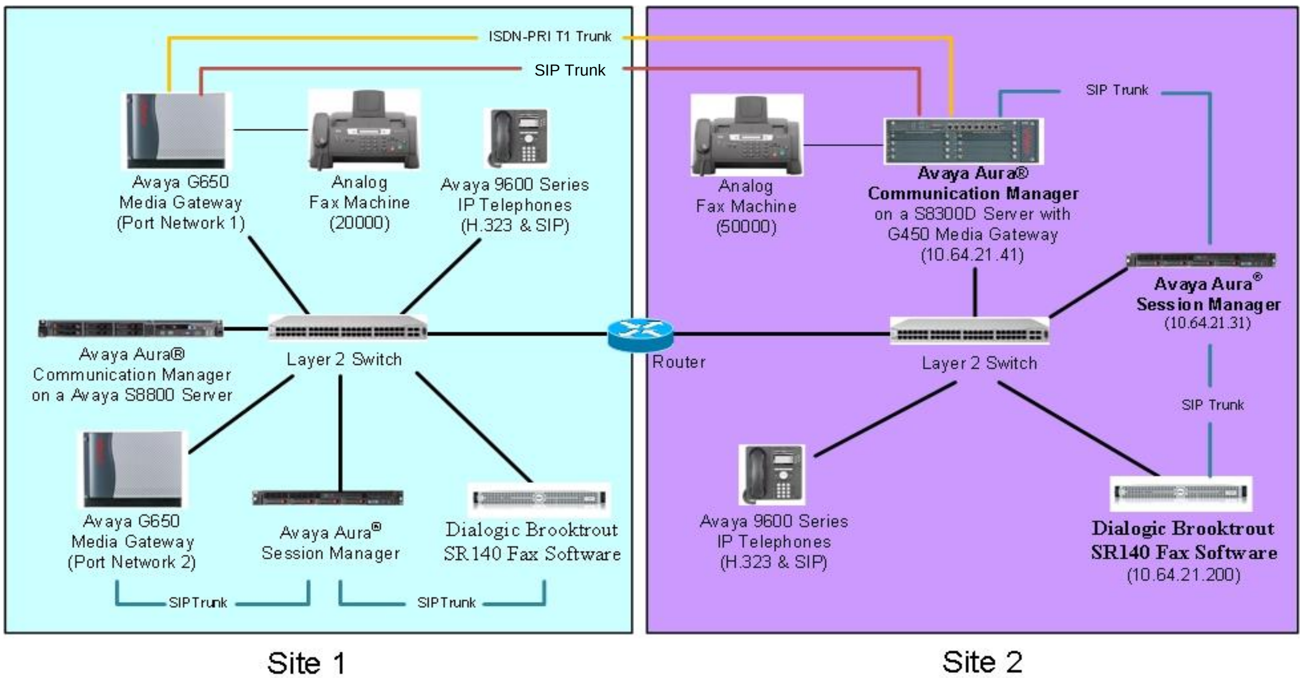


Figure 1: Dialogic Brooktrout SR140 Fax Software with Session Manager

Site 1 had an Avaya S8800 Server running Communication Manager with two Avaya G650 Media Gateways. Each media gateway was configured as a separate port network in separate IP network regions. The fax server at this site communicated with Communication Manager via an SIP trunk which terminated on a CLAN circuit pack in port network 2. IP media resources were provided by Media Processor (MedPro) circuit packs. Two versions of the MedPro circuit pack were tested in this configuration: TN2302AP and TN2602AP. Endpoints at this site included an Avaya 9600 Series IP Telephone (with SIP and SIP firmware), and an analog fax machine.

Site 2 had an Avaya S8300D Server running Communication Manager in an Avaya G450 Media Gateway. The fax server at this site communicated with Communication Manager via an SIP trunk. On the Avaya G450 Media Gateway, the signaling and media resources supporting the SIP trunk were integrated directly on the media gateway processor. Endpoints at this site included Avaya 9600 Series IP Telephones (with SIP and SIP firmware), and an analog fax machine.

The IP phones (SIP) at each site had no specific role in fax operations; therefore, this part of the configuration is not covered in these Application Notes. They were present in the configuration to verify VoIP telephone calls had on adverse impact on the FoIP faxing operations.

A fax call originating from a local fax server was sent to Session Manager via an SIP trunk. Based on the dialed digits, Session Manager and Communication Manager routed the fax call either to the local fax machine or to one of the trunks (ISDN-PRI or SIP) to reach the remote site. When the fax call reached the remote site, the Communication Manager at that site routed the call either to the local fax machine or to Session Manager for onward routing to the local fax server over the SIP trunk. The diagram below details the sample configuration used in connection with this document.

2.4 Avaya Aura® Equipment - Site 1

Equipment	Software/Firmware
Avaya S8800 Server	Avaya Aura® System Manager: 6.0.0 (Build No. – 6.0.0.0.668-3.0.7. 2) (Avaya Aura® System Platform: 6.0.2.1.5)
Avaya S8800 Server	Avaya Aura® Session Manager 6.0.2.0.602004
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0.1 (R016x.00.1.510.1 with Patch 19358)
Avaya G650 Media Gateway (at Main Site) - 2 CLANs - 2 MedPros – TN2302 - 2 MedPros – TN2602	TN799DP - HW01 FW38 & HW13 FW 38 TN2302AP - HW20 FW120 TN2602AP - HW02 FW057
Avaya 9600 Series IP Deskphones (SIP)	Release 3.1 Service Pack 3 (96x0) Release 6 Service Pack 5 (96x1G)
Avaya 9600 Series IP Deskphones (SIP)	Release 2.6 Service Pack 5 (96x0) Release 6 Service Pack 2 (96x1G)
Analog Fax Machine	-N/A-
Fax Server – Dialogic FaxDiagTool on a Windows 2008 Server	Compiled with SDK 6.2.4
Dialogic Brooktrout SR140 Fax Software – Boston Bfv API – Boston Driver – Boston SDK – Boot ROM	v6.2.4 (Build 12) v6.2.0 (Build 4) v6.2.4 (Build 12) 6.2.1B9

2.5 Avaya Aura® Equipment - Site 2

Equipment	Software/Firmware
Dell™ PowerEdge™ R610 Server	Avaya Aura® System Manager: 6.1.0 (Build No. – 6.1.0.0.7345-6.1.5.502), Software Update Revision No : 6.1.9.1.1634 (Avaya Aura® System Platform: 6.0.3.4.3)
HP ProLiant DL360 G7 Server	Avaya Aura® Session Manager 6.1.5.0.615006
Avaya S8300D Media Server	Avaya Aura® Communication Manager 6.0.1 (R016x.00.1.510.1 with Patch 19303)
Avaya G450 Media Gateway	31.18.1
Avaya 9600 Series IP Deskphones (SIP)	Release 3.1 Service Pack 3 (96x0) Release 6 Service Pack 5 (96x1G)
Avaya 9600 Series IP Deskphones (SIP)	Release 2.6 Service Pack 5 (96x0) Release 6 Service Pack 2 (96x1G)
Analog Fax Machine	-N/A-
Fax Server – Dialogic FaxDiagTool on a Windows 2003 Server	Compiled with SDK 6.2.4
Dialogic Brooktrout SR140 Fax Software – Boston Bfv API – Boston Driver – Boston SDK – Boot ROM	 v6.2.4 (Build 12) v6.2.0 (Build 4) v6.2.4 (Build 12) 6.2.1B9

2.6 Network Addresses

The following table lists the IP addresses and their descriptions used in subsequent sections.

Device #	Device Make, Model, and Description	Device IP Address
1	Avaya Aura Session Manager	10.64.21.31
2	SR140 Fax Server	10.64.21.200

3 Prerequisites

- None

4 Summary of Limitations

- None

5 Avaya Aura® Communication Manager Setup Notes

This section describes the Communication Manager configuration necessary to interoperate with the Dialogic Brooktrout SR140 Fax Software. It focuses on the configuration of the routing and SIP trunk between Communication Manager and Session Manager. All other components are assumed to be in place and previously configured, including the SIP and ISDN-PRI trunks that connect Sites 1 and 2 in **Figure 1**.

The examples shown in this section refer to Site 2. Similar steps also apply to Site 1 using values appropriate for that location.

The configuration of Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration, the **save translation** command was used to make the changes permanent.

The procedures for configuring Communication Manager include the following steps:

1. Verify Communication Manager License
2. Administer IP Network Region
3. Administer IP Codec Set
4. Administer IP Node Names
5. Administer SIP Signaling Group
6. Administer SIP Trunk Group
7. Administer Private Numbering
8. Administer Route Pattern
9. Administer AAR Analysis

Step	Description
1.	**Verify Communication Manager License** Use the **display system-parameters customer-options** command to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. 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 trunks permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to acquire the appropriate licenses.

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

 IP PORT CAPACITIES                                             USED
                      Maximum Administered SIP Trunks: 12000 57
            Maximum Concurrently Registered IP Stations: 18000 9
              Maximum Administered Remote Office Trunks: 12000 0
 Maximum Concurrently Registered Remote Office Stations: 18000 0
               Maximum Concurrently Registered IP eCons: 414   0
   Max Concur Registered Unauthenticated SIP Stations: 100   0
                         Maximum Video Capable Stations: 18000 0
                    Maximum Video Capable IP Softphones: 18000 1
                        Maximum Administered SIP Trunks: 24000 170
   Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
    Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                              Maximum TN2501 VAL Boards: 128   0
                      Maximum Media Gateway VAL Sources: 250   1
            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.)
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>Administer IP Network Region
Use the change ip-network-region command to administer the network region settings. The values shown below are the values used during compliance testing. Note that the IP-IP Direct Audio settings must be disabled.

Authoritative Domain: avaya.com
Name: Any descriptive name may be used (if desired).
Intra-region IP-IP Direct Audio: no
Inter-region IP-IP Direct Audio: no
By default, IP-IP direct audio (media shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Shuffling can be further restricted at the trunk level on the Signaling Group form.
Codec Set: 1 The codec set contains the list of codecs available for calls within this IP network region.

<pre>
change ip-network-region 1   Page   1 of 20
                                IP NETWORK REGION
   Region: 1
 Location: Authoritative Domain: avaya.com
     Name:
 MEDIA PARAMETERS   Intra-region IP-IP Direct Audio: 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   AUDIO RESOURCE RESERVATION PARAMETERS
 SIP IP ENDPOINTS   RSVP Enabled? n
   SIP Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
    Keep-Alive Interval (sec): 5
             Keep-Alive Count: 5
</pre></td></tr>
</table>

Step	Description
3.	**Administer IP Codec Set** Use the **change ip-network-set** command to administer an IP codec set. IP codec set 1 was used during compliance testing. Multiple codecs can be listed in priority order to allow the codec used by a specific call to be negotiated during call establishment. The example below shows the values used during compliance testing.

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

                           IP Codec Set

     Codec Set: 1

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

On **Page 2**, set the **FAX Mode** field to ***t.38-standard***. The **Modem Mode** field should be set to ***off***.

Leave the **FAX Redundancy** setting at its default value of ***0.*** A packet redundancy level can be assigned to improve packet delivery and robustness of FAX transport over the network (with increased bandwidth as trade-off). Avaya uses IETF RFC-2198 and ITU-T T.38 specifications as redundancy standard. With this standard, each Fax over IP packet is sent with additional (redundant) 0 to 3 previous fax packets based on the redundancy setting. A setting of 0 (no redundancy) is suited for networks where packet loss is not a problem.

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

                           IP Codec Set

                               Allow Direct-IP Multimedia? y
               Maximum Call Rate for Direct-IP Multimedia:  2048:Kbits
      Maximum Call Rate for Priority Direct-IP Multimedia:  2048:Kbits

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>Administer IP Node Names
Use the change node-names ip command to create a node name and enter the IP address of Session Manager. Enter a descriptive name in the Name column and the Session Manager IP address in the IP address column. Also note the node name of the processor (procr) as it will be used later to configure the SIP trunk between Communication Manager and Session Manager.
<pre>
change node-names ip   Page   1 of   2
                                    IP NODE NAMES
     Name   IP Address
 AES_21_46   10.64.21.46
 CM_20_40   10.64.20.40
 CM_22_12_CLAN1A   10.64.22.16
 CM_22_12_CLAN2A   10.64.22.19
 IPO_21_64   10.64.21.64
 SM_20_31   10.64.20.31
 SM_21_31   10.64.21.31
 default   0.0.0.0
 msgserver   10.64.21.41
 procr   10.64.21.41
 procr6   ::
</pre></td></tr>
</table>

Step	Description
5.	**Administer SIP Signaling Group** During compliance testing, a SIP signaling group and the associated SIP trunk group was used for routing fax calls to/from the fax server via Session Manager. Use the **add signaling-group** command to create a signaling group for use by the SIP trunk to the fax server. Signaling group ***1*** was configured using the parameters highlighted below. Default values may be used for all other fields. ▪ Set the **Group Type** to ***SIP***. ▪ The **Transport Method** was set to the recommended default value of ***tls*** (Transport Layer Security). As a result, the **Near-end Listen Port** and **Far-end Listen Port** are automatically set to ***5061*** ▪ Set the **Near-end Node Name** to the node name that maps to the IP address of the processor (i.e. ***procr***) used to connect to Session Manager (see **Step 4**). ▪ Set the **Far-end Node Name** to the node name that maps to the IP address of Session Manager configured in **Step 4**. ▪ The **Far-end Network Region** was set to ***1***. This is the IP network region which contains Session Manager and the fax server. ▪ Set the **Direct IP-IP Audio Connections** field to ***n***. This setting disables Media Shuffling on the trunk level. ▪ The default values were used for all other fields.

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

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



   Near-end Node Name: procr                   Far-end Node Name: SM_21_31
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1

Far-end Domain:
                                               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? n
Session Establishment Timer(min): 3                        IP Audio Hairpinning? n
         Enable Layer 3 Test? y
                                                    Alternate Route Timer(sec): 20
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>Administer SIP Trunk Group
Trunk group 1 was configured with the add trunk-group command using the parameters highlighted below. Default values may be used for all other fields.

On Page 1:
Set the Group Type field to sip.
Enter a descriptive name for the Group Name.
Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the TAC field.
Set the Service Type field to tie.
Set the Member Assignment Method to auto.
Set the Signaling Group to the signaling group shown in the previous step.
In Number of Members field, enter the number of trunks in the trunk group. This determines how many simultaneous calls can be supported by the configuration.
Default values may be used for all other fields.
<pre>
add trunk-group 1   Page   1 of 21
                                TRUNK GROUP

Group Number: 1   Group Type: sip   CDR Reports: y
  Group Name: to SM_21_31   COR: 1   TN: 1   TAC: 101
   Direction: two-way   Outgoing Display? n
 Dial Access? n   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 50
</pre></td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td></td><td>Administer SIP Trunk Group – Continued
On Page 3:
Set the Send Name field and Send Calling Number field to y. These settings enable the sending of calling party name and number to the far end.
Set the Format field to unk-pvt. This field specifies the format of the calling party number sent to the far-end.
Default values may be used for all other fields.
<pre>
add trunk-group 1   Page   3 of 21
TRUNK FEATURES
          ACA Assignment? n   Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: unk-pvt
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? Y
</pre></td></tr>
<tr><td>7.</td><td>Administer Private Numbering
Private numbering defines the calling party number to be sent to the far-end. Use the change private-numbering command to create an entry that will be used by the trunk group defined in Step 6. In the example shown below, all calls originating from a 5-digit extension beginning with 5 and routed across any trunk group (since the Trk Grp(s) entry is blank) will be sent as a 5-digit calling number.
<pre>
change private-numbering 0   Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext   Trk   Private   Total
Len Code   Grp(s)   Prefix   Len
 5 5   5   Total Administered: 2
                                                          Maximum Entries: 540
</pre></td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>8.</td><td>Administer Route Pattern
Use the change route-pattern command to create a route pattern that will route calls to the SIP trunk that connects to Session Manager.

A descriptive name was entered for the Pattern Name field. The Grp No field was set to the trunk group created in Step 6. The Facility Restriction Level (FRL) field was set to a level that allows access to this trunk for all users that require it. The value of 0 is the least restrictive level. The Numbering Format was set to lev0-pvt. The default values were used for all other fields.
<pre>
change route-pattern 1   Page   1 of   3
                    Pattern Number: 1   Pattern Name: to SM_21_31
                             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   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   lev0-pvt 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
</pre></td></tr>
<tr><td>9.</td><td>Administer AAR Analysis
Automatic Alternate Routing (AAR) was used to route calls to the fax server via Session Manager. Use the change aar analysis command to create an entry in the AAR Digit Analysis Table for this purpose. The highlighted entry specifies that if the dialed number is 75000 and is 5 digits long, to use route pattern 1. Route pattern 1 routes calls to Session Manager.
<pre>
change aar analysis 7   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
     75000   5   5   9   aar   n
</pre></td></tr>
</table>

6 Avaya Aura® Session Manager Setup Notes

This section provides the procedures for configuring Session Manager as shown in the reference configuration. All provisioning for Session Manager is performed via the System Manager web interface. System Manager delivers a set of shared, secure management services and a common console across multiple products in the Avaya Aura® network, including the central administration of routing policies, and a common format for logs and alarms. This section assumes that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

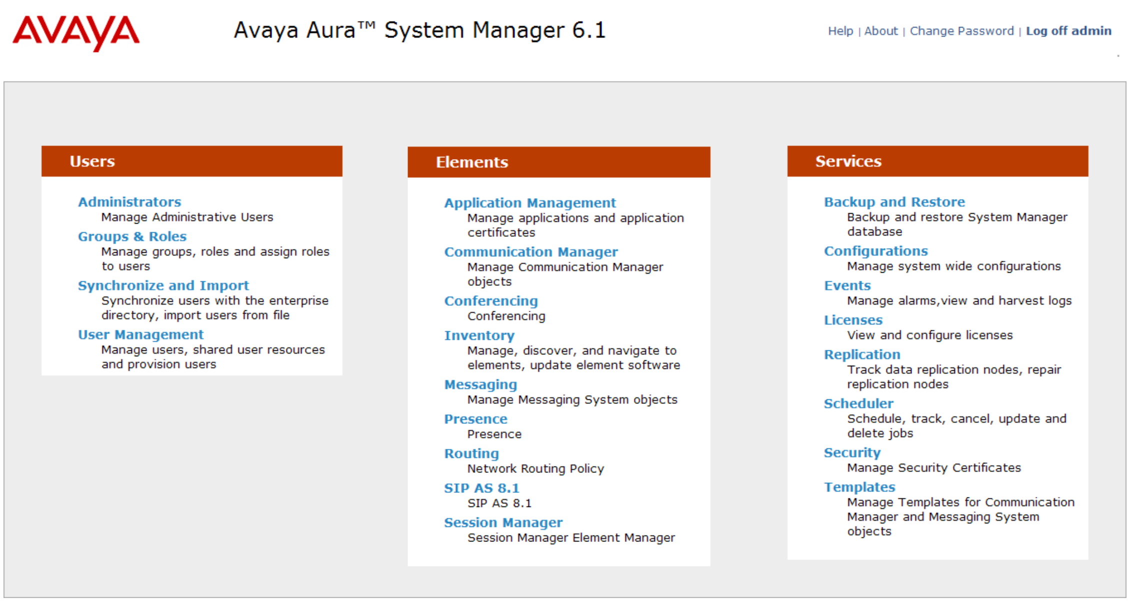
This section summarizes the configuration steps that are necessary for interoperating with Dialogic Brooktrout SR140 Fax Software. The test environment was previously configured to enable Communication Manager and Session Manager at each site to communicate with each other. Details of this configuration and the SIP endpoints are not described in this document. Additional information can be obtained from **Reference [3]**.

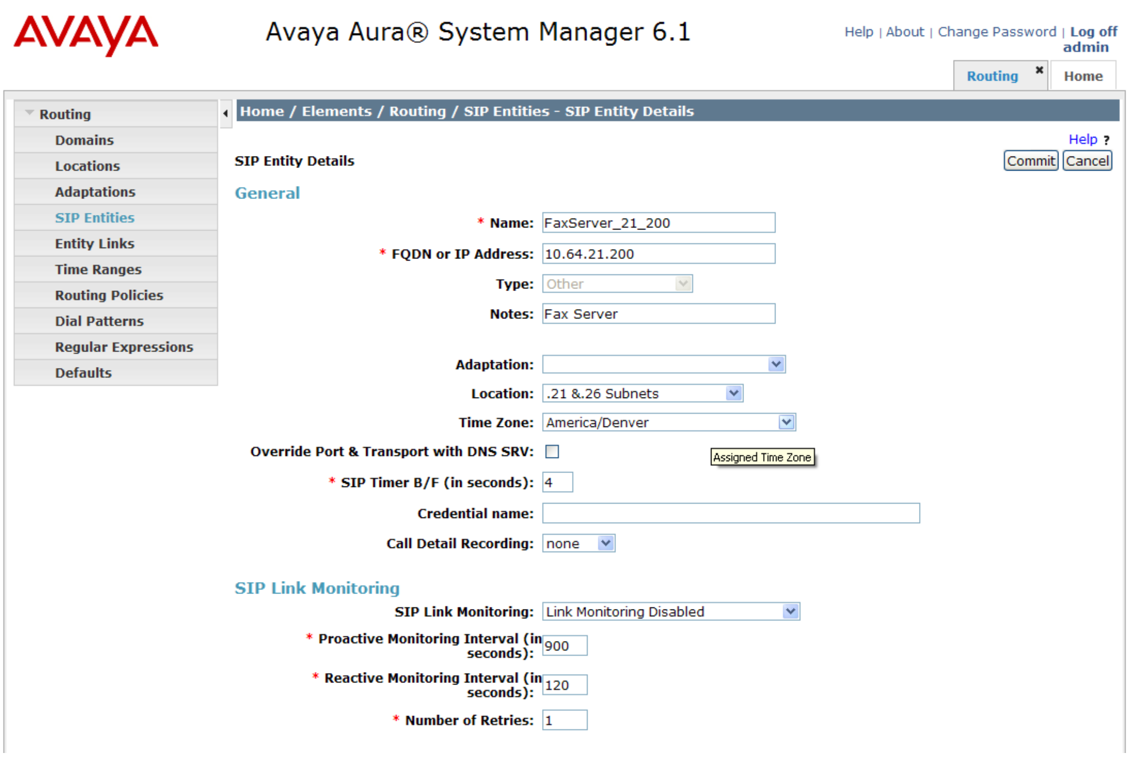
The examples shown in this section refer to Site 2. Similar steps also apply to Site 1 using values appropriate for that location.

The procedures described in this section include configurations for the following:

1. Login
2. Create a SIP Entity for the fax server
3. Create a SIP Entity Link for the fax server
4. Create a Routing Policy
5. Create a Dial Pattern

Step	Description
1.	**Login** Access the Session Manager administration web interface by entering [https://*<ip-addr>*/network-login/](https://<ip-addr>/network-login/) as the URL in an Internet browser, where *<ip-addr>* is the IP address of the System Manager server. Log in using appropriate credentials. The main page for the administrative interface is shown below.



Step	Description
2.	**Create a SIP Entity for the Fax Server** A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. During compliance testing, a SIP Entity was added for the Session Manager itself (not shown), Communication Manager (not shown), and the fax server. Navigate to **Routing→SIP Entities**, and click the **New** button (not shown) to add a SIP Entity. The configuration details for the SIP Entity defined for the fax server are as follows: Under **General**: • **Name**: a descriptive name • **FQDN or IP Address**: ***10.64.21.200*** as specified in **Figure 1**. • **Type**: select ***Other*** Default settings can be used for the remaining fields. Click **Commit** to save the SIP Entity definition. The screen below shows the SIP Entity configuration details for the fax server. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>

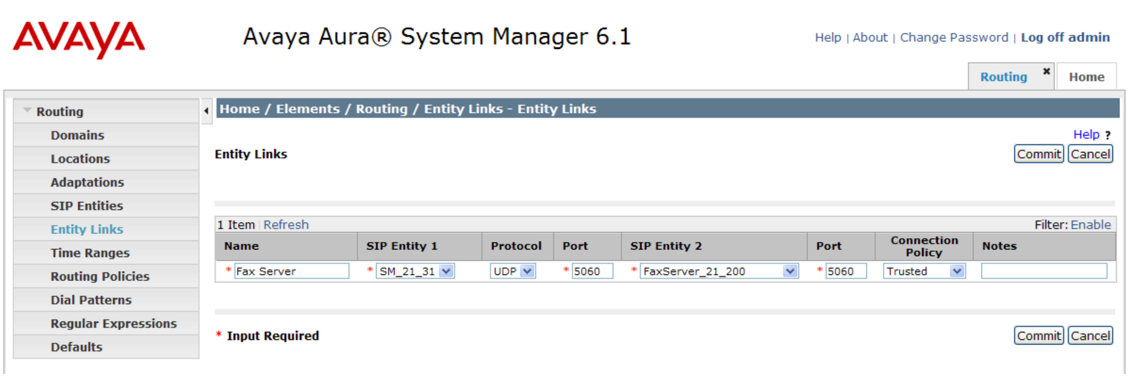
Create an Entity Link for the Fax Server
A SIP trunk between Session Manager and a telephony system is described by an Entity link. Two Entity Links were created:

- Session Manager ←→ Communication Manger (not shown)
- Session Manager ←→ Fax Server

Navigate to **Routing→Entity Links**, and click the **New** button (not shown) to add a new Entity Link. The screen below shows the configuration details for the Entity Link connecting Session Manager to the fax server.

- **Name**: a descriptive name
- **SIP Entity 1**: select the Session Manager SIP Entity.
- **Protocol**: select ***UDP*** as the transport protocol.
- **Port**: ***5060***. This is the port number to which the other system sends SIP requests.
- **SIP Entity 2**: select the fax server SIP Entity.
- **Port**: ***5060***. This is the port number on which the other system receives SIP requests.
- **Trusted**: check this box
- **Notes**: optional descriptive text

Click **Commit** to save the configuration.



</td></tr>
</table>

4.

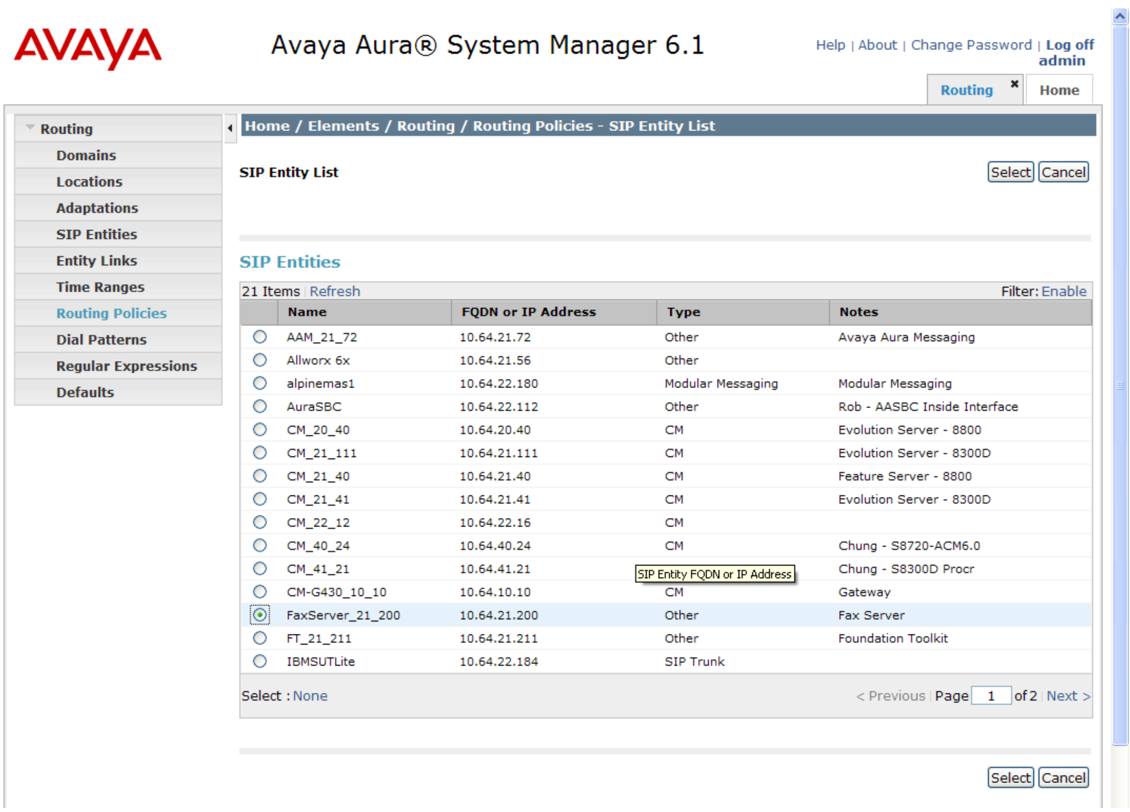
Create a Routing Policy

Routing policies describe the conditions under which calls will be routed to the SIP Entities connected to the Session Manager. Routing Policies were added for routing fax calls to the fax server and calls from the fax server to other SIP entities (not shown).

Navigate to **Routing→Routing Policies**, and click the **New** button (not shown) to add a new Routing Policy.
Under **General**:

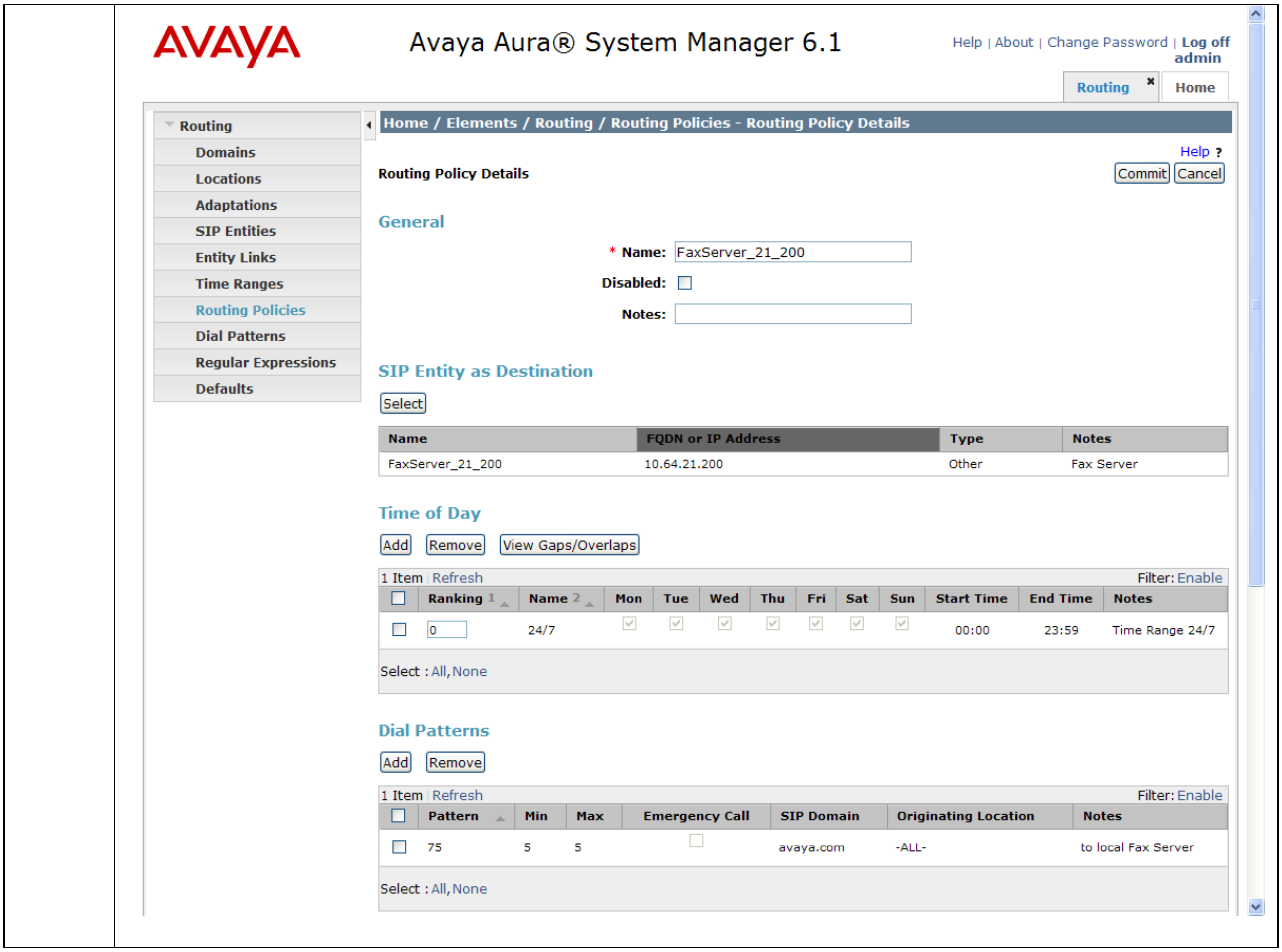
- **Name**: a descriptive name
- **Notes**: optional descriptive text

Under **SIP Entity as Destination**
Click the **Select** button and the screen below is displayed. Select the fax server SIP Entity, to which the routing policy applies, and click the **Select** button to return to the previous screen.



Under **Time of Day**
Click **Add** to select a Time Range (not shown since the default time range of 24/7 was used during compliance testing).

Default settings can be used for the remaining fields. Click **Commit** to save the configuration. The screen below shows the routing policy used during compliance testing.


Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
Routing
Home
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / Routing Policies - Routing Policy Details
Help ?
Routing Policy Details
Commit
Cancel
General
* Name: FaxServer_21_200
Disabled:
Notes:
SIP Entity as Destination
Select
Name
FQDN or IP Address
Type
Notes
FaxServer_21_200
10.64.21.200
Other
Fax Server
Time of Day
Add
Remove
View Gaps/Overlaps
1 Item | Refresh
Filter: Enable
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
Select : All, None
Dial Patterns
Add
Remove
1 Item | Refresh
Filter: Enable
Pattern
Min
Max
Emergency Call
SIP Domain
Originating Location
Notes
75
5
5
avaya.com
-ALL-
to local Fax Server
Select : All, None

5. **Create Dial Pattern**

Dial Patterns define digit strings to be matched against dialed numbers for directing calls to the appropriate SIP Entities.5-digit numbers beginning with “75” were routed to the fax server.

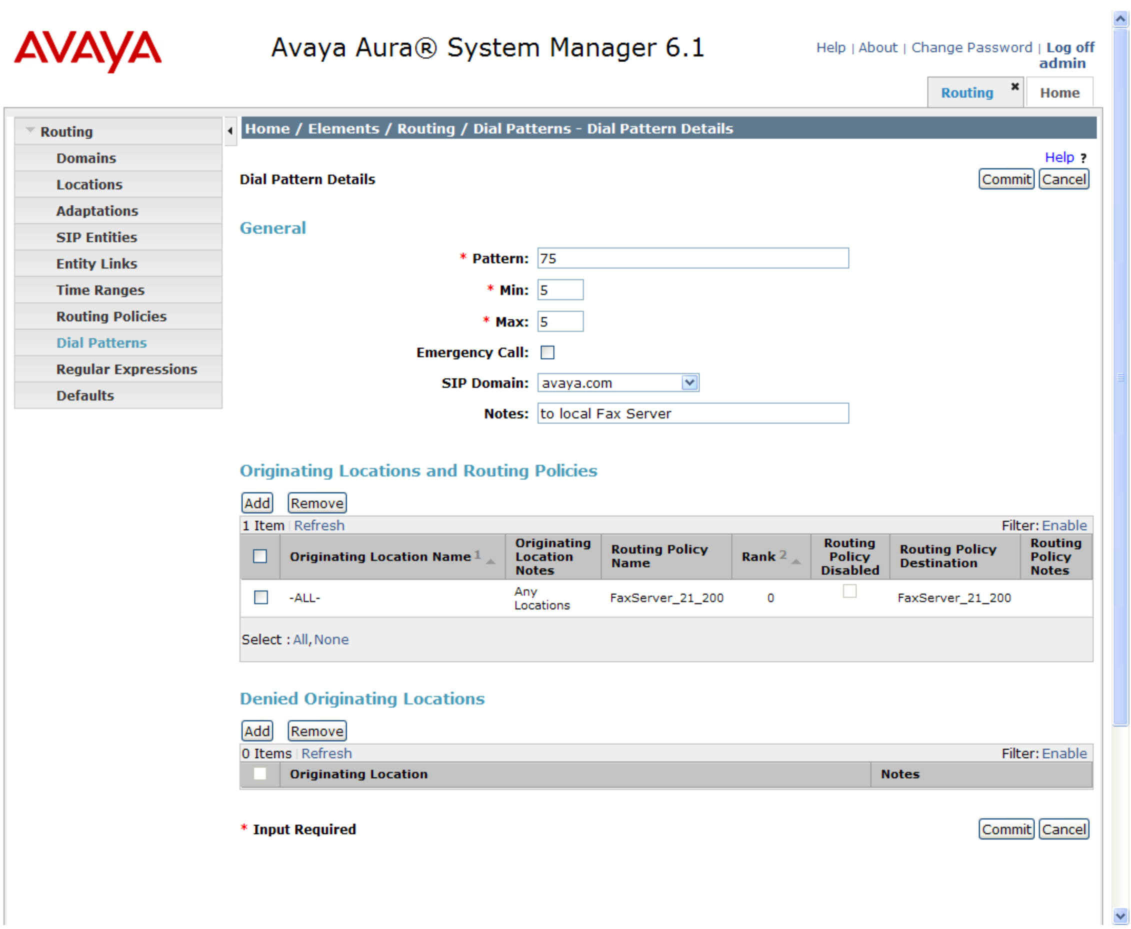
Navigate to **Routing→Dial Patterns**, click the **New** button (not shown) to add a new Dial Pattern.

Under **General**:

- **Pattern**: dialed number or prefix
- **Min**: minimum length of dialed number
- **Max**: maximum length of dialed number
- **SIP Domain**: select the SIP Domain created in **Step 2** (or select **–ALL–** to be less restrictive)
- **Notes**: optional descriptive text

Under **Originating Locations and Routing Policies**
Click **Add** to select the appropriate originating Location (e.g. ***–ALL–***) and Routing Policy (e.g. ***FaxServer_21_200***) from the list (not shown).

Default settings can be used for the remaining fields. Click **Commit** to save the configuration.



7 Dialogic® Brooktrout® SR140 Software Setup Notes

This section describes the configuration of the Dialogic Brooktrout SR140 Fax Software. It assumes that that a fax server application and all required software components, including Dialogic Brooktrout SR140 Fax Software, have been installed and properly licensed. For instructions on installing Dialogic Brooktrout SR140 Fax Software, consult the Dialogic Brooktrout SR140 Fax Software documentation (**Reference [4]**).

Note that the configurations documented in this section pertain to interoperability between the Dialogic Brooktrout SR140 Fax Software and the Avaya SIP infrastructure. The standard configurations pertaining to the Dialogic Brooktrout SR140 Fax Software itself (e.g., administering fax channels) are not covered. For instructions on administering and operating the Dialogic Brooktrout SR140 Fax Software, consult the Dialogic Brooktrout SR140 Fax Software documentation (**Reference [4]**).

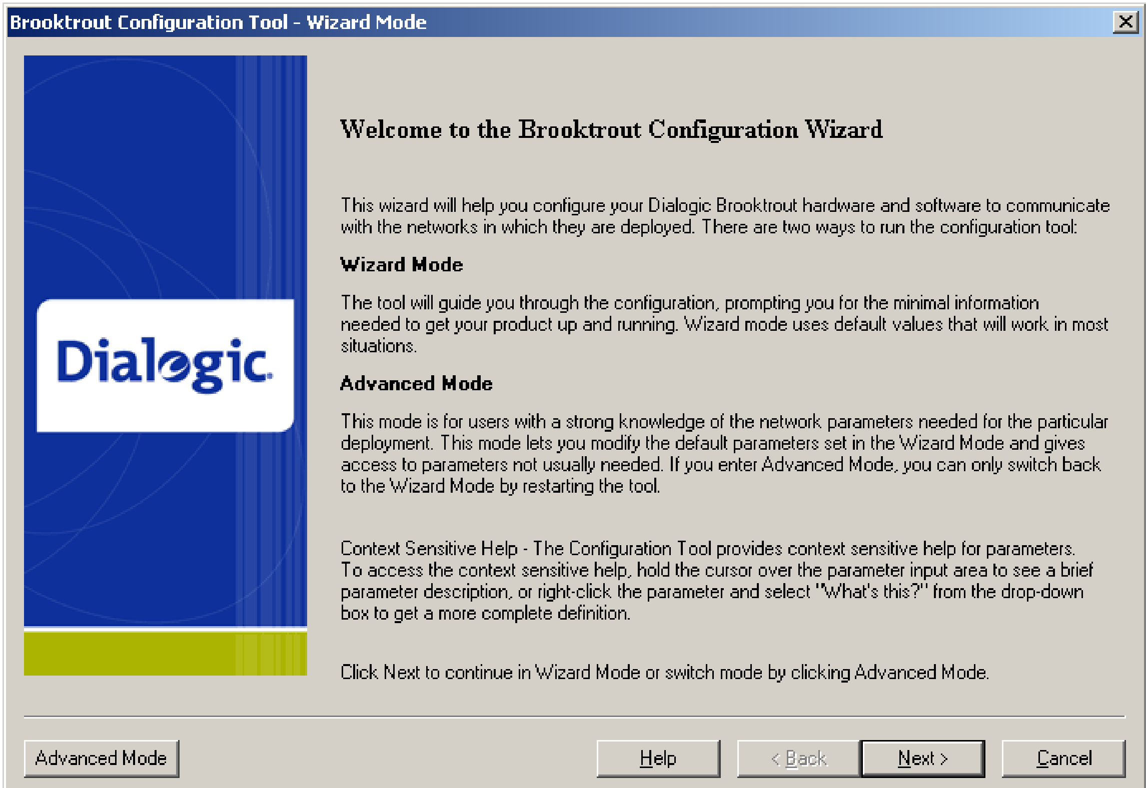
The examples shown in this section refer to Site 2 in **Figure 1**. Similar steps also apply to Site 1 using values appropriate for that location.

The configuration procedures covered in this section include the following:

1. Launch Brooktrout Configuration Tool
2. Configure IP Stack
3. Configure SIP IP Parameters
4. Configure T.38 Parameters
5. Complete RTP Parameters
6. Configure RTP Port Range
7. Complete Brooktrout SR140 Configuration

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>

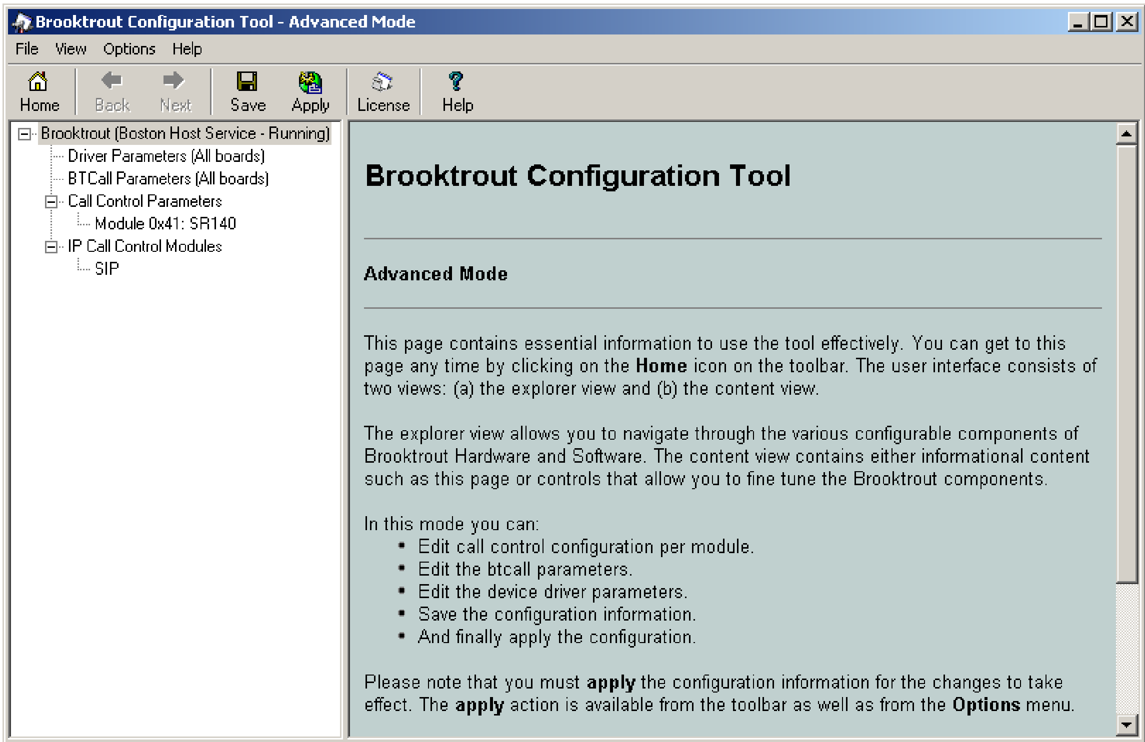
Launch Brooktrout Configuration Tool
Navigate to the path of the Brooktrout configuration tool (i.e. **configtool.exe**) and launch the tool. The **Brooktrout Configuration Tool – Wizard Mode** window gets displayed. Click the **Advanced Mode** button on the bottom left.



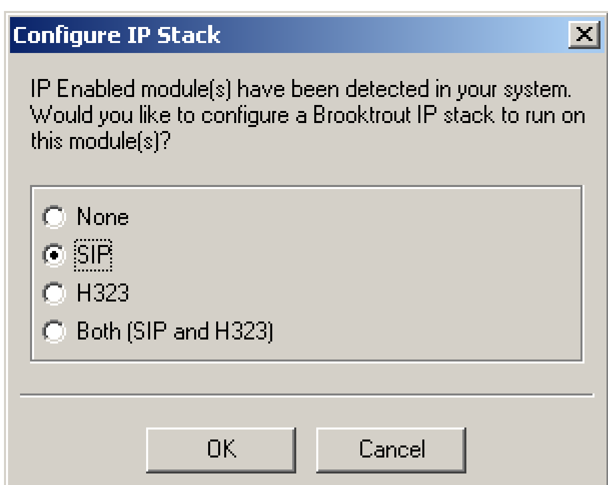
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>

Configure IP Stack
The following configuration tool window is displayed.



Select **Options → Configure IP Stack** from the top menu. The screen below is displayed.
Select **SIP** and click **OK**.



</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>

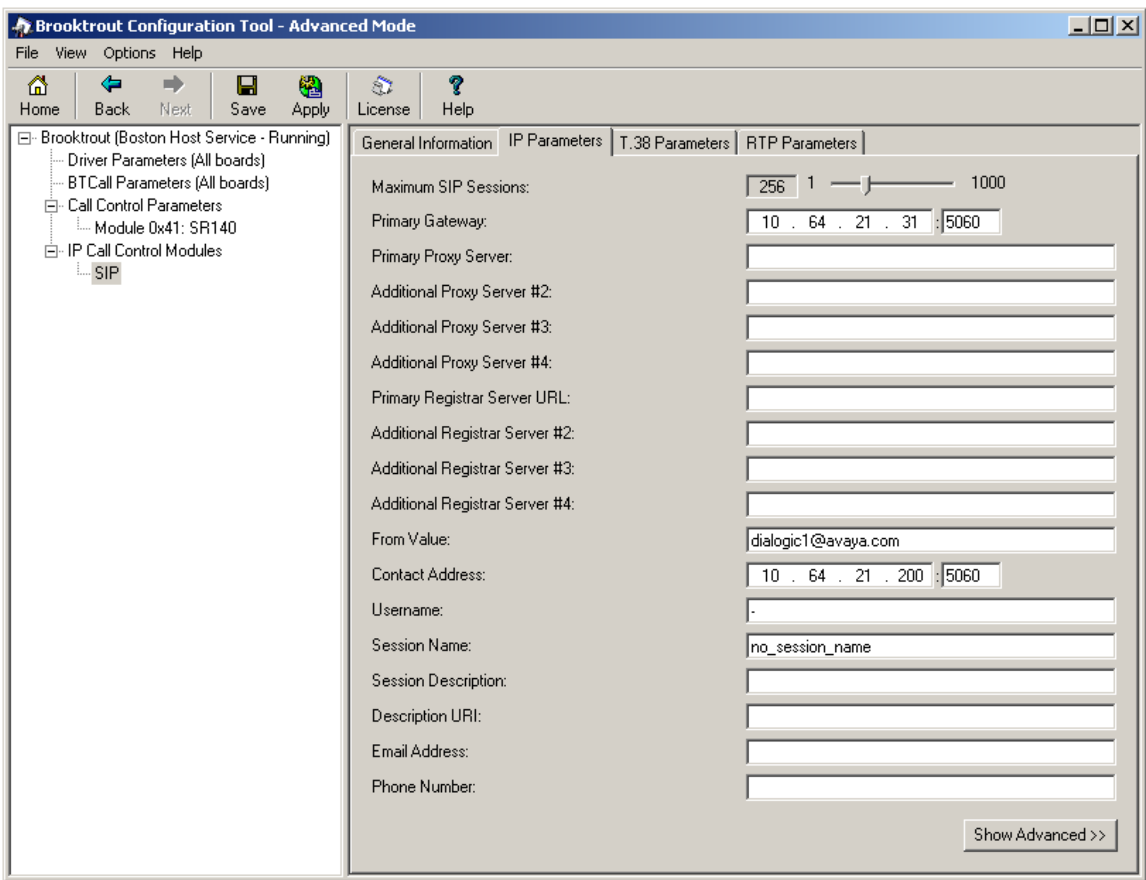
Configure SIP IP Parameters

Important*: This step describes configuring the Primary SIP Gateway address using the Brooktrout Configuration Tool. This method is sufficient if the fax server will communicate with a single SIP gateway. Refer to the Dialogic Brooktrout SR140 Fax Software documentation for configuration details if the fax server will communicate with multiple SIP gateways.*

From the pane on the left, navigate to **Brooktrout → IP Call Control Modules → SIP** in the left navigation menu. Select the **IP Parameters** tab in the right pane. Configure the fields as follows:

- **Primary Gateway** –set to the IP address of Session Manager, and port number ***5060***.
- **From Value** – set to appropriate ***UserInfo@DomainName***. The *DomainName* should be set to the authoritative domain as configured in Session Manager.
- **Contact Address** – set to the IP address assigned to the fax server and port number ***5060***.
- **Username** – Required. Default value is a dash ('-') character.

Use default values for all other fields.

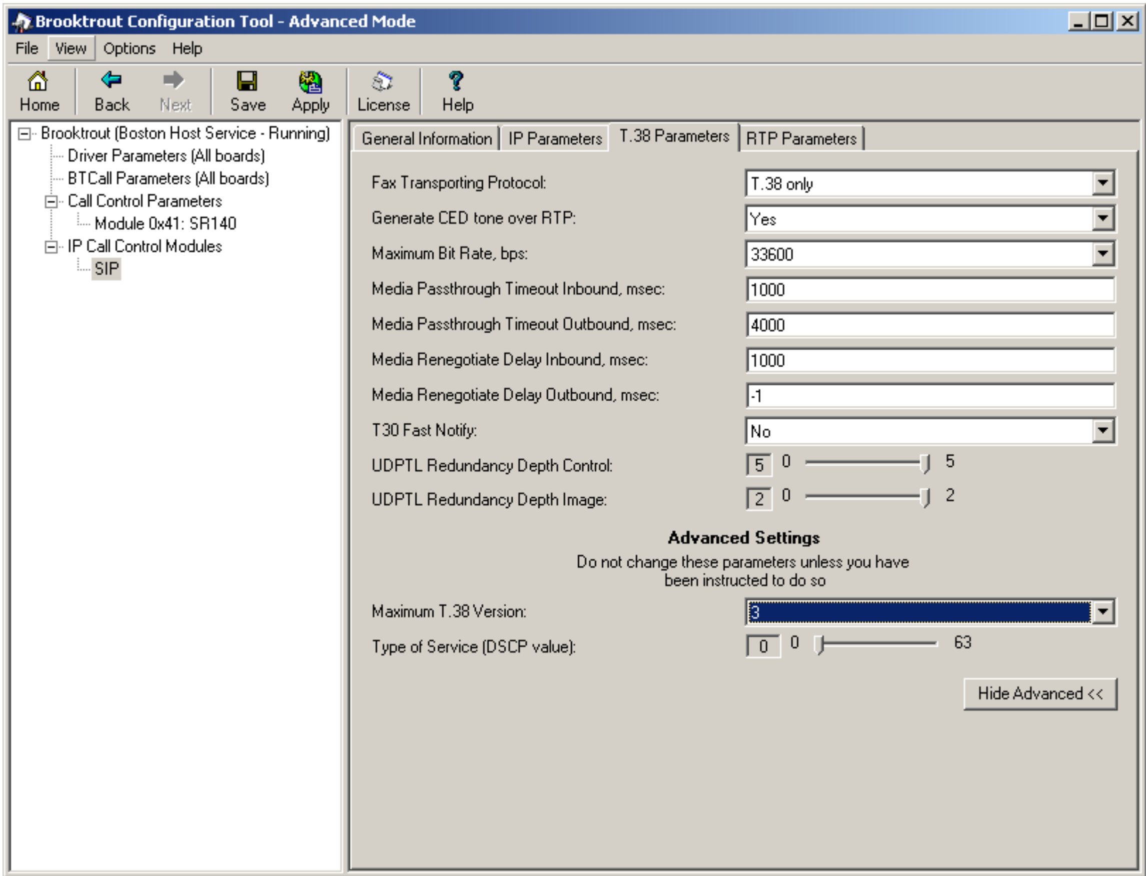


</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>

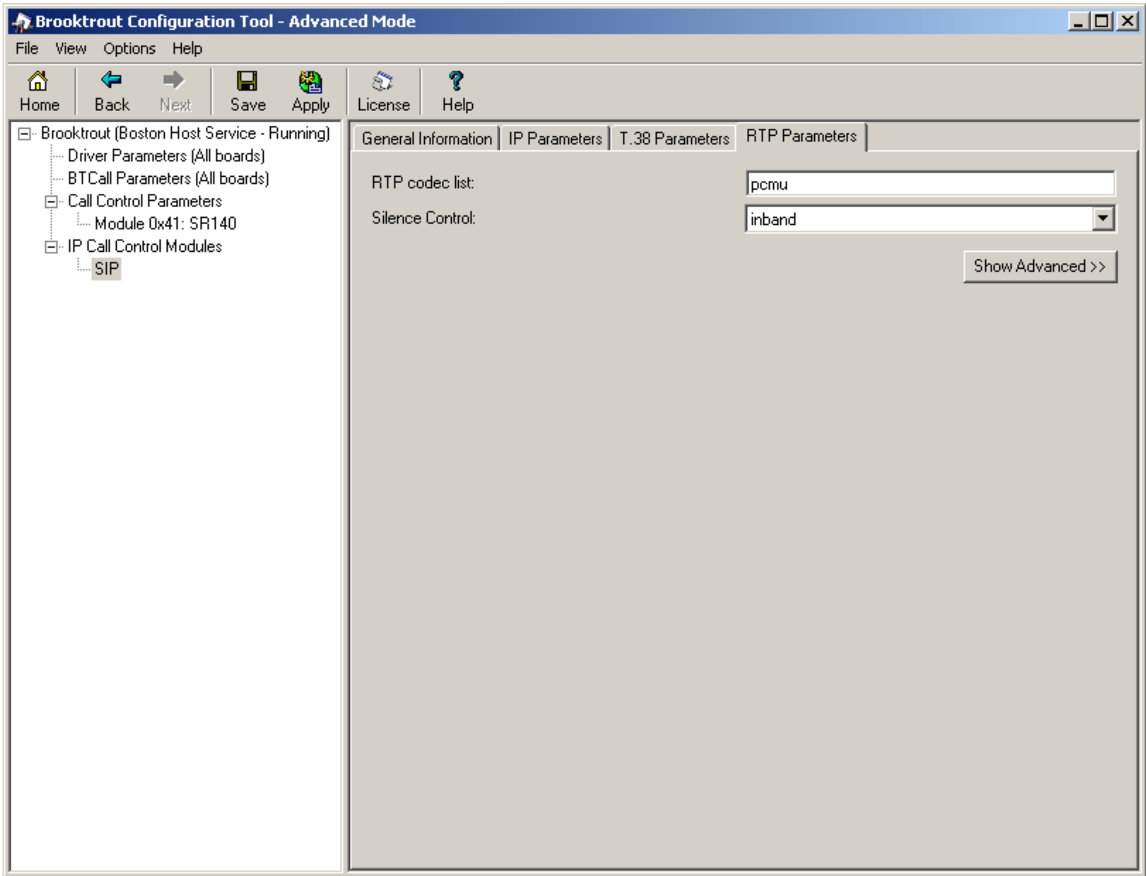
Configure T.38 Parameters

Select the **T.38 Parameters** tab. The screenshot below shows the values used during compliance testing.



</td></tr>
</table>

Step	Description
5.	**Configure RTP Parameters** Select the **RTP Parameters** tab. Set the **RTP codec list** value to use only a single codec, either ***pcmu*** or ***pcma*** to match the codec used in your region.

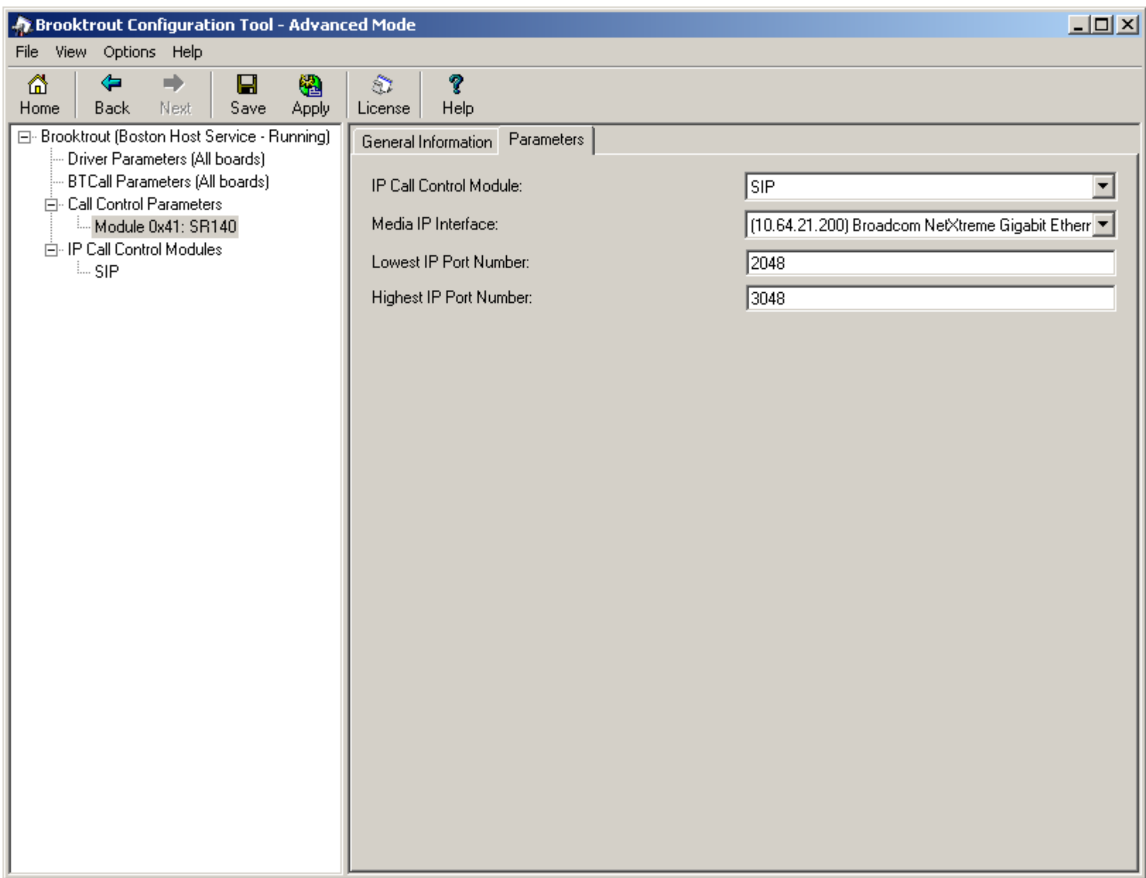


<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>

Configure RTP Port Range
From the pane on the left, navigate to **Call Control Parameters → Module 0x41: SR140**.

Select the **Parameters** tabs. Configure the **Lowest IP Port Number** and **Highest IP Port Number** values to match the **UDP Port Min** and **UDP Port Max** values in the **IP Network Region** configuration screen in Communication Manager.

Note*: The Communication Manager default port range is 2048 to 3329; however, the Brooktrout Configuration Tool range only spans 1000 ports. If you set Lowest IP Port Number to 2048, the Highest Port Number should automatically be set to 3048.*



</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>Complete Brooktrout SR140 Configuration
After verifying all the above parameters are properly set, click Save in the button menu and exit the Brooktrout Configuration Tool.

From Windows explorer, navigate to the path of the Brooktrout call control configuration file (i.e. callctrl.cfg). Open the callctrl.cfg file and verify the following (making any edits as necessary):

• Verify that the following configuration segment is present; and that the rtp_codec value under the [host_module.1/rtp] header matches the value specified in Step 5 above, either “pcmu” or “pcma”. (Note, <code>...</code> below indicates other entries under the header).

<pre>[host_module.1/rtp]
 ...
 rtp_codec=pcmu
 ...</pre>
• Verify that rtp_ced_enable is set to true under the [host_module.1/t.38parameters] header. (Note, <code>...</code> below indicates other entries under the header).

<pre>[host_module.1/t.38parameters]
 ...
 rtp_ced_enable=true
 ...</pre>
After making and saving any edits in the callctrl.cfg file, restart the fax server.</td></tr>
</table>

8 Verification Steps

The following steps may be used to verify the configuration:

- From Communication Manager SAT, use the:
  - **status signaling-group** to verify the signaling group to the fax server is in-service.
  - **status trunk-group** command to verify the trunk group to fax server is in-service.
  - **list trace tac** command to verify that fax calls are routed over the expected trunks.
- From System Manager, confirm that the Entity Link between Session Manager and the fax server is in service.
- Verify fax calls can be placed to/from the fax server.

9 Conclusions

These Application Notes describe the procedures for configuring Dialogic Brooktrout SR140 Fax Software with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using a SIP trunk interface. Dialogic Brooktrout SR140 Fax Software successfully passed compliance testing.

10 Additional References

This section provides references to the product documentation relevant to these Application Notes. Avaya product documentation may be found at http://support.avaya.com.

[1] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Doc # 555-245-205, Release 6.0, Issue 8.0, June, 2010.

[2] *Administering Avaya Aura™ Communication Manager*, Doc # 03-300509, Release 6.0, Issue 6.0, June, 2010.

[3] *Administering Avaya Aura® Session Manager*, Doc # 03-603324, Release 6.1, Issue 1.1, November, 2010.

[4] Dialogic Brooktrout SR140 Fax Software documentation may be found out http://www.dialogic.com/en/Products/fax-boards-and-software/foip/sr140.aspx.

[5] Product documentation for Avaya products may be found at: https://support.avaya.com/css/Products/

[6] *Avaya Solution & Interoperability Test Lab Report: Application Notes for Configuring Dialogic Brooktrout SR140 Fax Software with Avaya Aura® Communication Manager and Avaya Aura® Session using a SIP Trunk Interface - Issue 0.1*

11 Frequently Asked Questions

- ”*I’m configured as near as possible to this the sample configuration described in this document, but calls are still not successful; what is my next step?”*
  - ➔ Provide this document to your gateway support.
  - ➔ Ensure T.38 is enabled on the gateway.
  - ➔ Confirm that basic network access is possible by pinging the gateway.

- ”*How do I obtain Wireshark traces?*”
  - ➔ The traces can be viewed using the Wireshark network analyzer program, which can be freely downloaded from *http://www.wireshark.org*.
  - ➔ To view the call flow in Wireshark, open the desired network trace file and select “Statistics->VoIP Calls” from the drop down menu. Then highlight the call and click on the “Graph” button.